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ICONS 2014 Editors

Marko Jäntti, University of Eastern Finland, Finland
Gary Weckman,Ohio University, USA

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Foreword

The Ninth International Conference on Systems (ICONS 2014), held between January February 23rd-27th, 2014 in Nice, France, continued a series of events covering a broad spectrum of topics. The conference covered fundamentals on designing, implementing, testing, validating and maintaining various kinds of software and hardware systems. Several tracks were proposed to treat the topics from theory to practice, in terms of methodologies, design, implementation, testing, use cases, tools, and lessons learnt.

In the past years, new system concepts have been promoted and partially embedded in new deployments. Anticipative systems, autonomic and autonomous systems, self-adapting systems, or on-demand systems are systems exposing advanced features. These features demand special requirements specification mechanisms, advanced behavioral design patterns, special interaction protocols, and flexible implementation platforms. Additionally, they require new monitoring and management paradigms, as self-protection, self-diagnosing, self-maintenance become core design features.

The design of application-oriented systems is driven by application-specific requirements that have a very large spectrum. Despite the adoption of uniform frameworks and system design methodologies supported by appropriate models and system specification languages, the deployment of application-oriented systems raises critical problems. Specific requirements in terms of scalability, real-time, security, performance, accuracy, distribution, and user interaction drive the design decisions and implementations. This leads to the need for gathering application-specific knowledge and develop particular design and implementation skills that can be reused in developing similar systems.

Validation and verification of safety requirements for complex systems containing hardware, software and human subsystems must be considered from early design phases. There is a need for rigorous analysis on the role of people and process causing hazards within safety-related systems; however, these claims are often made without a rigorous analysis of the human factors involved. Accurate identification and implementation of safety requirements for all elements of a system, including people and procedures become crucial in complex and critical systems, especially in safety-related projects from the civil aviation, defense health, and transport sectors.

Fundamentals on safety-related systems concern both positive (desired properties) and negative (undesired properties) aspects. Safety requirements are expressed at the individual equipment level and at the operational-environment level. However, ambiguity in safety requirements may lead to reliable unsafe systems. Additionally, the distribution of safety requirements between people and machines makes difficult automated proofs of system safety. This is somehow obscured by the difficulty of applying formal techniques (usually used for equipment-related safety requirements) to derivation and satisfaction of human-related safety requirements (usually, human factors techniques are used).

We take here the opportunity to warmly thank all the members of the ICONS 2014 Technical Program Committee, as well as the numerous reviewers. The creation of such a high quality conference program would not have been possible without their involvement. We also kindly thank all the authors who dedicated much of their time and efforts to contribute to ICONS 2014. We truly believe that, thanks to all these efforts, the final conference program consisted of top quality contributions.

Also, this event could not have been a reality without the support of many individuals, organizations, and sponsors. We are grateful to the members of the ICONS 2014 organizing committee for their help in handling the logistics and for their work to make this professional meeting a success.

We hope that ICONS 2014 was a successful international forum for the exchange of ideas and results between academia and industry and for the promotion of progress in the field of systems.

We are convinced that the participants found the event useful and communications very open. We also hope the attendees enjoyed the charm of Nice, France.

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Andras Farago, The University of Texas at Dallas, USA

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A New Design Process to Reduce Resource Usage in SDR Systems

Sameh Yassin, Islam R Kamel, Hazim Tawfik
Department of Electronics and Communications
Cairo University Faculty of Engineering
Cairo, Egypt
syassin@eece.cu.edu.eg, islam.kamel@eng.cu.edu.eg, Hazim.Tawfik@eece.cu.edu.eg

***Abstract*—Software Defined Radio (SDR) is a recent trend in communication systems. Its primary goal is to flexibly handle different technologies using the same hardware platform by applying different software. To achieve this goal efficiently, the resource utilization should be minimized by implementing algorithms in fixed point arithmetic. This work is focused on the minimization of resource utilization by reducing the data width of the mathematical quantities inside the receiver chain. A design process is proposed to optimize the computational accuracy. The design process is used to link system performance and computational accuracy using simulation. The simulation approach is chosen because of its ability to cope with functional changes in SDR systems, and to satisfy the short time to market requirement. To verify the results, a case study is considered by reducing the resource usage in the SDR platform known as Universal Serial Radio Peripheral (USRP). The functions with highest resource utilization are implemented in fixed point. In addition, the optimized functions within the case study are implemented using Hardware Description Language (HDL) on a Field Programmable Gate Array (FPGA), to verify experimentally the reduction in resource utilization due to the proposed design process.**

Keywords*—*computational accuracy; finite precision; software defined radio; signal processing; USRP.

I. Introduction

Software Defined Radio is a flexible platform that can provide dynamic reconfiguration using software. In other words, the same hardware can be used to implement different transceiver functions, such as modulation, detection, and channel estimation. One motivation for the rapid development of SDR systems, is the need to to add new features to the radio equipment or to upgrade its functionality. This is needed by important sectors such as military and public safety [1]. In these sectors, special purpose radios were the norm rather than the exception. One radio device is needed for few number of functions or wave forms, and it is not straightforward to alter the device functionality.

A SDR system is a generic communication system starting with user data layer and ending with physical data layer [2]. The generality of a SDR system is due to its ability to reconfigure the system modules without changing the hardware. This is obtained by adding control to a communication system, as shown in Figure 1. A SDR system consists of radio front end, base band processing, control bus, and application. These modules are combined to map the required data from the application into physical signal at arbitrary carrier frequency. The application is modified during design time and/or execution time by sending reconfiguration messages through the control bus. Hence, to modify the functionality of the system, only the application is updated using software configuration.

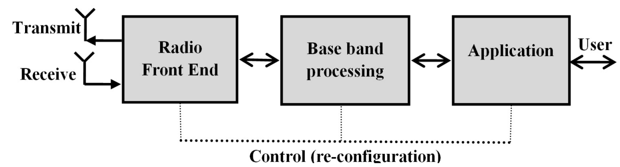

Fig. 1. Software Defined Radio Aspects

In SDR, it is frequently desired to add features to an existing system. However, there are three considerations while adding new features, namely execution time, power consumption, and resource utilization. The main focus of this work is to evaluate the increase in resource usage due to these added features. In addition, the increase in the resource usage should be optimized in order not to affect the system performance. In the next paragraphs, we are going to describe the related work that address computational accuracy and their effects on system performance.

In [3], the main motivation is to reduce the energy consumption, and the complexity requirements to satisfy an embedded system needs. Nguyen et al. use fixed point arithmetic to present an optimized digital receiver. The merit for this work is finding an analytical relation between the receiver performance, and the number of bits needed to satisfy energy consumption constraints. Using this approach, the minimum data width can be estimated with negligible performance degradation compared to floating point arithmetic. However, to deduce the minimum data width one can not reach closed form expression for some equations. For SDR systems, it is impractical to update the performance equations each time the system functions are modified.

In [4], Novo et al. highlighted the need to implement complex signal processing algorithms in fixed point. A metric was developed to compensate for the loss in accuracy due to the conversion from floating point to fixed point. That metric is system dependent, and should be dimensioned carefully for each application.

In [5], the concept of scalable SDR is introduced for battery powered devices. To achieve the two contradicting requirements of fast time to market and minimum energy consumption, a new method is presented to deal with fixed point arithmetic. The new method considers the changes in data format, such as modulation scheme, and number of antennas, and accordingly data width can be adjusted to save energy consumption of the battery. However, the savings in power consumption come at the expense of increased resource utilization.

The rest of this paper is organized as follows. In Section II, the problem formulation due to finite precision is highlighted. Then, a design process is proposed to discover the minimum computation cost required that maintains the system performance in terms of Bit Error Rate (BER). In Section III, a case study is presented to validate the proposed design process. In Section IV, computer simulations are developed to evaluate the system performance. In addition, the case study is implemented in FPGA to verify the simulation result experimentally. The paper is concluded in Section V.

II. Problem description due to finite number representation

A. Number Representation

The use of fixed point numbers to implement algorithms has some limitations. Due to the finite resources of machine number system, there exist many unavoidable issues such as overflow, underflow, scaling. There also exist some errors such as round-off error, and quantization error.

Due to these issues, the implementation of algorithms in fixed point representation can affect system performance considerably. There are two approaches to evaluate the degradation in system performance, namely analytical, and simulation. In this work, the simulation based approach is chosen because it is more adequate to the nature of programmable SDR systems in terms of supporting new features in short design cycle, and coping with fast market changes.

We propose a new process to design any new SDR system, or add features to an existing one. The added value of this process is the link between system performance and computational accuracy. A similar method is proposed Mehard et al. [6] using an analytical approach. The analytical approach was chosen there because it needed less execution time. However, obtaining closed form expressions for each application is not straightforward, and may have to be solved numerically. Therefore, the increase in problem complexity may outweigh the decrease in execution time. Moreover, SDR need to be flexible to changes in system functions. It is impractical to update the performance equations each time the system functions are modified.

B. Proposed Design Process

The aim of the design process is to calculate the optimum data width that can be used without affecting the system performance. A metric will be used to compare between system performance using floating point, and fixed point representation. The data width is optimized by allocating proper number of bits to the integer part i_{wl} and the fraction part f_{wl}. The proposed design process is outlined in the flowchart shown in Figure 2.

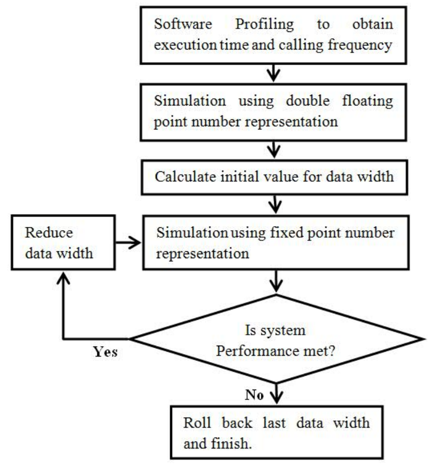

Fig. 2. Flow chart for the proposed data width minimization process

The first step is to determine the utilization of processing power by each system function. This is achieved using software profiling. For each system function, calculate its execution time and the number of times it is called. Both are needed to avoid optimizing one function with high execution time when it is only called few times. The second step is to simulate the system performance using double floating point numbers, which is efficient when a large dynamic range is required [4]. The dynamic range is defined as the ratio between largest and lowest signal amplitudes. The performance metric is chosen to be either BER [6], or Signal to Noise Ratio (SNR) [3]. In this work, BER is used for comparison because it can be accurately measured. The third step, is to determine an initial data width for fixed point simulation as will be discussed in Subsection II-C. This initial value will be optimized in the following steps. The fourth step, is to simulate the system and obtain performance curves using fixed point data width. The fifth step, is to compare between the BER curves obtained in steps two and four. By comparing both BER curves, the required increase in SNR to obtain the same BER value can be calculated. The required increase in SNR ϵ can be calculated using (1).

$$\epsilon = SNR_{fp} - SNR_0, \tag{1}$$

where SNR_0, SNR_{fp} are the SNR for floating point, and fixed point, respectively. The value of ϵ is a design parameter, that is chosen arbitrarily. When the system is desired to have the same BER value for both fixed point and floating point, ϵ is set to 0 dB. For further savings in resource usage, ϵ is increased. The fifth step proceeds by increasing SNR_{fp}. If the difference $SNR_{fp} - SNR_0$ is still smaller than ϵ, then reduce the data width (i_{wl}, f_{wl}). Otherwise, roll back previous value of data width and proceed to the end of the process. Note that the data width is reduced gradually first by reducing the integer part i_{wl} by one bit at a time, until its minimum value is obtained. Afterwards, the fraction part f_{wl} is reduced by one bit at a time until its minimum value is obtained.

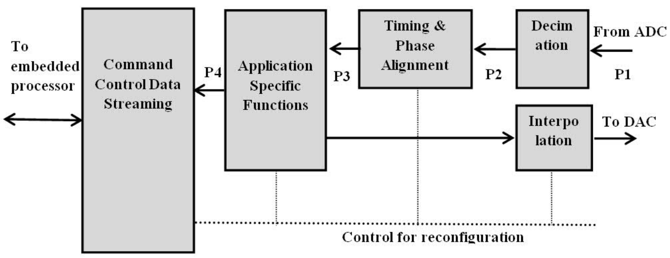

Fig. 3. Signal flow graph for receiver chain in generic SDR system

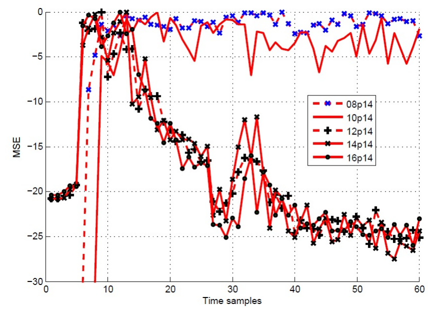

Fig. 4. MSE of ensemble of 50 trials. The integer part is variable

C. Dynamic range estimation

To estimate the initial value of data width a signal flow graph for the SDR system under development should be constructed, as shown in Figure 3. In related work [3][5], the data width and dynamic range are usually estimated for the system at one point. In this work, it is proposed to estimate the data width at each block's output. Afterwards, the dynamic range can be estimated for both receiver and transmitter chains. The initial value for data width will be calculated for receiver chain because it is more complex than the transmitter in terms of the required number of functions. Without loss of generality, the same method can be applied to transmitter chain to calculate its initial data width. For the receiver chain, the points of interest is highlighted and marked as P_i where $1 \leq i \leq N$, where N is the number of blocks with different data width. Next, the initial value of data width can be estimated by observing each point in the receiver chain. At point P_1, the data width is the precision of the front end Analog to Digital Converter (ADC), such that

$$wl_{P_1} = Accuracy_{ADC}. \tag{2}$$

At point P_2, the data is prepared for base band processing by reducing its sampling rate using the decimation process. The input stream sampling frequency is reduced by an integer factor called decimation factor R, even though only the sample rate is changed, and not the bandwidth of the signal. The input signal bandwidth must be filtered to avoid aliasing. Therefore, the decimation process requires an increase in the data width to maintain proper number of bits per sample. The increase in data width can be calculated, depending on the decimation method of choice. One example is the Cascaded Integrated Comb (CIC) filter. The input signal is fed through one or more cascaded integrator sections, then a down sampler, followed by one or more comb sections [7]. The increase in the data width will be dependent on the differential delay M of the comb section, and the number of blocks N as in (3), where $Ceil(A)$ rounds to the nearest integer greater than or equal to A.

$$wl_{P2} = wl_{P1} + Ceil(N * log_2(M * R)). \tag{3}$$

At point P_3, the timing and phase changes relative to the original transmitted signal are estimated. This includes correlation operation which is composed of addition, shift, and multiplication operations. The multiplication operation particularly leads to the most significant increase in the input data width which is proportional to the multiplier $n1$, and the multiplicand $n2$ as shown in (4).

$$wl_{P3} = wl_{P2} + \sum(n1 + n2 - 1). \tag{4}$$

At point P_4, application specific functions can also increase the input data width, and may alter the numerical stability due to the use of fixed point such as channel equalization. The data width for this block must maintain numerical stability of the SDR system. In case of channel equalization, the data width should result in quantization noise power that will not alter the computation of the Mean Square Error (MSE) of the equalization algorithm [8]. To illustrate how to maintain numerical stability of equalization algorithm, one channel equalization algorithm, namely the Recursive Least Squares (RLS) is considered.

To maintain numerical stability of RLS, data width should be increased as shown in Figure 4. A family of MSE curves is obtained for different data widths ranging from ($i_{wl} = 8$, $f_{wl} = 14$) bits, to ($i_{wl} = 16$, $f_{wl} = 14$) bits. This family of curves can be used to conclude that the minimum value is $i_{wl} = 12$, because it keeps MSE decreasing as the time samples advance. To obtain the minimum value for fractional part $f_{wl} = 14$, a similar family of curves is developed but with fixed integer part, and variable fractional part. Note that stabilizing the algorithm will increase the data width. Therefore, fast fixed order filters can be used to reduce the data width [8]. To choose an equalizer algorithm with lowest resource utilization, a fair comparison between different equalizer algorithms was proposed in [9]. Yassin and Tawfik proposed to weight the resource usage of equalizers by mapping the algorithm in terms of mathematical operations. This fair choice will compensate for the increase in data width. The data width for this block can be calculated as in (5), where δ is the required increase in the data width to maintain the numerical stability.

$$wl_{P_4} = wl_{P_3} + \delta. \tag{5}$$

The value wl_{P_4} can be used initially for the design process. To validate the design process, a case study will be considered in Section III.

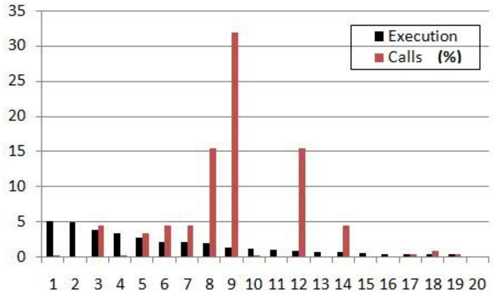

Fig. 5. The percentage of processor execution time, number of calls against the function identities

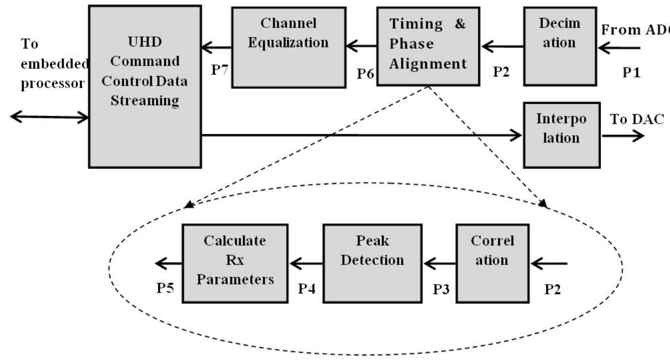

Fig. 6. Signal flow graph for the block under test “Analyze Traffic Burst”

III. Case study: Open BTS receiver chain

The case study is part of the Open Base Transceiver Station (OpenBTS) project [10], which is based on ETTUS research platform named Universal Serial Radio Peripheral (USRP). This platform provides a cheap alternative to standard BTS [10], [11]. Hence, rural communities can enjoy cheap and basic telecommunication services using Global System for Mobile communications (GSM).

The first step, is to perform software profiling for the OpenBTS system as shown in Figure 5. The function identities of the OpenBTS system appear on the x axis, while the y axis shows both the percentage of processor execution time, and the normalized number of calls of each function. As previously mentioned, only those functions with high execution time and high calling frequency should be optimized. It can be observed from Figure 5 that the functions with identities (8, 9, and 12) have the highest utilization. These functions are grouped into one large function named “Analyze Traffic Burst”. The purpose of the function is to calculate important parameters of the receiver, namely Time of Arrival (ToA), Valley Power (VP), and channel estimation coefficients. ToA, and VP are used for synchronization between mobile station and base station. The second step, is to simulate system performance using double floating point representation. The results of this step will be detailed in Section IV.

The third step, is to calculate an initial data width for simulation. A block diagram is constructed for the “Analyze Traffic Burst” function as shown in Figure 6. At $P1$, the accuracy of the ADC component in USRP is 12 bits. Then the input data width equals the ADC accuracy $wl_{P1} = 12$. At $P2$, decimation is implemented using CIC filter with $N = 4$ stages and a variable decimation rate $log_2(M \times R) = 7$ resulting in $wl_{P2} = (12 + 28)$. However, the original OpenBTS design chooses to truncate this value to be $wl_{P2} = (12 + 12)$ to save resource usage. This truncation will not affect the design process, the truncation can be ignored and same results will be obtained. At $P3$, the data width will be doubled due to the multiplication in the correlation function resulting in a data width of $wl_{P3} = (24 + 24 - 1)$. At $P4$, the peak detection is divided into two sub blocks, namely Coordinate Rotation Digital Computer (CORDIC) and interpolation. The basic operation is to rotate the x-y axes of the input data to eliminate the y component. Hence, the magnitude of the input vector is stored in the x component only. This will in turn reduce the required resources. On the contrary, the interpolation operation contains multiplications. Therefore, the data width will be proportional to the width of the multiplier and multiplicand. The multiplier is the output of the previous stage, while the multiplicand is a SINC function that is read from memory element. A SINC function has maximum integer value of 1. Hence, it can be represented with minimum accuracy, and it is chosen to be the same as $wl_{P2} = 24$. Now, the data width of multiplier equals $wl_{P3} = 47$, and the data width of multiplicand is 24. Therefore, the initial value for data width in the simulation will be $wl_{P_4} = (47 + 24) = 71$ bits.

The initial value will be optimized by repeating steps four and five until the performance condition is violated. This will be discussed in the following section.

IV. Simulation Results

The results obtained in Section III are validated by developing a simulation model using Mathworks tool Simulink Matlab $v7.12$ [12]. The model consists of transmitter, channel, and receiver. The transmitter consists of random data source generator, GSM burst formatter, and digital up converter. The channel is a Rayleigh fading channel, with variable fractional delay, and Additive White Gaussian Noise (AWGN). This channel model [14] can be used to simulate the system for the cases of typical urban, and rural areas. The receiver consists of digital down conversion, timing and phase alignment, channel equalization, burst deformatter, and bit error rate calculator.

The simulation is used to calculate BER curve for different values of SNR as shown in Figure 7. The graph with solid line and ‘o’ marker is calculated when all variables are represented in floating point. The design process starts with $\epsilon = 0.2dB$ to obtain system performance similar to the case of floating point arithmetic. This results in a minimum data width ($i_{wl} = 62$, $f_{wl} = 4$) bits, which is shown in the curve with the solid line and ‘x’ marker. It can be observed that for the same BER value, the SNR difference between the two curves is less than $1\ dB$. This was expected because ϵ is small. If the system performance is relaxed to reduce resource utilization, the same process can be run again with $\epsilon = 2\ dB$, resulting in data

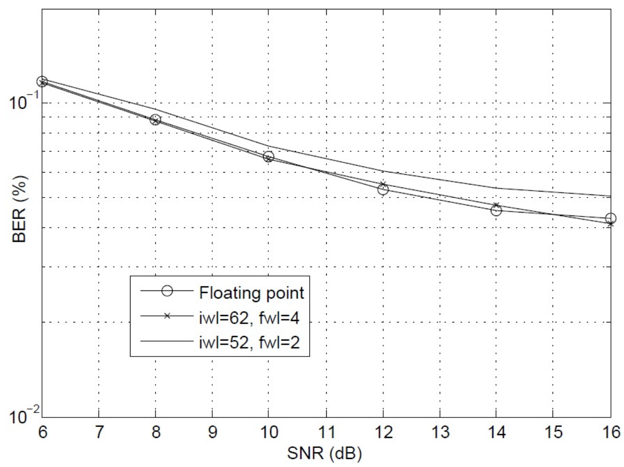

Fig. 7. BER curves for both floating point and fixed point

TABLE I
FPGA RESOURCE UTILIZATION FOR THE CASE STUDY

Unit	$i_{wl} = 52$			$i_{wl} = 62$		
	Used	Available	%	Used	Available	%
Slices	7538	23872	31	9548	23872	40
Flip Flops	9273	47744	19	15278	47744	32
4-LUTs	14074	47744	29	18620	47744	39
BRAMs	7	126	5	7	126	5
GCLKs	2	24	8	2	24	8
DSP48s	57	126	45	57	126	45

width of ($i_{wl} = 52$, $f_{wl} = 2$) bits. It should be mentioned that the advantage of using fixed point is two fold. First, because the data width can be minimized at different points of the receiver chain. Second, all fixed point mathematical operators will consume less resources than floating point operators [13].

To validate the results experimentally, a simulation is run on the design after mapping to physical FPGA cells. This is called timing simulation. Xilinx's design tools $v14.1$ are used, namely ISE, and ISIM. The USRP is equipped with Xilinx's FPGA named Spartan 3A-DSP 3400. The resource utilization due to the mapped design is reported in Table I. When the design is implemented using floating point arithmetic, the synthesis operation fails to map the design into the FPGA. This is expected because a floating point multiplier can consume one, or few FPGA units [9][15]. After applying the design process, the design can fit the into the FPGA with resource usage that is found to be less that 50%. The DSP48 blocks have relatively high utilization 45%, because they are required to implement multiplication operation without using the logic slices of FPGA. Note that, the multipliers have 36 bits operands, and hence they can be used for both cases of ($i_{wl} = 62$, $f_{wl} = 4$) and ($i_{wl} = 52$, $f_{wl} = 2$), with the same resource utilization. Finally, the experimental results verify the validity of the proposed design process.

V. CONCLUSION

SDR has a desirable nature of adding new features by reconfiguration. However, this will increase the resource usage and may affect system performance. One solution is to implement algorithms in fixed point number representation. In this work, a new design process is proposed to link between system performance and computational accuracy using simulation. To validate the proposed process, a case study of the OpenBTS project is considered. It was not possible to implement the case study into FPGA without applying the proposed process, while maintaining system performance. Moreover, the system performance was relaxed to obtain more savings in resource usage. Finally, the results were verified experimentally using FPGA implementation. It was shown that the utilization of FPGA did not exceed 50 % of the available resources .

ACKNOWLEDGMENT

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Efficient FPGA-Based Architecture for Spike Sorting Using Generalized Hebbian Algorithm

Wen-Jyi Hwang, Chi-En Ke, Sheng-Ying Lai, Jung-Gen Wu
Department of Computer Science and
Information Engineering
National Taiwan Normal University
Taipei, Taiwan
Email: whwang@csie.ntnu.edu.tw, gdset@hotmail.com, ddt0218@hotmail.com, jgwu@csie.ntnu.edu.tw

***Abstract*—An efficient VLSI architecture for fast spike sorting is presented in this paper. The architecture is able to perform feature extraction based on the Generalized Hebbian Algorithm (GHA). The employment of GHA allows efficient computation of principal components for subsequent clustering and classification operations. The hardware implementation of GHA features high throughput and high classification success rate. The proposed architecture is implemented by Field Programmable Gate Array (FPGA). It is embedded in a System-On-Programmable-Chip (SOPC) platform for performance measurement. Experimental results show that the proposed architecture is an efficient spike sorting design with high speed computation for spike trains corrupted by large noises.**

Keywords-Spike Sorting; FPGA; Generalized Hebbian Algorithm.

I. INTRODUCTION

Spike sorting [1] is often desired for the design of brain machine interface (BMI) [2]. It receives spike trains from extracellular recording systems. Each spike train obtained from the system is a mixture of the trains from neurons near the recording electrodes. The goal of spike sorting is to segregate the spike trains of individual neurons from this mixture. Spike sorting is a difficult task due to the presence background noise and the interferences among neurons in a local area. A typical spike sorting algorithm involves computationally demanding operations such as feature extraction. One way to carry out these complex tasks is to deliver spike trains to external computers. Because the delivery of raw spike trains requires high bandwidth, wireless transmission may be difficult. Existing spike sorting systems may therefore be wired, restraining patients and test subjects from free movement.

Hardware spike sorting is an effective alternative for BMI applications. It allows the spike sorting to be carried out at the front-end so that data bandwidth can be reduced for wireless communication. A common approach for hardware design [3] is based on Application Specific Integrated Circuits (ASICs). A major drawback of ASICs is the lack of flexibility for changes. With the wide range of spike sorting algorithms that already exist and the continual design and improvement of algorithms, the ability to easily change a spike sorting system for new algorithms is usually desired. However, the modification in ASIC is very difficult, especially when chips are implanted in the brain. In addition, the high Non-Recurring Engineering (NRE) costs and long design and verification efforts for fabricating ASICs can severely limit the applicability of emerging BMI applications. The field programmable gate array (FPGA) is an effective alternative to ASIC for hardware implementation with lower NRE costs. Moreover, the circuits in an FPGA are reconfigurable; thereby providing higher flexibility to a spike sorting architecture for future extensions.

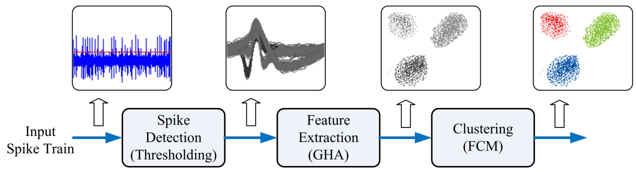

Figure 1: The operations of a typical spike sorting system.

The objective of this paper is to present an effective FPGA-based hardware architecture for spike sorting. The architecture is able to perform online training for feature extraction in hardware. The feature extraction is based on the generalized Hebbian algorithm (GHA) [4]. The proposed architecture is used as a hardware accelerator of a spike sorting system on a System-On-Programmable-Chip (SOPC) platform for performance evaluation. The computation time of spike sorting based on the SOPC is measured and compared with existing works. Experimental results reveal that the proposed architecture is able to perform feature extraction in real time with low hardware resource consumption.

The remaining parts of this paper are organized as follows: Section 2 gives a brief review of the spike sorting operations and the GHA algorithm. Section 3 describes the proposed GHA architecture. Experimental results are included in Section 4. Finally, the concluding remarks are given in Section 5.

II. PRELIMINARIES

A. Spike Sorting Operations

Figure 1 shows the operations of a typical spike sorting system, which consists of spike detection, feature extraction and clustering. The spike detection identifies and aligns spikes from a noisy spike train. A simple spike detection technique

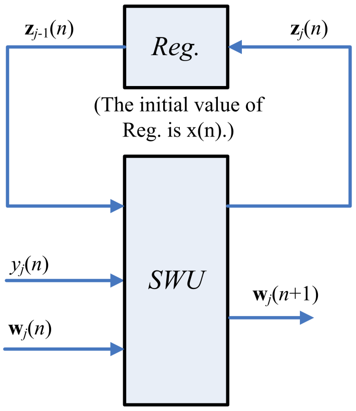

Figure 2: The hardware implementation of (10) and (12).

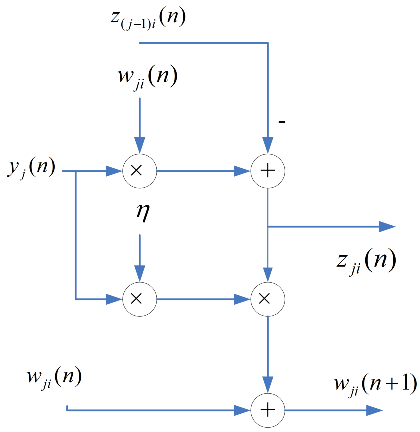

Figure 3: The architecture of each module in the SWU unit.

is to perform thresholding based on the absolute value of spike samples. The feature extraction finds the feature vectors from the detected spikes. The GHA algorithm has been found to be an effective technique for feature extraction. Based on the feature vectors, the final step of the spike detection is to perform clustering and classification using unsupervised clustering methods such as the fuzzy c-means (FCM) algorithm [5]. Detailed discussions of each step can be found in [1].

B. GHA Algorithm

Let

$$\mathbf{x}(n) = [x_1(n), \ldots, x_m(n)]^T, n = 1, ..., t, \quad (1)$$
$$\mathbf{y}(n) = [y_1(n), \ldots, y_p(n)]^T, n = 1, ..., t, \quad (2)$$

be the n-th input and output vectors to the GHA, respectively. In addition, m, p and t are the vector dimension, the number of Principal Components (PCs), and the number of input and output vectors for the GHA, respectively. The output vector $\mathbf{y}(n)$ is related to the input vector $\mathbf{x}(n)$ by

$$y_j(n) = \sum_{i=1}^{m} w_{ji}(n) x_i(n) \quad (3)$$

where the $w_{ji}(n)$ stands for the weight from the i-th synapse to the j-th neuron at iteration n.

Let

$$\mathbf{w}_j(n) = [w_{j1}(n), \ldots, w_{jm}(n)]^T, j = 1, \ldots, p \quad (4)$$

be the j-th synaptic weight vector. Each synaptic weight vector $\mathbf{w}_j(n)$ is adapted by the Hebbian learning rule:

$$w_{ji}(n+1) = w_{ji}(n) + \eta[y_j(n)x_i(n) - y_j(n)\sum_{k=1}^{j} w_{ki}(n)y_k(n)] \quad (5)$$

where η denotes the learning rate. Given an input vector $\mathbf{x}(n)$, the GHA algorithm involves the computation of $y_j(n)$ in (3), and $\mathbf{w}_j(n)$ in (5) for $j = 1, ..., p$. After a large number of iterative computation and adaptation, $\mathbf{w}_j(n)$ will asymptotically approach to the eigenvector associated with the j-th eigenvalue λ_j of the covariance matrix of input vectors, where $\lambda_1 > \lambda_2 > \cdots > \lambda_p$. A more detailed discussion of GHA can be found in [4].

C. GHA Algorithm for Spike Sorting

The GHA can be used for feature extraction of spikes. To use GHA for feature extraction, the $\mathbf{x}(n)$ in (2) is the n-th spike in the spike train. Therefore, the vector dimension m is the number of samples in a spike. Let

$$\mathbf{w}_j = [w_{j1}, \ldots, w_{jm}]^T, j = 1, \ldots, p \quad (6)$$

be the synaptic weight vectors of the GHA after the training process has completed. Based on $\mathbf{w}_j, j = 1, ..., p$, the GHA feature vector extracted from training vector $\mathbf{x}(n)$ (denoted by $\mathbf{f}_n$) is computed by

$$\mathbf{f}_n = [f_{n,1}, ..., f_{n,p}]^T, \quad (7)$$

where

$$f_{n,j} = \sum_{i=1}^{m} w_{ji} x_i(n) \quad (8)$$

be the j-th element of $\mathbf{f}_n$. The set of feature vectors $F = \{\mathbf{f}_1, ..., \mathbf{f}_t\}$ are then for subsequent classification and clustering.

III. Proposed Architecture

The proposed GHA architecture consists of three functional units: the memory unit, the synaptic weight updating (SWU) unit, and the principal components computing (PCC) unit.

A. SWU Unit of GHA

To reduce the complexity of computing implementation, (5) can be rewritten as

$$w_{ji}(n+1) = w_{ji}(n) + \eta y_j(n)[x_i(n) - \sum_{k=1}^{j} w_{ki}(n)y_k(n)] \quad (9)$$

The design of SWU unit is based on (9). Although the direct implementation of (9) is possible, it will consume large hardware resources. One way to reduce the resource consumption is by observing that (9) can be rewritten as

$$w_{ji}(n+1) = w_{ji}(n) + \eta y_j(n) z_{ji}(n), \quad (10)$$

where

$$z_{ji}(n) = x_i(n) - \sum_{k=1}^{j} w_{ki}(n)y_k(n), j = 1, \ldots, p. \quad (11)$$

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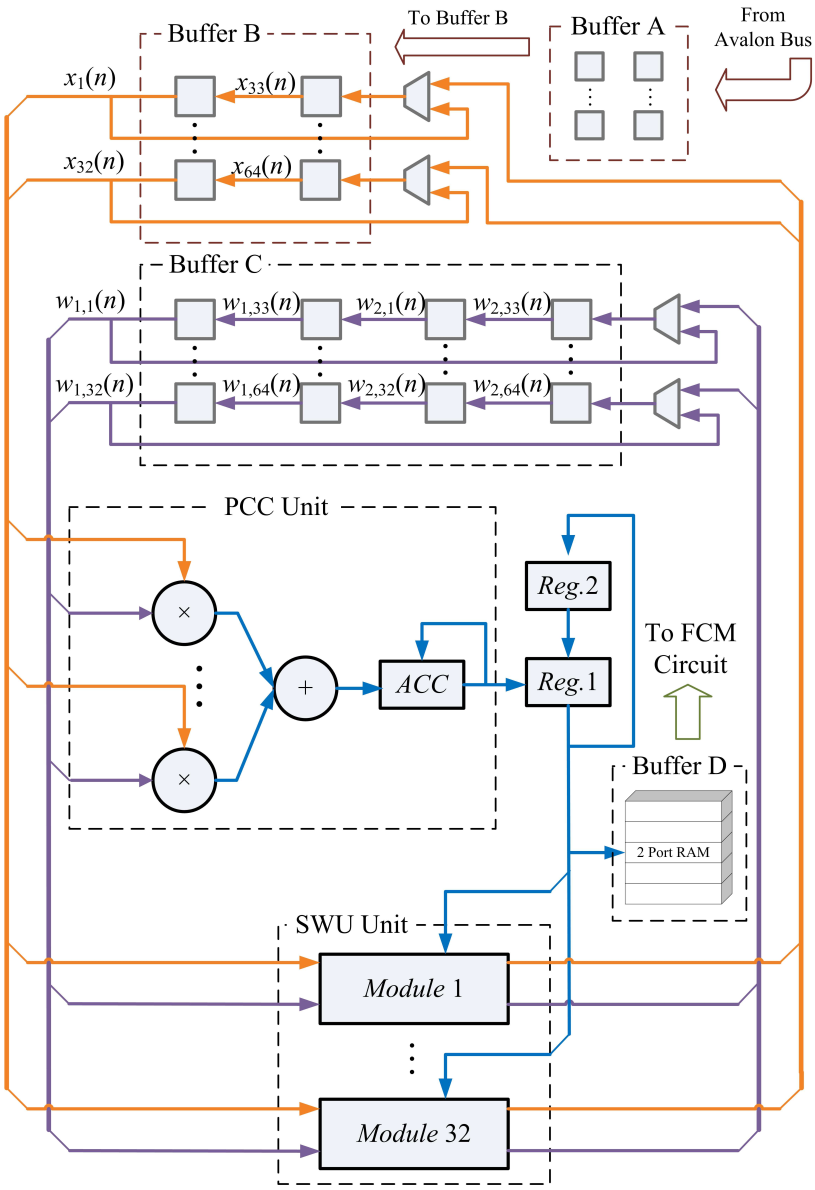

Figure 4: The GHA circuit for $m = 64$, $p = 2$, $b = 2$ and $q = 32$

and $\mathbf{z}_j(n) = [z_{j1}(n), \ldots, z_{jm}(n)]^T$. The $z_{ji}(n)$ can be obtained from $z_{(j-1)i}(n)$ by

$$z_{ji}(n) = z_{(j-1)i}(n) - w_{ji}(n)y_j(n), j = 2, \ldots, p \tag{12}$$

When $j = 1$, from (11) and (12), it follows that

$$z_{0i}(n) = x_i(n) \tag{13}$$

Therefore, the hardware implementation of (10) and (12) is equivalent to that of (9). Figure 2 depicts the hardware implementation of (10) and (12). As shown in the figure, the SWU unit produces one synaptic weight vector at a time. The computation of $\mathbf{w}_j(n+1)$, the j-th weight vector at the iteration $n+1$, requires the $\mathbf{z}_{j-1}(n)$, $\mathbf{y}(n)$ and $\mathbf{w}_j(n)$ as inputs. In addition to $\mathbf{w}_j(n+1)$, the SWU unit also produces $\mathbf{z}_j(n)$, which will then be used for the computation of $\mathbf{w}_{j+1}(n+1)$. Hardware resource consumption can then be effectively reduced.

One way to implement the SWU unit is to produce $\mathbf{w}_j(n+1)$ and $\mathbf{z}_j(n)$ in one shot. However, m identical modules, individually shown in Figure 3, may be required because the dimension of vectors is m. The area costs of the SWU unit then grow linearly with m. To further reduce the area costs, each of the output vectors $\mathbf{w}_j(n+1)$ and $\mathbf{z}_j(n)$ is separated into b blocks, where each block contains q elements. The SWU unit only computes one block of $\mathbf{w}_j(n+1)$ and $\mathbf{z}_j(n)$ at a time. Therefore, it will take b clock cycles to produce complete $\mathbf{w}_j(n+1)$ and $\mathbf{z}_j(n)$.

B. PCC unit of GHA

The PCC operations are based on (3). Therefore, the PCC unit of the proposed architecture contains adders and multipliers. Because the number of multipliers grows with the vector dimension m, the direct implementation using (3) may consume large hardware resources when m becomes large.

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Similar to the SWU unit, the block based computation is used for reducing the area costs. In fact, (3) can be rewritten as

$$y_j(n) = \sum_{k=1}^{b}\sum_{i=1}^{q} w_{j,(k-1)q+i}(n)x_{(k-1)q+i}(n). \qquad (14)$$

The implementation of (14) needs only q multipliers, a q-input adder, and an accumulator.

C. Memory Unit of GHA

The memory unit contains four buffers: Buffers A, B, C and D. Buffer A fetches and stores spike $\mathbf{x}(n)$ from the main memory. Buffer B contains $\mathbf{z}_j(n)$ for the computation in PCC and SWU units. Buffer C consists of the synaptic weight vectors $\mathbf{w}_j(n)$. The feature vectors $\mathbf{f}_1, ..., \mathbf{f}_t$ are stored in Buffer D. The Buffers A, B and C are shift registers. Buffer D is a two-port RAM for the subsequent access by the FCM unit.

D. Operations of the GHA unit

In typical spike sorting implementations [8], a spike may contain 64 samples. In addition, two PCs may suffice for feature extraction [1]. Therefore, without loss of generality, the GHA unit for $m = 64$ (i.e., vector dimension is 64) and $p = 2$ (i.e., number of PCs is 2) is considered in this subsection. In the GHA unit, each vector is separated into 2 blocks. Moreover, the dimension of each block is 32. Therefore, we set $b = 2$ and $q = 32$ for the circuit implementation. Figure 4 shows the resulting GHA circuit for $m = 64$, $p = 2$, $b = 2$ and $q = 32$. The operations of the GHA circuit can be separated into 4 states, as revealed in Figure 5. The most important operations of the GHA circuit are the PCC operations in State 3 and SWU operations in State 4. These two operations are further elaborated below.

Assume the input vector $\mathbf{x}(n)$ is available in Buffer B. In addition, the *current* synaptic weight vectors $\mathbf{w}_1(n), \mathbf{w}_2(n)$ are stored in the Buffer C. Based on $\mathbf{x}(n)$ and $\mathbf{w}_1(n), \mathbf{w}_2(n)$ the PCC unit produces output vector $y_1(n), y_2(n)$. The computation of $y_j(n)$ is separated into two steps. The first step finds $\sum_{i=1}^{32} w_{j,i}(n)x_i(n)$. The second step computes $\sum_{i=33}^{64} w_{j,i}(n)x_i(n)$, and then accumulate the result with that of the previous step to find $y_j(n)$. These two steps share the same circuit in the PCC unit.

Upon the completion of PCC operations, the SWU unit will be activated in State 4. Using $\mathbf{x}(n)$, $y_j(n)$ and $\mathbf{w}_j(n), j = 1, 2,$ the SWU unit computes the new synaptic weight vectors $\mathbf{w}_j(n+1), j = 1, 2,$ which are then stored back to Buffer C for subsequent training. Similar to the computation of $y_j(n)$, the computation of $\mathbf{w}_j(n+1)$ consists of two steps. The first step computes the first half of $\mathbf{w}_j(n+1)$ (i.e., $w_{j,1}(n+1)$,...,$w_{j,32}(n+1)$). The second step calculates the second half. These two steps also share the same circuit in the SWU unit. Moreover, the computation of $\mathbf{w}_1(n+1)$ also produces $\mathbf{z}_1(n)$, which is stored back to Buffer B.

After the training process of GHA circuit is completed, the Buffer C contains the synaptic weight vectors $\mathbf{w}_1$ and $\mathbf{w}_2$. Based on the synaptic weight vectors, the PCC unit can be used for feature extraction operations. The computation results are stored in Buffer D as feature vectors $\mathbf{f}_n, n = 1, ..., t,$ for the subsequent FCM clustering operations.

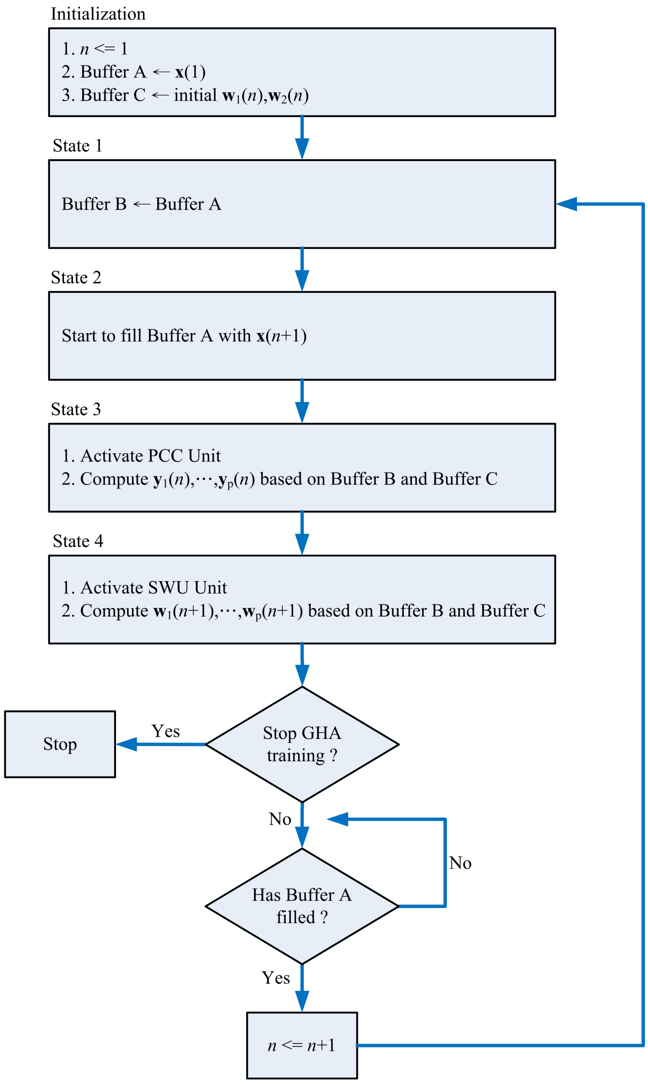

Figure 5: The training operations of the GHA circuit.

E. NOC-based GHA System

The proposed architecture is used as a custom user logic in an NOC system consisting of softcore NIOS CPU, DMA controller and on-chip RAM. An NOC is a new platform for implementing advanced SOCs in an interconnection network strategy, which solves the problems of traditional bus architecture, such as communication efficiency, latency and single clock synchronization. In a typical spike sorting system, delivery of spike signals, feature vectors, and classification results are required. The NOC therefore is beneficial for enhancing transmission speed and throughput of the proposed architecture.

In the NOC system, all the detected spikes are stored in the on-chip RAM and then transported to the proposed GHA circuit for feature extraction. The DMA-based training data delivery is performed so that the memory access overhead can be minimized. The softcore NIOS CPU coordinates different components in the NOC. It is responsible for circuit activation and control. The results of feature extraction are stored in the memory unit of the GHA circuit for subsequent clustering

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TABLE I: CCRs of the Proposed Architecture for the Spike Sorting with Different SNR Levels.

		SNR (dB)					
		1	2	4	6	8	10
$c=2$	t	1651	1638	1621	1656	1662	1653
	$\bar{t}$	1644	1632	1617	1654	1660	1651
	CCR	99.58%	99.63%	99.75%	99.88%	99.88%	99.88%
$c=3$	t	1850	1860	1842	1870	1873	1828
	$\bar{t}$	1571	1672	1737	1791	1812	1769
	CCR	84.92%	89.89%	94.30%	95.78%	96.74%	96.77%

operations.

IV. Experimental Results

In order to evaluate the performance of the proposed architecture for spike sorting, the simulator developed in [8] is adopted to generate extracellular recordings. The simulation gives access to ground truth about spiking activity in the recording and thereby facilitates a quantitative assessment of architecture performance since the features of the spike trains are known a priori. Various sets of spikes with different signal-to-noise (SNR) ratios and interference levels have been created by the simulator for our experiments. All the spikes are recorded with sampling rate 24000 samples/sec. The length of each spike is 2.67 ms. Therefore, each spike has 64 samples. The dimension of vectors for GHA training therefore is $m = 64$. The number of PCs is $p = 2$ for the circuit design.

We first consider the classification correct rate (CCR) of the proposed architecture. The CCR for spike sorting is defined as the number of spikes which are correctly classified by the total number of spikes. To show the robustness of the proposed architecture against noise interference, various SNR ratios are considered, ranging from SNR=1 dB to 10 dB. Table I shows the resulting CCRs for the spike trains with two target neurons ($c = 2$) and three target neurons ($c = 3$). The duration of the spike trains is 14 seconds. The spikes extracted from the spike trains are used for the GHA and FCM training, as well as spike classification. The total number of spikes used for training and classification (t), and the number of spikes which are correctly classified ($\bar{t}$) are also included in the table. Because the performance of FCM training may be dependent on the selection of initial vectors, each CCR value in the table is the average CCR values of 40 independent executions. From the table, it can be observed that the proposed architecture is able to attains CCR above 84 % for $c = 3$ when SNR=1 dB.

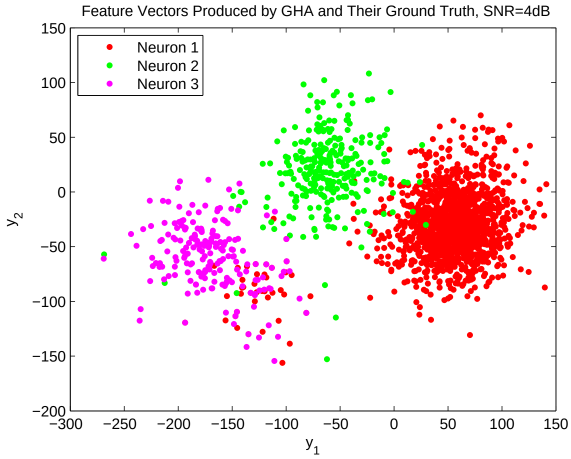
Table II compares the CCRs of the GHA- and PCA- based spike sorting algorithms. They all use the FCM method for clustering. It can be observed from the table that the GHA algorithm has slightly higher CCRs over the PCA for various SNR levels. The hardware implementation of the PCA may be difficult because it requires the covariance matrix of input data. Therefore, the GHA algorithm is well suited for spike sorting due to its simplicity for hardware design, and its high CCRs.

To further elaborate the effectiveness of the GHA and FCM algorithms for spike sorting, Figure 6 shows the distribution of GHA feature vectors of spikes for SNR=4, and the results of FCM clustering. The center of each cluster produced by FCM are also marked in the figure. By comparing Figure 6.(a) with Figure 6.(b), we see that the proposed GHA and FCM circuits are able to correctly separate spikes even for large noises.

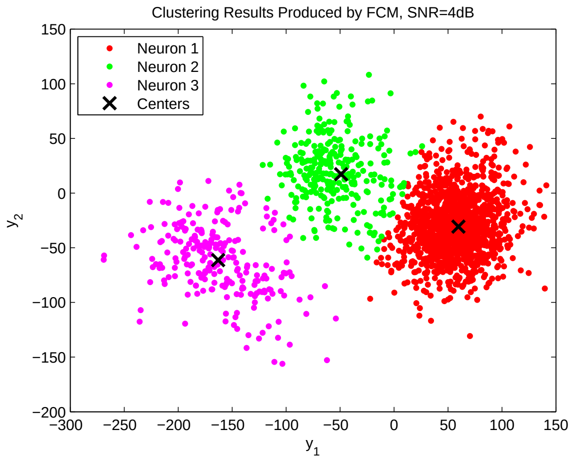
The proposed NOC-based spike sorting system features fast computation. Table III shows the training time of the proposed GHA system for various clock rates. The design platform for the experiments is Altera Quartus II with Qsys [9]. The target FPGA device is Altera Cyclone IV EP4CGX150DF31. The training time of its software counterpart running on Intel I7 processor is also included in the table for the comparison purpose. The training set for these experiments consists of 800 spikes. The number of epoches for GHA training is 100. It can

TABLE II: CCR Values of Various Spike Sorting Algorithms.

	SNR (dB)				
	1	4	6	8	10
GHA	84.92 %	94.30 %	95.78 %	96.74 %	96.77 %
PCA [5]	84.21 %	94.08 %	95.72 %	96.69 %	96.72 %



(a)



(b)

Figure 6: The distribution of GHA feature vectors of spikes, and the results of FCM clustering for SNR=10. (a) Ground truth of neuron spikes, (b) Clustering results produced by FCM.

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TABLE III: THE TRAINING TIME OF THE PROPOSED GHA CIRCUIT FOR VARIOUS CLOCK RATES.

Implementation	NOC-based GHA						Software GHA
Processor	Altera NIOS II						Intel I7
Clock Rate	50 MHz	200 MHz	400 MHz	600 MHz	800 MHz	1 GHz	2.61 GHz
GHA (ms)	35.60	8.92	4.46	2.97	2.23	1.78	181.38

TABLE IV: COMPARISONS OF THE PROPOSED GHA CIRCUIT WITH OTHER FPGA-BASED FEATURE EXTRACTION IMPLEMENTATIONS.

Arch.	FPGA Devices	Logic Cells or LEs	DSP elements or Multipliers	Embedded Bits	Max. Clock Rate	Throughput
Proposed GHA Arch.	Altera Cyclone IV EP4CGX150	15688	128	63488	1G Hz	4.50×10^7
GHA Arch. in [6]	Xilinx Virtex 6 XC6VSX315T	12610	12	0	100M Hz	1.60×10^6
GHA Arch. in [7]	Xilinx Cyclone IV EP4CGX150	9144	432	63448	50M Hz	2.75×10^6

be observed from Table III that the propose NOC-based spike sorting system is able to operate up to 1 GHz clock rate. In addition, the GHA training time decreases linearly with the clock rate. When clock rate becomes 1 GHz, the total training time of the proposed NOC-based spike sorting system is only 1.99 ms. By contrast, the training time of the Intel I7 processor is 193.18 ms. The speedup of the proposed hardware system over its software counterpart is therefore 97.08.

In Table IV, we compare the area costs and throughput of the proposed GHA circuit with those of other FPGA-based hardware implementations [6], [7] for feature extraction. The throughput is defined as the number of input training vectors the circuit can process per second. It can be observed from Table IV that the proposed GHA architecture attains highest clock rate and throughput at the expense of higher area costs. In fact, the proposed architecture has throughput 28.125 (4.50×10^7 vs. 1.60×10^6) and 16.3 times (4.50×10^7 vs. 2.75×10^6) higher than that of architectures in [6] and [7], respectively. The proposed algorithm has superior performance because it is based on shift registers for storing weight vectors and input vectors for high speed computation. In addition, although the proposed architecture has higher area costs, it only consumes a small fraction of the hardware resources available in the target FPGA. In fact, there are 149760 LEs, 6635520 embedded memory bits, and 360 multipliers in the target FPGA Cyclone IV EP4CGX150DF31. The proposed architecture utilizes only 10.78 %, 0.96 % and 35.56 % of the LEs, memory bits and multipliers of the target FPGA, respectively. All these facts show the effectiveness of the proposed architecture.

V. CONCLUSION

The proposed architecture has been implemented on the Altera FPGA Cyclone IV for physical performance measurement. The architecture is used as an hardware accelerator to the NIOS CPU in a NOC platform. Experimental results reveal that the proposed spike sorting architecture has advantages of high CCR and high computation speed. For SNR=10, its CCR is above 96 % for three target neurons. When SNR becomes 1 dB, it is still able to retain CCR above 84 %. The architecture is able to achieve 1 GHz clock rate. The speedup over its software counterpart running on Intel I7 processor is above 97. The GHA circuit has higher computation speed as compared with existing hardware implementations for GHA feature extraction. These results show that the proposed system implemented by FPGA is an effective realtime training device for spike sorting.

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Icy-Core Framework for Simulating Thermal Effects of Task Migration Algorithms on Multi- and Many-Core Architectures

Molka Becher, François Pacull
CEA, LETI, DACLE, LIALP
F-38054 Grenoble, France
{molka.becher, francois.pacull}@cea.fr

Saddek Bensalem
VERIMAG Lab, UMR 5104, UJF
F-38041 Grenoble, France
saddek.bensalem@imag.fr

***Abstract*—The design of complex many-core architectures represents a milestone for the next generation of mobile applications. But concentrating computing elements in a small area will raise the temperature of the chip very high. Consequently, the silicon is subject to overheating that leads to aging and damaging effects. Thermal-aware task migration algorithms could be a solution to this problem. In order to evaluate these algorithms, a test environment for analyzing dynamic thermal management solutions is required. In this paper, we present Icy-Core as a framework for simulating thermal effects of task migration algorithms on multi- and many-core platforms.**

Keywords*—*Many-Core Architectures; Thermal Management; Task Migration; Simulation.

I. Introduction

Currently, the trend for embedded system industry is to increase the number of computing elements in their platforms. This allows to design more powerful and flexible systems with a reduction of power consumption. However, this requires additional mechanisms to master power. One of the possible interesting aspects to study is the dynamic reconfiguration of the *application mapping* during its execution. By dynamic mapping of an application, we denote the reconfiguration of application task assignment to the platform cores at run-time, namely migration of tasks from one core to another.

In a first approach, we can consider two main reasons to trigger such a reconfiguration of the application:

Firstly, when we consider complex embedded systems (e.g., a smart-phone) which can run different type of applications or services, we need to dynamically manage the resources shared between the different programs according to unforeseen events. For instance, receiving a call while playing a game or displaying a video.

Secondly, due to the increase of the number of cores in a very reduced surface, the efficient management of thermal dissipation becomes more complex. As thermal stress can reduce the lifetime of the circuit or lead to its damage, dynamic thermal management of these platforms is necessary.

In general, to avoid this damage we use Dynamic Voltage and Frequency Scaling (DVFS), which acts on the frequency and voltage of cores in order to reduce the power consumption, and by consequence the temperature. This is part of the solution, however, this degrades performance in terms of execution speed since it makes some cores running slower. Dynamic reconfiguration, i.e., moving a task from one core to another, targets to keep the same level of performance. In fact, task migrations may come with overhead costs due to data transfer or frequent migrations. Therefore, there is a compromise to find between reducing migration costs and maximizing performance.

In this paper, we address only the thermal management of multi-core platforms. We present our framework, called Icy-Core, which has been designed to evaluate task migration algorithms on multi- and many-core platforms. The purpose of this paper is to discuss the framework capabilities. Thus, for the case study part, we use a naive task migration algorithm only to illustrate the framework features.

The paper is structured as follows. Section II is dedicated to related work. Section III presents the goal of Icy-Core framework, its overall architecture and its operating mode. In section IV, we present our case study through a simple example of task migration and the simulation time of Icy-Core. Section V introduces the advantages and limitations of Icy-Core. Finally, we conclude the paper in Section VI.

II. Related Work

In the literature, there is a variety of simulation frameworks. Simulation features and environment structure are different from one framework to another depending on simulation objectives. Heterogeneous Architectures and Networks-on-chip Design and Simulation framework (HANDS) [1] [2], Structural Simulation Toolkit (SST) [3], Multicore Power, Area, and Timing framework (McPAT) [4], Polaris [5] roadmapping toolchain and Wattch [6] framework are used at early design stages while Bartolini et al. virtual platform [7] is designed for evaluation of control strategies at run-time.

HANDS is a modular tool to simulate performance, power, temperature and reliability metrics for exploring network-on-chip interconnections. McPAT framework combines power, area and timing modeling to performance simulation to help architects evaluate the future of many-core architectures. Polaris, a system-level framework, focuses on the choice of network-on-chip interconnection, which preserves performance and physical constraints. Wattch and SST are architecture-level frameworks. The former is designed to evaluate architecture power consumption and optimize its power dissipation while the latter simulates power, area and temperature for exploring network-on-chip design. Bartolini et al. virtual platform evaluates the control strategies of many-core systems by simulation of power, temperature and reliability management. There is also a commercial tool called AceThermalModeler [8], which is paired to Aceplorer [8], a power simulator, with the intention of getting a thermal simulation.

Compared to these frameworks, Icy-Core has a different objective which is to simulate thermal effects of task migration algorithms. Thus, for the thermal simulators as stand-alone tools, we considered HotSpot [9] and 3D Interlayer Cooling Emulator (3D-ICE). HotSpot is a popular simulator that generates an equivalent circuit of a targeted 2D or 3D integrated circuit based on the thermal resistances and capacitances. As for 3D-ICE, it was developed on the same compact thermal model of HotSpot simulator. So, it uses the same format of inputs / outputs and the same model for thermal problem construction. 3D-ICE offers features for accessing the thermal profile details during the simulation. Moreover, it is easily encapsulable. This is the main reason of our choice.

III. Icy-Core Framework

As presented in the section before, several simulation frameworks exist for exploring many-core and network-on-chip design. However, none provides an integrated tool able to evaluate task migration algorithms according to a thermal constraint. We propose here to simulate application power consumption and targeted platform *thermal profile*. The thermal profile of a platform is the set of temperatures at a given time of the smaller units –*thermal cells*– in the chip depending on the granularity of the discretization specified. When the thermal profile is depicted by a graph, it's called *heat map*. Having an idea of the platform thermal profile helps us to make decisions of task migration and to evaluate later the decisions made. This is the idea of Icy-Core framework.

A. Icy-Core Goal

The purpose of Icy-Core is to assess task migration algorithms by drawing up the thermal profile of an application mapping upon a targeted platform. As the application mapping gives the location of tasks on cores, we can compute core temperatures and detect overheating that could damage cores. Then, we can apply on the initial mapping the required changes to avoid these undesired effects.

The main contribution of this work is the design of a modular framework which allows us to monitor and compare thermal behavior of task migration algorithms.

B. Icy-Core Architecture

Fig. 2 illustrates the overall architecture of our proposed tool. Icy-Core is composed of five modules. There is two existing components that we modified: the *Platform Simulator* and the *Thermal Simulator*. The *Task Migration Algorithm* is the module we want to test. We specifically developed the *Power Profile Generator* and the *Heat Map Visualization* modules. The role of each module is detailed here after.

1) Platform Simulator: For our study, we have chosen as target platform the ST-microelectronics Heterogeneous lOw poweR Many-core (STHORM) described in Fig. 1. A platform simulator provides performance estimation about the cores and the network interconnections.

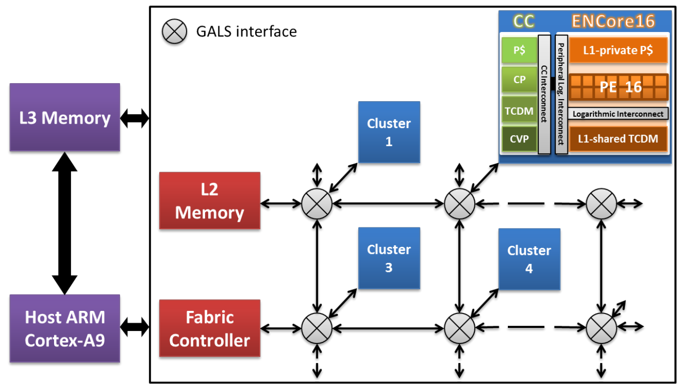

Fig. 1. STHORM Architecture

STHORM, formerly called P2012 [10], is a many-core accelerator. The platform is composed of a *fabric controller* that controls the system and four clusters connected by an *Asynchronous Network-on-Chip (ANoC)*. The routers of this *NoC* provide a *Globally Asynchronous Locally Synchronous (GALS)* scheme. There are up to sixteen *Processing Elements (PE 16)* per cluster. Each cluster is composed of a *Cluster Controller (CC)* and a *multi-Core computing ENgine (ENCore)*. The *ENCore* contains the cores (PE 16), a *L1-private Program cache (P$)* and a local *Tightly-Coupled Data Memory (TCDM)* shared by all the cluster processors. The *CC* consists of a *Cluster Processor (CP)*, a *Program cache (P$)*, a *local Tightly-Coupled Data Memory (TCDM)* and a *Clock Variability and Power (CVP)* module. The latter is in charge of dynamic frequency scaling.

The interest of this platform is firstly the number of cores (up to 64) and secondly its hierarchical and non-uniform architecture. STHORM platform offers different simulation environments at different accuracy levels. From the fastest to the slowest simulator we have:

- The Gepop-Posix environment for X86 functional simulation that provides fast results but less accuracy;
- The Gepop-ISS environment for first-level time approximation that is an Instruction-Set-Simulator on STxP70 code;
- The Gepop-ARM-ISS environment that is an Instruction-Set-Simulator on ARM processors.

In our case, we use the Gepop-ISS environment, which provides the core power consumption according to the power model [11] of the platform.

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2) Thermal Simulator: We have chosen 3D-ICE [12] [13], developed by EPFL, as our Thermal Simulator. Generally, a thermal simulator is responsible for computing the thermal profile of the application mapping on the targeted platform.

3D-ICE is a simulation platform able to analyze the temperature trace in three dimensional integrated circuits. This simulator is fast enough to allow quick feedback to the user. This is important for the global reactivity of our tool.

In our case, we use the software thermal library of 3D-ICE and we encapsulate it in such a way we can call it remotely to obtain the current heat map of the platform.

In addition, the *spatial thermal distribution* is modeled in 3D-ICE by a 3D resistance and capacitance network. Spatial thermal distribution means that heat moves in three directions depending on distribution of the platform materials, cooling mechanism (i.e., heat sink) and power consumption of the integrated circuit elements.

3) Power Profile Generator: A *power profile* corresponds to the power dissipation values of each functional block of a chip. The power profile of processing elements is needed by the Thermal Simulator to compute the thermal profile of the platform.

For the Icy-Core initialization phase (see Section III-C1), we obtain the power dissipation values of each core from the platform performance traces. These power dissipation values corresponds to the *core power profile*. The Platform Simulator takes too much time to run all the application and to compute performance traces. In addition, it does not allow us to modify the mapping of the application during the execution. Yet, to iterate the dynamic reconfiguration process (see Section III-C2), we need to modify the *task mapping*, i.e., which task runs on which core, during the execution of the application according to the thermal stress. Here comes the role of the Power Profile Generator module which computes the core power profile according to the *task power profile* and the task mapping. The task power profile corresponds to the power consumption values of each task of the application while the task mapping is the output of the Task Migration Algorithm. We compute the task power profile according to the information extracted from the Platform Simulator and the initial task mapping.

To sum up, the Power Profile Generator aggregates the task power profile and the task mapping to supply the Thermal Simulator with the core power profile. Moreover, the Power Profile Generator stores the different task mappings to cores for traceability.

4) Task Migration Algorithm: This module is a placeholder for the Task Migration Algorithm we want to benchmark. It takes as input the outputs of the Thermal Simulator and computes accordingly which tasks need to be migrated from one core to another. Thus, it takes the mapping decision for the dynamic reconfiguration of the application on the targeted platform given the current context. The task migration algorithm may follow different strategies from very simple to sophisticated ones.

In the literature, dynamic thermal management has been addressed through different strategies: combination of techniques to stop rising the chip temperature [14], thermal predictive scheme [15], distributed thermal balancing [16] with agent-based thermal management [17]. These environments are applied to simplified architectural models either because the number of cores is limited or because the architecture itself is very regular. They seem to be very far from the architecture of current multi-core trends. However, we will test these algorithms on our environment to understand their strengths and weaknesses and we will use them as baseline for our own algorithms.

5) Heat Map Visualization: The Thermal Simulator produces the simulation outputs as a sequence of numerical values that are not easily readable by human. To get more visibility on the simulation outputs, we developed the Heat Map Visualization module. It is used to plot heat map graphs corresponding to the distribution of the temperatures in the chip at a given time.

C. Icy-Core Operating Mode

As depicted in Fig. 2, our framework Icy-Core is decomposed into three parts: *Initialization*, *Main loop* and *Visualization* which use the five modules described above. The *Initialization* part corresponds to the Platform Simulator module. The *Main Loop* connects three modules: Power Profile Generator, Thermal Simulator and Task Migration Algorithm. The *Visualization* part corresponds to the Heat Map Visualization module.

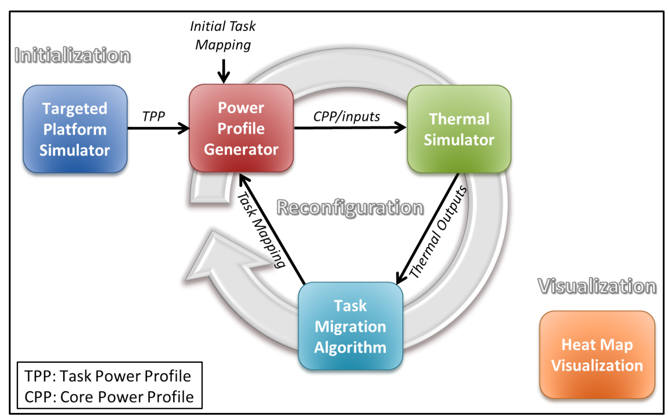

Fig. 2. Icy-Core Overall Architecture

1) Initialization: This phase purpose is to compute the application power consumption through the Platform Simulator. Thus, we run the application on the simulator to obtain performance estimation traces. From these traces, we obtain the average power consumption for each core during the execution of the application. In the implementation of the application, we specify which task runs on which core to have an idea of the initial task mapping. Knowing the initial task mapping and the execution time of each task, we compute the average power consumption of each task called the *task power profile*. This is the bootstrapping phase run only once at the beginning.

2) Main Loop: Dynamic Reconfiguration: This loop is composed of three sub-steps:

- Computation of the power profile of each core: It is the corresponding power consumption of a core deduced from the task power profile and the the task mapping. If the reconfiguration process is executed for the first time, the initial task mapping is taken into account. Otherwise, the new task mapping generated by the Task Migration Algorithm is used.
- Thermal simulation based on 3D-ICE simulator: It takes the power profile of cores and delivers the platform thermal profile.
- Execution of the Task Migration Algorithm: It detects from the *hot spots* (cores with thermal issues) which cores need to be managed. The objective of the algorithm is to find a solution of task migration such that the thermal issues decrease or disappear. The output is a new task mapping of the application on STHORM.

This loop runs in a regular time interval which is a parameter to define at the beginning of the simulation.

3) Visualization: At each iteration, the user may have a visual feedback of what is currently happening on the platform. The visualization phase is a separated module that can be called at each iteration to plot a heat map graph from the thermal outputs given by 3D-ICE. It represents the temperature of each thermal cell of the chip described by the Cartesian coordinates.

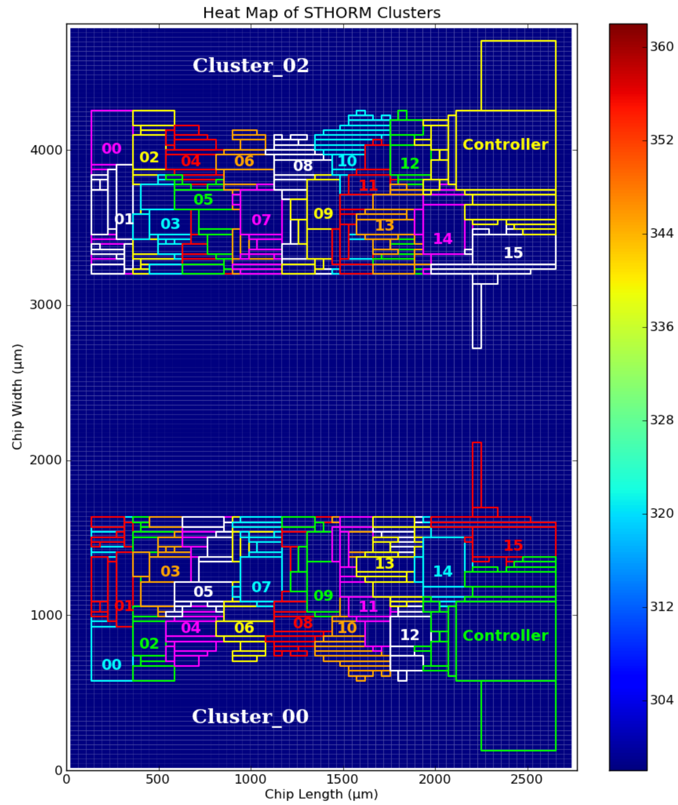

Fig. 3. Thermal Visualization of Two Clusters of STHORM When Idle

IV. Case Study

In order to illustrate how our system works, we consider to benchmark a naive task migration algorithm in the context of STHORM platform.

To simplify the case study illustration, we use only two clusters of STHORM: $Cluster\_00$ and $Cluster\_02$. Each cluster is composed of 16 cores and a cluster controller. Fig. 3 presents the thermal visualization of the two clusters when no task is running. The rectangles with same border color represent a functional block of STHORM. The initial temperature of the chip is 298K (24.85 °C).

By examining Fig. 3 more carefully, we can see that a core is decomposed into several sub-parts that are located at different places of a cluster. In our case, we manage only the core temperatures and we do not take account of the memories and the interconnections.

A. Transient Temperature Simulations

The task migration algorithm we have used as a toy example for the sake of simplicity is: At each iteration the hottest core above a given threshold is considered. We evaluate the tasks that are assigned to the core in order to determine the *greediest task*. The greediest task is the task that consumes more power than other tasks assigned to the core. Then, we migrate the greediest task to the coolest core of the platform.

As this algorithm sorts the core temperatures from the highest temperature to the lowest one, it has a linearithmic time complexity of $O(n\ log(n))$ in the average case. This algorithm is very simple, but it helps in the understanding of the framework.

The step by step execution is given from Fig. 5 to Fig. 7. At each step, we execute the main loop of our framework as shown in Fig. 4. It basically consists in pipelining the output of the previous module in the loop into the next one.

loop
  powerProfile ← PowerProfileGenerator(*taskMapping*)
  thermalProfile ← ThermalSimulator(*powerProfile*)
  heatMap ← Visualization(*thermalProfile*)
  hotspotList ← TaskMigrationAlgo(*thermalProfile*)
  if *hotspotList* not empty **then**
    taskMapping ←
      TaskMigrationAlgo(*taskMapping,* hotspotList)
  else
    exit loop
  end if
end loop

Fig. 4. Icy-Core Main Loop

As input, we suppose we have 20 tasks running on the platform. Step I of Fig. 5 is the result of the initial task mapping deduced from the Platform Simulator. According to this mapping, our algorithm detects two hot spots in the cluster number two: $Core2\_1$ and $Core2\_3$. Their respective temperatures 353.6K (80.45 °C) and 351.6K (78.45 °C) exceed the threshold fixed at 350K (76.85 °C).

As $Core2\_1$ is warmer than $Core2\_3$, the task mapping

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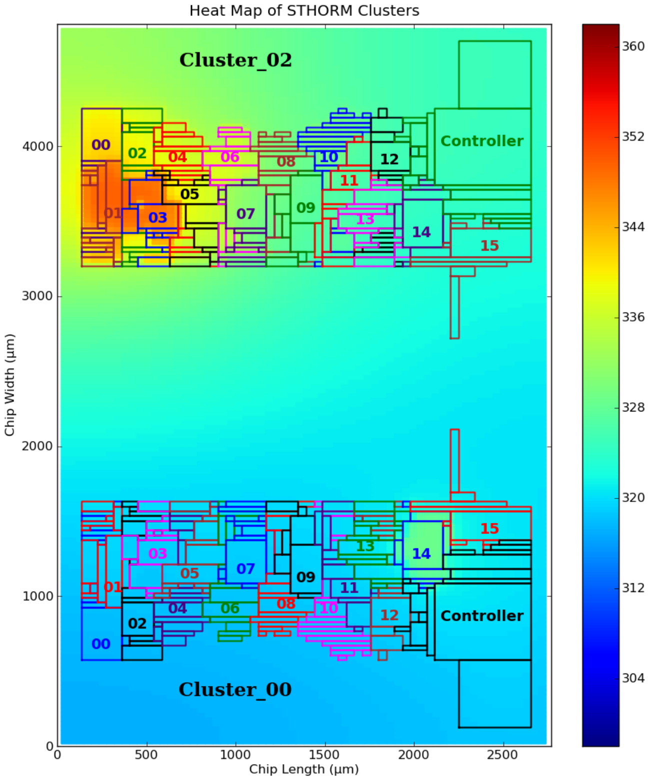

Fig. 5. Step I: Detection of $Core2\_1$ and $Core2\_3$ as hot spots

algorithm takes the decision to handle $Core2\_1$ first.

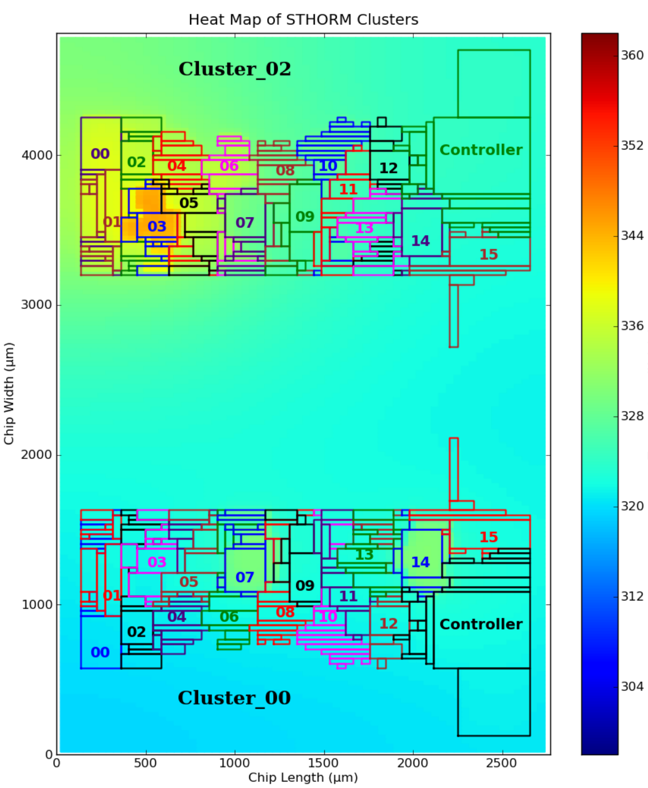

Fig. 6. Step II: Task migration from $Core2\_1$ to $Core0\_7$

Thus, the greediest task executed on $Core2\_1$ is migrated to the coolest core, in our example $Core0\_7$. This migration corresponds to Step II and it is represented by Fig. 6. When several destination cores having the lowest temperature coexist, one is chosen randomly.

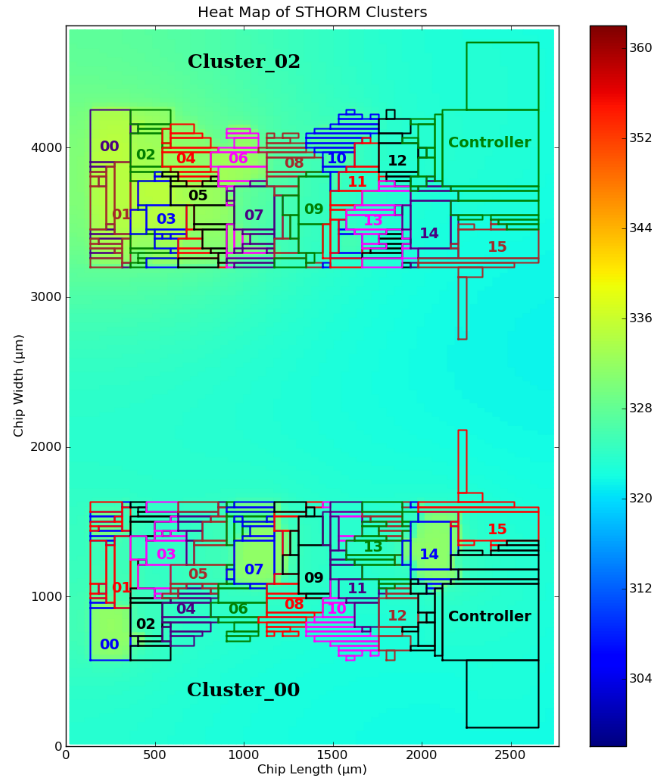

Fig. 7. Step III: Task migration from $Core2\_3$ to $Core0\_0$ – No core is above the threshold, the system is thermically balanced

In spite of this migration, $Core2\_3$ is still considered as hot spot (see Fig. 6). Thus, in Step III, the task running on $Core2\_3$ is migrated to $Core0\_0$. Following this decision, the system thermal profile is well balanced as there is no more core temperature above the threshold (see Fig. 7).

B. Icy-Core Simulation Time

Icy-Core has been tested by the initial task migration algorithm to determine the simulation time overhead. Table I shows the total CPU time for each module of Icy-Core and the total time scale of the loop.

TABLE I
SIMULATION TIME OVERHEAD

Module	Simulation Time (Seconds)
Heat Map Visualization	2.268
Power Profile Generator (PPG)	0.032
Thermal Simulator (TS)	0.102
Task Migration Algorithm (TMA)	0.028
Icy-Core Loop (PPG + TS + TMA)	0.162

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The Icy-Core loop has been customized to take effect each $200ms$.

V. Icy-Core Advantages and Limitations

To dynamically manage the chip temperature, an input of the current thermal state is needed to react when hot spots occurs. However, currently, the temperature of the chip and its localized elements is not provided by the platform sensors. Thus, this framework is required to have an idea of the platform thermal profile during the execution of the application. This helps us to take decisions of migration according to the thermal stress. In addition, it allows us to change the task mapping of the application during the execution. Currently, this is not possible in the platform simulator as the current execution model is run-to-completion.

Thanks to Icy-Core modular structure, we can test various task migration algorithms against the thermal effect. These task migration algorithms can later be compared according to different criteria, for instance, the task migration overhead, the maximum temperature reached, etc. We can so determine a task migration algorithm suited to our target platform. Icy-Core does not consider all the hardware constraints, however, it allows us to save the high cost of a chip.

VI. Conclusion

This paper presents a framework implementation to assess task migration algorithms on a targeted many-core platform. Differently from the existing simulation frameworks, the purpose of this framework is to ease the analysis of dynamic reconfiguration solution in order to face thermal issues. The proposed framework represents a combination of two existing modules: i) a Platform Simulator (Gepop-ISS), ii) a Thermal Simulator (3D-ICE), two specially built modules: i) a Power Profile Generator, ii) a Heat Map Visualization module and the Task Migration Algorithm we want to test.

The advantage of the Icy-Core architecture is its generic model. Thus, it is used to test various dynamic reconfiguration algorithms against the same constraint like thermal effect. This helps us to compare the reaction of these algorithms according to the same constraint. In addition, it allows us to apply different constraints than thermal one in order to manage different problems. For instance, to achieve resource management purposes, e.g., management of the platform memory, the thermal simulator module can be replaced by another module that estimates memory utilization.

We are currently investigating improvements made on the STHORM simulator power profile in the frame of management of low power consumption. This will favorably impact the accuracy of our tool and will help us in designing an efficient task migration algorithm for the targeted platform, which is our ultimate objective.

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Comparison of Receiver Architecture in Terms of Power Consumption and Noise Figure for Cochlear Implants Application

U. Cerasani, W. Tatinian
University Nice Sophia Antipolis
LEAT, Valbonne, France
umberto.cerasani@unice.fr
william.tatinian@unice.fr

P. Ndungidi, C. Dualibe, C. Valderrama
Université de Mons
Mons, Belgique
papy.ndungidi@umons.ac.be
Fortunato.Dualibe@umons.ac.be
Carlos.Valderrama@umons.ac.be

***Abstract*– Change in patients requirements has lead to new cochlear implants architecture specifications. In this paper, we proposed the receiver chain architecture study and design for new type of cochlear implants. Two receiver architectures were compared in terms of power consumption and Noise Figure, both of fundamental importance in biomedical embedded systems. SPICE simulations of the these architectures were carried out and transient results were presented for the solution retained. Furthermore optimization of the Low Noise Amplifier (LNA) using mathematical computations is presented, increasing the entire receiver performances.**

Keywords: SPICE Simulations, RF Receiver, LNA Optimization

I. INTRODUCTION

Cochlear implant is a device aiming to aid severely deaf people to partially recover hearing (electrical description of cochlear implant can be found in). A prototype aiming to make the implant less noticeable is presented in. In this new cochlear implant prototype, the transmitter is inserted inside the auditory canal, sending Radio Frequency (RF) waves (allowing reduced antenna size) to the receiver located inside the patient' s skull. As the auditory canal space is limited, the battery size and its charge are the main limiting factors for the emitter design, diminishing its performances.

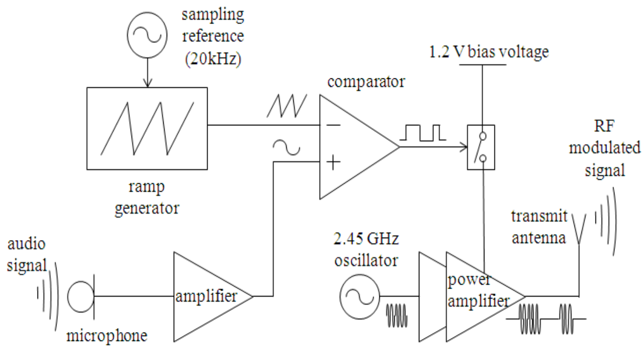

Figure 1: Transmitter architecture (redrawn from)

According to the work presented in, the Digital Signal Processor (DSP) that was previously located in the emitter, can be integrated on the receiver. This has the advantage of reducing the emitter consumption but on the other hand implies an electrical architecture reworking. The emitter has already been presented in and the architecture is recalled in Figure 1. The input signal coming from the microphone was compared with a ramp reference signal to create a Pulse Width Modulated (PWM) signal, then multiplied by a 2.45GHz oscillator to create the RF signal sent to the transmitter antenna.

The aim of this paper is to select the receiver architecture fitting the most with the cochlear implants specifications and then to optimize its critical block in terms of Noise Figure and power consumption as explained in Section III. The receiver architectures overview is presented in Section IV, followed by the mathematical optimization of the LNA. Two different types of receiver front end architectures are then discussed and compared in terms of added noise and power consumption (Section V.), before presenting our concluding remarks and preferences.

II. RECEIVER ARCHITECTURE

The communication channel between the transmit- and the receive antenna is presented in, leading to an estimated received power of -87dBm, and added noise in the propagation channel of -95dBm. Those values were computed from the maximal authorized emitted power in the ISM band (20dBm), the attenuation in human tissue with corresponding dielectric constants, the ISM bandwidth (80MHz) and assuming a white noise distribution. This fixes the LNA sensitivity and is used as basis for the receiver specifications and architecture selection.

The selected architecture for the receiver includes a low noise amplifier followed by an amplitude demodulation stage and a digital signal processing unit controlling the electrodes array, which is implanted inside the patient's cochlea.
Electrodes array are the most consuming part of the receiver (their power consumption estimation is around 50mW [*4*]) and this energy cannot be reduced for proper nerve fibers

excitation. The total electronic consumption should be significantly lower than the electrodes array consumption to increase battery lifetime

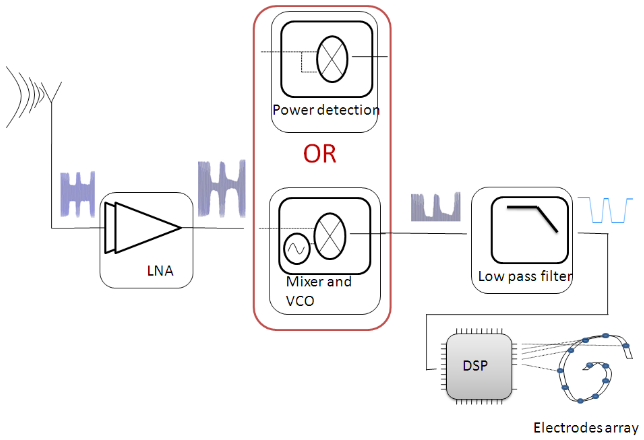

Figure 2 : Transmitter architecture overview, with signal outputs in color

Ultra low power processors currently available in market consume few µW at the 1MHz working frequency [5], [6]. Ultra-low-energy biomedical signal processors (BSPs) are nowadays being developed with overall energy consumption around 100 pJ/cycle with a working frequency of 1MHz (a state of the art review can be found in [7]). Therefore more constraints in term of power consumption are put in the RF/analog electronic devices of the receiver, which should not contain complex blocks for envelop detection.
Most particularly, a high consumption would lead to an increase of the battery reload frequency, which is inconvenient for the patients. Because of low power constraints, power repartition between blocks becomes fundamental. The unique block allowed to consume more power was the LNA in order to decrease the total Signal-to-Noise Ratio (SNR) of the entire receiver as explained in Section III.

As presented in Figure 2, two architectures are investigated for the AM demodulation: a power detection using a mixer with the RF and LO inputs connected together and a coherent demodulation composed of a mixer and a local oscillator at the same frequency as the input signal. The output signal is a PWM-like waveform and is low-pass filtered to recover the initial signal amplitude. Linearity is not significant as the PWM signal maximum frequency (around 2MHz) is substantially lower than the carrier frequency (2.45GHz).

III. LNA OPTIMIZATION

As a direct consequence of the Friss equation [8] (reminded in (1)) the Noise Factor of the entire receiver is greatly dependent on the LNA Noise Factor and gain. Consequently we decided to optimize the LNA design in terms of power consumption and noise figure.

$$F_{total} = F_{LNA} + \frac{F_{demod} - 1}{G_{LNA}} + \frac{F_{filter} - 1}{G_{LNA}\, G_{demod}} \qquad (1)$$

Where F_{LNA}, F_{demod} and F_{filter} are the Noise Factor of the LNA, the Noise Factor of the blocks composing each demodulating topology and the Noise Factor of the RC filter respectively. G_{LNA} and G_{demod} are the Power Gains with load of the LNA and the blocks composing each demodulating topology respectively.

TABLE I. : COMPONENT VALUES OF THE LNA OBTAINED BY MATHEMATICAL COMPUTATION

	Implemented values
Center Frequency	**2.475 GHz**
Voltage supply	**1.2 V**
Current Consumption	**10 mA**
L_L	**1.3 nH**
C_L	**3.18 pF**
$L_{M1,M2}$	**0.15 um**
$W_{M1,2}$	**165 um**
NF	**1. 4**
L_S	**0.14 nH**
L_G	**14.3 nH**

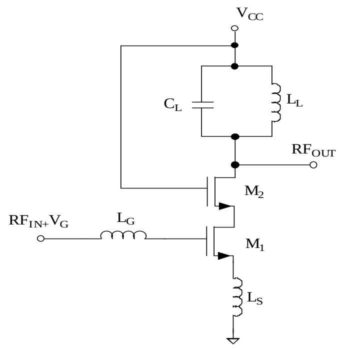

Figure 3: Circuit of the Low Noise Cascode Amplifier

Figure 3 shows the schematic of the implemented LNA. The cascode stage (M_2) is used to improve input-output isolation, reduce the Miller effect and, consequently, increase the bandwidth. The resonant circuit formed by L_L and C_L permits a high gain with a low voltage supply at the frequency of interest. With L_G and L_S a narrow band input impedance matching is obtained. The input transistor (M_1) is used in inductively degenerated common-source configuration.

TABLE II: PARAMETERS OBTAINED BY MATHEMATICAL COMPUTATION AND WITH CADENCE SIMULATION

	Mathematical computation	Cadence evaluation
RF Frequency	2.4GHz	**2.45GHz**
S21	>17dB	**24.7dB**
S11	-10dB	**-7.11dB**
S22	-10.1dB	**-1.5dB**
NF	<1.4 dB	**0.75dB**
Power Consumption	13.2 mW	**11.71mW**

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The width of the transistor and the other parameters of the circuit were mathematically computed using a similar development than the method presented by Thomas H. LEE [9], minimizing the Noise Figure while optimizing power consumption.

For both architecture, we evaluated the receiver performances using a 0.13µm RF CMOS technology. Table I shows the values obtained from calculations, after some minor modifications according to the target CMOS 130 nm process.The results of simulations are summarized in Table II.

In practice, equations used for this optimization were derived from first order modeling (SPICE level [10]) and provided general characterization of the LNA further refined by Cadence simulations (especially modifying the polarization of M1), explaining the improvements in Table II, especially in terms of NF and gain.

IV. Receiver Architectures Implemented

The two architectures presented in Figure 2 were implemented in SPICE and they were compared in terms of power requirements and noise addition. The LNA and the filter were common to both solutions. The polarizing circuitry is not shown in the figures.

A. *Solution 1: LNA and Power Detector*

The input signal is the one received by the antenna (2.4GHz carrier modulated in amplitude by a PWM signal) and the LNA output is connected to the Local Oscillator Input (LO +) as well as on the RF input of the Mixer. The Power Detector is therefore created using a mixer with its inputs connected to the same source. We designed a single balanced Mixer because of its low power consumption. However, as the DC offset is injected directly in the RF source, it degrades the mixer linearity [11], not of importance in our application.

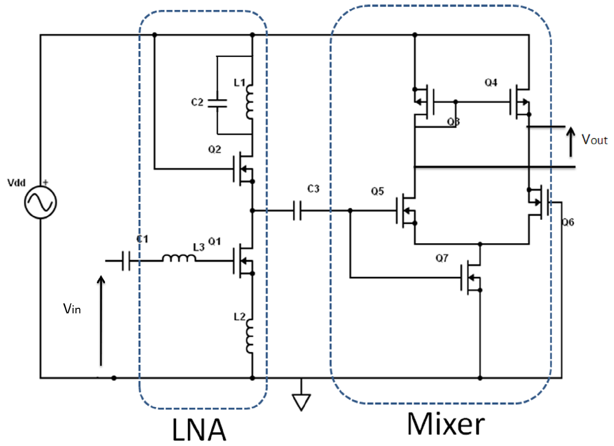

Figure 4: Receiver architecture solution 1 with a LNA and a power detector

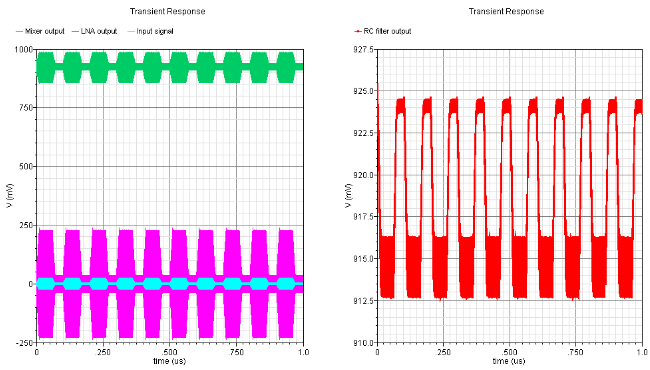

Figure 5: Transient simulations including the input signal, the LNA output, the mixer output (right) and the RC filter output (left)

The designed mixer has a NF of 20dB, and its output is directly connected to the RC filter. The simplified schematic of the Solution 1 is presented Figure 4 (with bias circuit removed for clarity reasons) and the associated transient simulations are presented in Figure 5.

For the test case, the PWM signal was created with a 1 kHz sine wave, which was sampled at 1MHz. The RF input signal is represented in light blue and the corresponding magnitude is low enough to make the LNA operate within the linear region. Removing the DC voltage, the output signal is represented in pink, where one can see that the ratio between the maximal and minimal magnitude is around 5, which is high enough for a further OOK demodulation. The effects of rising and falling times are also visible on the curve. As a result, after filtering, we obtained a signal that corresponds to the emitted signal, where the distortions and the low output magnitude are mainly due to the low power consumption of the mixer (100µA).

The main advantage of this architecture was the use of only two active devices (LNA and mixer) instead of three (in the other solution).

B. *Solution 2: LNA, Mixer and VCO*

The oscillator used is an L-C tank oscillator with a central frequency of 2.45GHz and frequency control (V_{tune}) to compensate the frequency shift due to temperature variations within the human body (presented in Figure 6).

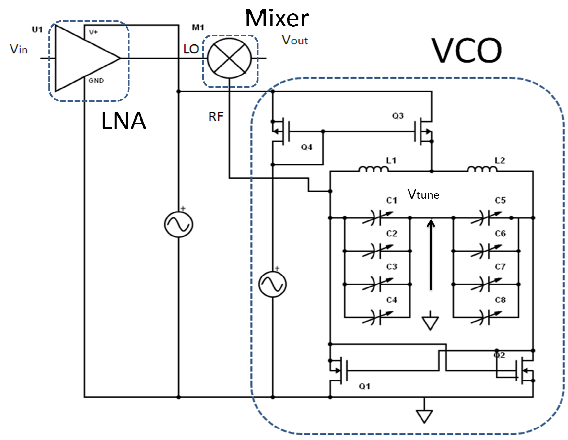

Figure 6: Receiver with mixer and VCO

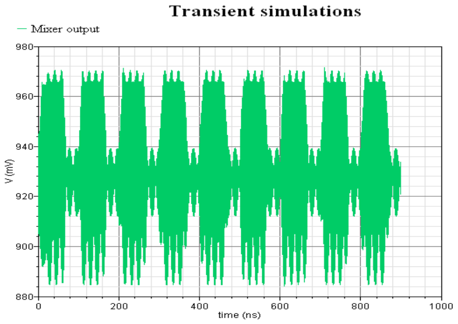

Figure 7: Transient simulation of the mixer output in the second architecture

The width of the NMOS of the differential pair is 210µm and the length is 150nm, which drives a current of 10mA. The Noise Figure of the cross coupled pair was obtained with SPICE simulations and was found equal to 7dB. Then, using Periodic Steady-State (PSS) analysis, we estimated the phase noise, which was around -120dBc/Hz at 1MHz from the carrier (PWM maximal frequency).

Transient simulation results are presented in Figure 7, where one can see that the output voltage magnitude of the mixer is much higher than in the previous architecture, at the cost of output oscillations due to the frequency shift between LO and RF. Indeed this shift is inevitable as low power constraints did not allow carrier recovery. Nevertheless, this distortion may be removed by subsequent Signal Processing.

V. ARCHITECTURE COMPARISON

The main requirements of our receiver was its low power consumption and a very reduced Noise Figure. Maximizing SNR at the receiver output (before DSP computation) is hence very relevant and consists on minimizing the total Noise Factor of the receiver.

TABLE III: NOISE FACTOR AND POWER COSUMPTION COMPARISON

	Power Detector	Mixer and VCO
Power Consumption (mW)	**11.91**	**21.4**
NF	**2.57**	**2.61**

Total Noise Factor computation is expressed by the Friss formula as previously stated in (1).

The Noise Factors comparison for each solution are shown in Table III, as well as their respective Power Consumption estimation.

The solution with VCO is consuming twice as much power as the solution with the Power Detector. This results in a total extra energy of about 100mA*h per day (assuming that the device is activated during 8 hours per day).

As discussed in Section II. the increase in power consumption of the RF front end represents between 20% to 40% of the power consumed by the electrodes array depending on the selected topology.

As stated in Section III.B., the effect of the VCO phase noise is not significant compared to the frequency shift associated with the absence of carrier recovery in the receiver. This may significantly corrupt the signal rectification as the signal translation into lower frequencies is followed by the RC filter.

In consequence, it is authors' belief that the first architecture should be preferred for this application. Optimization algorithms such as NSG2 presented in [12] are available and permits optimization of the overall architecture to maximize parameters of interest (still under work).

VI. CONCLUSION

This paper presented the study and design of the receiver front end architecture for new type of cochlear implants.

As the LNA block was compulsory in both the solutions and significantly impact the overall architecture Noise Figure, its optimization using mathematical computation was first performed before its implementation in Cadence and subsequent refinement.

Two receiver architectures were compared in terms of power consumption and Noise Figure, both of fundamental importance in biomedical embedded systems. The architecture with Power Detector was preferred for our application, as it consumed less and did not suffer of frequency shift, which could impact the signal demodulation.

The optimization of the overall LNA and Power detector architecture is ongoing, using a nonlinear multi-objective optimization, based on the Genetic Algorithm (GA) NSGA II.

This overall optimization may be necessary if the measured Noise Power inside the transmission channel would be greater that the estimated one, yet very close to the Signal Power received by the antenna.

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A Cartesian Methodology for an Autonomous Program Synthesis System

Marta Franova
LRI, UMR8623 du CNRS & INRIA Saclay
Bât. 660, Orsay, France
mf@lri.fr

***Abstract*— In this paper, we present the main difference between Newtonian and Cartesian approaches to scientific creativity when related to Program Synthesis (PS). The main contribution of the paper is a thorough discussion on the creative building of a theorem prover. We illustrate these ideas by an analysis of Peano's axioms defining the set of non negative integers, from the point of view of creativity. This analysis is then applied to the more complex case of the general framework for our own 'Constructive Matching Methodology' (*CMM*) as a Cartesian approach to the creation of an autonomous theorem prover for PS.**

Keywords-program synthesis systems; methodology; Constructive Matching methodology; creativity, symbiosis

I. INTRODUCTION

Automatic construction of programs is obviously a desirable goal. There are two main approaches to tackle with this task, namely inductive and deductive. In this paper, we are interested in the deductive approach to Program Synthesis (PS) introduced by Manna and Waldinger in the eighties [22] and followed by many authors, for instance [4], [5], [8], [10], [13], [21], [23], [26]. This problem is however undecidable as a consequence of Gödel's Theorems [19]. In this paper, we shall present an attempt to, as much as possible, approximate the automatization of the deductive approach to PS by introducing the conceptual switch of 'Cartesian Intuitionism', defined by Franova [16] and informally described in the book Franova [15].

This approach is, from an epistemological point of view, an interesting alternative and a complement to the more formal Newtonian approaches because it enables to handle informal specifications. Nevertheless, it is still too soon to compare these approaches on the basis of their relative performance. From a practical point of view, in building what we call Cartesian Intuitionism, we try to open the way to a creative approach that provides a frame of thought to the user of a theorem prover in the process of recovering from a failure.

Before going into the details of the structure of the paper, let us stress the role of Section IV of this paper. This Section contains an example illustrating and underlining the deep gap between creating a set of axioms, that is to say, Cartesian *creation* of these axioms and making use of a given set of axioms, that is to say Newtonian *construction* of a proof.

In Section IV, we illustrate Cartesian creation of the Newtonian theory of the non negative integers build using Peano's axiom. The idea is that each of its 5 axioms depends on the other ones to be justified. Besides, modifying one axiom modifies the others as we shall then illustrate. Another obvious example of Cartesian creativity, though at a much higher level, is provided by Lobachevski's geometry. Euclidian geometry is a very efficient Newtonian system. It becomes Cartesian when you try to play with the axiom relative to the parallels, where you 'create' universes where parallels in the same plane can cross once, several times, all cross at the same point or not etc. The creative aspects we deal with here are akin to these two examples: Our axiomatic system has to invent new axioms each time it meets a failure.

The paper is structured as follows. In Section II, we recall the formulation of the deductive approach to PS. In Section III, we recall the main features of Newtonian and Cartesian approaches to scientific creativity related to PS. In particular, we shall recall the basic notions of Cartesian intuitionism. Section IV has been already summarized. We shall devote Section V to the description of our Constructive Matching Methodology (*CMM*) in the light of Cartesian Intuitionism. In Section VI, we present a few epistemological remarks.

II. PROGRAM SYNTHESIS – DEFINITION OF THE PROBLEM

By program synthesis we call here the deductive approach to automatic construction of recursive programs introduced by Manna and Waldinger [22]. This approach starts with a specification formula of the form $\forall x\ \exists z\ \{P(x) \Rightarrow R(x,z)\}$, where x is a vector of input variables, z is a vector of output variables, P(x) is the input condition. R(x,z) is a quantifiers-free formula and expresses the input-output relation, i.e., what the synthesized program should do. A proof by recursion of this formula, when successful, provides a program for the Skolem function sf that represents this program, i.e., R(x,sf(x)) holds for all x such that P(x) is verified. In other words, program synthesis transforms the problem of program construction into a particular theorem proving problem. The role of the deductive approach is thus to build an inductive theorem prover specialized for specification formulas. There are two main problems with respect to this role:

1. Treatment of strategic aspects of inductive theorem proving system specialized for specification formulae.

We have illustrated this first problem on a simple specification theorem (a computation of the last element of a list) in [16] and a complex example (synthesis of

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Ackermann's function defined with respect to the second argument) is presented in [12].

2. Treatment of strategic aspects of creativity related to the design of such theorem prover.

The present paper is concerned with this second problem, that is, the one of building a system able to perform program synthesis. Strategic aspect can seldom been efficiently formalized. We moreover deal with creativity, the formal aspects of which start being explored. It is obviously too soon to present a general (Newtonian) theory of (Cartesian) creativity in the usual style of lemmas, theorems etc.; thus the nature of this paper is more epistemological than axiomatic.

III. NEWTONIAN AND CARTESIAN APPROACHES TO PS

In the previous Section, we have mentioned that we are here interested in the creative process of construction of an inductive theorem prover. This prover has to be specialized in specification formulae. There are two main styles in this creative process. For particular reasons presented in [16], we call 'Newtonian' the standard approach and 'Cartesian' the non-classical one. In [16], we have presented in detail the above two styles. In this Section, we shall recall the main features necessary for understanding the present communication.

A. Newtonian Approach

The specialists in the Newtonian approach to PS build their own theorem prover, one based on a logic of sequential research. Classically, the reference system of any theorem proving system consists in a set of axioms, rules of inference and control mechanisms devoted to finding a recursive proof for the specification formula. In the Newtonian approach, the various blocks composing the reference system are a composition of some tools chosen among the whole set of all existing tools. In the case where an author introduces a new block he/she invented him/herself, then this new block must be coherent with the existing tools. In other words, more formally, Newtonian approach considers creativity as a finite linear sequence:

beginning
advancement-1
advancement-2
…
advancement-n
end.

In this sense, Newtonian creativity is similar to essentialism within the frame of logics as defined by J.Y. Girard in [18].

Since this approach is based on standard mathematical knowledge, it inevitably inherits the negative results of Kurt Gödel [19]. The results of Gödel are said to be negative because they show that the objective of PS, as it is formulated in *beginning*, cannot lead to a successful end of the task in the classical framework. This happens because the classical approach focalizes on the problem

∃ formal framework
in which ∀ specification formula has a solution.

Gödel's results show the impossibility to define a formal logical framework, containing the natural numbers, allowing to deal with the automated resolution (confirm or counter) of specifications given in a general way. This is why Newtonian approaches react by placing themselves inside user-dependent theorem proving assistants, such as ACL2 [6], the system RRL [20], the system NuPRL [9], the Oyster-Clam system [7], the extensions of ISABELLE [24], the system COQ [3], Matita Proof Assistant [1] and Otter-Lambda [2].

B. Cartesian Approach

We have developed an alternative to this classical approach by taking into account the creativity necessary for designing a PS system from the point of view of what we call Cartesian Intuitionism [16]. This non-classical creativity

(a) focalises on the problem: {∀ specification formula ∃ formal framework in which the given specification formula has a solution}
(b) oscillates between the problems {∃ framework ∀ specification} and {∀ specification ∃ framework}
(c) considers the creativity process in its recursive cyclic version given by the scheme

$$\begin{tikzcd} \text{beginning} \arrow[rr, shift left] \arrow[dr, shift left] & & \text{end} \arrow[ll, shift left] \arrow[dl, shift left] \\ & \text{mean} \arrow[ul, shift left] \arrow[ur, shift left] & \end{tikzcd}$$

where the arrow means "steers".

These three points give to Cartesian Intuitionism the feature of a combination of what is called essentialism and existentialism within the frame of logics by Girard [18].

Points (a), (b), and (c) together mean that Cartesian approach to PS is based on a logic of recursive science where the reference system of the problem and the milestones of construction of the solution (i.e., the definitions and the rules of inference of a given specification formula) are formulated hand in hand with the development of the solution. Moreover, the exact demarcation of the reference system and the milestones of construction is the final stage of the process, and it is also a part of the solution. It follows that Cartesian approach specifies in an informal way the purpose to be reached, by a necessarily informal formulation of the reference system. For instance, *CMM* specifies the purpose of PS informally by the sentence: "Create a custom-made mechanism for proving specification formulae that automates PS as much as possible." We shall say more about our application of Cartesian approach to PS later in Section V, because we need first to clarify the model of creation for *CMM* in the following Section.

IV. NEWTONIAN CONSTRUCTION VERSUS CARTESIAN CREATION

In this Section, we shall be interested in the set of natural numbers N, seen here as a creation model for particular complex systems. More precisely, we shall point out the difference between the use and the creation of Peano's axioms. Peano's axioms define the arithmetic properties of natural numbers N. These axioms include a constant symbol

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0 and unary function symbol S. These axioms are usually used to build formal proofs about natural numbers. Our presentation does not deal with this topic, but with the one of reasoning about the construction of these axioms, that is the creation process involved in their building.

Supposing that the membership relation "$\in$" and the equality "=" are already defined, the basic Peano's axioms read:

(A1) $0 \in N$.

(A2) if $n \in N$ then $S(n) \in N$.

(A3) for all $n \in N$, $S(n) \neq 0$.

(A4) for all n, $m \in N$, if $S(n) = S(m)$, then $n = m$.

(A5) if M is a set such that

- $0 \in M$, and
- for every $n \in N$, if $n \in M$ then $S(n) \in M$

then M contains every natural number.

We shall tackle here, in this Section, with the difference between the use and the creation of these five axioms. To this purpose, we need to precisely specify the difference between synergy and symbiosis.

A. Synergistical Construction

An object is constructed *synergistically* when it can be considered as a result of the application of some specific tools from an existing tool-box**,** that is all the tools that have been developed in all scientific domains beforehand, for various purposes. These tools are not built in such a way that one calls another to solve one of its problems before this one has finished its computations. That is, tool B can call on tool A in one way only: the input of B contains a part of A computations, once A computations have been all achieved. It follows that these tools must be used independently of each other for the construction of other objects. During the construction process they do not lose their properties.

B. Symbiotical Construction

In contrast to this, an object is constructed *symbiotically* when its parts, maybe seemingly independent, have, during the construction process, no meaning as isolated entities and a slight change of one part influences the others and the whole as we illustrate later.

The main point we want to underline about Peano's axioms is that their ***use*** is synergetic, while their ***construction*** process is symbiotic. In other words, when ***using*** them, we can use several axioms as being independent entities and the constructing elements 0, S, and N can be considered as isolated from each other, though they are interdependent elements as show (A1) and (A2). The following example will show in which way Peano's axioms construction process is of symbiotic nature.

Let us first consider axiom (A1), which deals with 0 and N. This first axiom, however, does not say what is the full meaning neither of 0 nor of N. In particular, from this axiom we cannot conclude that 0 is a basic element and that N is the final object we want to define. The axiom (A1) expresses only an interdependence between two symbols 0 and N. The symbol $\in$, does not tell more than 0 is an "element" and N is one of sets to which this element belongs. There is no difference, apart substitution, between (A1) and (B1): "rose $\in$ garden". This means that the creator of Peano's axioms has already in mind a "vision" or an "informal specification" of what 0 and N mean for him in this first axiom. In other words, writing this first axiom, the axiom's creator intuitively knows what 0 and N will be once their description will be completed, i.e., when all the necessary (in this case five) axioms will be provided. In the creator's mind, the first axiom contains implicitly and intuitively all the remaining axioms and all the axioms are constructed from his/her intuitive vision of the "whole", i.e., N. Therefore, 0 and S do not belong to an already given tool-box and the meaning of 0, S and N in the construction process is custom-made. Moreover, 0, S, and N are symbiotic during the construction process and they are not synergetic parts. During the construction process, N steers the realization of 0 and S and vice versa, they cannot be considered as isolated already known elements. We shall present later an example illustrating this symbiotic character; but we now need at first to introduce some more notions.

C. Cartesian Creation

N is constructed with the help of three "elements", namely 0, S and N itself. Note that N self-reference is already acknowledged as a constructive recursive 'trick'. These construction parts are usually named 'the constructors'. We have already mentioned that these parts are symbiotic during the construction process, while when using the Peano's axioms for reasoning, we may consider them synergetic "*par la pensée*" (as Descartes puts it). In the following, instead of 'construction' we shall call this process 'Cartesian creation' in tribute to Descartes' §62 of *The Principles of philosophy* [11]. We shall use the following notation:

$$< A + B > = C,$$

where A, B are constructors and C is the created "whole" for this kind of a symbiotic Cartesian creation. This defines N in the following way:

$$< 0 + S + N > = N.$$

Now, we can illustrate the symbiotic character of the constructors 0, S and N. Let us consider Peano's axioms without (A3). In such a case we have the liberty to suppose that there exists $n \in N$ such that $S(n) = 0$. Let us suppose that $S(S(0))$ is such an element. We have then $S(S(S(0))) = 0$. Let us call (B3) this hypothesis. Then, (A1), (A2), (B3), (A4) and (A5) constitute a meaningful definition of the set that contains three elements, namely 0, $S(0)$ and $S(S(0))$. This new axiomatic definition defines a set, N3, that is finite and thus is different from the infinite set N defined by Peano's axioms. In other words, a little change in a property of one constructor altered the properties of all the constructors, including N which changed into N3. This is not the case in a synergic construction, where a change of one construction module may influence the behaviour of the whole but has no direct effect on the other modules. This explains why we so much stress the difference between symbiotic Cartesian creation and synergetic Newtonian construction. Once a symbiotic creation of a whole is completed, we may *think* of the constructors as being "unconnected" synergetic elements.

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We just have shown that this thinking is not valid during the creation process. This is why there is also a difference between a creation process and the use of the completed whole created by the same process.

An interesting feature of a symbiotic creation is that one cannot produce a sample or "architectural" miniature before the whole creation process is completed. Moreover, partial results are often incomprehensible outside the creation process which works mainly with informally specified problems that must be simultaneously solved. The drawbacks we just exposed must be one of the reasons why Cartesian creation is hardly reported in the scientific communications that concentrate on the result of the creation, not on their creative process itself. Researchers seem to prefer tool-box Newtonian progressive construction which provides the security of familiarity with such linear or modular processes. This may also explain why our original Cartesian approach is not used in the research on Program Synthesis.

Summarizing this Section, we can say that Cartesian creation focuses on building a system, a whole, by progressively inventing symbiotic constructors. Such a progressive process is possible since the first constructors and the whole are described by a 'mere' informal specification, as we shall show in the next Section. The standard Newtonian research is not accustomed to such an informal goal specification and it usually gathers already existing mechanisms that have been certainly not custom-designed for the given goal. This choice leads, during the construction process, to new problems, more often related to the chosen basic tools than to the given goal. These new problems ask for a new search for already existing tools and to attempts for adapting them to the given goal, a process that tends to fail when it is completely automated. In other words, in Cartesian creation, the basic tools, i.e., constructors and the whole system are custom-made, while in Newtonian construction, the basic words are "choice" and "adaptation" of already available tools.

V. *CMM* IN THE LIGHT OF CARTESIAN INTUITIONISM

The basic principle of Newtonian PS system is the use of a fixed set of specific strategies in order to solve the problems that are submitted to it. In case of failure, the user is requested to provide lemmas or axioms that lead to success.

The basic principle of Cartesian PS system is also the use of a specific strategy defined by the axioms upon which the system is built. But this is true only as long as the system meets no failure. In case of failure, we build a new PS system possibly with a new solving strategy. We already illustrated such behaviour by building the pseudo-Peano system by replacing (A3) by (B3) and N by N3. If this kind of incomplete natural numbers is used to prove a theorem containing the term, say S(S(S(S(0)))), the 'synthesis' will fail. In a Newtonian approach, the user would be asked for a lemma specific to S(S(S(S(0)))) that enables a success. In such a case our approach would propose to modify the system of axioms by changing (B3) and N3. We fully agree that, in this particular case, a human feels the needed modification as being trivial and would rather suggest to enlarge the solution to introducing N itself. See below a modification that is less easy to find.

Let us now provide a more complex example that illustrates a situation where modifying system of axioms defining PS mechanism is not trivial.

Newtonian system called Otter-Lambda is presented by Beeson [2], together with several examples of its execution. We have chosen among them a formula

$$\forall a\ \forall n\ \{\ S(0) < a \Rightarrow n < exp(a,n)\ \} \qquad (*)$$

that the Otter-Lambda system fails to prove when the basic information relative to (*) is given as a recursive definition of the exponentiation function exp (with respect to the second argument):

(1) $exp(u,0) = s(0)$

(2) $exp(u,S(v)) = u*exp(u,v)$

of the addition and of the multiplication with respect to the first argument:

(3) $0 + u = u$

(4) $S(v) + u = S(v + u)$

(5) $0 * u = 0$

(6) $S(v) * u = (v * u) + u$

The definition of < is also recursive and given as:

(7) $0 < y$, if $y \neq 0$

(8) $S(v) < y$, if $v < y$ & $y \neq S(v)$

Since the Otter-Lambda system fails, it requests some help from its human user. In [2], the user is able to provide the following lemmas that enable Otter-Lambda to complete the proof of (*).

(9) not(u<v) or (x*u < x*v) or not(0 < x)

(10) (x < y) or (y ≤ x)

(11) not(y ≤ x) or not(x < y)

(12) not(u < v) or not (v ≤ w) or (u < v)

(13) not(S(0) < z) or not(0 < y) or (S(y) ≤ z*y)

(14) 0 + x = x

We applied our Cartesian approach to the same problem, which does not suggest to get any user's help. The system determines n as the induction variable, since it occurs in recursive arguments of all the functions and predicates and the other possible candidate variable a occurs in the non-recursive first argument of the function exp which would stop the evaluation process in an inductive proof. Nevertheless, our system notices at once a probable source of trouble: the predicate < is recursively defined on its first argument, while, in (*), the induction variable n occurs also in second position of the predicate <. At this stage, the system could suggest the user to provide a definition of < with respect to both argument (this would actually fail), or to the second argument (this would fail as well), or else, a non recursive definition (that would succeed). As already claimed, our system does not call on its user, and it will proceed by calling a custom-designed constructor module we named "Synthesis of Formal Specifications of Predicates" described by Franova and Popelinsky [17]. The symbiotic

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system *CMM* with this constructor module included generates the following formal specification for predicate <:

(15) $x < y \Leftrightarrow \{ \exists z\ y = S(x + z) \}$.

With this new definition (*) is transformed into

$$\forall a\ \forall n\ \exists z \{ S(0) < a \Rightarrow \exp(a,n) = S(n + z) \}. \quad (**)$$

Note that this last formula is a specification formula by introducing the existentially quantified variable z. *CMM* is then able to prove it (without interaction with the user). *CMM* generates and proves autonomously the following lemmas:

L1. $\forall a\ \forall n1\ \forall b\ \exists z1 \{ S(0) < a \Rightarrow (n1 + b)*a + a = SS(n1 + z1) \}$.

L2. $\forall a\ \forall b\ \exists z2 \{ S(0) < a \Rightarrow b*a + a = SS(z2) \}$.

L3. $\forall a\ \exists z7 \{ S(0) < a \Rightarrow a = SS(z7) \}$.

L4. $\forall a\ \forall m\ \forall d\ \exists z5 \{ S(0) < a \Rightarrow (m + d) + a = S(m + z5) \}$.

L5. $\forall a\ \forall d\ \exists z3 \{ S(0) < a \Rightarrow d + a = S(z3) \}$.

L6. $\forall a\ \exists z4 \{ S(0) < a \Rightarrow a = S(z4) \}$.

This example illustrates all three points (a), (b), (c) of Cartesian Intuitionism in that, when meeting failure, a need for a complementary constructor transforming a recursive definition of a predicate into a non-recursive equivalent is informally specified. Then, the successful formalized design of this constructor enlarges the power of *CMM* and thus modifies the whole *CMM* which is ready, when necessary, to be once again modified.

The basic constructor of *CMM* is presented in [16] and the other constructors of *CMM* specified so far are described in our publications up to 2001. Some of these constructors were implemented in the system *Proofs Educed by Constructive Matching for Synthesis* (PRECOMAS) [14].

VI. A FEW EPISTEMOLOGICAL REMARKS

Accepting to use Cartesian Intuitionism as a way of creation of some complex systems (we exemplified here a Program Synthesis system) requires a deep transformation of our attitude together with an inevitable shift in thinking, because of changes, due to the new context, in vocabulary meaning, resonances and connotations. Newtonian theories and systems provide a kind of comfortable environment by the identified boundaries existing between each component of their architecture. Therefore, it is true that losing this comfort by accessing the new context we define here requires from the scientists a large change in their behaviour. In this, a Cartesian system requires from researchers the acceptance of open-ended research with its conceptual switches and a new propensity to deal with completeness and incompleteness. In a sense, such an open-ended 'technological' approach seems to be a natural answer to the open-ended theory of natural numbers and the open-ended 'bunch' of desires expressed as program synthesis problems.

Until now, the main technique used in the direction of such an opening to intuition has been carried out by the brainstorming techniques, in which several subjects relax enough to build unexpected mind connexions that might bring a new idea to the fore. In a sense, brainstorming could be an ideal way to define as precisely as possible what is the starting, informal, specification of the problem. The following steps of our proposal are still based on something similar to brainstorming, but the mind of each subject has to focus on ideas explicitly related to the informal specification of the problem. Ideas to find a path from informal to formal specification, then to solution, are triggered by each new problem arising at each failure to succeed in proving a step towards solution. In that sense, the collaboration between the members of a team working on the problem at hand, is enriched and much more focused by this problem than it is during a brainstorming session.

VII. CONCLUSION

Any design of a new complex system obviously requires, during its creative process**,** that its authors might be able to generate new ideas. In the field of program synthesis, our approach can be looked upon as a 'generator of new ideas'. We thus somewhat try to contradict Karl Popper who claims in [25] that "there is no such a thing as a logical method of having new ideas, or a logical reconstruction of this process." Our opinion is that Popper restricts here logical thinking to the linear one and his claim is perhaps valid in such a framework. On the contrary, our experience shows that Cartesian Intuitionism with its recursive features provides a method for having new ideas (and ones that are 'useful-for-solving-the-problem-at-hand)' as well as a model for a reconstruction of creative process, as we illustrated it in the study of creation of the Peano's axioms and its application to the design of an autonomous PS system.

By this paper, we have progressed in the direction of an adequate formalization of the first fundamental challenge met, as pointed out in [16], in the oscillatory design of the recursive system, namely, the challenge of understanding the symbiotic interrelation between a recursive whole, like N or *CMM*, and its parts (constructors) like S from N or "Synthesis of Formal Specifications of Predicates" from *CMM*. Understanding this first challenge will help to accelerate our future work on the three remaining problems described in [16], namely the 'chameleon' like behaviour of Cartesian systems, which are simultaneously static/dynamic, finite/infinite and complete/incomplete.

The ideas explained in the present paper are an illustration of our methodology that we plan to enlarge to problem-solving in general, not only to program synthesis.

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An Electricity Flow Optimization Problem for Effective Operation of Storage Devices in Microgrid

Hidenobu Aoki and Norihiko Shinomiya
Dept. of Information Systems Science
Soka University
Tokyo, Japan
shinomi@ieee.org

***Abstract*—An electrical energy storage device has been expected to alleviate negative effects of the integration of renewable energy resources into electricity supply systems; however, the appropriate storage technology along with the control characteristics as a component in the electricity supply system has not been clarified yet. As one of approaches, an effective control of electricity flow and an operation of storage devices have been developing. This paper formulates an electricity flow optimization problem in reducing transmission loss to analyze the fluctuation of it followed by the adjustment of the amount of electricity stored in storage devices.**

***Keywords*—electrical energy storage device; microgrid; renewable energy resource; transmission loss; transportation problem.**

I. Introduction

Smart Grid, a concept of incorporating ICT (Information and Communication Technology) into electricity supply system, is being developed from various perspectives, such as enhancing the reliability of electricity supply system and realizing low carbon society with renewable energy resources. On the other hand, to reduce the amount of transmission loss, physical loss of electrical energy while it is transmitted though transmission line is also main objective of Smart Grid technology [1].

In a traditional power grid, the large amount of electricity can be lost since electricity is supplied from a few large-scale power plants located in rural areas to consumer sides. In contrast, by exploiting renewable energy resources as distributed energy resources and controlling them effectively with the concept of Smart Grid, the reduction of transmission loss has been expected [2].

However, the generation patterns of those renewable energy resources fluctuate irregularly due to weather conditions and it may result in the fluctuation of frequency and voltage in a power grid. As a result, the degradation of the stability in power grid might be caused when a large amount of them are introduced in electricity supply system [1][2][3].

To deal with the drawback of the renewable energy resource, management strategies of them are studied in many research works [3]; Microgird (MG), forming a small-scale power grid composed of local electricity consumers and distributed energy resources has been proposed to realize a self-sufficient supply system in local areas [2]. In [4], it is stated that MG with its locality can make energy management of renewable energy resources improved as well as reducing transmission loss. In addition, as a way to mitigate negative effects of the integration of renewable energy resources, utilizing grid-integrated electrical energy storage devices has been considered as well [5][6]. This is because it can be used to stabilize the short-time frequency and voltage fluctuation, manage peak loads, and improve the power quality [7][8].

A. Related Work of Storage Device

A lot of research related to the use of the storage device in electricity supply systems have been developing in order to improve the efficiency and reliability of the storage technology [9]. Moreover, although there are various types and sizes of storage devices for different purposes, a large-scale storage device installed beside renewable energy resources has been developing. This is because if renewable energy resources and storage devices are installed in geographically separated places, transmission loss may arise to send surplus electricity from renewable energy resources to storage devices to store it. Furthermore, storage devices have been also expected as a backup source of electricity for an emergency, such as when natural disaster occurs from the perspective of the stability and durability of the electricity supply system [10].

Many of the current research related to storage technology mainly focus on the power balance for stable and reliable electricity supply [6][11]; however, a full potential of the storage device and an appropriate storage technology along with the control characteristics as a component of the electricity supply system have not been clarified yet and remained as a research task [11] [12]. As one of approaches to analyze the control characteristics, an effective operations of the storage device by optimizing electricity flow has been studied [9][13].

B. Research Objective

This paper assumes the situation that storage devices are installed beside renewable energy resources which generate surplus electricity in MG. Then, this paper tries to smooth the amount of electricity stored in storage devices by adjusting the amount of electricity supplied from renewable energy resources. This is to distribute the amount of surplus electricity as backup sources for the realization of a durable electricity supply system for an emergency.

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In this situation, the amount of electricity generated in a renewable energy resource can be divided into two types of use, the amount of electricity supplied from a renewable energy resource and the amount of electricity stored in a storage device; that is, there is a dependency between the amount of electricity supplied from the renewable energy resource and the amount of electricity stored in the storage device. Moreover, transmission loss depends on how much and how far electricity is transmitted. As a result, it might be indicated that smoothing the amount of electricity stored in storage devices may affect the total transmission loss. Therefore, this paper analyzes how transmission loss is influenced by various patterns of the scatter of the amount of electricity stored in each storage device.

The rest of the paper is organized as follows: In Section I I, the definition of MG graph, such as types of nodes and the amount of electricity, and problem formulation are mentioned. A simulation procedure and result are presented in section Ⅱ I. Section Ⅳ concludes the paper with future work.

II. Model Definition and Problem Formulation

In this section, firstly, the model of MG graph is defined. Then, an electricity flow optimization problem is formulated.

A. Definition of Microgrid Graph

This paper assumes MG graph composed of different types of node and edge. First of all, let a node G be Central Generation Facility (CGF) node utilizing the chemical energy stored in fossil fuels such as coal, fuel oil, natural gas which is converted into electrical energy. In reality, CGF will be assigned to each MG as a supplemental power resource for renewable energy resources;accordingly, this paper considers as though a single G node is included as a component of MG.

Secondly, for renewable energy resources, although there are different kinds of them, such as solar or wind power generation in reality, this paper does not consider those differences but assume them just as a renewable energy resource. Furthermore, this paper supposes an installation of the storage device beside renewable energy resources mentioned in Section I-B. Thus, let a node representing renewable energy resources contain the function of the storage device and be defined as renewable energy resource with storage device node, RS_i.

Finally, a node for electricity demand, such as houses, factories, hospitals and so on is defined as D_j. This paper assumes that the number of electricity demand nodes in MG graph, m, is greater than that of RS nodes, n.

Those three kinds of nodes are connected in the following manners: Every electricity demand node is connected to G so that they can receive electricity from the stable power resource. Furthermore, each electricity demand node is connected to all RS nodes in MG graph to simplify the problem discussed in the following subsections.

B. Problem Formulation

MG graph describing node types and electricity flows is shown in Figure 1. Let d_j be the amount of electricity demand

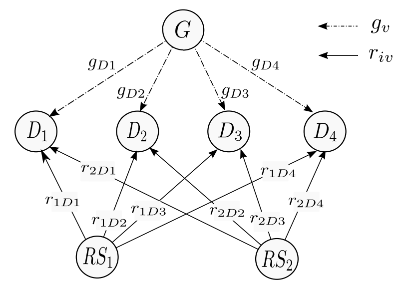

Fig. 1. A structure of MG graph.

in node D_j, and the amount of electricity derived from G node to D_j is g_j. The amount of electricity generated in RS_i is r_i, and the amount of electricity supplied from RS_i to D_j is r_{ij}. The surplus electricity generated in renewable energy resources can be stored in a storage device. Hence, let the amount of surplus electricity stored in RS_i be s_i which can be described as the subtraction of the total amount of r_{ij} for $j \in D$ from the amount of r_i as in (1). The sum of r_{ij} for $j \in D$ is also defined as r'_i as in (2). Finally, let a rate of transmission loss along with electricity transmission between RS nodes and D nodes be defined as $loss_{ij}$.

$$s_i = r_i - \sum_{j=1}^{m} r_{ij} \tag{1}$$

$$r'_i = \sum_{j=1}^{m} r_{ij} \tag{2}$$

There are several constraints to match demand and supply. Firstly, the amount of electricity demand in each D node has to be fully met with the total amount of electricity supplied from G node and RS nodes stated in (3).

$$d_j = g_j + \sum_{i=1}^{n} r_{ij} \tag{3}$$

This paper focuses on optimizing electricity flows from RS nodes to D nodes. Thus, let d'_j be the amount of the electricity subtracted g_j from d_j, denoting the amount of electricity that RS nodes need to supply to D_j. In order to match demand and supply in MG, the condition in (4) must be satisfied.

$$\sum_{j=1}^{m} d'_j = \sum_{i=1}^{n} r'_i \tag{4}$$

In this paper, for randomly given d_j and r_i, the value of s_i is set at first, and r'_i is determined accordingly. Afterward, r_{ij} is optimized.

1) Transportation Problem: To decide r_{ij}, this paper applies Transportation Problem (TP). TP is based on supply and demand of commodities transported from several sources to different destinations [14]. In general, TP tries to minimize total transportation cost for the commodities transported from source to destinations. Applying TP for an electricity flow

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optimization, this paper considers to minimize total transmission loss from RS to D nodes. The objective function of the electricity flow optimization can be formulated as in (5).

$$\begin{cases} \text{Minimize} \ \sum_{i=1}^{n}\sum_{j=1}^{m} r_{ij} loss_{ij} \\ \text{Subject to} \ \sum_{j=1}^{m} r_{ij} = r'_i, \quad for \ \ i = 1, 2, ..., n \\ \qquad\qquad\quad \sum_{i=1}^{n} r_{ij} = d'_j, \quad for \ \ j = 1, 2, ..., m \\ r_{ij} \geq 0 \quad for \ \ i = 1, 2, ..., n \ \ and \ \ j = 1, 2, ..., m \end{cases} \tag{5}$$

2) Method to Solve Transportation Problem: There are mainly two methods to obtain the solution for TP, Simplex method and Transportation method. This paper applies Transportation method since it is more efficient one that yields results faster and with less computational effort [14]. Transportation method consists of the following three steps.

1) Obtaining an initial feasible solution.
2) Testing Optimality.
3) Revising the solution until an optimal solution is obtained.

For conducting those three steps, in general, TP can be portrayed in a tabular form by means of a transportation table shown in (2). In a transportation table, the number of row represents that of RS node, n, and the number of column denotes that of D node, m.

$RS\ node(i)$	$D\ node(j)$				$Supply(r'_i)$
	1	2	$\cdots$	m	
1	r_{11} $loss_{11}$	r_{12} $loss_{12}$	$\cdots$	r_{1m} $loss_{1m}$	r'_1
2	r_{21} $loss_{21}$	r_{22} $loss_{22}$	$\cdots$	r_{2m} $loss_{2m}$	r'_2
$\cdots$	$\cdots$	$\cdots$	$\cdots$	$\cdots$	$\cdots$
n	r_{n1} $loss_{n1}$	r_{n2} $loss_{n2}$	$\cdots$	r_{nm} $loss_{nm}$	r'_n
$Demand(d'_j)$	d'_1	d'_2	$\cdots$	d'_m	$\sum d'_j = \sum r'_i$

Fig. 2. A transportation table.

3) North-West Corner Method: The first step is to obtain an initial feasible solution which satisfies the requirement of demand and supply. Although an initial feasible solution can be obtained by several methods, this paper applies North-West Corner method (NWC), a method to compute an initial feasible solution, which begins selecting a basic variable in the upper left-hand corner of the transportation table. The algorithm of NWC is described in Algorithm 1.

4) Modified Distribution Method: Once an initial solution is obtained by NWC, the next step is to check its optimality. Although there are several methods to find an optimal solution of TP, such as Vogel's method or Stepping Stone method,

Algorithm 1 North-West Corner Method (NWC)

Require: $\sum r'_i = \sum d'_j$
  $Start\ with\ the\ cell\ at\ the\ upper\ left\text{-}hand\ corner$
  $i, j \leftarrow 0$
  while $i < n$ **do**
    if $r_{ij} > d'_j$ **then**
      $r_{ij} \leftarrow r_{ij} - d'_j$
    else if $r_{ij} < d'_j$ **then**
      $d'_j \leftarrow d'_j - r_{ij}$
      $i \leftarrow i + 1$
    else
      $i \leftarrow i + 1$
      $j \leftarrow j + 1$
    end if
  end while
  $Calculate\ initial\ transmission\ loss$

Modified Distribution method (MODI) has been used as a standard technique for obtaining an optimal solution [14]. This paper applies MODI for obtaining optimal solution, and the algorithm of MODI is described in Algorithm 2.

Algorithm 2 Modified Distribution Method (MODI)

Require: $An\ initial\ feasible\ solution\ by\ NWC$
  while $all\ p_{ij}\ are\ not\ a\ positive\ value$ **do**
    $Determine\ the\ values\ of\ dual\ variables,\ u_i\ and\ v_j$
    for $each\ unassigned\ cell$ **do**
      $p_{ij} \leftarrow \ c_{ij} - (u_i + v_j)$
    end for
    if $all\ p_{ij} \geq 0$ **then**
      $Optimization\ is\ completed$
      $Calculate\ optimized\ transmission\ loss$
    else
      if $p_{ij} < 0\ and\ min\,(p_{ij})$ **then**
        $Draw\ a\ closed\ path\ starting\ from\ r_{ij}$
        $Assign$ + or - $sign\ on\ the\ cells\ in\ it\ alternately$
        $Find\ min\,(r_{ij})\ in\ cells\ with\ -sign$
      end if
      for $each\ cells\ in\ closed\ path$ **do**
        if $sign\ is\ +$ **then**
          $r_{ij} \leftarrow r_{ij} + \ min\,(r_{ij})$
        else
          $r_{ij} \leftarrow r_{ij} - \ min\,(r_{ij})$
        end if
      end for
    end if
  end while

III. Simulation and Result

In this section, the simulation procedure and result are presented.

A. Simulation Procedure

In this simulation, MG graph is created with 4 RS nodes, 8 D nodes, and a single G node. The sum of r_i and d_j are set as 1200 and 1000. Then, assigning a random value in each r_i for $i = 1$ to 4 and d_j for $j = 1$ to 8 such that the sum of r_i is equal to 1200, that of d_j, 1000, respectively. Note that it is assumed that about 80% of each d_j is supplied by G node in this simulation. As a result, d'_j is set as the subtraction of g_j from d_j. The total amount of surplus electricity is determined by the difference of the sum of r_i and that of d_j, which is 200, Based on this value, the value of s_i is given. This simulation creates 2000 patterns of the set of s_i for $i = 1$ to 4 such that the sum of s_i is 200. A pattern of s_i is, for instance, s_1=37.54045, s_2=62.13593, s_3=45.95469, s_4=54.36893. Once the value of s_i is given, set the value of r'_i, subtracting s_i from r_i. Then, decide r_{ij} such that transmission loss is minimized by means of NWC and MODI method discussed in Section II-C. Note that $loss_{ij}$ is randomly assigned to each edge between RS and D nodes with the value of the second decimal place in the range shown in (6). This is because about 5% of electricity would be lost during transmisison to consumer sides in general although it depens on the distance it is transmitted.

$$0 < loss_{ij} \leq 0.05 \qquad (6)$$

B. Simulation Result

A correlation between scattering of s_i and transmission loss in Figure 3. In Figure 3, the horizontal axis describes the standard deviation of the pattern of s_i, and the vertical axis represents the transmission loss optimized by MODI.

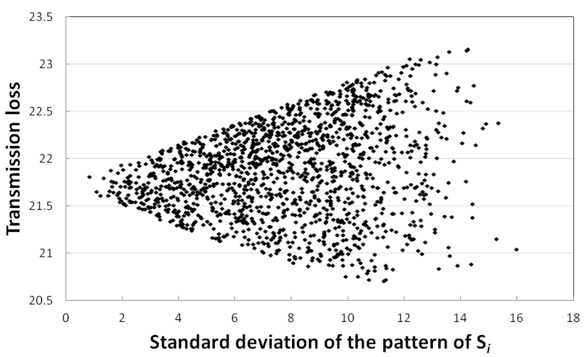

Fig. 3. Correlation between scattering of s_i and transmission loss.

As seen in Figure 3, when the value of standard deviation of the pattern of s_i is close to 0, the transmission loss gets the value close to the average and does not get the worst value. In addition, the difference of the best and worst value of the transmission loss is smaller as a standard deviation of the pattern of s_i is close to 0.

This result may indicate that even if the amount of electricity stored in several storage devices is smoothed by adjusting electricity flows, the transmission loss would almost take the average value although it is not the best improved value.

IV. Conclusion And Future Work

In summary, this paper formulates an electricity flow optimization problem, presents its solution, and analyzes the fluctuation of transmission loss followed by the adjustment of the amount of electricity stored in storage devices.

The simulation result implies that when the amount of electricity stored in storage device is smoothed by adjusting the electricity flows, the value of the transmission loss almost gets the mean value compared to other patterns. However, this result comes out under a certain situation given in this paper, and other resulst might be obtained in different situations. To examine it, therefore, further simulation in various situations, such as different node numbers and their values need to be conducted. Moreover, since this paper does not cover all aspects of prospective electricity supply system, there are remained issues to make this research more practical. For example, this paper assumes that each D node is connected to all RS nodes in MG for the simplification of the electricity flow optimization problem. Hence, the paper needs to reorganize the structure of the MG which has less transmission lines by considering the effective connections between D nodes and RS nodes.

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Construction Principles for Well-behaved Scalable Systems

Peter Ochsenschläger* and Roland Rieke*†
*Fraunhofer Institute for Secure Information Technology SIT, Darmstadt, Germany
†Philipps-Universität Marburg, Germany
Email: peter-ochsenschlaeger@t-online.de, roland.rieke@sit.fraunhofer.de

***Abstract*—We formally define *scalable systems* as *uniformly monotonic parameterised systems* and motivate this definition. With respect to such scalable systems, we focus on properties, which rely on specific component types and a specific number of individual components for these component types but not on the specific individuality of the individual components. We characterise *well-behaved* scalable systems by those systems which fulfil such a kind of property if already one prototype system (depending on the property) fulfils that property. Self-similar uniformly monotonic parameterised systems have the above desired property. Therefore, we define well-behaved scalable systems as self-similar scalable systems. This paper presents a formal framework that provides construction principles for well-behaved scalable systems. It gives sufficient conditions to specify a certain kind of basic well-behaved scalable systems and shows how to construct more complex systems by the composition of several synchronisation conditions.**

Keywords-uniformly parameterised systems; monotonic parameterised systems; behaviour-abstraction; self-similarity of behaviour.

I. Introduction

Scalability is a desirable property of a system. In [1], four aspects of scalability are considered, i.e., load scalability, space scalability, space-time scalability, and structural scalability. In our paper, we focus on *structural scalability*, which is "the ability of a system to expand in a chosen dimension without major modifications to its architecture" [1]. Examples of systems that need to be highly scalable comprise grid computing architectures and cloud computing platforms [2], [3]. Usually, such systems consist of few different types of components and for each such type a varying set of individual components exists. Component types can be defined in such a granularity that individual components of the same type behave in the same manner, which is characteristic for the type. For example, a client-server system that is scalable consists of the component types *client* and *server* and several sets of individual clients as well as several sets of individual servers. Let us now call a choice of sets of individual components an *admissible choice of individual component sets*, iff for each component type exactly one set of individual components of that type is chosen. Then, a "scalable system" can be considered as a family of systems, whose elements are systems composed of a specific admissible choice of individual component sets.

In this paper, we focus on the dynamic behaviour of systems, which is described by the set of all possible sequences of actions. This point of view is important to define security requirements as well as to verify such properties, because for these purposes sequences of actions of the system have to be considered [4], [5], [6]. For short, we often will use the term *system* instead of *systems behaviour* if it does not generate confusions. With this focus, scalable systems are families of system behaviours, which are indexed by admissible choices of individual component sets. We call such families *parameterised systems*. In this paper, we define *well-behaved scalable systems* as a special class of parameterised systems and develop construction principles for such systems. The main goal for this definition is to achieve that well-behaved scalable systems fulfill certain kind of safety properties if already one prototype system (depending on the property) fulfills that property (cf. Section III). To this end, construction principles for well-behaved scalable systems are *design principles for verifiability* [7].

Considering the behaviour-verification aspect, which is one of our motivations to formally define well-behaved scalable systems, there are some other approaches to be mentioned. An extension to the Murφ verifier to verify systems with replicated identical components through a new data type called RepetitiveID is presented in [8]. A typical application area of this tool are cache coherence protocols. The aim of [9] is an abstraction method through symmetry, which works also when using variables holding references to other processes. In [10], a methodology for constructing abstractions and refining them by analysing counter-examples is presented. The method combines abstraction, model-checking and deductive verification. A technique for automatic verification of parameterised systems based on process algebra CCS [12] and the logic modal mu-calculus [13] is presented in [11]. This technique views processes as property transformers and is based on computing the limit of a sequence of mu-calculus [13] formulas generated by these transformers. The above-mentioned approaches demonstrate that finite state methods combined with deductive methods can be applied to analyse parameterised

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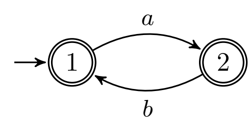
systems. The approaches differ in varying amounts of user intervention and their range of application. A survey of approaches to combine model checking and theorem proving methods is given in [14]. Far reaching results in verifying parameterised systems by model checking of corresponding abstract systems are given in [15], [16]. It is well known that the general verification problem for parameterised systems is undecidable [17], [18]. To handle that problem, we present (a) a formal framework to specify parameterised systems in a restricted manner, and (b) construction principles for well-behaved scalable systems.

In Section II, scalable systems are formally defined. Section III and Section IV give sufficient conditions to specify a certain kind of basic well-behaved scalable systems. Section V shows how to construct more complex well-behaved scalable systems by the composition of several synchronisation conditions. Concluding remarks and further research directions are given in Section VI. The proofs of the theorems in this paper are given in [19].

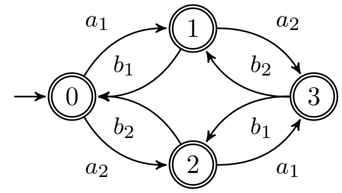
II. Characterisation of Scalable Systems

The behaviour L of a discrete system can be formally described by the set of its possible sequences of actions. Therefore, $L \subset \Sigma^*$ holds where Σ is the set of all actions of the system, and Σ^* (free monoid over Σ) is the set of all finite sequences of elements of Σ, including the empty sequence denoted by ε. This terminology originates from the theory of formal languages [20], where Σ is called the alphabet (not necessarily finite), the elements of Σ are called letters, the elements of Σ^* are referred to as words and the subsets of Σ^* as formal languages. Words can be composed: if u and v are words, then uv is also a word. This operation is called the *concatenation*; especially $\varepsilon u = u\varepsilon = u$. A word u is called a *prefix* of a word v if there is a word x such that $v = ux$. The set of all prefixes of a word u is denoted by $\mathrm{pre}(u)$; $\varepsilon \in \mathrm{pre}(u)$ holds for every word u. Formal languages which describe system behaviour have the characteristic that $\mathrm{pre}(u) \subset L$ holds for every word $u \in L$. Such languages are called *prefix closed*. System behaviour is thus described by prefix closed formal languages. Different formal models of the same system are partially ordered with respect to different levels of abstraction. Formally, abstractions are described by alphabetic language homomorphisms. These are mappings $h^* : \Sigma^* \longrightarrow \Sigma'^*$ with $h^*(xy) = h^*(x)h^*(y)$, $h^*(\varepsilon) = \varepsilon$ and $h^*(\Sigma) \subset \Sigma' \cup \{\varepsilon\}$. So, they are uniquely defined by corresponding mappings $h : \Sigma \longrightarrow \Sigma' \cup \{\varepsilon\}$. In the following, we denote both the mapping h and the homomorphism h^* by h. We consider a lot of alphabetic language homomorphisms. So, for simplicity we tacitly assume that a mapping between free monoids is an alphabetic language homomorphism if nothing contrary is stated. We now introduce a guiding example.

Example 1. *A server answers requests of a family of clients. The actions of the server are considered in the following. We assume with respect to each client that a request will be answered before a new request from this client is accepted. If the family of clients consists of only one client, then the automaton in Fig. 1(a) describes the system behaviour $S \subset \Sigma^*$, where $\Sigma = \{a, b\}$, the label a depicts the request, and b depicts the response.*



(a) Actions at a server with respect to a client



(b) Two clients served concurrently by one server

Figure 1. Scalable client-server system

Example 2. *Fig. 1(b) now describes the system behaviour $S_{\{1,2\}} \subset \Sigma^*_{\{1,2\}}$ for two clients 1 and 2, under the assumption that the server handles the requests of different clients non-restricted concurrently.*

For a parameter set I and $i \in I$ let $\Sigma_{\{i\}}$ denote pairwise disjoint copies of Σ. The elements of $\Sigma_{\{i\}}$ are denoted by a_i and $\Sigma_I := \bigcup_{i \in I} \Sigma_{\{i\}}$, where $\Sigma_j \cap \Sigma_k = \emptyset$ for $j \neq k$. The index i describes the bijection $a \leftrightarrow a_i$ for $a \in \Sigma$ and $a_i \in \Sigma_{\{i\}}$.

Example 3. *For $\emptyset \neq I \subset \mathbb{N}$ with finite I, let now $S_I \subset \Sigma_I^*$ denote the system behaviour with respect to the client set I. For each $i \in \mathbb{N}$ $S_{\{i\}}$ is isomorphic to S, and S_I consists of the non-restricted concurrent run of all $S_{\{i\}}$ with $i \in I$. It holds $S_{I'} \subset S_I$ for $I' \subset I$.*

Let $\mathcal{I}_1$ denote the set of all finite non-empty subsets of $\mathbb{N}$ (the set of all possible clients). Then, the family $(S_I)_{I \in \mathcal{I}_1}$ is an example of a monotonic parameterised system.

If the example is extended to consider several servers, which are depicted by natural numbers, then, e.g.,

$$\mathcal{I}_2 := \{\mathring{I} \times \hat{I} \subset \mathbb{N} \times \mathbb{N} | \mathring{I} \neq \emptyset \neq \hat{I}, \text{ with } \mathring{I}, \hat{I} \text{ finite}\}$$

is a suitable parameter structure.

$\mathcal{I}_2$ used in the example above shows how the component structure of a system can be expressed by a parameter structure using Cartesian products of individual component sets. The following Definition 1 abstracts from the intuition of a component structure.

Definition 1 (parameter structure)**.** *Let N be a countable (infinite) set and $\emptyset \neq \mathcal{I} \subset \mathfrak{P}(N) \setminus \{\emptyset\}$. $\mathcal{I}$ is called a* parameter structure *based on N.*

For scalable systems it is obvious to assume that enlarging the individual component sets does not reduce the corresponding system behaviour. More precisely: let I and K be two arbitrary admissible choices of individual component sets, where each individual component set in I is a subset of the corresponding individual component set in K. If S_I and S_K are the corresponding systems' behaviours, then S_I is a subset of S_K. Families of systems with this property we call *monotonic parameterised systems*. The following definition formalises monotonic parameterised systems.

Definition 2 (monotonic parameterised system)**.** *Let $\mathcal{I}$ be a* parameter structure*. For each $I \in \mathcal{I}$ let $\mathcal{L}_I \subset \Sigma_I^*$ be a prefix closed language. If $\mathcal{L}_{I'} \subset \mathcal{L}_I$ for each $I, I' \in \mathcal{I}$ with $I' \subset I$, then $(\mathcal{L}_I)_{I \in \mathcal{I}}$ is a* monotonic parameterised system*.*

As we assume that individual components of the same type behave in the same manner, S_I and S_K are isomorphic (equal up to the names of the individual components), if I and K have the same cardinality. This property we call *uniform parameterisation*. With these notions we define *scalable systems* as *uniformly monotonic parameterised systems*. Monotonic parameterised systems in which isomorphic subsets of parameter values describe isomorphic subsystems we call *uniformly monotonic parameterised systems*.

Definition 3 (isomorphism structure)**.** *Let $\mathcal{I}$ be a parameter structure, $I, K \in \mathcal{I}$, and $\iota : I \to K$ a bijection, then let $\iota_K^I : \Sigma_I^* \to \Sigma_K^*$ the isomorphism defined by*

$$\iota_K^I(a_i) := a_{\iota(i)} \text{ for } a_i \in \Sigma_I. \tag{1}$$

For each $I, K \in \mathcal{I}$ let $\mathcal{B}(I,K) \subset K^I$ a set (possibly empty) of bijections. $\mathcal{B}_{\mathcal{I}} = (\mathcal{B}(I,K))_{(I,K) \in \mathcal{I} \times \mathcal{I}}$ is called an isomorphism structure *for $\mathcal{I}$.*

Definition 4 (scalable system)**.** *Let $(\mathcal{L}_I)_{I \in \mathcal{I}}$ a monotonic parameterised system and $\mathcal{B}_{\mathcal{I}} = (\mathcal{B}(I,K))_{(I,K) \in \mathcal{I} \times \mathcal{I}}$ an* isomorphism structure *for $\mathcal{I}$.*
$(\mathcal{L}_I)_{I \in \mathcal{I}}$ is called uniformly monotonic parameterised with respect to *$\mathcal{B}_{\mathcal{I}}$ iff*

$$\mathcal{L}_K = \iota_K^I(\mathcal{L}_I) \text{ for each } I, K \in \mathcal{I} \text{ and each } \iota \in \mathcal{B}(I,K).$$

Uniformly monotonic parameterised systems for short are called scalable systems*.*

Example 4. *Let $\mathcal{I} = \mathcal{I}_2$.*

$$\begin{aligned} \mathcal{B}^2(\mathring{I} \times \hat{I}, \mathring{K} \times \hat{K}) := \{\iota \in (\mathring{K} \times \hat{K})^{(\mathring{I} \times \hat{I})} \mid & \text{it exist bijections} \\ & \mathring{\iota} : \mathring{I} \to \mathring{K} \text{ and } \hat{\iota} : \hat{I} \to \hat{K} \text{ with } \iota((r,s)) = (\mathring{\iota}(r), \hat{\iota}(s)) \\ & \text{for each } (r,s) \in (\mathring{I} \times \hat{I})\} \end{aligned}$$

for $\mathring{I} \times \hat{I} \in \mathcal{I}_2$ and $\mathring{K} \times \hat{K} \in \mathcal{I}_2$ defines an isomorphism structure *$\mathcal{B}^2_{\mathcal{I}_2}$.*

III. Well-behaved Scalable Systems

To motivate our formalisation of *well-behaved*, we consider a typical security requirement of a scalable client-server system: Whenever two different clients cooperate with the same server then certain critical sections of the cooperation of one client with the server must not overlap with critical sections of the cooperation of the other client with the same server. If for example both clients want to use the same resource of the server for confidential purposes, then the allocation of the resource to one of the clients has to be completely separated from the allocation of this resource to the other client. More generally, the concurrent cooperation of one server with several clients has to be restricted by certain *synchronisation conditions* to prevent, for example, undesired race conditions.

According to this example, we focus on properties which rely on specific component types and a specific number of individual components for these component types but not on the specific individuality of the individual components. Now, we want to achieve that a well behaved scalable system fulfils such a kind of property if already one *prototype system* (depending on the property) fulfils that property. In our example, a prototype system consists of two specific clients and one specific server.

To formalise this desire, we consider arbitrary I and K as in the definition of monotonic parameterised system. Then we look at S_K from an abstracting point of view, where only actions corresponding to the individual components of I are considered. If the smaller subsystem S_I behaves like the abstracted view of S_K, then we call this property *self-similarity* or more precisely *self-similarity of scalable systems*, to distinguish our notion from geometric oriented notions [21] and organisational aspects [22] of self-similarity. In [5], it is shown that *self-similar uniformly monotonic parameterised systems* have the above desired property. Therefore, we define *well-behaved scalable systems* as self-similar uniformly monotonic parameterised systems. We now formally look at $\mathcal{L}_I$ from an abstracting point of view concerning a subset $I' \subset I$. The corresponding abstractions are formalised by the homomorphisms $\Pi_{I'}^I : \Sigma_I^* \to \Sigma_{I'}^*$.

Definition 5 (self-similar monotonic parameterised system)**.** *For $I' \subset I$ let $\Pi_{I'}^I : \Sigma_I^* \to \Sigma_{I'}^*$ with*

$$\Pi_{I'}^I(a_i) = \begin{cases} a_i \mid & a_i \in \Sigma_{I'} \\ \varepsilon \mid & a_i \in \Sigma_I \setminus \Sigma_{I'}. \end{cases}$$

A monotonic parameterised system $(\mathcal{L}_I)_{I \in \mathcal{I}}$ is called self-similar *iff $\Pi_{I'}^I(\mathcal{L}_I) = \mathcal{L}_{I'}$ for each $I, I' \in \mathcal{I}$ with $I' \subset I$.*

Definition 6 (well-behaved scalable system)**.** *Self-similar scalable systems for short are called* well-behaved scalable systems*.*

A fundamental construction principle for systems satisfying several constraints is intersection of system behaviours. This emphasises the importance of the following theorem.

Theorem 1 (intersection theorem). *Let $\mathcal{I}$ be a parameter structure, $\mathcal{B}_\mathcal{I}$ an isomorphism structure for $\mathcal{I}$, and $T \neq \emptyset$.*

i) *Let $(\mathcal{L}_I^t)_{I\in\mathcal{I}}$ for each $t \in T$ be a monotonic parameterised system, then $(\bigcap_{t\in T} \mathcal{L}_I^t)_{I\in\mathcal{I}}$ is a monotonic parameterised system.*
ii) *Let $(\mathcal{L}_I^t)_{I\in\mathcal{I}}$ for each $t \in T$ be a scalable system with respect to $\mathcal{B}_\mathcal{I}$, then $(\bigcap_{t\in T} \mathcal{L}_I^t)_{I\in\mathcal{I}}$ is a scalable system with respect to $\mathcal{B}_\mathcal{I}$.*
iii) *Let $(\mathcal{L}_I^t)_{I\in\mathcal{I}}$ for each $t \in T$ be a self-similar monotonic parameterised system, then $(\bigcap_{t\in T} \mathcal{L}_I^t)_{I\in\mathcal{I}}$ is a self-similar monotonic parameterised system.*

Weak additional assumptions for well-behaved scalable systems imply that such systems are characterised by parametrisation of one well-defined minimal prototype system. More precisely:

Definition 7 (minimal prototype system). *Let $\mathcal{I}$ be a parameter structure based on N. For $I \in \mathcal{I}$ and $n \in N$ let $\tau_n^I : \Sigma_I^* \to \Sigma^*$ the homomorphisms given by*

$$\tau_n^I(a_i) = \begin{cases} a \mid & a_i \in \Sigma_{I\cap\{n\}} \\ \varepsilon \mid & a_i \in \Sigma_{I\setminus\{n\}} \end{cases}.$$

For a singleton index set $\{n\}$, $\tau_n^{\{n\}} : \Sigma_{\{n\}}^* \to \Sigma^*$ is an isomorphism and for each $n \in I \in \mathcal{I}$ holds

$$\Pi_{\{n\}}^I = (\tau_n^{\{n\}})^{-1} \circ \tau_n^I. \tag{2}$$

If now $(\mathcal{L}_I)_{I\in\mathcal{I}}$ is a well-behaved scalable system with respect to $(\mathcal{B}(I,K))_{(I,K)\in\mathcal{I}\times\mathcal{I}}$ with $\{n\} \in \mathcal{I}$ for $n \in I \in \mathcal{I}$ and $\mathcal{B}(I,K) \neq \emptyset$ for all singleton I and K, then because of (2) *holds*

$$\mathcal{L}_I \subset \bigcap_{n\in I} (\tau_n^I)^{-1}(L) \text{ for each } I \in \mathcal{I},$$

where $L = \tau_n^{\{n\}}(\mathcal{L}_{\{n\}})$ for each $n \in \bigcup_{I\in\mathcal{I}} I$.

L is called the minimal prototype system *of $(\mathcal{L}_I)_{I\in\mathcal{I}}$.*

Definition 8 (behaviour-family $(\dot{\mathcal{L}}(L)_I)_{I\in\mathcal{I}}$ generated by the minimal prototype system L and the parameter structure $\mathcal{I}$). *Let $\emptyset \neq L \subset \Sigma^*$ be prefix closed, $\mathcal{I}$ a parameter structure, and*

$$\dot{\mathcal{L}}(L)_I := \bigcap_{i\in I} (\tau_i^I)^{-1}(L) \text{ for } I \in \mathcal{I}.$$

The systems $\dot{\mathcal{L}}(L)_I$ consist of the "non-restricted concurrent run" of all systems $(\tau_i^{\{i\}})^{-1}(L) \subset \Sigma_{\{i\}}^*$ with $i \in I$. Because $\tau_i^{\{i\}} : \Sigma_{\{i\}}^* \to \Sigma^*$ are isomorphisms, $(\tau_i^{\{i\}})^{-1}(L)$ are pairwise disjoint copies of L.

Theorem 2 (simplest well-behaved scalable systems). *$(\dot{\mathcal{L}}(L)_I)_{I\in\mathcal{I}}$ is a well-behaved scalable system with respect to each isomorphism structure for $\mathcal{I}$ based on N and*

$$\dot{\mathcal{L}}(L)_I = \bigcap_{i\in N} (\tau_i^I)^{-1}(L) \text{ for each } I \in \mathcal{I}.$$

IV. Construction of Well-behaved Systems by Restriction of Concurrency

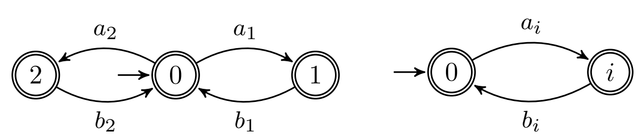
Now, we show how to construct well-behaved systems by restricting concurrency in the behaviour-family $\dot{\mathcal{L}}$. In Example 3, holds $S_I = \dot{\mathcal{L}}(S)_I$ for $I \in \mathcal{I}_1$. If, in the given example, the server needs specific resources for the processing of a request, then - on account of restricted resources - an non-restricted concurrent processing of requests is not possible. Thus, restrictions of concurrency in terms of synchronisation conditions are necessary. One possible but very strong restriction is the requirement that the server handles the requests of different clients in the same way as it handles the requests of a single client, namely, on the request follows the response and vice versa. This synchronisation condition can be formalised with the help of S and the homomorphisms Θ^I as shown in the following example.

Example 5. *Restriction of concurrency on account of restricted resources: one "task" after another. All behaviours with respect to $i \in I$* influence *each other. Let*

$$\bar{S}_I := S_I \cap (\Theta^I)^{-1}(S) = \bigcap_{i\in I} (\tau_i^I)^{-1}(S) \cap (\Theta^I)^{-1}(S)$$

for $I \in \mathcal{I}_1$, where generally, for each index set I, $\Theta^I : \Sigma_I^* \to \Sigma^*$ is defined by $\Theta^I(a_i) := a$, for $i \in I$ and $a \in \Sigma$.

From the automaton in Fig. 1(b), it is evident that $\bar{S}_{\{1,2\}}$ will be accepted by the automaton in Fig. 2(a).



(a) Automaton accepting $\bar{S}_{\{1,2\}}$ (b) Automaton accepting $\bar{S}_I$

Figure 2. Automata accepting $\bar{S}_{\{1,2\}}$ and $\bar{S}_I$

Given an arbitrary $I \in \mathcal{I}_1$, then $\bar{S}_I$ is accepted by an automaton with state set $\{0\} \cup I$ and state transition relation given by Fig. 2(b) for each $i \in I$.

From this automaton, it is evident that $(\bar{S}_I)_{I\in\mathcal{I}_1}$ is a well-behaved scalable system, with respect to each isomorphism structure $\mathcal{B}_{\mathcal{I}_1}$ for $\mathcal{I}_1$.

Example 6. *A restriction of concurrency in the extended example where a family of servers is involved is more complicated than in the case of $(\bar{S}_I)_{I\in\mathcal{I}_1}$. The reason for that is that in the simple example the restriction of*

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concurrency can be formalised by a restricting influence of the actions with respect to all parameter values (i.e., the entire Σ_I). When considering the restriction of concurrency in the extended example, the actions influence each other only with respect to the parameter values, which are bound to the same server.

Let the first component of the elements from $\mathbb{N} \times \mathbb{N}$ in the parameter structure $\mathcal{I}_2$ denote the server, then the actions from $\Sigma_{\{r\}\times\hat{I}}$ influence each other for given $r \in \mathring{I}$ with $\mathring{I} \times \hat{I} \in \mathcal{I}$ and thus restrict the concurrency.

For the formalisation of this restriction of concurrency, we now consider the general case of monotonic parameterised systems $(\dot{\mathcal{L}}(L)_I)_{I\in\mathcal{I}}$. As already observed in (2), for each well-behaved scalable system $(\mathcal{L}_I)_{I\in\mathcal{I}}$ there exists (under weak preconditions) a system $(\dot{\mathcal{L}}(L)_I)_{I\in\mathcal{I}}$ with $\mathcal{L}_I \subset \dot{\mathcal{L}}(L)_I$ for each $I \in \mathcal{I}$, where $L = \tau_n^{\{n\}}(\mathcal{L}_{\{n\}})$ for each $n \in I \in \mathcal{I}$. Moreover, in context of Definition 8 it was observed that $\dot{\mathcal{L}}(L)_I$ consists of the non-restricted concurrent run of pairwise disjoint copies of L.

In conjunction, this shows that an adequate restriction of concurrency in $(\dot{\mathcal{L}}(L)_I)_{I\in\mathcal{I}}$ can lead to the construction of well-behaved scalable systems. Therefore, the restricting influence of actions with respect to specific parameter values described above shall now be formalised.

Definition 9 (influence structure)**.** *Let $T \neq \emptyset$ and $\mathcal{I}$ a parameter structure. For each $I \in \mathcal{I}$ and $t \in T$ a* sphere of influence *is specified by $E(t,I) \subset I$. The family*

$$\mathcal{E}_\mathcal{I} = (E(t,I))_{(t,I)\in T\times\mathcal{I}}$$

is called influence structure *for $\mathcal{I}$ indexed by T.*

The non-restricted concurrent run of the pairwise disjoint copies of L will now be restricted in the following way: For each $t \in T$ the runs of all copies k with $k \in E(t,I)$ influence each other independently of the specific values of $k \in E(t,I)$. With respect to our extended example (several servers) with $\mathcal{I}_2$, the spheres of influence $E(t,I)$ are generalisations of the sets $\{r\} \times \hat{I}$, where $I = \mathring{I} \times \hat{I}$ and $t = (r,s) \in \mathring{I} \times \hat{I}$.

Generally, for each $t \in T$ the intersection

$$\dot{\mathcal{L}}(L)_I \cap (\tau^I_{E(t,I)})^{-1}(V) \tag{3}$$

formalises the restriction of the non-restricted concurrent run of the copies of L within $\dot{\mathcal{L}}(L)_I$ by the mutual influence of each element of $E(t,I)$.

Definition 10 (behaviour of influence and influence homomorphisms)**.** *In* (3)*, the* behaviour of influence *V is a prefix closed language $V \subset \Sigma^*$, and for $I, I' \subset N$ the homomorphism $\tau^I_{I'} : \Sigma_I^* \to \Sigma^*$ is defined by:*

$$\tau^I_{I'}(a_i) = \begin{cases} a \mid & a_i \in \Sigma_{I\cap I'} \\ \varepsilon \mid & a_i \in \Sigma_{I\setminus I'} \end{cases}. \tag{4}$$

The homomorphisms $\tau^I_{E(t,I)}$ are called the influence homomorphisms *of $\mathcal{E}_\mathcal{I}$.*

Definition 11 (behaviour-family $(\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I)_{I\in\mathcal{I}}$ generated by the minimal prototype system L, the influence structure $\mathcal{E}_\mathcal{I}$, and the behaviour of influence V)**.** *Because the restriction* (3) *shall hold for all $t \in T$, the restricted systems $\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I$ are defined by the prefix closed languages*

$$\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I := \dot{\mathcal{L}}(L)_I \cap \bigcap_{t\in T} (\tau^I_{E(t,I)})^{-1}(V) \text{ for } I \in \mathcal{I}.$$

Definition 11 shows how synchronisation requirements for the systems $\dot{\mathcal{L}}(L)_I$ can be formalised by influence structures and behaviour of influence in a very general manner. Since, similar to the well-behaved scalable systems $(\dot{\mathcal{L}}(L)_I)_{I\in\mathcal{I}}$, in the systems $(\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I)_{I\in\mathcal{I}}$ each $\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_{\{i\}}$ shall be isomorphic to L for each $\{i\} \in \mathcal{I}$, $V \supset L$ has to be assumed. Therefore, in general we assume for systems $(\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I)_{I\in\mathcal{I}}$ that $V \supset L \neq \emptyset$. Note that $\tau^I_{I'}$ are generalisations of τ^I_n and Θ^I, because

$$\tau^I_n = \tau^I_{\{n\}} \text{ and } \Theta^I = \tau^I_I = \tau^I_N \tag{5}$$

for each $I \subset N$ and $n \in N$.

Further requirements, which assure that $(\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I)_{I\in\mathcal{I}}$ are well-behaved scalable systems, will now be given with respect to $\mathcal{E}_\mathcal{I}$, $\mathcal{B}_\mathcal{I}$, L and V. Assuming $T = N$ and $\varepsilon \in V$ the scalability property is assured by the following technical requirements for $\mathcal{E}_\mathcal{I}$ and $\mathcal{B}_\mathcal{I}$:

Theorem 3 (construction condition for scalable systems)**.** *Let $\mathcal{I}$ be a parameter structure based on N, $\mathcal{E}_\mathcal{I} = (E(n,I))_{(n,I)\in N\times\mathcal{I}}$ be an influence structure for $\mathcal{I}$, and let $\mathcal{B}_\mathcal{I} = (\mathcal{B}(I,I'))_{(I,I')\in\mathcal{I}\times\mathcal{I}}$ be an isomorphism structure for $\mathcal{I}$. Let $\varepsilon \in V \subset \Sigma^*$, for each $I \in \mathcal{I}$ and $n \in N$ let $E(n,I) = \emptyset$, or it exists an $i_n \in I$ with $E(n,I) = E(i_n,I)$, and for each $(I,I') \in \mathcal{I}\times\mathcal{I}, \iota \in \mathcal{B}(I,I')$ and $i \in I$ holds*

$$\iota(E(i,I)) = E(\iota(i),I').$$

Let $E(t,I') = E(t,I) \cap I'$ for each $t \in T$ and $I,I' \in \mathcal{I}, I' \subset I$. Then $(\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I)_{I\in\mathcal{I}}$ is a scalable system with respect to $\mathcal{B}_\mathcal{I}$ and

$$\mathcal{L}(L,\mathcal{E}_\mathcal{I},V)_I = \dot{\mathcal{L}}(L)_I \cap \bigcap_{n\in I} (\tau^I_{E(n,I)})^{-1}(V)).$$

Example 7. *Let $\mathcal{I}$ be a parameter structure based on N, and for $I \in \mathcal{I}$ let $\bar{E}(i,I) := I$ for $i \in N$.*

$$\bar{\mathcal{E}}_\mathcal{I} := (\bar{E}(i,I))_{(i,I)\in N\times\mathcal{I}} \text{ satisfies the assumptions of Theorem 3 for each isomorphism structure } \mathcal{B}_\mathcal{I}. \tag{6}$$

It holds $(\Theta^I)^{-1}(V) = (\tau^I_{\bar{E}(i,I)})^{-1}(V)$ for each $i \in N, I \in \mathcal{I}$, and $V \subset \Sigma^*$.
Therefore, $\mathcal{L}(L,\bar{\mathcal{E}}_\mathcal{I},V)_I = \dot{\mathcal{L}}(L)_I \cap (\Theta^I)^{-1}(V)$ for $I \in \mathcal{I}$.
Especially, $\bar{S}_I = \mathcal{L}(S,\bar{\mathcal{E}}_{\mathcal{I}_1},S)_I$ for each $I \in \mathcal{I}_1$.

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Example 8. *For the parameter structure $\mathcal{I}_2$, and for $\mathring{I} \times \hat{I} \in \mathcal{I}_2$ let*

$$E^2((\mathring{n},\hat{n}),\mathring{I} \times \hat{I}) := \begin{cases} \{\mathring{n}\} \times \hat{I} \mid & \mathring{n} \in \mathring{I} \\ \emptyset \mid & \mathring{n} \in \mathbb{N} \setminus \mathring{I} \end{cases} .$$

$$\mathcal{E}^2_{\mathcal{I}_2} := (E^2((\mathring{n},\hat{n}),\mathring{I} \times \hat{I}))_{((\mathring{n},\hat{n}),\mathring{I}\times\hat{I}) \in (\mathbb{N}\times\mathbb{N})\times\mathcal{I}_2} \quad (7)$$

satisfies the assumptions of Theorem 3 for the isomorphism structure $\mathcal{B}^2_{\mathcal{I}_2}$.

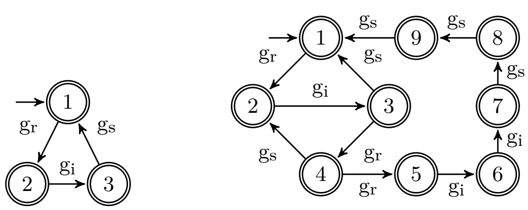
$(\mathcal{L}(S,\mathcal{E}^2_{\mathcal{I}_2},S)_I)_{I\in\mathcal{I}_2}$ is the formalisation of the extended example (several servers) with restricted concurrency.

In order to extend Theorem 3 with respect to self-similarity, an additional assumption is necessary. This is demonstrated by the following counter-example.

Example 9. *Let $G \subset \{\mathrm{g_r},\mathrm{g_i},\mathrm{g_s}\}^*$ the prefix closed language, which is accepted by the automaton Fig. 3(a). Let $H \subset \{\mathrm{g_r},\mathrm{g_i},\mathrm{g_s}\}^*$ the prefix closed language, which is accepted by the automaton in Fig. 3(b). It holds $\emptyset \neq G \subset H$ but $(\mathcal{L}(G,\bar{\mathcal{E}}_{\mathcal{I}_1},H)_I)_{I\in\mathcal{I}_1}$ is not self-similar, e.g.,*

$$\Pi^{\{1,2,3\}}_{\{2,3\}}(\mathcal{L}(G,\bar{\mathcal{E}}_{\mathcal{I}_1},H)_{\{1,2,3\}}) \neq (\mathcal{L}(G,\bar{\mathcal{E}}_{\mathcal{I}_1},H)_{\{2,3\}}$$

because $\mathrm{g_{r1}g_{i1}g_{r2}g_{r3}} \in \mathcal{L}(G,\bar{\mathcal{E}}_{\mathcal{I}_1},H)_{\{1,2,3\}}$, and hence $\mathrm{g_{r2}g_{r3}} \in \Pi^{\{1,2,3\}}_{\{2,3\}}(\mathcal{L}(G,\bar{\mathcal{E}}_{\mathcal{I}_1},H)_{\{1,2,3\}})$, but $\mathrm{g_{r2}g_{r3}} \notin (\mathcal{L}(G,\bar{\mathcal{E}}_{\mathcal{I}_1},H)_{\{2,3\}}$.



(a) Automaton accepting G (b) Automaton accepting H

Figure 3. Counterexample

Definition 12 (closed under shuffle projection)**.** *Let $L,V \subset \Sigma^*$. V is closed under shuffle projection with respect to L, iff*

$$\Pi^{\mathbb{N}}_K[(\bigcap_{n\in\mathbb{N}}(\tau^{\mathbb{N}}_n)^{-1}(L)) \cap (\Theta^{\mathbb{N}})^{-1}(V)] \subset (\Theta^{\mathbb{N}})^{-1}(V) \quad (8)$$

for each subset $\emptyset \neq K \subset \mathbb{N}$. We abbreviate this by $\mathrm{SP}(L,V)$.

Remark 1. *It can be shown that in $\mathrm{SP}(L,V)$ $\mathbb{N}$ can be replaced by each countable infinite set.*

Remark 2. *If L and V are prefix closed with $\emptyset \neq L \subset V$, then it is easy to show that $\mathrm{SP}(L,V)$ follows from self-similarity of $(\mathcal{L}(L,\bar{\mathcal{E}}_{\mathcal{I}_1},V)_I)_{I\in\mathcal{I}_1}$.*

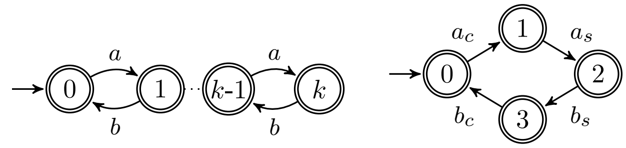
With Definition 12 we are now able to formulate our main result for constructing well-behaved scalable systems defined by a single synchronisation condition.

Theorem 4 (construction condition for well-behaved scalable systems)**.** *By the assumptions of Theorem 3 together with $\mathrm{SP}(L,V)$*

$$(\mathcal{L}(L,\mathcal{E}_{\mathcal{I}},V)_I)_{I\in\mathcal{I}}$$

is a well-behaved scalable system.

Example 10. *For $k \in \mathbb{N}$ let the prefix closed language $F_k \subset \{a,b\}^*$ be defined by the automaton in Fig. 4(a).*



(a) Automaton for $F_k \subset \{a,b\}^*$ (b) One client, one server

Figure 4. Automata at different abstraction levels

With respect to Example 1, $F_1 = S$ holds. It can be shown that $\mathrm{SP}(S,F_k)$ holds for each $k \in \mathbb{N}$. With Theorem 4 now, by (6) and (7) especially, the systems $(\mathcal{L}(L,\bar{\mathcal{E}}_{\mathcal{I}_1},F_k)_I)_{I\in\mathcal{I}_1}$ and $(\mathcal{L}(L,\mathcal{E}^2_{\mathcal{I}_2},F_k)_I)_{I\in\mathcal{I}_2}$ are uniformly monotonic parameterised and self-similar. These are the two cases of the guiding example where the concurrency of the execution of requests is bounded by k.

Theorem 4 is the main result for constructing well-behaved scalable systems defined by a single synchronisation condition. The following section shows how this result together with the Intersection Theorem can be used for constructing more complex well-behaved scalable systems defined by the combination of several synchronisation conditions, as for example well-behaved scalable systems consisting of several component types.

V. Well-behaved Scalable Systems Generated by a Family of Influence Structures

Up to now, the examples were considered at an abstraction level, which takes into account only the actions of the server (or the servers, depending on the choice of the parameter structure).

Example 11. *For a finer abstraction level, which additionally takes into account the actions of the clients, a finer alphabet, e.g., $\check{\Sigma} = \{a_c,b_c,a_s,b_s\}$ and a prefix closed language $\check{S} \subset \check{\Sigma}^*$ is needed, which, e.g., is defined by the automaton in Fig. 4(b).*

In general, a finer relation for system specifications at different abstraction levels can be defined by alphabetic language homomorphisms.

Definition 13 (abstractions)**.** *In general, let $\check{L} \subset \check{\Sigma}^*$ and $L \subset \Sigma^*$ be prefix closed languages. We call $\check{L}$* finer than *L or L* coarser than *$\check{L}$ iff an alphabetic homomorphism $\nu : \check{\Sigma}^* \to \Sigma^*$ exists with $\nu(\check{L}) = L$.*

For each parameter structure $\mathcal{I}$ and $I \in \mathcal{I}$ ν defines an homomorphism $\nu^I : \check{\Sigma}_I^* \to \Sigma_I^*$ by $\nu^I(a_i) := (\nu(a))_i$ for $a \in \check{\Sigma}$ and $i \in I$, where $(\varepsilon)_i := \varepsilon$.

Let now $\mathcal{E}_\mathcal{I}$ be an influence structure for $\mathcal{I}$ indexed by N which is the base of $\mathcal{I}$, and let $\emptyset \neq L \subset V \subset \Sigma^*$ be prefix closed. $(\mathcal{L}(L, \mathcal{E}_\mathcal{I}, V)_I)_{I \in \mathcal{I}}$ induces a restriction of the concurrency in $(\dot{\mathcal{L}}(\check{L})_I)_I$ by the intersections

$$\dot{\mathcal{L}}(\check{L})_I \cap (\nu^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E(t,I)})^{-1}(V)] \text{ for each } I \in \mathcal{I}. \quad (9)$$

If $\check{\tau}^I_{I'} : \check{\Sigma}_I^* \to \check{\Sigma}^*$ is defined analogously to $\tau^I_{I'}$ for $I, I' \subset N$ by

$$\check{\tau}^I_{I'}(a_i) = \begin{cases} a \mid & a \in \check{\Sigma} \text{ and } i \in I \cap I' \\ \varepsilon \mid & a \in \check{\Sigma} \text{ and } i \in I \setminus I' \end{cases},$$

then holds $\tau^I_{I'} \circ \nu^I = \nu \circ \check{\tau}^I_{I'}$. From this it follows that

$$(\nu^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E(t,I)})^{-1}(V)] = \bigcap_{t \in N} (\check{\tau}^I_{E(t,I)})^{-1}(\nu^{-1}(V))$$

and therewith

$$\dot{\mathcal{L}}(\check{L})_I \cap (\nu^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E(t,I)})^{-1}(V)] = \mathcal{L}(\check{L}, \mathcal{E}_\mathcal{I}, \nu^{-1}(V))_I \quad (10)$$

for each $I \in \mathcal{I}$. Notice that $\emptyset \neq \check{L} \subset \nu^{-1}(V) \subset \check{\Sigma}^*$ is prefix closed. So if $(\mathcal{L}(L, \mathcal{E}_\mathcal{I}, V)_I)_{I \in \mathcal{I}}$ fulfils the assumptions of Theorem 3, then this holds for $(\mathcal{L}(\check{L}, \mathcal{E}_\mathcal{I}, \nu^{-1}(V))_I)_{I \in \mathcal{I}}$ as well and the system

$$(\dot{\mathcal{L}}(\check{L})_I \cap (\nu^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E(t,I)})^{-1}(V)])_{I \in \mathcal{I}}, \quad (11)$$

which is defined by the intersections (9), is a scalable system. The following general theorem can be used to prove self-similarity of such systems.

Theorem 5 (inverse abstraction theorem)**.** *Let $\varphi : \Sigma^* \to \Phi^*$ be an alphabetic homomorphism and $W, X \subset \Phi^*$, then*

$$\mathrm{SP}(W, X) \textit{ implies } \mathrm{SP}(\varphi^{-1}(W), \varphi^{-1}(X)).$$

Generally, by (8), $\mathrm{SP}(\nu^{-1}(L), \nu^{-1}(V))$ implies $\mathrm{SP}(X, \nu^{-1}(V))$ for each $X \subset \nu^{-1}(L)$. Especially $\mathrm{SP}(\check{L}, \nu^{-1}(V))$ is implied by $\mathrm{SP}(L, V)$ on account of Theorem 5. So, by Theorem 5, if $(\mathcal{L}(L, \mathcal{E}_\mathcal{I}, V)_I)_{I \in \mathcal{I}}$ fulfils the assumptions of Theorem 4, then

$$\begin{aligned}(\mathcal{L}(\check{L}, &\mathcal{E}_\mathcal{I}, \nu^{-1}(V))_I)_{I \in \mathcal{I}} \\ &= (\dot{\mathcal{L}}(\check{L})_I \cap (\nu^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E(t,I)})^{-1}(V)])_{I \in \mathcal{I}} \quad (12)\end{aligned}$$

is a well-behaved scalable system.

The intersections in (9) formalise restriction of concurrency in $(\dot{\mathcal{L}}(\check{L})_I)_{I \in \mathcal{I}}$ under *one specific aspect* (one specific synchronisation condition), which is given by ν, $\mathcal{E}_\mathcal{I}$, and V. Restriction of concurrency under *several aspects* (several synchronisation conditions) is formalised by the intersections

$$\dot{\mathcal{L}}(\check{L})_I \cap \bigcap_{r \in R} (\nu_r^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E_r(t,I)})^{-1}(V_r)]$$

for each $I \in \mathcal{I}$ based on N, $R \neq \emptyset$ is the index set of the aspects. The family of aspects restricting concurrency is given by

- a family $(\nu_r)_{r \in R}$ of *alphabetic homomorphisms* $\nu_r : \check{\Sigma}^* \to \Sigma^{(r)*}$ for $r \in R$,
- a family $(\mathcal{E}^r_\mathcal{I})_{r \in R}$ of *influence structures* $\mathcal{E}^r_\mathcal{I} = (E_r(t, I))_{(t,I) \in N \times \mathcal{I}}$ indexed by N for $r \in R$, and
- a family $(V_r)_{r \in R}$ of *influence behaviours* $V_r \subset \Sigma^{(r)*}$ for $r \in R$.

From (10) it follows now

$$\begin{aligned}\dot{\mathcal{L}}(\check{L})_I \cap \bigcap_{r \in R} (\nu_r^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E_r(t,I)})^{-1}(V_r)] \\ = \bigcap_{r \in R} \mathcal{L}(\check{L}, \mathcal{E}^r_\mathcal{I}, \nu_r^{-1}(V_r))_I\end{aligned}$$

for each $I \in \mathcal{I}$. Because of the intersection theorem, the uniform monotonic parameterisation and self-similarity of the system

$$(\dot{\mathcal{L}}(\check{L})_I \cap \bigcap_{r \in R} (\nu_r^I)^{-1}[\bigcap_{t \in N} (\tau^I_{E_r(t,I)})^{-1}(V_r)])_{I \in \mathcal{I}} \quad (13)$$

can be inferred from respective properties of the systems

$$(\mathcal{L}(\check{L}, \mathcal{E}^r_\mathcal{I}, \nu_r^{-1}(V_r))_I)_{I \in \mathcal{I}} \text{ for each } r \in R.$$

Using (11) and (12), this requires the verification of the assumptions of Theorem 4 for

$$(\mathcal{L}(\nu_r(\check{L}), \mathcal{E}^r_\mathcal{I}, V_r)_I)_{I \in \mathcal{I}}$$

for each $r \in R$. If $\mathcal{I}$ is based on $N = \bigtimes_{k \in K} N_k$, where K is a finite set and each N_k is countable, then along the lines of $\mathcal{I}_2$, a parameter structure $\mathcal{I}_K$ can be defined for this domain. Such $\mathcal{I}_K$ fit for systems consisting of finitely many component types. Each subset $K' \subset K$ with $\emptyset \neq K' \neq K$ defines a bijection between N and $(\bigtimes_{k \in K'} N_k) \times (\bigtimes_{k \in K \setminus K'} N_k)$. By this bijection, for each of these K' an influence structure $\mathcal{E}^{K'}_{\mathcal{I}_K}$ is defined like $\mathcal{E}^2_{\mathcal{I}_2}$ that satisfies the assumptions of Theorem 3 with respect to an isomorphism structure $\mathcal{B}^K_{\mathcal{I}_K}$ defined like $\mathcal{E}^2_{\mathcal{I}_2}$.

VI. Conclusions and Further Work

This paper presented a formal framework to construct well-behaved scalable systems. The basic parts of that framework are formalisations of parameter structures, influence structures and isomorphisms structures. Together with so-called prototype systems and behaviours of influence these structures formally define scalable systems, if certain conditions are fulfilled. Scalable systems are called well-behaved, iff their behaviour is self-similar. A sufficient condition for such self-similarity is given in terms of prototype systems and behaviours of influence. A deeper analysis of this condition is subject of a forthcoming paper of the authors. One of our motivations for the formal definition of well-behaved scalable systems was the verification of behaviour properties. Usually, behaviour properties of systems are divided into two classes: *safety* and *liveness* properties [23]. Intuitively, a safety property stipulates that "something bad does not happen" and a liveness property stipulates that "something good eventually happens". In [5], it is shown, that for well-behaved scalable systems a wide class of safety properties can be verified by finite state methods. To extend this verification approach to reliability or general liveness properties, additional assumptions for well-behaved scalable systems have to be established. In [24], such assumptions have been developed for uniformly parametrised two-sided cooperations. To generalise these ideas to a wider class of well-behaved scalable systems is subject of further work.

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Analytical Model of an Eddy Current Retarder with Consideration of Nonlinear Magnetization Characteristics of its Ferromagnetic Material

Seongyun Park, Kyihwan Park
School of Mechatronics
Gwangju Institute of Science and Technology
Gwangju, Republic of Korea
baksg@gist.ac.kr, khpark@gist.ac.kr

Changhee Yoo, Jungkyu Hwang
Advanced Development Team
Sangsin Brake Co., Ltd.
Daegu, Republic of Korea
yooch@sangsin.com, teriushjg@sangsin.com

***Abstract*—The nonlinearity magnetization characteristics of the rotor material is a problem in the analysis of eddy current retarders (ECR) because the typical operating conditions of ECRs require high supplied currents. In order to address the nonlinearity of the rotor material, a proper method is needed. In this paper, we propose a process to express the effective magnetic flux considering the nonlinear magnetization characteristics of the ferromagnetic materials used in ECRs. Using a numerical iterative scheme to deal with nonlinear magnetization characteristics, we explain why the torque tends to increase less with higher supplied currents. Using the nonlinear analysis model, we discuss the effects of several design parameters on the torque-speed behavior in order to determine the parameters that are most important for torque-speed performance.**

Keywords-Eddy current retarder, Nonlinear magnetization characteristics, Magnetic saturation, Numerical analysis, Copper layer.

I. INTRODUCTION

With stricter safety regulations being imposed on large vehicles, there is an increasing demand for velocity retarders, which assist the primary friction brakes. In fact, the use of retarders for heavy-duty vehicles is a growing trend in Japan, Australia, Europe, and North America. Among several types of retarders, ECRs are preferred owing to their fast response time and the small installation space they require.

ECRs are classified into linear types, drum types, and disk types, depending on their rotor shapes, and into permanent magnet types and electromagnet types, depending on their flux supply sources. For permanent-type ECRs, the path of the magnetic flux through the rotor is controlled by mechanically moving the magnets, which alters the polarity between adjacent magnets. Although this type of retarder requires no external power source, it is mechanically complex and suffers from off-state leakage flux [1]. Electromagnet-type ECRs, on the other hand, in which the magnetic flux is controlled by a coil current, are mechanically simple and free from flux leakage.

The approach to analyze an ECR can be classified into Finite Element Method (FEM) [2-4] and analytical method [5-10]. Research on ECRs by FEM has mostly been conducted on industrial applications with high torques. Jang analyzed a linear-type ECR with a permanent magnet array using [2]. Choi discussed an optimum method for managing a Halbach array using FEM, as well for linear rotor-type ECRs [3]. For a drum-type ECR with permanent magnets, Alvaro took into consideration the thermal effect [4]. The merit of FEM analysis is the accuracy of its solution. However, the downside of FEM is a long computation time for even moderately complex geometry.

Without recourse to FEM analysis, analytical methods have been utilized to model the eddy current distribution and retarding torque of ECRs. The advantage of analytical methods over FEM is the reduced computation time and increased understanding of physical principles underlying the ECR. Smythe studied the effects of an induced magnetic flux on the retarding behavior [5]. Schieber analyzed the distribution of eddy currents in the vicinity of pole arrays using the magnetic vector potential [6]. By assuming constant power dissipation for all speed regions, Wouterse showed that the retarding force can be described in terms of the air gap and pole diameter [7]. Lee proposed a model by using an image charge method for a disk-type ECR [8]. A representative analytical model was proposed by Davies, who derived the torque equation from a diffusion equation of the eddy current density in a three-dimensional Cartesian coordinate system [9]. By applying the energy conservation, Malti derived a torque equation in three-dimensional cylindrical coordinates [10].

However, the problem of the analytical approaches is that an enormous deviation of the theoretical from the experimental result is produced when a high current is applied to the electromagnets of the ECR. The discrepancy is due to the fact that the magnetic flux density does not increase linearly with the coil current. The approach in this paper to address this difficulty is to find the effective magnetic density by calculating iteratively using the magnetization data of the rotor material.

Section II is devoted to explanation on the configuration of a drum-type ECR. The analytical procedure combined with iteration method is explained in Section III through V. In Section III, the eddy current density equation is derived. Section IV is divided by two parts. In Part A, the magnetic circuit is utilized to obtain an expression for the effective magnetic flux density. Part B explains the iteration steps to compute the effective magnetic flux. After that, Section V derives the torque equation by applying the Lorentz force formula. In result and discussion, the analytical and

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experimental results are compared, and the effects of some design variables on the torque-speed characteristics are evaluated.

II. CONFIGURATION AND PRINCIPLE OF A DRUM-TYPE ECR

Fig. 1 shows the structure and actual shape of the rotor and stator for the drum-type ECR that was studied in the present work. As shown in Fig. 2, the ECR is mounted at the end of a propeller shaft in a vehicle.

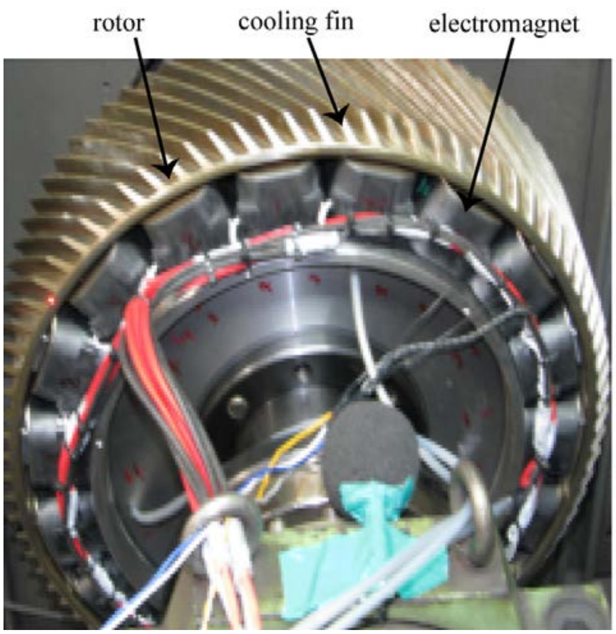

Figure 1. Configuration of drum-type ECR.

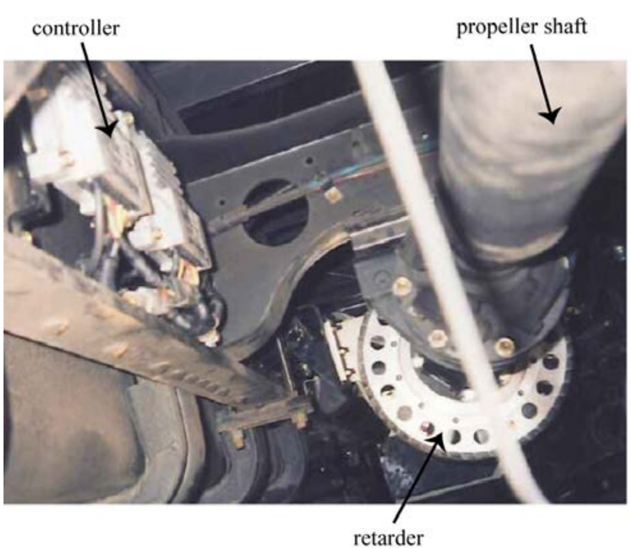

Figure 2. ECR mounted in a vehicle.

A pair of electromagnets was used to supply the magnetic flux in the radial direction through the rotor. Nine pairs of electromagnets were used to produce a peak torque of 800 N·m at a maximum current. The fins on the outer surface of the rotor dissipated the heat generated by the eddy current. The torque is controlled by varying the amount of coil current supplied from the retarder controller. A maximum current of 72 A can be supplied to 18 electromagnets.

The eddy current generated in the moving rotor and the applied magnetic flux interact to produce a force that opposed the direction of rotation, attributed to Faraday's law. In the next session, the physical model of an ECR is developed based on fundamental electromagnetic equations.

III. DERIVATION OF EDDY CURRENTS IN THE ROTOR

Table 1 lists the definitions of the design parameters and material properties. Fig. 3 shows the front, side, and isometric views of one pair of the electromagnets and the corresponding portion of the rotor. The rotor is usually made of steel to ensure its high stiffness. The surface of the rotor is coated with copper in order to generate a large amount of eddy current, because copper has a higher conductivity than steel. In this work, the permeability and conductivities of the steel and copper are considered separately in order to analyze the effect of the copper coating thickness on the generation of the eddy current.

TABLE I. DEFINITION OF THE DESIGN PARAMETERS AND MATERIAL PROPERTIES

Symbol	**Physical property**
μ_1, μ_1	permeability (N·A^2)
r_1, r_2	resistivity (Ω·m)
g	airgap (mm)
λ	pole pitch (mm)
d	depth of copper plating (mm)
L	axial length of rotor (mm)
L_a	axial length of pole (mm)
L_b	pole width (mm)
D	rotor diameter (mm)
p	number of pole pairs

The distribution of eddy current density is described by (1), which is obtained using Faraday's law, Ampere's law, and Ohm's law [9].

$$\nabla^2 \mathbf{J} = \frac{\mu}{\rho}\frac{\partial \mathbf{J}}{\partial t}. \tag{1}$$

Since we know that there is no eddy current density in the z direction at the edges of the rotor, there is no eddy current in the *x* direction at the center line of rotor. Hence, we have boundary conditions of

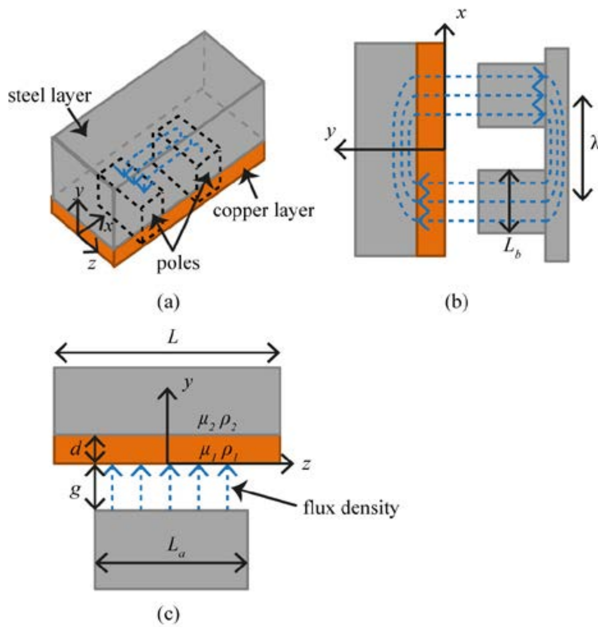

Figure 3. One pair of the electromagnets and corresponding portion of the rotor: (a) Isometric view, (b) side view, and (c) front view.

$$J_{z,copper}\left(z=L/2\right)=0,\ J_{x,copper}\left(z=0\right)=0,$$
$$J_{z,steel}\left(z=L/2\right)=0,\ J_{x,steel}\left(z=0\right)=0. \tag{2}$$

From the electric field boundary condition of $E_{z,steel}=E_{z,copper}$ and by using $\mathbf{E}=\rho\mathbf{J}$, we obtain

$$\rho_1 J_{z,copper}=\rho_2 J_{z,steel}. \tag{3}$$

From the magnetic field boundary condition of $H_{z,steel}=H_{z,copper}$ and by using $\nabla\times\mathbf{J}=-\dfrac{\mu}{\rho}\dfrac{\partial\mathbf{H}}{\partial t}$, we obtain

$$\frac{\rho_1}{\mu_1}\frac{\partial J_{x,copper}}{\partial y}=\frac{\rho_2}{\mu_2}\frac{\partial J_{x,steel}}{\partial y}. \tag{4}$$

Then, using (1), (2), (3), and (4), we obtain the eddy current density in the copper-and-steel layer [9] as follows:

$$\begin{aligned}
J_{x,copper}&=\mathrm{Re}\frac{\lambda}{2L}e^{j\left(n\left(2\pi x/\lambda-\omega t\right)-\pi/2\right)}\left\{\frac{e^{\gamma_{1mn}(2d-y)}+e^{\gamma_{1mn}y}}{1+e^{2\gamma_{1mn}d}}\right\}\frac{m}{n}\hat{J}_{mn}\sin\frac{m\pi z}{L},\\
J_{z,copper}&=\mathrm{Re}\,e^{j\left(n\left(2\pi x/\lambda-\omega t\right)\right)}\left\{\frac{e^{\gamma_{1mn}(2d-y)}+e^{\gamma_{1mn}y}}{1+e^{2\gamma_{1mn}d}}\right\}\hat{J}_{mn}\cos\frac{m\pi z}{L},\\
J_{x,steel}&=\mathrm{Re}\frac{\rho_1}{\rho_2}\frac{\lambda}{2L}e^{j\left(n\left(2\pi x/\lambda-\omega t\right)-\pi/2\right)}\left\{\frac{2e^{\gamma_{1mn}d}}{1+e^{2\gamma_{1mn}d}}\right\}e^{\gamma_{2mn}(d-y)}\frac{m}{n}\hat{J}_{mn}\sin\frac{m\pi z}{L},\\
J_{z,steel}&=\mathrm{Re}\frac{\rho_1}{\rho_2}e^{j\left(n\left(2\pi x/\lambda-\omega t\right)\right)}\left\{\frac{2e^{\gamma_{1mn}d}}{1+e^{2\gamma_{1mn}d}}\right\}e^{\gamma_{2mn}(d-y)}\hat{J}_{mn}\cos\frac{m\pi z}{L}.
\end{aligned} \tag{5}$$

where $\hat{J}_{mn}$ is the magnitude of the eddy current, and γ_{2mn} and γ_{1mn} are defined as

$$\begin{aligned}
\gamma_{1mn}{}^2&=\left(\frac{2n\pi}{\lambda}\right)^2+\left(\frac{m\pi}{L}\right)^2+j\omega n\frac{\mu_1}{\rho_1},\\
\gamma_{2mn}{}^2&=\left(\frac{2n\pi}{\lambda}\right)^2+\left(\frac{m\pi}{L}\right)^2+j\omega n\frac{\mu_2}{\rho_2}.
\end{aligned} \tag{6}$$

IV. THEORETICAL MAGNETIC CIRCUIT MODEL FOR THE PROPOSED ECR

Let us assume that the eddy current density $\mathbf{J}$ and effective magnetic flux density $\mathbf{B}_e$ are generated when the rotor is in motion. We then have Faraday's law as

$$\rho\left(\nabla\times\mathbf{J}\right)=-\frac{\partial\mathbf{B}_e}{\partial t}. \tag{7}$$

By inserting $J_{x,copper}$, $J_{z,copper}$ into (7), we obtain the effective magnetic flux density in the y direction, B_{ey}, as

$$\begin{aligned}
B_{ey}\left(x,y,z,t\right)&=-\int\rho_1\left(\frac{\partial J_{x,copper}}{\partial z}-\frac{\partial J_{z,copper}}{\partial x}\right)dt\\
&=\mathrm{Re}\,e^{jn\left(2\pi x/\lambda-\omega t\right)}\hat{B}_{mn}\left\{\frac{e^{\gamma_{1mn}(2d-y)}+e^{\gamma_{1mn}y}}{1+e^{2\gamma_{1mn}d}}\right\}\cos\frac{m\pi z}{L}.
\end{aligned} \tag{8}$$

where $\hat{B}_{mn}$ represents the amplitude of the effective flux density [9] and is defined as

$$\hat{B}_{mn}=-\frac{2\pi\rho_1}{\lambda\omega}\hat{J}_{mn}\left\{1+\left(\frac{m\lambda}{2nL}\right)^2\right\}. \tag{9}$$

$\hat{B}_{ey}$ is an essential variable in the eddy current analysis in this work, since it is ultimately used for calculating the retarding torque.

A. Magnetomotive relationship used for obtaining amplitude of effective flux density

As can be seen in (8), $\hat{B}_{ey}$ can be obtained only if $\hat{B}_{mn}$ is correctly calculated. The other parameters such as λ, ω, d, γ_1, and γ_2 can be easily obtained from the mechanical specifications of the retarder. For this purpose, we will consider the relationships associated with the magnetic circuit, which is composed of a pair of poles and a rotor, as shown in Fig. 4(a).

From the magnetic circuit model shown in Fig. 4(b), we obtain the first relationship as

$$\mathbf{F}_{sy}\left(0,0,0,t\right)=\mathbf{F}_{iy}\left(0,0,0,t\right)+\mathbf{F}_{ey}\left(0,0,0,t\right) \tag{10}$$

where $\mathbf{F}_{\mathbf{sy}}$ is the supplied magnetomotive force produced by the supplied current, $\mathbf{F}_{\mathbf{iy}}$ is the induced magnetomotive force produced by the eddy current, and $\mathbf{F}_{\mathrm{ey}}$ is the effective magnetomotive force.

Equation (10) is obtained from the magnetic circuit, as shown in Fig. 4(b). In Fig. 4(b), Φ_{ey} and R are the effective magnetic flux and the effective reluctance, respectively. We obtain $\mathbf{F}_{\mathbf{sy}}$, $\mathbf{F}_{\mathbf{iy}}$, and $\mathbf{F}_{\mathrm{ey}}$ as follows:

1) Supplied magnetomotive force

If we assume that the supplied magnetomotive force $\mathbf{F}_{\mathbf{sy}}\left(x,0,z,t\right)$ has a constant value of Ni above the pole area and 0 elsewhere, we can describe $\mathbf{F}_{\mathbf{sy}}\left(x,0,z,t\right)$ as a Fourier series as

$$\mathbf{F}_{\mathrm{sy}}\left(x,0,z,t\right)=Ni\,\mathrm{Re}\sum_{n=1,3,\cdots}C_n e^{j\left(n\left(2\pi x/\lambda-\omega t\right)-\frac{n\pi L_b}{\lambda}-n\phi\right)}\sum_{m=1,3,5,\cdots}C_m\cos\frac{m\pi z}{L} \quad (11)$$

where, $C_m=\dfrac{4}{m\pi}\sin\left(\dfrac{m\pi L_a}{2L}\right)$ and $C_n=\dfrac{4}{n\pi}\sin\left(\dfrac{n\pi L_b}{\lambda}\right)$ represent the coefficients of a Fourier series for a square wave, and ϕ is the phase delay with respect to the effective magnetomotive force.

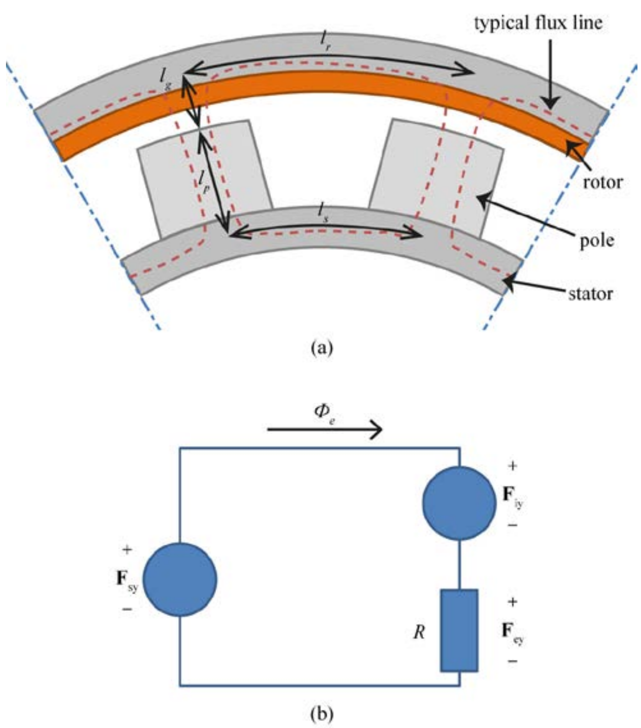

Figure 4. (a) Magnetic flux path of the ECR and (b) diagram of its magnetic circuit showing the relationship between magnetomotive forces.

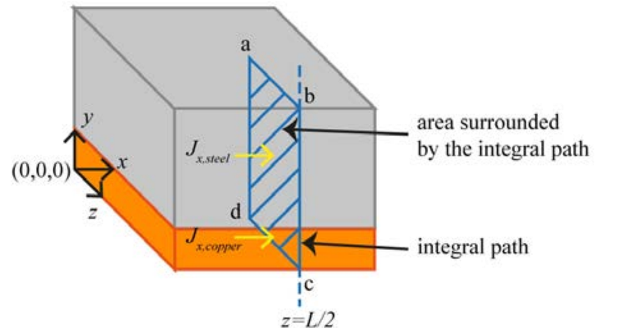

Figure 5. Rotor segment with eddy current density vectors indicated.

At $x=y=z=0$, $\mathbf{F}_{\mathbf{sy}}$ is

$$\mathbf{F}_{\mathrm{sy}}\left(0,0,0,t\right)=Ni\sum_{n=1,3,\cdots}\sum_{m=1,3,5,\cdots}C_nC_m\cos\left(n\omega t+\frac{n\pi L_b}{\lambda}+n\phi\right). \quad (12)$$

2) Induced magnetomotive force

Fig. 5 shows a rotor segment with eddy current density vectors indicated as $J_{x,copper}$ and $J_{x,steel}$. As $J_{x,copper}$ and $J_{x,steel}$ flow through the area enclosed by the integral path "abcd," the induced magnetomotive force $\mathbf{F}_{\mathbf{iy}}\left(x,0,z,t\right)$ can be calculated as

$$\mathbf{F}_{\mathrm{iy}}\left(x,0,z,t\right)=\int_{z'=z}^{\frac{L}{2}}\int_{y=0}^{d}J_{x,copper}\,dydz'+\int_{z'=z}^{\frac{L}{2}}\int_{y=d}^{\infty}J_{x,iron}\,dydz'. \quad (13)$$

By inserting $J_{x,copper}$, $J_{x,iron}$ from (5) and $x=z=0$ into (13), we obtain

$$\mathbf{F}_{\mathrm{iy}}\left(0,0,0,t\right)=\sum_{n}\sum_{m=1,3,5,\cdots}\hat{B}_{mn}\frac{\lambda^2\omega C_m}{4\pi^2 n\rho_1}\frac{d+\dfrac{\rho_1}{\sqrt{2}\rho_2\alpha_{2n}}}{1+\left(\dfrac{m\lambda}{2nL}\right)^2}\cos\left(n\omega t+\pi/2\right). \quad (14)$$

3) Effective magnetomotive force

The effective magnetomotive force $\mathbf{F}_{\mathbf{ey}}\left(0,0,0,t\right)$ can be expressed using the effective magnetic reluctance R and effective magnetic flux Φ_{ey} as

$$\mathbf{F}_{\mathrm{ey}}\left(0,0,0,t\right)=R\Phi_{ey}=RA_pB_{ey}=RA_p\hat{B}_{mn}\cos\left(n\omega t\right) \quad (15)$$

where A_p is the pole area.

From (12), (14), and (15), the phase relationship among magnetomotive forces can be represented as shown in Fig. 6. We note that $\mathbf{F}_{\mathbf{ey}}$ leads $\mathbf{F}_{\mathbf{iy}}$ by a phase angle of 90°. Hence, by applying the Pythagorean theorem to the rectangular triangle composed of $\mathbf{F}_{\mathbf{sy}}$, $\mathbf{F}_{\mathbf{ey}}$, and $\mathbf{F}_{\mathbf{iy}}$, we obtain the following relationship:

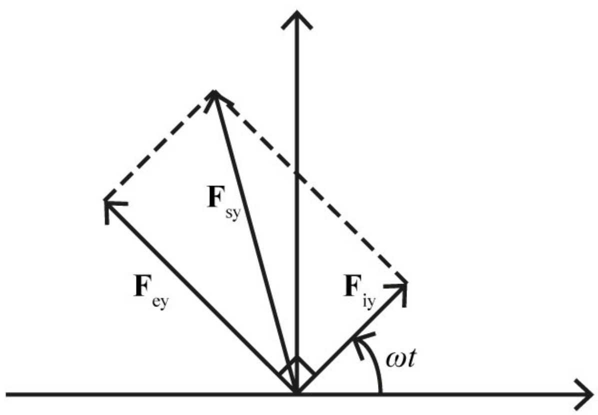

Figure 6. Phasor diagram of magnetomotive forces with phase relationships.

$$\hat{B}_{mn}{}^{2} = \frac{\left(NiC_nC_m\right)^2}{\left(RA_p\right)^2 + \left\{ \dfrac{\lambda^2\omega C_m}{4\pi^2 n\rho_1} \dfrac{d+\dfrac{\rho_1}{\sqrt{2}\rho_2\alpha_{2n}}}{1+\left(\dfrac{m\lambda}{2nL}\right)^2} \right\}^2}. \tag{16}$$

By using (16) [9], the effective magnetic flux density can be calculated.

B. Ampere's law used for obtaining the amplitude of the effective flux density

Since R is unknown in (16), as well as $\hat{B}_{mn}$, we need another relationship in order to obtain $\hat{B}_{mn}$. For this purpose, we apply Ampere's law $Ni = \oint Hdl$ to the magnetic flux path, which is denoted by a dashed line in Fig. 4(a). As a result, we obtain

$$\left|\mathbf{F_{ey}}\right| = 2H_pl_p + 2H_gl_g + H_rl_r + H_sl_s \tag{17}$$

where l_p, l_g, l_r, and l_s are the flux paths, as shown in Fig. 4(a).

Using the constitutive relationship $B = \mu H$, (17) can be represented in terms of the magnetic reluctance R and magnetic flux Φ_e as

$$\begin{aligned}\left|\mathbf{F_{ey}}\right| &= 2\frac{l_p}{\mu_p}B_p + 2\frac{l_g}{\mu_g}B_g + \frac{l_r}{\mu_r}B_r + \frac{l_s}{\mu_s}B_s \\ &= \left(R_p + R_g + R_r + R_s\right)\Phi_e = R\Phi_e.\end{aligned} \tag{18}$$

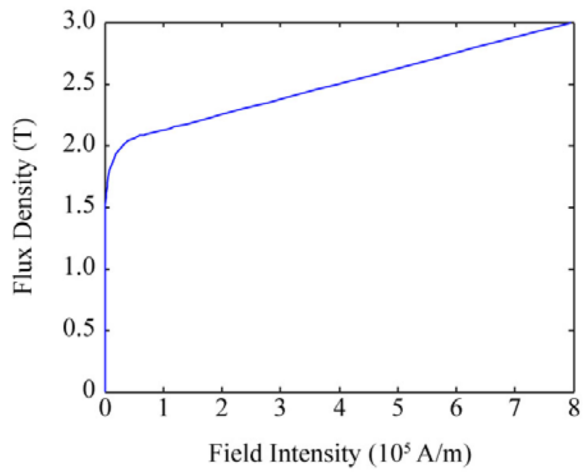

Figure 7. Magnetization curve for the rotor and stator.

Here, $R_p = \frac{2l_p}{A_p\mu_p}, R_g = \frac{2l_g}{A_p\mu_0}, R_r = \frac{l_r}{A_r\mu_r}$, and $R_s = \frac{l_s}{A_s\mu_s}$ are the reluctance of the pole, the rotor, and the stator, where A_p, μ_p, A_r, μ_r, A_s, and μ_s denote the areas and the permeability of the pole, the rotor, and the stator, respectively.

Since R_p, R_r, and R_s are negligible owing to the high relative permeability of the rotor and the pole, R becomes R_g if the rotor material is assumed to have a linear magnetization characteristic. By substituting the simplified reluctance into (16), we obtain $\hat{B}_{mn}$ as

$$\hat{B}_{mn}{}^{2} = \frac{\left(NiC_nC_m\right)^2}{\left(R_gA_p\right)^2 + \left\{ \dfrac{\lambda^2\omega C_m}{4\pi^2 n\rho_1} \dfrac{d+\dfrac{\rho_1}{\sqrt{2}\rho_2\alpha_{2n}}}{1+\left(\dfrac{m\lambda}{2nL}\right)^2} \right\}^2}. \tag{19}$$

Given that $\hat{B}_{mn}$ is obtained by (19), B_{ey} is also obtained by (8).

In practice, however, R is not analytically determined because the steel that composes the rotor and the stator exhibits a nonlinear magnetization characteristic for a flux density of around 1.5 T, as shown in Fig. 7. This means that $\hat{B}_{mn}$ cannot be analytically by (19), in which a linear magnetization characteristic is assumed. To address this difficulty, we solve (16) and (21) simultaneously, with the help of the numerical method given in the following description.

- We assume an initial value for $\hat{B}_{mn}$. From this value, $\left|\mathbf{F_{iy}}\right|$ and $\left|\mathbf{F_{ey}}\right|$ are obtained using (14) and (15).

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- Since the magnetic flux Φ_e is generated by $\left|\mathbf{F}_{\mathbf{ey}}\right|$, we can use (17) and check the equality using the magnetization curve shown in Fig. 7 and using the relationships $B_p = \Phi_e / A_p$, $B_g = \Phi_e / A_p$, $B_r = \Phi_e / A_r$, $B_s = \Phi_e / A_s$. Until both sides of (17) become the same, Φ_e keeps increased slightly. The final value of Φ_e is a numerical solution of the magnetic flux generated by $\left|\mathbf{F}_{\mathbf{ey}}\right|$. We then have

$$\hat{B}_{mn}' = \frac{\Phi_e}{A_p}. \tag{20}$$

- Check whether the calculated solution $\hat{B}_{mn}'$ is the same as the assumed value of $\hat{B}_{mn}$. When they are different, increase $\hat{B}_{mn}$ slightly and repeat the above process, beginning with 1), until the calculated value $\hat{B}_{mn}'$ becomes the same as the assumed value $\hat{B}_{mn}$. The final value $\hat{B}_{mn}$ obtained in this iterative process is the numerical solution of the effective magnetic flux density.

Similar to the linear case, since $\hat{B}_{mn}$ has been obtained, B_{ey} can be obtained using (8).

V. RETARDING TORQUE OF THE ECR

Fig. 8 shows the infinitesimally small volume in which B_{ey} and J_z interact. The Lorentz force F_x is obtained via the cross product of $J_z \times B_{ey}$. The eddy current retarding torque T is calculated by integrating the cross product of F_x and the distance from the center to the point of action $\frac{D}{2} + y$ over the entire rotor volume. Therefore, the total retarding torque is a summation of the torque generated at copper, T_{copper}, and the torque generated at steel, T_{steel}, as follows:

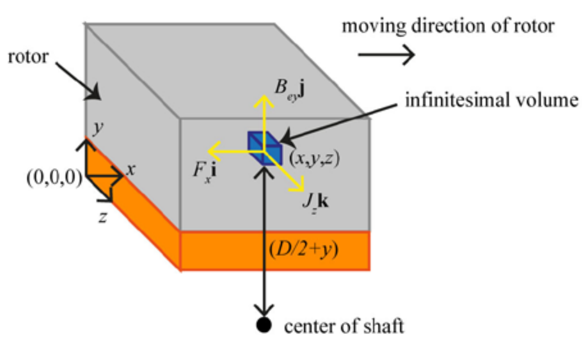

Figure 8. Lorentz force acting on the infinitesimal volume of the rotor.

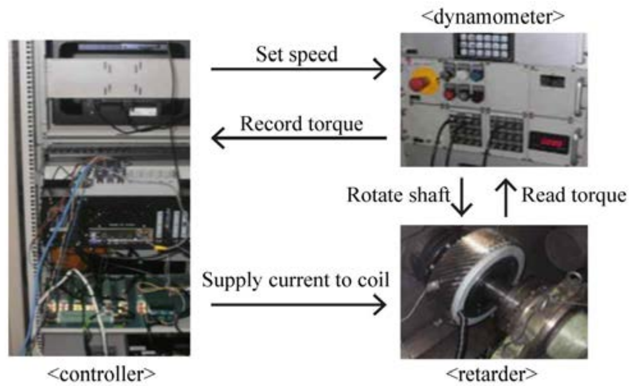

Figure 9. Schematic diagram of experimental setup.

$$\begin{aligned} T_{copper} &= \sum_n \sum_m \frac{n\omega}{2\pi} \int_{x=0}^{p\lambda/n} \int_{y=0}^{d} \int_{z=-L/2}^{L/2} \int_{t=0}^{2\pi/n\omega} \left(J_{z,copper}\right)\left(B_{ey,copper}\right)\left(y + \frac{D}{2}\right) dxdydzdt \\ &= -\sum_n \sum_m \frac{dLp^2\lambda^3\omega C_n{}^2 C_m{}^2}{16\pi^2\rho_1} \hat{B}_{mn}{}^2 \left\{1 + \left(\frac{m\lambda}{2nL}\right)^2\right\}^{-1}, \end{aligned} \tag{21}$$

$$\begin{aligned} T_{steel} &= \sum_n \sum_m \frac{n\omega}{2\pi} \int_{x=0}^{p\lambda/n} \int_{y=d}^{\infty} \int_{z=-L/2}^{L/2} \int_{t=0}^{2\pi/n\omega} \left(J_{z,steel}\right)\left(B_{ey,steel}\right)\left(y + \frac{D}{2}\right) dxdydzdt \\ &\cong -\sum_n \sum_m \frac{p^2\lambda^3\omega L C_n{}^2 C_m{}^2}{32\pi^2\alpha_{2n}\rho_2} \hat{B}_{mn}{}^2 \left\{1 + \left(\frac{m\lambda}{2nL}\right)^2\right\}^{-1}, \end{aligned} \tag{22}$$

$$T = T_{copper} + T_{steel} = -\sum_n \sum_m \frac{p^2\lambda^3 L\omega C_n{}^2 C_m{}^2}{16\pi^2\rho_1} \hat{B}_{mn}{}^2 \frac{\left\{d + \rho_1 / 2\rho_2\alpha_{2n}\right\}}{\left\{1 + \left(\frac{m\lambda}{2nL}\right)^2\right\}}, \tag{23}$$

where $\alpha_{2n} = \sqrt{\frac{n\omega\mu_2}{2\rho_2}}$.

It can be seen that torque is a function of variable $\hat{B}_{mn}$ at some speed because other symbols in (23) are given by mechanical specification.

VI. RESULT AND DISCUSSION

A. Experimental setup

Fig. 9 shows the experimental setup used for the eddy current retarding torque experiments. An engine dynamometer was used to supply the rotational velocity and to measure the retarding torque. A coil current was applied to the retarder from a power supply by a command from a computer, and the retarding torque was recorded in the computer. The rotational velocity was increased from 200 rpm to 1000 rpm. The supplied current was increased from 1 A to 4 A, making a total of 18 A to 72 A when multiplied by the number of poles. In order to exclude the effect of the thermal heat generated at the rotor, the rotor was cooled for more than 30 min. A total of 24 experiments were performed.

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B. Comparison between the experimental and theoretical results with the consideration of nonlinear magnetization characteristics of the material

Fig. 10(a) shows the theoretical and experimental results of the torque characteristics of the rotational velocity when the magnetization characteristics are assumed to be linear. When the applied current is low, the theoretical and experimental results agree well. However, when the applied current is increased, the real torque is substantially reduced as compared to the theoretical torque.

Fig. 10(b) shows the theoretical and experimental results for the torque characteristics of the rotational velocity when the magnetization characteristics are nonlinear, since this is practical. As can be seen in the figure, the real torque agrees well with the theoretical torque in the overall current ranges.

C. Dependency of torque characteristics on variations in the design parameter

Fig. 11 shows the torque characteristics with respect to the copper thickness. As shown in Fig. 11(b), the position of the maximum torque varies for different values of *d*. The torque in a high rotating speed region decreases with an increase in *d*. This result indicates that the reduction of the magnetic flux from the air gap increases due to increased copper thickness is more dominant than the increase in the eddy current. When the copper thickness increases, the maximum torque is obtained at a low rotational speed.

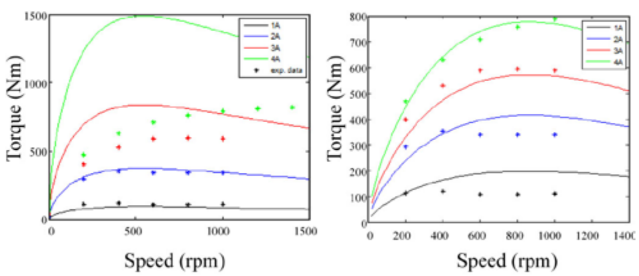

Figure 10. Torque-speed curves for (a) linear model and (b) nonlinear model.

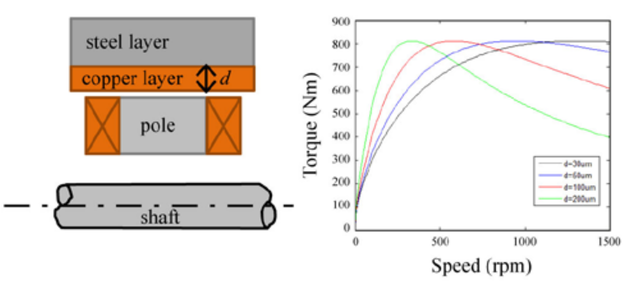

Figure 11. (a) Crosse section of the retarder with copper thickness indicated by d and (b) analytical torque curves obtained for I = 4 A when d varies by specified values of 30, 50, 100, and 200 μm.

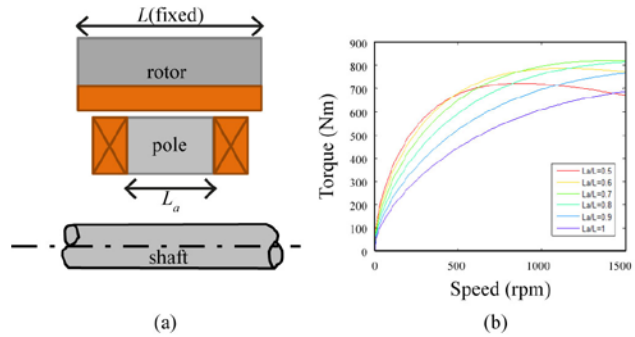

Figure 12. (a) Crosse section of the retarder with rotor and pole widths specified as L and L_a, respectively, and (b) analytical torque curves obtained for I = 4 A when L_a/L varies from 0.5 to 1.

From these results, it can be said that it is important for the thickness of the copper coating to be evenly distributed over the rotor, since the torque is strongly affected by even a slight variation in its thickness.

Fig. 12 shows how the torque varies with the rotor-width-to-pole-width ratio. In order to compare only the effect of the applied magnetic flux density, the total supplied current and the rotor length were assumed to be constant. As shown in Fig. 12(b), the torque-speed relationship varies for L_a/L.

When the pole area is decreased, even though the magnetic flux density is increased, the torque decreases owing to the reduced area in which the eddy current is generated. For the speed region from 1000 rpm to 1500 rpm, which is a typical driving condition, the ratio of 0.7 gives a maximum torque value.

VII. Conclusion

In this paper, we have proposed a process to express the effective magnetic flux in terms of the nonlinear magnetization characteristics of ferromagnetic materials, an issue that has not been considered in previous analytical methods of an ECR. Using a numerical iterative scheme to deal with the nonlinear magnetization relationship, we explained why the torque does not linearly increase in response to higher supplied currents. This result implies that the nonlinearity of the material should be adequately addressed in the analysis of ECRs. Using the iterative analytical model, we discussed the effect of several design parameters on the torque-speed behavior. It was discussed that the copper thickness and the rotor-width-to-pole-width ratio play an important role in determining the torque-speed performance.

For the sake of simple derivation of equations, the rotor surface was assumed to be flat. This assumption would be appropriate if the rotor diameter is much greater than the pole pitch. Further researches on deriving the model based on cylindrical coordinates and comparison it with the model based on Cartesian coordinates would be beneficial for the effect of rotor curvature on the analysis of an ECR.

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Optimization of HDF5 Performance for Virtual Reality Objects Enhanced by Implicit Features

Alexander Tzokev
Faculty of Industrial Technologies
Technical University of Sofia
Sofia, Bulgaria
alextz@tu-sofia.bg

Angel Bachvarov, Stoyan Maleshkov
Faculty of German Engineering Education and Industrial Management (FDIBA)
Technical University of Sofia
Sofia, Bulgaria
a_bachvarov@tu-sofia.bg, maleshkov@tu-sofia.bg

***Abstract*—The amount of data used in Virtual Reality environments can be very large especially when working on real-world engineering problems. The interaction between the environment and the user or enhancement the objects therein with implicit features require integration of a high performance data management system which allows an efficient data handling. The deployment of specialized scientific databases such as Hierarchical Data Format 5 (HDF5) offers certain advantages over more business-oriented relational database management systems. This paper presents results from a study for optimization of the storage efficiency of the HDF5 data base trough chunked datasets enabling effective handling of the large data amounts produced within and for virtual reality environments. Further, a method for predicting the optimal chunk size or arrays of native data types with rank 1 and 2 is discussed.**

Keywords – Virtual Reality; HDF5; Data Chunking; Chunk Size Prediction; Databases.

I. INTRODUCTION

Recently a concept of so called implicit features of the objects for enhancement of their immersive representation within multimodal virtual reality (VR) environments was proposed in [1]. The implicit features represent "hidden" properties of an object which normally are not part of the object model and cannot be perceived directly by the observer through his/her senses, such as magnetization, radiation, humidity, toxicity, surface roughness etc.

The VR paradigm assumes that the virtual objects are fully functional replicas of the physical originals. However, the commonly used approaches for building the virtual objects are related to some limitations in their presentation within the virtual world. Only the so called *explicit* features of the objects, meaning the (geometrical, structural and topological) characteristics which could be perceived directly by an observer are covered. Furthermore, the mechanism of integration the newly created objects in the VR environment imposes some additional restrictions due to the need for simplification of the data structure describing the model. All this leads to significant reduction of the informational content and the use of VR technologies for presentation of the objects does not give any substantial advantages in comparison to the much affordable conventional techniques, which depends mainly on the observer's visual channel. Normally, he/she can explore the object model trough a set of commands which modify its spatial position, size and attributes.

The new approach using the implicit features extends the information content and deepens the immersive representation of the object model through its enhancement with a set of already mentioned implicit features. An example for this is shown in Fig. 1 where beside the visually perceived attributes the observer can get additional information concerning the functional behavior and operating modes of the object. Depending on the specific needs the properties could be presented only visually or in combination with sonification, which means by aural and/or tactile channel for perceptualizing data. This approach extends the underlying general scene-graph structure of the VR software system incorporating additional effects nodes for implicit features representation. It is flexible and can be easily implemented on different hardware configurations on a single computer or on a computer cluster for immersive presentation. The use of implicit features not only enhances the informational content of the virtual objects, but also enables their exploration in different contexts enabling representation of different properties sets according, e.g. narrative, informative, marketing, technical, instructional etc.

The assignment (mapping) of the implicit features to the CAD model of the object and further manipulation of the relevant datasets are discussed in detail in [2]. In the general case, this is done using simulation input and configuration files from simulation solvers. The reason for this is the fact that the most of the researchers are still looking at VR mainly as a CAD/CAE post-processing tool only for viewing and assessing the design and simulations results in real time.

Within this context the deployment of a high performance parallel hierarchical data management and storing system, such as HDF5 [3] is significant precondition for successful implementation of the concept.

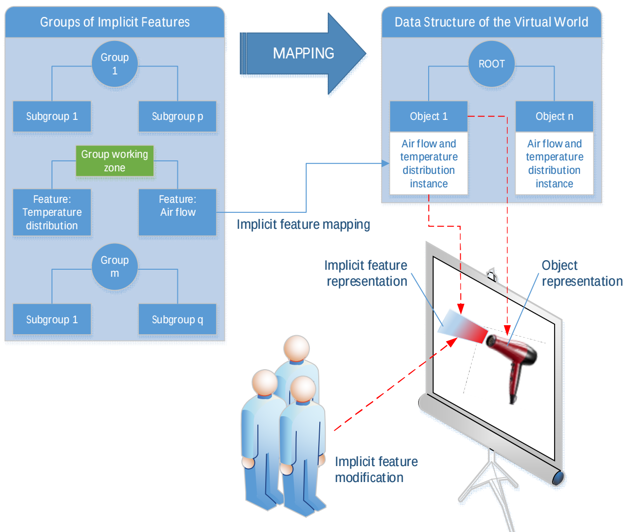

Figure 1. Implicit features mapping in VR

A preliminary study [4] analyzes the possibilities for using NoSQL database management systems (DBMSs) when storing and retreiving large ammount of scientific information in form of 2D arrays storing image infromation. The I/O performance of MySQL is compared to HDF5 and the results proofs that the HDF technology is faster in terms of reading and writing data when managing large datasets. In some cases the information query is faster with MySQL due to lack of builtin indexing in HDF5.

The computing power used at the scientific and engineering simulations and the generated ammount of data require a high degree of parallelism both within the software applications and at all levels of the underlying architectural platforms and storage systems [5]. The HDF model was design for efficient parallel usage and storage of very large multidimensional datasets of complex data types. To achieve optimal performance the HDF5 chunk and cache parameters must be fine-tuned and an algorithm for prediciting their values will be useful.

Further this paper contains overview of the HDF5, representing the software model and the implementation of the technology in VR objects enhanced by implicit features, a discussion of experiments, and results for the HDF5 performance with different chunk sizes is given. The next part of the paper presents a model based on the experimental data for predicting the optimal chunk size for 2D arrays of double data type. Finnaly, the conclusions and acknowedgments are given.

II. HDF5 Technology

In order to implement the mapping of the implicit features in VR as presented in Fig.1, a high performance database management system is required for managing a large amount of numerical data of different type (scalar values, multidimensional arrays, vectors, tensors, etc.). An easy and convenient way to define the object hierarchy is use of HDF5 data model or other type of specialized scientific hierarchical data files. Using relational DBMSs shall result in complex data and links descriptions and shall cause possible performance bottlenecks.

The HDF5 consists of a specialized data model, software libraries and a file format for data management, which supports different data types. Such a model is very flexible and efficient when working with high-volume and complex scientific datasets. An HDF5 file contains variables, arrays, groups and types which, based on the software model are known as Datasets, Group and Datatype. The model also defines simple and extending link mechanism for creating associations between information items in the file [6]. Fig. 2 shows the HDF5 models and their implementations between software objects [7].

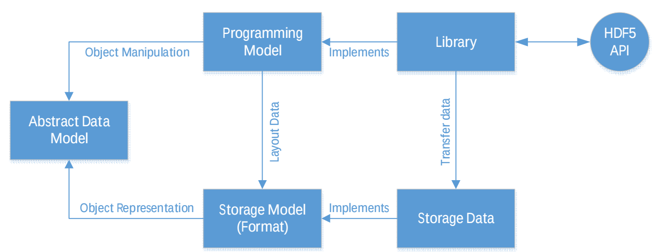

Figure 2. General HDF5 model and implementation

All these features determine the HDF5 as a favorable solution for data management system in VR environments.

The HDF5 datasets are array variables whose data elements are described and shaped as multidimensional arrays (up to rank of 32). A dataset can shrink and grow its limits up to predefined maximum extent. Depending on the storage layout strategy, this maximum extend can be virtually unlimited [6]. The latest version of the HDF model supports the following storage layouts:

- *Contiguous* – array elements are laid out as a single sequence in the file;
- *Chunked* – array elements are stored as a collection of sub-arrays with preliminary defined fixed-size, called chunks;
- *Compact* – used for small datasets with size less than 64 KB. All elements can be read as a part of variable’s metadata or header retrieval.

The performance and some other capabilities of HDF5 depend mainly on the used storage layout. For partial data manipulation the so called hyperslab HDF5 object is required and the storage layout should be chunked.

An important task is to define the optimal chunk size and shape for each dataset, since this parameter is related to the overall I/O performance and as stated before to the additional features (i.e. hyperslabs, data compression, encryption, etc.). A general metric for the storage efficiency is the expected number of chunks retrieved by queries under access workload. In [8], Otoo et al. have presented mathematical models based on geometrical programming and steepest descent optimization for defining the chunk parameters.

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The prediction of the optimal chunk size of HDF5 dataset based on the size and rank of the input array may improve the performance of the data storage system and shall lead to more fluent communication and workflow. Hence, one of the main purposes of this study is to define a model for adaptive prediction of the chunk size for HDF5 datasets based on the size and shape of the data.

Another possibility to improve the I/O performance of the HDF5 data set is to adapt the chunk cache size by its automatically resizing as needed for each operation. The cache should be able to detect when the cache should be skipped or when it needs to be enlarged based on the pattern of I/O operations. At a minimum, it should be able to detect when the cache would severely hurt performance for a single operation and disable the cache for that operation [7]. However, the chunk cache is relevant to the chunk size and defining the optimal size for the chunk will lead also to possible optimization of the cache parameters.

III. EXPERIMENTAL WORK

Four experiment series have been carried out under the same laboratory conditions (hardware, operating system, installed software modules and libraries and running processes):

1. Parsing input and storing data;
2. Sequential reading entire HDF5 dataset to memory object;
3. Reading partial information from the HDF5 dataset;
4. Partial modification of HDF5 dataset.

The experiment variables are described in Table 1.

TABLE I. EXPERIMENT VARIABLES

	Experiment Variables		
	Variable	***Description***	***Domain of possible values***
1.	Data type	Type of data stored in the array	integer or double
2.	Size of array	Number of elements in the array	**Rank 1:** 1 to 99 with step 1 and from 100 to 100100 with step of 500. **Rank 2**: 1 to 99 with step 1 and from 100 to 500 with step of 10.
3.	Rank of array	Rank of array with data	1 and 2
4.	Chunk size	Size of chunk	1 to 100
5.	Chunk shape	Rectangular or square	**Rectangular:** [1,x], where x=[1..100] **Square:** [x,x], where x=[1…100]

For testing purposes, a dedicated Linux application has been developed in C++ language and compiled as 64-bit executable with the latest available HDF5 static libraries (hdf-1.8.11). All tests were automated and with no user interaction.

The */usr/bin/time* Linux command was used to measure the execution time. This command runs the specified program with the given arguments. When the execution finishes, *time* writes a message to the standard output giving time statistics about this program run. These statistics consist of the elapsed real time between invocation and termination, the user CPU time (the sum of the *tms_utime* and *tms_cutime* values in a *struct tms* as returned by *times()*), and the system CPU time (the sum of the *tms_stime* and *tms_cstime* values in a *struct tms* as returned by *times()*) [9]. The study can be further extended with direct analysis of the IO operations performed by the HDF5, but this requires complex modification and new build of the HDF5 library. The input data for the tests represents results from a real-world mechanical and heat transfer simulations.

Fig. 3 shows the workflow for analyzing the test result data and observations.

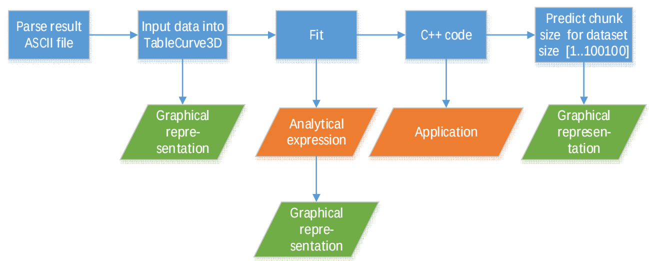

Figure 3. Analysis of the test result data

The two most important deliverables from the analysis are the analytical expression for predicting the optimal chunk size and the derived software application or libraries based on this model.

IV. DISCUSSION OF EXPERIMENTAL REULTS

A. *One Dimensional Integer Array*

Fig. 4 shows the results from automated writing test (data storage) with one dimensional array of integer data type with varying number of elements according to Table I.

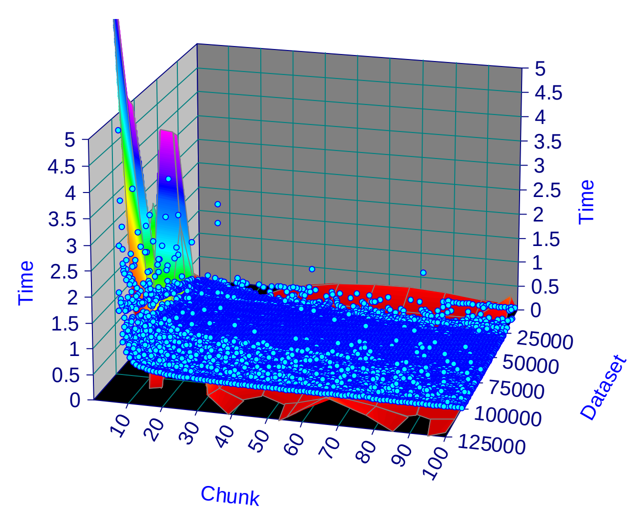

Figure 4. Results from writing one dimensional integer array (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

The resulting data represent a surface with higher values of analyzed parameter (real time) close to a small chunk size regarding the size of the dataset. This result is expected and the chunk size must be kept small in order to optimize the performance of the software modules and the cache that are used to store the data into HDF5 file. This will lead to smaller memory footprint but shall increase the number of I/O operations for large dataset or frequently modified data.

When using the HDF5 libraries there is a certain amount of overhead associated with finding chunks. When chunks are made smaller, there are more of them in the dataset. When performing I/O on a dataset, if there are many chunks in the selection, it will take extra time to look up each chunk. Moreover, since the chunks are stored independently, more chunks results in more I/O operations. The additional metadata necessary to locate the chunks also increases the size of the file as chunks are made smaller. In the most cases, making chunks larger shall result in fewer chunk lookups, smaller file size, and fewer I/O operations [6]. Here should be mentioned that the chunk size of 1 and the chunk size equal to the size of the dataset shall not be recommended by the HDF5 developers [7].

The distribution of the test data in Fig. 4 confirms the hypothesis that the chunk size have to be adaptive and the optimal value for the corresponding size of the dataset should be calculated before imitating the writing functions in the HDF5.

The results from the sequential read test for one dimensional integer array are presented in Fig. 5.

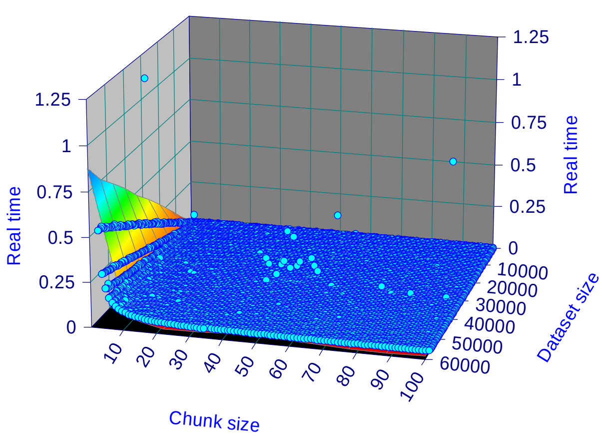

Figure 5. Results from sequentialy reading one dimensional integer array (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

Within the tests for partial reading (Fig. 6) and modifying (Fig. 7) 1% of the dataset is retrieved and altered. If the dataset size is less than 1, only one record is used.

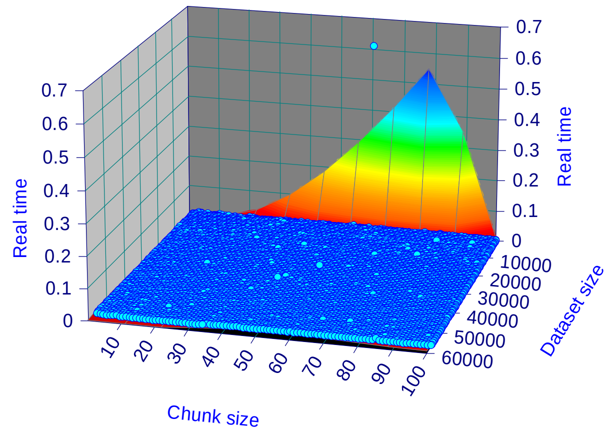

Figure 6. Results from partial reading of 1% of the size of one dimensional integer array (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

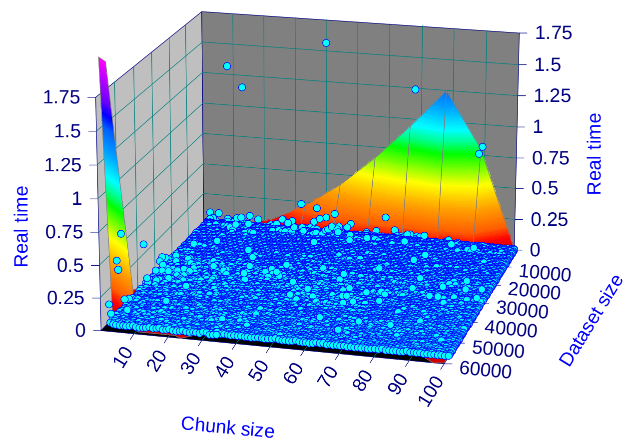

Figure 7. Results from modify test for one dimensional integer array (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

As can be seen from Fig. 4, Fig. 5, Fig. 6, and Fig. 7 the write test is the slowest test (in terms of real time parameter).

Another observation is that the chunk size has greater influence on the data storage than the reading and modifying tests.

B. *Two dimensional double array*

The same four experiments have been performed for two dimensional array of double data type. The results from the write test are shown in Fig. 8.

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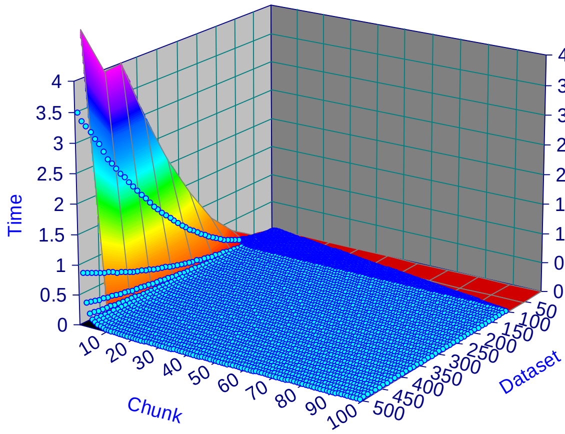

Figure 8. Results from writing a two dimensional double array (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

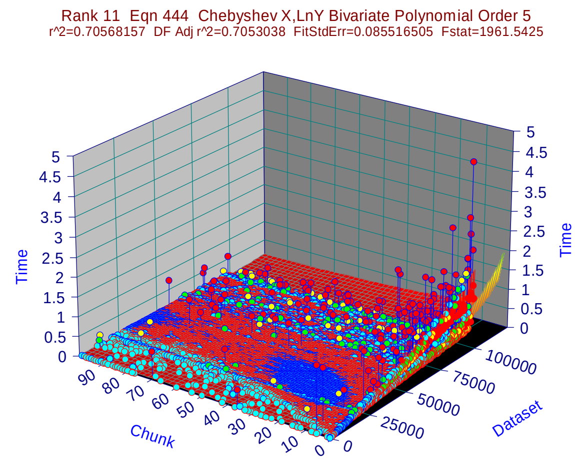

Figure 9. Results from approximation of experimental data for writing a one dimensional integer array (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

The tests for sequential and partial reading and the test for modifying show simmilar results in comparison to the resulst obtained for one dimensional array. Here, the slowest test again is the test for writing the datasets into the HDF5 file.

V. Predicting Chunk Size by the Expected Amount of Stored Information

The experimental results show that optimizing the chunk size for writing datasets in HDF5 file shall have positive influence on the overall performance of the data storage system integrated in a VR application.

The observed data from the automated tests is analyzed and approximated using Sigmaplot TableCurve3D software package. Several approximation functions were tested. For further use the Chebyshev series X, lnY Bivariate Order 5, which gives very good results for all analyzed data within the test cases.

$$
\begin{aligned}
&T_n(x') = \cos(na\cos(x')) \\
&z = a + bT_1(x') + cT_1(y') + dT_2(x') + \\
&+eT_1(x')T_1(y') + fT_2(y') + gT_3(x') + hT_2(y')T_1(y') + \\
&+iT_1(x')T_2(y') + jT_3(y') + kT_4(x') + lT_3(x')T_1(y') + \\
&+mT_2(x')T_2(y') + nT_1(x')T3(y') + \\
&+oT_4(y') + pT_5(x') + qT_4(x')T_1(y') + \\
&+rT_3(x')T_2(y') + sT_2(x')T_3(y') + tT_1(x')T_4(y') + uT_5(y')
\end{aligned}
\qquad (1)
$$

A. *Approximating One Dimensional Integer Array*

The approximation with the Chebyshev series for the write experimental data is presented in Fig. 9.

The resulting polynomial from (1) and the calculated parameters are used to create a C++ function for estimating the optimal chunk size based on the expected number of records in the dataset. The calculated values from calling the chunk size prediction function for dataset with size from 1 to 100100 is given in Fig. 10.

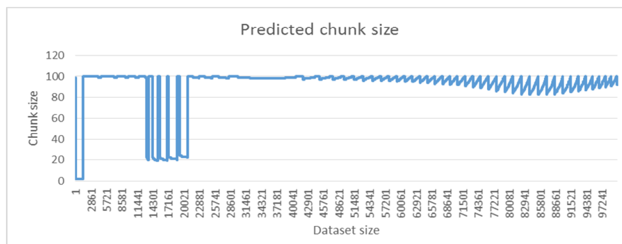

Figure 10. Predicted chunk size for optimal writing of one dimensional integer array (x – size of the input data, y – predicted size of the chunk)

The distribution of the suggested chunk size within range {1-100} is shown in Fig. 11.

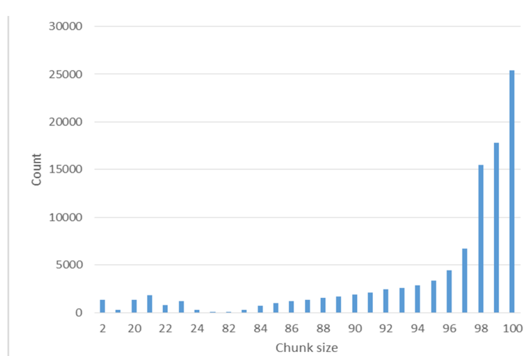

Figure 11. Distribution of the predicted chunk size for optimal writing of one dimensional integer array (x – predicted size of the chunk, y - count)

B. Approximating Two Dimensional Double Array

The same approach for results analysis used for one dimensional array is applied. The surface fit with Chebyshev series is shown in Fig. 12.

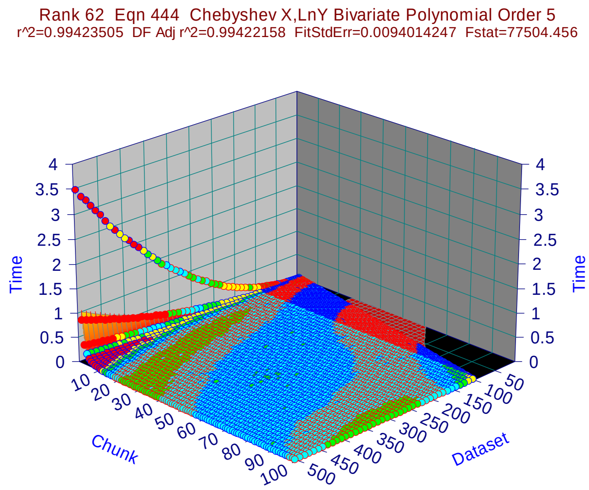

Figure 12. Results from approximation of experimental data for writing a two dimensional array of double data type (x – chunk size [number of elements], y – real time [s], z - size of dataset [number of records])

The predicted value for chunk size for dataset with 1 to 500 records for the two dimensional array is summarized in Fig. 13.

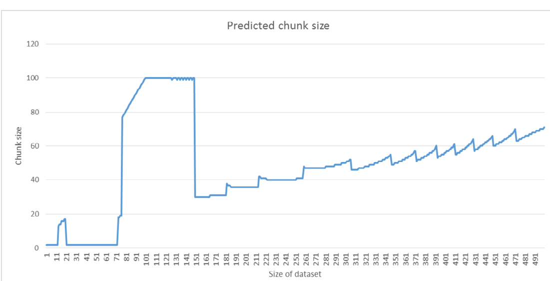

Figure 13. Predicted chunk size for optimal writing of two dimensional double array (x – size of the input data, y – predicted size of the chunk)

The distribution of the predicted chunk size values can be obtained from Fig. 14.

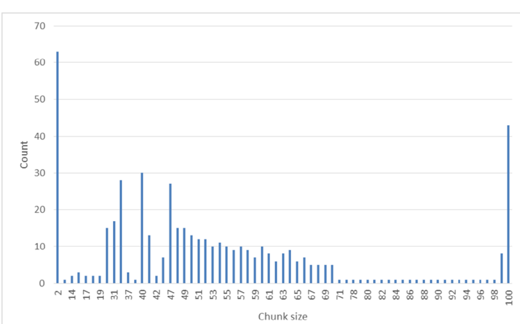

Figure 14. Distribution of the predicted chunk size for optimal writing of tow dimensional integer array (x – predicted size of the chunk, y - count)

A separate experiment has been carried out Based on the function for predicting the chunk size for two dimensional double array.

The results from 1022 executions of HDF5 dataset (two dimensional double array containing random values) write function with random sizes of the chunk and the dataset are compared to the same number of write operations with predicted chunk size. The graphical representation of the results can be obtained from Fig. 15.

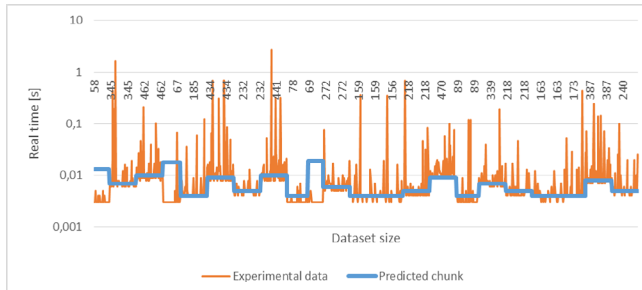

Figure 15. Comparing the real time for 1022 writes of two dimensional dataset with random values for chunk and dataset sizes with real time observed with predicted chunk size

The real time needed to finish all 1022 write cycles with random parameter values within the test is 19.15 s.

The same experiment but using the predicted chunk size took time of 7.14 s.

VI. CONCLUSIONS AND OUTLOOK

The chunk size has a significant influence on the performance of the HDF5 operations and since chunking is obligatory for data compression and for using hyperslabs finding the optimal chunk size for specific dataset is important task.

Based on the analysis of the experimental data an analytical model for surface and function approximation has been derived. This model is based on Chebyshev series X, lnY Bivariate Order 5.

e analytical equations have been used for nt of software functions for predicting the chunk size based on expected dataset size, reading the entire dataset in sequential and partial orders and for modification of the records in a dataset.

The proposed approach for chunk size prediction has been successfully validated by synthetic tests with network and local data storage. The overall performance improvement is of 65,56% (based on most important task – data storage).

A good practice is to predict the chunk size based on the amount of data before creating the dataset. Each dataset in HDF5 database should have the appropriate size of the chunks.

Using the HDF5 for data storage and retrieval has several benefits like very high performance when working with large datasets of complex data types, parallel access to the data (requires MPI and parallel file system) and the predicted optimal chunk size based on the expected amount of data can significantly improve the performance. All these makes HDF preferred over other SQL and NoSQL DBMSs when developing VR applications, especially if the objects are enhanced with implicit features.

The study can be extended to analyze the relation between the chunk size, chunk cache options and the overall I/O performance. The results could be used to derive a model for predicting their optimal values based on the size of the input data set.

Another important further study is to perform experiments with dataset dimension larger than 2 and with complex data types.

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A New Method for Haar-Like Features Weight Adjustment Using Principal Component Analysis for Face Detection

Ramiro Pereira de Magalhães and Cabral Lima
Federal University of Rio de Janeiro
Rio de Janeiro, Brazil
e-mail: ramiro.p.magalhaes@gmail.com, cabrallima@ufrj.br

Abstract—**This paper proposes a new weight assignment method for Haar-like features. The method uses principal component analysis (PCA) over the positive training instances to assign new weights to the features. Together with the method, a particular Haar-like feature that uses statistics extracted from positive training instances is employed. The method and the Haar-like feature were designed to verify if the distribution of points produced from the negative instances in the single rectangle feature space (SRFS) of each Haar-like feature could be modeled as an uniform distribution. Although negative instances may spread themselves in very different and chaotic ways through the SRFS, experiment with the method and the Haar-like feature has shown that the negative instance cannot be properly modeled as an uniform distribution.**

Keywords-pattern detection; Viola-Jones framework; Adaboost; Haar wavelet; principal component analysis

I. Introduction

Object detection is the task of automatically discovering the presence and location of a particular object in an image. It is usually the first step for additional processing over the target object. The detection of human faces in complex scenes, required for several applications, is a complex problem and has been the main subject of several researches, many of them surveyed by Zhang and Zhang [1]. The Viola-Jones framework [2] is probably the most known method to deal with this problem. The use of Haar-like features to develop an accurate frontal face classifier makes the process fast enough to detect objects in real time video. Some researches tried to improve the speed, accuracy and robustness of such approach [3] [4] [5] [6]. For instance, Pavani et al. [7] established that it is possible to assign better weights to Haar-like features by interpreting them as the inner product of the weights versus the rectangular areas average values. Their experiments showed superior results if compared to many other relevant works.

This paper, based on the approach of Pavani and colleagues, presents a new method for assigning weights to Haar-like features. PCA is used to find a vector of weights that befittingly identifies the positive training instances. This simple and fast method may be complementary to Pavani's approach and has the advantage that it can be applied on a step preceding the classifier boosting. This provides some relief to the boosting process, known as a lengthy phase. During the PCA processing, some statistics are extracted from the positive instance dataset in order to be employed in a new classifier similar to the one suggested by Landesa-Vazquez and Alba-Castro [5].

In this paper, Section II introduces the processes and frameworks used in the development of a face detector as shown in [2]. Section III reviews some recent researches that brought interesting ideas and enhancements to such detectors. Section IV details the main contributions of this paper. In Section V some experiments using the proposed method are detailed. Section VI concludes this paper and presents some future works.

II. The Viola-Jones face detector

In this section, Viola-Jones' face detector building blocks are described. Notions of boosting and the structure of the face detector along with the functions used to extract features are also presented.

A. Boosting

Boosting is a machine learning technique based on the idea that it is possible to form an accurate classification rule (named strong classifier) by merging many inaccurate classification rules (the weak classifiers). The most known boosting algorithm is Adaboost [8]. Roughly, a boosting algorithm must show the input, a labeled dataset with positive and negative instances, to another algorithm (generically named weak learner) pointing out that some instances are more important to be accurately classified than others. With this information, the weak learner must choose a weak classifier that best labels the input dataset while considering the importance of each instance. After that, the importance of each instance is updated with aid of the recently produced weak classifier, which is then pushed into the strong classifier along with a rating of its classification capability. This whole loop then is repeated for a certain amount of iterations until the strong classifier with its boosted weak classifiers reaches some stop criteria.

A boosting algorithm usually receives as input a set of m labeled instances $(x_1, y_1), \ldots, (x_m, y_m)$ where $x_i \in X$ represents the objects to be classified; and $y_i \in Y = \{-1, +1\}$ is the set of possible classes, where $+1$ indicates that the object belongs to the desired class and -1 the opposite. The main objective of a boosting algorithm is to generate a strong classifier $H : X \mapsto Y$ composed by some weak classifiers $h_t(x)$, where $t = 1, \ldots, T$ means the iteration in which the weak classifier was generated. For each iteration the boosting algorithm invokes another algorithm, generically referred as weak learner, that is responsible to produce the weak classifiers

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Input: $(x_1, y_1), \ldots, (x_m, y_m)$ where $x_i \in X$ e
$y_i \in \{-1, +1\}$
begin
Set $D_1(i) = 1/m$ where $i = 1, \ldots, m$;
for $t = 1, \ldots, T$ **do**
Provide D_t to the weak learner and from it get
$h_t : X \mapsto \{-1, +1\}$, such that h_t minimizes
$\epsilon_t = Pr_{i \sim D_t}[h_t(x_i) \neq y_i]$
Let $\alpha_t = \frac{1}{2}\ln(\frac{1-\epsilon_t}{\epsilon_t})$
for i = *1, ..., m* **do**

$$\begin{aligned} D_{t+1}(i) &= \frac{D_t(i)}{Z_t} \times \begin{cases} e^{-\alpha_t} & \text{if } h_t(x_i) = y_i \\ e^{\alpha_t} & \text{if } h_t(x_i) \neq y_i \end{cases} \\ &= \frac{D_t(i) exp(-\alpha_t y_i h_i(x_i))}{Z_t} \end{aligned}$$

where Z_t is a normalization factor chosen
so that D_{t+1} is a distribution.
end
end
return $H(x) = sign(\sum_{t=1}^{T} \alpha_t h_t(x))$.
end

Figure 1. The Adaboost algorithm used to boost a set of weak classifiers into a strong classifier.

$h_t(x)$ which will be added to the strong classifier. Figure 1 shows Adaboost, slightly adapted from [9].

B. Face detector

The goal of a face detector is to determine the presence and location of faces in an arbitrary image. If they exist, then the detector should also be able to determine the region they occupy in the image [1]. This has been seen as a challenging task for a machine due to the enormous variety of human skin, hair, eye colors, texture, facial features, accessories, expressions, rotations, and even environment lighting conditions.

The powerfull and fast face detector proposed by Viola and Jones in [2], and revised in [10], operates by classifying the contents found inside a window positioned over the image. This window slides in the vertical and horizontal directions until the whole image has been scrutinized. This process may repeat with windows having different sizes as first shown by Rowley et al. [11]. Such “sub-windows” form an overcomplete set of the examined image, but very few of them contain a face. Therefore, a detector that thoroughly inspects every sub-window consumes a lot of time evaluating background scenes in order to find a single face. To deal with this problem, Viola and Jones proposed that the final classifier should be built like a chain of increasingly complex strong classifiers. Each node in this chain should reject many background objects (around 50% or more) while rejecting very few faces (preferably none). This “rejection cascade”, originally proposed by Baker and Nayar [12], allows the quick discarding of uninteresting sub-windows because it is enough that a single node rejects the input for it to be classified as background (Figure 2).

The whole chain is the result of a bootstrapping process. To each node a threshold of maximum false positive and true positive rates are set. Similarly, a maximum false positive rate is also set to the whole chain. Positive and negative training instances are provided to Adaboost that will iterate as much as needed to reach the node thresholds. Once a node is ready it is added to the chain that is then tested for its maximum false positive threshold. If the threshold of the chain has not yet been reached, a number of false detections made by the chain over the negative instance set are then used to boost the next node. The set of positive instances always remains the same. Through this process the nodes closer to the end of the classifier will be trained with “harder” instances, hence they will be more complex, i.e., they will have more weak classifiers and be more precise.

C. Haar wavelets as a weak classifiers

A Haar wavelet is a function proposed be Alfred Haar [13] to transform a signal in a simpler (or more meaningful) representation to certain analysis procedures. Papageorgiou et al. [14] created a feature extractor that uses Haar wavelets to encode local differences of pixels in images. This Haar-like feature is a value in $\mathbb{R}$ obtained from the weighted sum of pixel intensities contained in the d rectangular regions of the Haar wavelet, where each region is associated with a weight $v \in \mathbb{R}, v \neq 0$. Usually, the weights of a Haar-like feature add up to 0, and are proportional to the amount of pixels contained in the rectangle that they refer to. Considering w a Haar wavelet, r a rectangular region of w, and l the pixels contained in r, it is possible to establish:

$$f(w) = \sum_{i=1}^{d} v_i(\sum_{l \in r_i} l). \tag{1}$$

The weak classifiers proposed by Viola and Jones [2] use such features. In fact, they are a function $h(x, f(w), p, \theta) \mapsto \{-1, +1\}$, where the value $+1$ means that the object belongs to the class of interest, and -1 the opposite. Considering $p \in \{-1, +1\}$ the polarity (or parity), θ a threshold, and x an image sub-window, [2] established:

$$h(x, f, p, \theta) = \begin{cases} +1 & \text{if } pf(x) < p\theta \\ -1 & \text{otherwise} \end{cases}. \tag{2}$$

p simply affects the orientation of the comparison. Sometimes, $h(x, f, p, \theta)$ is defined to return 0 instead of -1. The value used in this paper keeps (2) consistent with Algorithm 1.

Since its proposition, researchers are trying to improve the Haar-like features. Besides extending Papageorgiou and

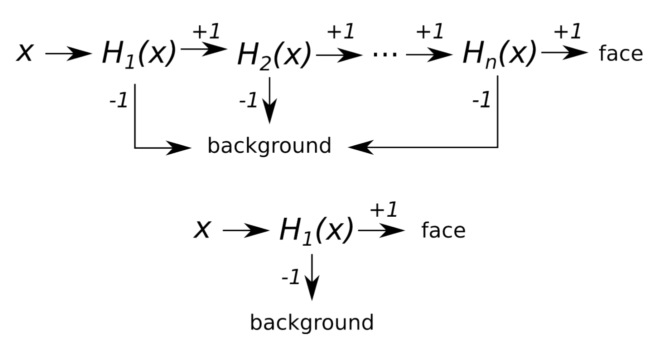

Figure 2. Rejection cascade composed of strong classifiers $H_i(x)$ (above) compared with a monolithic classifier (below).

colleagues set of features, Viola and Jones [2] developed a method to calculate any Haar-like feature value in constant time. Lienhart and Maydt [15] proposed $45°$ rotated features, also calculated in constant time. Disjoint rectangles, a more general way to produce features, were introduced by Li et al. [16]. Later, Viola and Jones [17] showed a set of "diagonal features". Figure 3 shows some examples of such features. A collection of other researches about this topic can be found in [1].

III. Related works

This section presents a review of some researches dealing with accuracy, performance and training time of strong classifiers through manipulation of Haar-like features.

Dembski [3] presents the result of some experiments carried out with the Lienhart and Maydt's extended feature set [15]. The main goal of this research was to verify if there is some pattern in the contribution of the features found in a strong classifier, i.e., to find if there is a set of features more useful than others. He demonstrated that the line features (rotated or upright) provide a better generalization error than both the center-surround and the border ones. Dembski compared the horizontal and rotated features and established that the latter generalize better than the former. He observed that larger features perform better than the smaller ones. Nevertheless, the generalization differences among the compared features are small, so it is possible that Dembski's results do not hold in other experiments.

In Baumann's work [4], a modification in the Adaboost algorithm to explore the human face symmetry was proposed. In their experiments, the time taken to boost a strong classifier was reduced by almost 40%, due to the selection of two weak classifiers per round instead of just one as usually occurs. The first classifier is chosen using the normal procedure [8] and a second symmetric feature is chosen and placed in a symmetric region of the sub-window, but its final position will still be target of a search in the close neighbouring area.

Landesa-Vázquez and Alba-Castro [5] developed a weak classifier slightly different from the one proposed by Viola and Jones [2]. Motivated by physiological studies on human vision, they modeled an apolar weak classifier that considers the Haar-like feature's absolute value. They compared their strong classifier with Viola and Jones and, although they did not observe any change in the detector precision, their final cascade had much less weak classifiers.

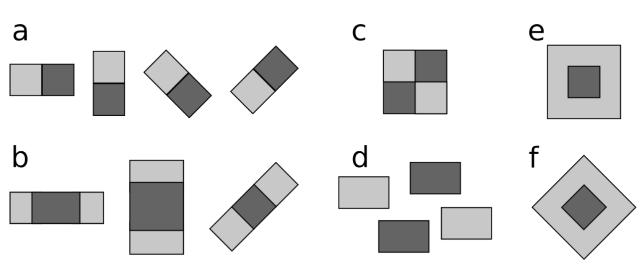

Figure 3. Examples of Haar-like features: (a) border; (b) line; (c) 4-dimensional feature proposed in [14]; (d) disjoint rectangles proposed in [16]; (e) e (f) center-surround features. Darker regions have different weight than the lighter ones.

Vural and colleagues [6] proposed a new set of Haar-like features with a very different composition of rectangular regions, and able to rotate in six angles. The upright features that serve as template for the rotated versions, were automatically generated through an iterative procedure that first adds a single rectangle and then evaluates the feature performance. Only those with the smallest error rate participated on the classifier boosting rounds. As a result, only a quite small amount of features, if compared to other researches, was used in the boosting rounds. This not only sped up the boosting procedure but also reduced the amount of features found in the final detector.

Pavani et al. [7] argued that the weights typically assigned to rectangles of a feature are suboptimal. They demonstrated this through the introduction of the SRFS, where vectors s of d dimensions contain the averages $s_i, i = \{1, \ldots, d\}$ of the pixels contained in each one of the Haar-like feature's rectangles. This is the linear algebra interpretation of the feature value calculation, as shown in (3):

$$f(w) = \sum_{i=1}^{d} v_i (\sum_{l \in r_i} l) = \sum_{i \in w} v_i s_i. \tag{3}$$

Therefore, $f(w)$ is the result of the inner product of s by the weights vector v, i.e., a Haar-like feature projects s in the direction of v.

Pavani evaluated the distribution of vectors in the SRFS. For some Haar-like features w they generated a set of vectors S_w^+ using only the positive training instances, and did the same to the negative instances, creating the set S_w^-. He established that S_w^+ results in a very concentrated point cloud, while S_w^- shows a much more varied spread. Since those classes spread over the SRFS in very particular ways, Pavani and collaborators observed that the projections made with typical values of v may not help to discern between classes as they should. For instance, consider a Haar-like feature w' with two rectangles ($d = 2$) and weight vector $v' = \{-1, 1\}$, and assume that the SRFS $S_{w'}^+$ of w' spreads like a bivariate Gaussian distribution, with the highest variance axis parallel to v', as seen in Figure 4. In this case, points in $S_{w'}^+$ are projected in the direction of v' mixing themselves more often with the $S_{w'}^-$ set. If v' is perpendicular to the highest variance axis formed by $S_{w'}^+$ distribution, then this mixture will be less frequent, and the Haar-like feature will be more discriminative. Pavani proposed then the optimization of vector v of all candidate Haar-like features, and used three different methods to this effect: brute-force search, genetic algorithms and Fisher's linear discriminant analysis (FLDA). These methods not only optimize v, but also select during boosting the parameters p and θ with the smallest classification error.

The three resulting detectors were tested and compared among themselves and with the ones of [2] [18] [19] [20]. The most accurate (given by the area over the ROC curve) was the genetic algorithm optimized one, although it took 20 days to be trained [7]. This detector was also considered the fastest since, in average, rejects more negative samples using less nodes of the classifier cascade.

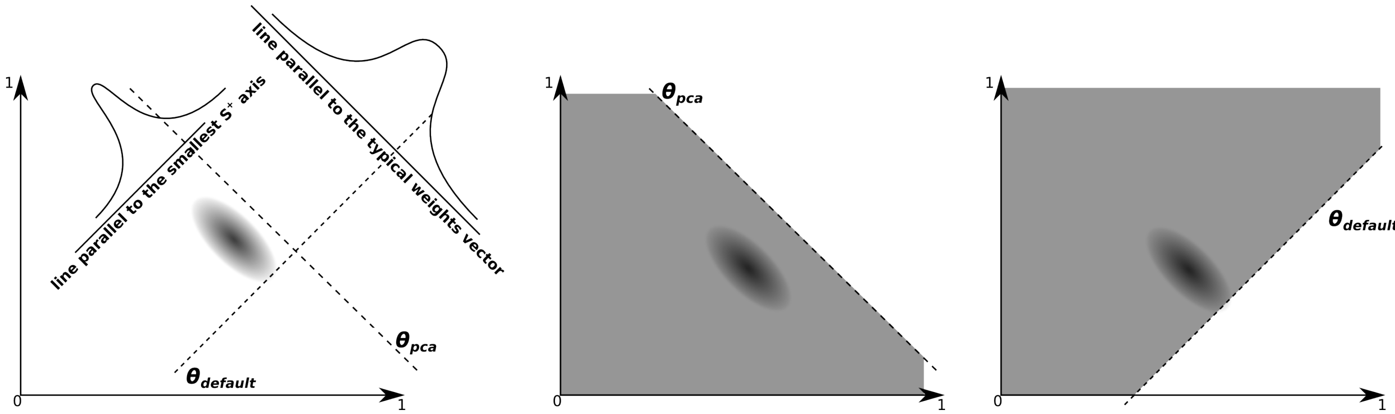

Figure 4. The left image shows a set of points in the SRFS formed only by faces. Projecting a point in the SRFS in a parallel direction to the optimal v yields a higher concentration of points.The application of the classifier threshold θ splits the SRFS in very distinct ways, as the remaining images show.

IV. A NEW OPTIMIZATION METHOD

In this section, an optimization method complementary to those shown in [7] is proposed. Additionally, Pavani's results are discussed in order to properly motivate the ideas used in the new method.

As shown in Section III, Pavani's results suggest that (a) the distribution formed by the face and background feature points in the SRFS do not have the same covariance; or (b) the spread of S^+ or S^- is not properly described by the Gaussian distribution. It is possible to reach such conclusions by recalling that FLDA is particularly effective when both classes it tries to separate behave as Gaussian distributions with the same covariance [21, p. 120]. In addition, through the analysis of data shown in [7], even though it looks like that S^+ spread can be modeled through the Gaussian distribution, the same seems hard to be stated about S^-. These observations were described in [7]. In fact, up to the moment this paper is written, only a few other researches considered Pavani's findings [22] [23]. Therefore, it is hard to state for sure how face and background feature values produced from the most used Haar-like features are spread over their respective SRFSs.

This research aims to provide some additional information about how features values are laid out in the SRFS. Admitting that a Gaussian distribution befittingly models S^+_w distributions, it is intended to verify if a uniform distribution better models the S^- spread. This assumption might seem naive, but it should be verified because a simpler and faster Haar-like feature weight assignment procedure could be employed if the assumption holds true. Hence, the method proposed here uses PCA to assign weights to each Haar-like feature w from S^+_w's principal component of least variance. To explore even further the fact that S^+_w is highly concentrated around its mean, a weak classifier similar to the one proposed in [5] is used. The method and the classifier are detailed in Section IV-A.

The proposed method indeed reduces the total training time. While FLDA is a fast method if compared to brute-force search or genetic algorithms, it needs to estimate the average and variance of both sets S^+_w e S^-_w in order to obtain the inter and intra-class spread. PCA is less complex than FLDA and, in this case, is applied only over S^+_w. Another important aspect that would also reduce the total training time is the moment when v optimization occurs. In [7], an optimization method must be invoked at least once per node of the chain of classifiers, after all, as shown in Section II-B, the negative instances set change between each cascade node boosting run. In the particular case of the FLDA optimization, each feature must be optimized once per node. The method proposed here allows the pre-optimization of the weak classifiers prior to boosting them. This is possible because the set of positive instances remains the same throughout the whole chain of classifiers construction process, so it is unnecessary to recalculate each feature's weight when the construction of a new cascade node begins.

Vural et al. [6] described an iterative construction technique of features. In the method proposed here, the features are chosen following Pavani's rules, as described in Section V-B. Through these rules, neither the rotated features, neither the center-surround are used, what conforms with Dembski's [3] considerations (see Section III).

A. *The proposed feature*

Let μ be S^+ mean. The following feature is proposed:

$$f'(w) = |\sum_{i \in w} v_i(s_i - \mu_i)|. \tag{4}$$

Combined with Viola-Jones' weak classifier threshold, this feature effectively creates a "band" over the SRFS perpendicular to v, that passes through μ, and has 2θ width. Figure 5 illustrates this.

The insight behind this features and classifier combination is very simple: the "band" should cover the maximum amount of points of S^+ and the minimum of S^-. It is important to note that S^- is assumed to be uniformly distributed, and S^+ spreads like a multivariate Gaussian distribution (Section IV). Hence, by projecting S^+ in the direction parallel to its axis of smallest variance, it is possible to get the highest concentration of projections of points of this set, therefore the smallest possible θ value. μ's value is set during the PCA execution together with v. Similarly to [10], p and θ is assigned by the weak learner during the boosting phase.

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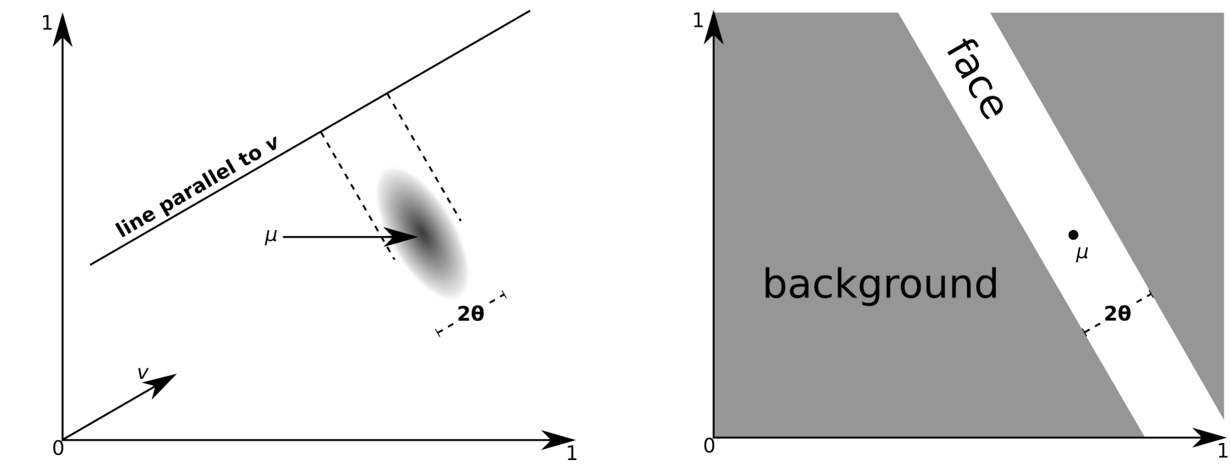

Figure 5. Representation of the effect caused by the proposed feature when combined with a weak classifier in an hypothetical SRFS. To the left, it is represented by a point cloud S^+, its mean μ, and θ. To the right, the effect of the classifier creates on that space if $p = +1$.

B. Feature parameters selection

While running PCA, both v and μ are set for each Haar wavelet. To achieve this, the first step is to produce the SRFS S_k^+ for each Haar-like feature k. Then, for each feature, the mean and the covariance matrix Σ_k^+ are estimated. While the mean is attributed to μ, Σ_k^+'s eigenvector of smallest eigenvalue is calculated and assigned to v_k. v_k must be normalized. This procedure is shown in Figure 6.

V. Experiments

This research hypothesis was experimentally verified. In order to do this, three monolithic strong classifiers (each one with 200 weak classifiers) were boosted. The first of them is similar to Viola and Jones' classifier; the second had only its weights assigned via PCA; and the third used the feature proposed in this paper, as seen in Section IV-A, with the relevant parameters set with the procedure shown in Section IV-B. Although the weights must be different, the rectangle templates used for each Haar wavelet were the same. Additionally, the same face and background images were used to boost all classifiers.

Input: $x_i \in X^+$, the set of positive instances.
Input: $w_k, k = 1, \ldots, K$, the Haar-like features.
Output: v_k, the new weight vectors w_k.
Output: μ_k, the mean of S_k^+.
begin
  foreach w_k **do**
    Use w_k to produce S_k^+ for every X^+
    Estimate μ_k and Σ_k, respectively mean and covariance matrix of S_k
    Find Σ_k eigenvalues and corresponding eigenvectors.
    Assign to v_k the eigenvector of smallest eigenvalue of Σ_k
  end
end

Figure 6. Assignment of v_k and μ_k to Haar-like feature k using PCA.

A. Positive and negative instances

Four different available face databases were used to create the positive instances dataset needed for all training procedures: the MIT-CBCL Face Database #1 [24]; the BioId face database [25]; the FEI Face Database [26]; and the AR Face Database [27]. A total of 4,938 faces were automatically extracted with programs especially designed to adequate each image to this requirements of the proposed method. Figure 7 shows some faces present in the trainning dataset.

The MIT-CBCL Face Database #1 contains 2,429 images in grayscale and width and height of 19 pixels of faces looking straight forward. It is the dataset that best fits this requirements for the present experiments because it only needed to be rescaled to 20 pixels.

The BioID Face Database contains 1,521 grayscale images with 384 pixels wide and 284 high. In general, subjects in this database are looking to the camera, but show some variations in facial expressions and face rotation, tilt and yaw. A file describing the position of the eyes accompanies each image. By using such descriptor, a program automatically estimated the position and rotation of the face and then aligned it with the horizontal, cut and rescaled the face region.

The original version of FEI Face Database contains color photographs of 640×480 pixels of 200 people (100 of each

Figure 7. Excerpt from the positive instances dataset mixing faces extracted from publicly available datasets.

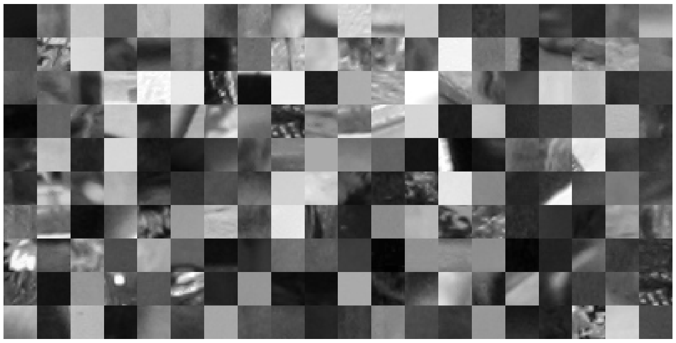

Figure 8. Excerpt of 200 samples from the negative instances dataset used to boost the three strong classifiers.

sex) in 14 different poses and lighting conditions looking straight to the camera in a controlled environment. Some other works were already made over FEI's images, so there are derivations of the original database. For the present experiments, the chosen version contains images in grayscale 250 pixels wide and 300 high of the same 200 individuals looking straight forward but in two poses: relaxed and smiling. The eyes were already aligned as described in [28].

The AR Face Database contains pictures of 126 faces taken on two different days and in controlled conditions. From a single person in a single day, 13 different photographs were taken, each with a particular facial expression, or with the subject wearing a particular accessory, or under certain lighting conditions. All subjects were looking straight forward. Only the subsets 1, 2, 3, 4, 7 and 8 were used. The extraction process is very similar to the one employed in the FEI Database, since these images from the dataset also had the eyes aligned.

Concerning the negative instances, a total of 114,865 samples were taken from around 2,000 digital colored pictures of nature, animals, landscapes, buildings, architecture, paintings, sculptures and people. From those samples, 6,000 were randomly chosen to be used in the classifier boosting procedure. Many of the original pictures had faces which were manually removed with the aid of an application specially designed for this purpose. Other parts of the human body, including hands, hair, feet, clothing and accessories, were not removed. Samples from rough artistic reproductions of the human face were also left in this dataset. Examples of the negative instances can be seen in Figure 8.

B. Haar wavelet set

The Haar wavelets rectangles size and position were chosen according to the same rules mentioned in [7]:

1) only 2 to 4 rectangles can be combined in a Haar wavelet;
2) the template of each Haar wavelet must fit in 20×20 pixels window;
3) rotated features like those proposed in [15] must not be used;
4) distances dx and dy between rectangles, as described in [16], are integer numbers multiples of the rectangle size in the respective directions;
5) all rectangles of a Haar wavelet have the same height and width;
6) the minimum sizes of any rectangle is 3×3 pixels.

By strictly following these rules, a total of 1,641,107 Haar wavelets were generated. From this set, only 218,544 were selected through the application of some additional rules.

C. The face detector operation

The face detector examines the test images, as shown in [10]: moving $[\Delta s]$ pixels in the vertical or horizontal direction, where s is a factor that scales the size of the detector itself. After scanning the whole image, the detector window size is increased by 25%, and the image is scanned again. This repeats until the detector is bigger than one of the image's sides. The initial value of those parameters are: $\Delta = 1.5$ and the initial scale is 1.5. The initial detector sub-window size is 20×20 pixels.

There are some ways to determine if a sub-window had been correctly classified. In [29], a face is considered correctly detected if the detected region contains all face annotations (eyes, nose and mouth) and the size of the detector is smaller than four times the distance between the eyes. In [7], a detection is considered correct when the size of the detected region is $\pm 10\%$ of the annotated face, and when the distance from the center of the detected region to the center of the annotated region is at most 10% of the size of the annotation. In the method proposed here, Pavani's approach is applied. The face region is calculated from the annotated eyes that comes with the test dataset. The height and width of a face region is 1.9402 times the annotated distance between the eyes. The region's top left point was positioned 0.2423 times the region width to the right of the annotated right eye, and 0.25 times the width of the window above the right eye. These were the same parameters used to extract faces from the BioId database. No detection sub-window integration was made.

The detector also does some pre-processing of the image. Viola and Jones' detector performed variance normalization on the image sub-window prior to evaluating them [10], but the authors did not clearly mention the parameters they used. Hence, the normalization implemented for the traditional Viola-Jones' detector was based in [15]. In [7], the images were intensity normalized, i.e., each pixel value was divided by the maximum value they could admit. The detectors that had their v vectors assigned through PCA employed this normalization procedure.

D. Results

The tools, training and testing software developed for this research were written in C++. All image manipulation operations were made with OpenCV [30], although certain image loading procedures were written with OpenImageIO [31]. Some algorithms were easily parallelized with the Intel Threading Building Blocks library [32]. The PCA optimization uses a slightly modified version of libpca [33] and the Armadillo [34] library. Some modules of Boost C++ Libraries [35] were also used for many different tasks.

Three monolithic detectors with 200 features were trained with the same datasets and under very similar conditions. The Haar wavelets rectangles' layouts were all the same, even though its weights were different. More specifically:

1) the first detector used the typical weights assigned to rectangles and was trained and operated exactly as described in [10];
2) the second detector had its weighs set via PCA, but did not use S^+ means for any purpose. Also, the images it scanned were intensity normalized;
3) the third detector was trained as proposed here, with every weight of the Haar-like features set as shown in the Algorithm 6. The feature values were calculated as described in Section IV-A) and images were also intensity normalized.

Detector (1) works as the control experiment while detectors (2) and (3) serve as means to verify the hypothesis. Monolithic classifiers were used instead of a cascade because they suffice to the present research's intention to further investigate the SRFS.

It took 4 minutes to run Algorithm 6 for all the 218,544 Haar-like features, each one consuming the 4,938 positive instances on a Intel Core i7 machine with 4 GiB of RAM memory. This algorithm ran in parallel using all the processor's cores. In the same machine, each boosting procedure took from 9 to 10 hours with the weak learner also running in parallel.

The detectors were tested against the MIT + CMU A, B and C datasets [18] [36], which contain pictures of many subjects whose faces are generally looking forward. A total of 19,024,094 sub-windows were scanned, and the detection acceptance criteria turned the 511 face annotations in 2,571 possible true positive sub-windows. Section V-C describes in details both the scanning method and the acceptance criteria.

The 200 feature detector ROC curve created by altering the detector threshold from $-\infty$ to $+\infty$, as shown in [37], is plotted in Figure 9. Considering the following: a) the three detectors performances; b) all weak classifiers were candidates to be part of the strong classifier in every iteration round; and c) the assumption that the Gaussian distribution is a good model for S^+'s distribution; then it is reasonable to conclude that the uniform distribution does not model adequately how S^- spread over the SRFS. This is an interesting observation since the data available about S^- suggest that it can spread itself in very different and chaotic ways.

The detectors using the proposed method have overfitted: their ROC curves near the perfect classification when tested against the training dataset. An exception to this occurs if the detector had its weights set via PCA and used the proposed weak classifier. In this case, the weak classifiers θ parameter was set to very small values, causing the "band" to be too thin, and allowing too many misclassifications to occur. The main cause of this problem is probably the lack of information about the negative instances in the weak classifiers. Indeed, when assuming that the negative instances behave as uniform distributions in the SRFS, one makes it impossible to use any additional information about the negative instances. On the other hand, the simplicity and speed of the training method proposed here are surely too compelling to be left untested. These observations points future researches towards the usage of weak classifiers that carries with them more information about the distributions of both classes in the SRFS.

It also seems that the rotations made in the BioId dataset and the choice of the AR datasets made the positive instances be too similar to each other, aggravating the overfitting. It is possible that the background instances used to boost the classifiers was relatively small. This observation comes from the comparison of the performance of the original Viola-Jones monolithic detector with the one produced in this work. Additional evaluations are being made in order to create a more diverse face database and to fine-tune the boosting parameters.

VI. Conclusion and future work

In this paper, a new weight adjustment method for Haar-like features (complementary to the ones shown by Pavani and collaborator's [7]) was proposed. This method uses PCA over the positive training instances to assign new weights to the features. It was also proposed the employment of a Haar-like feature that operates with parameters estimated from the same positive instances.

Both the weight assignment method and the Haar-like feature were designed to verify if the distribution of points produced from the negative instances in the SRFS of each Haar-like feature could be modeled as an uniform distribution. The motivation behind this, besides shedding light in a somewhat still unexplored concept, was the simplicity and speed of the proposed weight assignment method.

The weight adjustment method as well as the Haar-like feature were tested in the face detection task and compared with the Viola-Jones detector. Although the negative instances may spread themselves in very different and chaotic ways through the SRFS, the obtained results suggest that they cannot be properly modeled as an uniform distribution. This interesting finding is the most important contribution of this paper.

The future works concern to evolve the methods and classifiers to more complex ones while still exploring and bringing understanding about the SRFS. Experiments with other processes of feature weight adjustments using information of both positive and negative training instances are ongoing.

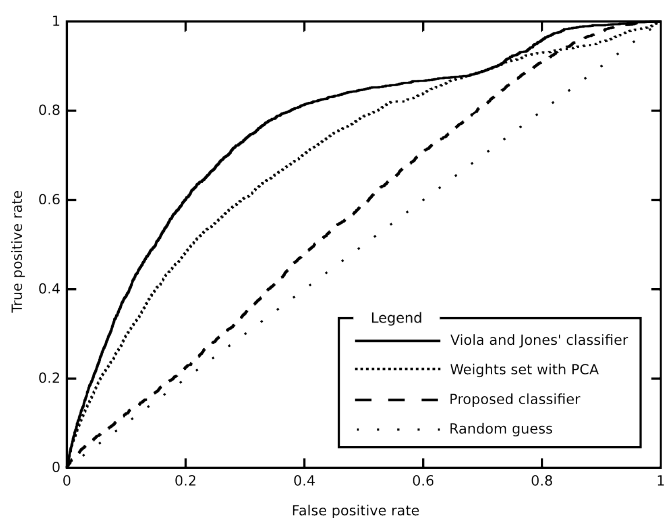

Figure 9. ROC curves of the detectors when tested with the MIT + CMU dataset. The vertical axis shows the true positive rate and the horizontal axis shows the false negative rate.

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Recognition of Haemolytic Transfusion Bags

Petra Rajmanová
Faculty of Electrical Engineering and Computer Science
VŠB - TUO
Ostrava, Czech Republic
petra.rajmanova@vsb.cz

Pavlina Núdziková
Faculty of Electrical Engineering and Computer Science
VŠB - TUO
Ostrava, Czech Republic
pavlina.nudzikova@vsb.cz

***Abstract*—This paper is about automatic recognition of haemolysis in transfusion bag. Currently, haemolysis in blood bag is recognized by a visual estimation. Nurses compare the colour of blood plasma with reference samples and according to their own opinion they evaluate whether blood plasma is damaged or not. The parameter for the comparison is colour. The best is to use image sensors – camera or video camera for analysis. Software that is designed in MATLAB programming environment is able to evaluate colour of blood plasma according to the reference samples. Determination of colour is done in three colour spaces, namely, RGB, Lab, and xyY.**

Keywords-method; colour recognition; haemolysis; blood plasma; references samples; camera.

I. Introduction

With the growth of treatments, there is increasing need for donated blood. This donated blood cannot be artificially manufactured or substituted. This process is indispensable for medicine. Blood may be obtained in two ways. The first possibility is to take of full blood with components. The second possibility is a targeted sampling of individual blood components. All the blood is processed into erythrocytes, plasma, and leukocytes. In plasmapheresis the donors took off only blood plasma and other blood components are returned to their body. During the sampling and subsequent processing of blood plasma [1, 2], it is possible to damage blood plasma, which is called haemolysis. It is impossible to use such plasma for next processing.

II. Taking of Blood and Processing

Donation of blood is a process. The blood is used for medical processing. There are two ways of taking of blood. The first possibility is to take of full blood with all components. The second possibility is a targeted sampling of individual components. The blood is taken into plastic bag. The plastic bag has anticoagulable solution inside. Individual bags are connected by tubes. The tubes have needles at the end.

A. *Taking of full blood*

Every time, the taken blood is processed on erythrocytes concentrate, on blood plasma, and on leukocytes. Full blood is divided in centrifuge. Blood is separated in individual components by special blood press; see Figure 1. Blood plasma is in upper part of blood bag. A buffycoat is in the middle of the blood bag. Erythrocytes are in lower part of transfusion bag.

Figure 1. Blood press.

Between bags is safeguard that has to be broken. After that is possible to separate blood components. Completed bags are stored in special fridge or freezer.

B. *Target sampling*

Target sampling is specific method of taking of blood, especially when it is donated only by one blood component. It is called trombocytopheresis, erytrocytapheresis, plasmapheresis.

III. Haemolysis

Erythrocytes are very sensitive. The membrane can be damaged by some physical and chemical agent. When the membrane is damaged haemoglobin flow out of the cell to blood plasma. This phenomenon is called haemolysis. This may occur for example during processing of blood plasma when it is centrifuged at high speed, after premature centrifugation or wrong break out of fuses or due to thermal effects [3]. When the fuse is broken wrongly, it is due to sharp edges. If the blood components are separated, erythrocytes can be damaged by these sharp edges. In current practice, the haemolysis in blood plasma bag is subjective analysis [4].

IV. Designed Method

Currently, haemolysis is recognized by comparison of blood plasma with reference samples; see Figure 2. This is done by nurses in laboratory. This method is very unreliable because human eye has different perception of colour and of brightness.

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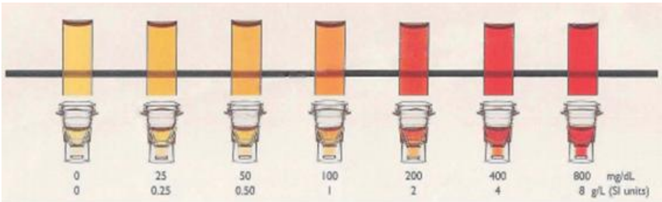

Figure 2. Colour scale for assessing degrees of haemolysis, expressed in g/l [mg/dl] haemoglobin.

Blood plasma can be used in hospital or it can be shipped to industrial factory for processing. Transfusion bags are checked between their usage; mainly, on colour, barcode or the integrity of the bag.

A. *Choice of method*

Haemolysis can be recognized by two methods. Spectrometry is recognition by optical method. Spectrometry gives only information about wavelength characterizing the colour of the plasma. Optical sensors are better for recognition of colours. Colour can be recorded by camera or video camera. Information is summarized on the image. This picture is processed on computer. Haemolysis is detected by the algorithm in Section C. Besides recognition of colour, there is possible to get information about barcode and check integrity of the bag.

B. *Block diagram*

The base of method is to design a measure chain; see Figure 3. It consists of light source, blood plasma, recorder and processing on computer.

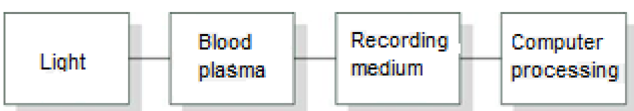

Figure 3. Block diagram for analysis of blood plasma by image sensor.

Light sources can be natural or artificial. Daily light is not so good for recognition of colour. The halogen ball is better than daily light or normal bulb. The worst source of light is fluorescence tube. Light imperfection can be minimalized by some filters.

Blood plasma has to be checked before its freezing because after freezing, blood plasma is covered by a white frost. White frost has its influence on recognition.

Recording medium can be camera or video camera. In this experiment, a Camera Canon EOS 450D was used.

Processing on computer is last step in recognition [5, 6].

C. *Algorithm of software*

The algorithm was designed in MATLAB. Pictures have been taken by camera in Blood Centre of the Faculty Hospital in Ostrava. Haemolysis has been recognized in three colour spaces. After that, it has been compared which model is the best. The colour spaces are RGB, Lab, xyY.

The RGB model is composed of R = red colour, G = green colour and B = blue colour. The model is based on knowledge reference colours. These references colours signify the level of haemolysis. These samples have been taken from the tab; see Figure 2. When the colour of blood plasma is the same like sample number one, it is represented by red colour. The second sample is represented by yellow colour. When the colour of blood plasma is the most similar to third sample, this area will be blue. The forth sample is represented by turquoise colour, fifth by green colour, sixth by pink and the last one is represented by black colour; see Figure 4.

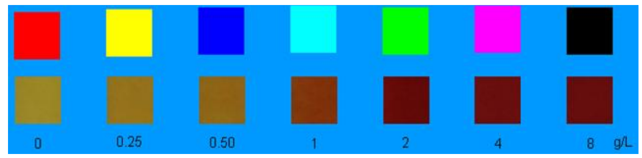

Figure 4. Colours of references samples.

The second model is the Lab model. L is brightness component "a, b" and represents colours of components. L component is unnecessary; the algorithm does not use it for recognition. The algorithm uses only component "a, b". The reference colours come from tab; see Figure 2. The values are transferred from RGB to XYZ and from RGB to Lab by transformational equation [7]:

$$X = 0.430574 \cdot R + 0.341550 \cdot G + 0.178325 \cdot B$$
$$Y = 0.222015 \cdot R + 0.706655 \cdot G + 0.071330 \cdot B$$
$$Z = 0.020183 \cdot R + 0.129553 \cdot G + 0.939180 \cdot B$$

$$L = 116 \cdot \left(\frac{Y}{Y_n}\right)^{1/3} - 16$$
$$a = 500 \cdot \left[\left(\frac{X}{X_n}\right)^{1/3} - \left(\frac{Y}{Y_n}\right)^{1/3}\right]$$
$$b = 500 \cdot \left[\left(\frac{X}{X_n}\right)^{1/3} - \left(\frac{Z}{Z_n}\right)^{1/3}\right]$$

The focus is calculated for every individual reference sample with coordinates "x, y". Blood plasma is classified to the nearest sample.

Last model is xyY model. This model includes the extent of human seeing. Y is brightness and colours components are "x, y". Transformational equation transfers values from RGB to XYZ and from XYZ to xyY. Algorithm does not use Y component [7].

$$x = \frac{X}{X + Y + Z}$$

$$y = \frac{Y}{X + Y + Z}$$

$$Y = Y$$

V. TESTING

Transfusion bags with blood plasma for testing are software coming from Blood Centre of the Faculty Hospital in Ostrava. The number of testing blood plasmas was 91. Two blood plasmas were all right and 2 were haemolytic. Pictures were taken with different light sources. Forty eight blood plasmas were taken with daily light. The other plasmas have been taken with artificial light source - Aputure AL-198 LED light and the light of LCD display which was put under the plasma. Nine blood plasmas were taken twice, with daily light and with LCD display and results were compared. All pictures have been tested in RGB, Lab and xyY colour space.

A. RGB

The algorithm evaluated five blood plasmas wrongly. It detected haemolysis for clear blood plasma. Mistakes were in the first half of testing in RGB colour space. In Figure 5, there is Software for recognition of haemolysis. There is also blood plasma, which is analysed in RGB colour space. Plasma is painted with colours of reference samples which occur in blood plasma.

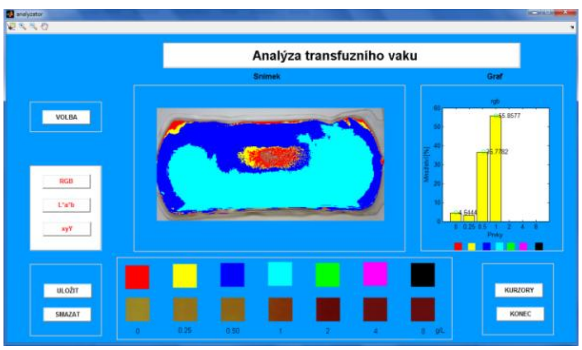

Figure 5. Software for analysing a blood bag with blood plasma - RGB model.

Plasma is painted with colours of reference samples that are occur in blood plasma. Reference samples are in RGB colour space, as shown in Figure 4.

- Software downloaded picture of blood plasma and reference samples.
- Picture with blood plasma was recognized by 3 models.
- Results were presented by colours on the picture with blood plasma and in graph; see Figures 5, 6 and 7.
- Accurate values can be shown by cursors.

B. Lab

The algorithm evaluated two blood plasmas wrongly. The example is in Figure 6.

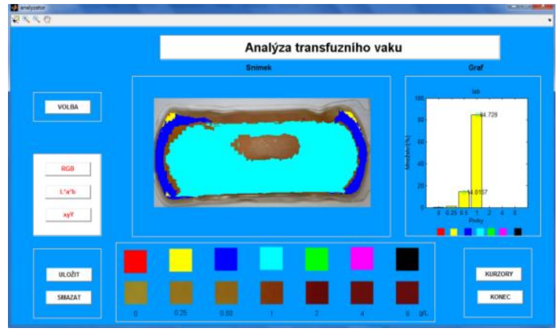

Figure 6. Software for analysing a blood bag with blood plasma - Lab model.

Plasma is painted with colours of reference samples which are occurred in the blood plasma. Reference samples are in Lab colour space too.

C. xyY

The model xyY is the worst way for recognition of haemolysis in blood plasma. The number of wrong results is eleven and software detected two haemolysis of blood plasmas as clear blood plasma; see Figure 7.

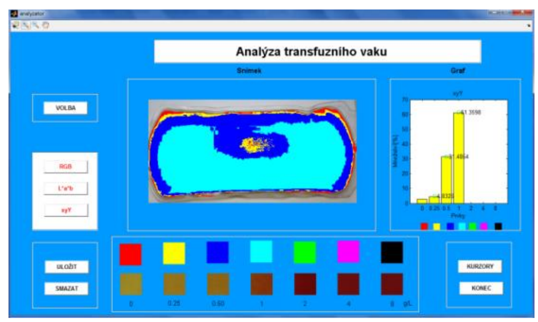

Figure 7. Software for analysing a blood bag with blood plasma – xyY model.

Plasma is painted with colours of reference samples that occur in blood plasma. Reference samples are in xyY colour too.

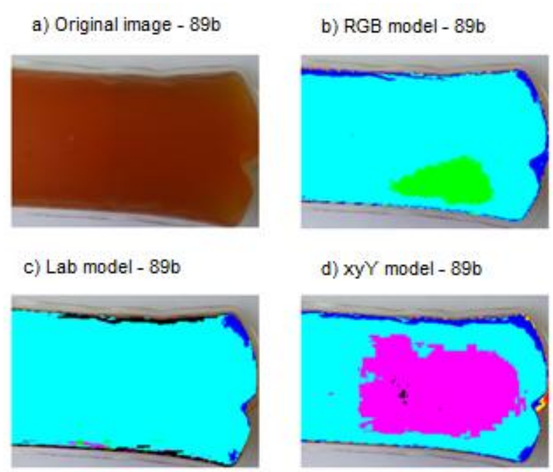

Figure 8. Blood plasma puts on white pad during daily light.

In Figure 9, there is blood plasma on white pad. The light source was daily light. Blood plasma has lightly colour at the end of transfusion bag. Bag has barcode in the middle. This barcode does not influence the recognition. Only Lab model made right detection. RGB model detected little amount of the fifth step of haemolysis. The model xyY recognized haemolysis in the half of transfusion bag.

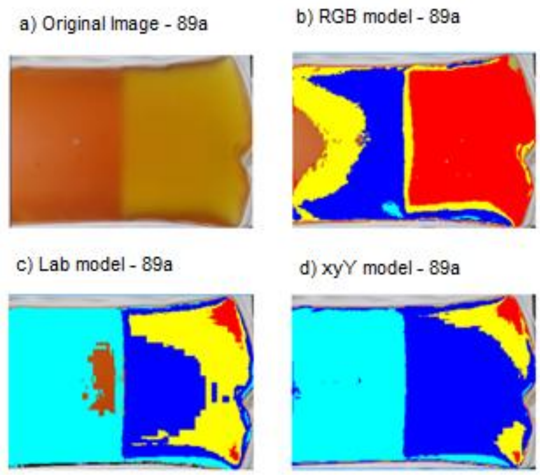

Figure 9. Blood plasma puts on LCD display.

In Figure 9, there is blood plasma put on LCD display. The source was light. There you can see barcode in the middle of the bag. The barcode was influenced by the results. The colour is darker than at the marginal part of transfusion bag. This is why this part is not important for classification. These conditions are better for classification of all colour models. There is no haemolysis in blood plasma.

VI. FUTURE PLAN

The algorithm can recognize colour of blood plasma on knowledge reference colours. These colours come from the table; see Figure 1. The plan uses real colour of blood plasma and does reference colours from these samples. When blood plasma was on the LCD pad, the barcode was in the middle. Haemolysis was detected from whole bag. The plan is recognize haemolysis just from peripheral parts without middle part, where the barcode is.

This method can be enlarged on other functions. One of the functions can be checking barcodes. Nurses do this checking. They are checking integrity of bag too. This can be done by algorithm too. The nurses could have more time for another more important work.

VII. RESULTS

The aim of this work was to design a method for automatic recognition of haemolysis in blood plasma. This method was designed in close cooperation with Blood Centre of Faculty Hospital Ostrava. Colour of blood plasma was analysed in three colour spaces RGB, Lab, xyY. Pictures have been taken by camera and light sources, such us daily light, LED light and LCD display light. The worst results were in xyY colour space. This colour space is based on human seeing. This can probably be the reason for the worst recognition. Majority of mistakes were caused by bad light source. When the transfusion bag was on the white table and light source was LED light.

Software was designed in MATLAB. It has many functions. Software can recognize haemolysis of blood plasma in three colour space. The result can be seen on picture with blood plasma. Detected colours are marked by colours of references samples. The quantity of colours is marked in the graph.

Future plan is to work out the software in detail, to make new reference samples from real plasma and to use light conditions at high quality. Finally, the aim is to make this method automatic in Hospital and in Industrial Factories.

ACKNOWLEDGMENT

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LonMaps: An Architecture of a Crime and Accident Mapping System based on News Articles

Hideaki Ito
School of Engineering, Chukyo University
101 Tokodachi, Kaizu-cho, Toyota, Aichi, Japan
Email: itoh@sist.chukyo-u.ac.jp

***Abstract*—LonMaps is an information system for a crime and accident mapping system based on news articles, which enables extraction of certain information items from the news article. To realize the system, four types of information items are extracted, which are crimes/accidents (incidents), places, dates, and personal names. For capturing incidents, a thesaurus consisting of two types of terms, daily and legal terms, is used. Daily terms are used in daily life, while legal terms are used in legal situations. Places, dates, and personal names are extracted on the basis of typical news report patterns. Finally, experiments are performed using LonMaps to evaluate its effectiveness of processing queries and of extracting information items.**

***Keywords*—*crime and accident map; news article; thesaurus; sentence pattern; extraction*.**

I. Introduction

Recently, several types of systems for information presentation and management have been developed by integrating maps [6]. When locations play an important role in such information and are presented on maps, a system that deals with such information becomes more useful. A collection of local news about crimes and/or accidents (incidents) is one such type of content that is published in newspapers. In order to annotate such news articles, it is required to extract key items from articles. Key items are elements such as the types of incidents, places, dates, and personal names. Extracted items are useful not only for mapping incidents on maps but also for managing and retrieving news articles.

To integrate local news articles and maps and to manage news articles, LonMaps (Local News Map System) is being developed [14]. Maps indicate places where incidents occur. The system analyzes a single news article to extract information items consisting of an incident, a place, a date, and a personal name. Local news articles have similar structures, so patterns for representing news are found relatively easily. News articles are analyzed by such patterns. After the analysis, the places of incidents are displayed on maps and their positions on the maps are obtained using geo-coding. LonMaps is implemented using GoogleMaps [11].

The type of incident is extracted from the news article. News articles consist of legal terms and daily terms. Legal terms are used in laws, courts, and police departments, i.e., legal situations, while daily terms are used in daily life. Therefore, it is necessary to reduce the gap between the two types of terms, and to annotate a news article using legal terms is needed for preventing ambiguous representation of incidents. Even if the article is described in only daily terms, suitable legal terms are captured from them. To reduce this gap and to correlate the two types of terms, a thesaurus is constructed. This thesaurus is used not only for annotating news articles but also for retrieving them.

Several types of location-centered and geographic information system are being developed with integrated maps [6]. Ilarriet al. [13] discussed that location-aware information is useful in our daily life. When a disaster happens, a crisis map becomes a social tool [10]. Some systems that indicate crimes on maps are developed and they are referred to as crime maps [2], [8], [17], [18]. Locations involving local news articles are indicated on maps available online [1]. A system that allows end users to note local information on maps has been proposed in [12]. A news article analysis systems have been developed in [9], [16]. Moreover, information extraction systems have been developed in [19], [20], where sentence analysis, grammatical knowledge, and templates are used. Some systems extract places from news articles, and they indicates the extracted places on maps [4], [21].

Many existing thesauri are built depending on the application domain. They are utilized for retrieval and annotation [3], [5]. The thesaurus of LonMaps is designed for crimes and accidents, and is used to correlate daily terms and legal terms collected from actual news articles. There are some types of links used to connect two terms. For retrieval and annotation, traversing of the thesaurus is controlled depending on types of terms and links.

Development of LonMaps is currently underway, and its overview, the details about the procedures of using the thesaurus, as well as place and date extraction are discussed in [14]. This paper shows some considerations about this system, description of news articles, name extraction patterns, and some experimental results of query processing and of extracting information items.

This paper is organized as follows. Section II describes design of components. Some mechanisms using a thesaurus are shown in Section III. Section IV shows sentence patterns for extracting information items. Some experimental results are shown in Section V. Finally, conclusion and future work are described in Section VI.

II. An overview of LonMaps

A. Design of LonMaps

When a news article is read, it is well-known that the '5W1H' (what, when, where, who, why, and how) of an

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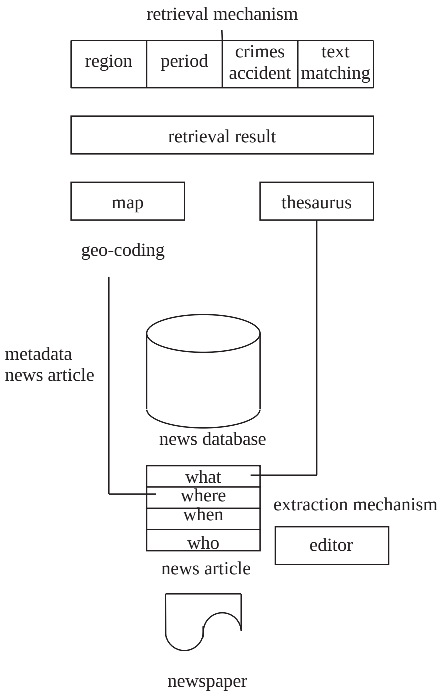

Figure 1. An overview of LonMaps.

```
<articles>
     <article id>  ; news id
          <issue year month day />  ; issued date
          <happen year month day /> ; occurrence date
          <newspaper which> </newspaper>  ; 'which' is a distinction
     of morning or evening editions of the newspaper. The value is the name of a
     newspaper.
          <accident>  ; annotation by incidents.
               <man-assigned> </man-assigned>   ; man assigned
     incident list, if required.
               <captured> </captured> ; a list of captured incidents in
     terms of legal terms.
          </accident>
          <address lng lat> </address> ; 'lng' and 'lat' are
     'longitude' and 'latitude', respectively.  The value is the places of an address
     and/or location .
          <headline> </headline> ;  the headline of a news article.
          <text> </text>  ; the body of a news article.
     </article>
          . . .
</articles>
```

Figure 2. A structure of a news article in XML.

incident are key items [15], [21]. We deal with four of these when using LonMaps: '4W' (what, when, where, and who).

The requirements of the system are as follows: (1) extraction of places where incidents occurred and indication of places on maps, (2) annotation of news articles in legal terms, and (3) clarification of the relationships between daily terms and legal terms using a thesaurus that consists of these two types of terms.

The features of the system are as follows:

- Four information items are captured from a news article: incidents, places, occurrence dates, and persons. These items are captured using the thesaurus and patterns that are defined by us. The dates, the places, and personal names are captured on the basis of patterns of the sentences. In many news articles, conventional patterns are typically used to describe the article.
- The thesaurus is provided for determining the relationship between legal terms and daily terms. This thesaurus includes not only terms representing incidents but also terms directly and/or indirectly related to incidents.

The thesaurus of the system is used to reduce the gap between daily terms and legal terms. For example, in retrieval of news articles, when a user specifies a query in terms that are a conventional representation of crimes or a usual representation of incidents, i.e., daily terms, some news articles cannot be retrieved, because they do not include recognizable daily terms. Daily terms are not always used for representing incidents, so legal terms may be required to retrieve the news article. In addition, it is necessary to annotate news articles in legal terms to describe incidents through formal and uniform representations. Since a news article may not include any legal terms, it is necessary to find suitable legal terms from a collection of daily terms that appear in the article.

Figure 1 shows the overview the architecture of LonMaps. The system consists of a retrieval mechanism, an extraction mechanism, an editor, and a news database. The retrieval mechanism retrieves news articles. The extraction mechanism analyzes news articles. Moreover, an editor is used to edit a news database, e.g., defining a news article and modifying elements of news articles. The extraction mechanism is a main component of this editor.

B. Description of News Articles

A news database is a collection of news articles described in XML. Figure 2 shows tags for describing a news article. In this description, metadata of the news article are included. The metadata include elements such as publication date and the newspaper name. An entire set of news articles is indicated by `<articles>`. Each news article is indicated by `<article>`. The `id` attribute is an article identifier. `<newspaper>` indicates the name of a published newspaper. `<issue>` specifies a publication date. `<happen>` specifies a date when an incident occurred. `<accident>` specifies a type of incident. `<man-assigned>` denotes a list of incidents assigned by a person. `<captured>` is a list of incidents captured by the system. `<address>` specifies a place, and its longitude and latitude are obtained using geocoding provided by GoogleMaps. `<headline>` specifies the headline. `<text>` specifies the main text. The system tries to capture the values of `<happen>`, `<incident>`, and `<address>`.

C. Description of Queries

Three types of queries are available in LonMaps: a keyword query, a time period query, and a region query. Figure 3 shows the structure of a screen for specifying

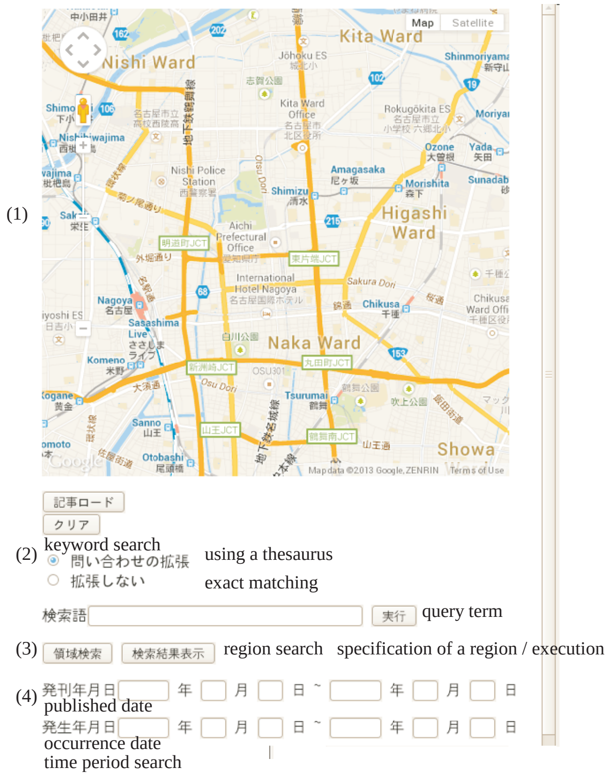

Figure 3. Structure of a screen for specifying a query in LonMaps.

queries. The first part of Figure 3 is a map. The second part is for a keyword query and a form to input query terms. If an expansion using the thesaurus is selected, given terms are expanded. Otherwise, when terms are given, exact matching is applied. A query is specified in query terms and logical connectives, i.e., and, or, and not. The third part of the screen is for a region query. A region is specified first, and then news articles within a specified region are sought. The fourth part is for a time period query. The occurrence date and/or a published date are specified as the query. After queries are specified, retrieval results are obtained. The retrieval mechanism shows the results on the map. Although markers are not shown in Figure 3, the retrieved articles are indicated by markers.

III. Utilization of the Thesaurus

A. Structure of the Thesaurus

Daily terms are usually used and easy to understand in daily life, while legal terms are defined explicitly. To retrieve and annotate news articles, it is necessary to correlate these two types of terms. The thesaurus is constructed from a collection of news articles and the Japanese compendium of laws, whose conceptual structure is shown in Figure 4. There are two types of legal terms. One is the formal names of laws. Laws are defined in a hierarchical structure. The other is the legal representation of particular crimes and accidents. In contrast, daily terms are informal names for incidents, relevant terms, verbs, and conjugations. Relevant terms do not directly represent incidents but frequently co-occur with other daily terms. Moreover, the original form of a verb and its conjugations are considered daily terms.

Four types of links are defined: “is-a”, “general-term”, “associated-with”, and “conjugation-form”. Two legal terms are connected by “is-a”. A collection of legal terms is organized into a hierarchical structure. Two related terms are connected by the “associated-with” link to each other. This link is used for connecting not only two daily terms but also a daily term and a legal term. A verb and its conjugations are connected by “conjugation-form.” Moreover, if an inverse link of a link is required, it is set explicitly. Furthermore, “general-term” and “associated-with” are used for connecting legal terms and daily terms. The former is used for converting the legal term to corresponding daily terms. The latter connects two terms that often co-occur and are strongly associated with each other.

A part of the thesaurus is shown in Figure 5. Its root is the node “root”. At the second level, the root node has two children: “law” for defining laws and “accident” for defining accidents.

B. Query Processing using the Thesaurus

When a term defined in the thesaurus is given as a query term, the given term is expanded by traversing connected links from the given term to others in turn. When a given term is the name of a law, at first, descendants of the given term are obtained by following “is-a” links recursively. Next, by following “general-term” links connecting legal terms, daily terms are obtained. Finally, relevant terms and conjugations are obtained by “associated-with” and “conjugation-form” links. A set of these obtained terms is treated as the response to the query terms.

When a daily term is given as a query term, a legal term corresponding to the given term is first obtained. Next, the terms that are connected to the given term by “associated-with” and “conjugation-form” links are obtained. For each obtained daily term, the procedure for processing a daily term is then applied recursively.

C. Annotation using the Thesaurus

To find an incident of a news article, it is examined whether terms defined in the thesaurus appear, or not. Consequently, some daily terms are founded. Legal terms related to these daily terms are sought. For example, assume a daily term appearing in a news article, as shown in Figure 6. The related legal terms are captured by the inverse link of “general-term”. Moreover, certain terms connected to the given term with “associated-with” links are also captured. Consequently, legal terms are selected as candidate terms for annotation.

In this example, some candidate legal terms are found. If a term is positioned lower than others in the hierarchical structure of the thesaurus, the term is considered as an incident, since a lower term tends to represent a specific incident more precisely than higher terms. For example, assume “murder” and “robbery” are obtained from the

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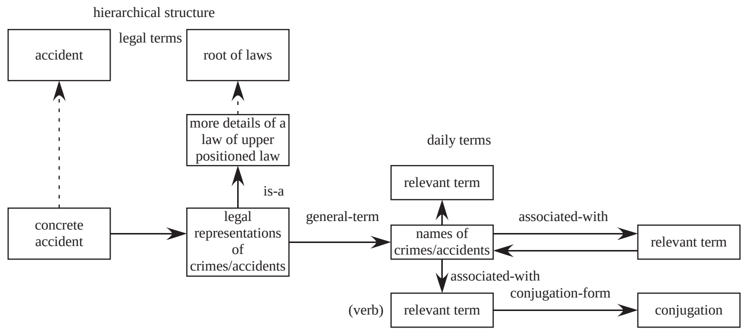

Figure 4. Components of the thesaurus.

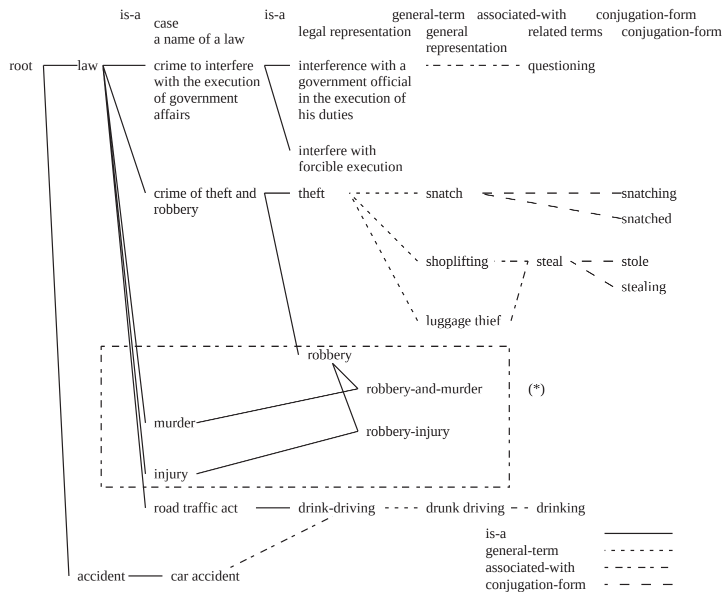

Figure 5. A part of the thesaurus.

original news article. The annotation mechanism tries to find more specific terms. Then, “robbery-and-murder” is selected rather than both “murder” and “robbery”, as shown in the part of the thesaurus marked by “(*)” in Figure 5. More details of the procedures used for traversing the thesaurus are described in [14].

IV. Extraction of Information Items

A. Extraction of Places

To extract places where an incident occurred, the following procedures are applied: (1) a part of a sentence that includes a place is extracted using sentence patterns, (2) area names are extracted by applying morphological analysis to the part of the extracted sentence, (3) when certain area names are omitted, they are complemented, and finally, (4) resolution of anaphora references is applied.

The patterns are shown in Figure 7. These patterns consist of three parts. The first part is a term representing a time period, the second a place, and the third a postpositional particle in Japanese. Here, words are separated by spaces in English but not in Japanese. For example, let a sentence including the place be “14 日午前 4 時 ごろ、名古屋市熱田区 1 番 3 の市道交差点 で(at the municipal road crossing in 3, Ichiban, Atsuta, Nagoya around 4 a.m. in the morning on the 14th)”. The string between ごろ、(around) and で (at) is extracted. ChaSen [7], the morphological analysis system for Japanese is applied to the string for obtaining nouns included in this string. ChaSen divides a string into words by filling spaces between words, and searches for nouns. Some nouns from the beginning of this are obtained, as long as possible. The identified nouns are then concatenated. The resulting sequence is seemed as a place. For example, “名古屋市熱田区 1 番 3” is extracted. Then, the complement of omitted area names and anaphora resolution are applied, if needed.

B. Extraction of Dates

Analogous to extraction of a place, extraction of an occurrence date is achieved on the basis of patterns and the published day. Figure 8 shows patterns for describing an occurrence date. First, a sequence of numbers and “日 (day)” are found. A modifier for representing time appears in the same sentence, e.g.,“午前 (in the morning)”. For example, let “14 日午前 (in the morning on the 14th)”

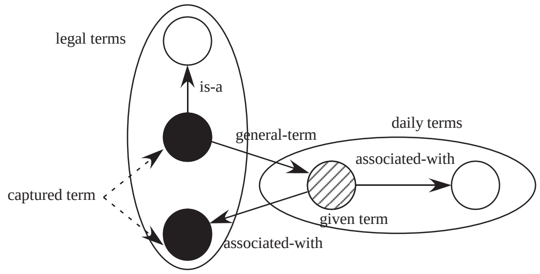

Figure 6. An annotation procedure when a daily term appears in an article.

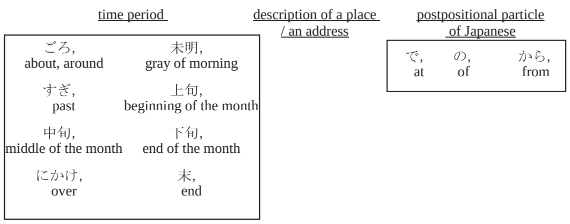

Figure 7. Patterns for extracting an occurrence place.

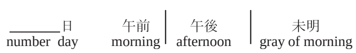

Figure 8. Patterns for extracting an occurrence date.

appear. The number 14 is extracted as an occurrence day. A month and a year are omitted in many cases. They are complemented on the basis of the published date of the news article, since the month of the occurrence day and the month of the published day are usually the same. The candidate day of the occurrence date and the published day of a published date are compared. If the candidate day is less than the published day, the month of the occurrence date and the month of the published date are the same. If the candidate day is greater than the published day, the month of the occurrence date is determined as the last month of the month of the published date in many cases.

C. Extraction of Personal Names

Personal names appear in a news article as victims, suspects, and other parties. Although such personal names are important in news, it is considered that presenting personal names in older news articles is unimportant over the long term. So, the system does not print personal names. To achieve this, personal names are extracted using patterns.

Figure 9 shows patterns that describe names of a suspect and/or a victim associated with crimes. Here, word order is different in English and Japanese. A personal name is a full name when the name is presented the first time. The full name is a sequence of his/her last name and first name in Japanese. A suspect and a victim are distinguished by the modifiers used with the names. Modifiers for a suspect and for a victim are, for example, ‘suspect’ and ‘title’, respectively. In Figure 9, path (a) is the most popular pattern. The sequence of this pattern is ‘occupation’, ‘name’, ‘modifier (suspect or title)’ and ‘(age)’. In (b), a comma between an occupation and a name is noted. In (c), a case particle is used. In (d), an address is shown. In cases where a name appears several times, the full name is omitted, afterwards. A personal name is referred by the last name and his/her modifier. Then, such references are treated as unprintable words. Moreover, as for occupation, some representations such as a worker, an office worker, or a therapist, are used.

TABLE I PRECISION OF QUERY PROCESSING USING A THESAURUS.

	precision
legal terms for crime names	0.92
daily terms for crime names	0.48
verbs in daily terms	0.84
nouns in daily terms	0.55

V. EXPERIMENTAL RESULTS

Query processing with the thesaurus was examined. As a query term, four types of terms were used: legal terms for crime names, daily terms for crime names, verbs in daily terms, and nouns in daily terms. Precision of their retrieval results were measured, as shown in Table I. The precision for legal terms representing crimes were better than the precision for other types of terms, since the information needs representing crimes in legal terms were more precise. The precision for verbs was better than the precision for nouns and crimes in daily terms. It appears that verbs were related to crimes more directly than nouns. Precision was worse for nouns in daily terms and crimes represented in daily terms. These terms were ambiguous and were not directly related to precise crimes. Recalls were high because the thesaurus was used to expand a given query term and many related terms were obtained.

Next, validation of annotation was measured. The results are shown in Table II. About one hundred news articles were examined for annotation. The average number of legal terms obtained using the thesaurus was 4.5 for one article. Among these terms, 2.8 suitable legal terms were included. The ratio of suitable terms to obtained terms was 0.21. This ratio was computed as (the number of common terms assigned by a person and by the system)/(the number of legal terms captured by the system). Person assigned terms included both legal and daily terms. Moreover, the ratio of suitable legal terms among obtained legal terms was computed to be 0.78. This was computed as (the number suitable legal terms captured)/(the number of legal terms captured). As described above, news articles were annotated by a person before extraction of legal terms using the system. Then terms that represented concrete incidents and appeared frequently in news articles were selected for annotation in many cases. It was difficult to annotate in legal terms since daily terms need to be converted to legal terms. The results of these two annotation experiments by a person and by the system indicate that annotation often involved daily terms when done by a person, whereas by the system, suitable legal terms were found from daily terms using the thesaurus.

The validity of places obtained from the appearance of written words was evaluated. Places where an incident occurred were extracted using description patterns, and places were obtained by reading a news article. When the

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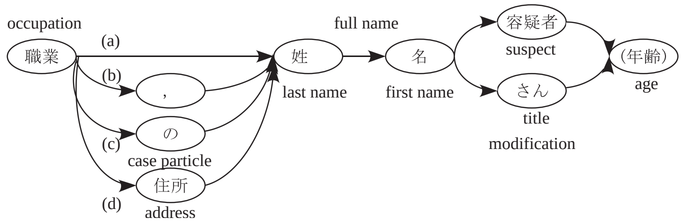

Figure 9. Patterns for extracting a personal name.

TABLE II VALIDITY OF ANNOTATION.

mean value of the number of captured legal terms	4.5
mean value of valid number of legal terms	2.8
ratio of suitable terms among captured terms	0.21
ratio of valid legal terms	0.78

TABLE III VALIDITY OF OCCURRENCE PLACE EXTRACTION.

	validity
pattern matching and concatenating nouns	0.78
pattern matching, ChaSen, complementing place names and anaphora reference resolution	0.91

place extracted by the system was the same as that extracted by a person, the system's response was assumed as correct. The validity is computed as (the number of news articles where the places are captured correctly)/(the number of treated news articles). The result of this experiment is shown in Table III. The validity of extracted place was 0.78 when only patterns were used. By applying morphological analysis, omission complement, and anaphora resolution, the validity improved to 0.91.

Personal name extraction was also examined. In our experiment, about 96% of appearances of personal names were covered by described patterns. When a personal name represented in a full name was obtained, the personal name described by only her/his last name was captured. However, there were cases where several names of suspects appeared in one sentence. LonMaps does not currently have the capability to classify such patterns.

VI. CONCLUSION AND FUTURE WORK

LonMaps is being developed as the first step toward building local news maps. This system provides mechanisms for retrieval of news articles and extraction of information items from them using a thesaurus and sentence patterns.

There are many cases that do not specify specific places in a news article. A place is specified as a division or an area of a town in many cases and this presents a problem in how to display a general area within a town. Moreover, the system captures incidents and retrieves articles on the basis of the thesaurus. When the thesaurus is used, the plausibility of relationships between terms is not introduced. When queries are processed and when incidents are extracted, the relationships between terms are treated as strict relationships. However, it is necessary to reflect plausibility of such relationships for introducing certainty of obtained news articles in retrieval and obtained terms in annotation.

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Generic Self-Learning Context Sensitive Solution for Adaptive Manufacturing and Decision Making Systems

Dragan Stokic, Ph.D., Sebastian Scholze, Oliver Kotte
Institute for Applied System Technology, ATB-Bremen,
Bremen, Germany
dragan@atb-bremen.de

***Abstract*— The paper investigates applications of context sensitivity to achieve high adaptivity of systems. A generic approach for context sensitivity, based on self-learning algorithms, is proposed aiming at a wide scope of systems. The approach is applied for high adaptation and integration of complex, flexible manufacturing systems as well as complex decision making systems, e.g., in software engineering. The proposed solution includes context extractor, adapter and self-learning modules allowing for adapting of the process and/or decision support systems depending on the extracted context. Both context extraction and adapter are continuously learning and improving their performance. Service Oriented Architecture (SOA) principles are used to implement these modules. The generic solution and specific applications in various manufacturing environments and decision making processes in software engineering are presented. The paper is one of first attempts to develop holistic context sensitive solution applicable to various systems.**

Keywords - context sensitivity;self-learning systems; adaptive manufacturing systems; context model; descision making systems; software engineering

I. INTRODUCTION

The objective of the work presented in this paper is to investigate how context sensitivity can be used to achieve a high adaptivity of various systems. In order to facilitate applicability of the approach to wide scope of systems, self-learning features are introduced. The idea is investigate application of a new generic context sensitive solution to wide scope of complex systems and decision support systems (DSS). The key assumption of the work is that the proposed generic self-learning context sensitive solution can be easily adjusted to allow for adaptation of wide scopes of systems. The context sensitivity allows for observation of changes in circumstances in which a system is operating, which in turn allows for a dynamic adaptation of the system to these varying conditions. In order to explore applicability of the proposed generic solution to a wide spectrum of systems, in this paper the emphasis is on its application for the systems in two very different domains: complex manufacturing systems and complex DSS in engineering domain.

Implementation of complex models and algorithms for such context sensitive systems require powerful Information and Communication Technology (ICT) infrastructures. A service oriented approach opens entire new perspectives for self-adapting systems. It is likely that approaches based on SOA principles, using distributed networked embedded services in device space (sensors, controllers, etc.), are the most appropriate for implementation of self-learning adaptive systems in general and specifically complex manufacturing systems and DSS.

The paper is organized in the following way. Section II provides a brief overview of the state-of-the art. In Section III, the overall concept is elaborated. In Section IV, the applications of the proposed concept are briefly described, while in Section V the future plans are indicated.

II. SURVEY OF THE STATE-OF-THE ART

Context Awareness is a concept propagated in the domains of Ambience Intelligence (AmI) and ubiquitous computing. It is the idea that computers can be both sensitive and reactive, based on their environment. As context integrates different knowledge sources and binds knowledge to the user to guarantee that the understanding is consistent, context modeling is extensively investigated within Knowledge Management (KM) research [1]. The current research on knowledge context is primarily oriented towards capturing and utilization of contextual data for actionable knowledge [2-4]. A number of systems to handle context awareness were proposed by the research community [5-7]. However, a holistic approach to application of context sensitivity to a wide scope of systems, as proposed in this paper, is not elaborated up to now.

One of the key problems is how to extract context from the knowledge process. In the research presented in this paper is decided (see the text to follow) to model context with ontologies, and, therefore, context extraction mainly is an issue of context reasoning and context provisioning: how to infer high level context information from low level raw context data. Based on the formal description of context information, context can be processed with contextual reasoning mechanisms [8-9].

Defining context for applying context awareness can be difficult [10-11]. Existing formal context models support formality and address a certain level of context reasoning [12 -13]. For example, the modeling of context in the case of DSS for Quality Assurance (QA) in Software (SW) development process presents an additional challenge, which has not been addressed up to now, as the services are highly dynamic and reside in distributed environments. With the emerging and maturing of semantic web technologies,

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Ontology based context modeling becomes a new trend both in academy and industry. Present research on context modeling is mostly focused on ontology. Compared to other methods, ontology based method has many advantages as it allows for context-modeling at a semantic level, establishing a common understanding of terms and meaning and enabling context sharing, reasoning and reuse [14].

In the area of self-learning systems, the research has demonstrated that the application of machine learning techniques, dynamic self-adaptation and operator's feedback in the loop promises to increase the intelligence of the overall system [15-17]. In production systems in particular, these methods have been proven to be especially useful for monitoring/diagnosis and control [18-20]. However, the applications of self-learning systems in, e.g., industrial practice are still in initial phase. In this paper a novel approach in such systems is presented, which is context aware, adaptive to contextual changes at run time and learns from adaptation and operator's action.

Of special interest for the work presented in this paper is SOA, i.e., the relation between context sensitivity systems and SOA. Scalable SOA holds promise for seamless integration, interoperation and flexibility in different environment [21-22]. But, there is a lack of adoption of overall SOA based self-adaptive systems in, e.g., discrete manufacturing environment and DSS. In this paper, SOA based context sensitive solutions are proposed as add-on features in existing infrastructure.

III. PROPOSED APPROACH FOR SELF-LEANING CONTEXT SESITIVITY

A. Proposed concept

The challenge is to define a solution able to handle wide scope of 'disturbances'/changes coming from either external conditions, or process/plant/DSS parameters changes, requiring control adaptations. The proposed approach is to (on-line) identify current dynamically changing context in which the system operates and to 'use' this identified context to adapt control. Therefore, the proposed approach includes a context extractor (as a generalized 'observer' providing current context) and an adapter (as 'active' control part) – see Fig. 1.

As explained above, SOA principles are the most appropriate for implementation of such systems. Context awareness, providing information about the process & plant or DSS and the circumstances under which the system operates, is a promising approach to assure, needed dynamic self- adaptation to changes in the context, including changes in processes & equipment parameters or parameters used within DSS. Wide applicability is enabled by self-learning features.

This approach has not been explored up to now. The basic assumption is that holistic, simultaneous and harmonized usage of self-learning context sensitivity, based on (on-line) extraction of a current context as indicated in Fig. 1, for adaptation of systems, is the effective way to assure considerable advantages regarding efficiency and availability of systems.

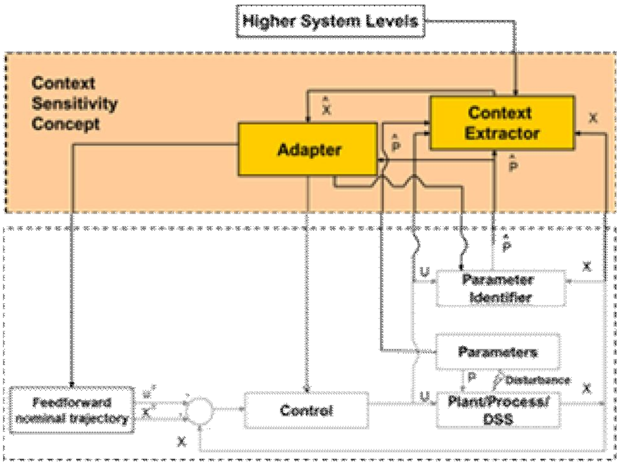

Figure 1. Approach proposed

B. General architecture

A generic solution for context based adaptation of systems is proposed which can be applied in various types of systems, specifically manufacturing systems and DSS. The features and functionality for the overall architecture are specified and developed as generic solutions easily adjustable to wide scope of systems. The system adapts to run time critical contextual changes and learns from it. Learning can also be enhanced by operator's feedback and experiences, especially in the cold start phase. The overall proposed reference architecture is illustrated in Fig. 2. The architecture is designed following SOA principles as an add-on to the standard control following the conceptual approach as presented in Fig. 1. The components of the proposed system include [23, 24]:

- Context Extractor, Adapter, Self-learning services - see the text to follow
- Validation module: the identified solutions are required to be validated, where the user can manually/automatically accepts/rejects any new solution. The validation sends the feedback to the other modules.

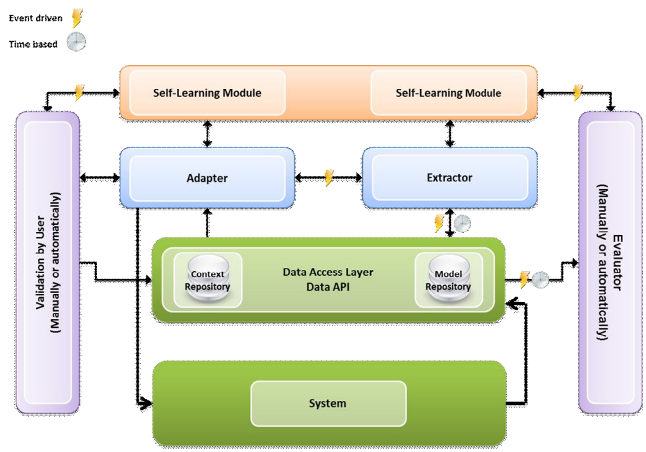

Figure 2. Proposed reference architecture [23, 24]

• Evaluator: Performance of adaptation and context extraction are measured by the evaluator either manually (in cold start) via operator's feedback or automatically via mapping against objective functions at run time. Evaluation results are sent to the Learning modules.

• Data access layer: Generic component, responsible for accessing the various system layers.

• Data Processing: These services are responsible for the bidirectional processing of information and perform, e.g., pre-processing of monitored raw data acquired via the data access layer, before the context is identified. Main functionality is to transform the raw data in a format which serves as basis for context identification. The Model Repository contains ontology based plant specific models for equipment, production processes/products and/or DSS. The models are shared by different software components at run time. The Context Repository allows update and storage of extracted/processed contextual information for later retrieval. Information flow among the modules is event driven in some cases and time based in other cases.

• Service Infrastructure: underpinning framework ensuring information is securely gathered from trusted context data sources and that the control updates are communicated to control systems with appropriate levels of authentication. The communication authentication components ensure seamless and secure connectivity with existing information system communication protocols and security mechanisms.

C. Context model

As indicated above, the basic assumption of the proposed approach is that monitoring the plant/process/DSS and its environment enriched with context can help to be aware of any potential change that may have an effect the system behaviors and/or decision making process. Therefore, a research key is the definition of a „holistic" and dynamic context model and ontologies to enable context awareness, allowing taking into account the context of the system operation and /or decision making processes (e.g., processes, equipment and product information, users, teams, etc.) [25]. As ontology allows for knowledge sharing, logic inference and knowledge reuse, it is a widely accepted approach for semantic-rich context modeling. Therefore, ontology is used for context modeling. Based on a context ontology, logic based context reasoning is realized, such as consistency validating, subsumption checking, etc. More importantly, domain specific rules are defined to infer implicit context from explicit context, and high level context from low level context. Other statistic and machine learning approaches can be adopted for non-logic context reasoning.

The definition of the context model is a key approach to assure usability of the proposed solution in different domains. The application of the solution to a specific domain normally requires adjustment of the context model. Therefore, a general and extensible context model is proposed. It is in a format that meets several requirements: help to describe and capture context easily; help to manipulate context; facilitate context consumption by KM services. The proposed Context Model defines two models as

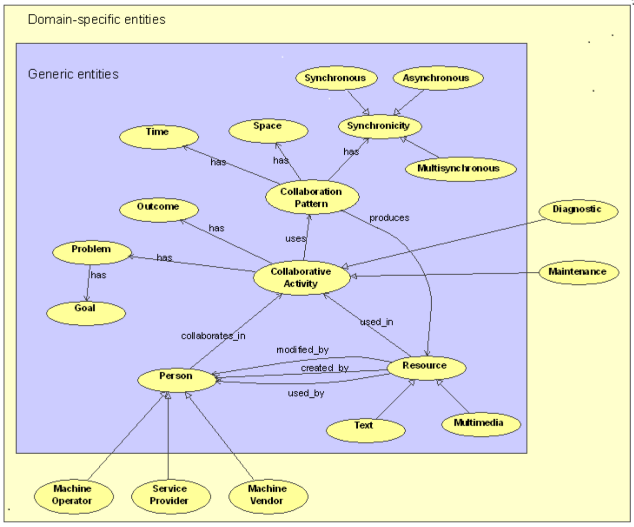

Figure 3. Context Model – ontology genereic and domain specific

sets of ontologies: Generic Context Model and Sector-Specific Context Model. Figure 3 presents an example of two models. The research is focused on development guidelines to effectively define context models for various applications. The basic principles for context modeling were followed: (1) Support description of main context: In practices, all context information cannot be modeled. The context model should consider those most relevant concepts and information needed to meet the requirement of context sensitive adoption. (2) Model the context that is (easy) acquirable: The context factors considered should be identifiable and acquirable, whether provided through computer monitoring automatically, or by user input explicitly. (3) Trade-off between investment of context modeling/extracting and effects of context sensitive adoption: Intuitively, if we could model as much context factors in as much details, the accuracy of context will be higher. However, this does not come for free: more time and efforts are needed for context modeling and more computing resources are needed to handle the context, which will bring deficiency to the adoption process.

D. Context extractor

The Context Extractor is the set of embedded services responsible for identifying changes in the context of the environment. The current identified context is used to extract available context knowledge. The results of the Context Extractor are used in the Adapter which is responsible for updating the system behavior. The Context Extractor uses all monitored "raw data" provided via the data access layer to derive the machine's current contextual situation. Using the ontology/context model the monitored data are evaluated and the context extracted. Based on the identified context, situations can be compared to previous ones and stored. A continuous process, coordinating with the monitoring and followed by the adaption process is built based on reasoning around the extraction of a contextual situation. The main elements of the context extractor are:

- Context Identification: It is responsible for the identification of the current context, based on monitored raw data, the adopted ontology and historic context information stored in the context repository.
- Rule Engine: Responsible for providing appropriate rules for the identification of context – context reasoning.
- User Interfaces: User interfaces for maintaining and administering the rules and the context repository.
- Application Specific Modules: This set of services includes application specific components and user interfaces.

E. Adapter

The Adapter is the set of services responsible for updating system behavior (locally and/or globally) in response to a change of context in the environment. The adaptation is based upon available context knowledge to identify the best set of rules to employ in each context. The Adapter is informed by the Context Extractor about a variation on the system (change of context) and adapts the system to handle the new situation. The Adapter is guided by a set of rules that describe how the system should behave in each particular context. These rules can be updated through learning based on lifecycle history data, context and user validation. Although triggered by the Context Extractor whenever a change of context is detected, the Adapter is able to periodically enquire the Learning module to check for a new set of rules that promise to improve system performance in accordance with production objectives. This information can be then presented to the user in order to validate or not the new proposal. In the positive scenario, this proposal will be saved and become the new de facto set of rules, which will delineate system behavior for that particular context. Each application case comprises a specific set of rules and output, but the Adapter structure is generic:

- Rule Engine: an engine framework that is able to process application specific rules.
- Context Action Selector: These services are responsible for the definition of the adaptation depending on a particular context. These services interpret the results obtained by running the rules in the Rule Engine.
- Context Action Optimizer: Set of services to update of the current set of rules, either due to a Learning module suggestion (validated by the user) or by direct input.
- Application Specific Module: This set includes services that need to be developed for each application individually. Application specific rules and information templates are also defined here.

F. Self-Learning

The proposed concept for self-learning represents the capability of a complete system to learn along its life-cycle based on the experience retrieved from past and current relations and share of information between its different elements in a distributed manner. Learning services allow the system to learn relying on Data Mining (DM) and operator's feedback to update execution of adaptation and context extraction at run time. The results (adaptation rules) are suggested to the Validation module, and must be backed up by the user (either upon request or automatically). Adaptation rules and context identification procedures are updated accordingly. DM has been considered to be appropriate approach for the concerned applications in complex production systems and DSS. The approach relies upon the extraction of previously unknown and potentially useful patterns from data sets. DM is often referred in the context of Knowledge Discovery in Databases (KDD), which consists in selecting data samples from a large database, treating it and analyzing it for pattern extracting. The sampling process is used because, most of the times, the total amount of data is too large to be fully analyzed. There are several types of methods for DM to discover the patterns in the data. The generic solution prosed in this paper includes a set of algorithms from which for each specific application the most appropriate one can be selected within system set-up. The cold start is applying normally supervised learning. The methods included in the generic solution are: Classification – learning of a function aiming to map the data inside a set of classes in the best way possible, Regression - focuses on the relationship between a dependent variable and one or more independent variable, Clustering – seeks to find a finite number of categories or clusters to classify the data, Association Rule Learning - Discovers relations between variables in databases and describes them as rules.

IV. APPLICATIONS

As indicated above, the proposed concept is applicable to wide scope of systems. Two specific applications were investigated in practice: application for the control of manufacturing processes and application for adaptation of decision support system within software engineering.

A. Application for adaptive manufacturing systems

The approach has been applied for adaptation of various manufacturing processes to environmental and parameters changes in several manufacturing companies in various sectors [25]. One of the applications was to achieve high adaptation of the machines in shoe industry to the changing conditions. Therefore, the proposed solution is applied to react to the changing situations/contexts associated with variations in different parameter sets. The parameter variations of both controlled plants (machines) and environments in which the systems are operating are in terms of pressure and temperature, speed frequencies of drives and pumps, proper material mix ratio and filling of materials into shoe forms. For example, one of the scenarios addresses synchronicity of the different valve circuits when dosing of several components. As the valve synchronization is designed by a mechanical system, and/or due to valve abrasion, with different valve opening times, caused by, e.g., different force requirements or different air supply, it may come to flaws in the product. Currently, the synchronization can only be adjusted by a technician during downtime of the machines. By implementation of the proposed self-learning solution an automatic adjustment of the valve switching to different conditions is achieved. The context monitoring serves as a basis for identifying valve adjustment parameters. Device centric infrastructure is based on machines, the corresponding PLC' and operational PCs. Ethernet, Field bus, CAN Open, Profibus based communication protocols are used to communicate within the equipment level. Main information sources from device layer are sensors for pressure/temperature, pumps/drives for speed frequencies and controllers for parameter sets. Enterprise level parameters are accessible in plant database, which are mostly associated with varied customization of shoe types and sizes related to the different production lot numbers. Operators adjust the boundary values or parameter sets through the operational PC of machines. The context extractor services extract and process relevant parameters from the database to input to the Adapter set of services. The adapted solutions, in terms of adjusted parameters, changes to speed frequencies of pumps/drives, changes in the process cycle time, etc., are sent to the Validation module for user's feedback. Based on adaptation results and operator's feedback, the learning services learn how changing cycle times and ambient conditions are influencing the production process (e.g., above explained valve synchronization) and also update the rules for context identification, adaptation and extension. From application point of view, the main advantages are that the operator is not forced to bring highest skills to run the equipment, but can more concentrate on the core processes while the production equipment is self-controlling all relevant parameters to keep the process running. Contrary to the classical adaptive control solutions, the generic self-learning modules integrating more intelligence into the production, are easily applicable to various machines/processes, gaining a higher benefit for the producer while less investment in the human resources are required.

B. Applicaton for descision support system in software engineering

The main objective of the application was to create context sensitive decision support services within flexible QA of SW development projects [26]. The new QA process is supported by the proposed solution composed of several knowledge, context sensitive services that are able to detect changes in the scope and requirements of a SSW component (or changes in its development process) and provide the adequate set of assessments as a basis for an accurate measurement of the quality of the process and product at any time and allow for effective decision making within QA. The Internet Services monitor the different stages of the software development process, interoperating with the existing applications and systems to provide quantitative information about the quality of each phase (i.e., project management, requirements gathering, functional and technical design, development and testing), the project as a whole and the resulting product. They also monitor context under which the SW is developed and decisions on QA have to be made. Data obtained in real-time by the monitoring services are used in an indistinctive way by software engineers, designers, developers, testers and managers alike for different collaborative decision making. In the case of services for monitoring, analyzing and enhancing SW development processes the notion of context refers to preferences and skills of users, physical capabilities of the equipment and environment conditions, coming from different kinds of information sources like bug-tracking systems, code repositories, etc. The key ontology, entitled Activity-Centric Collaboration Ontology (ACCO), allows for representing the context of collaborative work situations in the form of explicit machine interpretable knowledge. It enables intuitive representation of knowledge about collaborative work and serves as the base for further knowledge sharing, refining and reuse. It explicitly describes collaboration related activities, people, resources, and the relationships among them. The problem is how to better describe dynamic SW engineering tasks and decision making processes (i.e., which aspects regarding collaboration are relevant for decision making) in domain specific collaborative work, and how to integrate the context model into infrastructure and tools, in order to enhance context-sensitivity of these tools, and to facilitate context extraction from the content provided by the monitoring services. The appropriate relations between the concepts and various associated attributes in the selected context model are elaborated. The extractor extracts the context changes and informs the adapter to adapt the DSS parameters/rules accordingly. The approach is assessed in 2 different cases in order to validate the results under different conditions. The first case belongs to a large software company developing large Internet projects based on Rational Unified Process methodology. The second case belongs to a SME developing complex projects based on agile methodologies The analysis and testing indicate good potentials to improve QA process by the context sensitive decision support services. The benefits are highly depended on the complexity of the development project and their dynamics. The cost/benefit ratio asks for a deeper analysis of the specific company's development processes, but in the specific cases of complex SW processes the proposed approach is likely to bring benefits of more than 30% w.r.t. to classical DSS.

V. CONCLUSIONS

The research presented in the paper is one of the first attempts to apply a self-learning context sensitivity solution to support adaptivity of a wide scope of systems. The main benefit of the proposed generic solution is that it can be easily adapted to fit specific conditions of each system. The applicability of the solution to manufacturing systems and DSS in QA for SW development is demonstrated. The generic innovative context model has been proposed and developed, and the main adjustment of the solution to the specific systems is a definition of specific context models relevant for manufacturing systems and software engineering process. New applications of such approach for adaptations in flexible manufacturing and effective decision making in agile SW development are elaborated. These specific applications investigated in practice demonstrate that the proposed concept and ICT solution are promising to be applicable to wide scope of system, asking for minimal adjustments. As the applications of context sensitivity for both flexible manufacturing systems and DSS for software development processes have not yet been sufficiently researched, the presented solutions are also novelty in these two domains.

Many research problems, however, are still under consideration. The decisions which raw data are worthy to on-line collect/provide by monitoring services (which means efforts/coasts to integrate services with various systems, which include these data) in order to better extract the context and support decisions making, have to be made on case basis and are specific for each applications. The methodology on how to analyze cost/benefit ratio for various applications is developed. Alternatively, feature selection approaches fining the 'best' subset of features for the system adaptation have to be investigated [27]. The key research issues to be solved are how to refine the context models. Automatic update of the context model based on the observed changes in environment is a subject of the further research. Another problem under study is how to assure better automatic evaluation and validation of the results to make learning process more autonomous.

ACKNOWLEDGMENT

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A Matching Problem in Electricity Markets using Network Flows

Ryo Hase and Norihiko Shinomiya
Dept. of Information Systems Science
Soka University
Tokyo, Japan
{e1052105, shinomi}@soka.ac.jp

***Abstract*—This paper proposes a many-to-one matching algorithm between sellers and buyers in a deregulated electricity market model that can deal with the limited amount of supply of electricity. Our matching algorithm aspires safe supply of electricity and maximization of social welfare, which indicates overall satisfaction of participants in the markets. In order to satisfy these goals, the matching mechanism is developed based on the concept of a maximum flow problem in graph theory. Additionally, the piled knowledge in the field of economics about matching in markets is also applied into design of this algorithm. Simulation results shows that this algorithm can find whether there is the maximum many-to-one matching in a given model of electricity market or not.**

***Keywords*—bipartite graph; buyer-seller networks; many-to-one matching; maximum flow problem; social welfare.**

I. Introduction

Many countries have constructed electricity grids with centralized structure, in which a few sellers generate electricity and supply it. This structure has arisen from need for providing electricity to buyers safely. Generators in that system must be responsible for safe supply, and they can get adequate remuneration for this responsibility.

The main concerns of electricity buyers are both safe supply and low prices. Recently, failures in the electricity grids have been decreasing more than ever [1]. Thus, the requests of buyers for reducing the electricity prices become more remarkable. However, some problems exist in electricity markets on the centralized electricity grids. First, the sellers do not tend to reduce the prices because they have few competitors. Second, buyers are doubtful about transparency of current prices because no buyers have sellers who can be compared with current sellers.

Recently, there have been some actions for restructuring electricity markets in many countries [1], and deregulation of the electricity market is one of them. This action aims to construct more competitive markets, which are expected to bring technological innovation and lower prices.

In deregulated electricity market, a Power Producer and Seller (PPS) conducts retail of electricity by using electricity grids owned by other generators. The PPS has small capacity to generate electricity; hence, they must get electricity from another generator when demand is beyond own capacity. PPS cannot get electricity as much as they want because they will face a severe business situation if they continue to do that.

Although buyers want to get the electricity from a seller offering a lower price than the others, some buyers cannot contract with such a seller if demand for the seller exceeds its capacity [2]. These buyers must pay higher costs to get electricity from one of the other sellers. In this situation, it is realized that some buyers who can purchase the electricity from the cheapest seller may be satisfied, but the satisfaction of the others will be diminished.

This research proposes an algorithm to find a *matching* between sellers and buyers in a static model of an electricity market. By changing prices offered by sellers in high demand, the algorithm tries to find the prices that satisfy all supply and demand, and give the highest social welfare to us.

In Section II, some related works are introduced. Section III explains definitions of methods used in this research. Section IV proposes an algorithm that finds a matching between the seller and the buyer. Section V discusses simulation results, and Section VI consists of conclusion and future work.

II. Related Work

In 1962, Gale and Shapley discussed a *stable marriage problem* and a *college admissions problem* [3]. The stable marriage problem is a classic problem of *one-to-one matching*, and the matching is called *perfect matching* when the number of elements is the same on both sides of the matching. The objective of the college admissions problem is to find a proper matching between colleges and students. Colleges have the capacity to accommodate some students, and, on the other hand, students must be linked to only one college; thus, this type of matching is called *many-to-one matching*.

After the research done by Gale et al. [3], various researches on matching problems have been conducted. A model of matching problems can be represented as a bipartite graph; hence some researchers consider the matching problems based on *graph theory*.

Matching market is a method that constructs a matching between sellers and buyers in the networked market with a pricing mechanism, and this method is explained by Easley and Kleinberg [4]. The network used in matching market is described as a *buyer-seller network* that is discussed in [5]. The utilization of this type of network provides two merits. First, a buyer can choose a seller independently based on its evaluation of the seller. This is practical for the use in real markets because the markets do not ever have the central coordination of the choice of buyers. Second, efficiency of the market can be examined by social welfare, which is the total of utilities of all market participants. Even though both matching market and general economics deal with markets, they differ in network structure because matching market does not use

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anonymous networks. Matching market deals with networks that has typical structure, in which there are some buyers, sellers, and links between them. Thus, matching market shows how market participants affect each other in the network.

In an ordinary method of matching market, sellers and buyers deal with a single item, and the matching algorithm proposed by [4] constructs only perfect one-to-one matching. This paper proposes an algorithm finding a many-to-one matching in which all buyers and sellers deal with different quantity of electricity. For that reason, this paper integrates methods of a *feasible flow problem* and a *single-source unsplittable flow problem* with the matching market algorithm. A feasible flow problem is an application of a maximum flow problem that finds whether there is network flow that satisfies supply and demand of every node or not [6]. This problem is solved by a *circulation problem* explained by Jungnickel [7]. Besides, a single-source unsplittable flow problem finds the paths from a source to sinks in a network [8][9][10].

III. Definitions

This section explains definitions on network structure and methods used in this research. In this paper, it is assumed that electricity does not decrease anywhere in market models.

A. Market Settings of Sellers and Buyers

This subsection discuses property of the networked electricity market model. A buyer-seller network is represented as a bipartite graph containing two node sets. One is a set N_s containing sellers s_i $(1 \leq i \leq n)$, and the other is a set N_b consisting of buyers b_j $(1 \leq j \leq m)$; thus, the sizes of N_s and N_b are $|N_s| = n$ and $|N_b| = m$.

Each seller s_i and buyer b_j has its *supply capacity* c_i and *consumption quantity* q_j respectively. Let $c_{\min}$ be the minimum supply capacity of all sellers, and let $q_{\max}$ be the maximum consumption quantity of all buyers; then, as in other researches that consider the single-source unsplittable flow problem, $c_{\min}$ and $q_{\max}$ must satisfy $q_{\max} \leq c_{\min}$. In addition, for safe supply, q_j and c_i must satisfy $\sum_{j=1}^{m} q_j \leq \sum_{i=1}^{n} c_i$.

In the rest of this subsection, *valuation*, *price*, and *payoff* will be defined based on the theory of [4]. Definitions of these variables in [4] consider only single item. However, electricity is not a single item. Then, this paper extends definition of these variables to deal with different unit of electricity.

b_j decides valuation v_j $(v_j \geq 0)$, which is the maximum acceptable price of one unit of electricity for b_j. The valuation is also called *reservation price* [11]. Additionally, every buyer has distinct valuation value, and all participants of the market do not act cooperatively; hence, valuation of one buyer is private information for the other buyers. Furthermore, s_i offers a price of one unit of electricity p_i $(p_i \geq 0)$ to all buyers.

s_i and b_j have a payoff $u(s_i)$ and $u(b_j)$ respectively. As in [4], even though payoff of a seller is generally calculated by subtracting supply cost from sales, for simplicity, all sellers produce electricity at zero cost in our model. Therefore, if s_i sells $y(s_i)$ units of electricity to buyers, $u(s_i)$ will be

$$u(s_i) = p_i y(s_i). \tag{1}$$

In terms of buyers, $u_{b_j s_i}$ represents payoff of b_j for one unit of electricity offered by s_i; hence, $u_{b_j s_i}$ is defined as

$$u_{b_j s_i} = v_j - p_i. \tag{2}$$

Because b_j needs to calculate payoffs towards each seller s_i with valuation v_j, let $\boldsymbol{u}_{b_j}$ be a payoff vector of b_j, such that

$$\boldsymbol{u}_{b_j} := (u_{b_j s_1}\ u_{b_j s_2}\ \cdots\ u_{b_j s_n}). \tag{3}$$

Consequently, if b_j purchases $y(b_j)$ units of electricity from s_i, the payoff of b_j will become

$$u(b_j) = u_{b_j s_i} y(b_j). \tag{4}$$

B. Structure of a Preferred-Seller Graph

A *preferred-seller* of a buyer b_j is sellers s_i whose price p_i brings $\max(\boldsymbol{u}_{b_j})$. However, if $\max(\boldsymbol{u}_{b_j}) \leq 0$, there is no preferred-seller for b_j. Each buyer purchases electricity from its preferred-sellers; in addition, for simplicity, no buyer purchases electricity from more than one seller in our model.

A *preferred-seller graph* is an undirected bipartite graph denoted by G $(N_s \cup N_b, E)$, in which an edge set E contains edges between every buyer and its preferred-sellers; therefore, a preferred-seller graph represents possible pairs between buyers and sellers in a market represented by G.

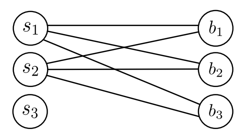
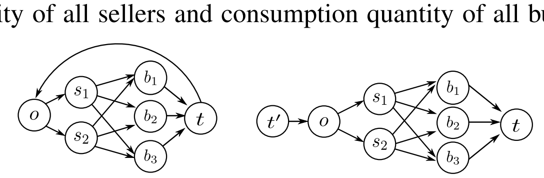


Figure 1. A preferred-seler graph ($|N_s| = 3, |N_b| = 3$).

In the graph G of Figure 1, the payoff of all buyers for both s_1 and s_2 is the maximum payoff, and the payoff of all buyers for s_3 is lower than the payoff for s_1 and s_2. As a result, Figure 1 has edges between every buyer and s_1 or s_2, and does not have edges between every buyer and s_3.

C. Flow Maximization on a Graph

Our algorithm realizes search for a many-to-one matching M on G. For this purpose, algorithm converts G into a directed graph H $((o \cup t \cup N_s \cup N_b), A)$, and considers a feasible circulation problem on H to find the flow that satisfies supply capacity of all sellers and consumption quantity of all buyers.



(a) A model of H. (b) A model of H'.

Figure 2. Graphs changed from the preferred-seller graph in Figure 1.

Any G can be converted to H by three steps. First, remove nodes whose degree is equal to zero, and add two nodes o and t. Second, make arcs (o, s_i) for all i, arcs (b_j, t) for all j, and an arc (t, o). Finally, change all edges in E to arcs $(s_i, b_j) \in A$ (let A be an arc set including all arcs in H). For instance, G in Figure 1 can be converted to H in Figure 2 (a).

Feasible circulation on H must satisfy following two constraints. One is a capacity constraint on every arc in H, and the other is a mass-balance constraint on every node in H.

First, as the capacity constraint, every arc $(v, w) \in A$ must have capacity of x_{vw} that is a nonnegative value of *flow* on (v, w). Additionally, let $ub(v, w)$ and $lb(v, w)$ be upper and lower bound of x_{vw} respectively; therefore, x_{vw} must satisfy

$$lb(v, w) \leq x_{vw} \leq ub(v, w). \tag{5}$$

Figure 3 (a) shows a flow value x_{vw} on an arc (v, w).

(a) The flow on (v, w). (b) A model to show (10).

Figure 3. The flow notation and mass-balance constraint.

For any i, $lb(o, s_i) = 0$, and $ub(o, s_i) = c_i$. By (5),

$$0 \leq x_{os_i} \leq c_i. \tag{6}$$

In addition, for any j, $lb(b_j, t) = q_j$, and $ub(b_j, t) = q_j$; hence,

$$x_{b_j t} = q_j. \tag{7}$$

In terms of (t, o), $lb(t, o) = 0$, and $ub(t, o) = +\infty$. Therefore,

$$0 \leq x_{to} \leq +\infty. \tag{8}$$

As in (t, o), $lb(s_i, b_j) = 0$, and $ub(s_i, b_j) = +\infty$; Thus,

$$0 \leq x_{s_i b_j} \leq +\infty. \tag{9}$$

Second, to consider the mass-balance constraint, difference function $d(v)$ is utilized. $d(v)$ is denoted by

$$d(v) = \sum_{\{w:(v,w) \in A_k\}} x_{vw} - \sum_{\{w:(w,v) \in A_k\}} x_{wv}. \tag{10}$$

For any node $v \in (o \cup t \cup N_s \cup N_b)$, $d(v)$ represents difference between flow values current into v and current from v, and $d(v)$ must satisfy $d(v) = 0$. For instance, in Figure 3 (b), $d(o)$ must satisfy $d(o) = x_{os_1} + x_{os_2} - x_{t'o} = 0$.

The feasible circulation problem on H can be solved by using general methods of a maximum flow problem by changing H into $H'((t' \cup o \cup t \cup N_s \cup N_b), A')$. Any H can be converted to H' by following steps. First, add a node t' to H, and remove an reverse arc from H; then, add an arc (t', o) to H. For instance, H in Figure 2 (a) is converted to H' in Figure 2 (b). In terms of capacity bounds, $lb(t', o) = 0$, and $ub(t', o) = \sum_{j=1}^{m} q_j$. Hence, $x_{t'o}$ must satisfy

$$0 \leq x_{t'o} \leq \sum_{j=1}^{m} q_j. \tag{11}$$

To obtain the maximum flow on H', the algorithm solves following objective function with (6), (7), (8) and (11).

$$\text{Maximize} \quad x_{t'o}. \tag{12}$$

D. Construction of Unsplittable Flow

Even though the algorithm finds flow that satisfies (12), this flow does not necessarily construct a many-to-one matching. In Figure 4, both $t' - o - s_1 - b_2 - t$ and $t' - o - s_2 - b_2 - t$ are

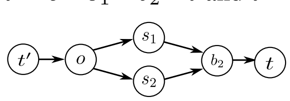


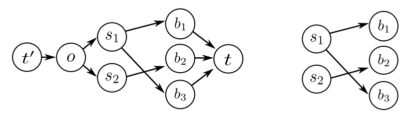
Figure 4. A model shows the constraint of unsplittability.

possible t'-t paths, and this flow is called *splittable flow* if both of these paths supply electricity to b_2. However, only one t'-t path must be selected to supply electricity to b_2 because no buyer can purchase electricity from more than one seller in our model; therefore, if the maximum flow is splittable flow, this flow must be changed into *unsplittable flow* by an algorithm proposed in [9]. This algorithm finds *alternating cycles* and adjusts flow on these cycles to get unsplittable flow.

In Figure 4, let $o - s_1 - b_2 - s_2 - o$ be an alternating cycle. In this alternating cycle, $o - s_1 - b_2$ is a forward path, and $o - s_2 - b_2$ is a backward path. Then, the algorithm decreases flow on the forward path, and augments flow on the forward path until flow on the forward path becomes zero.

E. Social Welfare with Flow Notation

Figure 5 (a) shows unsplittable flow on H' in Figure 2 (b). Because of the assumption that electricity does not



(a) Unsplittable flow. (b) A matching.

Figure 5. Unsplittable flow and a matching on H' in Figure 2 (b).

decrease anywhere in our model, in Figure 5 (a), t'-t path through (s_i, b_j) represents a transaction in which b_j purchases electricity from s_i. Furthermore, $x_{s_i b_j}$ represents quantity of electricity in the transaction between s_i and b_j. Thus, a many-to-one matching M between sellers and buyers can be acquired by extracting all arcs (s_i, b_j) from the unsplittable flow. The matching M is shown in Figure 5 (b).

In Section III-A, both $y(s_i)$ and $y(b_j)$ are defined as units of electricity in transactions of an electricity market. By using the flow notation, $y(s_i)$ can be described as

$$y(s_i) = \sum_{\{v:(s_i,v) \in A\}} x_{s_i v}. \tag{13}$$

By assigning (13) to $y(s_i)$ in (1), payoff of s_i is described by

$$u(s_i) = p_i \sum_{\{v:(s_i,v) \in A\}} x_{s_i v}. \tag{14}$$

Moreover, if b_j purchases electricity from s_i, $y(b_j)$ will be

$$y(b_j) = x_{s_i b_j}. \tag{15}$$

Therefore, (4) can be changed by

$$u(b_j) = u_{b_j s_i} x_{s_i b_j}. \tag{16}$$

Social welfare is defined as the total of all payoffs of buyers and sellers in a market; therefore, by summing up (14) and (16) for all buyers and sellers in an electricity market model, the social welfare of the model can be derived. Thus, social welfare of an electricity market model is represented by

$$W = \sum_{i=1}^{n} u(s_i) + \sum_{j=1}^{m} u(b_j). \tag{17}$$

For instance, social welfare W of the matching M in Figure 5 (b) is calculated in the following manner. s_1 supplies electricity to b_1 and b_3. Hence, $u(s_1) = p_1 x_{s_1 b_1} + p_1 x_{s_1 b_3}$. In terms of s_2, $u(s_2) = p_2 x_{s_2 b_2}$. b_1 and b_3 purchase $x_{s_1 b_1}$ units of electricity from s_1; thus, $u(b_1) = u_{b_1 s_1} x_{s_1 b_1}$ and $u(b_3) = u_{b_3 s_1} x_{s_1 b_3}$. Besides, $u(b_2) = u_{b_2 s_2} x_{s_2 b_2}$. Therefore, the social welfare will be $W = u(s_1) + u(s_2) + u(b_1) + u(b_2) + u(b_3)$.

IV. A Matching Algorithm

This section proposes an algorithm described below, which brings us to a matching between sellers and buyers.

Input: $|N_s|$, $|N_b|$, c_i, p_i, q_j, and v_j for all i, j.
Output: Updated prices and a many-to-one matching.
Seven steps of the algorithm:

1. Set the round number $k = 0$ when the algorithm starts. Construct G at round k, which is denoted by G_k, and let E_k is an edge set E of G_k. The algorithm terminates if there is one or more buyers that have no incident edges.
2. Convert G_k into H'_k that is H' at round k, and let A_k be an arc set A in H'_k.
3. Let M_k be the matching at round k. Discover M_k by solving the maximum splittable flow problem on H'_k. If there is feasible splittable flow, let this flow be called F. If there is not any feasible splittable flow, proceed to step 7.
4. Find alternating cycles in splittable flow F until no alternating cycles can be discovered in F; subsequently, augment or decrease flow along the alternating cycles.
5. Let $M_{\max}$ be a many-to-one matching that is feasible and maximizes social welfare. If M_k is $M_{\max}$, the algorithm terminates. The prices bring us to M are called *market-clearing prices*. If M_k is not $M_{\max}$, raise prices of sellers in H'_k by one unit.
6. Let W_k be social welfare derived from (17) on M_k.
7. Set $k = k + 1$, and back to the step 1 to start next round.

Before the algorithm starts, input must be initialized. The algorithm repeats its *round* that consists of seven steps and does not stop until discovering whether $M_{\max}$ exists or not in the preferred-seller graph with the given pattern of inputs.

The algorithm has two termination conditions; one is described in step 5, and the other is denoted in step 1. First one is trivial because it is the objective of this paper. Correctness of second one is described below. The prices offered by sellers do not decrease in the algorithm; for that reason, a buyer cannot purchase electricity from any seller if no seller offers a price that is lower than valuation of the buyer. Therefore, in that case, the algorithm terminates and finds that $M_{\max}$ does not exist in that graph and input pattern.

V. Simulation Results

To analyze accuracy of the proposed algorithm, a simulator of the algorithm has been developed with JAVA. This simulator was used to collect data about the transition of prices, preferred-seller graphs, and social welfare at every round.

Table I
THE PATTERN OF INPUT FOR THE SIMULATION.

$\|N_s\|$	$\|N_b\|$	c_1	c_2	c_3	q_1	q_2	q_3	q_4	q_5	q_6	q_7	q_8
3	8	160	160	160	10	20	30	40	50	60	70	80

p_1	p_2	p_3	v_1	v_2	v_3	v_4	v_5	v_6	v_7	v_8
4	2	1	5	6	7	8	9	10	11	12

Table I shows an example pattern of input for the simulation, and Table II displays one of the heuristic results on the given

Table II
TRANSITION OF THE VARIABLES IN THE SIMULATION.

k	p_1	p_2	p_3	$y(s_i)$			$y(b_j)$							
				s_1	s_2	s_3	b_1	b_2	b_3	b_4	b_5	b_6	b_7	b_8
1	4	2	1	0	0	160	10	0	30	40	10	0	70	0
2	4	2	2	0	160	160	10	20	30	40	50	20	70	80
3	4	3	3	0	170	150	10	20	30	40	50	20	70	80
4	4	4	4	70	150	140	10	20	30	40	50	60	70	80

k	$u(s_i)$			$u(b_j)$								W
	s_1	s_2	s_3	b_1	b_2	b_3	b_4	b_5	b_6	b_7	b_8	
1	0	0	160	30	0	150	240	70	0	630	0	1120
2	0	320	320	20	60	120	200	300	140	560	720	2760
3	0	510	450	10	40	90	160	250	120	490	640	2760
4	280	600	560	0	20	60	120	200	300	420	560	3120

input. In Table II, prices at round 4 satisfy all supply capacity and consumption quantity, and these prices maximize social welfare. Therefore, the matching M_4 is $M_{\max}$, and the prices at round 4 are called market-clearing prices for the given input. Figure 6 shows the structure of M_4. Every arc in M_4 indicates

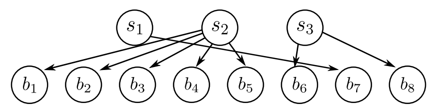


Figure 6. The matching M_4 in the simulation.

a transaction between a buyer and a seller, and flow on each arc is equal to the units of electricity dealt with in the transaction.

VI. Conclusion and Future Work

This paper proposed an algorithm discovering a many-to-one matching between buyers and sellers in a static electricity market model with the set of electricity prices. By the computer simulation, accuracy of the algorithm was examined.

In our definition of social welfare, the payoff of each seller is equal to the price offered by the seller. Nevertheless, in the real markets, sellers must pay the production costs that reduce a profit of the sellers; hence, more appropriate settings at that point is required to obtain more accurate social welfare in an electricity market. In addition, the concept of decrease of electricity on power lines will be integrated into the model of our algorithm. These assignments are our future work.

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Telemetric Data Visualization of Vehicle's Crew in Geographic Context

Pavlina Nudzikova
Faculty of Electrical Engineering and Computer Science
VŠB - TUO
Ostrava, Czech Republic
pavlina.nudzikova@vsb.cz

Petra Rajmanova
Faculty of Electrical Engineering and Computer Science
VŠB - TUO
Ostrava, Czech Republic
petra.rajmanova@vsb.cz

***Abstract* - The work is focused on information systems and design of web applications that visualize telemetric data and track of vehicle in geographic context. All the measured values and actual location are shown on maps provided by Google. For this purpose, a web application was designed and implemented. This application is able to show data of vital functions in real-time (electrocardiography, heart rate, respiration, oxymetry, etc., as an example of future needs) and basic information about the vehicle, such as location of the vehicle, temperature of interior, acceleration, etc. Collection of data in database with appropriate tools for its acquisition and indication is the aim of the work described in this article. This is starting point for post-processing and data analysis for specialists and experts.**

Keywords* - *Biological signal; Telemonitoring, GPS; GIS; ASP.NET; MySQL.

I. INTRODUCTION

If we talk about the means of transport, everyone would agree that the most important value is the vehicle's location in this particular context. It would be useful if the geographical coordination was drawn into map on computer and thanks to this you will obtain some important information, such as driver's vital signs, and crucial data about the vehicle itself.

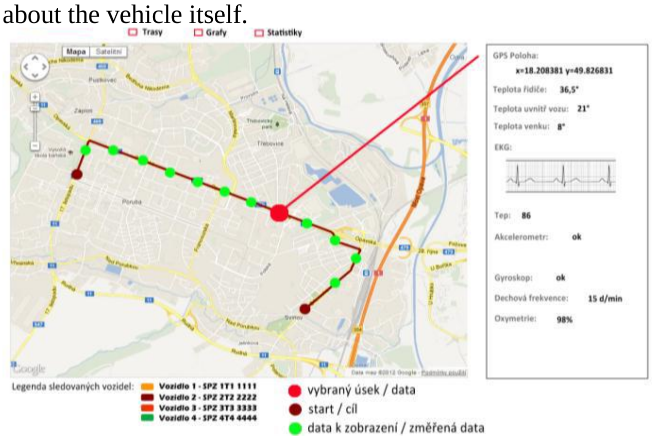

Figure 1. Concept of designed application.

Let us consider an information system that allows us vehicle tracing in real-time, to show current biological data of driver and important data about the vehicle itself for each particular time interval. In the other words, for the current vehicle's position on the map, there will be shown following measured health-status values, such as breath rate, heart rate, temperature or ECG curve, in the system. Of course, for somebody, these details could be insignificant, but for doctors, paramedics, or similarly educated people it could represent very important information. In addition to this, other information obtained from accelerometer and gyroscope could be provided. This information should be indicative of the fact that, for example, particular vehicle had an accident. The usage of such information system is quite extensive, both for fire-fighters, racing drivers, professional drivers, etc., and moreover, it is not necessary to use the system in cars only.

Section II reports the state of the art, following by three sections covering theory of the data sources (biological signals in Section III, geographic informations in section IV, vehicle and its parts usage technological data in section V). In the VI section there is described application design. Then there is measurement and data collecting chain description section. Software solution is presented in section VIII where are used components described. Section IX shows working user interface with a few examples. The last section of this article contains list of used technologies with appropriate description.

II. STATE OF THE ART

Nowadays, it is possible to buy interesting Global Positioning System (GPS) trackers that are able to record position and work with these data offline or online. Real-time behaviour, integrated into a single embedded system and openness of these solutions might be a problem. Another problem could also occur with operating parameters of monitored cars that are often very specialized and manufacture-oriented. Currently, there is a lot of work on current status of the driver [2], [3], who has major impact on the road traffic.

The aim of this work is to present an embedded platform that would enable collecting data from the heterogeneous resources and make the real-time visualization of data. The department of cybernetics and biomedical engineering created several prototypes (Kaipan Voltage) of electric cars in 2009-2012 with industry partners from the Czech Republic (see Fig. 8).

The next step was to implement an experimental system for above mentioned problematic areas.

We prepared solutions for all the data collection and processing that provide interesting results in improving safety of car driving with crew's interaction, looking for driver's stress factor and dynamics of load during driving. It is open source solution in a strong experimental phase where the driver is depersonalized at that time. It means that the work is oriented on data measurement, transmission and processing.

In [4], an embedded sensor system that collects physiological data, such as electrical activity of heart muscle, heart rate, breath rate, temperature and mechanical data as acceleration with built-in sensors, was designed. The acquired data are transmitted to the vehicle via Bluetooth communication interface. The data visualization is designed for Personal Digital Assistant (PDA). This application is usable for both, for displaying sensor data in real-time and for recording measured data to internal memory of PDA or to some other data medium.

Application for PDA was designed in [7]. The PDA is used for displaying actual values of process variables – actual speed, average speed, distance, battery's voltage and current, hydrogen fuel cells voltage and electrical motor current. It is also connected to programmable controller by serial communication link.

On the other hand, our work can indicate the vehicle location on the map, together with driver's biological signals in real-time. This is the advantage because we are able to see where the vehicle is at any given time and what the driver's vital signs are, at the same time. The acquired data are transmitted to database via GSM (originally Group Spécial Mobile) network.

III. BIOLOGICAL SIGNALS

Biological signal (biosignal) is a signal that is used in biology and medicine to represent information about the observed biological system (human body). These signals may be courses of electrical voltage, variable magnetic fields, changes in chemical concentration, mechanical motions, sounds, temperature changes, etc. We can record those signals as results of spontaneous activity of biological system (native signals), or as results of any intentional stimuli (evoked signals, provocation, etc.) [1, 2, 4]. The most often measured biosignals are:

- ECG
- Blood pressure (invasive or non-invasive measuring method)
- SaO2 / plethysmography
- Parameters related to breathing
- Temperature
- Analysis of anesthetic gases

A. Electrocardiography (ECG)

ECG is diagnostic method used to capture and record electrical activity of heart. It is basic method for examination in cardiology. It allows us to detect any abnormal heart rhythms (called arrhythmias), ischemic changes in myocardium, check effectiveness of cardio drugs, etc. The entire activity of heart is accompanied by emergence of an electrical signal. The graphic record is called electrocardiogram (ECG). Signals are sensed by electrodes placed on patient's skin. Any disorder of production or distribution of nerve impulses can affect not only the mechanical activity of heart, but also the shape of electrical signal. [3]

B. Heart Rate

Heart rate is one of basic and most frequently monitored physiological data, especially in clinical medicine, work and sports medicine. It is an accurate indicator of activity and performance of heart. Heart decreasing below certain limit or increasing above certain limit, or even irregular changes might indicate serious disturbance in the activity of heart, so called arrhythmia.

Pulse is a pressure wave which is caused by expulsion of blood from the left ventricle into aorta, from where it spreads to other arteries throughout the entire body. In medicine, series of these waves correspond to cardiac rhythm and heart rate. The heart rate indicates the number of beats of heart within one minute.

C. Respiratory

Respiration is a process of gas exchange between an organism and environment. During normal calm breathing an average adult breathes 6-7 litres of air per minute and the respiration rate is 12-14 breaths per minute. The respiration rate is evaluated mainly on the basis of other measured biological signals. For example, it can be evaluated from electrocardiogram that is modulated in the rhythm of breathing. Respiration is affecting and closely related to ECG records, e.g., it affects the change of amplitude of ECG.

D. Body Temperature

Measurement of body temperature is one of the oldest methods in medical diagnostics. We differentiate between contact and contactless temperature measurement of human body. In contact methods, the measuring device is (thermometer or only its sensor) directly touching the tissue whose temperature is measured. The heat is transferred from the tissue to the thermometer, directly by this contact. The opposite way is the contactless method where heat transmission from the tissue to the thermometer passes through the surrounding environment.

The body temperature of each individual person varies over the day, and it is usually higher in the evening. The temperature can be increased due to physical activity, mental effort, after a meal or, using certain drugs, and the height of temperature (even humidity of the surrounding environment) can impact.

E. Oxymetry

There are two ways of blood oxygen saturation measurement – invasive and non-invasive. Oxygen is transferred physically into the blood, dissolved in blood plasma, and chemically bound to blood pigment haemoglobin. Four oxygen molecules can bind themselves into one haemoglobin molecule. This creates oxy-haemoglobin, the reaction is reversible and repeatable. Within the chemical bond 70-times more oxygen is transferred than in physical dissolution. Oxygen is transported primarily by chemical bonds.

The amount of oxygen transferred by chemical bonds is expressed by blood oxygen saturation SaO2 (in arterial blood) and SvO2 (for venous blood). Their units are given in %.

IV. GEOGRAPHIC INFORMATION SYSTEM

The concept of Geographic Information System (GIS) is commonly used to refer to computer-based systems for processing geographic data, presented mainly in the form of various maps. The advantage of GIS in comparison to common maps is that it consistently separates functions of the maps, the storage of geographic data and their presentation, and adds even more options, such as spatial data analysis. Then the same data can be easily updated, analyzed and represented in different ways, so it can satisfy different user requirements at much lower need for compromise.

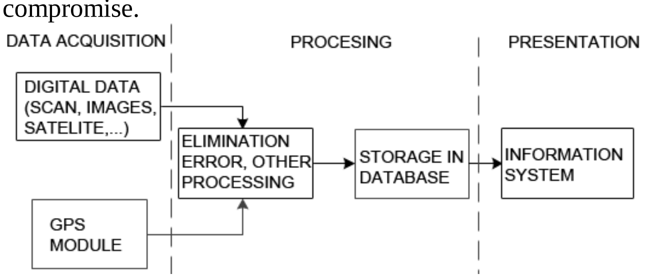

Figure 2. Flowchart GIS.

V. VEHICLE DATA

As well as human biosignals determine their health status, information about vehicle can also determine its characteristics and its condition. The overview of health status of vehicle's driver together with the characteristics and location of vehicle displayed on the map at the same time allow us to track these important parameters and to take actions if it is necessary.

A. *Location of the Vehicle*

The most important information about vehicle is definitely its location [5, 8]. The latitude and longitude of vehicle can be determined by using GPS coordinates. This information is obtained from the GPS module installed in the vehicle.

B. *Temperature of Interior*

Another important variable is the interior ambient temperature since changes occurring in the environment temperature are affecting thermoregulation of human body. The change of ambient temperature has high serious impact on human body and it can result even in overheating or hypothermia.

People belong to warm-blooded animals in which the core temperature of inner body is kept at 37° at normal conditions. Maintaining of temperature is possible only if the heat production in equilibrium dispense. This is done by system that is able to control body temperature by various thermo regulation mechanisms – (thermoregulation). Body temperature depends on the generation of heat and on other the external and internal factors.

C. *Accelerometry*

Accelerometry is a method of sensing acceleration using accelerometer's sensors. Accelerometers are used to measure acceleration. Generally, acceleration characterizes the rate of velocity change of mass point or the entire system of mass points in time.

D. *Gyroscope*

Usage of gyroscope is another way how to measure movement in space, change of position and angle or rotation. Gyroscopes are well known and used for measuring and determining the change in position or rotation in a perspective of any object to which they are attached. The gyroscope is a device for navigation and direction. Actually, it is the flywheel; heavy wheel rotating on bearings with little friction. The rotating flywheel has momentum, so that its axis without external forces is kept in the same direction. It means that it is a device that can determine its orientation in space (flywheel never changes its position during the rotation and thus gives information about the orientation of the vehicle in space). Gyroscopes are widely used for measuring angular velocity in units of degrees per second (° / sec), for example, to see how quickly the measured object is rotated.

VI. APPLICATION DESIGN

The application will display biosignals that are gathered by the sensor systems. Signals will indicate the driver's health and the current load of his organism together with the signals reporting the status of the vehicle. These signals are quite important since we know where the vehicle is, how the driver's body reacts in different situations, and in the case of health problems an immediate assistance can be provided. It should be noted down that all these values are gained from devices that are placed on-board of the vehicle.

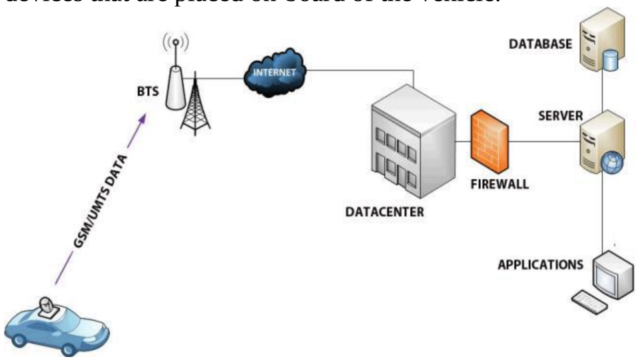

Figure 3. Data procesing.

First of all, the process of acquiring and processing data in general is described there. Then data processing will be particularly described in the application.

In Figure 3, there can be seen the whole process of communication. Data obtained from the modules located in the vehicle using GSM (or Universal Mobile

Telecommunication System) are transmitted via internet to central location (data-centre). In data-centre, servers and applications are located. Then, on these servers data are processed and stored in database. The application is finally accessing those processed records and performs and offer further necessary operations (calculations, visualization, etc.).

VII. MEASUREMENT CHAIN

Measurement chain consists of three basic blocks, namely, remote devices, server, and the client. These individual blocks communicate with one another via Internet network.

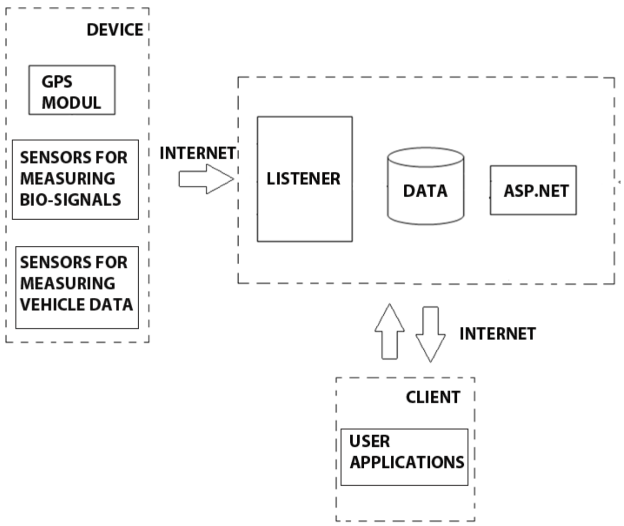

Figure 4. Measuring chain for the acquisition, processing and displaying data.

A. Remote Devices Block

This block contains modules for data measurement. In our case it will include GPS module that allows us to get information about where the vehicle is located. We could use Smartphone for localization and some other information, such as acceleration, but free applications did not meet our requirements. It is necessary that the data string contained to have a unique identification key.

Then sensors for measuring biological signals by which data indicating the driver's vital signs are obtained. Finally, sensors for measuring vehicle's data are needed. These sensors can be selected according to the purpose (temperature inside, temperature outside, tire pressure etc.). There is an attempt to make the application universal as much as possible for different usages so that it is necessary to choose the measured data (sensors).

It is possible to use it generally, but this work was focused on the vehicle Kaipan Voltage.

B. Server block

This block contains a service called "Listener", then relational database and robust applications implemented using ASP .NET framework [6].

Listener service allows us to listen to the particular server port and to have connection handling of one or more clients. The purpose of this is to receive pre-process data- streams. This means that data is not only received and stored into selected database "telemetriedb" but there is parsing of data performed to the individual data segments on beforehand. With such prepared records, we can then proceed with further processing and examine these in the application.

Data are stored in the standard MySQL database called telemetriedb. The database uses standard web interface phpMyAdmin for access to database. Communication with the DB is using the SQL query language.

Both blocks, the remote devices and the server, are communicating as client-server. The remote devices represent the client in the meaning of the network architecture and the server block is obviously the server.

Protection of personal data is not necessary because we do not indicate sensitive information. That would point out on a concrete user. Database is secured with password.

C. Client Block

The final block is the client. It is representing user application (user interface) and it is the only thing from the whole solution that the user will be able to see and work with. The goal was to create application's user interface as much clear and intuitive as possible, and in the way to have good view of the current health status of driver. On the welcome page, there is user guidance what applications you are allowed to see and what operations can be performed there.

VIII. SOFTWARE SOLUTIONAPPLICATION DESIGN

A. Listener Service

Service "Listener" is used for receiving and parsing data strings that come from each device. After Listener service starts, first it initializes required data and begins to run an infinite loop that waits for a client connection. The algorithm flowchart is shown in Figure 5.

When the device or client connects to the server (Listener service), a new thread is created. This thread is handling the processing, the requirements of any particular client. The thread is waiting for receiving data from the client. When the data are received, it converts them from a stream to a text format, the received data "received telegram" starts to process. Then, the data are stored in database. After that based on IMEI identifies the device type (which type of device was sent). Based on the type, it begins with parsing of the data. Once data are parsed to an object, then the object is stored into database.

B. Data Sending

Due to the unavailability of functional devices that could send real data directly to the server, we were forced to simulate this data. The concept of simulated data is very simple.

For the data that describes biological functions of driver and vehicle data (NOT its position) a simple simulation tool based on limited random values that create artificial data strings (received telegrams) was created. Those data strings are sent to the service Listener.

To be able to display the vehicle's path, it was needed to simulate the whole track. The values of its trajectory are gradually stored in data string (received telegram) as well. This is sent towards the server, as well.

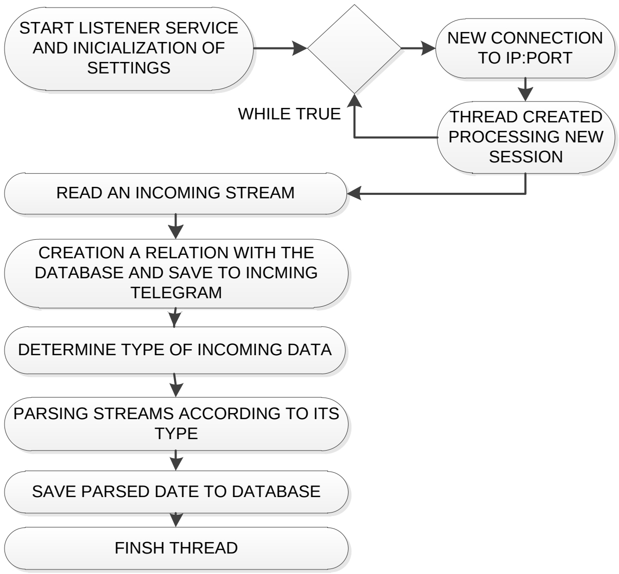

Figure 5. Flowchart Listener service.

C. *Loading Map*

Maps are implemented by busing the Google Maps API that allows us to load the maps into the web application using JavaScript. In this work, we use the Google Maps API v3, that is free and available for non-commercial purposes.

IX. USER INTERFACE

Here, one can see how the application looks like and what its functions are. In Figure 6, we can see a part of the application that could be interesting for the end-users. Tabs Car, Car type, Contact persons and Equipment are used for user’s communication with the database. It is a user-friendly interface for users who are not aware of SQL query language since the data can be inserted, edited or deleted directly in the web application by simple mouse clicking.

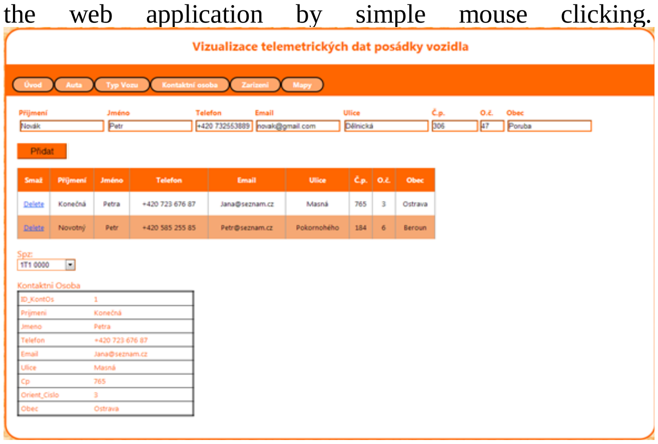

Figure 6. Edited data from the database in application.

For example, in Figure 6, there is shown contact person tab where you can add or delete a new contact person. There is also a selection box for the license plate that can help the user find contact persons related to the particular registered vehicle. Other tabs are designed in very similar way, everything straight and easy for the end users.

The most important part of the application is shown, if you open the Maps tab. It is practically the main part of the application that allows the user to view vehicle on the map, and displays all the driver’s important biosignals and the current information about the vehicle.

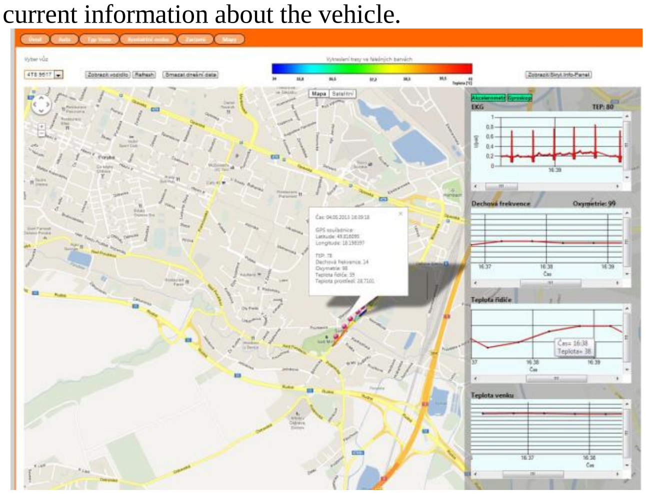

Figure 7. Edited data from the database in application.

X. SYSTEM IMPLEMENTATION

One of the main parts and the most essential task of this work was to design and create database that stores all data and is able to process them later. For this purpose, MySQL database was used because its function adequately covers the requirements of the problem. When using an application together with using all the client devices huge amounts of data that limit the processing speed and could clogging up the database are stored. Therefore, it was necessary to solve automatic or manual data clean-up that are no longer required, and solved by the application itself. To redeem the adequate processing speed of the application, the individual procedures were highly optimized within the database (creating additional indexes for parameters that could be searched out of the data).

Once this important task was resolved, it was decided what data are needed to be registered (in what format they should be obtained), it was necessary to design and create a service called “Listener”. This service was created by using the Microsoft .NET Framework as a service for Windows OS (the server). Its main task is to listen to a specific IP address and port. These parameters can be easily changed because it is possible to use the service in any network, on any server where software tools (Microsoft Framework & IIS) are available. Service can receive and handle multiple concurrent connections (called threads) simultaneously. Unfortunately, there is fixed International Mobile Equipment Identity (IMEI) parameter set up that is used to differentiate the device type, the service that should react appropriately and find out the type of communication. In practice, this means that the distinction between the types of devices is made on the basis of fixed IMEI. In other words, according to the IMEI chain the service will recognize that it is, e.g., GPS module and for that purpose there is set particular parsing routine. However, this can be removed by expanding data structure that would allow detecting the type of device based on IMEI automatically.

For the visualization of map data, the GIS system from Google, available for free, was used. More specifically, the "Google Maps API v3" were used where the application is written in ASP .NET with JavaScript and C# programming language. For the visualization elements, "chart.components" were used. We have chosen the type of graphs that can display static charts only. The application allows users to view route of one selected vehicle (however it is possible to switch among vehicles). Route is drawn in false-colours according to driver's body temperature.

It is possible to view the measured data in the given time points, for both the car and the driver, and also in every point of measurement along the route. It is also possible to display the information panel where graphs of the measured values related to time are.

The application can be further improved and expanded. First of all, the expansion of visualization with providing of information about other parameters (e.g., engine temperature, vehicle speed, etc.) could be. Then, there can be implemented feedback between the vehicle and the visualization part which would enable automatic detection and response to changes in essential signal characteristics. It might be needed to use dynamic graphs that could display the transparent visualization of biological data, as well. Another important improvement could be auto-call, the pre-defined contact person in case of sudden changes in vital signs or during an accident. To react on the current world trends, creating of suitable application for mobile phones with the Android OS or Apple iOS, which could allow drivers to check their vital signs and conditions of their cars directly in their vehicle, would be useful.

XI. CONCLUSION

The main goal of this work was to create an application for displaying Telemetric Data of vehicle's crew at map background. Since we did not have any devices that would generate real data and send them to the database server, the data were simulated.

This work follows [4] and is extending it by the amount of captured data. Not only information about drivers are captured, but also vehicle's data. All the values are displayed on the map then.

The system was created to show biometric and operating data on the map. For testing purpose, the system was designed for electric Kaipan. It shows information about crew's health status during vehicle testing. System is used for ergonometric measurements of vehicle's prototype during driving tests. The system is able to interconnect telemetric data and crew's biometric data together for prototype of electric vehicle Kaipan Voltage and shows joined data in geographic context. The result is that driver's subjective feelings are extended by objective measurement data.

Figure 8. Kaipan Voltage prototype.

ACKNOWLEDGMENT

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Smart Metering based on Wireless Networks for Improved Water Management

Ho-Hyun Lee, Sung-Taek Hong, Gang-Wook Shin
K-water Research Institute, K-water(Korea Water Resources Corporation), Daejeon, South Korea,
dlghgus@gmail.com, sthong@kwater.or.kr, gwshin@kwater.or.kr

***Abstract*— In the field of water resources management, the application of Smart Water Grids (hereafter SWG) has rapidly gained traction as a popular method for achieving stable water supplies and minimizing the wasting of water. However, unlike the smart grids that have been applied to electricity and gas sectors, Smart Water Grids are more complicated due to the fact that water meters are typically installed under the ground making power supply and data transmission more difficult. This study takes full advantage of analog meters by using L(inductor)/C(Capacitor) sensors in the water meter counters for the purpose of developing various wireless communication methods in order to find an optimal transmission method. Wireless communication systems were examined according to their specific frequency environmental conditions to optimize the pipeline network operation and management.**

Keywords - Smart Meter; Smart Water Grid; Wireless Communication.

I. INTRODUCTION

Smart grid technologies are being studied actively for the efficient production and supply of energy against climate change and large-scale power shutdown [1-2]. In the field of water resource management, the popularity of smart metering systems is increasing based on their abilities to obtain water usage data in real time which are needed to ensure a stable water supply, reduce water treatment costs, and meet unexpected water demand increases.

The price of tap water is very low compared to the cost of electricity and gas in Korea, and the cost of installing mechanical water meters is relatively cheap. On the other hand, the use of mechanical type meters presents many challenges in terms of implementing real-time data transmission for smart metering. In addition, smart metering systems do not work well due to the additional constraints of difficult installation conditions which are much different from the electricity and gas sector [3-4].

In Korea, pipeline networks are mostly installed with the meters but they are very slowly developed. There is still work to be done so these networks become efficient and to the point where they can ensure the safety of water quality and the consistency of water pressure and structural integrity. Technical approaches, such as monitoring and sensor technologies and optimization techniques, are needed in order to overcome the weak points [5].

The concepts and technologies applied to achieve intelligent water distribution systems have sparked a global movement to maximize the efficiency and to optimize the operation of water distribution networks. Various systems are being developed by combining a variety of technologies like water quality and quantity sensors and pipeline network software. To achieve a sophisticated and intelligent operation and management distribution network, the most important priorities are real-time leak detection and water quality monitoring in cases of emergencies. Thus, the development of new types of meters, sensors and integrated operation systems should be developed using cutting edge information and communication technologies.

II. DEVELOPMENT OF SMART METERS

A device used in each home’s water meter diameter 50mm or less can be classified into direct and indirect measurement. The indirect method depends on the rotation speed of the actual flow and the direct method measures the constant volume of water, which is used primarily for testing.

Half-electronic meters have the mechanical type of sensing and transmission to convert electrical signals, but do not have any internal microprocessors. There are two kinds of mechanical types of water meters. One of them is the pulse counting method which uses a lead switch and the other is the camera method which takes and sends photos, which through image processing, can be analyzed. A digital device should be equipped with a microprocessor in order to transmit real-time data.

An electronic water meter consists of sensors and additional indicators for flow measurement. The meter has a microprocessor which allows for internal data processing and storage by detecting electronic signals and displaying them in numerical values. A variety of electronic meters have been developed using lead switches, hall sensors, and magnetic sensors worldwide [6].

Electronic meters, in general, still have some limitations. For example, they are relatively expensive, frequently damaged, and their use might not be extended nationwide. Therefore, a half-electronic meter is considered as an alternative, which can use the robustness and cheap cost of the mechanical meter by adding a sensing unit. In addition, while preserving the advantages of an analog meter, it could transfer the data from three kinds of wireless transmitters, which are as follows: 424MHz Single, 424MHz Multi, and 2.4GHz ZigBee transmitter.

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A. Development and Performance Test of Flow Meters

Performance analyses for water meters were carried out empirically, and their impellers, upper plates and lower plates, nozzles and adjustment screws were the decisive factors for determining performance [7]. In case of calculating hourly flow from the meters to check their accuracy, the following equation should be applied to multiple their diameter and velocity.

$$Q = A \cdot V = \frac{\pi \cdot D^2}{4} \cdot V \qquad (1)$$

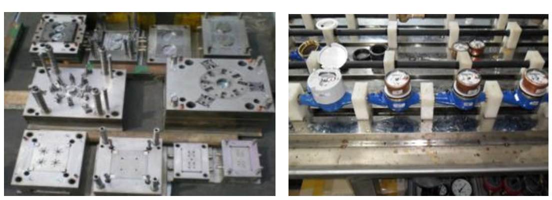
where, Q : flow(m^3 / h), A : pipe unit area(m^2)
V: Velocity(m / s), D: pipe diameter(m)

In case of the mechanical water meter, hourly flow was not used, but the integral flow was used instead in order to check the rotation of the impeller. Its equation is as shown in (2).

$$Q = \frac{1}{k} \cdot N \qquad (2)$$

where, Q : integral flow(m^3),
k : criteria constant(rpm / m^3)
N: rotation speed of impeller

In this study, the final mold of the water meter was determined through 12 experiments to investigate the effects of measurement accuracy by each factor or combination of factors with each other. Finally, the mold for the water meter was completed like in Fig. 1.



(a) Final Mold (b) Water Meter

Figure 1. Overview of the Water Meter

The developed meter should transfer its data to a main receiving center through wireless communication. Fig. 2 shows that a sensing and communication module can be added to the water meter for transmission. The module can be used only as a mechanical meter as shown in the left figure, but if necessary, the special module can be added to transmit the sensed data by wireless communication as shown in the right of Fig. 2.

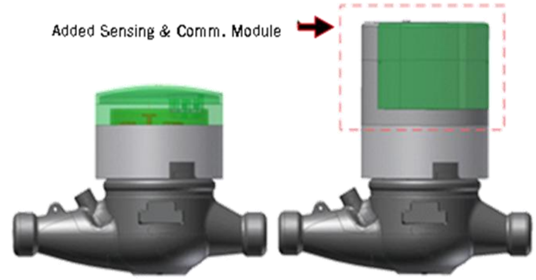


Figure 2. Configuration of Smart Meter

The sensing part of the meter adopts a L/C sensor in order to receive the data from the mechanical water meter on the bottom. Fig. 3 shows the configuration of the sensing coil and rotating plate. The L/C sensor is divided into two parts, which are metallic and non-metallic, to sense the variation of waves being rotated by the mechanical meter. Variations can finally be easily digitized and sent to remote sites.

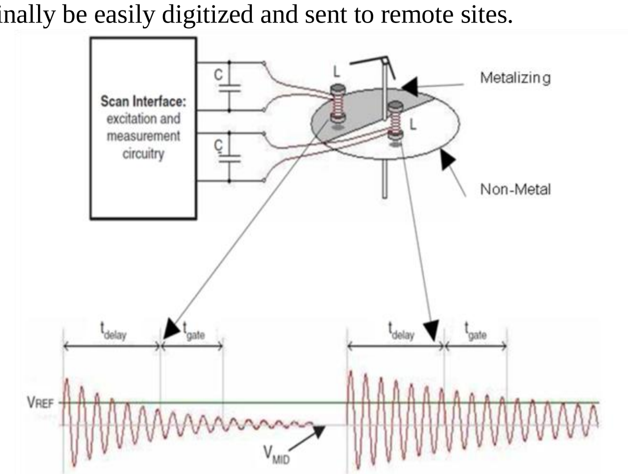


Figure 3. Configuration of Sensing Coil and Rotation Plate

The L/C sensor signal gives two data, as shown in Fig. 4. In case that the sensed level is lower than Vref(Reference Voltage), it emits a signal value of zero which means it is going through a reflection panel. In case of a higher level than the reference voltage, it emits a signal value of one which means it is going through a non-reflector. Signal 1 has more voltage compared to the others owing to not having a reflection plate. Even though signals 2 and 3 have the same conditions as having a reflection plate, they have different attenuations which can be caused by their distance, size and material. Thus, signal 2 in the middle of the figure shows it senses one even in a reflection plate. To prevent this situation, the reference voltage has been designed to automatically change in order to compensate the error by the distance between the L/C sensor and reflection plate.

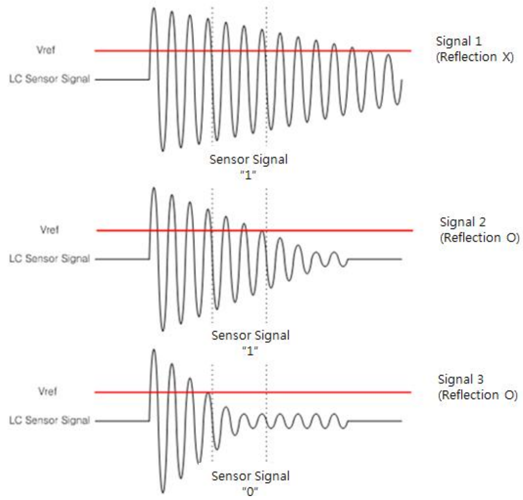


Figure 4. Sensor Signal from L/C sensor

B. *Development and Performance Test of the Transmitter*

In this study, three kinds of wireless communication methods, 400 MHz single-band, 400 MHz multi-Band, 2.4 GHz ZigBee were selected to find the optimal communication method by analyzing the communication characteristics of difficult water meter installation sites.

400 MHz Single-band adopts the module with a central frequency of 424.750 MHz and 2 level FSK modulation as shown in Table 1. In addition, 400 MHz Multi-Band applies the frequency range of 424.700 MHz ~ 424.950 MHz and channel spacing is 8.5 KHz, resulting in a module which has 20 channels. 2.4 GHz band ZigBee adopts the IEEE 802.12.3 standard module with O-QPSK modulation scheme and data rate of 250 KBPS, as shown in Table I [8].

At first, these three kinds of wireless modules, two 400 MHz band and one 2.4GHz band, are designed and manufactured through laboratory tests in order to guarantee stable transmission. Personal Data Assistant (PDA) was also developed with a specific inside module to receive transmitted data from the three modules for field testing. The PDA also has a special software program for monitoring data and analyzing communication environment from water meters, which makes it possible to conduct field tests with mobility and convenience.

The software, based on Win CE, has been designed to allow for selective input data such as weather, distance, location and cover type. Inputted and measured data are saved in the database to transfer to an analytic computer program, by which we can know the best results among the three modules. In Fig. 5, an image of the PDA screen shows the test program to input measuring conditions and calculate its transmission results. The image on the left side in Fig. 5 shows the parameter setting screen such as weather, position, distance, casing, and test number to do performance test. The image on the right side shows the results of a ping test. These tested data were accumulated in the database in order to analyze them.

TABLE I. CHARACTERISTICS OF WIRELESS COMMUNICATION

Type	Item		Specification
400 MHz Single -band	Front-End	Central Freq.	424.750 MHz
		Freq. Deviation	2.5 KHz
		Modulation	2 Level FSK
		Output	Less than 10 mW
		Sensitivity	-115 dBm
		ANT. Gain	Less than 2.3 dBi
	Comm. Method	Front-End	PWM
		Ext. I/F	UART 9600 BPS
400 MHz Multi -band	Front-End	Freq. Range	424.700 MHz ~ 424.950 MHz
		Freq. Deviation	2.5 KHz
		Modulation	2 Level FSK
		CH. Interval	8.5 KHz
		CH. Number	20 Ch
		Output	Less than 10 mW
		Sensitivity	-115 dBm
		ANT. Gain	Less than 2.3 dBi
	Comm. Method	Front-End	Manchester
		Ext. I/F	UART 9600 BPS
2.4 GHz ZigBee	Front-End	Freq. Range	2.4 GHz ~ 2.483 GHz
		Modulation	O-QPSK
		Data Rate	250 KBps
		Channel Num.	11 Ch ~ 26 Ch
		Output	Less than 10 mW
		Sensitivity	-85 dBm
	Comm. Method	Front-End	SPI
		Ext. I/F	UAT 115 KBPS

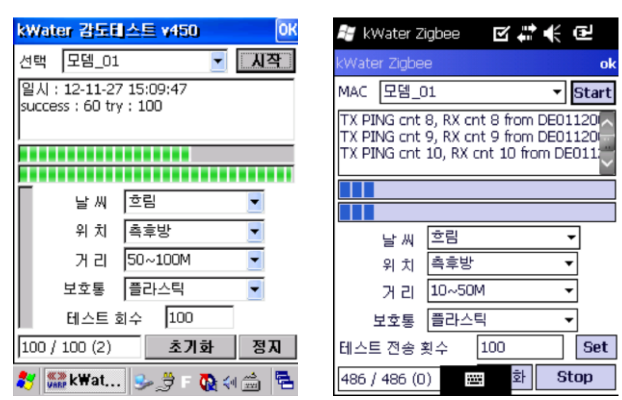


Figure 5. Operation Display of the PDA Program

Because the industrial PDA was designed for a low powered wireless communications (424MHz, 2.4GHz) module to receive data from smart meters, it was very easy to test the three kinds of developed modules by changing real distances, lab and open space environments, and then, field tests were carried out after each RF module was set to have the same performance. Table II shows each module's electric field strength and communication length using a spectrum analyzer.

TABLE II. ELECTRIC FIELD STRENGTH AND LINE OF SIGHT

RF Module	Electric Field Strength	Distance	Remark
2.4 GHz Zigbee	-21.94 dBm	640 m	Height : 2 m
424 MHz Single	-22.10 dBm	630 m	
424 MHz Multi	-22.06 dBm	630 m	

Most of the water meter covers are made of polyethylene, galvanic, or iron casing, as shown in Fig. 6. Each case has its own propagation characteristics. They were examined by conducting field tests.

(a) PE Case (b) Galvanic Case (c) Iron Case

Figure 6. Cover Types for Water Meters

Fig. 7 shows the four installation sites in Gwang-Ju City, Kyeung-gi Province, South Korea. Each section has its own unique characteristics such as congestion, located in quite areas, or being nearby to main roads. Performance testing was done after installation of the three kinds of RF modules at 40 points and selecting the possible direction among the front, side and rear position.

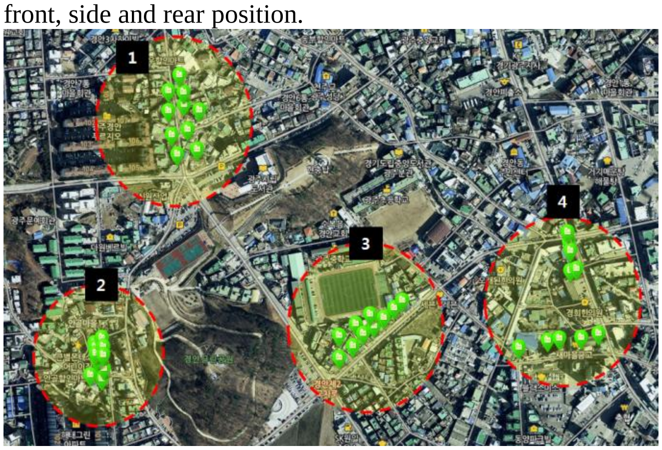


Figure 7. Testbed for Performace Test

Data and field work photos was collected into the database server to analyze the performance, as shown in Fig. 8. The database was constructed using a web program for the purpose of making accessibility easy for authorized persons when onsite.

(a) Analysis of Distance (b) Characteristics

Figure 8. Analyzing Program for the Smart Meter

Performance results by distance and direction are shown in Fig. 9. The figure on the left side had no differences in terms of distance and direction. But the images in the middle and on the right side had the worse success rates in rear areas. This was because the water meters were installed in blind spots owing to building structures and relatively poor propagation environments.

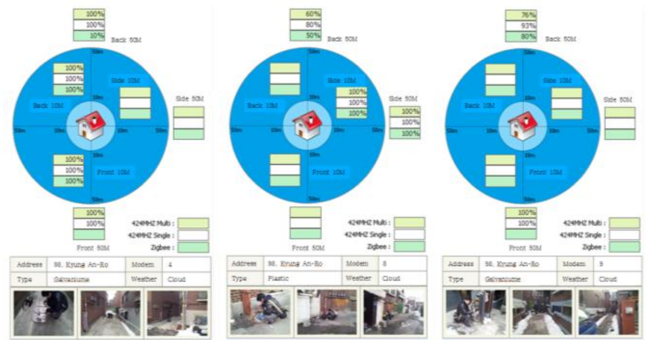


Figure 9. Result of Performace Test by distance and direction

A transmission distance comparison is shown in Fig 10, in which the success rate decreased as the distance became farther. But the success rate suddenly increased at 90m, which was caused by as small portion of data, 10 among 700 data. Most of the data at 90m were measured from the side of the water meter, whereas only one was collected from the rear and it is believed to have the worst success rate. Each of the success rates of communication modules are as follow: 424Mhz Multi module was 78%, 424Mhz Single module was 88%, and the Zigbee module was the lowest at 77%.

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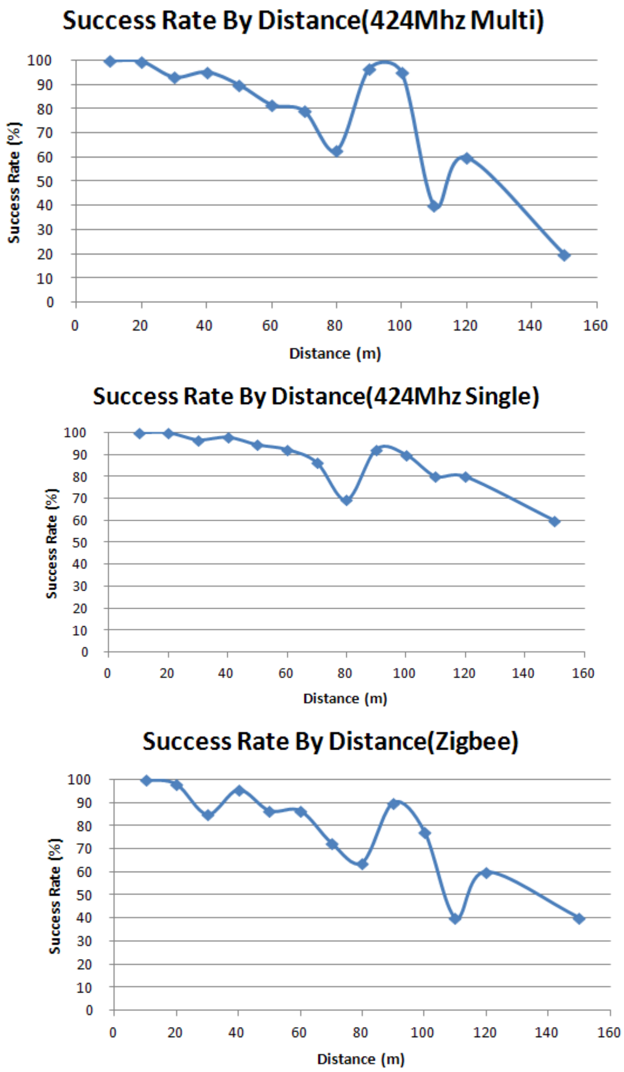

Figure 10. Success Rate by Distance

The transmission distance was almost similar in the 424 MHz Multi and 2.4 GHz Zigbee modules. However, it was analyzed that the 424 MHz Single module was 10 percent more efficient than the other modules in regards to the transmission distance. Besides, the casing for the water meter was analyzed depending on the type of casing, plastic or galvanic. The results show that the plastic casing had a 93% success rate and the galvanic casing had a success rate of 90%. Both of the casings have limited influence on data transmission.

III. CONCLUSION

As the application of smart technologies is expanding in urban infrastructure systems, the developed smart meters for distribution networks can be used to operate and manage water supply efficiently and economically. A newly developed half-electronic water meter combines the standard mechanical water meter with an electrical signal transmission. Thus, it can remove some of the weak points of existing electronic water meters such as the risk of data loss, which is believed to be a major impediment of smart metering.

Also, three kinds of wireless modules were tested in various communication environment conditions to determine which modules were best suited for specific terrain characteristics. Thus, it is expected that customized transmission services can be provided. Finally, this research will be applied to large demonstration sites in the near future.

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Support System for Caregivers with Optical Fiber Sensor and Cleaning Robot

Junko Ichikawa, Tetsuya Kon, Kazuhiro Watanabe and Norihiko Shinomiya
Dept. Information Systems Science
Soka University
Tokyo, Japan
shinomi@soka.ac.jp

***Abstract*—Population aging is one of the serious issues in developed nations. Because of the increase in the number of senior citizens, the demand for care of them has been enhanced. Consequently, some facilities such as group homes are provided in order to support their life. However, shortage of caregivers in group homes provides lack of attentions to situations or accidents of the elderly. Therefore, this research aims to propose a supporting system with sensor network technology and robots for caregivers in such facilities that can reduce workload of nurses and assure safety of facility residents.**

Keyword—monitoring system; sensor network; robot; nursing.

I. Introduction

Society is aging globally, and the number of the elderly dramatically is increasing especially in developed countries. Japan also encounters this serious problem. The rate of aging of Japan, which is defined as the percentage of the population of over 65 years old has monotonously been rising since 1950 and exceeded 24% in 2012 [1]. Furthermore, it is expected to exceed 30% in 2025. Following the increase in the number of elderly residents, some social problems have emerged such as insurance structure issues and health problems. Because many of them suffer from physical issues, the demand for health caret enhances.

In Japan, the number of facilities that support elderly people is over 25,000 sites [2], and *group homes* account for 47% of these facilities. A group home is a facility where caregivers stay overnight and nurse the elderly with senile dementia. In these facilities, one caregiver is usually required to take care of approximately 10 elderly people during night time. In this situation, caregivers are possibly not aware of accidents of the elderly. For example, falling from a bed and wandering aimlessly in a facility might not be found, and elderly people can be fatally injured.

In order to solve the problem, the video monitor systems in group homes have been developed [3]. Caregivers can monitor elderly people behaviours through cameras which are installed in living rooms, bathrooms and entrances in group homes 24 hours of day. However, such monitoring systems directly display private behaviours with images and sounds. Thus, it may make elderly people stressed because of invasive surveillance.

Using sensor network is conceived as a solution for prevention of the invasive surveillance. Sensor networks can indirectly retrieve environmental information and situations by setting many sensor nodes. An advantage of sensor networks in the fact that they detect the information without use of videos. Nonetheless, detection of abnormal behaviours with sensors can be a misjudged recognition. For instance, something falling on the floor can be regarded as a fall of an elderly person, and this incorrect reports confused caregivers. Thus, another technique is needed in order to confirm a reported situation by sensor networks. Robots that function autonomously are expected to collect more detailed information, which cannot be found only with wired sensor networks because they can move to the incident location and conduct interactive confirmation processes using sounds.

Hence, this research proposes a Support System for Caregivers (SSC) with optical fiber sensors and a cleaning robot, which monitors behaviours for elderly people and notices the caregiver about an abnormal behaviour. Section III summarizes some requirements for SSC that gained from our hearing researches to caregivers. This section also discusses necessary elements for the systems based on the requirements. Furthermore, section IV describes the design and utilized technologies in our prototype system. Finally, the progress of development and the results of preliminary experiments are demonstrated in sectionV.

II. Related Work

In order to assist the elderly to safely spend their daily life, some monitoring systems have been proposed. For example, a homecare monitoring system is proposed by Bourennane et. al. [4]. The research suggests that the utilization of multi-sensor networks realizes behaviour observance of an elderly person, who lives alone at home. Due to the observance, the proposed system provides the alert function in case of dangerous accidents. However, this research does not suppose the use of such a system in welfare facilities where many elderly people inhabit.

Another instance is a monitoring system in group homes [3]. The paper suggests a method to monitor facility residents with videos and install such equipments into the group homes. Nevertheless, the use of cameras is hesitated in terms of privacy problems. Moreover, such a camera system cannot be sited in bathrooms where accidents such as tumble frequently occur.

Both the proposed systems partly solves the problems in care of the elderly; however, a support system that considers the use in facilities where many incidents simultaneously happen and privacy issues is required. Thus, this paper proposed the support system, which is to find an abnormal behaviour of

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residents in welfare facilities and prevent invading privacy of the elderly.

III. Support System for Caregiver

This section summarizes some requirements for SSC that gained from our hearing researches to caregivers. This section also discusses necessary elements for the systems based on the requirements.

A. Assumed Situation

Our system presupposes the situation that elderly person suddenly fall down on the floor, and caregivers does not become aware of the accident. This system aims at the goal which are to detect a fall quickly and to confirm safety for elderly person without cameras by using sensors and robots. In addition, the system informs caregivers about the place where the elderly fells down and the information of safety confirmation. Thus, the system supports caregivers to notice accidents that are not aware so far. Moreover, it is expected to reduce the load of care because even though caregivers do not watch the elderly people carefully all of the day, they were able to know when the elderly person needs care.

B. System Overview

In this system, sensors embedded in a floor are used to detect fall. A situation of fall is acknowledged by that more than two sensors continue to simultaneously retrieve pressure over certain time interval. Furthermore, a robot with touch sensor is used to confirm safety of elderly person.

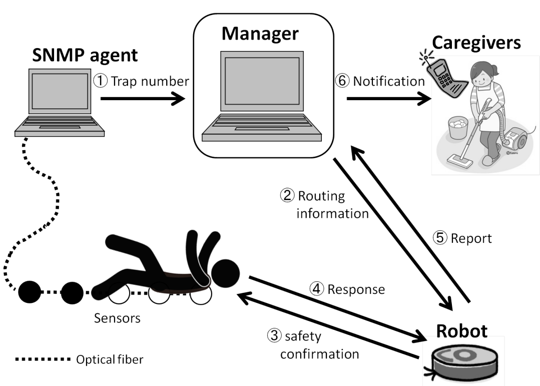

Fig. 1. System component diagram

The system overview of support system for caregivers is shown in Figure 1. An agent of Simple Network Management Protocol (SNMP) regularly observes the state of sensors and sends trap number to a manager. The manager retrieves the information and monitors a behaviour of an elderly person. When the manager recognizes the fall, it calculates a route to the incident location for a robot. The manager sends the route to a robot, and the robot moves toward the location following the route. After the robot arrives the location, it beeps in order to check whether or not the elderly person is unconscious. If the elderly does not push the touch sensor as the response to the beep, a robot reports on the state of the elderly to the manager. Then, the manager notifies caregivers about the abnormal situation.

IV. Prototype of SSC

This section presents details of system components, such as sensor device, network management, SSC manager and robot in our prototype system.

A. Sensor Device

The system uses sensor technologies to detect the accident. Generally, wireless sensor networks are set to observe the motion such as falling [5]. A wireless sensor gathers a lot of information to distribute the sensor nodes among a wide area. However, wireless sensor networks demand to supply an electric power to all sensor nodes.

To solve this issue, the *hetero-core spliced optical fiber sensors* have been developed. The fiber sensor does not need to be supplied power, and furthermore, the sensor can work as not only sensing device but also communications links [6].

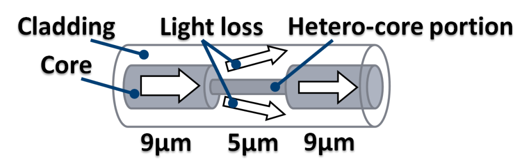

Fig. 2. Structure of hetero-core spliced optical fiber sensor

A structure of a hetero-core spliced optical fiber sensor is shown in Figure 2. It is composed of two single mode transmission fibers. One fiber has 9 μm core diameter, the other is 5 μm diameter. A few millimeters of 5 μm core diameter fiber is inserted into 9 μm core diameter fiber segmented into some parts. The part where the 5 μm core fiber is inserted is called *hetero-core portion*. When the hetero-core portion is bent from outside by pressures, light waves leak into the cladding region. The hetero-core spliced optical fiber sensor can work as sensor by measuring the light leakage. By utilization of the sensor, the smart pressure sensing mats to monitor human activities can be developed. The mats detect motions of walking and tumbling by surveillance of the change of an optical signal attenuation [7].

B. Sensor Network Management

Optical Sensory Nerve Network (OSN) with hetero-core spliced optical fiber sensors is a network that realizes communication and sensing simultaneously [8]. The OSN uses SNMP to manage sensor network equipments and distinguishes the sensor states by trap [9].

Figure 3 shows the structure of the binary switch sensor. The switch sensors are one of the hetero-core spliced optical fiber sensor module, and one hetero-core portion is inserted in the sensor. When the button is pressed, the curvature of hetero-core portion changes, and the sensor is given the optical loss. Figure 4 shows the component of sensor network management system. First, the media converter (A) converts from voltage

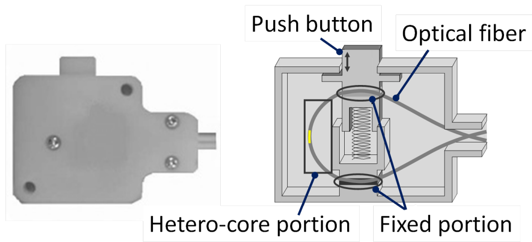

Fig. 3. Binary switch sensor module

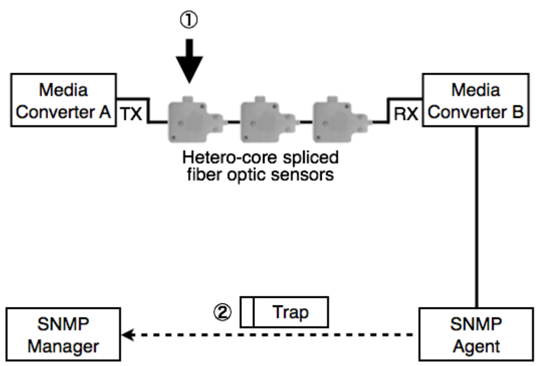

Fig. 4. Sensor states detection by SNMP

to light signal, and it sends the signal. Next, the signal went through the sensors to the media converter (B), and it is converted to voltage. When the binary sensors are pressed, the optical loss is occurred. An SNMP agent measures the optical loss and issues trap to an SNMP manager. The trap previously was set to a threshold voltage per each sensor. Thus, the manager can detect the sensor pressed location to check the trap.

C. SSC Manager

An SSC manager fulfills two types of roles: detecting the state of sensors and routing for robot. As a sensor manager, it handles sensor information; additionally, it calculates a route for guiding a robot with its routing function.

1) Detecting the State of Sensors: The manager regularly analyzes the sensor information provided by an SNMP agent that observes state of each sensor and detects an abnormal situation. The abnormal situation is defined as the state which is more than two sensors continue to simultaneously retrieve pressure over the time interval, which is set 5 seconds. If the manager finds the situation, it sends routing information to robot.

2) Routing for Robot: The manager also has a routing function to give a route to a robot. In the routing process, the manager adopts a shortest path algorithm that has been studied in the field of graph theory. A shortest path algorithm is a solution to obtain a path with the minimum weights from a given source vertex to a destination vertex in a graph consisting of vertices and edges.

First, the manager makes a map which represents the layout of a facility and obstacles, such as furniture, walls and doors. On the map, it marks possible turning points for a robot and two specific location points. One of the location points describes the position where an elderly person falls down, and the other is the present location of the robot. The manager regards these points as vertices and calculates the shortest path from the current location to the incident location which passes though the turning points based on the Dijkstra's algorithm [10].

After a robot receives the routing information from the manager, it moves along the given shortest path. However, the robot does not move precisely on the given path because the tires of the robot may slip on floor. Thus, the route given to a robot needs to be revised.

The revision of the route is conducted by certain time intervals. When the gap between the present location detected by sensors and the given path exceeds a certain distance, the manager provides the robot a route to return from the current location to the closest turning point on the given path. When the robot returns on the given track, it resumes moving along the path.

D. Robot

The system uses a robot to confirm safety for the elderly in order to reduce misinformation of the sensor, and the following is the requirements for a robot that is suitable for our system.

- Appearance of the robot does not offend a user.
- Any cameras are not attached on the robot.
- The robot can autonomously move.
- The movement can be controlled remotely.
- In order to confirm safety, a robot has beep function and a bumper.

As a robot which fulfills these requirements, a vacuum cleaning robot, such as Roomba [11] is likely to be selected. Recently, Roomba has penetrated in general households; thus, it is easily configured in our system. Roomba designed simply that does not threaten the elderly. Additionally, it can beep sounds as its alert, and a bumper is installed on ahead of the body, which is used as a button in our system. Besides, though it autonomously moves around rooms in its cleaning mode, its movements are able to be controlled by programs from remote locations.

The following describes the behaviour of confirming safety by Roomba. Roomba that is provided the routing information by the manager shifts to the location where a elderly person falling down. After Roomba arrives the location, in order to check whether the elderly has consciousness, it beeps a sound during certain time. The manager observes the reaction from the elderly person to the beeps. When the bumper on the Roomba body is pushed within the certain time, the manager judges that the person is conscious. In contrast, when the bumper is not pushed, the person is judged as unconscious by the manager.

V. Experiment Results

This section explains results of SSC trial experiment, examining the sensor management and robot control function. SSC

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is developed by Java Universal Network/Graph Framework (JUNG). This framework is an open source library for Java in order to analyze and visualize the structure of graphs. The library contains many algorithms in graph theory such as a shortest path algorithm.

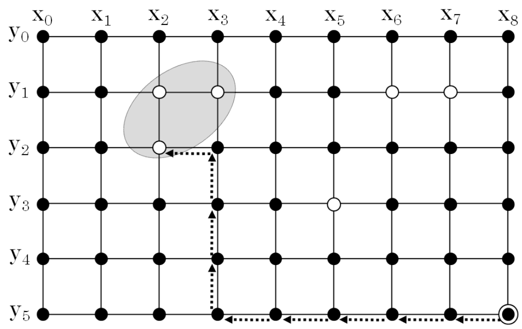

Fig. 5. Logical map and robot path calculated by manager

TABLE I
SEPARATED COMMANDS FOR A ROBOT IN A MANAGER

Order	Command
1	Go Straight 30cm
2	Go Straight 30cm
3	Go Straight 30cm
4	Go Straight 30cm
5	Go Straight 30cm
6	90 right spin
7	Go Straight 30cm
8	Go Straight 30cm
9	90 left spin
10	Go Straight 30cm

In this experiment, 54 sensors are embedded in a floor at intervals of 30cm. At first, manager recognizes connection relationship of the sensors and model it with a graph as shown in Figure 5. When arbitrary sensors are pressed, the manager detects the change of the sensor states. The pressed sensors are shown as white nodes in Figure 5. When more than two sensors are adjacent each other, the manager judges the place as the accident location where an elderly person falls down. In Figure 5, adjacent sensors (x_2, y_1), (x_5, y_3) and (x_6, y_1) are pressed, so the manager set the place of the sensors as an incident location. Then, the manager recognizes the present place of robot (x_8, y_5). Furthermore, the manager calculates the shortest path between the place of the robot and one of the sensors. The path calculated by the manager is indicated as arrows of broken line. Finally, the manager generates routing information as a set of commands to send the robot. Table I describes the set of commands, which corresponds to the shortest path in Figure 5.

VI. CONCLUSION AND FUTURE WORK

In summary, this paper proposes a support system for caregiver, which uses a optical fiber sensor and cleaning robot. This system aims to monitor the elderly without the use of video recording in order to prevent privacy invasion. By using the hetero-core optical fiber sensor network and a vacuum cleaning robot, the network realizes detection of tumble of a elderly person and confirmation of the detection. First, some requirements to realize the assumed system are summarized, and the system overview and components are also mentioned. Furthermore, the concrete technologies such as hetero-core fiber sensors and Roomba that suits for our system requirements are introduced, and the prototype system consisting of the equipments is described. In addition, two types of roles of an SSC manager, which are the sensor management system and robot control function are shown, and an experiment results of these functions are presented. However, the experiment is not considered an initial direction of the robot. Furthermore, to decrease the gap between an distance calculated and a distance moved robot also remains an issue as future works.

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Remote Monitoring System with Optical Fiber Sensors and the Internet-Standard Protocol

Koki Fuchigami, Lee See Goh, and Norihiko Shinomiya
Dept. of Information Systems Science
Soka University
Tokyo, Japan
shinomi@ieee.org

***Abstract*—Services exploiting environmental information collected by sensor networks are in great vogue. Even though many sensor networks adopt wireless sensor devices, wireless sensor networks hold the battery life problem of sensor devices. Therefore, Optical Sensory Nerve Network (OSN) using the hetero-core optical fibers can simultaneously conduct sensing and communications without batteries. In this paper, the data collection method with the Internet-standard protocol in OSN is proposed. Additionally, this paper describes a remote monitoring system and development results of the remote monitoring system in OSN.**

***Keywords*—sensor network; optical fiber sensor; simple network management protocol.**

I. INTRODUCTION

In recent years, information of human behaviors collected by sensor networks has been widely used for various services. The networks with capacity of sensing are utilized for security purpose in houses and environmental monitoring in forests and rivers [1][2].

Although wireless sensor devices are commonly used for monitoring systems, they need to implement batteries in themselves or generate electricity by themselves with solar power because each sensor is mutually independent. However, the power generation is unstable and limited because the amount of generation is influenced by climate conditions. As a result, the wireless sensors are forced to save power by limiting functions and communications among sensors. In order to solve this problem, *Optical Sensory Nerve Network* (OSN) using the hetero-core spliced fiber optic sensors (hetero-core sensor) has been developed, and this wired sensor technology simultaneously realizes sensing and communications with a fiber [3].

This paper proposes a method to collect data in OSN and acknowledge conditions of hetero-core sensors with Simple Network Management Protocol (SNMP) [4]. Furthermore, we also propose a remote sensor identification system from outside of OSN using a database and a web application. The objective of this study is to remotely monitor the multiple sensor conditions in OSN from outside of OSN.

In Section II, hetero-core sensors and the characteristics of OSN are explained based on previous researches. Moreover, this section proposes the identification method of multiple sensor conditions. Finally, remaining issues of the sensor identification method related to SNMP are described, and the remote monitoring system for OSN that reduces complexity of access from distant locations caused by SNMP is proposed in Section III. In addition, an example behavior of the developed system is described.

II. OPTICAL FIBER SENSOR NETWORK

This section explains the previous work related to OSN. Furthermore, the multiple sensors identification method with SNMP is proposed.

A. Hetero-core Spliced Fiber Optic Sensor

A hetero-core spliced fiber optic sensor is utilized in OSN. This sensor plays two roles of sensors and communications links in OSN [5][6]. Figure 1 shows the structure of a hetero-core sensor. The sensors can be simply fabricated by fusion splicing. The sensors in our system are made of two kinds of single mode fibers whose diameters are 9 μm and 5 μm. The portion consisting of a thinner fiber is called *hetero-core part*. If the hetero-core part bends by pressure, a part of light is lost as leakage in the hetero-core part. Therefore, pressure on hetero-core part can be detected by measuring amount of the light leakage. Besides, this attenuation of light from hetero-core parts does not negatively affect data communications [7].

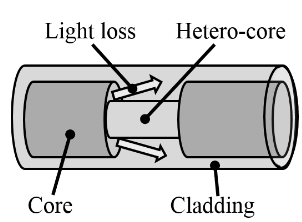

Figure 1. Structure of hetero-core sensor.

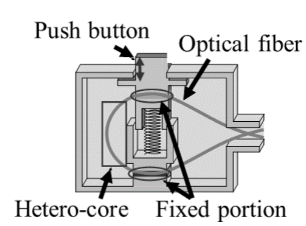

Figure 2. Binary switch sensor.

In this paper, a binary switch sensor using a hetero-core sensor is utilized, and Figure 2 shows the structure of a binary switch sensor. When the button at the upper part is pushed, the hetero-core part bends. As a result, the amount of light leaking to outside increases, and ON/OFF status of the sensor is identified by measuring the attenuation amount. Note that, the sensor condition is defined as *ON status* when the button is pushed.

B. Optical Sensory Nerve Network

The OSN is a sensor network that is developed by adding sensing function into Ethernet [3]. Hence, optical fibers in

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OSN are used for not only data communications but also sensing. Because a hetero-core sensor detects environmental information by using transmitted light in the optical fiber, provision of electric power is not needed for sensing. Thus, OSN solves the problem related to limitation of sensor functions in wireless sensor networks.

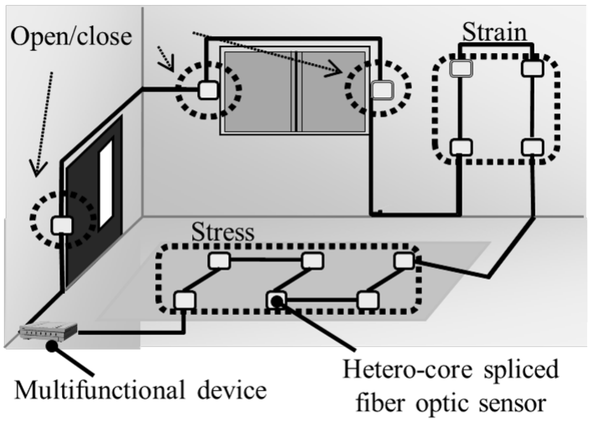

Figure 3. Example of a monitoring space using OSN.

Figure 3 indicates an example of a monitoring space using OSN. Sensors are embedded in walls, floors, doors, and windows in order to gather state information, such as strain of walls, pressure on floor, and open-close state of doors and windows.

C. Sensor Identification Method with SNMP

In order to identify sensor conditions in a sensor network, SNMP is utilized [4]. SNMP is a protocol that is implemented in general network devices to monitor and control network devices. A network with SNMP consists of *an SNMP manager* and *some SNMP agents*. An SNMP agent monitors a network device and it sends device information to the SNMP manager. The SNMP manager controls some SNMP agents, and it receives device information from the SNMP agent. If an exceptional event, such as a communication failure occurs, the SNMP agent sends notification messages called *Trap* to the SNMP manager. The sensor identification method utilizes this notification function to identify change of sensor states.

Figure 4 describes design of our system that identifies sensor conditions. This system contains three hetero-core sensors named A, B, and C. When the condition of any sensor becomes ON, the transmitted light in a fiber attenuates. The attenuation is always monitored by a measuring instrument which contains the SNMP agent function. In this measuring instrument, a script program that is defined to issue the Traps can be set.

Because attenuation of each sensor differs from each other, the total amount of attenuation also diverge depending on combinations of reacted sensors. There are seven patterns of attenuation amount in total because this system has three sensors. Attenuation amount is different in each combination, so the program in the measuring instrument can link each attenuation with different Trap numbers. When such an issued Trap is received by the manager, the manager can identify every sensor condition.

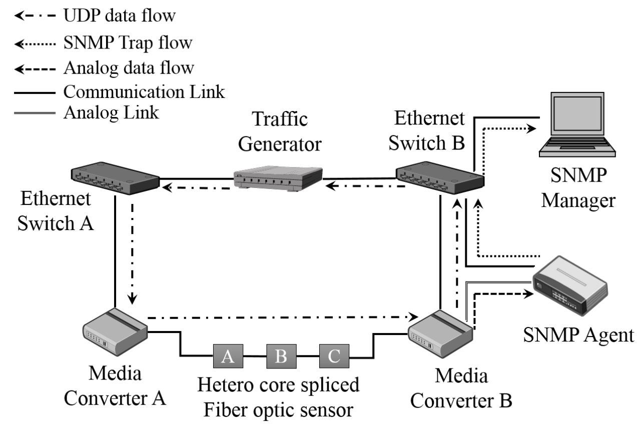

Figure 4. System configuration diagram.

Next, the identification results of this method are indicated and evaluated. TABLE I shows identification results of sensor conditions. The SNMP manager distinguishes sensor conditions by analysing the Trap number issued by the SNMP agent. Figure 5 indicates attenuation patterns of seven sensor combinations in decibel notation. The figure represents the different amount of attenuation among seven sensor combinations, which means that the sensor conditions are able to be identified.

TABLE I
EXPERIMENTAL RESULTS OF THIS METHOD.

Time	Trap number	Description
10:00:00	21	Switch A ON
10:00:10	28	OFF
10:10:00	23	Switch A&B ON
⋮	⋮	⋮

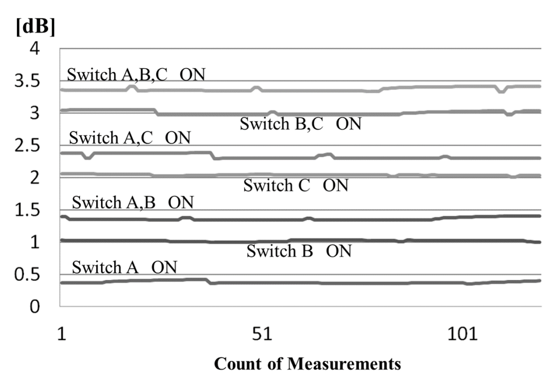

Figure 5. Attenuation of 7 patterns in dB notation.

III. REMOTE MONITORING SYSTEM

This section describes remaining issues of the sensor identification method. Furthermore, this section proposes a remote

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monitoring system for OSN as solution of the issues.

A. Remaining Issues and System Requirements

As mentioned in Section II-C, it is possible to detect sensor conditions with SNMP. However, there are two remaining issues to be solved when the size of the system expands, or this system is utilized in outdoor [8].

The first remaining issue is the remote monitoring function. In the identification method with SNMP, sensor conditions can be monitored only within the OSN. However, it is assumed that observers connect to the OSN from outside and hope to remotely monitor the sensor conditions. Therefore, a method to confirm sensor information from remote locations is required.

In order to monitor the sensor network, the current identification method with SNMP requires Virtual Private Network (VPN) connection and an SNMP manager software at a computer of observers. Therefore, all network equipments need to be adjusted for VPN and SNMP in order to get sensor information. When a sensor network becomes larger in scale, the installation of the settings into all equipments may impair usability of OSN. Thus, a method to monitor sensor conditions from remote locations without complex settings is needed.

The second remaining issue is management method of sensor information. In the current identification method, the sensor information is stored in reaction time order as shown in TABLE I. This stored data format is a text file, so it does not have a search function, and the manipulation of data is difficult. Therefore, the identification method does not suit for long term monitoring. Hence, the data processing and search function are required.

In order to satisfy the listed requirements in this section, we have developed and designed the remote sensor identification system, which can provide sensor information on a web page using a web application and a database.

B. System Overview

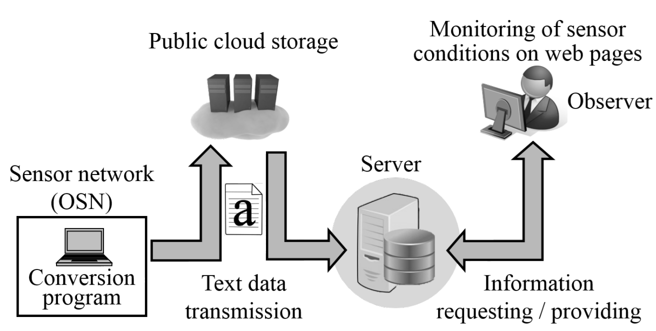

Figure 6. Remote monitoring system.

Figure 6 indicates the developed system. As a solution to the remote monitoring issue of Section III-A, text data containing the Trap numbers and sensors reaction time are sent to a server for sensor condition identification without using the SNMP manager software at a computer of observers. In order to remove the SNMP manager software from a computer of observes, a software that is called *a conversion program* receives an issued Trap from an agent in OSN. The conversion program creates text data contains a Trap number and sensor reaction time, and it sends text data to a server. Therefore, a manager software, which receives the SNMP messages is not needed. Furthermore, observers do not need to use VPN connections because sensor information is provided to observers on a web page. In other words, if observers can connect to the Internet, they can monitor sensor information from anywhere.

Next, as a solution to the sensor information management problem discussed in Section III-A, the database (DB) and the web application are utilized in the developed system. The use of the DB enables this system to manage sensor information, and the search and process of data also become easier. Moreover, the web application enables observers to deal with the data easily and to monitor in a long term even without any knowledge about DB.

C. System Components

This section explains software components of our remote monitoring system.

1) Conversion program: This program receives an SNMP message and converts an SNMP message to text data in a sensor network. A trap number and sensor reaction time are written in the converted text data for sensor identification. This program is executed at each time when it receives a Trap.

2) Database: The database in our system satisfies the following conditions.

- Sensors may be installed in some different locations.
- Trap number and sensor reaction time need to be saved.

In order to store the sensor information to the DB, *the DB storage program* is implemented in the server. This program has following functions.

- The program checks additional sensor information from the previous program execution. If there are any added text data, the program generate a trigger file.
- The program stores sensor information to the DB. If the storing process is completed, the program delete the trigger file.

The trigger file plays a role to inform the new added text data. The program will be automatically executed on the server at any scheduled time. The minimum execution interval is one minute.

3) Web application: In this system, the web application, which works together with the DB provides the sensor information to observers on a web page. Therefore, observers monitor the sensor information without any knowledge about DB. Furthermore, if the observers can connect to the Internet, they would monitor the sensor information from anywhere.

D. System Operation

In this section, the flow of the system operation is described.

First, attenuation of transmitted light in a optical fiber increases when the sensor becomes ON status. This attenuation is detected by an SNMP agent, and the agent issues a Trap that

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is unique number for each attenuation. After the conversion program receives an issued Trap, it converts a Trap to text data that indicate a Trap number and sensor reaction time. This created text data are sent to the server and stored. At this time, sensor information has not been stored in the DB yet.

Next, the process of the system is shifted the server. In the server, the DB storage program is executed every minute. The program checks the additional sensor data from the previous execution. If there are any additional text data, the program creates the trigger file in the server. When the storing process is successfully finished, the program delete the trigger file. If the storing process is failed, the trigger file is left. Therefore, the data, which is not stored in the DB can be easily checked.

Finally, the web application and the DB work together and provide the sensor information to observers on a web page.

E. Operating Results of the System

In this section, the operating results of the system are indicated and evaluated. The sensor system used for this operation is composed as same as the system architecture shown in Figure 4. It was verified that all sensor conditions are correctly monitored on a web page. Figure 7 indicates a display example of the web page.

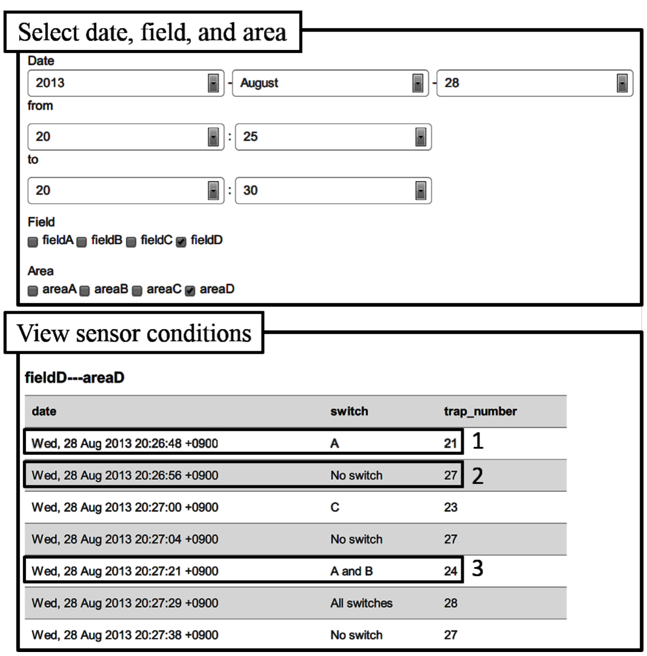

Figure 7. Display example of the web page.

This system enables the observers to search information more easily than checking all data with a list like TABLE I. Furthermore, the observers can monitor the OSN from outside of it. Results of this system operation indicates that state of each sensor can be represented through our sites. For example, the line **1** in the Figure 7 indicates the detection of switch A reaction. The line **2** indicates the completion of switch A reaction. Furthermore, the line **3** indicates the reaction of the switch A and B; therefore, it is also possible to display multiple sensor conditions by this system.

IV. CONCLUSION AND FUTURE WORK

In this paper, the SNMP is used to identify conditions of hetero-core sensors in the proposed OSN. Moreover, we proposed and designed remote monitoring system to solve the identified remaining issues. In order to monitor the sensor conditions from remote locations, we developed a system that utilized the web application and the DB. The sensor conditions can be remotely monitored on a web page by this developed system.

In the future, we will consider the utilization of OSN for outdoor and the construction of larger scale of sensor networks. For example, in the field of nursing, sensor network informs behavioral patterns of elderly people to caregivers. Moreover, in the field of agriculture, farmers monitor their scattered farmlands from remote locations. Therefore, this system would play a role to provide some practical usages for OSN. This system develops the capability of OSN because it has a remote monitoring function and a database function of sensor information. Moreover, we should increase the number of sensors in the sensor network because only three sensors are identified by the current system. This restriction is because the script capacity of the agent is limited. Thus, a development of an agent, which has no limit of the capacity is needed.

ACKNOWLEDGEMENT

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Still Picture Internet Broadcasting System with Audience-oriented Bandwidth Control for Smartphone Broadcasters

Yuki Nakano, Yoshia Saito and Yuko Murayama
Graduate School of Software and Information Science
Iwate Prefectural University
Iwate, Japan
Email:{y.nakano@comm.soft.iwate-pu.ac.jp, y-saito@iwate-pu.ac.jp, murayama@iwate-pu.ac.jp}

Abstract— **We propose PictCast, which is a still picture Internet broadcasting system with dynamic picture quality adjustment based on audience requests for smartphone broadcasters. Live Internet broadcasting services, which are available on smartphones are increasing. However, the smartphone users need to decrease data traffic because they perform Internet broadcasting via 3G networks. To tackle this issue, we propose an effective bandwidth usage by reducing amount of data used to the minimum corresponds to audience satisfaction. We study the minimum quality of still pictures and temporarily determine the percentage of quality improvement requests from audience. Then, we conduct a field study during a festival at our university and find out whether there are pictures which the audience requires more high quality or not, the possibility to reduce amount of data traffic and what type of content the audience is interested in. Finally, we discuss our findings based on the result, and end up with the conclusion and future work.**

Keywords: Internet broadcasting; Narrowband network; smartphones

I. INTRODUCTION

Live internet broadcasting services, which are available on smartphones are increasing in demand, along with the popularization of Internet broadcasting and smartphones. Example of these services such as Bambuser, Qik, Stickam, and Ustream. 3G Internet connections are required for Internet broadcasting using smartphones in outdoor areas where wireless LAN connections are unavailable. However, there are also bandwidth limitations in rural areas and developing cities. Thus, we should consider the smartphone users' broadcast in these environments because the audience would like to look at a place where they have not been before. In some cases, smartphone users cannot perform Internet broadcasting adequately due to lack of 3G transmission speeds for video streaming in real time. Engström [1] states that effective use of bandwidth is needed because the study shows delay problems with live broadcasting systems on mobile networks. Therefore, data traffic should be reduced to achieve stable Internet broadcasting via 3G networks without delay. Smartphone users frequently use LTE to stream live video in real time. However, cell phone carriers often restrict the upload and download speeds of LTE after a certain amount of data has been transmitted. Data traffic should be reduced as much as possible even if users use LTE. Moreover, cell phone carriers nowadays tend to shift flat-rate data services to pay-as-you-go data services. Thus, 3G data traffic should be optimized to save packet communication fees.

Our research goal is to minimize the amount of data traffic by effective use of bandwidth which satisfy the audience.

S.McCanne [2] adjusts video quality depending on available network bandwidth as a method to achieve effective use of bandwidth. Similarly, recent research in mobile broadcasting have studied how to maximize bandwidth utilization [3]. However, these schemes may use redundant network bandwidth in excess of audience satisfaction. The quality should be determined from audience satisfaction point of view, instead of available network bandwidth. In our work, we introduce new aspect of dynamic quality adjustment picture with audience participation during the live broadcast. The amount of data traffic is optimized by dynamic quality adjustment which could change still picture quality based on audience requests. In the proposed system, we adjust the picture, not video quality to reduce data traffic as much as possible and take a closer look at the user interaction. For the use of pictures to reduce data traffic, Okada [4] realizes conferencing system use still pictures because it can work in a narrowband network. The conferencing system detects users' eyes direction and assists users by changing still pictures of users' faces without using videos. The use of still pictures has proved to reduce data traffic significantly. Moreover, the still picture view can be adjusted in case audience request for high quality picture. For future work, we will adjust video quality instead of picture quality.

In this paper, we present PictCast, a prototype system of still picture Internet broadcasting with dynamic picture quality adjustment. The innovation of our research is audience-oriented bandwidth control based on audience requests. The PictCast achieves stable Internet broadcasting via 3G networks using still pictures instead of video and optimizes the amount of data traffic by allowing the change of picture quality based on audience requests. For dynamic picture quality adjustment function, first it sends the minimum quality picture which satisfy the audience. Secondly, it will retransmit picture data to the audience from the broadcaster after it receives a certain number of quality improvement requests. For the preliminary study, we studied and predefined the minimum quality and percentage of quality improvement requests from the audience. We carried

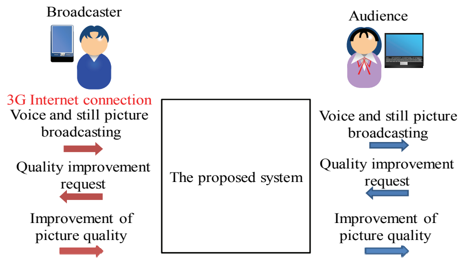

Figure 1. System model.

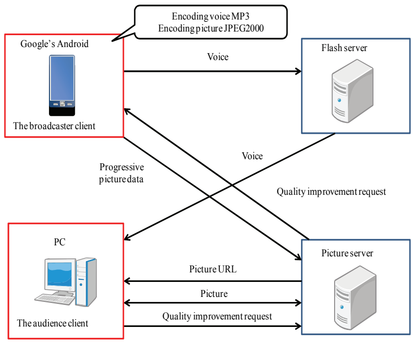

Figure 2. System Architecture.

out a field study during a festival at our University. As a result of the study, we discussed our findings and continue with the PictCast project.

The paper is organized as follows. In the next section, we show a model of our proposed system and present design and implementation of the prototype system. Section 3 describes a preliminary study. Section 4 presents the results of a field study. Section 5 gives some conclusions and our future work.

I. PICTCAST

Figure 1 shows the model of our proposed system. A broadcaster sends still pictures and voice to the audience via 3G wireless network. The audience watches a broadcast program from their PC via high-speed network. The voice is compressed and broadcast to the audience in real time. The still pictures are compressed in a progressive format. To reduce data traffic, the broadcaster sends the still pictures to the audience at the minimum picture quality which is acceptable to the audience at first. If the audience requires higher quality of still pictures, they can send quality improvement requests to the broadcaster. The broadcaster then retransmits higher quality still pictures correspond to a certain number of audience requests. In this case, the first low-quality still picture will be replace by retransmitting with a higher quality one. Therefore, we use progressive decoding, which is represent by progressive JPEG. The progressive decoding makes a picture gradually sharper as the download progresses. To apply the mechanism, the broadcaster can improve the quality of a still picture progressively without wasting transmitted picture data.

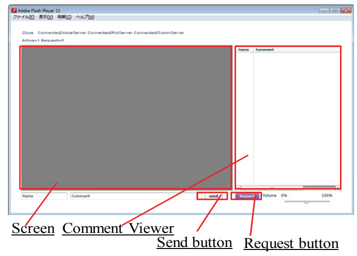

Figure 3. The user interface of the audience client.

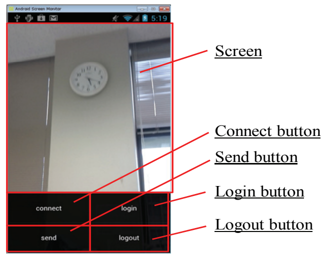

Figure 4. The user interface of the broadcaster client.

A. *System Architecture*

We implement a prototype system which provides still picture broadcasting with progressive decoding. The prototype system consists of ~~a~~ voice server, picture server, broadcaster clients and audience clients. The broadcaster client sends voice and still pictures to voice and picture server respectively. The audience will receives voice and still pictures from each server. Figure 2 shows architecture of the prototype system.

We developed the broadcaster client on a Google android terminal. The broadcaster client compresses voice in MP3 format by using the LAME library. We used Red5, which is an open source streaming server for the voice server. The broadcaster client sends the compressed voice data to the voice server over Real Time Messaging Protocol (RTMP). Then, audience clients receive the voice data from the voice server. The audience client is implemented in Adobe Flash. JPEG2000 format is used for the progressive decoding. The JPEG2000 format provides various progressive decoding functions which can change resolution and compression rate of the picture. In this implementation, the progressive decoding by resolution is used. First, the broadcaster client sends a low-resolution still picture to the server. Then, it sends additional data to the server responding to audience requests, so that the resolution of the still picture can be doubled. To compress the still picture in JPEG2000 format, we use a library OpenCV, which can be used on Google android terminal.

The picture server functions as a Web server using Apache. When the picture server receives a picture with JPEG2000 format from the broadcaster client, it changes from JPEG2000 to JPEG by using the OpenJPEG library. The picture server provides a URL for the compressed JPEG picture to the audience clients. The audience client shows a still picture, downloading it from the picture server and it

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also has a button to send a quality improvement request to the picture server. When the picture server receives a certain number of quality improvement requests, it requests additional data of the picture from the broadcaster client.

B. User Interface

We design the user interface for both broadcaster and audience client. Figure 3 shows the broadcaster client user interface. There are two broadcasting function in the design of the interface. The first function allows the broadcaster to take a picture with a touch on the smartphone screen. Meanwhile, the second function allows the broadcaster to interact with voice and picture server. The broadcaster client consist of four buttons: “CONNECT” establish connections with voice and picture server, “LOGIN” button authenticates the broadcaster with picture server based on specified username and password to start the broadcasting session. “SEND” button transfer the still picture to picture server and “LOGOUT” disconnect user with both voice and picture server. These buttons will not appear when the broadcaster snaps a picture and it will be displayed again by pressing MENU button on Android. Figure 4 shows the audience client user interface. There is a screen with 640x480 resolution for still picture on the layout of the interface. “SEND” button will send comments from client to both audience and broadcaster. The audience can read the comments on the comment view. “REQUEST” button sends a request for picture quality improvement to the picture server. Therefore, the still picture quality will be improved step-by-step based on audience request.

II. Preliminary Study

We conduct a preliminary study to predefine the still picture minimum quality and quality improvement request percentage from audience. The following section describes the method used and its result.

A. Method

There are many subjective assessment methods suggested by International Telecommunication Union Telecommunication Standardization Sector (ITU-T) and International Telecommunication Union Radio communications Sector (ITU-R). We choose Absolute Category Rating methodology (ACR) with 5-point scale on total assessment time and ease of evaluation criteria. The ACR methodology defined by ITU-T Rec. P910. In ACR, subject need to evaluate the picture quality within 10 seconds after they see the picture displayed randomly. Randomly ordered picture can eliminate the possibility of affecting the assessment by indicating the picture sequence. For example, a picture with medium quality will have the high rating if the subject watches it after the low quality picture. The picture quality will be given an average score evaluated by Mean Opinion Score (MOS).

Besides, for the percentage of quality improvement request, subjects were asked on the subjective picture quality assessment by the question: “Would you request for a higher quality picture?” for all types of picture resolutions.

TABLE I. THE RESULT OF THE ASSESSMENT

	MOS	The percentages of quality improvement requests
640x480	4.41	30%
320x240	3.20	64%
160x120	2.18	95%
80x60	1.56	100%
40x30	1.24	100%

B. Environment

We conduct an assessment following an environment described by ITU-R Rec. BT.500. The pictures were displayed on a 17-inch LCD monitor (LCD-A173KW) with the resolution 1024x768. Subjects performed the assessment at a distance of 120cm away from the display screen.

The pictures are in JPEG format with 5 resolutions, from 40x30 to 640x480. The size of the picture display field is 640x480 pixels because it is a most frequently size for Internet broadcasting sites.

We choose ten kinds of picture for the picture quality assessment which consist of person, group, painting, cat, paper, meal, building, tree and standard image pictures. "Standard Image" picture is one of the pictures standardized by The Institute of Image Electronics Engineers of Japan (IEEJ). The picture was rate by International Standards Organization (ISO). Those picture were selected based on typical categories of picture in live video broadcasting for mobile device over the Internet [5]. The number of subjects should be more than fifteen according to ITU-R Rec. BT.500. There are 23 university students in IPU participated in the assessment.

C. Result

Table 1 shows the result of the assessment. For the minimum quality of still picture, we have decided to choose 160x120 resolution picture because of the MOS score is around 2.5, while for 320x240 resolution, the MOS score is more than 2.5 for all assessment. The MOS score of 2.5 is the acceptable value because it represent over 50% of subjects who give 3 points. Therefore, for the prototype system, three steps offered for the progressive decoding from 160x120 to 640x480 picture resolution.

Meanwhile, the percentage of quality improvement request for 160x120 and 320x240 resolution are above 50% from the audience. From the result, we temporarily determined a threshold for the picture quality improvement of 50% of the total unique users.

III. Field Study

We carried out a field study to answer the following questions: (1)Does the audience send requests that they would like higher quality pictures?; (2)Is the proposed system possible to reduce the amount of data traffic?; (3)What type of content is the audience interested in?

We set three experimental conditions based on the preliminary study as follows: 1) the first low-quality still picture is 160x120 resolution, 2) the progressive decoding is

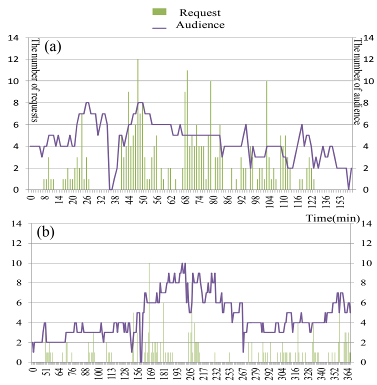

Figure 5. The changes in the number of the requests (a) On the first half of the broadcast and (b) On the second half of the broadcast.

offered in three steps from 160x120 to 640x480, and 3) the broadcaster retransmits higher quality still pictures responding to the requests of more than 50% of the total unique users.

A. Environment

The festival was held at Iwate Prefectural University campus. We broadcast the festival scenes at various places around campus by using a smartphone, which is available to communicate up to 300kbps over the Internet. Anyone can view the broadcast content on the website embedded in the audience client.

B. Broadcast contents

The field study was conducted on October 28th from 10:00 to 13:00 and from 13:00 to19:00. During the first half of the broadcast, we interviewed people who served food at the street stands. During the second half of the broadcast, we introduced some of the festival events such as concerts, onstage entertainment, and fireworks.

C. Results

We counted the total quality improvement requests in order to confirm the audience really sent the requests if they were not satisfied with the minimum picture quality. The total number of the requests was 283 on the first half of the broadcast and 175 for the second half of the broadcast. As a result, we found that the audience was not satisfied with the minimum picture quality and there were times when the audience wanted to see the pictures with higher quality.

Moreover, we analyzed changes in the number of the requests. Figure 5 shows the changes per 60 seconds. From this figure, we found there were pictures which the audience wanted to see in higher quality and the audience that was not satisfied with 160x120 resolution sent the requests. We measured the amount of data traffic for the system on that day. We compared the proposed system with a system which sends still pictures only at 640x480 resolution to the audience in terms of the amount of data traffic. The amount of the data traffic for the proposed system was about 15 Mbytes. The alternative system however required about 120 Mbytes. We found the proposed system was able to reduce data traffic significantly compared to the alternative system.

We analyzed when the audience sent requests. As a result, the audience sent many requests when we reported about the weather. For example, a picture of a puddle under the main stage only became high quality from the onstage entertainment broadcast. It is possible for weather report to be one of the features for this system.

D. Issues

A threshold value of total unique requests for picture quality improvement was set to 50 %. However, one of the audience commented, "I'm so stressed because my requests are often ignored". It is possible there were inactive audience members who just listened to the voice streaming for back-ground music or did not feel like aggressively sending the requests. Considering these inactive audience members, it would not be an adequate threshold to satisfy the audience. We should study the threshold to improve average audience satisfaction.

IV. CONCLUSION AND FUTURE WORK

We present PictCast, which is a prototype system with still picture Internet broadcasting and dynamic picture quality adjustment functions. As a result from the field study, we found that there were times where the audience request for higher quality pictures. The proposed system was able to reduce data traffic and the audience interested in weather report. We will design a future system which can be used in disaster area because the audience would like to see the current conditions, such as Weather report and surrounding in that area which have a narrowband network area.

We also found that there was a problem with the threshold value of total unique request for picture quality improvement. We will review the threshold value to improve the average of audience satisfaction in disaster area environment. For future work, we will release the broadcaster client on Google Play Store in order to get feedback on how users use PictCast in disaster area.

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Towards Organizational Modules and Patterns based on Normalized Systems Theory

Peter De Bruyn, Herwig Mannaert and Jan Verelst
Normalized Systems Institute (NSI)
Department of Management Information Systems, University of Antwerp
{peter.debruyn,herwig.mannaert,jan.verelst}@uantwerp.be

***Abstract*—Normalized Systems Theory provides prescriptive guidance on how to design modular software structures so that they exhibit a high degree of evolvability and low degree of diagnostic complexity. The theory basically suggests to first break up the modular structure in a very fine-grained way based on “concerns” (in terms of change drivers or information units) and then aggregate these concerns in a structured way into patterns or “Elements”. While the relevance of this theory for analyzing and designing organizational artefacts has been shown previously, only the first step of the reasoning has already been performed in the past, i.e., identifying a set of organizational concerns to be separated. In this paper, a first attempt to proceed to the second step (i.e., aggregating concerns into organizational “Elements”) is proposed. We formulate some meta-requirements for such organizational Elements (i.e., having exhaustive interfaces, aggregating several basic constructs into one “Element” as well as including and identifying a relevant set of cross-cutting concerns). We also propose a tentative set of five organizational Elements: Party, Product or Service, Compensation, Work Unit and Asset or Resource. The relevance of these Elements is shown by briefly discussing some (theoretical) illustrations.**

***Keywords*-Normalized Systems; Organizational patterns; Interface definition.**

I. Introduction

Modularity —dividing a system into a set of interacting subsystems— has proven in the past to be a powerful concept in order to describe or build artefacts (such as software, organizations, etc.) so that they exhibit a set of advantageous characteristics, such as evolvability or lower diagnostic complexity. Increased evolvability, as it allows to adapt only a part (i.e., one module) of the artefact while leaving the other parts unchanged. Lower diagnostic complexity, as for instance erroneous outcomes can be traced to a particular subsystem (i.e., one module) instead of leaving the whole system to be analyzed for its cause. However, prescriptive guidance regarding how to design software systems or organizational artefacts as modular structures exhibiting these properties is rare. Normalized Systems Theory (NST) has recently proposed such prescriptive guidance to develop highly evolvable systems [1] with low complexity [2]. As a first step, the theory proposes a set of theorems which basically prescribe how to fragment a modular system (i.e., based on concerns such as change drivers or information units) for this purpose. In a second step, a set of patterns (called “Elements”) are proposed to realistically apply the theorems in practice. Both steps have already been performed and documented at the software level (on which NST was originally applied). At the organizational level however, only the relevance of the theory ([3], [4]) as well as some initial guidelines to partition business processes [4] as organizational artefacts (i.e., step 1) have been suggested. Therefore, no set of patterns (or “Elements”) at this organizational level are available.

In this paper, the authors propose a first attempt regarding the formulation of such potential Elements at the organizational level in order to make an initial contribution to this research gap. We will first briefly discuss some essential aspects of NST, showing how the Elements at the software level have been conceived (Section II). From this discussion, we derive a set of meta-requirement which should be fulfilled when formulating Elements at the organizational level in Section III and propose a potential initial set of five organizational Elements in Section IV. We then provide some illustrations (Section V) and offer our final conclusions (Section VI).

II. Normalized Systems Theory

In a quest for making systems more evolvable and less complex, NST has applied the concepts of stability from systems theory ([1], [5]) and entropy from thermodynamics [2] to modular systems and derived a set of formally proven *design theorems* prescribing how to identify and delineate individual modules. For instance, from the evolvability point of view, the Separation of Concerns theorem prescribes that every change driver (i.e., part of system which can independently change) should be isolated into its own modular construct (e.g., class in a object-oriented software environment). NST further shows that each of its principles should be systematically applied throughout the whole modular structure in order to truly realize its benefits. As this results in highly fine-grained modular systems, it is a very challenging effort to manually design a system in reality without any theorem violations.

Consequently, as a second step in its reasoning, NST proposes to employ a set of patterns (called “*Elements*”),

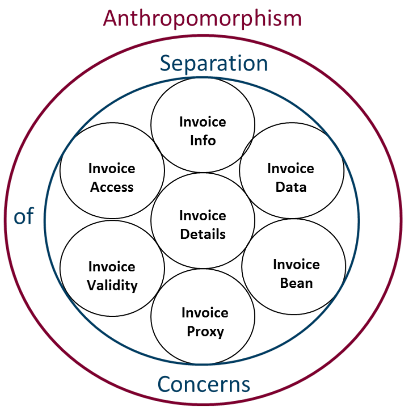

Figure 1. A NST Data Element (based on [5]).

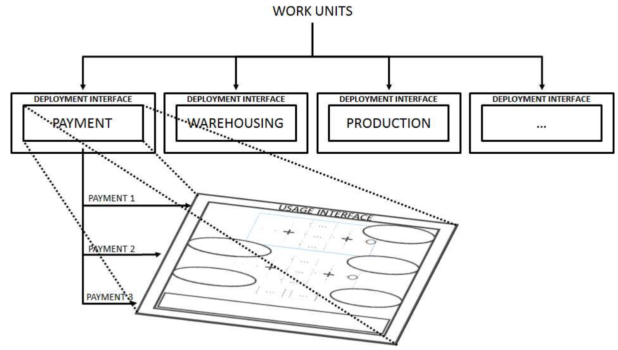

Figure 2. A schematic representation of an Organizational Work Unit deployment.

which are a structured aggregation of modular structures, conforming with the NST theorems and which can be more readily used to build real-life applications. At the software level, five elements were proposed: Data Elements, Action Elements, Workflow Elements, Connector Elements, and Trigger Elements. Experience has demonstrated that this set of elements indeed enables the creation of real-life (business or administration oriented) software applications [6]. Figure 1 represents the internal structure of one of these Elements, i.e., a Data Element for storing and exchanging information regarding an Invoice. Typically, such Element consists out of one core construct (here: software class), which performs or stores the main functionality of the Element (in this case: a set of data attributes, for instance the class “InvoiceDetails” for an Invoice Data Element). On top of this core construct, several other constructs are included within the Element taking care of the cross-cutting concerns. In case of this software Data Element, such classes are added in order to take care of persistency (e.g., “InvoiceData”), transactional integrity (e.g., “InvoiceBean”), etc. In essence, each of these classes incorporating such cross-cutting concern represents a “mini-bus” acting as a proxy or facade to the (external) technology used for this cross-cutting concern (e.g., JPA, EJB). Due to the NST theorems, the parts implementing a cross-cutting concern should be separated within their own construct inside the Element. Such design also ensures version and implementation transparency regarding functional changes (e.g., adding a new data attribute or updating the technology of one of the cross-cutting concerns) as the impact of these changes remains isolated within one instantiated Element.

While originally applied at the level of software systems, it has been shown that NST reasoning can be applied to organizational artifacts, such as business processes [4] or enterprise architectures [3] as well. More specifically, a set of guidelines has been proposed in order to help identifying typical concerns at this organizational level, which should therefore be separated [4]. This again results in a very fine-grained modular structure, difficult to arrive at each time “from scratch”. Consequently, a set of Organizational Elements might be beneficial at this level as well. However, such Elements have not yet been proposed.

III. The Meta-Requirements of Organizational NST Elements

Elements at the organizational level would imply the creation of organizational modules, which exhibit the same characteristics as the Elements at the NS software level. Therefore, we discuss three necessary characteristics for organizational “chunks” to become modules and even Elements eventually exhibiting high evolvability and diagnostability.

A. Defining exhaustive interfaces

Earlier, we defined modularization as “*the process of meticulously identifying the dependencies of a subsystem, transforming an ambiguously defined ‘chunk’ of a system into a clearly defined module (of which the borders, dependencies, etc. are precisely, ex-ante known)*” [7]. In doing so, we differentiated our definition from that of Baldwin and Clark [8] in which a module should already exhibit high intramodular cohesion and intermodular coupling. While we acknowledge that such properties are clearly highly valuable characteristics for modular structures, we stress the importance of exhaustively defining a subsystem’s interface (i.e., its interaction with its environment) before engaging in more sophisticated optimization efforts such as maximum cohesion and minimum coupling. In order to arrive at such complete interfaces for modules at an organizational level, we highlight the following aspects which should — at minimum— be taken into account when one’s aim is to arrive at blackbox modules having a fully defined interface.

First, a distinction could be made between those parts of an interface which are related to the “set-up” of the module (the so-called “*deployment interface*”) versus those parts related to the calling or invocation of the module (the so-called "*usage interface*"). The deployment interface can be understood as (mostly) a "one-time" interaction to get the module initially "deployed" and operational in its working environment (e.g., the necessary infrastructure), whereas the usage interface is related to the repetitive interaction each time a request (input) to produce a certain result (outcome) is directed towards the module. The difference between both types of interfaces is visually depicted in Figure 2. Second, both the "functional" or "front-end" interface as well as the "non-functional" or "back-end" interface should be recognized. The *"functional" interface* typically relates to the input arguments given to the module to provide a certain product or service (e.g., the amount of money to be transfered by a payment module). The *"non-functional" interface* concerns these interactions (or parameters) which are not necessarily viewed as directly linked to the product related input or output provided by a module, but are nevertheless necessary to obtain its correct execution (e.g., the presence of a well-functioning Internet connection). Often, this functional interface aligns with the usage interface whereas the non-functional interface mostly corresponds to the deployment interface. As we see that, in practice, the "deployment" and "non-functional" interfaces are often not explicitated, while necessary to obtain true black box modules, we suggested in the past some interface dimensions which should be taken into account in order to obtain exhaustively defined interfaces for organizational modules such as:

- *Supporting technologies* (e.g., SWIFT, the Internet);
- *Knowledge, skills and competences* needed by people carrying out tasks in the organization;
- *Money and (financial) resources* required to operate the module;
- *Human resources, personnel and time* defining how many (concurrent) people are required (and for how long) in order for carry out certain activities;
- *Infrastructure and real estate* required to perform the activities (e.g., offices, machines);
- *Other modules or information* needed for a module to operate correctly.

B. Aggregating modular building blocks into organizational Elements including cross-cutting concerns

Experience with NST at the software level has shown that obtaining evolvable and diagnosable modular structures, requires a very thorough separation of concerns resulting in a very fine-grained modular structure. However, obtaining such systems in practice only seems realistic when the designer has a set of "Elements" at his disposal, each being a structured aggregation (i.e., pattern) of modular building blocks. At the software level, the modular building blocks available to compose Elements are typically classes and objects in object-oriented environments or structures, functions, procedures in structured programming languages. In organizations, such basic modular building blocks might constitute people, the actions they perform (and their order), as well as the materials they need and/or produce. The formulation of the NST Elements based on these building blocks is likely to be an inductive process (i.e., the Elements eventually have to comply to the NST theorems, but are not directly derivable from them) and the union of all NST Elements should provide all basic functionality to build state-of-the-art modular structures within its application domain. At the software level, Data, Action, Flow, Connector and Trigger Elements were suggested which allow to store data, perform actions with them, define an order of sequence between these actions, connect with external systems in a stateful way and trigger the necessary Data and Flow Elements on a time-based manner if needed, respectively. Therefore, structured aggregations of modular building blocks might equally result in Elements at the organizational level, for instance the five Elements we will suggest as potential candidates in Section IV.

However, we showed in Section II how NST incorporates a set of classes in these Elements as proxies or facades to a number of cross-cutting concerns. The concept of cross-cutting concerns was already well-known in the software engineering field when adopted by NST. For instance, in the "aspect-oriented" programming paradigm, attention to the principle of separation of cross-cutting concerns was already given. Nevertheless, in many other application domains in which modular designs are present, such cross-cutting concerns seem to be useful as well. Consider — as a simple example— the modular design of a house. Each of the rooms (e.g., living, kitchen, bedroom) could be considered as a separate module, each having their own functionality. However, while there might be some central and separated water, heating and electricity supply and management systems, these facilities typically have to be available throughout the whole house. Each room wants to have the possibility of tapping water, using the heating system and employing electrical devices. As a consequence, electric cords with sockets, water pipes with taps and heating pipes with radiators are incorporated in each module (here: house room), connected with the central facility provision units. Therefore, the water, heating and electricity facilities in a typical house can be considered as quite adequate illustrations of cross-cutting concerns in the construction area. As organizations have been argued to be modular structures, several modular structures seem to have "cross-cutting concerns" and NST states that these cross-cutting concerns should be incorporated in a structured way into Elements in order to make the structures evolvable and lowly complex, it seems interesting to incorporate such cross-cutting concerns in organizational Elements as well. The question then obviously becomes: can we identify cross-cutting concerns at the organizational level, and of what kind are they? Once we have some insight into this aspect, we can start to formulate Elements at the organizational level.

C. Identifying cross-cutting concerns

NST reasoning suggests that identifying cross-cutting concerns at the organizational level would allow us to obtain highly evolvable and lowly complex artefacts in this domain. First, the concept of "cross-cutting concerns" in organizations seems an appealing thought intuitively. Indeed, one can easily imagine the necessity for most parts or modules in an organization to communicate with items external to the module, or report relevant items in the bookkeeping system. Second, certain business modeling languages seem to acknowledge the importance of such cross-cutting concerns in an implicit way: to keep the modeling and mental complexity manageable, they make abstraction from certain aspects during their modeling efforts while concentrating on mostly one relevant aspect regarding each functional domain of an organization, thereby reflecting their "view" on the functioning of organizations. For instance, regarding the bookkeeping aspects of an organization, the "Resources, events, agents" (REA) approach provides a method to design accounting information systems [9]. As a consequence, an organization is modeled as a set of economic events (performed by economic agents) resulting in stock flows in terms of economic resources. As this is the primary focus of the modeling method, other organizational aspects are mainly put out of scope. Another example, regarding the communication aspects of an organization, involves the Enterprise Ontology (EO) theory and its accompanying DEMO method, claiming that the essence of an organization resides within a generally recurring communication pattern regarding ontological actions (i.e., involving human decisions or the creation of products) [10]. As a consequence, an organization is primarily modeled as a set of actors engaging in the request-promise-execution-statement-acceptance (standard) pattern of ontological (communication) acts and facts, abstracting away from other organizational aspects (e.g., the implementation of the execution step).

Without claiming to have the optimal or even an exhaustive set of cross-cutting concerns, we suggest the following concerns as being some suitable candidates of cross-cutting concerns for the "Elements" at the organizational level (all of them will be incorporated in one of our proposed Elements, i.e., the Work Unit Element):

- **Registration or logging**: As tasks and their encompassing flows are executed, relevant information should be logged and registered. Multiple different kinds or "dimensions" of relevant information could be thought of in this respect, e.g., throughput time, resource consumption, quality metrics, etc. Therefore, in order to reduce complexity during and after execution time, this information should ideally be logged at the fine-grained level of information units, i.e., each part within a business process design of which independently traceable information according to these dimensions is assumed to be needed later on [11]. This information should be persisted in one way or the other, which is the responsibility of the constructs making up this cross-cutting concern. Such information may then be used (and possibly aggregated) later on for diagnostic, KPI-reporting, business intelligence and other purposes.
- **Transactional integrity (including cancellations)**: On top of registering certain properties regarding information units during the execution of task(s) (flows), some modules should be able to handle requests from outside, as well as internally triggered actions based on stateful transactions (i.e., interacting life cycle information objects going through their respective life cycles). Amongst others, this means that a flow can only be in one state at a time, no conflicting or contradicting states should be able to occur and cancellation requests are handled in an appropriate way (e.g., no blind interruption of the regular or the "happy" path). These predetermined transaction structures should be enforced one way or the other, which is the responsibility of the constructs making up this cross-cutting concern.
- **External communication**: As the organizational modules are expected not to work in isolation, they should be able to communicate via incoming and outgoing information streams with other modules. However, as communication issues (such as message format, fault handling, background technology) clearly do not belong to the "core" responsibility of each genuine (i.e., non-communication dedicated) module, (the connection to) these issues should be handled in different construct instances than those which do handle this core responsibility. Consequently, to enable the interaction of a module with external modules, communication should be incorporated as a cross-cutting concern within organizational elements.
- **Authorization policy**: Typically, flows and their constituting tasks are only allowed to be executed by certain actor(s) (roles) due to safety, legal and company defined regulations. In addition, in certain cases, the actual identity of these actor roles should be verified. As the common role definitions and identity verifications should be accessed and used throughout several organizational modules, these concerns seem to depict a genuine cross-cutting concern as well, handling these functionalities.
- **Bookkeeping adapter**: Many executed tasks and flows result in one way or the other to changes in the bookkeeping ledgers or financial reporting systems of an organization. However, bookkeeping standards as well as the way in which certain goods and assets are valued, are clearly a different kind of concern as the "core" activities of, for instance, a procurement or assembly module. Therefore, some constructs should

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be able to extract the information needed for several bookkeeping standards (based on the logged or registered information) as a cross-cutting concern within many organizational modules, and provide it to central bookkeeping and financial reporting modules [12].

IV. Towards Normalized Systems Elements at the Organizational Level

Looking for the Elements which might be needed in order to fully describe the functioning of organizations, we adopt the following perspective. As depicted in Figure 3(a), the behavior of organizations can basically be described as a number of interacting Organizational Work Units, executing (flows of) tasks by employing a set of internal Assets or Resources. Eventually, these Work Units deliver Products and Services to Parties external to the organization (such as customers). The delivery or procurement of certain Products or Services is generally associated with an act in return: a Compensation. To the extent that this set of concepts allows us to model a domain (in this case: organizations) and explain some reasoning about it, it could be regarded as a kind of new, modest ontology regarding the functioning of organizations in terms of Elements. However, before being able to make this claim, we should more specifically define each of these concepts and articulate the relationships between them.

- **Party**: A Party is a (natural) person, company or legal entity which is doing business. Whereas we identify an Entity as anything from a company to an organizational department, we consider a Party as an identifiable "actor" which has the authorization to act on behalf of an entity. The core of the Party Element is the listing of its identification and idiosyncratic properties. Additionally, a set of cross-cutting concerns is added, which are typically needed for each Element of this type (e.g., external communication). For instance, a company procuring Bike Parts, assembling them to Bikes and delivering them to customers, could be considered as a Party (as well as its suppliers, customers and employees under contract).
- **Product or Service**: A Product (typically tangible) or Service (typically non-tangible) is something an organization (which is a Party) is capable of providing to its customers (which are also Parties). The core of the Product or Service Element comprises the definition (its characteristics) and nature of the Product or Service (i.e., production details, marketing details, etc.). Moreover, it contains some links to Organizational Work Units (cfr. infra) performing certain (sets) of tasks necessary to actually provide the Product or Service according to its characteristics defined in the "core". For instance, in the case of a bike producing company, a Product could constitute a Bike so that Customer ABC could request Company XYZ to produce a particular Bike instance with chassis number 123. While delivering this Bike instance, the Bike company will most likely rely upon several of its Work Units (e.g., Procure, Assembly).
- **Compensation**: A Compensation is an act in return from one Party to another Party, generally because the latter Party has received or will be receiving a certain Product or Service from the first one. As a consequence, a Compensation is typically linked with a Product or Service provided by one Party. It is important to note that a Compensation might take on several forms (e.g., an invoice, followed by a payment, etc.), not necessarily being of a financial kind. Additionally, a Compensation might have instantiation frequencies which are not necessarily completely aligned with the instantiation frequency of the associated Product or Service. Therefore, the core of the Compensation Elements consists of the definition and properties (i.e., the "terms") of the Compensation, supplemented with a set of cross-cutting concerns to complete this Compensation according to its specified terms. For instance, a Bike producing company could issue a Compensation instantiation with a customer as a result of delivering a Bike to that Customer earlier. Alternatively, the company might have to fulfill a monthly recurring payment procedure with its bank as part of its loan for its real estate ownership. As a consequence, also a Compensation is typically associated with one or more Organizational Work Units (cfr. infra).
- **Asset or Resource**: Assets and Resources are defined as both human (e.g., employees) and non-human parts (e.g., product parts, consumable raw materials, buildings), which are used for carrying out tasks and/or flows. We consider Assets or Resources as internal to an organization. However, these may be acquired or hired via contracting or purchases between Parties earlier in time. Therefore, the core of the Asset or Resource Element consists of the Asset or Resource identification and its characteristics, as well as a set of cross-cutting concerns to generally employ the Asset or Resource (e.g., communicate with it). For instance, the Assets or Resources for a Bike producing company might constitute of a set of employees (sales people, assembly people, etc.) and infrastructure (building, manufacturing equipment). In order for the company to employ these Assets or Resources, the company might have bought the building and have hired the employees resulting in a Compensation instantiation.
- **Organizational Work Unit**: An Organizational Work unit is the whole of a set of tasks, task flows and accompanying assets/resources needed to execute these tasks. A *Task* is a small part of work (performed by resources/assets such as human actors or machinery, and potentially consuming other resources/assets), iden-

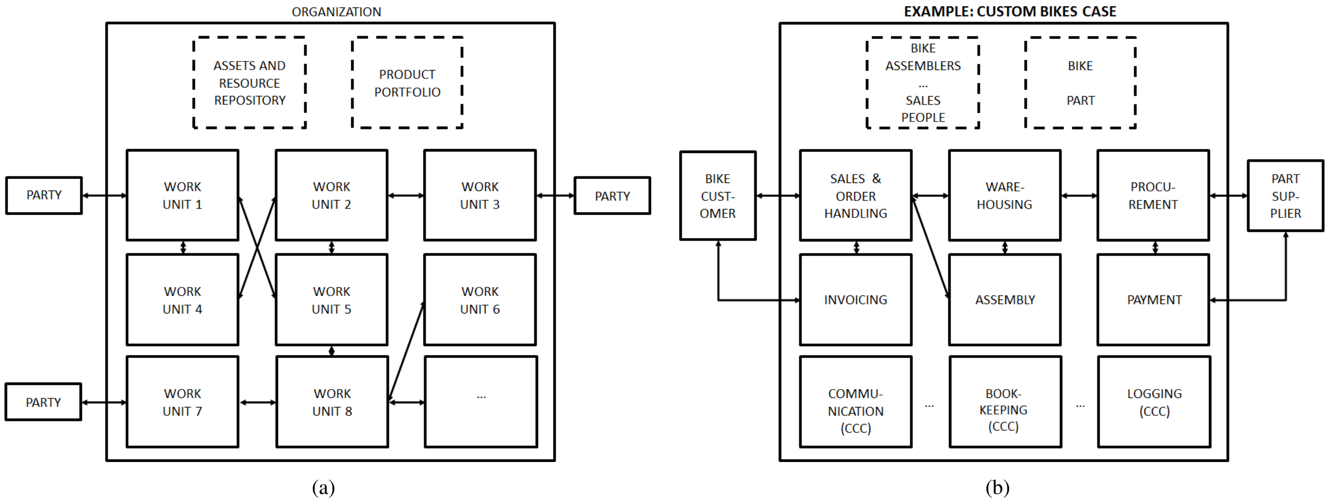

Figure 3. An organization modeled as the interaction between a set of deployed Organizational Work Units, using Assets and Resources, producing Products and Services and generating Compensations. Eventually, the Products and Services, as well as Compensations are exchanged with Parties external to the organization.

tified as being a separate change driver or information unit. A *Task Flow* is defined as a set of tasks in a particular order (including sequences, iterations and selections) all operating on one lifecycle information object. The internal design of the flows and tasks within the Organizational Work Unit should adhere to the guidelines for designing evolvable [4] and lowly complex business processes ([11], [13]), as documented before. Ideally, this core of the Organizational Work Unit (i.e., tasks and flows, complemented with a set of Assets or Resources) should also constitute a logical whole which is externally loosely coupled. However, in contrast with an exhaustively defined interface, this is not a prerequisite as a full interface enables the analysis and optimization of the delineation of these Work Units. Some relevant cross-cutting concerns for an organizational Work Unit were already discussed in Section III-C.

To sum up, we propose a Party, Product/Service, Compensation, Assets or Resources and Organizational Work Unit as a tentative set of Elements within the application domain of organizations. The Work Units use Task (flows) and Assets/Resources to perform their function. A potential internal structure (i.e., content) regarding the Products or Services, Compensation, Assets or Resources and Parties is shown in the various panels of Figure 4. Each of the Elements contain some core and basic information regarding that Element type (such as identification, description, etc.) as well as a set of cross-cutting concerns (visualized by the ovals) depicting some viable cross-cutting concerns for each of them. It is important that the proposed sets of cross-cutting concerns should not be considered as exhaustive or restrictive. Rather, our goal is to illustrate reasoning behind the aggregation mechanism for obtaining Elements, while incorporating cross-cutting concerns at the organizational level. As it is clear that the major part of organizational results is produced by the Organizational Work Units, the internal working of external organizations or people (i.e., Parties) is not to be manipulated, and Products/Services as well as Compensations largely depict the relation between multiple Parties, let us take a closer look at the internal structure (i.e., design of once it is deployed) of the Organizational Work Unit as depicted in Figure 5. As can be seen from the figure, we conceive an Organizational Work Unit as a set of individual tasks, a set of flows consisting of tasks, consuming assets and resources (e.g., raw materials, people performing the actions), incorporating the above enumerated set of cross-cutting concerns and surrounded by the appropriate (exhaustive) usage interface.

There are basically two main reasons why we chose to start from our own set of ontological concepts, instead of starting from one of the several already existing ones in extant literature. First, as we already argued in Section III-C, many existing business modeling methods in literature seem to mainly focus on one particular aspect (e.g., one of our suggested potential cross-cutting concerns for organizational modules). Second, we did not want to start our discourse on NST Elements at the organizational level by criticizing or claiming to ameliorate one particular existing organizational ontology. Nevertheless, we are aware that other ontological frameworks might have similar or related concepts (e.g., many frameworks acknowledge the existence of things like “actors” or “parties”), but the set of our five proposed Elements and their accompanying interpretation is —to the best of our knowledge— new. Also, we think that our proposed ontology largely aligns with the “common denominator” or a

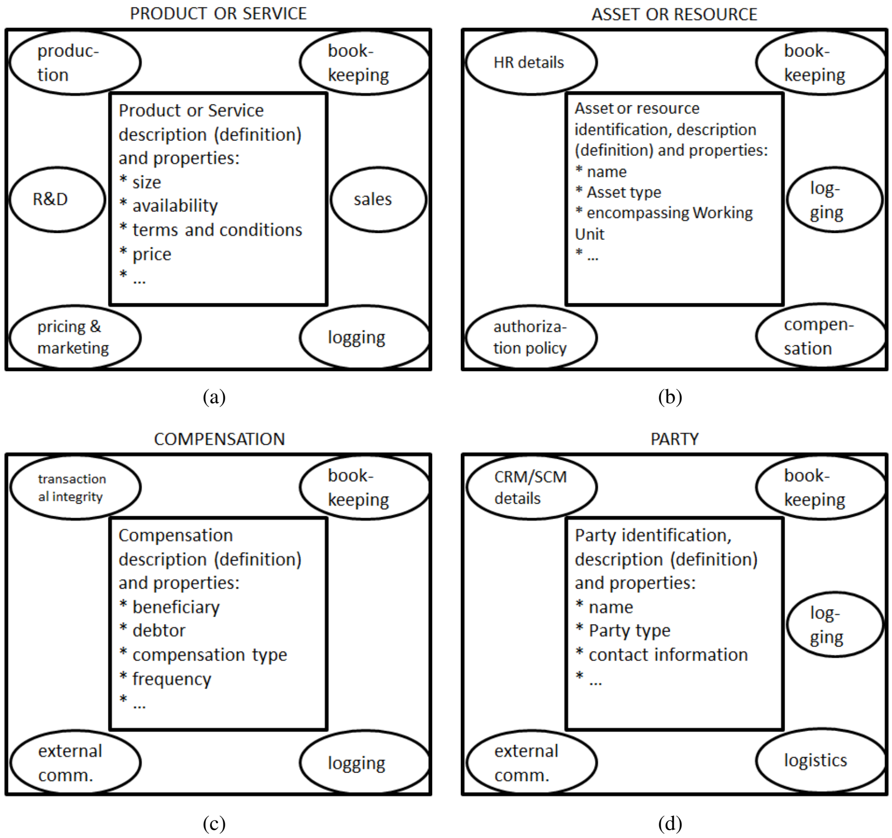

Figure 4. Organizational Product/Service, Asset/Resource, Compensation and Party Elements.

lot of "heuristic feeling" when conceptualizing about organization (just as our NST ontology at the software level did). An organization basically consists of Assets or Resources carrying out task and their flows (eventually aggregated into modules), in order to deliver Products or Services to Parties in return for some Compensation. However, we encourage other researchers to compare our NST reasoning with their own business modeling efforts and look for opportunities of cross-fertilization between both. For instance, we hypothesize that it should be perfectly possible to combine (certain parts of) our NST reasoning on organizational Elements with some of the already existing ontological frameworks (such as EO, REA, etc.).

Consequently, equally as for the "Elements" previously proposed at the software level, we are fully aware that our initial ontology (both the set of elements as well as their internal structure) is only one possibility of modeling and modularizing a company, and that other ontologies or Element structures are possible, and might potentially even be more optimal. Rather, the main purpose of our proposed organizational Elements is to provide a constructive proof-of-concept regarding the feasibility of modeling and designing organizations realistically in a way adhering to the NST theorems. However, the authors want to emphasize that our organizational elements seem to allow modularization in the three dimension as mentioned by Campagnolo and Camuffo (i.e., product design, production system and organizational design) [14]. This indicates that realistically deployable organizational modules seem to be at the intersection of business processes and organizational departments, thereby combining several viable modularization dimensions within organizations (e.g., according to functional domains, mainly process-oriented, etc.). Consequently, modularization as conceived by NST Elements does not necessarily imply extreme specialization due to its fine-grained separation: at the level of an Element, each module has its own "core" responsibility, supplemented with a link to a variety of cross-cutting concerns.

V. Illustrations

In this section, we will illustrate our reasoning from above in two ways. First, from a more high-level approach, we will show how nearly all aspects of an exemplary organization (labeled as "the Custom Bikes case" by Van Nuffel [4]) can be described or categorized while employing our proposed ontology. Second, from a more bottom-up approach as suggested by NST, we will elaborate on the possible internal design of one possible Organizational Work Unit, a Payment

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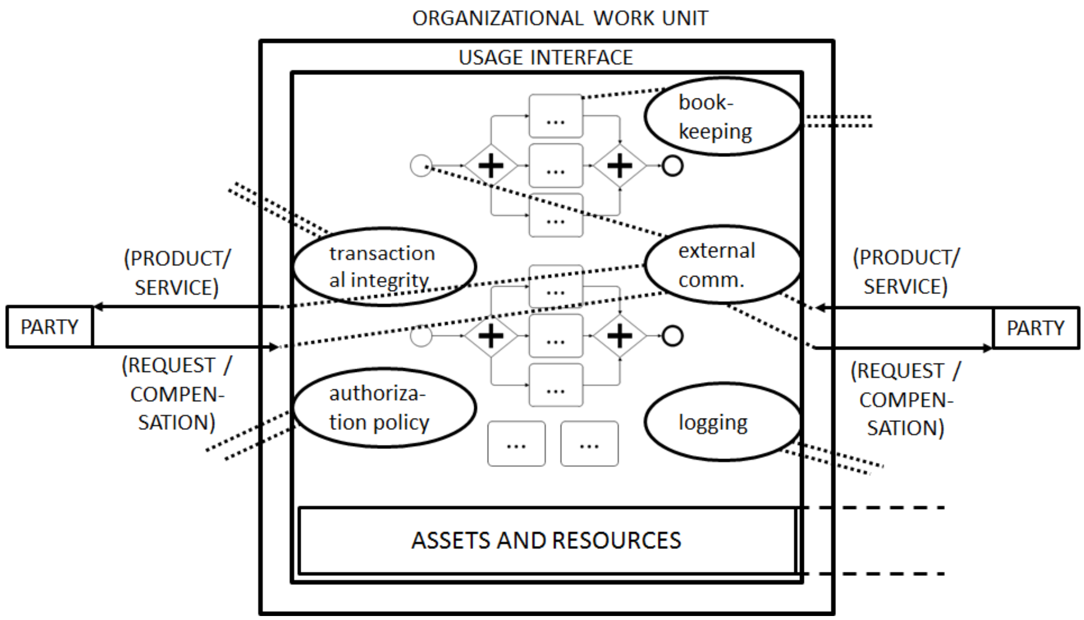

Figure 5. A schematic representation of an Organizational Work Unit, an Element at the organizational level according to NST reasoning.

Element.

A. Applying the Organizational Constructs and Elements

The “Custom Bikes case” [4] concerns a small company producing customized bicycles. Basically, for each order which is received, an order handling process starts (e.g., checking the customer details, evaluating and categorizing the order, manager approval, order scheduling). If needed, a process can be executed to order some missing parts, after which the assembly of the bike is done, shipped and invoiced. The result of applying our organizational constructs to this organization is shown in Table I and Figure 3(b). We can notice that nearly all aspects of the case can be categorized within our set of ontology constructs. In the work of Van Nuffel [4], most efforts have been invested in the identification and modeling of the tasks and flows. While one can argue that some initial attention has also been given to Products or Services, Compensations and Assets or Resources, the modeling of Parties and (often interface-related) Assets or Resources seems to be abstracted away in the work. Additionally, no Organizational Work Units according to our definition can be discerned. In essence, this means that a thorough fine-grained separation of concerns was performed regarding task (flows), but that no patterns or organizational “Elements” were explicitly identified. As we explained earlier, this type of fine-grained modularization is a first important step in the normalization of modular structures. However, the systematic definition of an exhaustive interface, incorporation of cross-cutting concerns, as well as a structured aggregation of the flows of activities (together with its Assets or Resources) should, in our opinion, be performed in a next stage in order to arrive at more “realistic” organizational modules and “Elements”.

Table I
APPLYING THE ORGANIZATIONAL CONSTRUCTS AND ELEMENTS

Construct type candidates	Manifestations in Case
Product or Service	Bike Part
Compensation	Invoice Payment
Party	Customer Employee Part Supplier Custom Bikes Company
Asset or Resource	Sales people Bike Assemblers Warehouse employer ... Manufacturing plant building Manufacturing equipment Electricity supply ...
Tasks and their flows	Order Handling flow Customer Entry flow Purchase Order flow ...
Organizational Work Units	To be identified

B. A Design Case: Towards a Payment Element

Let us consider the inner design of a possible Payment Element, responsible for carrying out the payments in an organization, to demonstrate the viability of an organizational Work Unit in reality. If we start by drafting a **deployment interface** of such Element, we see that some Assets/Resources are required to operate the module. First, we need a *human person* to operate some of the task(flows) within the Element, for instance expressed as a percentage of full-time equivalents. Some *IT infrastructure* (e.g., servers, PC) might be needed, as well as a connection to *external services* completing the payment (e.g., Internet, Swift, Isabel). Finally, we probably need some *infrastructure* as well (buildings, desk, etc.). Ideally, this deployment interface exhibits version transparency, meaning that if the Payment Element or the Elements using the Payment Element are changed (i.e., they go to a new version) the deployment interface remains unchanged (i.e., no additional Assets/Resources need to be deployed in order to ensure the proper functioning of the Element).

Realizing the deployment interface is similar to a "constructor" in a software environment, in the sense that these Assets/Resources become available for the deployed and implemented module. As we consider Assets/Resources as Elements as well, each of their instantiations is again encapsulated within their own set of cross-cutting concerns. For example, the employment contract with the employee is handled by the cross-cutting concern making the link to a Compensation. The needed bookings in the bookkeeping records for employing an Asset/Resource are handled by the bookkeeping adapter cross-cutting concern (e.g., the yearly depreciation of a building) and any changes in their "state" (e.g., extra courses followed by an employee) are followed up and tracked by the logging cross-cutting concern. This also implies that a particular Work Unit Element could make a request to such Asset/Resource Element instance to receive the information about its yearly costs (e.g., in terms of depreciation or wage) for (for instance) their own cost accounting purposes.

In terms of the **usage interface**, each individual payment to be carried out is most likely associated with a *Compensation* stipulating a certain amount to be paid by the company at a certain point in time in a certain currency to a certain account. Also this interface should ideally be version transparent so that a new version of a Compensation can still be processed by the existing Payment Element and that new implementations of the Payment Element can still process the existing Compensations. The actual execution of this payment can be typically seen as a set of (orchestrated) *tasks*, being part of a *(sub)flow*. While some of these tasks will be performing a genuine part of the payment execution logic (e.g., perform a data check, verify amount of current bank account, initiating the electronic transaction, etc.), some of them will use the proxy or facade constructs within the Element triggering a cross-cutting concern. For instance, the *authorization* cross-cutting concern can be used to verify whether only entitled persons carry out payments (e.g., only managers are allowed to trigger payments over 1000 euro). The *logging* cross-cutting concern would log (for instance in a database) for every flow instance and task instance, when it is performed, by whom, how long it took, which resources were consumed etc. The *transactional integrity* cross-cutting concern might use some rules to make sure that cancellation requests for a Payment are consistently rejected or might be performed when certain conditions are met, but would in any case not allow the uncontrolled interference of any cancellation request with the stateful and ongoing transactions within the Element. Finally, the *bookkeeping* cross-cutting concern of the Payment Element could make sure that the outflow of monetary resources is registered in the central bookkeeping and balance sheet. Informed about the completion of the Payment, the bookkeeping cross-cutting concern of the associated Compensation can make sure that this debt is removed from the balance sheet.

Internally, the modular design of such Element and its flow(s) is compliant with the NST guidelines for attaining stability and avoiding entropy. It should be clear that such element is not merely restricted to contain only one core flow or process. For instance, in case we should decide that the considered payment Element would not only contain a process to pay debts but also to check whether a claim from accounts receivable has been fulfilled or not. In that case, the "core functionality" of the Element would contain at least two processes.

Consequently, from a dynamic perspective, this Work Unit might cope with several **changes** in a stable way. First, the *resources* executing the task(flow)s might change (e.g., a human operator which gets replaced by a machine or software application), without affecting other Work Units as long as the usage interface remains unchanged. Second, the way a *cross-cutting concern* is implemented or the actor performing it, can easily be modified. For instance, if another bookkeeping standard should be adopted or a specified Party is appointed to take care of this cross-cutting concern (e.g., an external bookkeeper), the changes are isolated within this cross-cutting concern (and therefore not spreaded out among the core functionality flows of other Elements, which are just making the link via the proxies in which they are encapsulated). Third, specialization and optimization of this Work Unit could lead to the *re-use* of this Element within multiple organizational departments or even on cross-organizational scale. Indeed, one can imagine that such Payment module might not only cope with internal Compensations to be settled (i.e., coming from one's own company), but in fact any Compensation (as this could be the only prerequisite listed in the usage interface) including those from other organizations as well. Therefore, "performing payments" could become a re-usable Element or even a stand-alone business in the long term if wanted. Once again, this confirms our view on organizations as being essentially a (structured) aggregation of Work Units, coupled via exhaustive interfaces which are ideally loosely coupled.

Also, as the design requires very fine-grained information tracking, the degree of entropy can be controlled. For instance, additional cross-cutting concerns could be

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imagined to be added to the Elements, later on. Consider a company wanting to perform *cost accounting*. In such case, an additional cross-cutting concern "cost accounting" could be incorporated, performing cost calculations based on the logged information (e.g., "500 invoices have been processed in one month, needing 1 FTE") and bookkeeping information (e.g., "this 1 FTE has cost of 3000 euro a month").

VI. Conclusion

This paper addressed for the first time explicitly the research gap regarding the NST Elements at the organizational level, similar to the Elements which exist at the software level. While we do not claim to have solved the research gap completely, the main contributions of this work should be situated in advancing the application of NST reasoning at the organizational level. More specifically, we discussed the general rationale and mechanism of such Elements, listed a set of three necessary meta-requirements for Elements at the organizational level (i.e., exhaustive interfaces, the aggregation of fine-grained modular building blocks, and the identification of cross-cutting concerns which should be included), and proposed an initial set of five Organizational Elements in this respect. In further (extended) work, we aim to discuss some more in-depth case studies based on our reasoning and look for potential issues during their realization. For example, while we are convinced that our reasoning regarding the formulation of exhaustive interfaces is theoretically solid, several issues might arise when trying to actually describe it in reality, determining the overall potential for realistic application. Additionally, it would be interesting to complement our work with domain related knowledge and expertise so that some of the Elements might become best-practice organizational artefacts which might be adopted by several organizations in the long term.

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SmartModels – A Framework for Generating On-line Assessment Systems

Emanuel Ţundrea
Griffiths School of Management
"Emanuel" University of Oradea
Oradea, Romania
emanuel@emanuel.ro

Abstract **- Globalization has set a strong mark on the way education can be provided. Every knowledge institution provider has to publish and offer his expertise for a wider readership in order to stay on top. One of the prime-time opportunities is to build web-based software solutions in order to support distance learning. There are so many ways to organize a class and professors can imagine so many ways to evaluate students that the complexity of such a software system can be hard to implement and it can hardly foresee future education forms. The Model-Driven Architecture (MDA) project from OMG promotes the use of meta-modeling in order to drive the system's design and implementation. In this context, this paper presents (i) an approach – it reviews the SmartModels approach briefly introducing its principles, basic entities and main elements when defining a business-model; and (ii) a prototype – SmartFactory, which is based on Eclipse platform and its role is to validate the new approach. It addresses the paradigm of how to practically implement MDA principles and rules for software engineering. Therefore, it deals with important implementation issues based on Eclipse Platform. The examples in this paper target the process of developing e-learning deployment tools for on-line assessment solutions.**

Keywords-meta-modeling; SmartModels; SmartFactory; software product lines; on-line learning assessment system.

I. INTRODUCTION

SmartModels aims to address in a practical way the MDE [6] principles. It is an approach which integrates these concepts and proposes a way of developing domain-specific software based on models as a more flexible option to the Meta-Object Facility (MOF) [6] plus Unified Modeling Language (UML) [11] approach. It gains know-how from a previous research also on meta-modeling [5]; together with it, a prototype called SmartFactory was developed in order to validate and get feed-back from users.

MDA approach, as Object Management Group (OMG) [6] established it, has the advantages of stability and platform-independence through defining business functionality and behavior in a base Platform-Independent Model (PIM) technology-independent way. This means that an approach based on models has many advantages primarily from the design point of view, but also from the implementation point of view (application coding, management and maintenance). There are many proven examples on developing standards like SQL, GUI builders, HTML or regular expressions.

One of the main problems companies face today is that even if there is a perfect model, the programmers have to make a lot of compromises when trying to implement the model using a specific programming language, when mapping to a platform and when needing to adapt to new requirements and redeploy.

This is one of the main concerns of the Object-Oriented Programming (OOP) principles which did not cure important issues faced by software companies these days on developing complex software for reuse and protecting the more and more evolving applications against technological obsolescence.

This problem applies to the process of developing e-learning deployment tools, when trying to encapsulate all type of possible knowledge presentations or questions from an assessment.

A very interesting technical solution comes from adaptable Service-Oriented Architectures (SOA) [14] which integrates the Aspect-Oriented Programming (AOP) [4] as a new design solution. For example, Monfort et al. [12] have implemented an infrastructure to dynamically reroute messages according to changes without redeployment. This way, companies gain flexibility, but it is still much to be done in order to obtain genuine flawless applications, and current market implementations still do not provide means for easy adaptable application behavior.

Section III presents SmartModels response to this challenge by the way it encapsulates meta-information of complex entity families. Thus, the meta-level defined in the design phase can dynamically control the reification level and even the instance level of an application. More objective, section III.C shows how SmartModels integrates AOP technology through dynamic aspects called actions.

Another interesting trend is well ascertained by Pohl et al. [13]. This solution makes use of Data-Flow Diagrams (DFD) [11] as a basis for a model-driven development framework for embedded target platforms. New target platforms can be easily added using platform specifications based on Orthogonal Variability Models (OVM) [11]. The downside is the fact that in reality these tools are tailored to produce code for a specific hardware platform or virtual machine. Tailoring is mainly done with hand-written code or generation through restrictive and complex formal specifications.

There is no broader vision on what is to be done if the platform evolves (which is the case so often).

SmartModels tries to reduce this gap between the design and implementation and to ensure the independence between the model, future family of applications and the pressing need for technological adaptability. Through its small kernel and a set of basic entities, it provides a framework to describe models and a software factory to automatically generate code as much as possible. This means that the applications may be re-generated at any time if the model or the technology evolves and also the model instances can drive the behavior of the application at code generation time or at run-time.

For the time being, one last argument for choosing SmartModels is that, together with its SmartFactory prototype, it forms a possible and feasible way to apply these principles. The tool is implemented on Eclipse platform which has prime time today in the realm of researchers on software engineering.

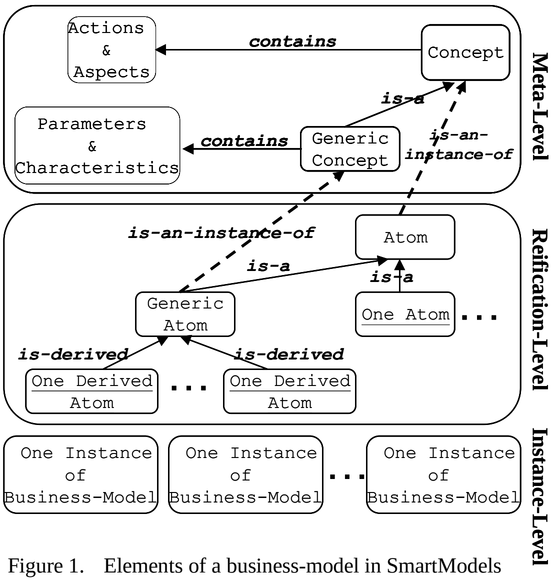

Figure 1. Elements of a business-model in SmartModels

Therefore, SmartModels applies MDA [6] and Domain-Driven Development (DDD) [3] principles through reusing the know-how of promising platforms for building Integrated Development Environments (IDEs) like SmartTools [1] or Eclipse [10].

The main objective of SmartModels is, on the one hand, to clearly identify, thanks to a meta-level, the semantics of concepts used for the modeling of a given domain, and, on the other hand, thanks to approaches by separation of concerns and generative programming [2], to equip, in a modular way, the applications related to this domain [9].

Next section will present each entity of SmartModels with respect to the level it manifests. Section III presents the process of deploying these models built around e-learning tool examples using the SmartFactory prototype. Section IV summarizes the results of the experiment and provides insights on potential perspectives on how to further develop the expressiveness of both the approach and the tools.

II. SMARTMODELS – AN APPROACH BASED ON MODELS

Figure 1 distinguishes between the different levels of the architecture of the meta-model: the main elements proposed by SmartModels in order to define business models.

A. SmartModels Meta-Level

First of all, the *meta-level* is the top level of SmartModels business-model reification and it handles the meta-information through concepts. A concept participates to the definition and the management of the meta-information of a business-model. A separation between the meta-level and the reification-level is important in order to clearly identify the relevant common and variable concerns of a family of entities which compound a domain and even identify inter-related concerns of a set of closely-related domains. The meta-level contains the model's entities information (abstract) and rules which define their behavior and set the foundation for describing their infrastructure requirements.

The definition of the meta-information of a model is under the responsibility of an expert of the business domain to which the model is dedicated. Therefore, it encapsulates the semantics of entities and their treatments. It can be related to one or a number of *atoms* and drives their behavior. Just as a forward-looking it is important to mention that in SmartModels' approach an atom is the structure which encapsulates the description of an entity.

The main entity which points out the clear separation between the semantics (meta-level) and their reification (reification-level) of the SmartModels model entities is the *Concept*. The choice to encapsulate the meta-information at the meta-level has a couple of very important advantages:

- The support for reuse of the semantics in other (closely related) models;
- The maintenance of the semantics (updating and redefining of the semantics) deals only with the meta-level (concepts);
- The model transformation, which is one of the goals of the approach.

The semantics of a business-model stored in a concept are reified through a set of hyper-generic parameters and characteristics [5] (which form the meta-information) and a set of actions (which perform treatments on the entities according to their meta-information). The identification of the parameters and characteristics and their possible values is the job of the meta-programmer which addresses the know-how of the business-domain. Based on the set of parameters corresponding to a SmartModels' concept, the approach can make a differentiation between the families of entities of a domain and based on their values it can distinguish the entities within a family.

The hyper-generic parameters customize the behavior of the entities (it refers to generic atoms, see section II.B, and not to their instances) of a business-model. Their role is to capture and express the properties which compound the definition of the generic entities. A parameter expresses a basic type property, e.g., a boolean or an integer value, an enumeration, a tuple type or a collection of those values. A characteristic expresses a property whose value is defined by atom(s) which are defined within the model (enumeration, tuple or collection). The programmer has to set those values to describe the behavior of a generic entity. For example, a business-model built to encapsulate the structures (entities) and semantics of a tool to create on-line assessment (quizzes) solutions may present parameters like:

- *MultipleAnswerCardinality,* which tells if a question corresponds to a single (1) or multiple possible correct answer (2..*) or
- *ForceExactAnswer,* which expresses the requirement to accept only precise answers (TRUE) in case of expecting a name or checking for spelling mistakes, or if it is interpreted together with the first parameter it can have the meaning of accepting an answer only if all choices are correctly set, or
- *TimeLimit,* which adds the aspect of time limitation for the specified assessment or question,

and characteristics like:

- *PossibleImageTypes,* which indicates the list of accepted picture file types (let's assume that images are reified through basic atoms in the model).

Actions are "first-class" entities described by concepts in order to dynamically manage the behavior of atoms according to their meta-information. The body of an action encapsulates the execution which can be performed by that action. This execution usually takes into account the set of parameters and characteristics of the generic atom to which the action is attached and optional can present a set of aspects [4], invariants, preconditions and post-conditions. For example, an action can check the remaining time to limit the work on a question or can verify the image links that the user tries to import in the project.

This is the line of demarcation between semantics (*the meta-level*) and data of a business-model (*the reification-level*). As it has been anticipated in the previous section, an atom is the reification of entities of a business-model. Identifying the atoms of a domain is an important task of a programmer.

B. SmartModels Reification-Level

The entity of a business-domain is encapsulated in a model hierarchy through *atom* – a structure which holds the entity data and, which is similar to the MOF [6] "class" notion, is available in most of the object-oriented programming languages (OOPL). It can be used to factorize the data of a domain and has instances within the applications which rely on the given business-model.

In this context, it is important to learn about the SmartModels distinction between *basic and generic atoms*. An atom is generic if its meta-information presents parameters and/or characteristics. If an atom does not need additional semantics besides its data-model, it is called *basic* and it will have direct instances within applications.

Now, it is the time to introduce the notion of *derived atom* (Figure 1), which is an instance of a generic atom obtained through relevant combination of values associated with the sets of characteristics and parameters that participate to the definition of its *generic* atom.

Going back to the previous example of a tool to create on-line assessment solutions, the user can identify:

- *basic atoms*: image types, answers
- *generic atoms*: a question with hyper-generic parameters, characteristics and actions like those presented in Section I.A. One can imagine its heirs being all sorts of question types: hot-spot (allow student to answer by selecting an image from a set), forms to entry text, drag and drop images (set up a draggable image over a list of possible correponding images), labelling (label a set of images to match their text descriptions), text identification (for example to identify each spelling mistake in a passage), true/false questions. The list of generic atoms can be designed so it can benefit by the advantages of polymorphism. This way, the model can continue to be enriched by adding other heirs to describe new types of questions at the meta-level (to enhance their semantic information) and at the reification level (to enhance their structural properties).
- *derived atoms*: examples of questions having different properties: i.e., a free-form text edit derived atom is a text form question atom with the following meta-information (Section III.D for a detailed diagram and presentation):
    - forced exact answer: FALSE;
    - time limit: NONE.

  or a type of image labeling can be described by:
    - multiple-answer cardinality: 4 (the correct choices can be up to four);
    - forced exact answer: TRUE;
    - time limit: "00:05:00" (exactly five minutes);
    - image types accepted: basic atoms like JPEGImage, PNGImage, BMPImage.

Actions will check the conformance of the derived atoms' parameter values during the creation of new questions or at run-time checking the constraints (time limit, mandatory exact answer).

C. SmartModels Instance-Level

At this point, the benefits of the SmartModels approach can be observed. A possible application to the example mentioned above can be a tool for easy developing and deploying flexible and reusable on-line learning software systems, which describes and presents knowledge and also which can generate assessments and provide assistance on the evaluation process.

This section summed up the design of the approach. Next section presents SmartFactory – a practical implementation of SmartModels principles. It will try to demonstrate its interest through building the above mentioned domain model and generating its applications.

III. SMARTMODEL OF THE ON-LINE LEARNING ASSESSMENT SYSTEM

This section explores more the example of building a model through the SmartModels' approach for deploying a tool to develop on-line learning assessment system. It is based on a previous experience experimented in the I3S research laboratory [7]. Figure 2 presents the UML class diagram of the concepts and atoms which participate to the semantic and structural description of the model.

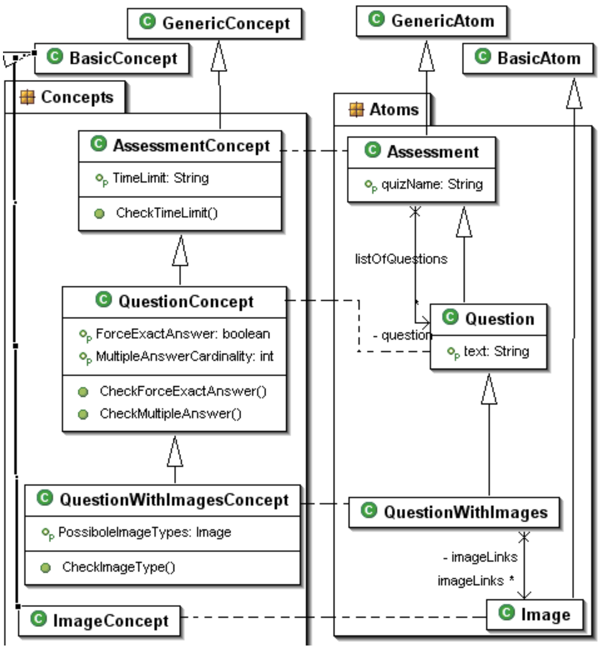

Figure 2. On-Line Assessment SmartModel Class Diagram

The *BasicConcept*, *BasicAtom* and the related generic atom and concept pointed out as supertypes for the model entities are presented just for the purpose to underline the way it attaches each new model to SmartModels kernel. They represent the abstract entities from the built-in metamodel.

Before going further it is important to see the SmartModels methodology to describe a business model. This is a five-step process:

1) to identify the basic atoms of the model,
2) to identify the generic atoms,
3) to define the criteria of genericity (the hyper-generic parameters) – typically, this is a step that must be performed by an expert of the domain. It represents a part of the knowledge of the business model;
4) to specify the actions attached to generic and non-generic atoms,
5) to specify the instances of the generic atoms (derived atoms).

Following this methodology, the next sections will highlight several important features of SmartModels and their advantages applied to this specific use-case.

A. Define Meta-Information of Complex Entity Families

One of the hardest part of creating a flexible tool for developing on-line assessment is that there are many ways a professor can imagine the evaluation of students. One can decide to create multiple-choice questions and require correct answers on all choices to mark all points. Another professor can decide to mark just the good answers and offer some points; others may even think of a weight for each answer and subtract points if a student makes wrong choices.

A professor can also imagine a requirement for evaluating a quiz to get an exact answer. This can mean checking for spelling mistake in case of a free-form text question or labeling correctly a set of images.

Should the structural entities of the model which describe the different types of questions of the quiz be equipped with all the information about all possible ways to evaluate them? SmartModels is a framework which makes a clear differentiation between the semantic information and reification of the families of entities of a domain. In this paper's example, the model can unambiguously separate the description of the structural features of each question type from the concerns that deal with the process of evaluating them.

This means that the user will define atoms only to encapsulate the different structural features of each question type without having to concentrate on the way they will be evaluated or mixing each type of question with all its possible evaluation manners. On a meta-level, the user can concentrate on creating rules to evaluate a quiz. More than that, he can create rules which apply at the level of a single question or a set of question types (i.e., how are multiple-choice questions marked) or even rules for the whole quiz (i.e., setting a time limit).

B. Carry on all Benefits of Polymorphism

This paragraph underlines a simple truth about SmartModels, but very powerful: it is built in the context of object-oriented technology, and therefore, it makes use of the notion of polymorphism.

More than that, this statute of SmartModels applies both at the level of concepts and atoms. This is one of the main reasons it can create families of entities. As a consequence, for both the semantic information and the structure of the entities of a domain it can easily:

- extend or refactor the model;
- reuse entities and their properties in order to describe more specialized entities in the same model or even in other closely related models (inheritance at the level of models).

As it has already been mentioned, a quiz may contain questions dealing with text and/or images. To keep things simple this paper considered only the case of questions with text and images, but a look in the second diagram one can observe that with a little change it can encapsulate meta-information of the different question concepts in different trees.

Figure 2 presents the *QuestionConcept*, which addresses the semantic information of a general text type question, but also its specialization *QuestionWithImagesConcept*, which concentrates on more specific characteristics dealing with image manipulation. This way, the instances of this concept (the corresponding questions with images atoms) will

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use semantic information about the text column of the question through inheritance.

PossibleImageTypes characteristic indicates the collection of accepted types of images which can be handled by the tool. Notice that the type of information that this characteristic uses is also an entity of the model: the basic atom *Image*. Again, this is the distinction that SmartModels makes between parameters and characteristics (Section I.A). For example, the parameter *MultipleAnswerCardinality* is of type integer with possible value between 1 and usually 4 (most of the quizzes have questions which include 4 choices, but it should not be limited); and parameter *ForceExactAnswer* of type boolean.

C. *Insert Dynamic Aspects: Actions*

What if after deploying the first version of the tool for developing on-line assessments there is a need to add a new mode of organizing the exam which was not planned at the model design stage, for example: “to enforce a time-limit”. Typically, a professor should be able to stipulate when he creates the quiz even if it was not included in the original model because it should be checked at run-time. In a classical object-oriented approach, it would lead to considerable changes in the structure of entities and in their behavior in order to implement this enhancement.

Thanks to the aspect-oriented approach proposed in SmartModels, it provides the opportunity to attach actions (*CheckTimeLimit*) to each concept to dynamically control the behavior of entities. This opportunity combined with the fact that it can benefit from inheritance, the actions increase the level of flexibility of the model: a professor may think either to set a time limit on individual questions if he likes or a global timer for the whole exam if he places the time-limit parameter at the level of the assessment question (Figure 2).

D. *Derived-Atoms Explosion*

Derived-Atoms are another mean provided by SmartModels to enrich the model and capture in the modeling phase as much as possible the commonalities and variabilities of the domain entities.

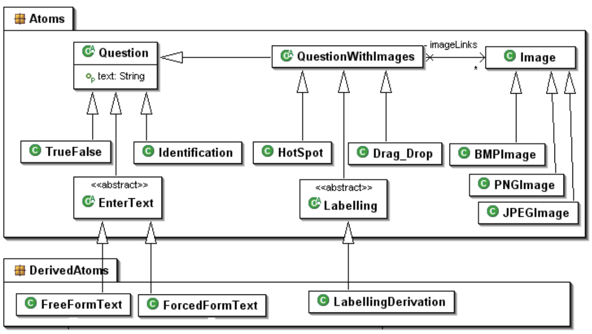

Figure 3. Atoms and Derived-Atoms

Returning to the above example, Figure 3 presents a couple of possible types of text questions and text and image questions. Now, the on-line assessment deploying tool can be equipped with more question types in two ways:

- either to create new atoms (creating new hierarchies of atoms in case of new entities form the domain or creating heirs of existing atoms to obtain specialized atoms through inheritance) or
- to derive new atoms from generic atoms in order to create new entities through a relevant combination of parameter and characteristic values. The number of combination possibilities is therefore limited only by the richness of the semantic information described through parameters, their type ranges and relevance in relation with other parameters from the same conceptual tree.

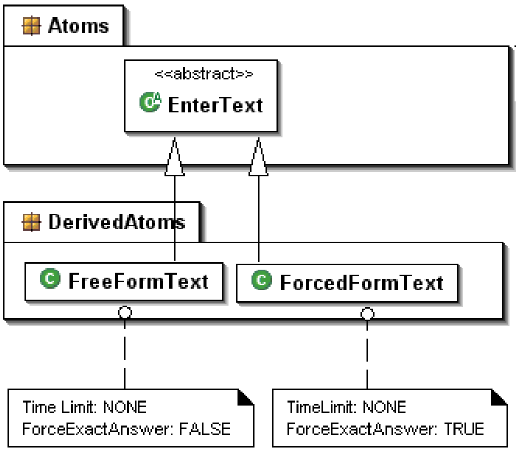

Figure 4. Edit-Text Derived Atoms

Now, the effects of setting the *ForceExactAnswer* parameter value to true or false can be seen when creating a new instance of *ForceFormText/ FreeFormText* derived-atom at run-time (Figure 4). Other variation of this edit-text type questions may have a time limitation, which can also be specialized to be an exact period of time or a fixed date and time value (a professor may not want to set a timer on the exam editing, but to set date and time limit until the assignment may have to be submitted).

Another interesting illustration would be to consider a labeling type of question when the student has to associate to a set of images a set of names or statements (Figure 5). A derived labeling atom can be obtained through a combination of values of its concept parameters. If *MultipleAnswerCardinality* is set to “4” then this means that there will be a question where there will be four choices to be presented to the student. A timer can be attached to this question and this tool can enumerate the types of supported images (*{JPEGImage, BMPImage, PNGImage}*).

Similar to the previous example, a professor can imagine other possible derivations. Adding a new parameter to specify the number of labels and the number of images to match, the user can obtain an even more flexible way to manage the creation of labeling questions. This way, a professor may create a question with ten labels from which to choose only the four valid names for the right column images.

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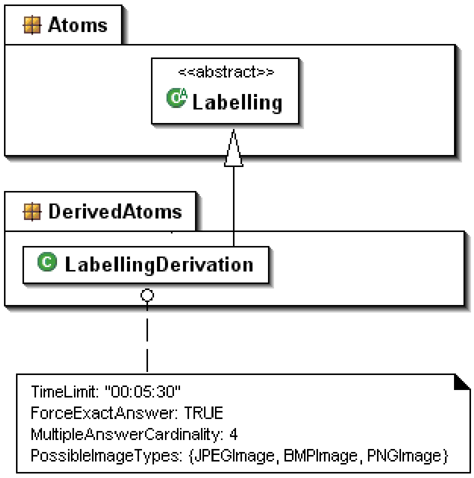

Figure 5. Labelling Derived-Atoms

Of course, the choice of the concept and/or atoms, as well as the associated parameters and characteristics may be discussed by an expert of the domain.

The aim of these examples is only to show the expressiveness of SmartModels approach to capture within a model as much as possible information which can then be automatically generated on a specific platform and mapped to an up-to-date technology.

IV. Conclusion and Future Work

It is very important to understand that in the world of software everything evolves: technologies, methodologies and applications.

It is believed that in order to provide an approach centered on models, which capture the know-how of a domain, it is of primary importance to ensure the independence between both the model and the software platform and between the model and the possible applications. This article promoted the idea that model-oriented programming is a better approach to solve these new challenges.

These ideas have been around for a couple of years, but today there is no major vendor which gets behind OMG's MDA initiative and makes it happen in software development. SmartModels' approach together with the SmartFactory prototype wants to form a possible and feasible way to apply these principles. It can happen in JetBrains or Eclipse and based on this last experience, paper [8] proposes a way to address meta-modeling issues extending their know-how. SmartModels is a MDA approach which provides a framework to create models that capture information about a business-domain independent from a technology, platform or programming language.

Because of the growing interest around educational technologies this paper experiments the SmartModels' approach for the description of various models and their applications of this domain. Therefore, it investigates the business model of a development and deploying tool for on-line learning solution and evaluation. The objective is to get feedbacks in order to improve the expressiveness of SmartModels – how to ease the job of a meta-programmer to describe a model, as well as a better automation.

Together with its prototype, SmartFactory, this paper presents a holistic solution to the issues raised. The objective is to validate the new approach. The experiment is made on the process of developing e-learning deployment tools for on-line assessment applications. It addresses the paradigm of how to practically implement MDA principles and rules for software engineering. Therefore, it deals with important implementation issues based on Eclipse Platform.

Thus, SmartModels inspired a solution which fills the gaps between the modeling solutions used by architects, software quality that engineers hope for, quantity of source-code that programmers have to write, and productivity targets that companies have to reach:

- an easy understandable and (friendly) usable approach for creating a coherent group of software artifacts for a domain (easy to encapsulate the know-how of a domain);
- flexible adaptation as a response to technological changes: a clear separation between the model and the technologies, but also a solid foundation to map on any software platform;
- a straightforward methodology to model and then to automate code generation for implementing and deploying family of software products;
- simple ways for prototyping as an extension of standard tools accepted on a large scale by the current research communities;
- an architecture designed for reuse integrating ideas from Domain Driven Development (DDD) [3], Aspect Oriented Programming (AOP) [4], Unified Modelling Language (UML) [11], Model Driven Architecture (MDA) [6], and generative programming [2].

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Electrical Adjustable Beds Testing Automation to Identify Mechanical and Electrical Defects

Mohamed M. El Rayes
Research and Development Dept.
ECTI Co.
6th of October City, Giza, Egypt
mohamed-rayes@hotmail.com

Tarek M. Tawfik
Walden University
Minneapolis, Minnesota, USA
tarek@mustemail.com

Tamer M. Nassef
Computer and Software Engineering Dept.
Misr University for Science and Technology
6th of October City, Giza, Egypt
tamer@ieee.org

***Abstract*— Designing and tuning a system to automate testing and identifying both mechanical and electrical defects appears to be conceptually intuitive, but can be hard in practice, if multiple objectives are to be achieved. This paper addresses the design of a system to detect the main design and industrial problems and to test reliability on an adjustable bed. The test can be processed with different types of data on multiple configurations that will allow tracing backward the exact actions, which the tester performed to the point, where the bug was discovered.**

Keywords—adjustable beds testing automation; electrical testing; mecanichal faults detection; Actuator testing.

I. INTRODUCTION

Traditional mechanical and electrical testing methods are performed by utilizing electrical and mechanical measuring devices and require the tester to be physically present to analyze the captured data [1]. Recently some studies especially in bioengineering field are using electrical sensors to build automated systems, these systems are used to increase computational analysis accuracy by processing the medical, biological, chemical, electrical, and materials information with both hardware and software sensory setup for their studies [2, 3] and other studies are using only numerical methods to test and evaluate the mechanical information of an artificial organs and implants with software based on biomechanical theories [4, 5]. Both of these methods can be merged to test and evaluate a realistic product based on electrical and mechanical theories.

The purpose of this study is to reduce the human factor in testing and design a tool to automate testing and to identify mechanical and electrical defects in the adjustable bed by analyzing the recorded data of the motor electrical measurements, where most of other methods are based on semi-automated tool that cannot ignore the human factor.

This tool allows to record and playback manual tests as the playback allows us to reproduce bugs. This test can be run with different types of data on multiple configurations and then we can trace backward the exact actions that the tester performed to the point where the problem was discovered. The tool can record also many test cases separately and rerun them according to the need to regression test a specific part or function of the mechanical system. This system combines software and hardware systems to provide a fully automated test. The testing system includes an automatic wireless control unit, voltage and current measurement for motors in the actuator and video/audio recording. The tool supports care and feeding of regression tests by the ability to automate a test and repeat it indefinitely according to the input change and automatically verify the test results and compare it to the expected results [6].

Test automation guarantees sustainability in the regression testing process which accelerates bugs and error detection. Mechanical problems appear as changes in the actuator motors current, so if we can measure the current passing in the motors with high speed, we can identify the locations of mechanical faults. Mechanical designers are interested in analyzing the dynamics of moving parts during the test and hear the noise coming out of them, which can indicate problems in the product. These operations must be synchronized precisely and do not take much of the computer resources. To automate the whole process, the wireless control unit must be under control of the software. A device was designed to simulate human presses on control unit buttons and that device can be configured through a USB interface and store that configuration during the test to minimize required computer resources. The following sections will give a quick description of the system and its specifications.

An adjustable bed as shown in Fig. 1 refers to the "superstructure" or underlying support base for a sleep surface. The mattress rests on the adjustable bed, just like it would on top of a foundation.

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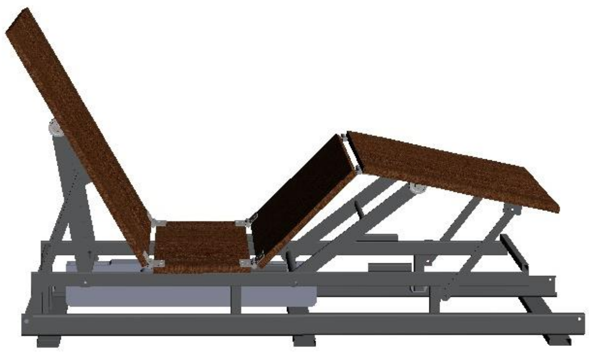

Figure 1. Adjustable Bed

Adjustable beds typically feature a lever system that allows for adjustable movement at the head and foot. The bed is segmented into three portions that move independently to accommodate individual sleeping preferences. The head & foot portions should be able to achieve an incline of 65 degrees to the head section and 45 degrees to the foot section. The Adjustable bed is equipment with a remote control with 6 buttons to adjust the required bed position as shown in the following figures.

Figure 2. Adjustable Bed Remote Control

Shown below is the mechanical functions related to the numbering of the remote control in Fig. 2.

1: Head Up
2: Head Down
3: Leg Up
4: Leg Down
5: Both Up
6: Both Down

II. MATERIALS AND METHODS

A. *SYSTEM HARDWARE*

The hardware consists of three modules, first ones is used for activating buttons of the wireless control unit with a pre-defined sequence and uses low current and no mechanical parts. The second is a current sensing circuit that produces a voltage proportional to the current flowing in the actuator. It also includes a voltage divider to enable voltage measuring, and the third module is a multi-channel A/D converter that has a Universal Serial Bus (USB) interface. It can take up to 100 readings per second for both current and voltage and angular position feedback, which is enough to detect any mechanical or electrical errors. All these modules are under control of a Personal Computer (PC) application that configures and stores all that data and synchronizes them with the video and audio recording. The software has functions to display the data and analyze them.

1) First Module: Automatic Remote Control Unit

This device consists of an MCU, which has an embedded USB transceiver, transistor switches to simulate buttons of the control unit and other components required for the circuit operation. It connects to the wireless control unit directly and the user can disconnect it and use the wireless control unit as an ordinary remote control.

The USB transceiver in the MCU is configured to operate at full speed (USB 2.0) and the connection is accomplished using HID USB protocol [7]; it provides easy and fast communication between the MCU and the PC. The software sends orders to the MCU, which always checks the received data while being connected to the PC. The MCU reads the configuration from the software and stores them in its EEPROM to avoid reconfiguration if the device is turned off. The configuration contains a sequence of button functions with different pressing time and delays between them and the number of cycles (number of repetition to that sequence). The MCU stores the number of completed cycles in its EEPROM every 20 cycles to prevent data loss in long tests if the power goes down.

TABLE I. COMMANDS USED TO CONTROL THE MCU OF THE FIRST MODULE

Command	Effect
Start	Starts the test sequence
Stop	Stops the test sequence
Read Sequence	Reads the saved sequence and the current step
Button Pressed Time	Sets the time of pressing a single button in manual control mode
Continue	Resumes the stopped sequence
Restart Current Sequence	Restarts the current sequence. It does not affect the cycle
Disconnect and Save	Disconnects the device from the PC and saves sequence to EEPROM
Set Cycles	Sets the cycles number
Reset All Operations	Restarts the sequence and resets the cycles counter.
Read Saved Cycle Counter	Read the cycles counter saved in the EEPROM
Press a Button	Press one of the six buttons in the remote control unit (manual control)

.

To simulate the six buttons on the control unit, we need six transistor switches [8]. The used transistors are MOSFET transistors which are turned on by the MCU. Bipolar transistors can be used, but MOSFET ones almost do not draw current form the MCU, so they save power. The idea is that transistor drain-to-source impedance is low when it is in the on mode and very high in the off mode, so it simulates the buttonsoperation sequence very well.

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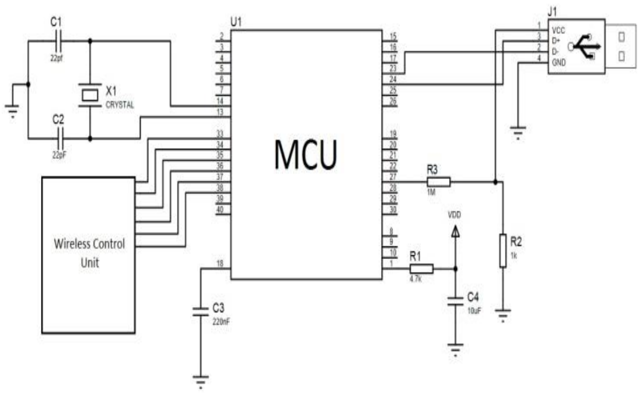

Figure 3. Circuit Diagram for the first module.

2) Second Module: Current and Voltage sensing

The main function of this module is to condition the analog signals to be measured by other digital device, by converting these signals to suitable forms.

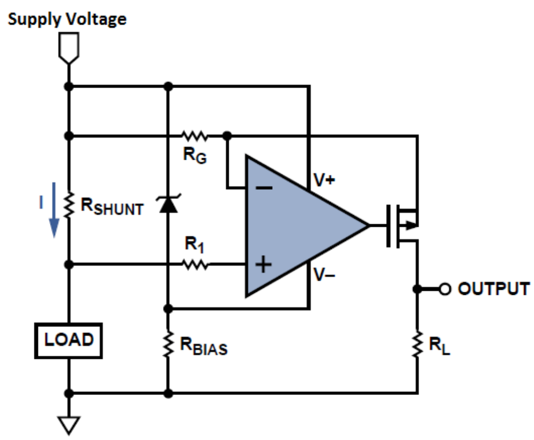

Figure 4. High-current sensing solution using an operational amplifier.

Current sensing circuit is simply a current to voltage converter. As A/D converters accept analog voltage only as an input, we have to convert the current to voltage with proper value.

The circuit topology is easy and familial [9]. The Op-Amp is rail-to-rail input/output type. The values of the resistors are selected to provide an output, which will not exceed 5V for our application that has a maximum continuous current of 3 amperes. This value will enable us to use the embedded A/D converter in the MCU. The Op-Amp is selected to have a fast response to make sure that the output will quickly settle at the right value. The current passes through the transistor and is controlled by the Op-Amp which is converted to a voltage value by a resistor.

Voltage sensing is easier than current sensing in general. In our application, the maximum allowed voltage is 30V, so the values of the voltage divider resistors are selected to provide a maximum of 5V at the output when the input is 30V. The resistors values should be high to minimize power dissipation in the resistors.

3) Third Module: High Speed Multichannel A/D Converter with USB interface

The third device consists of two low-cost MCUs operating at their maximum speed (48MHz). The first MCU is responsible for converting the analog signals to digital ones. For our application, we just need four A/D channels for our application, but there is one additional channel at the circuit for future use. The second MCU is responsible for USB communication. The device uses the USB HID protocol and it is optimized to send short reports at the maximum available speed.

The first two input channels are used for current and voltage sensing. The other two input channels are used for angular position feedback. It is critical to monitor the angular position of head and foot part of the bedduring the test to determine the effect of each test profile on the specified angle. The job is simply done by using two low cost potentiometers fixed on the axes of the moving parts. The outputs of these potentiometers are connected to the channels three and four of the A/D converter.

The communication between the two MCUs is done through a non-standard parallel connection optimized to guarantee maximum data transfer rate.

This configuration allows fast and reliable measuring of the current and voltage values.

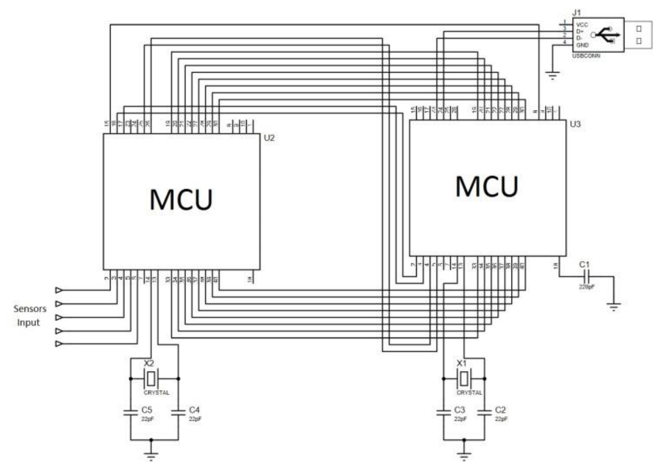

Figure 5. Circuit Diagram for the third module PCB.

B. SYSTEM SOFTWARE

The testing software contains all the features we need for various tests in one package. The program was written using C# programming language.The first section is the automatic remote control with its configuration, controls and readings of the stored data. The table in the middle of the user interface window is used to enter the required sequence of buttons. The same table is used for reading the stored sequence in the device and indicating the current step. The buttons on the left, as shown in Fig. 6, are used to manually press a button on the wireless control unit through the software. There are other fields to configure the button pressing time and the required number of cycles and reading the current number of cycles.

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The second section contains the controls of the available tests. The tests are: running the testing sequence only, current and voltage measuring and video and audio recording. The user can choose one or more of these tests and click on the start button and program will send orders to all required devices using USB HID protocol [10] (except the camera and the microphone) and return error message if any device is not connected or has a problem. The other control buttons are stop, resume, restart or reset the test. The outputs of the program are stored into an AVI extension for video and audio and TXT files for current and voltage measurements.

The challenging and most important things in this section are timing, synchronization and computer processor utilization. The application uses the multi-threading concept to provide fast and efficient processor power utilization. The software was carefully designed to make sure that the tests are running simultaneously and their data are recorded with precise timing. Tests data are saved to output files every 10 minutes to allow easy review.

An especial test that will help in the review process is the live measuring test. It gives order to the required button and measures the current in the actuator while graphing the received data on the screen. This can help the testing engineer to see the motion of the tested part and the current of the motor at the same time, so detecting mechanical errors may become easier with this tool.

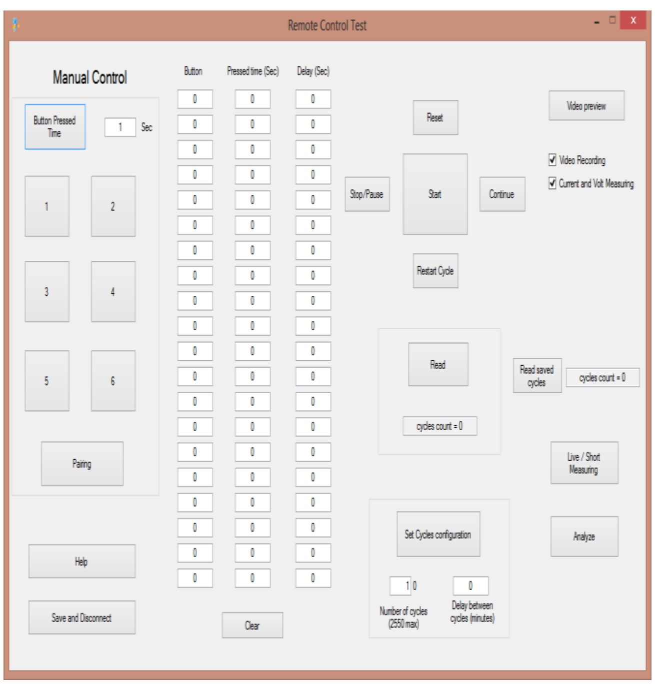

Figure 6. GUI of the computer application.

The program contains a special section for reviewing the tests data. The program reads the output current and voltage value and graphs them with high resolution, which can be adjusted to show more details if required. It detects the beginning and ending of every actuator stroke and calculates the time elapsed during each stroke and shows it in the graph. It can calculate the average of current or voltage values during a specific time.

III. Results

Examples of mechanical and electrical problems that can be detected are:

- Mechanical Misalignment of assembled parts.
- Linear guides Surfaces defects.
- Actuator Gear reduction system mechanical defects.
- Actuator motor Armature-related anomalies which can include commutator bar defects, riser defects and shorted turns or coils in the armature circuit

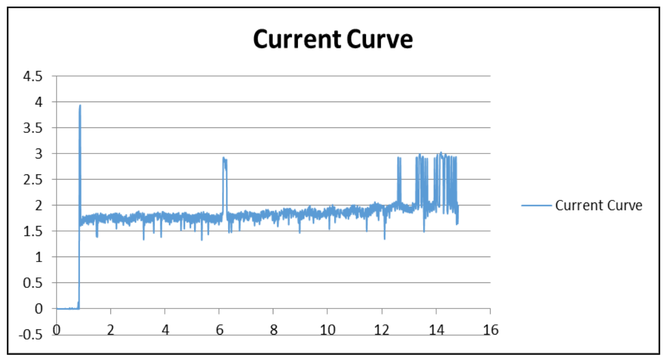

Figure 7. Current Curve Diagram

The above diagram shows the DC Current versus time curve concerning the Actuator motor measurements to the leg movement section of the adjustable bed, the bed leg bend to 45 degrees over an average time of 15 seconds.

By visually analyzing the curve pattern, we can diagnose the mechanical and electrical defects to the adjustable bed shown in Fig. 7 as follows:

a) at t=1 the curve spike to near 4 A which is considered a normal behavior to the peak in rush/peak starting current of a brushless DC motor which typically will be a function of circuit resistance and power resource.

b) The ascending pattern of the curve indicates the load variations with regard to the angular position of the adjustable bed foot section which is also considered a normal behavior.

c) At t=6 there is a sudden spike in the current consumption which is an unusual behavior indicating a mechanical defect. By mechanically analyzing the defect at the given point of time we found that this was caused by a defect in surface geometry in the linear actuator bracket.

d) From t= 12.5 until the end of the cycle there is a disturbed pattern in the curve, we found that there is a problem in the linear guide which gives the bed the wall hugger capability which slides the bed toward the wall, leaving the bed close to nightstand.

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IV. CONCLUSION

The use of data-driven automation approaches and frameworks further increases testing efficiency and can underpin effective configuration testing [11]. This tool is useful in various mechanical systems that uses a DC motor actuator as it support automated testing which serve in estimating product reliability, analyze current and voltage curves to diagnose mechanical and electrical defects, perform aging tests on different operational profiles, perform regression testing by reproducing defects at any given time or mechanical position, perform usability and load testing. Test automation can accelerate the testing cycle and promote product quality. Testing automation gave us a guarantee that each test case is always executed in the same way and in the same environment. When executing a test case manually, some steps could be missed, or intentionally omitted. This can result in not only missing some defects, but also can lead to reporting invalid defect which can result in not only missing some defects, but also can lead to reporting invalid defect.

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A Robust Wind Estimation Scheme for Simple Tethersonde

Hong-Soon Nam and Sang-Sung Choi
WPAN Application Section
ETRI
Daejeon, Korea
hsnam,sschoi@etri.re.kr

Dae-Young Kim
Department of InfoComm. Eng.
Chungnam National University
Daejeon, Korea
dykim@cnu.ac.kr

Seung-Gu Yang
R&D Center
Jinyang Industrial Co., Ltd.
Ansung, Korea
hl3eezy@jinyangind.com

***Abstract*—This paper presents a robust wind estimation scheme for a simple tethersonde system, which is to use a commercially available radiosonde for recalibration and recovery. A radiosonde system determines wind speed and direction by keeping track of the radiosonde's position as it rises. However, the method used in the radiosonde system is not suitable for the tethersonde system due to tethering to the ground. Hence, the proposed scheme introduces a drag force caused by the movement of the tethersonde to estimate the wind speed. This paper also evaluates the scheme through simulation in terms of wind speed estimation. Simulation results show that a radiosonde can be utilized as a tethersonde for the low and middle atmospheric observation without modification.**

Keywords-tethersonde; atmosphere; wind speed; anemometer; radiosonde;

I. INTRODUCTION

A radiosonde is used in measuring vertical profiles of atmospheric temperature, humidity, pressure, wind speed and wind direction as a balloon ascends through the atmosphere. The balloon typically rises up to an altitude of 35 km from the ground at a rate of a few meters per second. As the balloon ascends, it expands and eventually bursts, and then the radiosonde falls to the ground. Therefore, a radiosonde is not desirable for repeated calibration checks and repeated measurement because the system is discarded after once use [1], [2], [3].

On the other hand, a tethersonde is attached to the ground so that the limitations of a radiosonde can be mitigated by tethering to the ground. A tethersonde, in general, is to measure near the surface atmosphere up to an altitude of 1500 m and provides repeated calibration and measurement by adjusting the length of the tether line. Thus, detailed vertical profiles of meteorological atmosphere in the boundary layer can be obtained [4], [5]. A radiosonde system determines wind speed and direction by keeping track of its position through the use of global positioning system (GPS) receiver [6], [7]. However, unlike other parameters, the wind measurement method used in a radiosonde system is not appropriate for a tethersonde system because the tethersonde cannot move along the wind due to the tethering.

For this reason, this paper presents a simple and robust wind estimation scheme for a tethersonde system. The scheme firstly computes elevation and azimuth angles from the position of the tethersonde. Then, the wind speed is determined based on the drag force and the elevation angle calculated from the force balance principle. The rest of this paper is organized as follows. In Section II, we describe system overview. Next, the wind speed and direction estimation algorithm is described in Section III. After that we analyze the proposed scheme in terms of wind speed estimation by simulation in Section IV. Finally, we conclude this paper in Section V.

II. SYSTEM OVERVIEW

A tethersonde system consists of a tethersonde, a balloon and tether line, and a ground receiver system, as shown in Fig. 1. The scheme is to utilize a radiosonde as a tethersonde with or without modification. The tethersonde is equipped with sensors that measure temperature, air pressure, humidity, and GPS receiver to find its location. It periodically measures atmospheric parameters and transmits the parameters to the ground system in the 400 to 406 MHz meteorological band. As the tethersonde ascends, it reports in real time the temperature, humidity, pressure, and its position every second.

The ground system receives the data transmitted from the tethersonde and processes them in a personal computer (PC). Then, the PC estimates wind speed and direction by keeping track of the position of the tethersonde. The balloon and tether line are to suspend the tethersonde and to control its height by winding and unwinding the line. After observation, the tethersonde and balloon are recovered by winding the line.

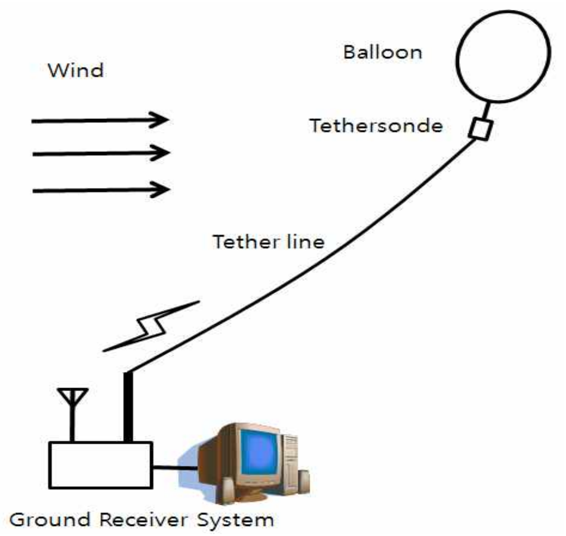


Figure 1. Tethersonde system consisted of a tethersonde, a ground receiver system, and a balloon and tether line

III. Wind Speed And Direction Estimation Algorithm

To analyze the movement of a tethersonde caused by wind, a spherical coordinate system is considered. The coordinate system consists of the radius r, the elevation angle θ, and the azimuth angle φ, as shown in Fig. 2. In the figure, the tethersonde is released at the origin point at time 0, $p(x,y,z)_0$, and after a time t the tethersonde is at a position, $p(x,y,z)_t$.

Once a tethersonde is launched, it travels horizontally at the same speed of the wind while the tether line is unwinding. When the tether line is released up to its maximum length, the tethersonde travels on the surface of the sphere whose radius is equal to the length of the line. Thus, the movement of a tethersonde can be classified into two stages, releasing stage and tethering stage.

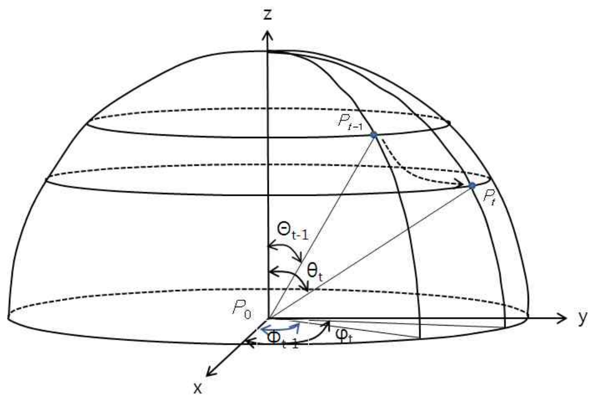


Figure 2. Spherical Coordinate system (r, θ, φ), where r, θ, and φ represent radial distance, elevation angle, and azimuth angle , respectively.

A. Releasing Stage Analysis

Once a tethersonde is released into the atmosphere, the sonde is to rise by a lift force. The lift force consists of buoyant force and drag force, F_b and F_d. Consider that F_b includes the gravity force mg of the balloon, tethersonde and tether line. Then, the buoyant and gravity forces are calculated from approximately simultaneous tethersonde soundings.

Similarly, the drag force is equal and opposite direction of the buoyant force. Since $F_d = -F_b$, the drag force can be determined by evaluating the buoyant force, which is readily calculated from the available data. The average density of the air is obtained from approximately simultaneous radiosonde soundings, and the cross sectional balloon is known. Then, the average lift force and lift speed of the balloon are defined as,

$$F_b = \frac{1}{2}\rho C_d A V_b^{\,2}, \quad (1)$$

$$V_b = (2F_b / \rho C_d A)^{1/2}, \quad (2)$$

where ρ is the density of air, V_b is the lift speed caused by the buoyant force, C_d is the drag coefficient, and A is the cross sectional area of the balloon.

Let V_l be the average lift speed over the measurement period T, which denotes the difference between time t and t-1. Then, V_l is given by the equation,

$$V_l = \left\| p(z)_t - p(z)_{t-1} \right\| / T. \quad (3)$$

According to (2) and (3), it is clear that the wind has an ascending air current if V_l is greater than V_b and a descending air current if $V_l(t)$ is less than V_b .

On the other hand, when the tethersonde is moved from $p(x,y,z)_{t-1}$ to $p(x,y,z)_t$, the horizontal wind speed and direction, V_w and φ_w can be expressed as,

$$V_w = \sqrt{(x_t - x_{t-1})^2 + (y_t - y_{t-1})^2}\,/T, \quad (4)$$

$$\varphi_w = \arctan[(y_t - y_{t-1})/(x_t - x_{t-1})]. \quad (5)$$

The values of $p(x,y,z)_t$ are known since they measured every second. With the information, the horizontal wind speed and direction can be estimated by (4) and (5).

B. Tethering Stage Analysis

When the tether line is released up to the maximum length, the tethersonde travels on the surface of the sphere whose radius equal to the maximum length of the line. Fig. 3 represents the relation among buoyant, wind, and tension forces in the tethering stage, where the tethersonde stays at a point or moves on the surface of the sphere. The tethering stage can also be divided into two states, steady state and dynamic state, depending on the position changes of the tethersonde.

Fig. 3 (a) shows the relation of the string tension F_T to wind force and buoyant force in the steady state, which is represented by $F_T^{\,2} = F_b^{\,2} + F_w^{\,2}$. In the steady state, the wind speed V_w can be determined from the buoyant force and the elevation angle. From the relation, the wind force and speed can be calculated by

$$F_w = F_b \tan\theta, \quad (6)$$

$$V_w = (2F_w / \rho C_d A)^{1/2}. \quad (7)$$

The cross sectional areas in (2) and (7) may be different depending on the directions, vertical side and horizontal side. On the other hand, in the dynamic state, the balloon is being moved from $p(x,y,z)_{t-1}$ to $p(x,y,z)_t$ during the measurement period, as shown in Fig. 3(b). When the tethersonde moves through the air, a drag force acts opposite direction to the movement of the tethersonde. The relation between acceleration and net force is given by Newton's second law of motion,

$$\sum \vec{F} = \vec{F}_b + \vec{F}_w + \vec{F}_T + \vec{F}_d = m\vec{a}. \quad (8)$$

Let $\vec{V}_s$ and $\vec{F}_d$ be the velocity and drag force of the tethersonde on the surface of the sphere, respectively. In this paper, the impact of tether line was ignored for simplicity. Assuming that azimuth angle is constant, $\vec{V}_s$ and $\vec{F}_d$ can be written as

$$\vec{V}_S = R\dot{\theta}a_\theta \,, \tag{9}$$

$$\vec{F}_d = \frac{1}{2}\rho C_d A V_s^{\,2} . \tag{10}$$

Let m and $\vec{a}$ denote the mass and acceleration of the tethersonde, respectively. The acceleration of a tethersonde may be expressed in the spherical coordinate system by taking into account the associated variation in the unit vectors,

$$ma_r = m(\ddot{r} - r\dot{\theta}^2) = F_b \cos\theta + F_w \sin\theta - F_T \,, \tag{11}$$

$$ma_\theta = m(r\ddot{\theta} + 2\dot{r}\dot{\theta}) = -F_b \sin\theta + F_w \cos\theta - F_d \,. \tag{12}$$

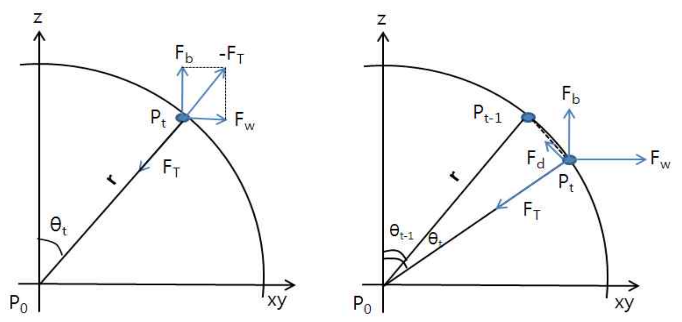
Where, $\dot{x}$ and $\ddot{x}$ denotes the first and second derivative of x. The wind speed may then be estimated by (8) or (12). Equation (8) can be modified to

$$\vec{F}_w = m\vec{a} - \vec{F}_d - \vec{F}_b - \vec{F}_T \,, \tag{13}$$

$$V_w = (2F_w / \rho C_d A)^{1/2} \,, \tag{14}$$

$$\varphi_w = \arctan(\frac{y}{x}) . \tag{15}$$

When a tethersonde stays at a position, $m\vec{a} = \vec{F}_d = 0$, equation (14) is equivalent to (7). Therefore, the wind speed and direction can be determined by (14) and (15), respectively.



(a) Steady state *(b) Dynamic state*

Figure 3. Relationship among lift, wind, and tension force in the tethering stage, where a tethersonde moves on the surface of the sphere.

C. Wind Estimation Algorithm

From the analysis, the wind speed and direction can be determined in the tethersonde system, as shown in Fig. 4. In the system, the GPS receiver in the tethersonde periodically reports its longitude, latitude, altitude, and wind speed. The ground receiver system then calculates the distance R between the origin point and current point from the received data. Depending on R, the algorithm goes into either the releasing stage or tethering stage. When the variation of wind speed is small enough, the algorithm may perform the wind estimation method of the steady state. On the other hand, although the algorithm may cause some error whenever the stages are changed, the error can be recovered in the steady state.

-- Wind speed and direction estimation algorithm from position tracking of a tethersonde

-- V_W : horizontal wind speed

-- Φ_W: wind direction

-- $P(x,y,z)_t$: tethersonde's position at time t

$$R_t = \sqrt{(x_t - x_0)^2 + (y_t - y_0)^2 + (z_t - z_0)^2}$$

If ($R_t = L_{max}$ and $R_{t-1} = L_{max}$) -- tethering stage

$$\theta = \arctan(\sqrt{x^2 + y^2} / z)$$

$$\vec{V}_S = R\dot{\theta}a_\theta$$

$$F_d = \frac{1}{2}\rho C_d A V_s^{\,2}$$

$$\vec{F}_w = m\vec{a} - \vec{F}_d - \vec{F}_b - \vec{F}_T$$

$$\varphi_w = \arctan(y / x)$$

Else -- releasing stage

$$V_w = \sqrt{(x_t - x_{t-1})^2 + (y_t - y_{t-1})^2} / T$$

$$\varphi_w = \arctan[(y_t - y_{t-1})/(x_t - x_{t-1})]$$

End;

Figure 4. Proposed wind estimation algorithm

IV. SIMULATION RESULTS

To verify the proposed scheme, we considered simulation parameters similar to a radiosonde as described in Table I. From the parameters, we can obtain buoyant force, lift speed caused by buoyant force and the radius of the sphere.

Fig. 5 shows the wind speed versus flight time and the trace of tethersonde in a spherical coordinate system. It was tested with the actual measurement data taken by a radiosonde system in a fine day at a location of longitude 127 °(E) and latitude 37°(N). The measurement data include measurement time, latitude, longitude, altitude, and wind speed. The distance from origin to the tethersonde, elevation angle, azimuth angle, and lift speed variation are calculated from the measurement data. In Fig. 5(b), until flight time 270 seconds, the position trace is plotted from the measurement data. After then, the position trace is plotted from both the distance and altitude that are calculated from the measured wind speed.

TABLE I. SIMULATION PARAMETERS

Parameters	Value	Description
Vol	7000 liters	Volume of balloon
ρ_{helum}	0.178 g/e	Density of helium gas
ρ_{air}	1.293 g/e	Density of air
m_{sonde}	370g	Mass of the balloon and radiosonde
$A_{balloon}$	$\pi(\frac{3}{4\pi}Vol)^{2/3}$	Sectional Area
L_{max}	1.5 km	Maximum length of tether line

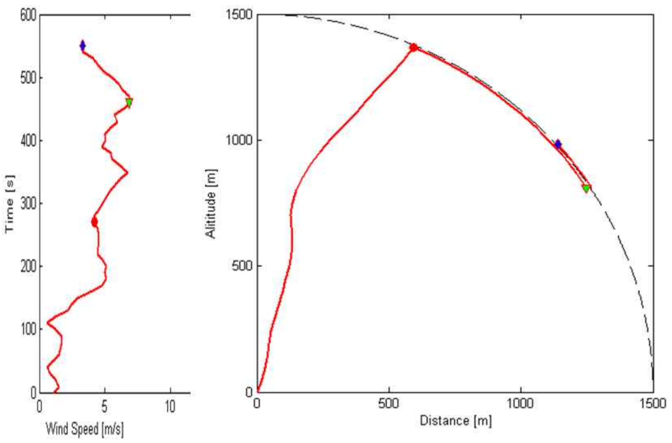
Fig. 5 (a) and (b) have three markers, red circle, green triangle, and blue diamond. The same markers in each plot correspond to each other. Table II represents the detailed values of the three points. For example, the red circle indicates 270 seconds of flight time, 611 m of horizontal distance, 1370 m of altitude, and 4.2 m/s of wind speed. After 270 seconds, the tethersonde moves on the surface of the sphere according to the wind speed and direction. For example, the green triangle shows that the wind speed is increased up to 7.0 m/s at time 469 seconds.

The figure shows that the tethersonde moves up and down along the surface of the sphere with the wind force.

TABLE II. SIMULATION RESULTS AT THREE POINTS

Marker	Time [s]	Distance [m]	Altitude [m]	Elevation angle[deg]	Wind speed [m/s]
circle	270	611	1370	24	4.2
triangle	469	1284	775	58	7.0
diamond	550	1134	982	49	3.3

Accordingly, the radius of the sphere is adjustable by changing the length of the tether line. In a similar manner, the scheme provides more detailed vertical profiles in the low and middle atmosphere by repeatedly winding and unwinding the tether line.



(a) Wind speed vs. flight time (b) Position trace

Figure 5. Wind speed vs flight time and position trace of tethersonde on the sphere.

V. CONCLUSIONS

In this paper, a robust wind estimation scheme for a simple tethersonde system has been presented, which utilizes a commercial radiosonde as a tethersonde for the low and middle atmospheric observation. A radiosonde is not desirable for recalibration and recovery due to discarding after use. To mitigate the limitations, the proposed scheme is to use the radiosonde by tethering to the ground for ascending and descending. However, although the atmospheric measurement techniques in the radiosonde are useful, the method for wind speed and direction measurement is not suitable for the tethersonde system.

The proposed scheme determines wind speed and direction by adopting a drag force in the dynamic state when the tethersonde moves on the surface of the sphere. Simulation results show that the scheme can be used in a simple tethersonde system for wind speed and direction measurement without additional wind sensor. The scheme also provides more detailed vertical meteorological profiles of the low and middle atmosphere by winding and unwinding the tether line. In addition, the observation altitude is extendable more than 1500 m just by increasing of the length of the tether line.

Further studies are needed to analyze the impact of tether line on wind speed estimation and verify the scheme under various environment conditions.

ACKNOWLEDGMENT

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Mobile Agent Synchronizer for a Real-time Architecture

Ali Esserhir and Fabrice Mourlin
Algorithmic, Complexity and Logic Laboratory
Paris-Est University (U-PEC)
Créteil, France
{ali.esserhir, fabrice.mourlin}@u-pec.fr

***Abstract*—In this paper, we use mobile agents in order to synchronize real-time distributed system. We present a way to exchange mobile agents from an agent server to agent hosts. Imported agents are used to apply actions in their host context. We solve security problems by a negotiation step between the host and the server. Locally, we use a specific thread to execute the imported agent. Its mission uses time vector for event synchronization. When the mission of the mobile agent is ended, it goes back to the server and waits for a future demand. We apply this strategy to implement mobile agents, which import the time of the server, and synchronize the actions between source and target hosts and keeps the real-time properties.**

Keywords-agents; mobile code; real-time system; task synchronization

I. INTRODUCTION

Mobile agents provide a new approach of distributed systems. They give a solution for moving part of code from an agent host to another one where the data are installed. If developers have the habit to transfer data from a software layer to another one, this technique has limits. When the size of data exceeds a limit, the exchange cannot be possible. Again, security reasons can filter data transfers. It depends on the semantics of the data: results of a company, hybrid mesh for numerical simulation, etc.

Many works are about the subject of mobile agents. Java Agent DEvelopment Framework (JADE) toolkit [1] is one of the most famous solutions for the development of software agents. Jade is a middleware that facilitates the development of Agent Management System (AMS). The need of mobility becomes greater in network management and other platforms exist, such as Tracy toolkit [2] or Mobile Computation Architecture (MCA) platform [3]. They allow moving agents from one node to another node of a network. MCA is designed for building a fault tolerance layer into a numerical application. When a distributed application is deployed over a set of processors, events can occur during the runtime. This set of computing resources is not fixed, the new processors can be free and few nodes can fail. Also, the execution should continue despite these events. MCA allows moving a part of the execution if the new computing resource appears or disappears. The mobile agent can be considered as a vehicle which leads a piece of code towards a computing resource. This approach can be done because the parts of the application are loosely coupled. The exchanges of data are few and predictive.

When the exchanges are based on time synchronization, this means that a time control algorithm is applied to a business application, such as a distributed numerical simulation. The main constraint of a real-time application is to respond to an event at a scheduled time. We did not find an existing work which groups mobile agent and real-time. For example, this requirement occurs when specific events happen, such: hardware interrupts, local clock, communication, etc. The traditional operating systems cannot guarantee that the latency is below a certain threshold. In fact, during some very resource-intensive operations (such as inputs / outputs), these systems can be temporarily blocked. This leads to the use of real-time operating systems, such as QNX [4], which ensures latency known by its micro-kernel structure. Deployment of applications on to such operating system allows users to exploit real-time features.

Our work is about distributed architecture and the use of mobile agents for synchronizing tasks in real-time context. Section 2 is about the requirements of our work. Section 3, we present our approach of mobile agent under the development constraints. In Section 4, we describe our algorithm and our implementation. Section 5 is about our measures and results. Finally, we review the main features of our contribution and the future directions.

II. REQUIREMENTS AND CONSTRAINTS

A. Operating system and libraries

A real-time system is a combination software / hardware where the software allows adequate management of material resources to complete certain tasks or functions in very specific time limits. Real-time applications are often embedded applications. Constraints of standard software, add the notions of response time, latency, clock, timing tasks,f etc. We use an operating system with a real-time kernel. A real-time kernel is the minimal implementation to make real-time scheduler, task management, and inter-task communication.

There are differences between a true and a real-time system. The first kind of system has completed a real-time kernel with modules and libraries to facilitate the design of real-time application: memory management, management of input/output management, timer management, network access, file management. A real-time operating system is the special case where the host system communicates with the target system. So, here we have chosen a development environment natively based on RT-Linux [5].

B. Programming languages

Our experience about mobile agent development has shown that object interpreted languages are useful for exchanging between two parts of code: a requestor and a server. Many programming languages exist, which are interpreted with object paradigm. We have already developed projects based on mobile agent architecture. Our approach has been validated in the distinct domain, such: web server monitoring, numerical analysis, business process management, etc. We have chosen Java language because it possesses useful features such: serialization, garbage collection, class loading, network, Just-in-time compilation, and thread scheduling. But, traditional implementations are incapable of running applications with real-time behavior.

Fortunately, real-time extensions to Java technology, based on the real-time specification for Java (RTSJ) [6], enable JVMs with real-time features. The RTSJ provides an API is enabling real-time scheduling, advanced memory management, high-resolution timers, asynchronous event handling, and asynchronous interruption of threads. Of course, standard Java applications can run without modification in a real-time JVM, but some APIs ,such as the threads and timers APIs, APIs are enriched. Also, when developers would like to create a new project with the real-time Java development kit [7], they can set their own configuration.

Because mobile agents move from a Java virtual machine towards another one, network protocols are necessary. Default implementation (provided by Sun, Oracle) of RTSJ does not provide any solution. Research works exist about real-time Remote Method Invocation (RMI) framework, but the libraries are not maintained or are not available. For instance, the RT-RMI framework supports timely invocation of remote objects. The thread classes defined by the RTSJ are used to provide the client and server threading mechanisms in the invocation process [8]. But, this framework is not available for four years

Also, we have selected a Jamaica virtual machine [9] because it provides a complete implementation of the RMI protocol. This software product is frequently updated, its roadmap is public and reference documentation is available. The use of RMI protocol means a rich protocol where object can be serialized on an RMI socket. Also, this feature allows developers to separate the concerns: agent server, agent host and mobile agent.

In the next section, we give details about our mobile agent design and implementation with this restricted development context.

III. MOBILE AGENT IN AN RT CONTEXT

The software architecture of a distributed mobile system is based on two main strategies. In the first strategy, agent hosts provide a remote service for reaching them. Next, an agent server configures a mobile agent to visit a set of hosts with these remote stubs. Then, the mobile agent starts its visits by using the remote services. By the end of its mission, it comes back to the server through a call to a remote service on the server.

A second strategy consists in the creation of mobile agents which have a remote call. Then, the agent hosts can send a demand to a specific mobile agent if is needed. Because, we believe that a mobile agent has to be autonomous, we have selected the first approach. Thus, the agent host is passive until the agent decides to reach it. Then, the host will allow the mobile agent to realize its mission in the context of the agent host.

A. Necessary conditions

Whatever the strategy is, it is necessary to have technical skills. The migration of agent can be realized onto the network with the following concerns:

- Common execution in virtual machine on all the nodes,
- Process persistence: saving and spawning
- Communication mechanism between agent hosts
- Security to protect mobile agents and agent hosts

In a heterogeneous environment, many different system architectures can be connected. An interpreted language that is capable of executing machine code solves the problem of a common execution language. A Java virtual machine answers to that constraint even if the versions are not the same. The use of timers is a first need to settle task management. Existing *Timer* class in standard JDK is not powerful enough for clock synchronization. RTSJ provides new timer with a richer API.

For agents to migrate to remote machines, they must be capable of saving their execution state, or spawning a new process whose execution state will be saved. This persistently means the conversion of the object's state into a data form suitable for transmission over a network. Mobile agents should not have to be responsible for achieving this themselves. Again, Java language takes into account the serialization concern and of course the deserialization.

Some communication mechanisms exist to transfer mobile agents across networks. A mobile agent might be transferred using TCP/IP, or by using a higher level of communication such as RMI, Internet Inter-ORB Protocol (IIOP), Simple Mail Transfer Protocol (SMTP) or even HyperText Transfer Protocol (HTTP). RMI is a sub part of Java language since its first version. This is also true for the real-time implementation.

Security is critical when mobile agent is transferred across a network. Depending on the mission of the agent, it could write local resources of the agent host. Mobile agents themselves need protection against hostile hosts. Java runtime forces the creation of a security manager which observes and checks all the actions of incoming agents. So, by default, a mobile agent has no permission at all and when an agent leaves for a new host, extreme care must be taken to prevent unauthorized modification or analysis of the agent. This can be done during the code loading of mobile agents.

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Under this set of constraints, we propose into the next section an object approach of RT mobile code. We stress the importance of the state of the agent which ensures that a mission starts with a first agent host and continues on the next one and so on.

B. *Object description*

1) Architecture descriptions.

Our approach is based on the use of the object model. First, we provide a high level deployment diagram, where main nodes are presented. The Server node plays the role of the agent factory. It supports component called Agent Server in the following section. The Host nodes play the role of agent client, which waits for the visit of mobile agents. Figure 1 gives a picture where the links support RMI protocol and nodes support Jamaica virtual machine.

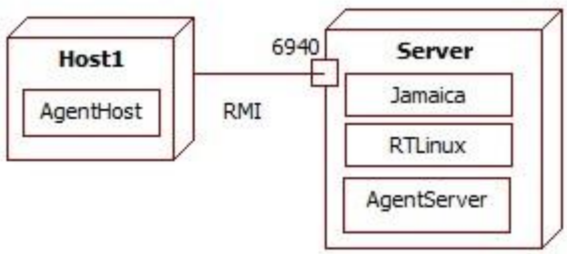

Figure 1. deployment diagram for a minimal network

This hardware architecture supports several components. For our design, all Host nodes have the same deployed components called *AgentHost*. This choice can evolve whether an agent host filters mobile agents through its type. The server node receives an agent server at deployment step. It will create mobile agents as demand. It configures them by injection of remote accesses and mission description. Then, mobile agents start and are autonomous during their whole activity. Figure 2 shows the dependencies of a mobile agent component. It depends on the provided interfaces of all the agent host components and also the remote interface of the agent server for the end of its mission.

The mobile agent provides two local interfaces. At first one, called *IAgent* is used by the agent server to configure locally the agent. Another one is called *IMobileAgent,* which is called by the agent host to start or to suspend the activity of the mobile agent locally to the host. This software architecture leads to the creation of three artifacts which will be assigned to the nodes of the Figure 1. Globally, all nodes which wish mobile agents needs to support an *AgentHost* component.

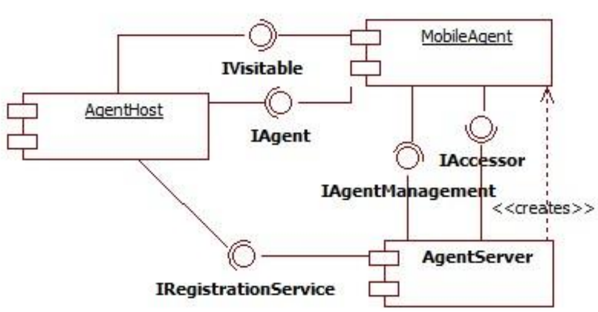

Figure 2. component diagram with provided interfaces

2) Class structure.

a) Mobile agent component structure.

Minimal structure of *the Mobile Agent component* contains a class which implements two main interfaces (*IMobileAgent* and *IAgent*) (Figure 3). This class has a behavior called *configure* method: it allows an agent server to provide remote stubs about the hosts to visit, a set of properties and a task. The task describes the activity of a mobile agent. This can be a data collection. For instance, at the end of a distributed application, a mobile collector can gather the partial results on each node. Each new task type is defined by a subset of classes. The main one implements *Task* interface. All tasks are placed on the agent server in the workspace directory. The properties are useful to provide a specific working context. So, the description of the features of agent *host1* is set into a file called *host1.properties* on the agent server. In the configuration step, the URL of this input stream is provided by the server. Next, the mobile agent extracts the data set into a *Properties* object. These data will be used during the activity of the mobile agent.

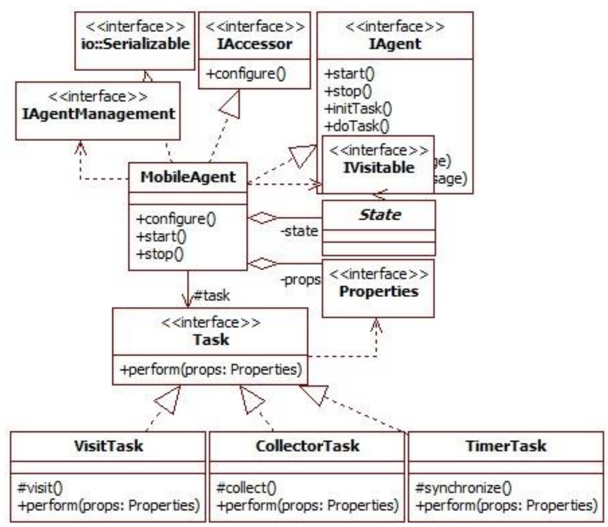

Figure 3. minimal class diagram of *MobileAgent* component

A mobile agent is composed with a *State*instance, which manages local data during its activity. This state is a data structure which is encoded during a transfer from on the agent host to another one or the agent server.

b) Agent host component structure.

The *AgentHost* component plays the role of the requestor. Its interaction sequence of the demand is as follows. First, the agent hosts are registered in the locate registry of the agent server. Then, it configures one or more mobile agent with the remote stubs. Next, the mobile agents use the remote access to move onto the agent hosts through a serialization step. It invokes a remote call through RMI protocol. This is why the *IHostService* interface is a generalization of *Remote* interface. In Figure 4, the implementation, called *AgentHost* is an extension of a technical RMI class. It generates automatically a remote stub for the publication into the server registry. So, it exposes a remote interface called *IHostService*, for mobile agents.

When the mobile agent is received by the agent host, it performs a check of the byte code and permissions useful to execute its activity. Then, it loads the code of the mobile agent and assigns needed permissions. It launches its mission. By the end of its job, the host invokes the *stop method* which triggers the continuation of the mission onto another agent host.

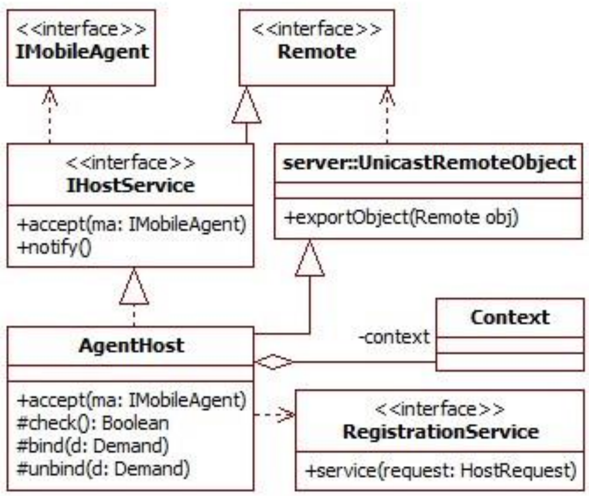

Figure 4. class diagram of *AgentHost* component

The runtime context of a mobile agent is restricted by the agent host itself. It does not run into another virtual machine. So the security is managed by an access controller, which is created by the host. This one is also responsible of the permissions which are defined locally to the host with a code base and a signature. They are not downloaded from the agent server. Each agent host has a context where the activities of the mobile agents are interpreted. So, after the execution of a mobile agent, the agent host suppresses its demand from the registry of the agent server.

c) Agent server component structure.

The agent server is a clever agent factory. It exposes two remote interfaces, one for the agent hosts and another for the mobile agents. The first interface (*Registration Service*) is used to collect or remove the demands of agent hosts. They are stored in a local registry.

The second interface (*Server Service*) is used by mobile agents in two situations; (i) on the one hand, when their mission is

completed and they return to their starting point, and on the other hand, when a mobile agent needs additional resources, it can ask the server that created it.

The interaction sequence for a mobile agent creation is as follows. First, a mobile agent is instantiated by the use of an extension of *MobileAgentFactory* class. A task is assigned to the new mobile agent. Next, it is configured with remote stubs of agent hosts. Then, its behavior is launched. So, it can begin its mission by moving onto the first agent host.

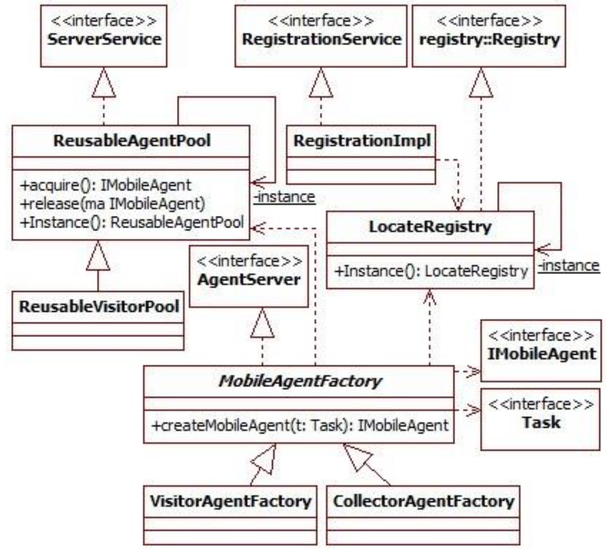

Figure 5. class diagram of *AgentServer* component

The agent server tracks the activities of mobile agents through incoming messages. By the end of the mission, the mobile agent uses the interface called *Service Server*. So, it returns to the pool of mobile agents pending mission. In a next request of an agent host, this mobile agent can be used to receive a new task and receive new information and thus make a new mission.

Because agents returning to the pool are already preconfigured with a business task, this can be useful to select an agent in the pool among all agents. Thus, the pool is indexed to the task of the agent. This is illustrated by the class, called *ReusableVisitorPool* that collects *Visitor* agents to be reused.

The creation and management of objects are a tricky business. It is often useful to create agents only when necessary. There are situations where it is useful to reuse objects (reusable agents), in other situations the duration of activity of an object is important to keep its state. This is the case of the services offered by the agent server where the concept of threads is crucial. We develop this notion in the next section.

C. *Real-time description*

A real-time system must meet certain performance constraints expressed in the execution time of tasks, even in the worst case. A real-time system is not as fast overall system, but a system fast enough or not too slow.

1) Management of server activity

a) Thread strategy

The RTSJ specification defines two categories of real-time threads. The difference between these two categories is the use of memory and whether or not a thread can be interrupted by the garbage collector. In the first category, defined by the *RealTimeThread* class, threads can use the heap allocation and regions to create objects. Therefore, these threads can be interrupted, but the garbage collector is not responsible for the automatic is responsible for the automatic management of allocated objects. The second

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category is composed of real-time threads where the worst execution time must be deterministic. These threads are defined by the class *NoHeapRealTimeThread* have a higher priority than the garbage collector to prevent any interference with its performance. Therefore, they are not allowed to access the heap allocations. They are forced to use regions for the allocation of objects.

The agent server component is structured into three parts which are independent. So, we have developed three classes which extend *RealTimeThread* because they can prevent the garbage collector from running, but would not preempt the collector if it is already running. These are the reusable agent pool, the registration implementation and the mobile agent factory. They run at a priority higher than the garbage collector. The three classes implement indirectly *Schedulable* interface. This means that they use the concepts of cost or duration, maturity period (for periodic tasks) overrun handler, if exceeded runtime and miss handler if the deadline is exceeded. The default scheduler is a priority scheduler and

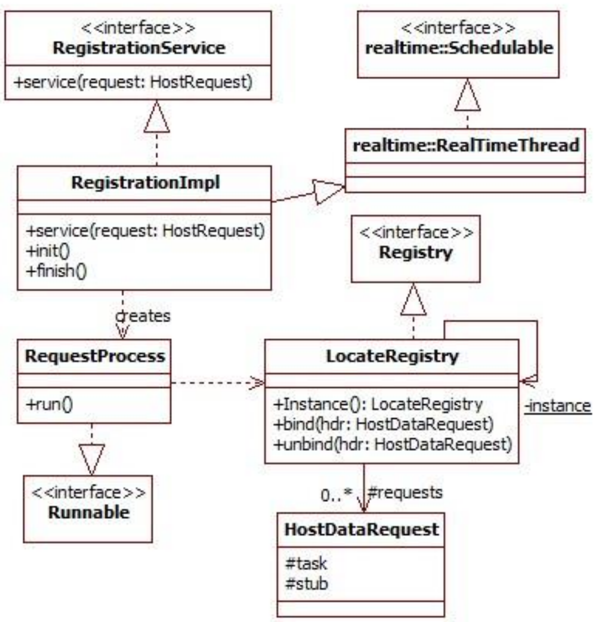

Figure 6. advanced class diagram of registration service

each real-time thread has its own priority

We have decided that requests from the agent hosts have to be registered first, next the return of mobile agents, and finally, the creation of new mobile agents. Also, we have assigned decreasing priorities. We use an aperiodic parameter to specify the cost and the deadline of threads. This set also a deadline missed handler to the schedulable servers. Then, these three threads are added to the scheduler to determine whether the system is schedulable. The scheduler is responsible of the whole runtime as a conductor in front of an orchestra. By the end of the registration thread, its callback (*finish* method) is invoked and this triggers the end of the whole agent server (figure 6). It means that the reusable pool of mobile agents is cleaned and the agent factory is stopped.

b) Memory management strategy

Each of the three threads (registration, agent pool, and agent factory) has its own memory area. A standard Java distributed application uses a distributed memory area that is hidden from the programmer and managed by the garbage collectors. In our application using real-time extensions, object creation is performed in the exact same manner, but now we can control which memory area the new operator uses to construct objects, and in turn, how that memory is reclaimed. We use subclasses of *ScopedMemory* for the registration requests. They indicate how the memory area will behave during allocations.

We use the *LTMemory* class which ensures that the allocation time will be linear. Moreover, the initial memory size, specified when the memory area is constructed, is contiguous. As a result, *LTMemory* areas pre allocates the minimum size specified when constructed. This memory area is not subject to garbage collection.

As an example, consider a request from an agent host about a future data collection. This object contains a data record with: the type of the wished task, reference of the remote agent host and a validity period for this demand. The registration is received by the registration implementation server. As a consequence, a *LTMemory* area is defined for this data record. In traditional object programming, this data record is available for garbage collection when the request is satisfied. This could perturb the behavior of the whole server component. The registration implementation is written so that it can reclaim the space for the temporary request without garbage collection. First, a scope memory area is needed; in our case *LTMemory* is used. Next, the code that is going to create and use the temporary request is constructed into a class that implements the *Runnable* interface; then, that class is passed as a parameter to *enter* method. To avoid creating a public class, a named inner class is used (called *Request Process*).

Our approach reclaims the temporary memory every time the *enter* method of scope memory is finished. However, as mentioned above, the creation of a named inner class is better than the creation of an anonymous class because it will still use memory in the surrounding memory area. With our solution, there is just the initial cost of creating the *HostDataRequest* object. We can remark that the only way to pass parameters to the run method (*Request Process* class) is by setting attributes of the object from the registration implementation. This main real-time thread dispatches the request process with the creation of an instance of *ProcessRequest* class which is runnable. So, it is available to treat next event.

c) Event management strategy

We use an approach to keep plain RMI and follow the event notification model similar to AWT/Swing. In those models, an object implements a simple listener interface and is then added to an appropriate event notification list. When that event occurs, every object in the list becomes notified of the event. The object takes whatever action necessary in

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response to the event. Such a model obviously is a better solution than polling, and solves the problems that polling has. There is no unnecessary network traffic and clients are notified instantly when a change in the data has occurred.

Applying this model to RMI is not trivial. Consider the RMI-based agent server as an example, where agent hosts registered with this agent server (interface *Registration Service*). After creating and configuring a mobile agent, the RMI agent server fires an event (called *AgentCreationEvent*) to inform the agent hosts of the arrival of a new mobile agent. To successfully do this, first, we enrich the remote interface named *IHostService* (Figure 3) with a method called *notify()*. The remote interface (called *RegistrationService*) with the *service()* method is analogous to the *setActionListener()* methods in Swing only instead of implementing the *ActionListener* interface, an object that implements the *IHostService* interface is needed. Just as the *ActionListener* interface serves to link an event with the application code that processes the event.

The *HostRequest* parameter serializes to a byte stream and is transmitted over the network to the server. The agent server needs to know how to deserialize the stream and reconstitute the object. Second, the *RequestProcess* instance registers the object by calling the *bind()* method of the *LocateRegistry*. Then the request is saved as a *HostDataRequest* instance.

When the same request comes from several agent hosts, only one mobile agent is created and configured with the set of remote stubs. It will execute its mission as far as the server has received corresponding request. All the hosts are notified by the server, before receiving the mobile agent. Now two cases can be considered: an aperiodic task as mentioned in the visit task (Figure 3), a periodic task as a data collection.

2) Configuration of periodic task

When mobile agents are prepared by a task and properties, some of them are executed periodically at different rates for most agents. Some of the agents may be scheduled for simultaneous execution though. Also, a mobile agent may need to start executing while another is currently executing. Many constraints are taken into account when scheduling, real-time mobile agents.

The first constraint is the amount of available resources; whether there are mobile agents into an agent host using existing resources. The second constraint is task precedence; to avoid ambiguous state. In the present work, all the tasks are considered independent but a task graph could be useful to organize structured activities. Certain tasks may need to be executed before others. The third constraint is timing; each task has its own deadline, some tasks execute longer than others, some may execute on a steady period, and others may vary between release times. Each constraint contains many of its own factors that need to be considered further when scheduling tasks.

The availability of system resources, such as the processor, is important to a scheduling multi thread application. But this goes beyond just the processor; other resources, such as shared objects that require synchronization across mobile agents, are to be considered. On agent host, we need to lock shared resources to avoid errors due to concurrent access. This limits a resource to being updated atomically, by one mobile agent at a time. Since resource locking synchronizes access to a shared resource, one mobile agent may become blocked when attempting to access a resourced that is currently locked by another mobile agent or host itself. In distributed real-time application, we ensure that high-priority (*RegistrationService*), real-time threads continue to make progress towards completion. Resource locking is commonly a problem since priority inversion can cause threads to execute out of order. When a request is received from a host, this host request is treated with care of resource lock.

A mobile agent is also defined by timed properties. The three main features are deadline, period and execution time. The deadline is requested by an agent host; its value is critical to determining if it can be scheduled feasibly. It is expressed with a timer local to the host. The host can request real-time tasks at regular time intervals. The mobile agent is configured with a value (periodic or aperiodic) along with the timing characteristics that might apply. The execution time of the request is the task cost used to know the interval execution times for a task. All these features are kept for the configuration step. The periodic feature is particularly useful by the end of the mission of a mobile agent. It means that the mobile agent has to start again the tour and performs its task. The controls are done to know whether the duration of the tour is permitted by the agent hosts.

3) Exchange management

All RMI services are sources of unbounded interference and their interaction with the real-time Java virtual machine and runtime libraries is not clearly defined. Also, delays occur without reason. Jamaica-RMI implementation avoids some of these constraints, but RMI communication is always a source of unpredicted measure. The non-deterministic behavior has induced many researchers to avoid or forbid their use in specific environments like high-integrity applications. Other researchers have developed their own implementation. First, we can cite the work of A. Ahern and N. Yoshida [10]. They present an object-oriented, Java-like core language with primitives for distributed programming and explicit code mobility. But, this is a formulation to prove the correctness of distributed programs; it does not provide an extension a RTSJ specification.

D. Holmes explained in [11] the contributions of the RTSJ specification, particularly the sequencing, memory management and asynchronous events. More specifically, it describes the importance of managing priorities and resources between threads introduced in [12]. In [13], S. Rho explains the design a real-time Java remote method invocation. In their work, remote method invocations are modeled as sporadic events. So, the overhead of their employment seems to be negligible. J. S. Iyilade provides a comparative study of the performance of the modern system, multi agents with the same criteria to try to limit the cost of RMI [14].

P. Basanta-Val describes the spectrum of approaches for distributed real-time [15]. Many projects around RMI and real-time are born, but often without reaching frameworks used by people other than the authors. The approach seems to Jamaica by far the most successful [16].

We used the results of S. Rho for the management of data exchange, especially for event notifications. Remote method invocations are modeled sporadic events and treated by a specific configured server. The latency of real-time RMI becomes stable without interference.

IV. Synchronization Algorithm

A. *Need of the control algorithm*

In a distributed application, a question remains often: how to time the execution of a Java program. Several answers are possible, depending on the used platform. There are some issues where *System.nanoTime()* cannot be reliably used on multi-core CPU's to record elapsed time. Each core maintains its own Time Stamp Counter (TSC): this counter is used to obtain the nano time (it is really the number of ticks since the CPU booted).

Hence, unless the OS does some TSC time warping to keep the cores in sync, then if a thread gets scheduled on one core when the initial time reading is taken, then switched to a different core, the relative time can sporadically appear to jump backwards and forwards. Because, a simple use of timers is not allowed because of the distribution of code over the network, we developed an approach based on timer vector. Our approach is implemented through the development of a new task called *TimerTask* as an extension of the Task (Figure 3).

B. *Distributed task synchronizer*

In this section, we present our implementation of task synchronizer based on clock vector. This approach is not new, for conflict resolution. Some NoSQL databases use this strategy too. L. Lamport defined the main concerns of this approach [17]. C.J. Fridge provided an application in the domain of the message passing system [18]. F. Mattern developed the clock vector idea for distributed applications. He proved correctness properties about partial ordering.

1) *Initial step*

In our context, agent hosts need to synchronize local activities. So, they need to receive mobile agents to do that. Mobile agents are initialized with a *TimerTask* instance and input data about one agent host. This *TimerTask* instance, will be the behavior of mobile agent after its migration onto the agent host. First mobile agents use accept service of their agent host. This host controls as mentioned previously. Next, it starts its mobile agent which starts its task. Each agent host has its own identifier. Each mobile agent has its local timer to manage.

2) *Core step*

The objective of the deployment is to apply a scheduling to the distributed host. Our scenario is the simplest: it is a step by step execution. Each time an agent host fires a local event, it calls the increment method of the mobile agent. Next, its own logical clock is incremented in the vector by one. Because, the role of the mobile agent is to synchronize the set of hosts, it needs to send its data to the other mobile agents.

Before sending the messages, it increments its own logical clock in the vector by one and then sends its entire vector. Each time a mobile agent receives a message, it increments its own logical clock in its vector by one and updates each element in its vector by computing the maximum of the value in its own vector clock and the value in the vector in the received message (Figure 6, arrows 11 and 12). Then, when the agent host (called *agentHost2*) fires a local event, its mobile agent (called *maB*) uses the updated vector to add a tick to its clock (Figure 6, arrows 15 and 16). As before, this time data is sent to all active mobile agents (only *maA* on Figure 6)

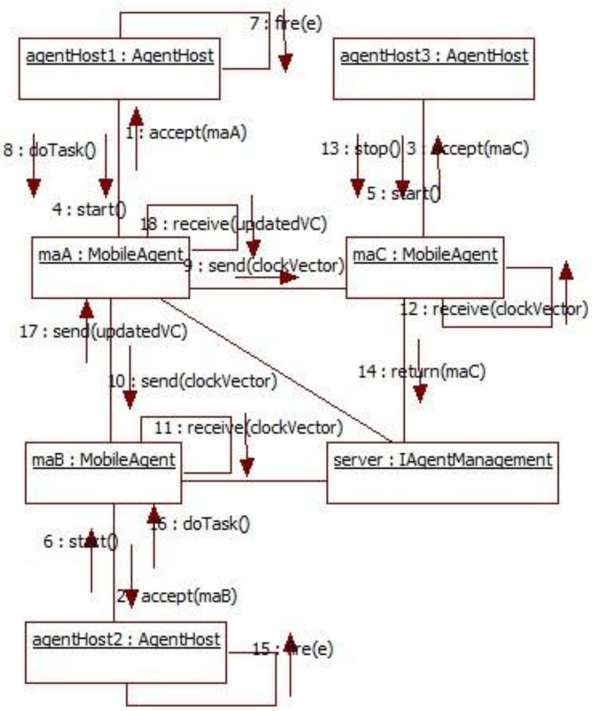

Figure 6. collaboration diagram of Task Synchronization

3) *End step*

The pilots of the distributed synchronization are the agent hosts. When a host decides to stop its partnership with the group, it just invokes the *stop* method of the mobile agent. Then, the mobile agent informs all the other that it does not accept future clock vectors. Next, it uses the *ServerService* interface of the agent server to go back to the server.

Figure 6 shows an agent host (called *agentHost3*) which stops the behavior of mobile agent called *maC*. So, it won't receive the next update from the other mobile agents. All steps of a task are released into the class called *TimerTask, which* implements the interface Task. This one requires three methods called: *init*, *doWork*, *end*. All the methods are not visible in Figure 6 because of visibility. But, this design corresponds to a Template design pattern which is wrapped into the class *MobileAgent*. In the next section, we present the results of this scenario and how it is possible to realize more complex interaction sequence.

V. Observations and Results

This experiment mixes real-time constraints and test construction. Because a distributed system is a set of local properties, it is not easy to discover global properties. Any observation involves perturbation on the whole program.

A. Information description

The great thing is that this time control extends to any number of agent hosts. It's just easier to understand in a context with two agent hosts (Figure 1), but the same mechanism works for any number of vector dimensions. Missing agent host is assumed to have version 0. The clock vectors can be used to describe temporal relations between events in a distributed system. A vector clock can be considered as a list of (place, transition) pairs, in which each place occurs at most once.

The clock vectors have a partial ordering, which captures the idea of before/after a given point in clock vector. It's partially because there exist pairs of clock vectors so that one is neither before nor after the other; they are "concurrent".

But this partial order stresses only on test path. The *TimerTask* instances can be created to stress other test paths. The initial properties of a mobile agent have consequences onto the interaction with its agent host. So, test coverage can be designed by setting precise properties for mobile agents before migration. After the execution, the observation of the state of each host allows the developer to validate or not the test. For instance, a given execution can achieve in a blocking state because a resource is not available. Such timer task is a helper class to detect anomalies into a distributed system.

B. More complex interaction development

This paper presented a simple scenario of test construction, but the more specific timer task is defined. This is particularly crucial when data injections are useful for the agent hosts. This can be occurred when the data environment of the host is set by the mobile agent.

In the previous example, the task is simply an automaton with three states called *init*, *doWork* and *end*. New kinds of task classes define such infinite loop behavior or with a specific skeleton based on a given interaction with en agent host. For instance, a message format can be forced or a volume of data.

VI. Conclusion

In this paper, we designed and implemented a real-time distributed application based on mobile agents. Java Remote Method Invocation is a valid transport protocol for mobile agent. By using a server centric approach with developer task classes, we achieved efficient and effective real-time test construction based on event orders. We believe that this work is an important and interesting step toward distributed real-time systems. We defined open frameworks, which can be completed by new contributors, especially who have experience with real-time concepts. The clock vectors are results which can be collected through the use of other mobile agents configured with *CollectorTask* class (Figure 3).

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Dynamic Pattern Utilization for Automatic UAV Control Support

Florian Segor, Chen-Ko Sung, Rainer Schönbein, Igor Tchouchenkov, Matthias Kollmann
IAS – Interoperabilität und Assistenzsysteme
Fraunhofer IOSB
Karlsruhe, Germany
{florian.segor, rainer.schoenbein, igor.tchouchenkov, matthias.kollmann}@iosb.fraunhofer.de
chen-ko.sung@sung.de

***Abstract*—Within the research done at Fraunhofer IOSB in Karlsruhe in the area of civil security, various types of sensor/sensor systems, ground vehicles and Unmanned Aerial Vehicles (UAVs), have been used in the project AMFIS for some years. When it comes to aerial situation overview and reconnaissance, the research is focused primarily on electrically operated Vertical Takeoff and Landing (VTOL) systems which can be operated easily by police or rescue forces on account of the simple use and the good maneuverability even during applications in urban areas. One of the main research intentions of AMFIS is the further reduction of workload for the operator in scenarios where multiple and complex networks of different sensors and sensor carriers are used. That leads directly to the need for a high level of automation of the single sensor carriers. To further improve this automation the use of a dynamic and adaptable ground pattern as well as the detection and extraction of the information content of the displayed ground pattern onboard a flying vehicle is examined. The central objective of this investigation is the technical advancement of the dynamic ground pattern and the evaluation of the present test results as well as the use limits and the possibilities of the presented solution.**

Keywords-automatic UAV guidance; adaptive pattern detection; security and reconnaissance; visual communication; civil rescue forces

I. INTRODUCTION

As for the technological advance today, there are many systems and sensors to support rescue forces in their work to manage natural or manmade disasters. One focus of the research done at Fraunhofer IOSB is the application of modern sensors and sensor carriers to support police and firefighters in such situations. The project AMFIS [1] is concerned with developing an adaptable modular system for managing heterogenic mobile, as well as stationary sensors. The main task of its ground control station is to serve as an ergonomic user interface and a data integration hub between multiple sensors mounted on light UAVs or Unmanned Ground Vehicles (UGVs), stationary platforms (network cameras), ad hoc networked sensors, and a super-ordinated control center.

Within the amount of different sensor carriers already integrated in the laboratory test bed, micro UAVs, especially small VTOL systems, play a special role. An application of multi-rotor systems within rescue or security scenarios had become more realistic in recent years because of the rising usability and higher levels of automation and has in some cases already become reality. The research done in AMFIS focuses also on the extension of the application ability and automation of these sensor carriers. The aim is a ground control station permitting a single operator to control a complex heterogeneous reconnaissance system, not only sequentially by dealing with one sensor carrier at a time, but in parallel with reduced workload and supported by a high level of automation.

Our experiments have shown that the achieved level of automation is sufficient in most cases for the automated application of multiple sensor carriers with a minimum of operator interaction [2], [3], [4].

Only the landing process needs the unlimited attention of the user or a manual steering pilot because the navigation based on GPS and pressure sensors is not precise enough for a secure unattended automatic landing when space is the limiting factor.

Though, the automatic take off process of a GPS supported VTOL UAV is possible without supervision, however, the flight sequence is far away from an absolutely secure procedure and can be further improved therefore.

To remove this restriction and to protect the aircraft as well as the personnel and the material near the lift off and landing site, procedures were developed to provide an on board detection of a ground pattern to use this information for an exact automatic landing [5].

To use a static pattern, some problems have to be considered. Flying on different altitudes, the size of the pattern varies and a partial coverage of the pattern is inevitable on low altitudes making it hard to provide constant pattern detection. In addition, we wanted to use the visual information to add a new communication channel to control the UAV. For these reasons, the basic detection algorithms were designed to be also capable of detecting different patterns and to extract additional information from the ground pattern as for example deviation from the approach path or the direction and speed of a potential movement of the landing platform (if, e.g., mounted on a vehicle).

II. RELATED WORK

With the advance of the technological progress, UAVs can be successfully used for more and more applications.

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Hence, during the last 10 years, varied research results concerning UAV-swarming, independent navigation behavior, sense-and-avoid procedures and also work within the topic of autonomous or automatic landing and lift off were published. Within the field of research about the automatic landing of a VTOL UAV, the principle of using a ground pattern and visual pattern recognition for navigation and position extraction has been treated extensively. The usability of this approach is undoubted according to the achieved success and the application ability of such a system is beyond all questions.

S. Sharp et Al. [6] presented a test bed for onboard detection of a defined ground pattern using Commercial Of The Shelf (COTS) camera and hardware components. Saripalli examines a very interesting application in [7] using a pattern detection algorithm on board of a small unmanned rotary aircraft. A theoretical approach to track and to land the UAV on a co-operative moving object is presented. Zhou et al. [8] as well as Yang et al. [9] examined the possibilities of an autonomous landing on a static "H"-shaped pattern. Especially, Yang pays special attention to the high noise immunity and the rotation independence of the detection algorithm. Xiang et Al. [10] describe very interesting set up with low-cost COTS components (IR Cam of the Wii remote). The components are used to build an active IR pattern for the positioning system of a multi-rotor UAV. Lange et al. [11] also address the landing of an UAV on a ground pattern. They concentrate on handling the problem of the discrete scaling of the pattern independent of the different flight altitudes of the UAV by introducing a special designed circular ground pattern. Through different circles, which are becoming smaller to the centre of the pattern, the algorithm is capable of detecting the landing site also during the final flight stage of an approach without the need to adapt the absolute magnitude of the pattern. A similar approach is followed by Richardson et al. in [12], describing the landing of an autonomous UAV on a moving ground platform by using a pattern detection algorithm in co-operative surroundings. As in [11], a multistage pattern, which enables the complete visibility of the pattern for on board recognition also at a low flight level, is used.

All these researchers have shown good success in addressing very similar purposes. However, the suggested solutions suffer from some limitations as for example the restrictions due to the missing discrete pattern scaling during landing and takeoff. Additionally, we assume that each static mark approach will react on a pattern-like natural or man-made structure with miss-interpretation or detection errors. The dynamic pattern introduced in this research allows the construction of an additional communication link to the UAV and, besides, solves problems which are not handled yet and therefore differs from the present proposals.

III. Application Scenario and Motivation

One of the central application scenarios of the AMFIS system is to deal with the support of rescue forces in disasters or accidents. The varied application of different sensors on board of a UAV can provide support for the rescue forces and make their work more safe and efficient. Derived from the experiments done with the AMFIS system, the missing capability of the UAVs used within these experiments to precisely take off and land automatically on a designated position was identified as one of the main challenges for the professional application – especially when multiple UAVs are deployed at the same time. To deal with this problem the pattern recognition was developed and tested with a UAV system experimentally.

A dynamic pattern is not necessary compelling for the solution of the primary problem and quite good results were achieved using non-dynamic static patterns. Indeed, a dynamic pattern offers additional advantages which extend the application possibilities of such a system. Just by using the access to an, in principle, almost unlimited pool of different signs and symbols, the abilities of a pattern concept can be clearly enlarged. By that, the detection capability of the algorithm is not limited to the pure localization of the pattern any more. It can be extended by the capability to extract information content hidden within a detected pattern. Besides, a dynamic pattern still offers some other advantages. As already Lange et al. [11] stressed out, an essential problem within using ground patterns originates from the detection of a static pattern at different flight altitudes. Even when using a fish-eye lens during an approach of the sensor to the pattern, the probability rises that parts of the pattern are not grasped by the sensor because of the limited aperture angle and the increasing appearance of the image or pattern. The use of a dynamically adaptable pattern allows resizing the shown pattern. Thus, the size of the pattern can be adjusted matching the current flight altitudes raising the chance that the sensor is capable of viewing the shape completely. Though the algorithm is designed to be rotation and scale independent, nevertheless, the result quality of the detection algorithm can be further improved by aligning the orientation of the pattern with the direction of the UAV as well as considering its point of view and distorting its perspective.

However, the introduction of an additional visual communication channel provides even more advantages. Unfortunately the widely used radio data connections between UAVs and their dedicated ground stations can be very easily disturbed - intentionally or unintentionally. The detection of a used radio frequency can be done using COTS systems and even if it is not so easy to break into the communication to take over the UAV, in most cases it can be overlaid leading to a complete communication breakdown between the ground control and the aerial system. Using a visual communication system, interfering with the communication becomes more difficult because a potential disrupter stays hardly unnoticed if applying a permanent influence on the pattern providing ground platform.

IV. Adaptive Pattern and Onboard Detection Chain

The currently used setup for development, evaluation and demonstration of the conceptual design consists of the mobile AMFIS system as a control station, on the one hand, and the dynamic ground platform and the camera on board of the UAV for visual information extraction, on the other hand. A central object for further development is the technological realization of the dynamic ground platform which will be integrated into the AMFIS communication backbone for information exchange and to receive control commands in the future (see Figure 1).

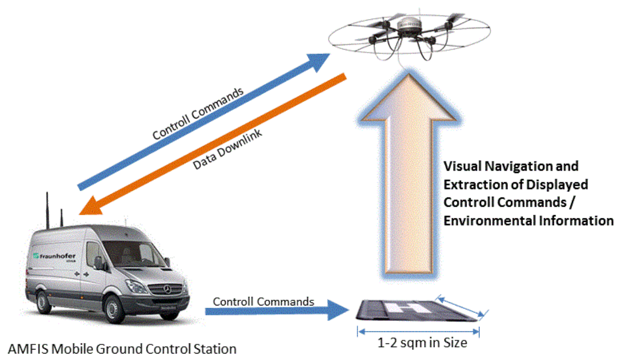

Figure 1. Sktech of the final target system.

For the initial non-dynamic testing of the algorithm, a static ground pattern with the shape of a white "H" on a black background was used. This test setup was designed to experimentally deploy the developed algorithm in a realistic test scenario under real conditions and environmental factors. However, on account of the long-term aim of developing and applying a dynamic pattern, the adaptability and expendability of the detection interpretation algorithm was emphasized. Hence, the developed dynamic pattern should show exactly the same static pattern (a white sign on black background) to achieve the highest possible contrast in the first experiments. Because the detection should be functional under bad lighting conditions also, a mechanical solution with flipping parts was excluded. An additional requirement was the demand for a simple solution to display different symbols or patterns. To cope with this, different Light-Emitting Diode (LED) matrices were examined and tested for their suitability. The experimental used ground patterns are all slightly different in technology and size. The originally used prototype based on single low cost LED panels and reached a size of 65cm x 65cm. Tested under realistic conditions, it shaped up that the low cost image display matrix, which provides control over every single LED, is not suitable on account of the used Pulse Duration Modulation (PDM) and the low fixed refresh rate. The PDM controlled LED cause a flickering not visible for the human eye, but for the camera. Experiments showed that this flickering troubles the algorithm in detecting possible blobs for the pattern in the video.

To reach a non-flickering representation, small 3x3 illumination LED matrices were used and assembled to an 18x21 experimental matrix even smaller than the original test system (see Figure 2). This pattern matrix turned out to be absolutely flickering free and can, therefore, be detected by the algorithm as one structure without any problems. The second advantage is that the assembled platform was luminous strong and provided the capability to see and detect the ground pattern even in bright sun light.

But, on account of the restrictions of the used 3x3 LED pluggable modules as missing control technology, limited displaying possibilities, difficult handling and the need for a more flexible test bed, the current research in this project is focusing on the use of a commercial high-end LED display for outdoor application, which is suitable to solve the problems of the low-cost display systems and can be deployed as a fully dynamic pattern projecting ground platform. For the new testbed, a panel of 1.57 square meter of high end SMD (Surface-Mounted Device) LEDs was purchased and is about to be included in the experimental setup. This more advanced system is designed to allow also a detection of the optical signals at higher flight levels of approx. 30 – 80 meters due to its size.

In addition, the panel offers the possibility to control the single LEDs again. This allows to scale the shown pattern and to adapt it to the flight altitude of the drone. The full 1.5 sqm can be used for maximum scaling and, therefore, reaches a size that allows the pattern to be recognized at an altitude of about 80 meters. It remains to be examined whether a pattern extraction is still possible in this distance.

If the UAV approaches the pattern or reduces its flight altitude over the pattern, the scaling can be adapted and the size of the shown pattern is reduced. Therefore, the full visibility of the pattern can be guaranteed on a lower flight level. Nevertheless, a short distance between camera and LED screen, as it happens on every final landing approach, can be seen as critical to functionality, because the low distance to the projection screen (d<1 meter to the surface) leads to the detection of single LEDs or rows of LEDs by the camera. In this case, the process chain is no longer capable of finding a coherent pattern area on which a verification and classification of the pattern is possible. This leads directly to the conclusion, that a workaround or an extension of the process chain is necessary to obtain the precise navigation during the last seconds of the final landing approach.

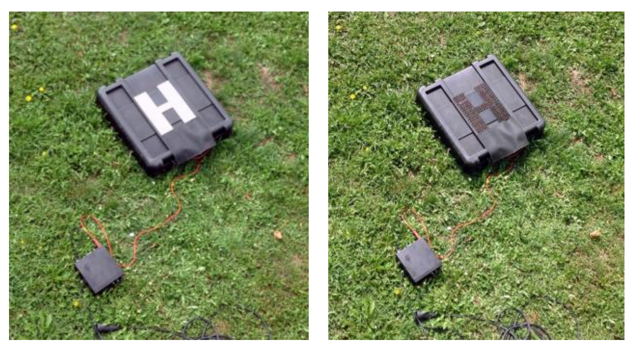

Figure 2. Illuminated an non-illuminated ground pattern.

The basic functions for adaptive pattern recognition have been reported in [5]. For the pattern recognition, there are two major tasks which must be solved. One task is the separation and extraction of possible pattern sub images from image sequences as pattern candidates for the recognition

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and interpretation of manmade landmarks. The implemented process chain with an adaptive threshold operation for this task works well and has not been modified for the present investigation. Another task is the recognition of patterns or manmade landmark images. The challenge of this task is that the onboard process chain for image evaluation must be robust, non-compute-intensive, expandable and fast. For that reason, we developed a so-called "zigzag" method which analyzes how many binary values of relevant parts of an object image are correlated with the expected values. The present investigation has shown that the method is easy to extend.

The algorithm inherits some serious advantages, as for example the rotation and scaling independence (see detection in Figure 4). At the same time it was designed not only to detect a pattern on the ground to calculate correct and GPS independent navigation information, but also to extract information from the different pattern sequences. The used "zig-zag" method has great advantages because of the fast and simple logic, used to recognize a single pattern. The procedure is quick and efficient and, hence, suited to deliver usable results with limited hardware capacity onboard which has been proven in the past attempts [5].

To achieve a sufficient information density, the different patterns have to be enlarged to reach the capability to transmit more complex information (see Figure 3).

Figure 3. Examples of used patterns.

This can be seen as a key feature of the dynamic pattern detection beside the improvement of the navigational information for the automatic landing. As already mentioned above, different patterns are shown at the same projection plane sequentially and can be recognized by the camera / algorithm on board the UAV. On the one hand, by flipping the patterns, errors occurring due to the detection of similarly looking natural structures should be avoided in future, because the system expects a regular change in the detected area. On the other hand, dedicated information will be linked to the single symbols. Orders or important information, as for example the current wind direction or a possible movement of the ground platform, can be encoded and transferred using the pattern sequences.

Therefore, the palette of used symbols was complemented with additional signs to extend the capability of encoding more complex information into a pattern sequence by switching between the introduced signs. Nevertheless, the used pattern pool is held small at the present time, because for every new introduced pattern the algorithm needs to be adapted in order to "learn" the new shape and to recognize it during the detection sequence. Additionally, an enlargement of the pattern pool also requires more logical operations during the scan process of possible pattern blobs found in the images which leads directly to an enlargement of process workload during the classification of the pattern in flight. It remains to optimize the balance between size of the pattern pool (for information encoding) and duration of the pattern classification process.

V. RESULTS

With the application of a static pattern, the functionality of the algorithm and its suitability has been proven for the integration onboard the UAV under the aspect of the limited computing capacity within this mobile system. The work based on these results has shown furthermore that even a simple active pattern which is reduced in its adaptability and displaying capacities is capable of improving the detection process. In [13], the test construction was described proving that the theoretical concept is functional and such a system could be applied successfully. The main objectives of these tests focused on the suitability of the concept mainly under strong external light influence and the enlargement of the detection capability to more than a single pattern as well as the differences in the detection results of the algorithm under differently strong self-illumination of the ground pattern (see Figure 2).

These experiments have shown that the used matrix can cope also with direct solar irradiation and emits enough light to produce a homogeneous pattern detectable by the algorithm.

Because of the promising results, the next step is to improve the possibilities of the ground pattern. Due to some restrictions of the used LED matrices like a slightly too big pixel distance and the difficult control of the LED sub elements of the pattern, there is a need for a platform with a higher usability. Therefore, the current step is to change the technology of the ground platform to a highly efficient commercial LED display that comes with the capability of high brightness and low pixel distance as well as multi-color representation possibilities. This promises a further improvement of the test bed and allows extending the experiments to test a bigger number of patterns without additional expenditure.

For the onboard component a fish eye lens was selected in the presented work to increase the detection area on one hand and to generate a stronger distortion in the picture particularly in the edge areas with the aim to test the algorithm also under more complicated conditions on the other hand. We assumed that the pattern must always be clearly visible from the image sensor, independent of the flight level of the UAV during the in-flight detection process. Numbers and characters are good land marks, because they have a system behind them, and can be encoded with high information content. To test the generality of our algorithm and process chains for the landmark detection and pattern recognition, two ground patterns "H" and "L" were used. These two ground pattern consists of the same hardware and were assembled currently by manually re-plugging the 3x3 LED modules.

Figure 4 shows the results of the pattern recognition while maneuvering the sensor carrier over the landing site. The width-to-height ratio of the ground pattern and even their shape may be strongly distorted by the fish-eye and rotation motion. The process chain works also well with the ground pattern "L" that is also captured with the fish-eye lens. To avoid ambiguity in the pattern recognition, a minimal size of the detected object image was used. This means that small object images can still be detected and extracted, but they are not suitable for pattern recognition.

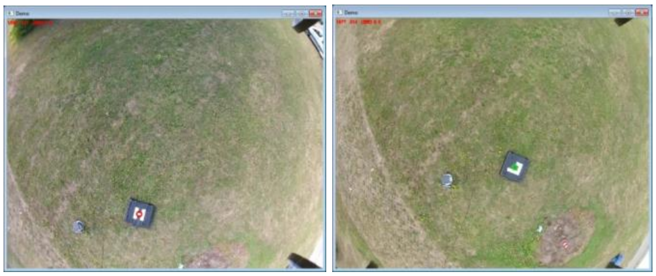

Figure 4. In-flight detection of shape "H" and "L" marked by colored circles at the center of the pattern ("H" is marked red; "L" is marked green)

Beside the fact that the algorithm cannot recognize the ground pattern in some frames the experiments showed that structures similar to the pattern can falsify the results. Though the used patterns were chosen that natural counterparts are rare, nevertheless, the attempts with static or partially static patterns showed that faulty detections are possible.

This leads to great danger for the flight system and ground crew and is therefore an essential point which needs to be solved in future works by introducing the final fully dynamic pattern projector.

VI. Conclusion and Future Work

On account of the conceptual change from a luminous-strong but not very easily to adapt matrix LED to an outdoor suited high-resolution LED display, the test system can be essentially extended in close future.

New possibilities are arising to improve the abilities of the algorithm in its detection quality and further extend its error tolerance as well as to develop completely new draughts for intercommunication between platform and UAV. For this purpose, the introduction of an initial recognition sequence which is visualized cyclic before each information transmission on the platform should be also examined.

Essential research topics will cover investigations in resolution and adaptation of the projected pattern in dependence of the altitude of the UAV. The focus will be to find the optimum way of scaling the pattern and identifying the thresholds at which a homogeneous pattern projection is no longer possible and how this problem can be avoided. That includes also tests about the practicability of different geometrical projections and rotations of the pattern and to determine the lowest possible angle of view to admit the most precipitous angles of approach. On account of the changed pattern technology a renewed test sequence to determine the efficiency of the high end panels with regard to the environmental conditions in particular to the solar irradiation has to be conducted. Due to the extended capabilities new possibilities of interaction have to be evaluated. The investigation of color coding as well as a negotiation or automatic calibration of the pattern can provide interesting new capabilities. For example, the adaptation of the luminous strength would be conceivable as a reaction to the current environmental conditions taking into account the feedbacks of the drone forwarding the information which kind of pattern coding can be recognized with the highest success rate.

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A Tool Architecture for Diagnostic in Power Electric Network Using Method Engineering and Multi Agent Systems

Jean-Pierre LIENOU
Department of Computer Engineering
ENSPY/LIMSS, University of Yaounde I,
Yaounde, Cameroon
lienou@gmail.com

Emmanuel TANYI
Department of Electrical Engineering
Faculty of Engineering, University of Buea,
Buea, Cameroon
emmantanyi@yahoo.com

Marcellin NKENLIFACK, Thierry NOULAMO
Department of Computer Engineering
IUTFV/LAIA, University of Dschang,
Bandjoun, Cameroon
marcellin.nkenlifack@gmail.com, thierry.noulamo@gmail.com

***Abstract*—Electrical networks in developing countries suffer from several shortcomings. One can cite the scarcity of well-trained professionals to carry out maintenance operations, the ageing of equipments, the lack of spare parts and especially the deadlines for the supply in case of purchase. Diagnosis is an operation often perilous, arduous, long, semi-formal and sometimes leading to long calculations or to not satisfactory results. Activities carried out during the diagnosis are sometimes related to particular components of equipment or "modular" steps of a diagnostic algorithm. These different diagnostic modules can be formalized or not. Since recently, situational method engineering is improving its formalization and it allows designers to reuse fragments in the design process in software life cycle for example. The purpose of this paper is to present an environment based on multi agent systems which allows you to apply method engineering in other areas (here the diagnosis in the electrical networks) in order to solve problems specific to subject area. The building process is illustrated by the decomposition of certain diagnostic procedures in simpler activities achievable in parallel on the ground or in a software application and thus, reduce diagnostic time while simplifying procedures. The final goal of this on-going work is to create depending on the context, self-own diagnostic procedure and instantiate it.**

Keywords- Situational Method Engineering; FDI Diagnostic; Multi Agent System; CIM; meta diagnostic

I. INTRODUCTION

Since a decade, Situational Method Engineering (SME) has evolved and is highly appreciated by practitioners in information system to design methodologies for complex software from “chunks” of methodologies. However, outside of its use in process engineering and software engineering, SME is not visible in other areas of research. In effect, the SME is going in the opposite direction of the creation of a standard way to do things. It is focused on how to identify and document fragments from existing sources by including the practical recommendations of the industry; on how to store these fragments and ensure their quality and how to build a methodology for a specific situation from these fragments. Any constructed method thus uses fragments from a database of fragments that leads to reusability [1] which is very important. We are trying to check if we can move this experience to the field of control engineering and more precisely in diagnosis. The problem is complex and we propose to solve using agents. Agents are used in electrical networks on the theoretical level and they are applied in various features. In [2], authors use Intelligent Electronic Device (IED) as an element of measurement to propose a model for protection of equipment. In [3], authors propose Multi Agent System (MAS) for the diagnosis of faults in a distribution network of a decentralized energy production. Note that the international community is working to produce an ontology specific to power field and namely the Common Information Model (CIM) of the IEC that we recommend.

This paper is structured as follows: Section II presents the some key concepts in the diagnostic field, SME, and the MAS in power electrical networks. Section III presents the proposed approach for the construction of the method with a scenario on a hypothetical example, as well as the procedure for the selection of an elementary fragment to form a methodology for diagnostic. Section IV presents the platform model with the features of main agents.

II. BASIC CONCEPTS

A. *Diagnostic*

Diagnostic is a process that determines whether a device or a system is in good or bad state. There are several types of diagnostic and several communities i.e., Fault Detection Isolation (FDI), Dx, and Bridge were formed in the recent past and each develops its conceptual diagnostic tools. Bad state according to FDI community is when the residue calculation does not match with the model. For the Dx community, it is when the observations are not coherent with the description of at least one component or of the system; therefore, an R-conflict can help to diagnose a system. To support our remarks, we will use a diagnostic approach as

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proposed by the FDI community. By applying the diagnosis to the grid, it must be noted that the network must be modeled in the state space model. Several classes of FDI approach exist. These classes include:

- Methods based on residuals using the model to predict the values to be measured. The parity space method that eliminates the unknown variables and the estimation method by state observers is to estimate the unknown variables
- The parameter estimation methods use estimation techniques to calculate a value of a parameter of a model when its structure and design parameters are well known.

We can list some diagnostic methods of heating wire or equipment (electrical panel) in Table 1.

TABLE I. SHORT LIST OF DIAGNOSTIC METHODS TO DIAGNOSE HEATING IN EQUIPMENTS

Faults	Diagnostic principle
Diagnostic of heating	Infra-red Thermography
	Load Flow Analysis
	Radio Frequency analysis
	Dissolved Gas Analysis (DGA)

We use the term meta- diagnostic to specify the construction of a diagnostic approach by SME.

B. *Situational method Engineering*

Literature on SME shows the existence of several approaches to describe the process cycle. Researchers helped to formalize SME and tools are created to facilitate their implementation. Existing libraries appear and are enriched every day. They include OPFRO, Praxos, etc.. Several standards address issues of SME. The most common standard is the ISO/IEC 24744, OMG SPEM, OPF and many others. Most used component types in SME are "fragments", "components", "chunks", "OPF fragments" and services.

a) The Brinkkemper approach

In this approach, there are two types of method fragments. The "process fragment" which describes the steps, activities and tasks and the "product fragment" for the structure of the product at the end of the process to be put in place. A fragment can have relationships with other fragments and can be composed of several fragments; an elementary step description of the steps for successful description of the components to put in the database and extract to recompose a method.

b) The OPF approach

The approach is based on "Object oriented Process, Environment, and Notation" (OPEN), which generally produces five meta classes each producing a fragment method (process or fragment method).

c) The FIPA approach

In [1] and [2], SME is used for the construction of MAS and is now considered standard in the field of SME. It is important to define the meta-model of each fragment method to identify the architecture of the basic methods and to define how to connect these fragments of methods and techniques to describe integration method.

d) Ralyté & Rolland approach

In this approach, also called "chunks method" the process is based on the fact that any fragment of a new method to be build has an intermediate result [5]. The approach is composed of two parts: the "process aspect", which is called either map or direction of realization of intentions and the "product appearance".

e) "Method service" approach

This method based on services is according to us, the one that can be adapted to diagnostic although we have not yet investigated all others for suitability. Its meta model is shown in Fig 1.

In general, methods can be constructed in several ways: ad hoc (intuitive building from scratch) by changing (abstraction, instantiation or adaptation of an existing method, taking into account the current situation) by extension/reduction a method (by complicating or simplifying assembly) and taking into account possible overlaps between fragments. The main objective is to use a quick way for the construction of methods corresponding to the specific needs of each project by:

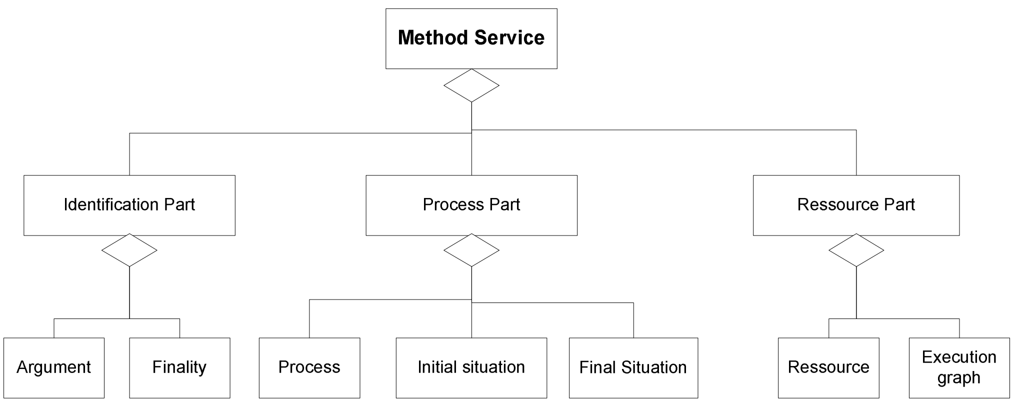

Figure 1. "Method Service" meta model in SME.

- guiding the method engineer in the requirements definition method for a specific project to develop a diagnostic system.
- helping the method engineer to select the components satisfying her method's needs.
- helping the method engineer to assemble selected fragments to build a new method or adapt an existing components method.

C. Electrical Power Network and Multi Agents System

An agent will be considered as a software component to build the system and get a multi-agent system. The choice of agent technology is due to a few of their properties: pro activity, sociality and especially the emergence from the activities of individual agents that is not provided in advance. This is actually the emerging functionality contrary to the program that is attractive to MAS because the designer does not have to consider all scenarios but the MAS can adapt functionalities: the analysis is less difficult in this point of view. But the task of distribution of subtasks on the agents remains complicated.

Equation (1) shows that emergence is due to the part of the collective function. The sum of the functions of the various agents of MAS should logically be less than the function of the MAS for collaborative agents

$$F\ (MAS) = \text{collective function} + \sum f\ (\#Agent) \qquad (1)$$

Fig 2 shows the use of a hypothetical system using a FDI diagnostic approach.

Telemetry data collection used to construct the current system state space model is compared to the nominal model stored in the physical system. A residue is calculated and if there are discrepancies, the diagnostic process is triggered and if not, the system proceeds to the next data collection.

III. PROPOSED APPROACH

The fragments of diagnostic methods depositories are stored in a database that can be queried. The granularity of selection can locate, isolate or identify the faulty component. According to written recommendations by experts, the result of diagnostic is displayed and a maintenance plan is proposed. Fig 3 shows the creation environment of the diagnostic methodology we call meta- diagnostic.

A. Environment for meta- diagnostics creation

The diagnostic engineer perceives a need to diagnose a component, equipment, network or even an embedded or not piece of software. He works in collaboration with the method engineer in diagnostic that will use (" method service" here) SME approach to populate the basic methods and simple compositions that allow not get deeper into details in the construction of the new method of diagnostic by the field engineer in diagnostic. Once the methodology is ready, the results are monitored and there is feedback to enrich the base with statistics and then the fragment method that failed to better diagnose is modified. With these statistics, future queries made by the engineer will show best diagnostic methods deemed efficient and with what tools you need to build the aggregated method.

B. Selection procedure and construction methods

In a typical case, where by example the calculation of the residue alerts, Fig 4 shows how to diagnose and if possible, calculate a new control law and control the industrial process. In case the process has a model in the state space, we can first select with a coarse granularity a method that will enable you to locate the faulty equipment and continue diagnosis with an identification method of the component in question within the equipment.

IV. MULTI-AGENT SYSTEM SUITABLE FOR PLATFORM

The platform must have a set of agents that will remove from the fragments base a set of consistent fragments that will diagnose the network an overall goal or in a goal - oriented equipment manner. We identify five agents for the realization of the system as shown in Fig 2 above.

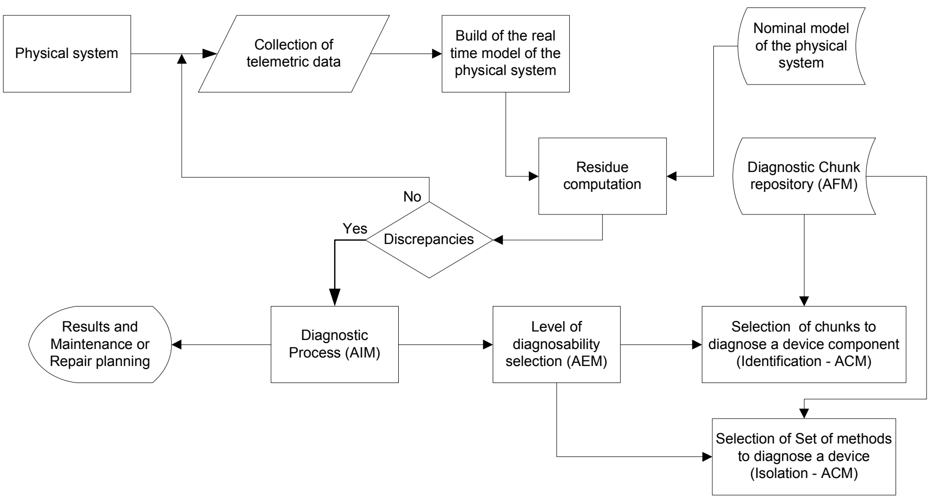

Figure 2. Main components of a hypothetical system using an FDI approach of diagnostic.

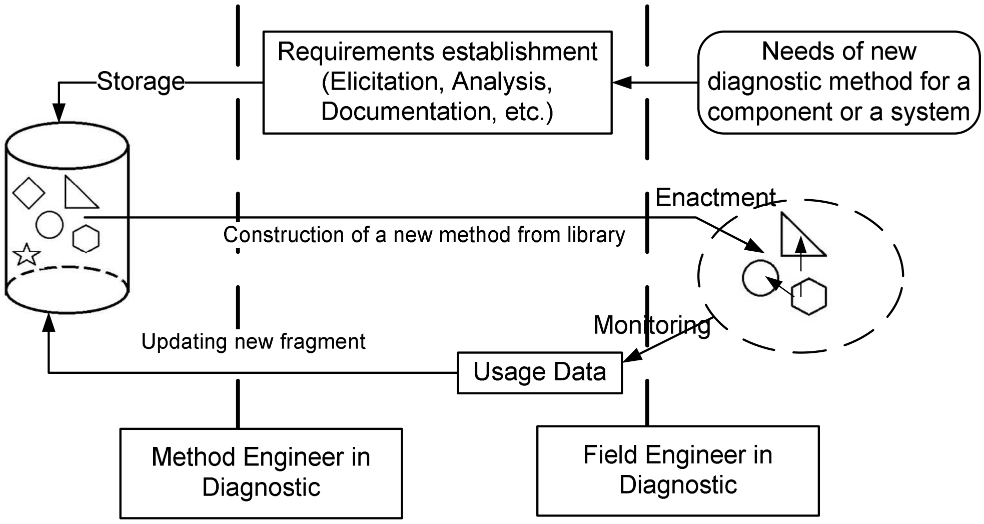

Figure 3. Actors' activities in diagnostic by method engineering

A. Agents functionalities

Agent Fragments Methods (AFM): This agent whose role is to interface with the basic methods used to describe a method of one of the approaches to SME and inserts it into the base methods.

Agent Construction Method (ACM): This agent from fragments and descriptions of engineering methods, selects the basic methods of the fragments that will develop new methods. It is responsible for the development, optimization, and many other features to generate the final methodology.

The Execution Methods (AEM) Agent: Under the orders of the Field Engineer Method, it monitors that developers have all the necessary need to develop their applications

The Interface Methods (AIM) Agent: Its role is to examine possible generated methods and modify the proposed agent construction choices if necessary.

B. Plausible scenario

Suppose part of the electrical network formed of bays, transformers, boxes and busbars. If the telemetry data create an alert, the engineer must create the diagnostic procedure based on telemetry. If he decides to go sequentially and check if the transformer is functioning and if not, he checks by Dissolved Gas Analysis (DGA), he will have the choice between the following basic methods: Duval's Triangle Roger's Ratio, CIGRE, IEEE, etc. These methods may not be feasible. He chooses a basic analysis radiofrequency method. If he suspects the box, he can think of a resistive or impedance short circuit. He moves on by studying the temperature in the cabinet or sensor noise, humidity, etc. If against the calculation of the residue shows that the voltages across the transformer is normal, it can check the safety equipment on the busbars without using selections of elementary algorithms.

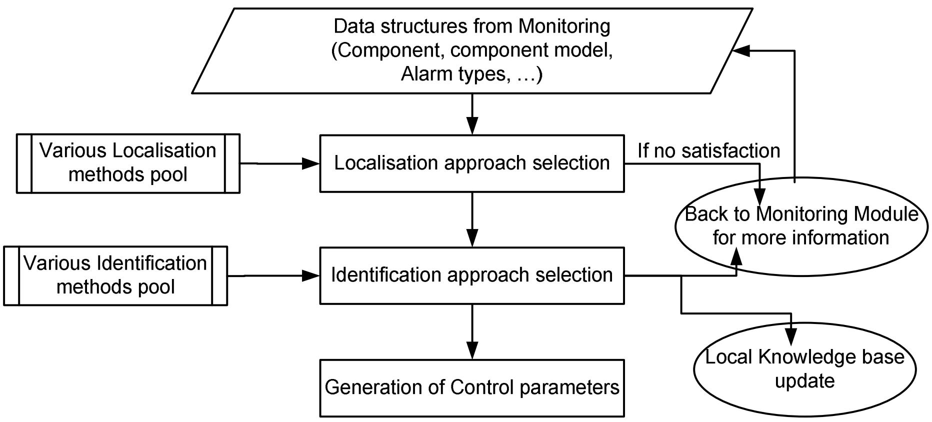

Figure 4. Fragments method selection process for diagnostic method construction.

V. Conclusion

We presented in this work the possibility of using SME to create diagnostic methodologies called here meta-diagnostic. We started with a background on SME and introduced briefly five approaches. We presented the model of the power grid based on the CIM and some uses of MAS in electrical networks. Our approach has been to propose a framework for describing the basic diagnostic methods (implemented by the method engineer using CAME tool) that can be recombined to form a complete method and instantiate (by a field engineer). As perspective, we think to formalize our concepts, expand the FDI diagnostic approach to Dx approach, refine descriptors methods and tools to use and implement it using Eclipse environment.

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Decision Support System for Offers of Fleet Availability Services

Andre Pozzetti, Ioana Bilegan, David Duvivier, Abdelhakim Artiba
LAMIH UMR CNRS 8201, University of Valenciennes and Hainaut-Cambresis
Valenciennes, France
andre.pozzetti@univ-valenciennes.fr

Barbara Matrat, Sébastien D'Arco, Sébastien Forato
Performance Based Services, Eurocopter
Marignane, France
Barbara.matrat@eurocopter.com

***Abstract*—The growing competitiveness within the aeronautic industry is inspiring companies to implement and offer Performance-Based Services. Whilst not necessarily a new concept, it is relatively new to the commercial aeronautic industry and presents prime suppliers with difficulties in predicting performance, risks and associated costs when performing whole of life costing during a service offer bid competition. In this paper, we propose a research to develop a novel methodology for a decision support system during the offer of Fleet Availability Services, remaining within the scope of Performance Based Contracting, achieved through the use of modeling, simulation, and optimization.**

Keywords - Performance Services; Aeronautic; Whole Life Costing; Performance Based Contracting

I. INTRODUCTION

As the Total Cost of Ownership (TCO) increases with complex aeronautic systems, companies within the industry are undertaking various strategies to gain a competitive edge. This is driving aeronautic customers to focus their needs on performance of the acquisition and sustainability of such systems. Whilst performance based sub-services, such as Parts By the Hour (PBH), Repair By the Hour (RBH) etc., are well-defined and relatively well-optimized throughout industry and literature [1 - 4], the ability of the industry to optimize global services, such as Fleet Availability, are yet to be established, and therefore competitive advantages are there for those whom provide Performance-Based Services (PBS) ensuring fleet operability.

Performance service methodologies, such as Performance-Based Contracting (PBC), or Performance-Based Logistics (PBL), are not new concepts to numerous differing industries globally [5][6], however the demanding military and increasingly demanding civilian aerospace acquisition and sustainment requirements present both industries with a significant challenge. This is inherent in the design of PBC and generally results from each contract being different [5]. Additionally, the services, which can be considered as a support strategy, are completed and comprised of integrated discipline sectors, such as logistics, maintenance, and engineering support [5], [6], [7], [8]. Generally measured through Key Performance Indicators (KPIs) [5], [6], [9], each covering different aspects, such as process engineering, the maintenance of the support systems, and the supply chain. It is however difficult to calculate the impact that decisions may have whilst operating within a complex system, which then contributes to the uncertainty for operations and costs.

The goal of this research is to develop a novel methodology for decision support in the offer of Fleet Availability Services (FAS), in the frame of Performance Based Contracting using modeling, simulation and optimization.

The structure of this paper is as follows: First, we provide an introduction to the problem with a literature brief to both define, and validate the problem, and topic. The middle sections discuss the key objectives of the research, following with our proposed methodology. Finally, preliminary results and concluding remarks are discussed at the end of this paper.

II. LITERATURE

Performance Based Contracting can be considered as a support strategy, utilized to achieve measurable outcomes for a product of service. In the aerospace environment, common result areas involved are: Availability, Reliability, Maintainability and Supportability [5], [6], [7], [8]. Generally, these are interpreted as:

- Availability – The availability of an aircraft at a point in time at a defined location.
- Reliability – The measure of time between system failures.
- Maintainability and Supportability – Engineering, Maintenance, Logistics and Supply Chain efficiency.

Outcomes are comprised of integrated Key Performance Indicators, and are usually defined in a negotiation process between both parties, during both the contract tender phase and again throughout the contract award negotiation phase. For example, the Maintainability and Supportability outcomes, as displayed above, may be comprised of KPIs covering the various aspects of engineering and maintenance processes as well as a mixture of supply chain measures, such as:

- Demand Satisfaction Rate
- Turnaround Time on Parts Delivery
- Engineering Change Requests
- Design Modifications
- Cannibalization, et cetera.

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With these outcomes and KPIs, described above are well-established and understood [5], [6], [7], [8], [9], the majority of literature has focus on the availability of sub-services rather than a fleet availability service as a complete system [2], [3], [4], [10]. Furthermore, little knowledge is developed on the various interactions within that complex system. Datta and Roy [10] proposed a methodology on the effect of customer-focused risks in cost estimates, highlighting other potential problems in the costing and development of availability type service contract. The highlighted problems are:

- The reliability of data,
- Assumptions regarding equipment failure,
- Considerations of uncertainties through the system life-cycle,
- Uncertainties of the customer's contribution to performance,
- Communication between the customer and supplier,
- Prediction of future maintenance and the inability to understand the cost impact of customer risks.

Due to the uniqueness of Performance Based Contracting and/or Performance Based Logistics contracts, one can assume that corporations are exposed to a significant amount of risk and potential over cost for managing, maintain and supporting such contracts across multiple platforms, with multiple customers. Consequently, a need exists to develop a methodology, design and implement a system for decision support, which may integrate the information communication systems environment of the supplier and customer. Additionally, the methodology should allow for the uncertainties and, in order to meet the needs for decisions in various scenarios, the methodology must use a multitude of modeling and optimization techniques.

III. OBJECTIVE

The main objective of this proposed research is to develop a process of decision support to achieve a performance goal of overall availability using modeling, simulation and optimization of a service delivery system in aeronautics. In particular, it is proposed to conduct a research work aimed at developing a methodology for optimizing the performance of services ensuring that fleets of aircrafts are operational.

The preliminary step is to analyze what are the achievable performances of availability, given the intrinsic characteristics of the system and its associated support. Performance indicators need to be formally identified and a method for calculating the level of performance will be proposed. Next, functional architecture will be defined, which will provide the framework for decision support. Each element of the architecture will correlate with the various features identified as mandatory, and it is envisaged that a module type information system be designed:

- A module allowing the collection and analysis, either periodic or real-time,
- A module for the pre-processing of the data characterized by the system,
- A module for the modeling phase of the defined problem,
- A management module for optimization for maintenance and logistics, both strategic and operational,
- A simulation phase will also integrate the process of decision support in the specific operating environment.

The simulation will be used for non-deterministic study decision behavior of human operators, an essential element in this complex system. System architecture for decision support will also be developed, which is intended to be flexible, responsive and robust.

IV. METHODOLOGY

This research will require a multidisciplinary approach, comprising a variety of methodologies. Due to the evolving nature of a fast, highly competitive industrial environment, the overarching methodology employed will be considered as 'learning by doing', or 'agile', however more specific methodologies will be employed during various phases of this research.

Many of the systems and processes are implemented across several operating entities, and are required for the monitoring and measurement of performance indicators, and therefore, constitute to an essential element in the study. The focus of this element is on the problem of aggregation of dispersed data into a central knowledge information system. This first step will rely on the use of the following methodologies:

- Systems Engineering,
- Data Modeling, and
- Hybrid OnLine Analytical Processing (HOLAP)

A second important step of the research is to define formalized models for simulation, incorporating various elements such as the penalties and incentives. As the research is conducted in partnership with Eurocopter, the tools utilized for any modeling, simulation, analysis and optimization are already pre-determined; additionally the methodology developed, must be compatible with existing technologies currently employed amongst the company.

The third phase of work will be concerned with the analysis of the resulting model, so validation and feedback from an aeronautic perspective can be obtained. For this part, a study will be based on Systems Engineering methodologies, but also include comparative analysis and case studies with existing platforms offered at Eurocopter.

Potential benefits of the implementation of our proposed methodology are suggested:

- Possibility for suppliers to deliver and manage several types of performance services based on defined availability,
- Possibility for both the supplier and customer to have a system providing traceability, monitoring and optimization of service delivery, and
- Find the best compromise between technical performance as expected by the customer, and the commercial supply of the industrial supplier.

V. PRELIMINARY RESULTS

Following the proposed methodology, two initial models were developed in one tool each, Microsoft Excel, and Systecon's SIMLOX. Let us describe the Excel model as "Model A", and the SIMLOX model as "Model B". Model A was developed as a statistical based model, focusing on the breakdown of measurable time under which the aircraft is either considered available, or unavailable. Model B on the other hand, is by nature of the tool, a Discrete Event Simulation (DES), and focuses on the differing states the aircraft can be considered under each moment in time, which can be measured in Hours, Days, Months or Years.

To gain an understanding of the impact of decisions during the development of Model A, a sensitivity analysis was performed on the main parameters, and added to the functionality of the model. Selected initial results in Figs. 1 - 3 show the sensitivity these three input parameters have on model output parameter, Availability. The input data originates from a hypothetical emergency services operational environment for a fleet of 18 helicopters, covering logistics inputs (Fig. 1), unscheduled events (Fig. 2) and additional helicopters (Fig. 3).

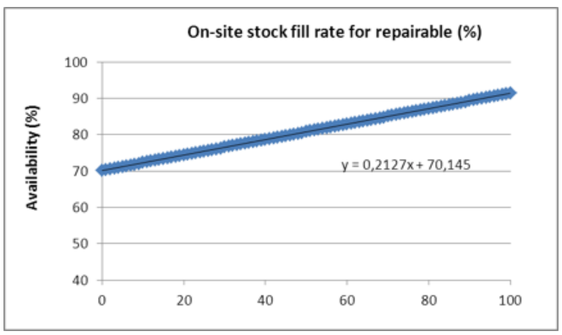

Figure 1. Sensitivity of the Global Fill Rate for Repairable

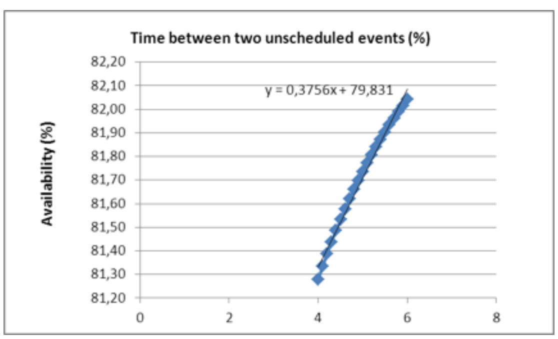

Figure 2. Sensitivity of the Time Between Unscheduled Events

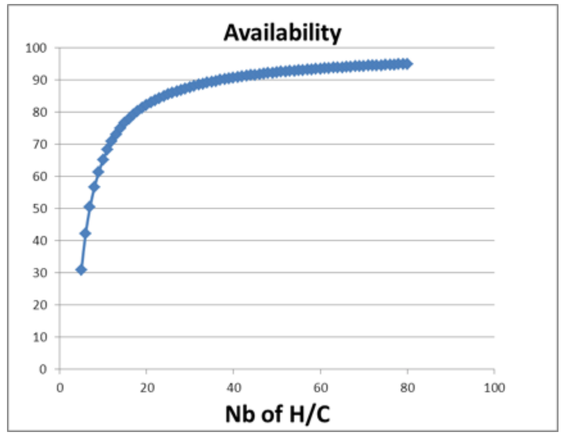

Figure 3. Sensitivity of the Availability

The initial results from the current development on Model A demonstrate how the model can allow an expert user to quickly ascertain sensitivity, and therefore identify the key drivers for the development of Fleet Availability Services.

VI. CONCLUSIONS AND PERSPECTIVES

This is a preliminary work and further development and analysis is required on penalty estimation and the impact based on the sensitivity of key parameters in the model. The efforts must also be replicated in Model B, however assumed to be more complex due to the nature of the SIMLOX tool. We anticipate the ability to import/export raw data for this step, therefore an interface and sub-tool must be developed for the sensitivity and impact analysis of Model B.

Once these steps have been formalized, and the models validated with Eurocopter, the new tools and methodology will be tested with chosen real world case studies.

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Authoritative Authors Mining within Web Discussion Forums

Kristína Machová, Michal Sendek
Dept. of Cybernetics and Artificial Intelligence
Technical University of Košice
Košice, Slovakia
kristina.machova@tuke.sk, michal.sendek@student.tuke.sk

Abstract—**The paper is focused on authoritative users of some web discussion searching and their authority measure estimation. The paper describes design of a method for authority calculation for all discussants of some selected web discussion forum. The designed method can be used in the process of web authority mining. This method involves both the conversational content mining and the conversational structure mining. The resulting implementation can be used for simulation of a dynamic change of the authority measure of social web users. The implementation of the presented method can be used by some firm or other organization for searching authorities (experts) in a special field, in the case, when the organization needs some experts – employees from this field and so such implementation can be used for preselecting on some position in the organization.**

Keywords-authority identification; web mining; conversational content; conversational structure; discussion forums

I. INTRODUCTION

Nowadays, the Web has become the phenomenon of our age. Mainly the social web and its platforms (chats, discussion forums, blogs and so on) enable interactions between actors. The actor, within this paper, is a web user, who has added one or more contributions to a given web discussion and so participated in the conversational content creation. Such web user is also called a contributor within this paper. Within these interactions, web users (contributors) communicate and influence each other. This communication creates so called conversational content, which is an important source of large-scale databases of information about knowledge, opinions, and attitudes of particular users. These data offer many possibilities for web mining from *conversational content*, *conversional structure* and from *conversional web usage*. The social web mining can be focused on different analysis tasks:

- Social networks analysis
- Opinion analysis
- Safety issues analysis
- Authority analysis

The social net analysis is usually focused on some social network structure analysis from the point of view of searching suitable metrics for dynamic analysis, prediction of changes in social networks, developing algorithms for social networks monitoring, dynamic visualization of social networks, research of social networks and their time characteristics. The social net analysis represents mining from conversional structure.

The opinion analysis is concerned with the classification of some web discussion to positive or negative opinion in an automatic way. The summarizing information about positivity or negativity of major part of users opinions about some object (a product, person, event, organization, etc.) can be obtained without necessity to read all discussions and can be useful in the course of a decision making process. The opinion classification represents mining in conversational content.

The safety issues analysis comprises two subtasks: theme modelling and authorship identification. Conversational content can be a source of some information connected with safety issues, for example suspicious activities and identification of authors of the conversation about these subtle themes.

The authority analysis represents the identification of authorities of some web discussions. The contribution of the paper is the design of a method for the authority identification using web mining from a conversational content and also from a conversional structure. There are no known approaches to authority identification from conversational content, which would be compared with our solution. The most known approaches determine the authority degree only from the conversation structure [1][3][9]. Our approach is not based on Natural Language processing (NLP) and it is not based on some known information retrieval algorithm as well.

The contributions of the paper are the following: at first the list of authorities of a given web discussion, ordered according to their estimated measure of authority, at second simulation of dynamic changes of the measure of these authoritative web users. Authority identification can be used in various real situations. For example some web user searches for some authority, which is able to give him/her some advice or information for decision making support. Another example – some Information Technology (IT) organization needs specialists - authorities in the given fields and is able to offer them an interesting position. The paper is related to the conference topics, especially to the track “Complex systems – Simulation and data mining” because: at first, the resulting implementation can be used for simulation of a dynamic change of the authority measure of social web users and at second, the designed method can be used in the process of data mining, especially for web authority mining.

Section II discusses various types of web authorities. The analysis of various kinds of discussion forum contributors is introduced in Section III. Our design of the approach to authority estimation is described within Section IV and next Section V presents the refined model of authority estimation.

Section VI is concerned with dynamic changes of the estimated authority of related social web users.

II. AUTHORITY ANALYSIS TYPES

An authority is reputable competence of some person, society or organization to affect somebody. Authoritative opinions and attitudes have been approved in real situations. The authority can be informal and formal.

The *informal (natural) authority* is a person with naturally authoritative behaviour. Other persons are willing to respect informal authority. Such authority is the result of a personal profile, capability, adequate self-confidence and social activities, which ensure the status for a person – authority. In the field of authorities, a mechanism of associating dynamics and psychology can be found. The people, who let an authority to lead them, enforce the weight of this authority.

The *formal (functional) authority* is a person, which other persons do not want to respect, but they have to. Such authority is the result of a position, title or function of some person within an organization (an arbiter, teacher, politician and so on). The formal authority can be at the same time the informal one as well. The formal authority can sometimes change his/her status. A leader could require submission, although his authority is missing honesty, braveness, predictability and ability of quick decision making.

Different problems can be formulated in the field of authority searching within the Internet. We can search for friend authority, for influence authority or for authority within a given web discussion or a given social network. The *friend authority* is a user with great number of relationships with other users of the Web. The *influencer authority* is a person, who impresses others because of his/her opinions and knowledge on some subject. Our attention was after all concentrated on the problem of searching for *authorities within web discussions*. We took into account the number of relationships - communications between web discussion contributors (mining in the structure of web) as well as the strength of influence in the form of estimation of opinion impression – influencer authority (mining in the content of web).

A. Searching for Authorities within Given Field of Research

The problem of authorities searching was solved by a method for searching scientific papers of a given research field within the Association for Computing Machinery Digital Library (ACM-DL) and Institute of Electrical and Electronics Engineers (IEEE) database. Consequently, we have found authorities between citations presented in the reference part of the searched papers. The Tag Cloud method was used for our results visualization. This approach can be used also for authoritative sources searching. This approach has been implemented and the application was named “Tag Cloud Authority” (TCA). The expected input of the system is a key word or a key phrase characterizing the given research field. The system searches for relevant documents within the ACM-DL and IEEE databases. All authors who have been mentioned in the reference part of each selected paper are considered. The authority degree of an author is increased if he/she is the first author mentioned in the processed reference. The complication is variability of citation standards. Many institutions create their own standards or let authors to choose the form of references.

The orientation on the first author simplifies the processing of selected documents and it has the following interpretation. The first author has usually the greatest share (portion) on the paper creation so he/she is the highest authority from the co-authors. The authority degree of a particular author represents the number of his/her publications citations related to the given research field. More information about this approach can be found in [4].

B. Searching for Authorities within Web Discussions

But, the main attention of this paper is on the analysis of authorities within web discussion forums. Each social web user can establish the discussion on some theme which is interesting for him/her. Other actors can add their contributions to this theme. Many people can adjoin this discussion but not all of them are experts in the discussed field. It is very important to let us be influenced only by authoritative contributors of the discussion forum. Thus, the recognition of authorities within the contributors is a matter of principle.

III. WEB DISCUSSION FORUMS

Users of the Internet can play roles of producers of web content within various platforms of the social web. The attention of this paper is focused on discussion forums. Discussion forum represents the area on a web page, which is created by the given page owner interactively. In the case the discussion on some theme is established (see Figure 1), other users can express their opinions within their contributions to the given theme.

These contributions (“Cont 1”, … , “Cont n” in Figure 1) create the discussion forum, which can be represented by a graph – an acyclic tree (in the right part of Figure 1).

People have various reasons for contributing to some discussion forum. Great majority of contributors are people, who want to find answers on their questions or want to obtain informed advices from more experienced people for decision making. They expect truthful information. These contributors create a core of discussion but they are not very authoritative ones, and so not very interesting for us.

A smaller group of contributors are actors, for whom the discussion is the opportunity to express their knowledge, to ensure about truth of their ideas or to revise their opinions. These users access to the discussion seriously, add only truthful information, and join discussion only when they are acquainted with the topic. They are really authoritative contributors and they are interested to be distinguished from the other actors. Therefore, an approach for these contributors identification was designed. For these identified authorities, a measure of their authority should be estimated. This authority value should be represented by a numerical value. Design of this numerical value - estimation of the authority is presented in the Section IV.

The last group of contributors is the group of troublemaking actors. They are provocateurs, who are not reputable and they only seek for an opportunity to present their opinions on web discussions. They often contribute not truthful information, invoke conflicts and they want to present their significance.

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They usually degrade all web discussions. These actors are not authoritative. They should be eliminated from the discussions.

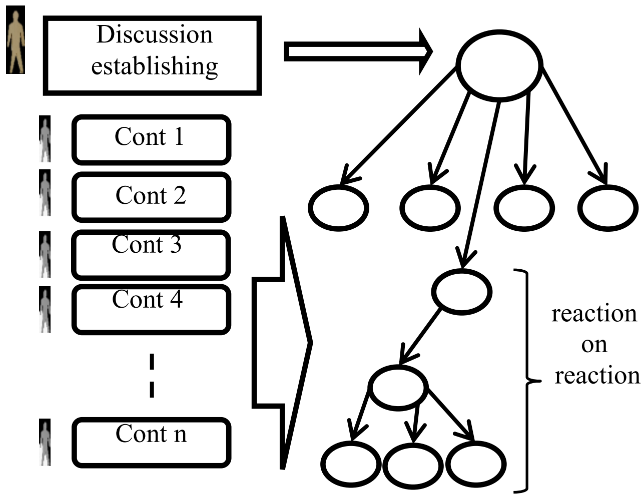

Figure 1. Tree representation of a discussion forum. ('Cont' represents a contribution. Arows lead from a contribution to all reactions on this contribution.)

The elimination of unsuitable contributions can be provided in the web discussion control. There are two ways for discussion control: manual and semiautomatic. *Manual* discussion control is made by a moderator. The moderator checks each new contribution from the point of reliability for the given discussion, law-breaking, whether it is not abusive and so on. Only suitable contributions are consequently added to the web discussion. This kind of discussion control is time consuming and difficult mainly in the case of discussions with great amount of contributions. Thus, semiautomatic discussion control was designed. It uses programs, which filter unsuitable contributions with the aid of words recognition. All contributions, which are denoted by such program as faux or abusive, are redirected to the moderator to decide about deletion of the contribution from the given discussion. There is another way for this problem solving. The moderator can enable actors to denote improper and abusive contributions by a special mark and this marking causes an automatic redirection of the contribution to the moderator for evaluation. It enables to integrate all actors of the web discussion into the process of discussion control. The combination of these three approaches creates so called the three level discussion control. It can ensure nearly zero occurrences of improper contributions. Faux and abusive contributions are forbidden because they can damage good reputation of some firm or organization. But, negative and serious contributions are accepted.

IV. Design of the Authority Estimation

The controlled web discussions can be used for opinion mining and authorities mining. The results of opinion mining can be a part of information, which is necessary for authority estimation.

A. Opinion Mining

The Opinion mining was used for acquisition of opinion polarity of all contributions of the web discussion. These opinions polarities were used for one parameter of a function of authority estimation named "polarity matching".

A web discussion carries a lot of information, for example various themes, opinions and attitudes concerning to various objects (a product, political situation, book, film, physician and so on). Actors who have created some discussion but also other actors who simply have the similar problem as the given discussion is about, want to know the whole opinion of all contributors to the given theme. If the discussion consists from a great number of contributions, reading the whole discussion is time consuming process. There arises the need of automatic classification of the web discussion to positive or negative opinion. This classification has to be based on classification of each particular contribution to positive or negative opinion. More about opinion mining can be seen in [2][7][8]. Opinion classification can be used in those fields where the aggregation of a large amount of opinions into integrated information is needed. The input to opinion classification can be represented by a large amount of conversational content (e.g., content of a discussion forum, blog, chat and so on) and the output of the classification is summary information about opinion polarity. It was solved also within our previous works [5]. This aspect of conversational mining – opinion mining of particular contributions – was used in our design and application of authority estimation (authority mining – Subsection IV.B) in the form of "polarity matching" parameter (Subsection IV.C).

B. Authority Mining

The search for authoritative web users within some web discussion represents mining within data about the web discussion as a whole. We are interested not only in contributions' content but also in the structure of the whole discussion.

The authority is related to contributors not to contributions. Thus, from the beginning it is necessary to collect all data about one contributor together. This process is illustrated in Figure 2. It can be seen in this figure, that all information (selected values) about the "Author A" is completed within first item of all selected values repository.

The authority value estimation was designed as a function of particular selected parameters. These parameters were chosen, because they have influence to authority identification. This function was designed to be composed from two parts: primary part and secondary part.

The result of the process of authority estimation of all discussion actors is the rating of authorities ordered in a descending way, which should indicate:

- contributors who showed the best knowledge concerning to the given theme,
- contributors who invoked most reactions,
- contributors who initialized diversion from the given theme most often.

Testing of this approach was oriented on web discussions, which contain more than 30 contributions, their contributions are not too short, which have more than one level of reactions, their discussion is controlled and they support creation of discussion trees. The implementation of this approach has been designed in such way, which enables using it for any theme. Thus, two different themes in two different web discussions were chosen for testing. The first one was a technically oriented discussion about Windows7 and the second one was a discussion about TeleVision (TV) serial “Neighbours”.

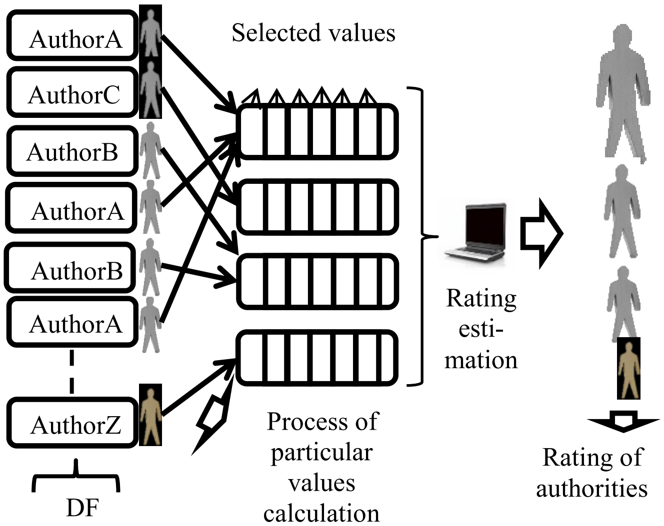

Figure 2. The process of authority estimation from the discussion forum (DF). The human-shape figures represent rated authorities.

C. Design of the Function for Authority Estimation

Our design of the function for authority calculation has the following parameters:

- *Number of discussion contributions (NC)* of the given contributor. It seems, that somebody who understands the theme (authority) will contribute to the discussion more often than other actors. A specific group of contributors are users, who are not so knowledgeable but they use to join discussions to put questions and to learn from answers. They are valuable contributors, because they shift the discussion on higher levels.
- *Number of reactions* (NR) on the contribution(s) of the given discussant. This parameter represents the number of reactions which support or negate a statement of the user, whose authority is examined. It is assumed, that more authoritative contributor could evoke higher number of responses and more reactions.
- *Number of occurrences on the bottom level (NBL)* of the discussion tree. The contributor, whose contributions are located on the bottom level of the discussion tree, usually has added the exhaustive commentary which answered all questions. This kind of contributors can be authoritative.
- *Polarity matching (PM)*. Within this parameter, the whole polarity of all contributions of the given actor is compared with the polarity of the whole web discussion. The polarity of particular contributions was determined using the Opinion Classification Application (OCA) [6]. This application uses the highest degree of positive polarity equal to 3 and the most negative contribution is marked by -3 degree. The greatest difference between the polarity of all users contributions and the polarity of the whole discussion can be 6. The polarity matching in the form of value from the interval <0, 3> is interpreted as an agreement between polarity of discussant’s opinions and opinion of the whole discussion. Similarly, polarity matching in the form of value from the interval (3, 6> is interpreted as a disagreement. Greater authority should be assigned to the users, polarity of whose contributions agrees with polarity of the whole discussion. The less authority should be assigned to actors with significant opinion disagreement between their opinions and overall discussion opinion.
- *Position in the discussion tree (PT)* expresses the number of all levels of the discussion tree the contributions of the user are situated in. Each level is considered only once regardless the number of contributions on this level. Exactly, PT is the ratio of this number and the number of maximal possible occurrences of contributions in the discussion tree.
- *Words number (WN)* represents ratio of all words within all discussant’s contributions to all words of the whole discussion.

All these parameters, taking separately, indicate rather chatty contributors than authoritative ones. But, taking them together as one entity, the emergency phenomenon arises. This phenomenon can indicate the authoritative contributors.

The first three parameters (number of contributions, number of reactions and number of occurrences on the bottom level) create the primary part of actor’s authority value, which influences final authority value in a significant way. The primary part has greater weight in the process of authority estimation. The last three parameters (polarity matching, position in the tree and words number) create the secondary part of actor’s authority value. The secondary part serves on precise tuning of the authority value of very similar authorities or is used for refining this value to prevent the case, when the primary parts of the authority value of different actors are the same.

We have experimented with a function for authority estimation in the form of a simple sum of all selected parameters. But, the final precision was very low (from 10% to 20%). The precision was counted as a ratio of the number of all correctly stated authorities (by our application in comparison with an expert opinion) to the number of all recognized authorities by our application (including these authors, which the expert do not consider as authorities). Equation (1) represents the design of the function for estimation of the Authority of the Contributor (AC).

$$AC = 4NC + 2NR + 4NBL + PM + PT + WN \qquad (1)$$

The greater weight was connected with those parameters, which appeared more significant during the testing phase. In this case, the testing precision was higher (54.8% for technical domain and 52.6% for real life domain) but not overly satisfying. So this function was refined to the following one (2):

$$AC = 4NC^2 + 2NR^3 + 4NBL + PM + PT + WN \qquad (2)$$

In this case, the testing precision was more satisfying (77.4 % in technical domain and 80.7 % in real life domain). Testing was provided in technical and life domains. Technical domain was represented by web discussions on the theme Windows 7 (http://www.verejnadiskusia.sk, http://www.kulman.sk and http://www.warxtreme.eu). Life domain was represented by web discussion on the TV serial story "Neighbours" (http://www.warxtreme.eu).

This form has been implemented. The implementation provides final rating of authoritative users, what is illustrated in Figure 3. This rating is situated in the left dialog window. This window obtains names of web discussion actors. These names are followed by numbers, which represent their authority values.

V. Refined Model of the Authority Estimation

The refined version of the function for authority calculation has been developed. Within this new version logarithmic functions were used. In addition, two new parameters were involved into our method of authority estimation:

- *Number of reactions of the contributor* (NRC) represents the number of reactions of a given contributor on contribution(s) of other discussants.
- *Frequency (F)* represents the number of the given contributor reactions within a time period.

The result was a modified model of the contributor authority estimation (3):

$$AC = 5(1+\log_{10}(NC))+13(1+\log_{10}(NR))+15(1+\log_{10}(NRC)) +(1+\log_{10}(NBL))+3(F+PT+WN) \quad (3)$$

This modified model of the authority estimation has been tested on three various discussion forums related to the following themes:

- The TV "PLUS" discusses about moderators' authorities according to the number of their "likes" on the Facebook.
- The presentation of a well known Slovak politician has been accepted inconsistently.
- Rockets, air attacks and sirens. Near east region drifts toward conflicts.

The results of these tests are presented in table 1.

VI. Dynamic Change of the Authority

Within our research, the dynamic change of the authority value was considered as well. It is an important aspect, related to authority identification. Somebody, who is searching for authorities in some given field, wants to know a dynamic change of the estimated authority value of candidates on authority in recent time.

A natural tendency to increase or decrease the strength of the estimated authority according to activities of the given user – discussant within a web discussion forum has to be considered. Any user of our application would like to have actual information about contributors to the given discussion forum.

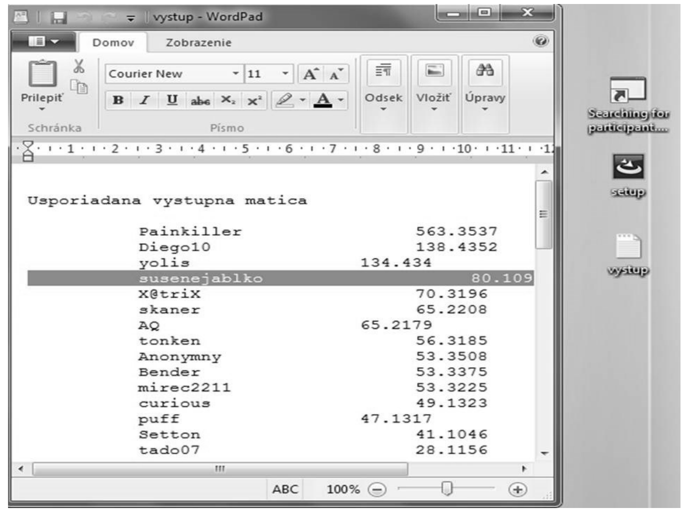

Figure 3. The implementation of the designed method of authority estimation with the resulting rating of authoritative actors.

TABLE I. Results of the Authority Estimation Testing

Theme of the Discussion Forum	***Precision***
Authority and the number of "likes"	0.94
Slovak politician	0.96
Rockets, air attacts and sirens	0.93

The approach to dynamic authority (DA) determination can be modelled by (4):

$$DA = (AC - D)T*P \quad (4)$$

Where:

AC is authority of the contributor according to (3)
D is dynamic change according to (5)
T is time characteristics, which represents the value of the percentage of authority change. It represents increasing or decreasing of the authority on the basis of his/her activity within particular day.
P is penalty (P = 1 for the number of banned contributions from interval (-∞;0>, P = 0.5 for the number of such contributions from interval (0;2> and P = 0 for the number of deleted contributions from interval <2; ∞)).

$$D = 1{,}7\sqrt{1 + 2 * \log AND} \quad (5)$$

Where: AND is the average number of days of web discussion continuance.

The visualization of the dynamic authority for 5 the most authoritative contributors is illustrated in Figure 4. You can see, that the authority value of some contributors is rising, but authority of some is degreasing during a period of seven days.

VII. Conclusion and Future Work

The paper introduced the novelty approach to authority estimation of actors of web discussions and implementation of this approach, which provides the resulting rating of authoritative actors. The resulting implementation is able to simulate the dynamic change of the authority measure of social web users. This approach has been implemented in the programming language Java and in the development environment NetBeans IDE. The results of this implementation were valuable, but they were connected with selected domains from a real life and from a technical world. The novelty of this approach is in estimation of an authority on the base of various parameters obtained not only from the structure but also from the content of the conversational content.

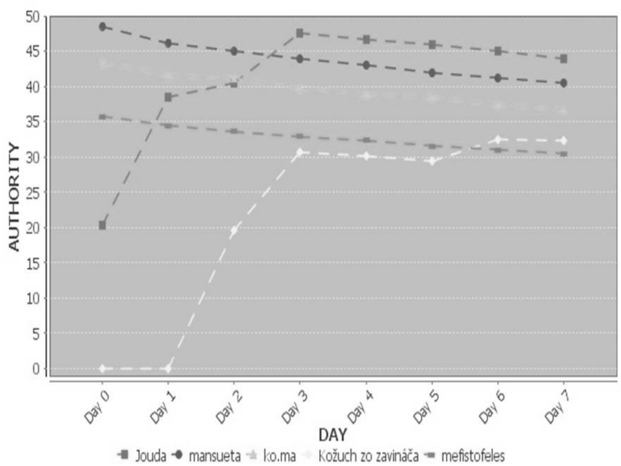

Figure 4. The dynamic authority of the five most significant contributors. Each of them has its own color.

The presented approach can be used for refining the process of opinion classification of some web discussions to positive or negative opinion. Within known approaches to opinion classification of the whole web discussion, the resulting classification to positive (negative) opinion is made when there are more positive (negative) contributions in this discussion and each contribution has the same weight within creation of the resulting polarity. The novelty approach could multiply the positivity value "1" (negativity value "-1") of the given contribution with the weight represented by the estimated authority value of the contributor, who is the author of the given contribution.

The authority identification can be used also by common people, who are interested in some web discussions because of decision making about some purchase, a holiday destination choosing and so on. They can obtain information about authorities and consequently put a question directly to the most authoritative actors of the web discussion. The presented application of authority identification can be used also by organization searching for skilled and valuable employees. The organization can establish some professional discussion and consequently appeal on persons interested in this work position to join the established web discussion concerning to key problems and tasks, this organization has to face up, or to key technologies, this organization uses. Final rating of authoritative actors of this established discussion can serve as the result of preselecting. Or simply, a responsible person of this organization can search for authoritative actors on various web forums, which are focused on technologies and tasks, which are concerned to this organization. Thus, the research in the field of authority identification has big importance for the future.

In future, we would like to refine the model of the authority identification to achieve higher precision. We tend to test our implementation in an extended web space. Also, we would like enrich possibilities of dynamic analysis of the web actor authority change.

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Formal Methods for Comparing Behavior of Procedures in Different Languages

David J. Musliner, Michael W. Boldt, Michael J. S. Pelican, Daniel J. Geschwender
Smart Information Flow Technologies (SIFT)
Minneapolis, USA
email: {dmusliner,mboldt,mpelican}@sift.info

***Abstract*—We are developing software tools to compare the behavior of two executable procedures, written in the same or different languages. These tools are useful in situations such as verifying that an automated procedure does the same thing as its manual backup, ensuring that a change to a procedure impacts only the intended behavior, and verifying that a procedure converted to a new language maintains its behavior. Our newest tool translates each procedure into a common language, links the initial conditions of the procedures together, and uses symbolic execution to explore the behavior of every execution path, comparing the behavior of the two procedures on each path. In this paper, we describe our approach in detail and present a case study of applying our tool to NASA procedures that had previously been automatically translated from the Procedure Representation Language (PRL) to the Timeliner scripting language. Our tool easily scaled to these real-world procedures, and identified several bugs in the prototype translator.**

***Keywords*— procedure equivalence, symbolic execution**

I. Introduction

Like any organization with complex, hazardous equipment, NASA operates manned spacecraft according to rigorously-defined standard operating procedures (SOPs). These procedures carefully define what steps should be taken to accomplish a wide variety of goals, including changing the operating modes of equipment, starting up and shutting down components or subsystems, and conducting various joint machine/crew activities such as spacewalks [1]. The procedures may include sequences of primitive commands that change the state of onboard equipment, tests that verify certain state changes via input telemetry, or branching logic that varies the procedural behavior depending on the state of the spacecraft or other environmental factors. Some procedures can be executed both automatically and manually.

Automatically comparing the behavior two such procedures proves useful in many situations, such as verifying that an automated procedure does the same thing as its manual backup, ensuring that a change to a procedure impacts only the intended behavior, and verifying that a procedure converted to a new language or platform maintains its behavior. Comparing two procedures can be difficult because they may be written in different languages, and they may use different symbols and control structures to represent the same desired behavior.

In this paper, we describe our tool for automatically comparing the behavior of two procedures. We also present a case study applying this tool to verifying the equivalence of NASA procedures that had previously been automatically translated from PRL to the Timeliner scripting language. Our tool easily scaled to these real-world procedures and identified several bugs in the translator.

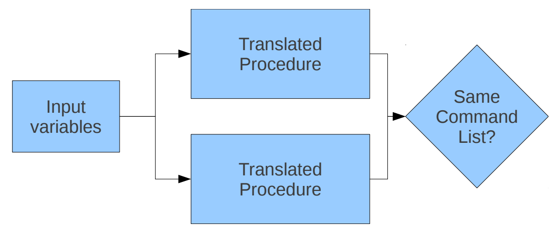

Fig. 1. Procedures are considered equivalent if their command outputs are the same.

II. Approach

A procedure has *inputs*, in the form of telemetry values that the procedure's *control logic* examines to determine what output *commands* should be issued. Our tool must determine whether, given a range of possible inputs, two procedures will issue the same output commands, in the same order. If not, the tool should identify the differences in a compact way.

To verify the functional equivalence of procedures, our approach includes the following steps:

1) Translate each procedure to C to enable the use of symbolic execution tools [2], [3].
2) Combine the translated procedures with driver code that sends them the same inputs and compares their output.
3) Using symbolic execution, repeatedly execute each procedure until all reachable combinations of paths through each procedure are covered, collecting and comparing the sequence of commands from each execution.

A. Translation to C

Procedure languages contain variables, operators, statements, and blocks, just like programming languages. *Steps* or *sequences* in procedures map nicely to functions in programming languages, as they represent parameterized, callable units of execution.

However, some procedure languages contain event-based statements, which trigger when certain conditions are met. For example, a procedure may take an action every 10 seconds, or when a sensor reading exceeds some threshold. Time-based

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triggers can be easily translated to multi-threaded timer-based operations in standard programming languages. One way of dealing with other event triggers is to check for their trigger conditions between each statement of the translated procedure, though this seems inefficient and may explode the verification state space. Our case studies have not included event-based statements, so we have not needed to solve this problem yet.

To identify and compare the commands issued by the procedures, we map each command to a unique integer. We also use this mapping technique to handle command parameters. When a procedure issues a command, we append the corresponding integers for the command and its parameters to a list. This allows us to compare the commands issued by the two procedures by simply comparing the list of integers.

B. *Composition and Execution*

Once translated, the procedures are inserted into a framework that provides identical inputs to each procedure and compares their output, as shown in Figure 1. Then a symbolic execution tool repeatedly runs each procedure with different inputs, ensuring coverage of all combinations of reachable branches between the two procedures.

In symbolic execution, a user identifies a set of variables that he is concerned with and the tool rewrites the target program so that those "symbolic" variables can be controlled and tracked. The tool iteratively executes the instrumented program and updates its internal state with the results of the run. Each time the instrumented program executes, it initializes the symbolic variables with the values provided by the tool. As it executes, the target program records its execution, including the loads, stores, assignments, branches, function calls, and returns that relate to the symbolic (traced) variables. The tool iterates until it executes every code path in the target program or meets other stopping criteria.

In our case, we define a set of symbolic global input variables which can be accessed by both procedures. The symbolic execution engine will run the procedures numerous times, each time following a different execution path and generating output command lists. Comparing these lists effectively compares the behavior of the two procedures. Over the course of the runs, every reachable combination of paths through the procedures is executed, so all potential differences in behavior are considered.

III. Case Study: NASA Procedures

NASA is developing a translator to convert a set of PRL procedures to Timeliner. We created a software tool to verify the translation, using an implementation our approach in Section II. Our tool verifies that the original PRL procedure and the translated Timeliner procedure behave identically, or it identifies how their behavior can differ. Here we describe the implementation details and present the results of this case study, including translation errors identified by our tool.

A. *The Procedure Representation Language (PRL)*

PRL is a still-evolving language designed to capture procedures that may be executed either by automation or by humans [4]. Defined by an XML schema, PRL allows a programmer to construct top-level *procedures* that are decomposed into *steps*, each of which may execute blocks of primitive *instructions* and control statements. Instructions can include spacecraft commands, tests of telemetry values, calls to other procedures, and `wait` instructions that block for some time or until a boolean expression becomes true. Instructions may be specified as manually-executable, or may include *automation data* to help describe how an automatic PRL executive should run the procedures. Automation data can include the expected `StartConditions` that must be true to enable the procedure, `InvariantConditions` that must remain true during execution (or the procedure fails), and `EndConditions` that wait until they are true to allow the procedure to end.

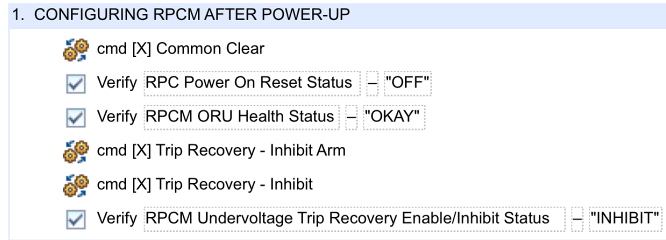

Fig. 2. The PRIDE integrated development environment supports relatively painless PRL editing and visualization.

PRL is being developed to support a gradual transition from fully-manual, textual procedures towards automation. As a result, some elements of a fully-automatic PRL system are not yet defined, including a complete formal semantics for the language and an automatic PRL executive. However, initial steps towards both have been taken in an experimental translation of PRL into the PLEXIL language [4].

An Eclipse-based development environment, PRIDE, has been developed to simplify procedure authoring [5]. Figure 2 shows a fragment of PRL as it appears in the PRIDE environment. This example is one part of a procedure to configure an electrical component by issuing a series of commands and verifying assorted telemetry values.

As illustrated by the example in Figure 3, translating PRL into C is relatively simple. The scope of every block and command is well established, so the control flow is very easy to understand. PRL steps are analogous to functions. Most of the PRL blocks have straightforward conversions to C loops and conditionals. Only the "unordered" block has no direct equivalent. The PRL language contains many types of instructions, though we only encountered `CommandInstruction` and `VerifyInstruction` in this procedure set. A `CommandInstruction` issues a command, which is collected in a list for comparison in our translation. A `VerifyInstruction` is intended to test whether input telemetry shows that the state of the controlled system has reached some expected value. In our translation, we map this to a test of an input value; if the test fails, the procedure will halt.

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```
int a_step1(node **commands) {
  //procedures/example.prl

  push(commands,x_rpcmcommonclearcmdtype,
      "instr11864393570120");

  if (! (x_rpcmpoweronresettype == off))
      { //=instr11864393688691
    return -1;
  }

  if (! (x_rpcmoruhealthtype == okay))
      { //=instr11864393820572
    return -1;
  }

  push(commands,x_rpcmundervoltagetrip
      recoveryinhibitarmcmdtype,
      "instr11864393957873");

  push(commands,x_rpcmundervoltagetrip
      recoveryinhibitfirecmdtype,
      "instr11864393982804");

  if (! (x_rpcmundervoltagerecovery
      inhibittype == inhibit)) {
      //=instr11864394053505
    return -1;
  }

  return 0;
}

node * a() { //=5.420
  //procedures/example.prl
  node *commands = NULL;
  if(a_step1(&commands) < 0) return
      commands;
  return commands;
}
```

Fig. 3. The PRL procedure from Figure 2 translated into C.

B. Timeliner

Timeliner is a suite of tools for building and deploying automated and semi-automated control systems. Timeliner has been utilized in several components of the International Space Satation, including the Command and Control Multiplexer-DeMultiplexer (MDM) and the Payload MDM. Timeliner's Logic Engine executes Timeliner Bundles, which are sets of procedures written in a language that combines fully autonomous and human interactive activities.

Timeliner has English-like code and many time-oriented keywords and control structures. This makes the language more accessible to system specialists without a background in computer programming. A Timeliner procedure is specified within a source file known a *Bundle*. A single Bundle may contain numerous *Sequences* and *Subsequences*, which are both groupings of Timeliner statements to be executed together. Sequences will execute in parallel by default, but can be made to execute serially. Subsequences may be called upon by any sequence to perform some common task. Sequences and even entire Bundles may be started and stopped at any time with the proper commands.

To illustrate some of the constructs of the Timeliner language, Figure 4 shows an approximate Timeliner translation of the partial PRL procedure shown in Figure 2. The `Master`

```
BUNDLE b5_420

  DECLARE bErrFlag BOOLEAN

  SEQUENCE Master ACTIVE

    -- STARTing step1
    START step1
    WHEN step1.SEQSTAT = SEQ_FINISHED THEN
      MESSAGE "Exiting Sequence Master
          successfully."
    END WHEN

  CLOSE SEQUENCE Master

  SEQUENCE step1

    COMMAND X.RpcmCommonClearCmdType

    SET bErrFlag = TRUE
    IF X.RpcmPowerOnResetType = "OFF" THEN
      SET bErrFlag = FALSE
    END IF
    IF bErrFlag = TRUE THEN
      WARNING "ERROR:X.RpcmPowerOnResetType"
      HALT b5_420
    END IF

    SET bErrFlag = TRUE
    IF X.RpcmOruHealthType = "OKAY" THEN
      SET bErrFlag = FALSE
    END IF
    IF bErrFlag = TRUE THEN
      WARNING "ERROR:X.RpcmOruHealthType"
      HALT b5_420
    END IF

    COMMAND X.RpcmUndervoltageTripRecovery
        InhibitArmCmdType

    COMMAND X.RpcmUndervoltageTripRecovery
        InhibitFireCmdType

    SET bErrFlag = TRUE
    IF X.RpcmUndervoltageRecoveryInhibit
        Type = "INHIBIT" THEN
      SET bErrFlag = FALSE
    END IF
    IF bErrFlag = TRUE THEN
      WARNING "ERROR:X.RpcmUndervoltage
          RecoveryInhibitType"
      HALT b5_420
    END IF

  CLOSE SEQUENCE step1

CLOSE BUNDLE b5_420
```

Fig. 4. The PRL procedure from Figure 2 translated into a Timeliner bundle.

sequence in our Timeliner encoding is the first sequence to execute and is responsible for beginning other sequences. In this example the `Master` sequence executes the sequence `step1`. Within `step1` are the four instructions present in the original PRL procedure. PRL's `CommandInstruction` is easily represented by a Timeliner `COMMAND` statement. PRL's `VerifyInstruction` is represented by a test of an input value; if the test fails, the translation sets an error flag and halts execution.

Generating C to represent Timeliner is generally straightforward. For example, Figure 5 shows the translation of the Timeliner procedure in Figure 4. Expressions are similar to C expressions, so they require little processing. Most control

```
int b_step1(node **commands) {
  //procedures/example.tls

  push(commands,x_rpcmcommonclearcmdtype,
      "23");

  berrflag = 1;
  if (x_rpcmpoweronresettype == off) { //=29
    berrflag = 0;
  }
  if (berrflag == 1) { //=32
    fprintf(stderr, "warning: error:x.rpcm
        poweronresettype\n");
    return -1;
  }

  berrflag = 1;
  if (x_rpcmoruhealthtype == okay) { //=41
    berrflag = 0;
  }
  if (berrflag == 1) { //=44
    fprintf(stderr, "warning: error:x.rpcm
        oruhealthtype\n");
    return -1;
  }

  push(commands,x_rpcmundervoltagetrip
      recoveryinhibitarmcmdtype,"52");

  push(commands,x_rpcmundervoltagetrip
      recoveryinhibitfirecmdtype,"57");

  berrflag = 1;
  if (x_rpcmundervoltagerecoveryinhibit
      type == inhibit) { //=64
    berrflag = 0;
  }
  if (berrflag == 1) { //=67
    fprintf(stderr, "warning: error:x.
        rpcmundervoltagerecoveryinhibit
        type\n");
    return -1;
  }
  return 0;
}

int b_master(node **commands) {
  //procedures/example.tls
  if (b_step1(commands) < 0) return -1;
  fprintf(stderr, "message: exiting sequence
      master successfully.\n");
  return 0;
}

node * b() { //=b5_420
  //procedures/example.tls
  node *commands = NULL;
  if(b_master(&commands) < 0) return
      commands;
  return commands;
}
```

Fig. 5. The Timeliner procedure from Figure 4 translated into C.

statements and instructions present no translation problems. However, there are a handful of language features that could be difficult to translate if they were encountered.

One such feature is the ability of Timeliner sequences to run in parallel with each other. A Timeliner bundle will generally have one master sequence that can execute other sequences. In addition to the possibility of issuing interleaved commands, parallel sequence execution also allows for "contingency" sequences that constantly monitor a condition in order to respond with a series of commands. These constructs could be problematic to translate but they were not encountered in our case study. The case study procedures run all of their sequences serially.

Timeliner also includes several time-based control statements. Our current procedure comparison only compares the *sequence* of commands, maintaining no temporal information. If this timing is considered an important part of the procedure execution, then the comparison may be inaccurate, as this timing will be lost.

C. CREST

We use the CREST concolic execution tool to verify all execution paths without having to exhaustively check every input value. CREST is referred to as a "concolic" execution tool because it executes code both concretely and symbolically. CREST uses the C Intermediate Language (CIL) [6] to instrument a target program to simultaneously perform symbolic and concrete execution. To use CREST, a user identifies a set of variables that he is concerned with and CREST rewrites the target program so that those "symbolic" variables can be controlled and tracked. During each test iteration, CREST writes an inputs file, executes the instrumented program and updates its internal state with the results of the run. Each time the instrumented program executes, it initializes the symbolic variables with the values from the inputs file. If there are not enough inputs or if no inputs are specified, the target program initializes each symbolic variable with a random value. As it executes, the target program records its execution, including the assignments and branches that relate to the symbolic variables. User-specified search modes, such as random or depth first, use this execution information to choose inputs for subsequent test iterations. CREST runs test iterations until it executes every code path in the target program or meets other stopping criteria.

D. Testing Framework

To test procedure equivalence, our tool provides identical inputs to each procedure and compares the sequences of commands they issue. Figure 6 shows the structure of our code surrounding the two procedures. We use globally-defined symbolic variables for to represent the procedure inputs. Since they are symbolic variables, CREST will follow all branches involving them.

CREST executes this code numerous times, each time following a different execution path. Each translated procedure will generate its own command list. The two lists should match exactly along every execution path. If they do not, there is some difference between the two procedures, which the tool reports.

E. Results

We tested our tool on 61 different real-world procedures used by one of NASA's experimental manned space program platforms. The procedures totalled almost 18000 lines of PRL, containing 172 steps, 347 commands, and 394 verify statements. The largest procedure consisted of 28 steps with 102 commands and 136 verification statements The PRL

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```
//Command list functions
struct node {...}
void push(node **head, int invalue,
    const char *location) {...}
int pop(node **head) {...}

//Command enumeration
enum mtype{
  Command1,
  Command2,
  ...}

//Input variable declaration
int Input1;
int Input2;
...

node * a() {
  //Translated code from Procedure A
}

node * b() {
  //Translated code from Procedure B
}

void main(void){
  //specify inputs to be handled
  //symbolically by CREST
  CREST_int(Input1);
  CREST_int(Input2);
  ...

  //run each procedure
  fprintf(stderr, ``Procedure A:\n'');
  node *a_commands = a();
  fprintf(stderr, ``Procedure B:\n'');
  node *b_commands = b();
  if (a_commands == NULL) {
    fprintf(stderr, ``A null!\n'');
  }
  if (b_commands == NULL) {
    fprintf(stderr, ``B null!\n'');
  }

  //loop through the commands until both
  //lists are empty
  while(a_commands!=NULL ||
      b_commands!=NULL) {
    //if the two commands are not the
    //same, the procedures differ and
    //this run is stopped
    int a_cmd = pop(&a_commands);
    int b_cmd = pop(&b_commands);
    fprintf(stderr, ``A: %d\nB: %d\n\n'',
        a_cmd, b_cmd);
    if(a_cmd != b_cmd) {
      fprintf(stderr, ``Failure - '');
      exit(1);
    }
  }
  fprintf(stderr, ``Success - '');
}
```

Fig. 6. The main program runs both procedures and compares the output commands to ensure their equivalence.

was converted automatically by a NASA-sponsored prototype translator into almost 15000 lines of Timeliner script.

Our tool easily scaled to these real-world procedures; testing the entire set takes less than 30 seconds on a modern laptop. Through this analysis, we identified four bugs in the prototype translator.

Two of the bugs were syntactic in nature. First, the translated Timeliner compares operator input to string values, but Timeliner does not accept string input. This was carried over from PRL, which *does* accept string input from the operator. Second, the Timeliner translation failed to declare the variable used to hold the results of some operations.

The remaining two bugs caused subtle behavioral differences between the procedures. The first occurs in procedures where the PRL contains an `IfThen` tag with no `Condition` tag. This is legal PRL syntax; it requires manual execution. The `title` attribute of the `IfThen` tag serves as the human-readable condition query, and an operator must input its value. The prototype PRL-Timeliner translator tool did not correctly take this into account. The Timeliner translation of such a construct has no `if` statement or condition. It does retain the instructions in the PRL `IfThen`, however, executing them even when the condition is false. The structure of this bug is shown in Figure 7.

The other behavioral difference also involves the `IfThen` tag. In PRL, an `IfThenBlock` may contain one or more `IfThen` tags, optionally followed by an `Else` tag. PRL semantics dictate that each `IfThen` and its `Else` are mutually exclusive, like an `if...elseif...else` construct. So, for the translation into Timeliner, the first `IfThen` should become an `IF` statement, and any following `IfThen` tags in the same `IfThenBlock` should become `ELSIF` statements. The prototype PRL-Timeliner translator instead translated all PRL `IfThens` to Timeliner `IF` statements, allowing for more than one to execute in a single run. Additionally, the translator left out the `ELSE` clause, so those statements will always execute. The structure of this bug is shown in Figure 8.

While these translator bugs were easily repaired, they illustrate the value of our approach to automatic comparison of procedure behaviors.

IV. Related Work

This work is related to a wide variety of prior and ongoing research in verification of high-reliability systems including work on performed for NASA's ongoing Automation for Operations (A4O) project [1]. Previous work on verification of procedures for NASA missions has largely focused on verifying the *internal* consistency, safety, and semantics of individual procedures and scripts, rather than comparison between two implementations of a procedure. For example, recent NASA research on verification of procedures written in PRL has addressed static verification to ensure well-formed Program Universal Identifier references, as well as dynamic verification of assertions such as "after the state 'abort plan' is set to true, no node in the plan repeats (loops)" [7]. Similarly, verification methods have been used to ensure static and limited dynamic properties of executable scripts coded in PLEXIL.

The verification of procedures has been explored in other contexts, such as nuclear power plant operation. For example, Zhang [8] has used SPIN model-checking to verify properties of operator procedures (*e.g.*, liveness), and developed an incremental approach for the construction of system models with increasing complexity in order to reduce the cost of finding mistakes. These techniques may be useful for spacecraft procedures when spacecraft models become more available,

```
<prl:IfThenBlock>
  <prl:IfThen title="Condition">
  <!    Action -->
  </prl:IfThen>
</prl:IfThenBlock>
```

(a)

```
Action
ENDIF
```

(b)

```
q = QUERY "Condition"
IF q THEN
  Action
ENDIF
```

(c)

Fig. 7. The conditionless `IfThen` semantics bug found by our tool. (a) Source PRL. (b) Incorrect Timeliner from translator. (c) Correct Timeliner translation.

```
<prl:IfThenBlock>
  <prl:IfThen>
    <prl:Condition>Cond1</prl:Condition>
    <!    Action1 -->
  </prl:IfThen>
  <prl:IfThen>
    <prl:Condition>Cond2</prl:Condition>
    <!    Action2 -->
  </prl:IfThen>
<prl:Else>
    <!    Action3 -->
  </prl:Else>
</prl:IfThenBlock>
```

(a)

```
IF Cond1 THEN
  Action1
ENDIF

IF Cond2 THEN
  Action2
ENDIF

Action3
```

(b)

```
IF Cond1 THEN
  Action1
ELSIF Cond2 THEN
  Action2
ELSE
  Action3
ENDIF
```

(c)

Fig. 8. The multiple-`IfThen` semantics bug found by our tool. (a) Source PRL. (b) Incorrect Timeliner from translator. (c) Correct Timeliner translation.

depending upon the complexity of the models and procedures and the performance of the verification tools.

The program equivalence problem for general programming languages attracts extensive research from the verification community. Program equivalence is formalized as several forms of *bisimulation*. In general, bisimulation refers to the idea that two programs have the same state transition structure. CADP is a popular suite of tools that can analyze abstract programs (formulated as Labelled Transition Systems (LTSs)) and verify complex properties expressed in specifications such as temporal logic or mu-calculus [9]. Given two programs formulated as LTSs (in our case, two procedures from different sources), the CADP bisimulation tool can check to see if the procedures are equivalent, modulo one of several *equivalence relations*. These relations, including strong equivalence, observational equivalence, and safety equivalence, provide different levels of guarantees about how the procedures correspond. High licensing fees prevented us from investigating CADP for our case study.

V. Conclusion and Future Directions

We have presented an approach to comparing the behavior of two procedures by translating them into a common language, linking their inputs, and using symbolic execution to explore all execution paths. We have illustrated the feasibility and usefulness of this analysis through a case study, in which we successfully identified bugs in a prototype language translator operating on real NASA procedures.

However, we have not yet modeled significant aspects of the Timeliner and PRL languages, such as parallel execution of Timeliner sequences and PRL's unordered statement blocks.

Adding more expressive correctness criteria would also increase the utility of our approach for procedure verification. In some cases, requiring an identical sequence of commands is too strict. For example, some commands may be irrelevant or order may not matter.

We are currently completing an integration of our procedure comparison tool into the PRIDE procedure editing environment, so that procedure authors can compare procedures and interact graphically with any identified behavioral differences.

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Out-of-Step Protection System Testing by Means of Communication Network Emulator

Antans Sauhats, Aleksandrs Dolgicers, Andrejs Utans, Dmitrijs Antonovs, Gregory Pashnin
Riga Technical University
Faculty of Power and Electrical Engineering
Riga, Latvia
e-mails: {sauhatas@eef.rtu.lv, dolgicers@eef.rtu.lv, utan@eef.rtu.lv, dantonov@eef.rtu.lv, pashnin@gmail.com}

***Abstract*— This paper presents the testing methodology for out-of-step protection system operation validation. The protection under consideration is a wide-area measurement-based system that consists of several parts: intelligent electronic devices, GPS measurement synchronization and communication network. The entire protection system must be tested in the laboratory before installation on site. The problem is, that communication network is hardly available at the laboratory testing stage and, at the same time, the communication network is a critical part of the system, which directly influences the entire system operation. To overcome this problem, the communication network emulator was elaborated that allows to test the entire protection system in real-time and in the presence of various, potentially vulnerable conditions.**

Keywords-out-of-step regime (OOS) protection; wide-area measurement system; wide-area protection testing; communication network time delay; communication network emulator

I. INTRODUCTION

A modern power system is a very complex structure comprising a huge number of equipments. A power system is a subject to a whole range of disturbances, contingencies and equipment faults that should be eliminated as soon as possible in order to guarantee power system reliable, secure and effective operation [1]. Power system protection's functions are accomplished by means of protection and automation devices. It is possible to subdivide all types of protection and automation systems on two separate groups:

1. Local protection and automation devices, whose main task is to protect only one of the power system objects (generator, transformer, transmission line, substation buses). This type of devices uses only locally obtained measurements (voltages and currents) to accomplish the object protection task;
2. Wide-area protection and automation systems, whose task is to mitigate contingencies, which, if ignored, may lead to power system instability and blackouts. These types of protection systems use the information from several, geographically distant, power system points. High speed communication channels used for real-time information exchange between the local devices comprising the wide-area protection.

Wide-area protection structure can be presented like on Fig. 1. The protection system consists of several elements: Intelligent Electronic Devices (IED) - local devices that measures voltages and currents (U, I), pre-process measurements in vector polar or rectangular form and exchange the information by means of digital communication network [2][6]. Because the physical distance between devices may be too large, the dedicated point-to-point communication channels are not always available. In this case, the private communication network is used for data exchange where switches and multiplexers are access points to the virtual communication circuits. Virtual communication circuits define the logical connection between network clients but the actual data path is defined by the current state of the network and several paths possible for one and the same logical connection.

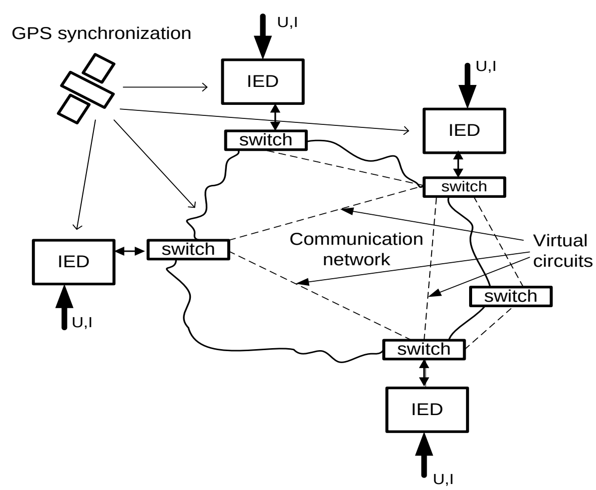

Figure 1. Wide-area protection structure.

To be able to process the measurements of such widespread system, the measurement synchronization should be accomplished. Global Positioning System (GPS) disciplined time sources are used for measurement synchronization. Each IED device receives one pulse per second (1pps) signal from substation GPS receivers [13]. Thus, all measurement are synchronized within several

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microseconds and supplied with appropriate time tags when transmitted through communication network. Unlike the local protection system, the reliability of wide-area protection systems is highly dependent of the performance and reliability of each element, comprising the system. To guarantee the effective and trouble-free operation of the whole system, the extensive testing should be carried out and all possible functional problems should be identified, preferably, at the system design and laboratory testing stage.

The reminder of this paper is organized as follows: Section 2 provides background information about out-of-step regime in power system and describes the out-of-step protection system structure and operation principles. The protection system operation in real-time along with the influence of the communication network time delay is analyzed in Section 3. Section 4 describes the methodology of protection system testing using a communication network emulator. Conclusion is drawn in Section 5.

II. OUT-OF-STEP REGIME IN POWER SYSTEM

A power system is always a subject to small or large disturbances and equipment faults. Local faults, such as short-circuits, are successfully mitigated by means of fast disconnection of the faulted object from the healthy grid. But regimes do exist in power system that can lead to much worse consequences than the local equipment faults because of their influence on the stable operation of the power system. Generally, this type of regimes arises as a result of power generation/load imbalance. One of such hazardous regimes is the Out-of-Step (OOS) regime. When generated power cannot be successfully delivered to the load (because of the transmission line limited capacity or short circuit) or, conversely, there is insufficient power (because of the sudden loss of generation or excessive load), then some part of the system generators start acceleration/deceleration in response to the generation/load imbalance. As a result, part of the power system operates asynchronously (loss synchronism) with the remaining part. Fig. 2 shows typical voltage and active power waveforms (effective values) in OOS regime. The situation may become even worse because of the uncontrolled load shutdown in response to the voltage deeps near the power swing electrical center. To avoid equipment damage and widespread power outages, the OOS protection should take appropriate control actions:

1. Try to restore generation/load balance of the system (add generation capabilities or remove excessive load);
2. In case the first step was unsuccessful, the power system should be split in several parts with a goal to preserve power balance within each peninsula. When the power balance within each part is achieved, the power system restoration should be accomplished by the system operator.

The OOS relaying principles are well-known and described [10][11]. At least, several electrical quantities (measured or derived) can be used for power swing detection: power, currents, impedance and impedance rate of change, power swing center voltage.

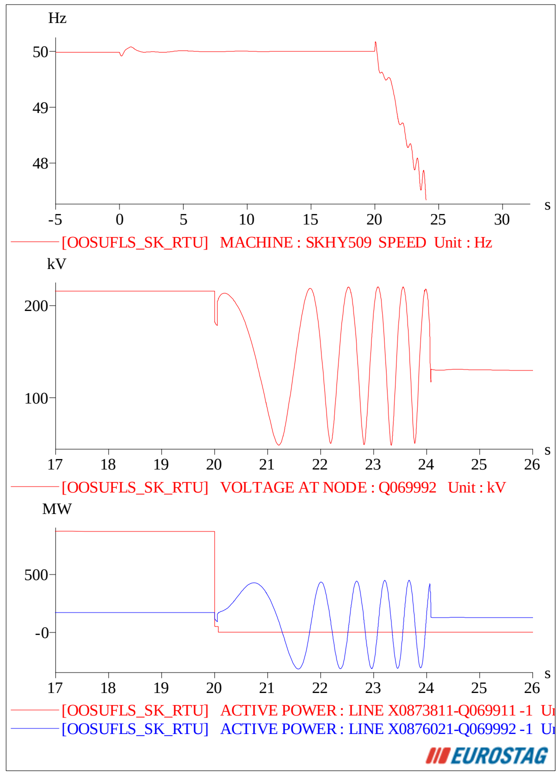
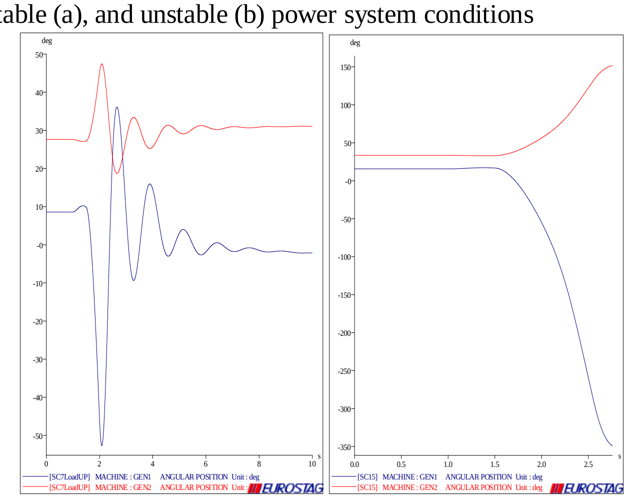


Figure 2. Out-of-step regime waveforms.

The primary reason of the OOS regime is generator (or group of generators) pole slip with respect to the rest of the system. Fig. 3 shows the generators rotors angle variation for stable (a), and unstable (b) power system conditions



(a) (b)

Figure 3. Generators rotor angle variation for stable (a) and unstable (b) power swing.

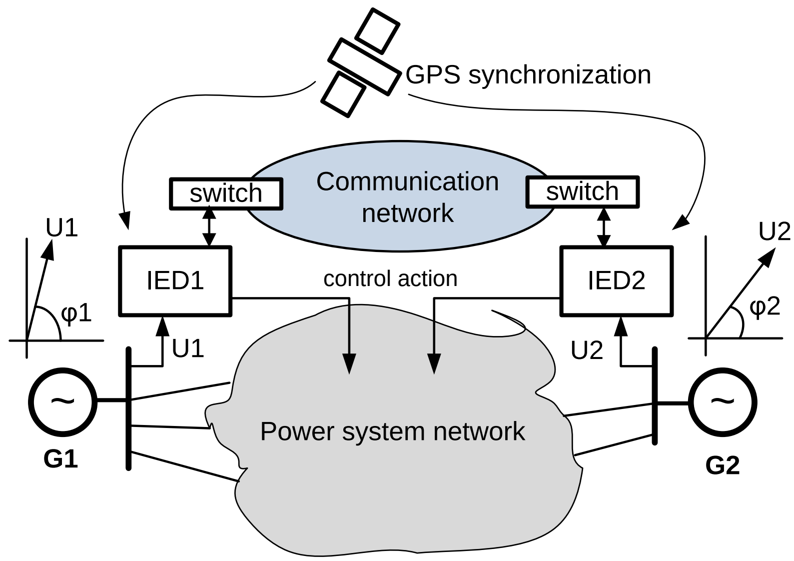


Figure 4. OOS protection system structure.

The generators electromagnetic force (EMF) vectors angle difference value can serve as an indicator of OOS regime. Direct measuring of the generators EMF vectors hardly available, but, the closest available approach is to measure the voltage phasors at the nearest (electrically) nodes. The simplified (with two generation sources G1, G2) OOS protection system structure is shown in Fig. 4. Local devices IED1 and IED2 measure the voltage phasors U1, U2, calculates the voltage phasors angles φ1, φ2 and exchange with information through the communication network. Each IED calculates the angle difference $\Delta\varphi = \varphi1 - \varphi2$ and recognize the OOS regime if three conditions are met:

1. The angle difference exceeds the system stability angle setting (derived from the results of power system regimes simulation):

$$\Delta\varphi > const1 \tag{1}$$

2. The rate of change of the angle difference does not exceed the value of const2 (this condition allows to distinguish OOS regime, when voltage angle changes smoothly, from the short circuit regime, when voltage angle can change abruptly):

$$d(\Delta\varphi)/dt < const2 \tag{2}$$

3. The negative sequence voltage does not exceed const3 setting value (this conditions allows to distinguish the OOS regime, which is three-phase balanced regime, from all others unbalanced regimes):

$$U2 < const3 \tag{3}$$

When Δφ starts approaching to the const1 value, the OOS protection should issue the command for load shedding or launch additional generation resources. If Δφ still increases and exceeds the value of const1, then the command should be generated to split the power system in predetermined place.

III. OOS PROTECTION REAL-TIME OPERATION ISSUE

It should be noted that distributed measurements and control systems are already in use in power system utilities. The Wide-Area Measurement System (WAMS) is an example of such system [4]. WAMS structure is very similar to the one presented in Fig. 1, except that instead of IEDs, the Phasor Measurement Units (PMU) are used across the system. PMUs are placed in critical power system points [2]. Each PMU calculates line frequencies, voltage and current phasors and streams those data over the communication network, along with the associated GPS time tags. Data from PMUs are collected in power system utility dispatch center and can be used to create wide-area visibility across the power system in ways that let grid operators understand real-time conditions, see early evidence of emerging grid problems, and better diagnose, implement and evaluate remedial actions to protect power system stability [3]. Several publications [4][5][7][8] dedicated to PMU real-time application for protection and control tasks. Wide-Area Monitoring, Protection and Control systems (WAMPAC) can cope successfully with relatively slow processes like inter-area oscillations, state estimation, under frequency load shedding, power system restoration after islanding. Typical PMU provides output data at rate 10-50 samples per second (for 50 Hz system) [7]. For the proposed OOS protection system, it is mandatory to trace not only the angle value, the voltage phasor rotation should also be tracked with, at least 5 electrical degree resolution (Fig. 5). This requirement could be fulfilled with a signal sampling rate of 500-1000 samples per second (depending on the implemented algorithm), that significantly exceeds the PMU output data rate.

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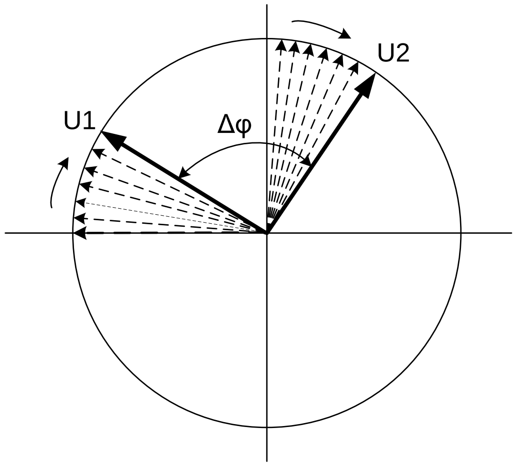

Figure 5. Voltage phasors sampling.

One more factor should be taken into account. WAMS can not be used in the absence of GPS synchronization, but OOS protection should be operable (possibly, with reduced precision) even if the GPS measurement synchronization is not available at the moment. In the absence of synchronization, the data transmission time delay, introduced by the communication network, can be calculated and taken into account before the angle between voltage phasors is calculated. The time delay calculation is based on well-known ping-pong method [12] and the result is valid only if transmitting and receiving time delays are equal (Fig. 6).

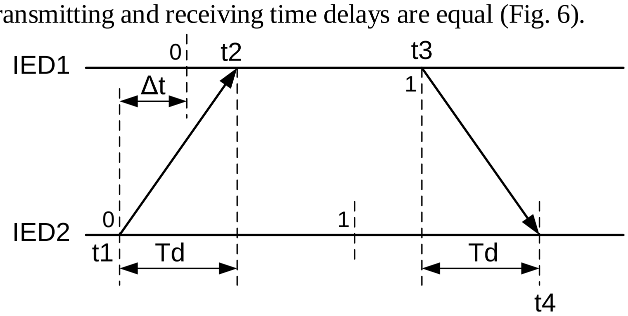

Figure 6. Time delay calculation using ping-pong method.

Then, the measurement synchronization could be achieved in assumption that time delays of the communication network are symmetrical (5). The data transmission time delay *Td* is

$$Td = (t4 - t1 - t3 + t2)/2 \tag{4}$$

Time difference Δt between IED1 and IED2 sampling instances is

$$\Delta t = Td - t2 \tag{5}$$

All time marks: t1, t2, t3 and t4 are local and devices attach these local times to the data frame when data exchange processed. Thus, devices can synchronize their measurements and are operable even without GPS synchronization. Time delay asymmetry will introduce an error in time delay calculations and this error will lead to additional angle error between voltage phasors. To mitigate this effect, IEDs automatically rearranges their settings to less sensitive. Despite the fact that the protection system will operate with lower precision, the system is still capable to detect OOS condition and take appropriate control action. The degree of device settings sensitivity should conform to two requirements:

1. Reliable operation of the protection for the majority of possible OOS regime scenarios;
2. Avoidance of false operation for all possible OOS regime scenarios.

Protection blocking was implemented to fulfill the second requirement - protection will be blocked if communication time delay exceed the maximal theoretically possible time delay for a given communication network. Then, a set of experiments should be carried out to be sure that the first requirement is also fulfilled.

IV. OOS PROTECTION SYSTEM TESTING

Any complex control system should be tested to validate the correctness of the implemented algorithms and to define the value of system operation reliability and robustness. This is especially true for protection and automation systems, whose correct and reliable operation largely determine the entire power system operation. Testing procedures for the local protections can be successfully accomplished in the laboratory, prior the device installation on site. Testing of the OOS protection system under consideration is much more complex task because not only each element of the system should be tested. Interaction of individual parts and correct operation of the entire system in the presence of various, potentially vulnerable conditions, should be checked. OOS protection system consists of three main parts: IED units, GPS synchronization system and digital communication network. It should be mentioned, that validation of the system operation on the site could be problematic because system elements are widespread geographically. Correct functioning of the system should be considered under several conditions: presence/loss of GPS synchronization, short-term unavailability of the communication channel, transmitted data integrity violation, operation of the system with different data transmission rates, communication time delay volatility and time delay asymmetry. The communication network is a critical element of the system and, at the same time, it is hardly available at the laboratory testing stage.

Testing of the considered protection system using simulation and modeling technique will not give us the valuable results because of the several reasons:

1. Inappropriate level of details/unavailability of IEDs models.
2. Communication network topology and hardware environment not always defined at the laboratory testing stage.
3. Each element of the system can be a subject to malfunction due to the hard-to-find programming errors, which can not be simulated.

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4. Protection system under consideration is a real-time application and assumes that each element of the system should operate in real-time.

An emulator of the Digital Communication Network (CNE) was developed and produced to overcome the problems of protection system testing in real-time. In contrast with a typical communication network simulator, CNE is not hosts/data packets oriented device. Also, the device operation principle is not communication protocol-dependent. CNE emulates communication network parameters and states that can potentially affect the protection system performance and reliability. CNE is a dedicated, microprocessor based device, which internal structure and principle of operation presented in Fig. 7. The input data are sampled and stored inside the memory buffer which is organized in a manner of first-in first-out (FIFO) register. The entire contents of the FIFO memory is shifted at one position prior each next sample. The size of the memory buffer N defines the time delay Δt between data input and data output moments (6). The data transmission time delay is

$$\Delta t = N / Fclk \qquad (6)$$

Two dedicated memory buffers per communication channel were implemented to provide full duplex data exchange. To minimize the jitter effect between the input and output signals, the signal sampling frequency *Fclk* should be significantly greater than data transmission rate *Fdata* (7):

$$Fclk >> Fdata \qquad (7)$$

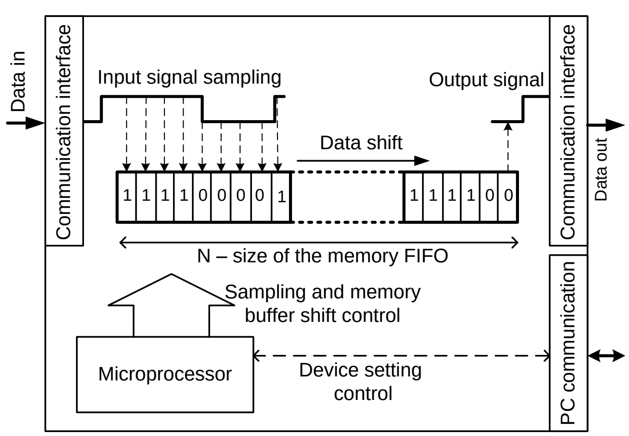

Figure 7. CNE principle of operation.

Device settings are controlled by means of PC with appropriate software. User can control the communication time delays for each communication channel and in both directions independently. One of possible features of the device is operation on a basis of predefined scenario: short-term communication interruptions, introducing random errors in transmitted data, abrupt variation of time delays.

The structure of OOS protection system testing using CNE is presented in Fig. 8. After OOS condition simulation by means of power system regimes modelling software (ATP [14], Eurostag [15], ETAP [16]), the data are uploaded into Relay Test Systems (RTS) and analogue signals can be replayed in real time. At least several tens scenarios of OOS regime should be generated to fulfil the requirement of reliable testing. Then, analogue signals (3-phase voltages) are supplied to each IED and replayed by means of RTS - Freja300.

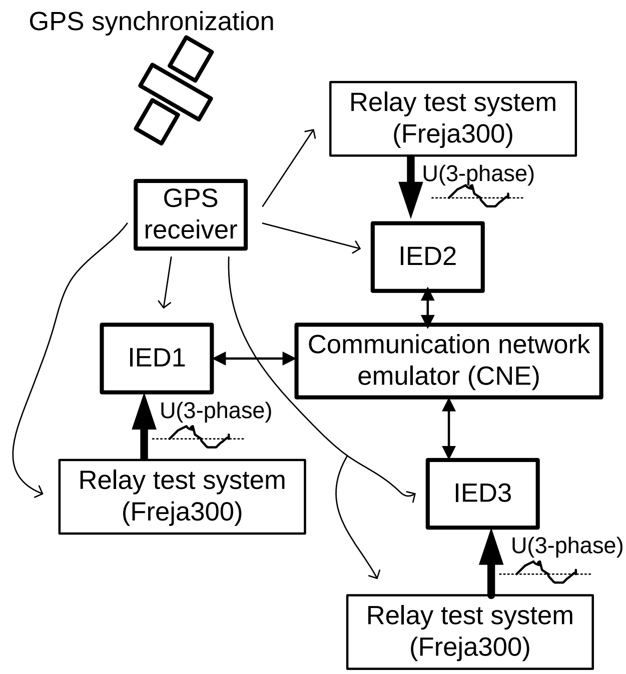

Figure 8. Laboratory testing of the OOS protection system.

IED measurements as well RTS output signals are synchronized by means of GPS receiver. Data exchange between devices is accomplished through the CNE that emulates time delays of the communication network according with predefined scenario. In Fig. 9, we can see an example of one of the experiments.

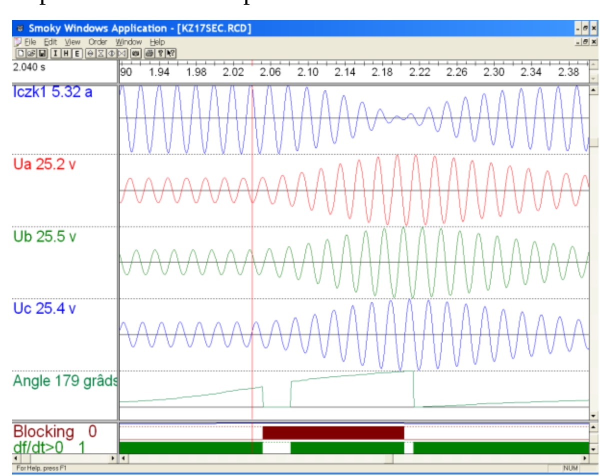

Figure 9. OOS protection system testing example.

CNE abruptly increases the time delay at time 2.05 s and change it back at 2.08 s. Protection operation blocking observable within this time interval in response to time delay variation. This testing structure allows test the whole system in real-time and in close-to-real conditions. After extensive testing, the conclusion is drawn about the validity of protection system settings as also about the entire system operation.

V. Conclusion

A secure and reliable operation of the power system is largely determined by correct operation of the protection relay systems. Appropriate testing of relay systems was always performed to evaluate relay reliability, conformance of the settings as also to validate various algorithmic issues, implemented in relays. Relay testing methodology significantly changes with an introduction of wide-area measurements and wide-area protection systems, because several additional components can influence the entire protection system reliability. The GPS measurement synchronization and communication networks now become the critical links and protection system should be tested for several, potentially vulnerable, conditions. Typically, communication network is hardly available to perform the wide-area protection system testing in laboratory. At the same time, it is possible to define communication network parameters, which directly influence the protection system reliability. The communication network emulator was developed and created to provide the OOS protection system testing in the laboratory.

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A Novel Technique for Retrieving Source Code Duplication

Yoshihisa Udagawa
Computer Science Department, Faculty of Engineering
Tokyo Polytechnic University
Atsugi-city, Kanagawa, Japan
udagawa@cs.t-kougei.ac.jp

Abstract—**In this paper we propose a new approach for the detection of clones in source code for improving safety of software systems. The main contributions of this paper are development of a mining algorithm to explore program structure and the definition of a similarity measure that is tailored to sequentially structured texts for retrieving similar source code fragments. Retrieval experiments were conducted using Apache-Tomcat 7, which is a large-size open source Java program. The results show that the proposed mining algorithm finds a set of frequent sequences within one minute, and the proposed similarity measure is a better indicator than the Sorensen-Dice index.**

Keywords- Java source code, Control statement, Method identifier, Similarity measure, Derived sequence retrieval model, Sorensen-Dice index

I. INTRODUCTION

Reusing source code via copy and pasting rather than rewriting a similar code from scratch is a common activity in software development because it is very easy and can significantly reduce programming effort and time. Hence, software often contains duplicated code, known as a software clone. Previous research shows that between 7% - 23% of source code in a typical software system can be cloned code [1][11].

A software clone may have an adverse impact on the quality, productivity, reusability, and maintainability of a software system [10][12][13]. Tool support is necessary to facilitate code change tasks, because the number of source code may reach several hundred thousand lines for a maintenance engineer based on the author's experience in the industry.

Code clone detection has been actively researched for approximately two decades. Many approaches for identifying similar code fragments have been proposed in the literature. Generally, these techniques can be classified into four main groups, i.e., text-based, token-based, structure-based, and metrics-based.

(1) Text-based approaches

In text-based approaches, the target source program is considered as a sequence of strings. Baker [1] described an approach that identifies all pairs of matching "parameterized" code fragments. Johnson [6] proposed an approach to extract repetitions of text and a matching mechanism using fingerprints on a substring of the source code. Although these methods achieve high performance, they are sensitive to lexical aspects such as formatting and renaming of identifiers, including variables.

(2) Token-based approaches

In the token-based detection approach, the entire source code is transformed into a sequence of tokens and control statements, which is then analyzed to identify duplicate subsequences. A sub-string matching algorithm is generally used to find common subsequences. CCFinder [19] adopts the token-based technique to detect "copy and paste" code clones efficiently. In CCFinder, a similarity metric between two sets of source code files is defined based on the concept of "correspondence." CP-Miner [10] uses a frequent subsequence mining technique to identify a similar sequence of tokenized statements. Token-based approaches are typically more robust against code changes compared to text-based approaches.

(3) Structure-based approaches

In this approach, a program is parsed into an abstract syntax tree (AST) or program dependency graph (PDG). ASTs and PDGs contain structural information about the source code; thus, sophisticated methods can be applied to ASTs and PDGs for clone detection. CloneDR [2] is a pioneer among AST-based clone techniques. Wahler et al. [18] applied frequent itemset data mining techniques to ASTs represented in XML to detect clones with minor changes. DECKARD [5] also employs a tree-based approach in which certain characteristic vectors are computed to approximate the structural information within ASTs in Euclidean space.

Typically, a PDG is defined to contain the control flow and data flow information of a program. An isomorphic subgraph matching algorithm is applied to identify similar subgraphs. Komondoor et al. [7] have also proposed a tool for C programs that identifies clones. They use PDGs and a program slicing technique to find clones. Krinke [9] uses an iterative approach (k-length patch matching) to determine maximal similar subgraphs. Structure-based approaches are generally robust to code changes such as reordered, inserted, and deleted codes. However, they are not scalable to large programs.

(4) Metrics-based approaches

Metrics-based approaches calculate metrics from code fragments and compare these metric vectors rather than directly comparing with the source code. Mayrand et al.[11] proposed several function metrics that are calculated using ASTs for each functional unit of a program. Kontogiannis et al. [8] developed an abstract pattern matching tool to measure similarity between two programs using Markov models. Some common metrics in this approach include a set of software metrics called “fingerprinting” [6], a set of method-level metrics including cyclomatic complexity, and a characteristic vector to approximate the structural information in ASTs.

The proposed approach is classified as a structure-based comparison [15]. It features a sequence of statements as a retrieval condition. We have developed a lexical parser to extract source code structure, including control statements and method identifiers. The extracted structural information is input to an extended Sorensen-Dice model [3][14] and the proposed source code retrieval model, named the “derived sequence retrieval model” (DSRM). The DSRM takes a sequence of statements as a retrieval condition and derives meaningful search conditions from the given sequence. Because a program is composed of a sequence of statements, our retrieval model improves the performance of source code retrieval.

In comparison with our previous paper [15], the main contribution of this paper is the development of a mining algorithm to explore a program’s structure. Without knowledge of the frequency of a sequence of statements, we could not issue a query to the point. The other contribution is a set of experiments using Apache-Tomcat 7 source code, which is considered as large-scale software.

The remainder of this paper is organized as follows. In Section 2, we present a source code pre-process to extract interesting fragments. In Section 3, we present an algorithm for mining program structures and define source code similarity metrics. Experimental results are discussed in Section 4. Section 5 presents conclusions.

II. EXTRACTING SOURCE CODE SEGMENTS

At the beginning of our approach, a set of Java source codes [4] is partitioned into methods. Then, the code matching statements are extracted for each method. The extracted fragments comprise class method signatures, control statements, and method calls.

(1) Class method signatures

Each method in Java is declared in a class. Our parser extracts class method signatures in the following syntax.

<class identifier>::<method signature>

Our parser extracts a method declared in an anonymous class in the following syntax.

<class identifier>:<anonymous class identifier>:
<method signature>

Generic data types are widely used in Java to facilitate the manipulation of data collections. Our parser also extracts generic data types according to Java syntax. For example, List<String> and List<Integer> are extracted and treated as different data types.

(2) Control statements

Our parser also extracts control statements with various levels of nesting. A block is represented by the "{" and "}" symbols. Hence, the number of "{" symbols indicates the number of nesting levels. The following Java keywords for control statements [4] are processed by our parser:

if, else if, else, switch, while, do, for, break, continue, return, throw, synchronized, try, catch, and *finally*.

(3) Method calls

From the assumption that a method call characterizes a program, our parser extracts a method identifier called in a Java program. Generally, the instance method is preceded by a variable whose type refers to a class object to which the method belongs. Our parser traces the type declaration of a variable and translates a variable identifier to its data type or class identifier, i.e.,

<variable>.<method identifier>

is translated into

<data type>.<method identifier>

or

<class identifier>.<method identifier>.

We selected Apache-Tomcat 7.0.42 as our target because Apach-Tomacat [17] is one of the most popular Java web application servers. We estimated the volume of the source code using file metrics. Typical file metrics are as follows:

Number of Java Files	----	1,100
Number of Classes	----	1,681
Number of Methods	----	10,640
Number of Code Lines	----	177,724
Number of Comment Lines	----	108,167
Number of Blank Lines	----	50,344
Number of Total Lines	----	334,457

Apache-Tomcat 7.0.42 consists of 334,457 lines of source code. Relative to the number of lines, Apache-Tomcat 7.0.42 is classified as large-scale software in the IT industry.

III. RETRIEVING SIMILAR SOURCE CODE

A. Code Retrieval Approach

Our experiments consist of two stages: (1) mining structures in the whole extracted program structures; (2) performing retrievals for the mined structures using the DSRM similarity model, which are defined in Subsection III-C.

B. Mining Structures in Source Code

Initially, we mine the structures of source code using the algorithm shown in Figure 1. This algorithm shares many concepts with the well-known *Apriori* algorithm for mining frequent itemsets [16][18]. It takes the minimum support number minSup as an argument, and has its control structures similar to those of the *Apriori* algorithm. However, our algorithm essentially deals with a sequence of statements, while the *Apriori* algorithm deals with a set of items.

The major difference between the two algorithms can be found in the candidate generation process. In the *Apriori* algorithm, new candidate k-itemsets are generated based on the (k−1)-itemsets found in the previous iteration. The order of the items is ignored because the *Apriori* algorithm focuses on finding a set of itemsets that occur in a dataset or transactions having a frequency greater than a given threshold, i.e., minSup.

Our algorithm is designed to find a set of sequences that occur frequently. It should be noted that several matchings can be detected in a sequence for a sub-sequence given as a matching condition. For example, the two matchings of a sub-sequence A→B are detected in a sequence A→B→A→C→A→B→D. The *Retrieve_Cand (MS, T_k)* function shown in Figure 1 finds a set of sequences of (k+1)-statements that includes k-statements found in the previous iteration in method structures extracted from Java source code "MS."

Because most of important methods are invoked in a control structure, the first element of the sequence T_1 is assumed to be a set of the "control statements," i.e., *if, else if, else, switch, while, do, for, break, continue, return, throw, synchronized, try, catch*, and *finally* statements. The assumption is considered as customization for retrieving source code duplication.

Figure 2 shows the number of retrieved sequences and elapsed time in milliseconds for each minSup. The total number of methods is 10,640; thus, for example, minSup 0.0070 corresponds to approximately 75 methods. We measured the elapsed time using the following experimental environment:

CPU: Intel Core i3 540 3.07 GHz
Main memory: 4.00 GB
OS: Windows 7 64 Bit
Programming Language: Visual Basic for Applications

Table I shows the 18 sequences mined with minSup 0.0070, which were used as retrieval conditions of the code similarity retrieval experiments.

```
// Mining statement sequences that begin with a control statement.
List<statement> MS;   // MS is a list of a "sequence" of statements
                      // that are extracted by a lexical parser.
int k;                // k defines the number of repetition
int minSup;           // minSup defines the minimun number of occurences
List<statement> LS;   // LS is a list of a "sequence" of statements
List<statement> Tk;   // Tk is a list of a "sequence" of statements
Map<statement, int> Ck;   // Ck is a map of a "sequence and
                          // the number of occurrences of statements
   LS= Φ;
   k=1;
   Tk= { i | i ∈{Control Statements} };
   repeat
      k=k+1
      Ck= Retrieve_Cand( MS, Tk );  // A set of sequences that begins Tk
      Tk= Φ;
      for ( each c∈Ck.key() ) {
         if ( at least one element of c is not a control statement
           and Ck.value() ≥ N×minSup ) {
              LS.add( c );
              Tk.add( c ) ;
         }
      }
   until Tk= Φ;
   Result= LS;
```

Figure 1. Algorithm for mining frequent sequences

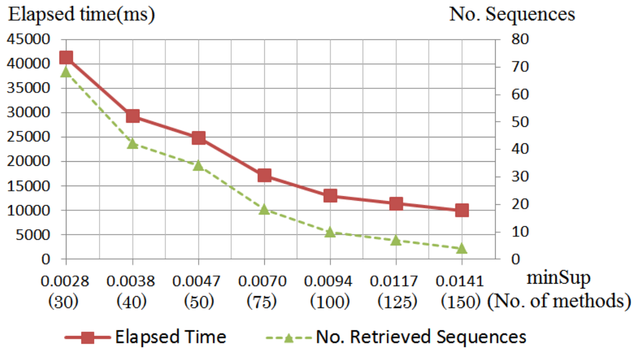

Figure 2. Number of retrieved sequences and elapsed time for each minSup

TABLE I. MINED SEQUENCES USED AS RETRIEVAL CONDITIONS

No	Sequences	Number of occurences
1	if{ → Log.debug	267
2	if{ → Log.debug → }	259
3	if{ → StringBuilder.append	218
4	catch{ → Log.error	150
5	catch{ → ExceptionUtils.handleThrowable	130
6	if{ → IllegalArgumentException	127
7	if{ → IllegalArgumentException → }	127
8	catch{ → Log.error → }	112
9	if{ → org.apache.juli.logging.Log.debug	101
10	if{ → StringBuilder.append → }	100
11	if{ → org.apache.juli.logging.Log.debug → }	99
12	if{ → StringBuilder.append → StringBuilder.append	97
13	if{ → IOException	95
14	if{ → IOException → }	95
15	catch{ → MBeanException	94
16	catch{ → MBeanException → }	94
17	if{ → StringBuilder.append → StringBuilder.append → }	92
18	catch{ → Log.error → } → }	75

C. Extending Sorensen-Dice Index

The Sorensen-Dice index is originally defined by two sets and formulated as follows.

$$\mathrm{Sim}_{\,\text{Sorensen-Dice}}(X_1, X_2) = \frac{2|X_1 \cap X_2|}{2|X_1 \cap X_2| + |X_1 \cap \neg X_2| + |\neg X_1 \cap X_2|} \quad (1)$$

Here, $|X_1 \cap X_2|$ indicates the number of elements in the intersection of sets X_1 and X_2.

In software, the Sorensen-Dice index is known to experimentally produce better results than other indexes such as a simple matching index that counts the number of features absent in both sets [3][14]. The absence of a feature in the two entities does not necessarily indicate similarity in software source code. For example, if two classes do not include the same method it does not mean that the two classes are similar. Our study takes the Sorensen-Dice index as a basis for defining the similarity measure between source codes. The extension of the Sorensen-Dice index on N sets is straightforward and is expressed as follows.

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$$\mathrm{Sim}_{\text{Sorensen-Dice}}(X_1, X_2, \ldots, X_n) = \frac{n \mid X_1 \cap X_2 \ldots \cap X_n \mid}{\sum_{r=0}^{r=n-1} (n-r) \mid SetComb(X_1 \cap X_2 \ldots \cap X_n,\ r) \mid} \qquad (2)$$

The function *SetComb*($X_1 \cap X_2 \cap \ldots \cap X_n$, r) defines intersections of sets $\{X_1, X_2, \ldots, X_n\}$ whose r elements are replaced by elements with the negation symbol. The summation of r = 0 to n−1 of *SetComb*($X_1 \cap X_2 \cap \ldots \cap X_n$, r) generates the power set of sets $X_1, X_2, \ldots, X_n$, excluding the empty set. (n−r) indicates the number of sets without the negation symbol. $|X_1 \cap X_2, \ldots, \cap X_n|$ indicates the number of tuples $<x_1, x_2, \ldots, x_n>$ where $x_1 \in X_1$, $x_2 \in X_2$, …, $x_n \in X_n$.

D. Similarity Metric for Source Codes

In our study, the similarity measure has been tailored to measure the similarity of sequentially structured text. We first define the notion of a sequence. Let S_1 and S_2 be statements extracted by the structure extraction tool. $[S_1 \rightarrow S_2]$ denotes a sequence of S_1 followed by S_2. In general, for a positive integer n, let S_i (i ranges between 1 and n) be a statement. Thus, $[S_1 \rightarrow S_2 \rightarrow \ldots \rightarrow S_n]$ denotes a sequence of n statements.

The similarity of the DSRM can be considered the same as the extended Sorensen-Dice index except for symbols, i.e., using the → symbol in place of the ∩ symbol. The DSRM similarity between two sequences is defined as follows.

$$\mathrm{Sim}_{\mathrm{DSRM}}([S_1 \rightarrow S_2 \rightarrow \cdots \rightarrow S_m], [T_1 \rightarrow T_2 \rightarrow \cdots \rightarrow T_n]) = \frac{n \mid [S_1 \rightarrow S_2 \rightarrow \cdots \rightarrow S_m],\ [T_1 \rightarrow T_2 \rightarrow \cdots \rightarrow T_n] \mid}{\sum_{r=0}^{r=n-1} (n-r) \mid\mid [S_1 \rightarrow S_2 \rightarrow \cdots \rightarrow S_m],\ SqcComb([T_1 \rightarrow T_2 \rightarrow \cdots \rightarrow T_n],\ r) \mid\mid} \qquad (3)$$

Here, without loss of generality, we can assume that $m \geq n$. In the case $m < n$, we replace the sequence $[S_1 \rightarrow S_2 \rightarrow \ldots \rightarrow S_m]$ with $[T_1 \rightarrow T_2 \rightarrow \ldots \rightarrow T_n]$.

The numerator of the definition, i.e., $| [S_1 \rightarrow S_2 \rightarrow \ldots \rightarrow S_m], [T_1 \rightarrow T_2 \rightarrow \ldots \rightarrow T_n] |$ indicates the number of statements in the sequence where $S_{j+1} = T_1$, $S_{j+2} = T_2$, ... , $S_{j+n} = T_n$ for some j ($0 \leq j \leq m-n$). The denominator of the definition indicates the iteration of the sequence match that counts the sequence of statements from r = 0 to n−1. Note that the first sequence $[S_1 \rightarrow S_2 \rightarrow \ldots \rightarrow S_m]$ is renewed when the sequence match succeeds, i.e., replacing the matched statements with a not applicable symbol "n/a." *SqcComb* ($[T_1 \rightarrow T_2 \rightarrow \ldots \rightarrow T_n]$, r) generates a set of sequence combinations by replacing the r ($0 \leq r < n$) statements with the negation of the statements.

A simplified version of the algorithm used for computing the DSRM similarity is shown in Figure 3. It takes a set of method structures M and a sequence $[T_1 \rightarrow T_2 \rightarrow \ldots \rightarrow T_n]$ as arguments, and returns an array of similarity values for the set of method structures.

We assumed that the *getMethodStructure(j)* function returns a structure of the *j*-th method extracted by the structure extraction tool. The function abstracts the implementation of the internal structure of the method, which is represented as a sequence of statements.

The *Count*(MS, TN, R) function returns the number of "positive statements" that matches the (n−R)-combinations of statement sequences TN in the method_structure MS. The *SqcComb*(TN, R) function generates (n−R)-combinations of statement sequences that replace the R statements with the negation of the statements in the sequence TN.

```
/* Datatype "method_structure" is a set of "sequence" of statements.
 *  A "sequence" is represented by an array of statements.
 *     Input: set_of_method_structure M;
 *     Input: sequence [T1→T2→...→Tn];
 *     Output: Sim[M.length];                              */
/* --- Definition of the SimDSRM function --- */
double[] SimDSRM( set_of_method_structure M, sequence [T1→T2→...→Tn])
{
 double Sim[M.length];
 int Nume;
 int Deno;
 for ( int j=0; j < M.length; j++ ) {
    Nume= Count( getMethodStructure(j), [T1→T2→...→Tn], 0 );
    Deno= 0;
    for ( int r=1; r < [T1→T2→...→Tn].length; r++ ){
       Deno= Deno + Count( getMethodStructure(j), [T1→T2→...→Tn], r );
    }
    if ( (Nume + Deno) == 0 ) { Sim[j]= -1; }
    else { Sim[j]= (double) Nume / (double)( Nume + Deno ); }
 }
 Return Sim;
}

/* --- Definition of the Count function. --- */
int Count( method_structure MS, sequence TN, int R)
{
 statement[] S;      // Type of S is a "sequence" of statements
 statement[][] DS;   // Type of DS is a set of a "sequence" of statements
 statement[] SV;     // Type of SV is a "sequence" of statements
 int CT=0;
 // Generate derived sequence replacing R statements with negations
 DS= SqcComb( TN, R );
 for ( each S[] ∈ MS){
   for ( int j=1; j<=MS.length–TN.length; j++ ){
     for ( each SV[] ∈ DS ){
       for ( int k=1; k<= TN.length–R; k++ ){
         if ( S[j] = SV[k] for all k-th elements that are not negative )
                 { CT= CT + (TN.length–R) };
       }
     }
   }
 }
 Return CT;
}
```

Figure 3. Algorithm to compute the similarity for the sequence $[T_1 \rightarrow T_2 \rightarrow \ldots \rightarrow T_n]$

IV. EXPERIMENTAL RESULTS

Table II shows omits some of the results of the retrieval experiments owing to space limitations. The retrieval condition is *if{ → IOException → }* (No.14 in Table I). Let a "boundary method" be a retrieved method whose DSRM similarity is greater than 0 and whose extended Sorensen-Dice index is the minimum among retrieved methods. The *read()* method in the *NioBlockingSelector* class, which is shown at No.15 in Table II, is the boundary method. The number of retrieved methods for the extended Sorensen-Dice index is defined by the boundary method. The number of retrieved methods for DSRM similarity is defined by the number of methods with similarity greater than 0. The degree of improvement of DSRM over extended Sorensen-Dice index is calculated by the following formula.

$$\frac{(No.methods\ by\ Sorensen-Dice)-(No.methods\ by\ DSRM)}{(No.methods\ by\ Sorensen-Dice)} \quad (4)$$

For the retrieval condition *if{ → IOException → }*, the number of retrieved methods for the extended Sorensen-Dice index is 89, whereas the number of retrieved methods for DSRM similarity is 71. The degree of improvement is (89 −

TABLE II. SAMPLE OF RETRIEVAL EXPERIMENTS

No	Method Name	Derived Sequence Retrieval Model			Ext. Sorensen-Dice Model		
		Similarity	Exact Match	Partial Match	Similarity	Exact Match	Partial Match
1	SSIServletExternalResolver::getAbsolutePath()	0.857	6	1	0.857	6	1
2	InputBuffer::readByte()	0.750	3	1	0.750	3	1
4	WsOutbound::flush()	0.750	3	1	0.750	3	1
5	UpgradeAprProcessor::write()	0.750	3	1	0.750	3	1
6	SecureNioChannel::close()	0.667	6	3	0.667	6	3
7	Conversions::byteArrayToLong()	0.600	3	2	0.600	3	2
8	BioReceiver::start()	0	0	11	0.545	6	5
9	FileUtils::forceDelete(File file)	0	0	9	0.333	3	6
10	InputBuffer::skip(long n)	0.200	3	12	0.200	3	12
11	ClassParser::parse()	0.158	3	16	0.158	3	16
12	MemoryUserDatabase::save()	0.097	3	28	0.290	9	22
13	InternalAprInputBuffer::fill()	0	0	23	0.261	6	17
14	SecureNioChannel::read()	0	0	19	0.158	3	16
15	NioBlockingSelector::read()	0.103	3	26	0.103	3	26

TABLE III. SUMMARY OF 18 RETRIEVAL EXPERIMENTS

No	Retrieval Sequence	Number of Methods by Derived Sequence Retrieval Model	Number of Methods by Extended Sorensen-Dice Model	Degree of improvements
1	if{ → Log.debug	143	151	5.3%
2	if{ → Log.debug → }	141	151	6.6%
3	if{ → StringBuilder.append	102	143	28.7%
4	catch{ → Log.error	112	131	14.5%
5	catch{ → ExceptionUtils.handleThrowable	97	111	12.6%
6	if{ → IllegalArgumentException	107	133	19.5%
7	if{ → IllegalArgumentException → }	107	133	19.5%
8	catch{ → Log.error → }	87	128	32.0%
9	if{ → org.apache.juli.logging.Log.debug	70	73	4.1%
10	if{ → StringBuilder.append → }	66	141	53.2%
11	if{ → org.apache.juli.logging.Log.debug → }	68	71	4.2%
12	if{ → StringBuilder.append → StringBuilder.append	41	140	70.7%
13	if{ → IOException	71	89	20.2%
14	if{ → IOException → }	71	89	20.2%
15	catch{ → MBeanException	30	31	3.2%
16	catch{ → MBeanException → }	30	30	0.0%
17	if{ → StringBuilder.append → StringBuilder.append → }	37	135	72.6%
18	catch{ → Log.error → } → }	66	116	43.1%
			Average	23.9%

71) / 89 = 20.2%.

Note that there are some methods whose DSRM similarity is 0, whereas the extended Sorensen-Dice index similarity is greater than 0. This occurs when a program structure does not include a given sequence of statements but includes some elements of the statements. This means that the DSRM imposes a more severe retrieval condition than the extended Sorensen-Dice model. Consequently, the results of the DSRM are a subset of the results of the extended Sorensen-Dice model.

Table III shows a summary of 18 retrieval experiments using the retrieval conditions shown in Table I. Column three of Table III represents the number of methods retrieved by the DSRM with similarity values greater than 0. Column four shows the number of methods retrieved by the extended Sorensen-Dice model.

The degree of improvement ranges from 0% to 72.6% and is 23.9% on average over the extended Sorensen-Dice model. The 0% improvement occurs for the case No.16 in Table III. The retrieval condition for the case No.16 includes the term "*catch{* → *MBeanException* → *}*," which is so rare in the collection of code that all *MBeanExceptions* are preceded by the *catch* clause. Thus, both retrieval models produce the same results. With the exception of No.16, the DSRM similarity outperformed the extended Sorensen-Dice index.

V. CONCLUSIONS

Many different similarity measures have been proposed to detect similar source code fragments. However, defining similarity measures should be carefully performed because the similarity measures may influence the detection of similar fragments more than other processes such as parsing structures, and normalizing identifiers.

Source code is essentially a sequence of statements; therefore, we have defined a similarity measure that is tailored to sequentially structured text to retrieve similar source code fragments. We also developed a mining algorithm to mine a set of sequences of statements with a frequency greater than a given threshold. Prior to similar source code retrieval, determining the frequency sequence of statements was essentially performed to issue a query to the point.

Our similarity measure was evaluated using Apache-Tomcat 7, which is a large-size open source Java program. The results show that the degree of improvement over the extended Sorensen-Dice model is on average 23.9% for the 18 retrieval conditions detected by our mining algorithm.

The results are sufficiently promising to warrant further research. In future, we intend to improve our algorithms by combining Java-specific information such as inheritance of a class and method overloading. We also plan to develop an improved user interface and conduct experiments using various types of open source programs available on the Internet.

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An Intrusion Detection Approach Using An Adaptative Parameter-Free Algorithm

Mourad Daoudi, Mohamed Ahmed-Nacer
Electronics and Computer Science Faculty, Laboratory LSI, USTHB
BP 32 16111 El Alia, Bab-Ezzouar, Algiers, Algeria
m-daoudi@usthb.dz, anacer@cerist.dz

***Abstract*—In intrusion detection from audit security, the volume of data generated by the auditing mechanisms of current systems is very large. It is important to provide security officers with methods and tools to determine predefined attack scenarios in the audit trails. The problem is Non-deterministic Polynomial-time hard (NP-hard). Metaheuristics offer an alternative to solve this type of problems. Unfortunately, many parameters have to be tuned for any metaheuristic, and their values may have a great influence on the efficiency and effectiveness of the search of good solutions. The exploration of an optimal combination of such values may be difficult and big time consuming. Clerc et al. have defined an adaptative parameter-free algorithm, called TRIBES, issued from Particle Swarm Optimization. It is developed to solve continuous problems. In this paper, we propose to adapt TRIBES to solve our combinatorial optimization intrusion detection problem. Modifications in different mechanisms and formulae adaptations in original TRIBES are made, like in the generation process of the particles and in the displacement strategies. The experimentations results show the good behavior of our approach. Comparisons with a basic genetic algorithm are provided.**

Keywords-Metaheuristics; NP-hard Combinatorial Optimization Problems; Particle Swarm Optimization; TRIBES; Genetic Algorithm; Security Audit.

I. INTRODUCTION

An important problem encountered when dealing with metaheuristics [1], [2] is the determination of good parameters values. Indeed, many parameters have to be tuned for any metaheuristic, and they may have a great influence on the efficiency and effectiveness of the search. The optimal values for the parameters depend mainly on the problem and even the instance to deal with and on the search time that the user wants to spend in solving the problem. A universally optimal parameter values set for a given metaheuristic does not exist, and it is not obvious to define which parameter setting should be used. Moreover, in real problems, the parameters are often correlated, which makes the choice of parameters harder. The difficulties still remain when using algorithms with a lower number of parameters. These algorithms are called *adaptive* algorithms, because the information gradually collected during the optimization process are used to determine the values of the parameters.

Particle Swarm Optimization (PSO) [3], [4], [5] is a stochastic population-based metaheuristic of biological inspiration. Several adaptive methods have already been defined for PSO [6], [7], [8], [9]. A parameter free metaheuristic, called *TRIBES* [10], [11], has been developed and showed good results. It was designed to act as a black-box and the user has just to define his problem and to run the process. Such an algorithm exists among genetic algorithms [12]. Several metaheuristics of combinatorial origin are however adapted to the continuous case like in PSO and TRIBES.

The problem under study: “Security Audit Trail Analysis Problem” (SATAP) is of discrete nature [13]. Therefore, we reconsider different mechanisms in TRIBES like the generation process of a particle or the displacement strategies developed in the structural and behavioral adaptations, so that they can be used, with the definition of a distance in the search space to solve our problem.

This paper is organized as follows: after an introduction, the parameter free metaheuristic TRIBES is presented in Section II. Section III describes the “Security Audit Trail analysis problem”. Section IV is dedicated to our contribution, adaptating TRIBES to solve SATAP problem. We provide some experimentation results that show the performances of our approach in Section V, where comparisons to a basic genetic algorithm-based approach are also given.

II.TRIBES

In PSO, each particle moves in the search space and updates its velocity according to best previous positions already found by its neighbors (and itself), trying to find an even better position. This approach has been proved to be powerful but needs parameters predefined by the user, like swarm size, neighborhood size, and some coefficients, and tuning these parameters for a given problem may be, in fact, quite difficult.

In TRIBES [11], the swarm is divided in tribes of individuals. Each tribe acts as an independent swarm, i.e., each tribe has its own global behavior and explores a particular region of the search space. All the tribes exchange information about regions they are exploring. As shown in Fig.1, the swarm is presented as an interconnected network of tribes, which are themselves interconnected networks of individuals. Each particle is informed by itself (cognitive memory *P*, by all the other elements of its tribe, called internal informers) and if this particle is the best particle of the tribe, called a shaman, then it is also informed by the other tribes’ shamans (called external informers). The social memory, noted *g*, of a particle is the informer with the smaller value of the objective function.

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The swarm must be generated and modified automatically, by means of creation, evolution, and removal of the tribes without defining any parameters. To make this possible, rules have been set up. Each particle has a current position and a best position, and then, particles and tribes' qualifiers are defined. A particle is said to be good or neutral: good if it has just improved its best performance, neutral, if not. As for a tribe, it is said to be good, neutral or bad, the larger the number of good particles in the tribe the better the tribe itself. In TRIBES, particles are added or removed according to tribes behaviors. Structural and behavioral adaptations of the swarm may then occur.

A. *Swarm's structural adaptations*

Bad particles are removed from good tribes. The removal of a particle implies a change in the information network. All the informer particles of a removed particle are directed towards the tribe's best particle. Particles are generated by bad tribes and form a new tribe. The bad tribe will keep contact with the new tribe and will try to use it to improve its best location.

Two types of particles are generated, free particles and confined particles, given that the particle type is randomly selected.

- Free particles: The particle is randomly generated according to a uniform distribution either in the whole search space, or on a side of the search space or on a vertex of the search pace. Once the method is selected (at random), then the particle is generated as follows:

$$X_j = \mathrm{U}(x_{\min(j)}, x_{\max(j)}), j \in \{1, \dots, D\} \qquad (1)$$

$$X_j = \begin{cases} \mathrm{U}(x_{min(j)}, x_{max(j)}), j \in I \subset \{1, \dots, D\} \\ \mathrm{U}(x_{min(j)}, x_{max(j)}), j \in J \subset \{1, \dots, D\} \end{cases} \qquad (2)$$

$$X_j = \mathrm{U}(\{x_{\min(j)}, x_{\max(j)}\}), j \in \{1, \dots, D\} \qquad (3)$$

  where $\mathrm{U}(x_{\min(j)}, x_{\max(j)})$ is a real number uniformly chosen in $[x_{min(j)}, x_{max(j)}]$ and $\mathrm{U}(\{x_{\min(j)}, x_{\max(j)}\})$ is a real number uniformly chosen in the list $\{x_{min(j)}, \dots, x_{max(j)}\}$. The two sets I and J are randomly defined for each particle and determine a partition of $\{1 \dots D\}$. Generating free particles aims at provide diversity in the population.

- Confined particles: They aim at intensify research inside an interesting area. Let X_i be the best particle of the generating tribe and i_x its best informer, and let P_x and P_{ix} be the best locations of X_i and i_x. The new particle will be generated in the D-sphere of center P_{ix} and radius $|| P_x - P_{ix} ||$ such that:

$$X_{generated} = alea_{sphere}(P_{ix}, || P_x - P_{ix} ||) \qquad (4)$$

  where $alea_{sphere}(P_{ix}, || P_x - P_{ix} ||)$ is uniformly chosen in the hyper-sphere of center P_{ix} and radius $|| P_x - P_{ix} ||$.

B. *Swarm behavioral adaptations*

The second way in view of adapting the swarm to the results found by the particles is to choose the strategy of displacement of each particle according to its recent past. Between two iterations, a particle can improve its performance, denoted by **(+)**, or can deteriorate it which is denoted by **(-)**. There can be no change as well, namely a status quo that is denoted by **(=)**. Then, the choice of the strategy of displacement is made according to the two last variations as shown in the following table:

TABLE I. STRATEGIES OF DISPLACEMENT IN TRIBES

Gathered status	**Strategy of displacement**
(= +) (++)	*Local by independent Gaussians*
(+ =) (- +)	*Disturbed pivot*
(- -) (= -) (+ -) (= =)	*Pivot*

- The pivot strategy is inspired from published works as in [13]. Let p be the best position ever reached by the particle and let g be its best informer and f the objective function. Then, the displacement is determined by:

$$X = C_1 * alea_{sphere}(H_p) + C_2 * alea_{sphere}(H_g) \quad (5)$$

  where $C_1 = \frac{f(p)}{f(p)+f(g)}$, $C_2 = \frac{f(g)}{f(p)+f(g)}$, $alea_{sphere}(H_p)$ is uniformly chosen in the hyper-sphere of center p and radius $\|p-g\|$ and $alea_{sphere}(H_g)$ is uniformly chosen in the hyper-sphere of center g and radius $\|p-g\|$.
- The disturbed pivot strategy is similar to the precedent one, adding a noise.
  In practice, for each component of the last computed position, a random number b is generated using a centered Gaussian distribution with standard deviation $\frac{f(p)-f(g)}{f(p)+f(g)}$. Then the component is multiplied by $(1+b)$.
- In local by independent Gaussians strategy, if g = $(g_1,..g_D)$ is the particle best informer, then the displacement is determined by:

$$x_j = g_j + alea_{normal}(g_j - xj, \|g_j - x_j\|), j \in \{1, \dots, D\} \quad (6)$$

  where $alea_{normal}(g_j - x, \|g_j - x_j\|), j \in \{1, \dots, D\}$ is a point randomly chosen with a Gaussian distribution with mean $(g_j - x_j)$ and standard deviation $\|g_j - x_j\|$.

III. SECURITY AUDIT TRAIL ANALYSIS PROBLEM

A. Introduction

Computer security has become in recent years a crucial problem [14]. It rallies the methods, techniques and tools used to protect systems, data and services against the accidental or intentional threats, to ensure: Confidentiality; Availability, and Integrity. Nowadays, different techniques and methods have been developed to implement a security policy: authentication, cryptography, firewalls, proxies, antivirus, Virtual Private Network (VPN), and Intrusion Detection System (IDS) [15].

IDSs are software or hardware systems that automate the process of monitoring the events occurring in a computer system or network, analyzing them for signs of security problems [16-19]. The intrusion detection system was introduced by James Anderson [20], but the subject didn't have great success. After that, Denning defined the intrusion detection system models [21], where he exhibits the importance of security audit, with the aim to detect the possible violations of system security policy.

According to Intrusion Detection Working Group of IETF an intrusion detection system includes three vital functional elements: information source, analysis engine, and response component [22]. There are five concepts to classify intrusion detection Systems, which are: The detection method; The behavior on detection; The audit source location; The detection paradigm; The usage frequency.

The detection method is one of the principal characters of classification they describe the characteristics of the analyzer. When the intrusion detection system uses information about the normal behavior of the system it monitors, we qualify it as behavior-based. When the intrusion detection system uses information about the attacks, we qualify it as knowledge-based.

The Security Audit is as medical diagnosis, in order to determine the set of conditions, which may explain the presence of observed symptoms (in IDS: the recorded events in the audit trail). For this reason, expert uses specific knowledge (the scenarios of attack) based cause at an effect. The expert uses its knowledge to develop assumptions that confront the reality observed. If there are still observed symptoms than the made hypothesis made is wrong. On the other hand, if there are more symptoms than those observed in the reality, a new hypothesis more relevant must be tested [23].

B. Specifications

We want to achieve a system that can tell whether an intrusion has occurred or not when analyzing the trace file security audit. Among the different existing approaches to develop such a system, we consider a posterior approach based on attack scenarios recorded in the audit trail. To establish these different scenarios (different attacks or attack signatures), we first define a certain number of so-called auditable event. The selection of auditable events in a real case is left to the administrator. Thus, each attack will be defined by the number of occurrence of auditable events. The audit file for analysis will also be defined by the number of occurrence of auditable events. The temporal order of sequence of events will not be considered (in this case the system can function in a heterogeneous distributed environment where the construction of a common time is impossible). In informal terms, the problem is to find the combination of attacks that maximizes the incurred risk, while possible under number of events of each type recorded in the audit file.

In this approach, the attack scenarios are modeled as a set of couples (e, N_e), where e is the type of event and N_e is the number of occurrences of this type of event in the scenario.

Formally, SATAP can be expressed by the following [22]:

$$\begin{cases} Max \sum_{1}^{Na} R_j . H_j \\ (AE \,.\, H)_i \le O_i,\ i \in \{1 \ldots N_e\} \end{cases} \tag{7}$$

where:

- N_e : the number of type of audit events
- N_a : the number of potential known attacks
- AE : a N_e x N_a attack-events matrix which gives the set events generated by each attack. AE_{ij} is the number of events of type i generated by the attack j, ($AEi\ j \geq 0$)
- R : a N_a-dimensional weight vector, where R_i ($R_i > 0$) is the weight associated with the attack i (R_i is proportional to the inherent risk in attack scenario i)
- O : a N_e-dimensional vector, where O_i is the number of events of type i present in the audit trail (O is the observed audit vector)
- H : a N_a-dimensional hypothesis vector, where $H_i = 1$ if attack i is present according to the hypothesis and $H_i = 0$ otherwise (H describes a particular attack subset).

To explain the data contained in the audit trail (i.e., O) by the occurrence of one or more attacks, we have to find the H vector which maximizes the product RxH (it is the pessimistic approach: finding H so that the risk is the greatest), subject to the constraint $(AE.H)_i \le O_i$, $1 \le i \le N_e$.

Because finding H vector is NP-complete, the application of classical algorithm is impossible where N_a equals to several hundreds.

The heuristic approach that we have chosen to solve that NP-complete problem is the following: a hypothesis is made (e.g., among the set of possible attacks, attacks i, j and k are present in the trail), the realism of the hypothesis is evaluated and, according to this evaluation, an improved hypothesis is tried, until a solution is found.

In order to evaluate a hypothesis corresponding to a particular subset of present attack, we count the occurrence of events of each type generated by all the attacks of the hypothesis. If these numbers are less than or equal to the number of events recorded in the trail, then the hypothesis is realistic.

To derive a new hypothesis based on the past hypothesis, several approaches have been proposed by researchers, including Genetic Algorithms [22-25] and Biogeography Based Optimization [26], [27].

IV. SECURITY AUDIT TRAIL ANALYSIS USING TRIBES ALGORITHM

The approach aims to determine if the events generated by a user correspond to known attacks, and to search in the audit

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trail file for the occurrence of attacks by using a heuristic method because this search is an NP–complete problem. The goal of the heuristic used is to find the hypothesized vector H that maximizes the product R*H, subject to the constraint $(AE.H)_i \leq O_i$,$1 \leq i \leq N_e$, where R is a weight vector that reflects the priorities of the security manager, AE is the attack-events matrix that correlates sets of events with known attacks, and N_e the number of types of audit events.

TRIBES is an optimum search algorithm issued from PSO. A swarm is divided into tribes of artificial particles. Each particle changes its position, moving from a position to an other. These positions are strings of length l coding a potential solution to the problem to be solved. We are in a situation where the coding of positions is immediate since the solution of the problem is expressed specifically in the form of a binary sequence. A position is considered as a series of N_a integers with values 0 or 1. Each position is a particular instance of the vector H. In other words, the element i in the position will be 1 if the position is a solution in which the attack i is declared present. Otherwise, this element takes the value 0. We note that the sum of the component elements in a position indicates the number of attacks that are detected.

The TRIBES-based method should return a binary N_a –vector H = (H_1, …, H_{Na}), where the value H_i = 1 indicates the presence of the attack i in the audit file O and 0 its absence. As we have to solve a maximization problem, the best solution is associated with the hypothesis H of larger value of the selective function

$$F(H) = R*H = \sum_{1}^{Na} R_i\, H_i \tag{8}$$

which represents the total risks incurred by the system under surveillance.

In addition, as we deal with a constrained problem, any solution to the problem must verify the inequalities: $(AE\times H)_I \leq O_i$ avec $0<i<N_e$. Thus, we have to eliminate the solutions that do not comply, setting to zero the value of the corresponding objective function (that is a harmony in which each note has a zero value). There combination process is repeated until a solution satisfying the constraints is generated. Recall that R_i is the weight of the attack i, that is the risk incurred to the system if the attack is not detected. For simplicity, the R_i value is taken equal to 1 for all i, i=1 … N_a. In this case, the objective function F resumes in computing the number of detected attacks.

We recall here that TRIBES is initially defined to solve continuous problems. The problem under consideration in our paper is an NP Hard combinatorial problem. Therefore adaptations of TRIBES are needed to solve our SATAP problem.

Let us define first a distance in our particular search space of binary N_a-vectors. We consider the well known Hamming distance, well suited to our problem. If *x* and *y* are two positions, then we note: *dist(x, y).*

After these specifications, we are now able to present the different changes that occur in the structural and behavioral adaptation mechanisms in our approach.

A. *Swarm's structural adaptations*

As defined in original TRIBES, the swarm is an interconnected network of tribes of different size, which are themselves interconnected networks of particles. In intra-tribe communication, each particle is informed by all the other elements of its tribe, and in inter-tribe communication, links are specified at their generation; each new tribe keeps contact with its generating tribes.

Because of the discrete solutions search space, the generation process of the free particles and confined particles seen in Section II is modified.

- The free particles are binary random vectors, randomly generated in the whole search space. If $X=(x_1,.., x_D)$ denotes the position of a particle, x_i is 1 or 0, indicating that the ith attack is detected or not.
- Let x be the best particle of the generating tribe and i_x its best informer, and let P_x and P_x be the best locations of x and i_x. Then, the confined particle, *Xnew* is such that: $dist(p_{ix}\ ,\ Xnew\ ) \leq$ d where d=*dist*(p_{ix} , p_x). To do so, we generate randomly an integer N between *0* and d, and then proceed randomly to N changes in P_x. The obtained new binary vector is the current position of the created new particle.

B. *Swarm's behavioral adaptations*

1) Pivot strategy: The pivot method concerns particles with bad behavior in the last two iterations. The method given in the continuous case is still maintained in its principle. However, the new position is determined from *p* et *g* using the distance **dist** to create their neighborhoods in place of the two hyper-spheres H_p and H_g. We recall that *p* is the best position of the particle and *g* the best position of the informers of the particle, and f the objective function.

The proposed procedure is:

a) Compute the distance dist(p, g). Let d its value.

b) Generate two uniformly distributed random integer numbers u_1 and u_2 in (0, d)

c) Generate two uniformly distributed random Na-vectors X and Y such that: dist(X, p) = u_1 and dist(Y, g) = u_2

d) Compute the attraction coefficients:

$$c_1 = \frac{f(\vec{p})}{f(p)+f(g)} \text{ and } c_2 = \frac{f(g)}{f(p)+f(g)}$$

e) For each element i, $0 \leq i \leq N_a$, of the new position vector Xnew, generate a random number u_3 , from a Bernoulli distribution of probability c_1.

If u_3 =1, then Xnew(i)=X (i)

If u_3 =0, then Xnew(i)=Y(i)

2) Disturbed Pivot strategy: The disturbed pivot method concerns particles with medium performances in the last two iterations. The proposed method preserves the original disturbed pivot.

The proposed procedure is :

a) Generate a position $X^{'}$ using the adapted Pivot method seen above.

b) Generate a random number *b* from a centered Gaussian distribution with standard deviation: $\frac{f(p)-f(g)}{f(p)+f(g)}$

c) Associate a number of k components of X to b,

d) Choose randomly k elements in *X***.** Let $X_{i1},\ldots,$ X_{ik} these elements.

If X_{ij} = 1 then *Xnew(*ij)=0, and

If X_{ij} = 0 then *Xnew*(ij)=1.

3) Local strategy with independent Gaussians: In this strategy, the principle is to intensify the search around the best informer $\vec{g}$. So, the neighborhood $\vec{g}$ will be determined using the distance d defined earlier. Then, a neighbor is generated from a Gaussian distribution as follows:

a) Compute the distance dist(X, g), *X* is the current position.

b) Generate an integer random number k from a centered Gaussian distribution with standard deviation dist(X, g).

c) Choose randomly k elements in *X*. Let $X_{i1},\ldots,$ X_{ik} these elements.

If X_{ij} = 1 then Xnew(ij) = 0, and

If X_{ij}=0 then Xnew(ij)=1.

Remark: The steps given earlier in TRIBES and not redefined in the present section, still remain unchanged.

V. EXPERIMENT RESULTS

A. Adapted-TRIBES approach

During the simulations, all the attacks actually present in the analyzed audit file must be known in advance. Thus, the events corresponding to one or more attacks are included in the observed audit vector O.

Ratios TPR, FPR, Accuracy and Precision [28] are used to evaluate our approach intrusion detection quality, with:

- TPR (True positive rate): TP / (TP+FN)
- FPR (False positive rate): FP / (TN+FP)
- Accuracy: (TN+TP) / (TN+TP+FN+FP)
- Precision: TP / (TP+FP)

where True negatives (TN) as well as true positives (TP) correspond to correct intrusion detection: that is, events are successfully labeled as normal and attacks, respectively. False positives (FP) refer to normal events being predicted as attacks; false negatives (FN) are attack events incorrectly predicted as normal events.

To evaluate our approach, many tests were performed using Attack-Events matrices of different sizes. All results are obtained as the average of 10 executions carried out for the same data and same number of injected attacks.

The following figures show the quality of our intrusion detection approach.

Reported results in Fig. 1 concern tests performed on an attack-events matrix of size (28x24) issued from [23], with 24 attacks and 28 types of events, and 15 attacks are injected.

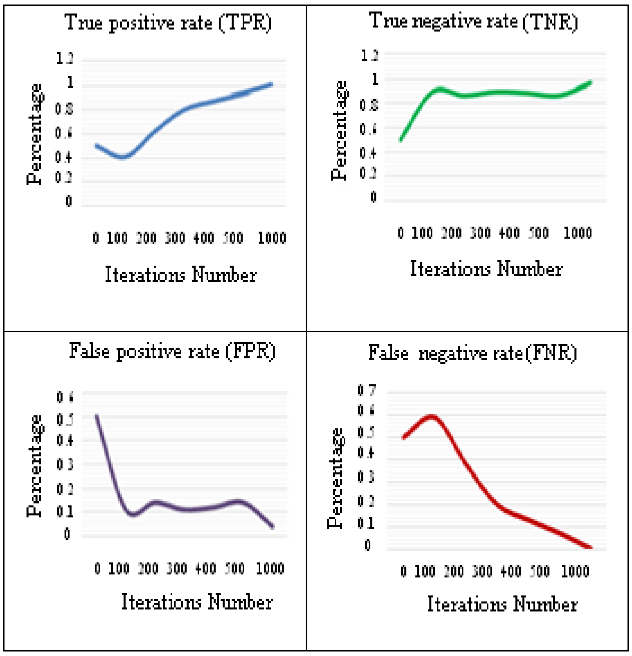

Figure1. Intrusion Detection Quality Measures: (AE: 24x28)

Fig. 2 shows results of tests performed on larger data, randomly generated, with an attack-events matrix of size (100x200), and 200 injected attacks.

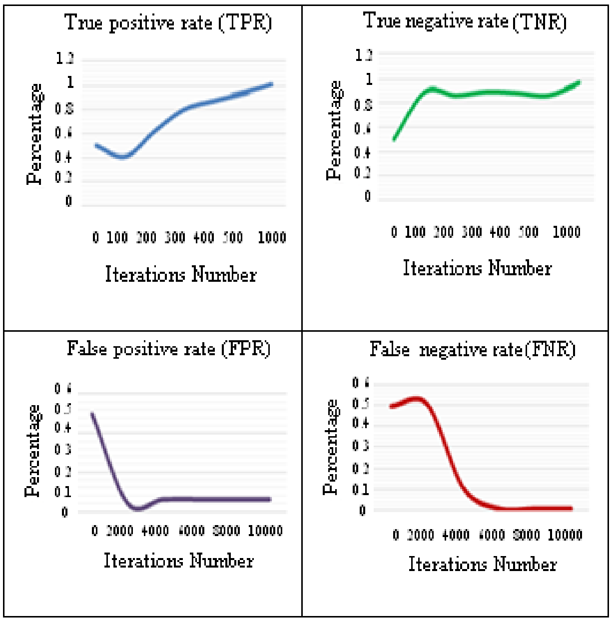

Figure 2. Intrusion Detection Quality Measures: (AE: 100x200)

We observe that after a certain number of generations, all injected attacks are detected, and no false attack (TPR= 1 and FPR =0). In addition, the number of attacks injected has no influence on these results. Indeed, we observe in both figures Fig. 1 and Fig. 2 that all attacks are detected (TPR=1 and FPR=0) after 1000 iterations (respectively 10000). Further, no false positive nor false negative attacks are detected (FNR.= FNR = 0) after 1000 iterations (respectively 10000).

B. *TRIBES vs. Genetic Algorithms*

An intrusion detection approach using Genetic Algorithms (GA) has been developed and tested in order to compare the results with those obtained when using our Tribes based approach.

A genetic algorithm handles a group formed of a population of individuals, of constant size, initially randomly generated. This constant size induces a competition between individuals representing the potential solutions to the problem at hand. The population evolves in successive generations. The strongest individuals survive and reproduce to create new individuals, while the others are gradually disappearing. To enable this change in population, genetic operators have been defined such as selection, crossover and mutation. However, to evaluate the different individuals in a population and allow differentiating between a "strong" individual and a "weak" one, a so-called selective function (fitness) is used, to associate a value on each individual in the population.

Results reported in Fig. 3 concern a (28x24) - Attack-Events matrix issued from [23].

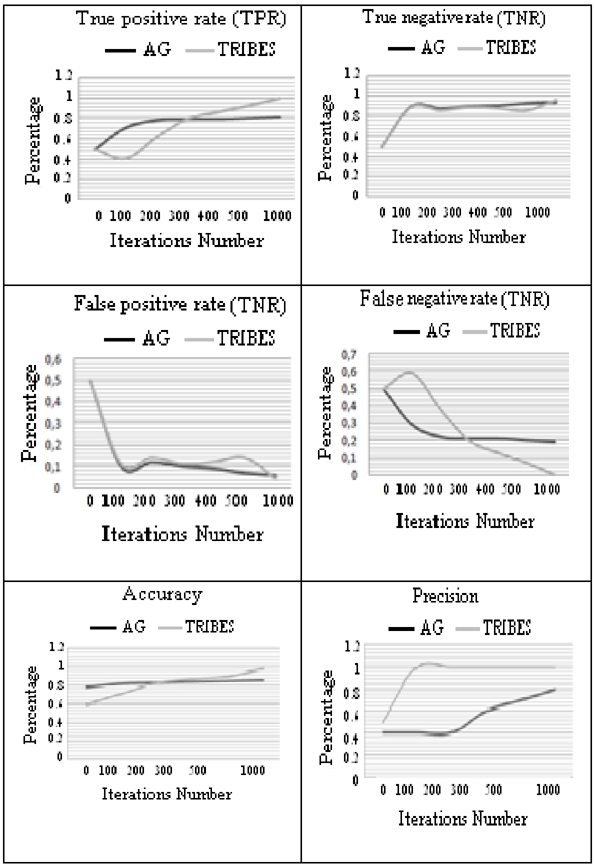

Figure 3. Tribes vs. AG: Intusion Detection Quality Measures (AE-matrix: 24x28)

The number of injected attacks is 15. We observe that all attacks are detected in both approaches. We note that GA detected the attacks after more than 1000 iterations, and obtained more false negatives.

Table 2 shows the different parameters involved in GA and their values obtained through simulation.

TABLE II. GENETIC ALGORITHM PARAMETERS SET

Population Size	500
Generation Number	1000
Mutation rate	0.01
Crossover rate	0.5

Now, let us consider the execution time for both methods. Tests are performed on the randomly generated attack-events matrix of size (100x200), and 200 injected attacks (used in section A above). Results are reported in Fig. 4.

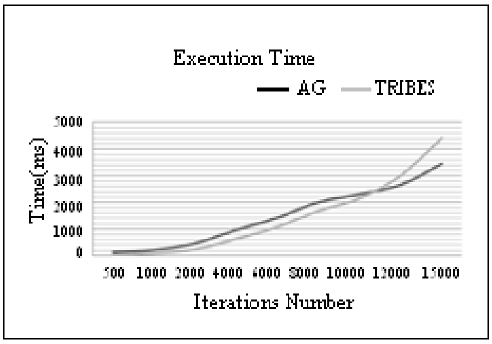

Figure 4. TRIBES vs. AG: Intrusion Detection Execution Time (AE-matrix: 100x200)

We observe that TRIBES-based method presents higher execution time with the number of iterations.

5. CONCLUSION AND FUTURE WORK

Security Audit trail Analysis can be accomplished by searching audit trail logs of user activities for known attacks. The problem is a combinatorial optimization problem NP-Hard. Metaheuristics offer an alternative for solving this type of problem when the size of the database events and attacks grow. We proposed to use an adaptative parameter-free algorithm as detection engine. Originally conceived to solve continuous problems, we had to we reconsider different mechanisms in TRIBES like the generation process of a particle or the displacement strategies developed in the structural and behavioral adaptations, so that they can be used, with the definition of a distance in the search space to solve our NP-Hard combinatorial optimization SATAP problem.

Experimental results of simulated intrusions detection are given. The effectiveness of the approach is evaluated by its ability to make correct predictions. It proved to be effective and capable of producing a reliable method for intrusion detection.

Comparisons with Genetic Algorithms inspired approach are provided, showing for our approach a good behavior. However, these systems are usually developed for predefined environments and do not offer a solution to some network characteristics such as changes in behavior of users and services, the increasing complexity and evolution of the types of attacks that they may be subject, the speed of attacks that can occur simultaneously on several machines, etc.

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SSICC: Sharing Sensitive Information in a Cloud-of-Clouds

Rick Lopes de Souza, Hylson Vescovi Netto, Lau Cheuk Lung, Ricardo Felipe Custodio
Department of Informatics and Statistics
Federal University of Santa Catarina
Florianopolis, Brazil
Email: {rick.lopes, hylson.vescovi, lau.lung, custodio}@inf.ufsc.br

***Abstract*—The need to share and manipulate sensitive data is a challenge for most content providers using the cloud for storage. The objective of this work is to propose an architecture to ensure a distributed access control to secure sharing of sensitive electronic documents in the cloud. This paper will explain the architecture of the model, details of the protocols, implementation, and analysis on security, usability, and performance. The main features of the proposed model are: the use of Identity-Based Encryption, byzantine fault tolerance, reliable integrity check, secure user revocation, low complexity in the management of cryptographic keys and secure sharing of sensitive electronic documents.**

Keywords*-*Privacy; Sensitive; Cloud.

I. Introduction

The possibility of decreasing investment in infrastructure for data storage become real through the use of cloud computing. A user can send, edit, save, and access his data in the cloud using any device. These characteristics are attractive for many corporations due to economic facilities provided. The cloud service provides savings in acquisition and configuration of hardware and software, making corporations to pay only for what they actually use.

The advantages of using cloud services bring with them the problem of ensuring the security of stored data. Typically, cloud services allocate more than one application or data in the same structure and, therefore, problems such as attacks coming from other corporations or even employees of cloud providers become real. Today, one of the main concerns of companies adopting cloud services is ensuring security and privacy of sensitive data. It is not difficult to find cases of data theft, as the case of SalesForce in 2007, where criminals have succeeded in stealing information about customers, such as e-mails and addresses [1]. In order to allow users to control the access to their sensitive data stored in a public cloud, a suitable access control is required. The access policies must restrict data access to only those intended by the data owners. These rules must be guaranteed by the cloud providers. If the system is allocated in just one cloud provider, the data owners have to assume that the cloud providers are trusted and they will prevent the access from unauthorized users. Thus, storing unencrypted data in public clouds can expose sensitive information to a malicious third party. To provide the necessary security, the cloud provider must not have access to unencrypted data. Therefore, the action of sharing a sensitive document to groups or roles is still considered a challenge.

Many techniques have been proposed to address these problems, but there is not a singular better solution. The traditional public key encryption techniques that uses public certificates with a public key infrastructure can not resolve all the problems involved in the sensitive document sharing and it is infeasible when the system grows in number of users. Another technique that can be used is called Identity Based Encryption (IBE) [2] and it was first introduced by Shamir in 1985. The IBE consists in three entities: sender, receiver, and a trusted authority. The sender of a message specifies an identity (a set of characters) such that only a receiver that matches that identity can decrypt and read it. The trusted authority is responsible for the authentication process and to supply the necessary private keys. These private keys are directly tied to the users identities.

The IBE trusted authorities can access the users private key and, for this reason, the IBE technique faces some resistance to be implemented in some systems. To overcome this limitation, multiple trusted authorities can be used, such that any of them can posses the users private key. The work of Aniket [3] proposes a multi authority to IBE systems. In his work, a set of authorities execute a modified algorithm of Joint Feldman Distributed Key Generator (JF-DKG) [4] to generate the master secret in a distributed manner. Users can contact a subset of authorities to request a part of the private key and then, reconstruct the entire user private key.

Another IBE limitation is the user revocation. Since one user has an identity 'ID', if the related key is compromised, there is no way to generate another private key to the same identity 'ID' without affecting other users. There are many proposes to overcome this problem. One of the proposals is to use Attributed Based Encryption [5]. This technique was proposed by Amit Sahai and Brent Waters and it posses the concept that every user has some attribute in a specific company or entity. The private keys are tied to users attributes that are given to then by a singular trusted authority. Recent works [6], [7], [8], [9], [10], [11] are based on multi-authorities but they still have some problems with user revocation due to re-encryption of documents and rekeying. Zhou et al. [12] proposes a modified algorithm of Boneh and Fraklin Identity Based Encryption (BF-IBE) [13] proposing a role based encryption. Our work has a similar idea of the Zhou et al. paper, using the concept of roles and groups to identify the users and encrypt sensitive documents.

This paper has as main focus to propose an architecture

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to ensure secure sharing of sensitive documents in a cloud of clouds. To make it possible, this paper uses Identity Based Encryption with multi-authorities to manage the cryptographic keys. The main contribution of this paper is an architecture to store sensitive documents and share it in public or private clouds. This architecture makes practical the maintenance of users and groups, by using multi-authorities with IBE, secret sharing and erasure codes. This work touches the areas of Secure User Revocation, Reliable Integrity Check, Backward and Forward Secrecy, Byzantine Fault Tolerance, Storage Economy, and Efficient Document Sharing.

This paper is organized as follows. In Section 2, we present the math preliminaries and the related work. Section 3 presents the architecture of the proposed solution. Section 4 presents a detailed system model. In Section 5, we will analyze the security and performance of the system model. Finally, Section 6 presents the conclusions of the study and discussion of possible future works.

II. Preliminaries and Related Work

A. Bilinear Pairing

Let G_1, G_2 be additive groups and G_T a multiplicative group, all of prime order p. Let $P \in G_1$, $Q \in G_2$ be generators of G_1 and G_2 respectively. A pairing is a map: $e : G_1 \times G_2 \rightarrow G_T$ for wich the following holds:

- **Bilinearity:** $\forall a, b \in Z_p^* : e(aP, bQ) = e(P, Q)^{a,b}$
- **Non-degeneracy:** $e(P, Q) \neq 1$
- **Computability:** There is an efficient algorithm to compute $e(P, Q)$ for any $P \in G_1$ and $Q \in G_2$.

The IBC protocols used in this work have a special form of pairing called symmetric pairing which has: $e(P, Q) = e(Q, P)$. The security of the techniques used in this work are based on the Decisional Diffie-Hellman (DDH) problem and the Decisional Bilinear Diffie-Hellman (DBDH) problem. Our work rely on the assumption that no probabilistic polynomial-time algorithms can solve the DDH and DBDH problem with non-negligible advantage.

B. Distributed Key Generation

We use a completely distributed key generation based on the Joint-Feldman distributed key generator (JF-DKG), the distributed key generation proposed by Aniket [3]. The JF-DKG requires a number $n \geq 3t + 1$ nodes to run correctly, being the simplest and most efficient DKG. We use the BF-IBE technique due to it's simplicity of setup and methods. In BF-IBE Setup, a Private Key Generator (PKG) generates private keys (d) for clients using their known identities (ID) and master-key (s). We seek an (n, t) distributed key generation over an elliptic curve group G of order q and generator U, where n are the total nodes involved and $t + 1$ honest nodes are sufficient to generate it correctly. Let F(z) = $a_0 + a_1 z + ... + a_t z^t \in Z_q[z]$ be the current shared polynomial and s = a_0.

The protocol proposed by Aniket uses an improved version of Feldman Verifiable Secret Sharing (Feldman VSS) algorithm to generate in a distributed manner the master-key. It has a bulletin board that generates the public parameters of BF-IBE Setup, publishes the values and than initializes the A_k and A_{ik} values to zero, for i = 1, ..., n and k = 0 ... t, where $A_k = a_k U$ and $A_{ik} = a_{ik} U$. The master key is set to zero. The nodes initiates the Feldman VSS. After t + 1 nodes run their protocols successfully, the distributed shares are considered safe. These nodes are called qualified nodes. Here, we denote their set as ∂. The bulletin board computes and broadcasts the coefficients A_k (for k = 0... t) for the implied shared polynomial F(z).U as A_k = $\sum_{P_j \in \partial} A_{ik}$. After verifying the new A_k values, nodes send confirmation signatures to the bulletin board. On receiving t + 1 confirmation signatures, the A_k values are finalized. Each node then computes their secret share as $s_i = \sum_{P_j \in \partial} s_{ji}$. Mode details can be obtained in the work of Aniket [3].

C. Private Key Extraction

To extract the private key in a distributed way, the client must contact the nodes and send a specific ID. After receiving the ID, the PKGs $P_i \in O$ authentic and authorize the user and then returns a private-key share $S_i H(ID)$ over a secure channel. The H represents a hash function $H : (0, 1)^* \rightarrow G^*$. Upon receiving t + 1 correct shares of her private key, the client can reconstruct the private key D_{id} as $D_{id} = \sum_{P_i \in O} \lambda_i s_i H(ID)$, where the Lagrange coefficient $\lambda_i = \prod_{P_j \in O, j \neq i} \frac{j}{j-i}$.

D. Related Work

Recent works propose the use of privacy services [14], [15] to address the privacy documents storage problem, as well as some others works [16], [17] propose not to encrypt the files and just split it and then send it to different cloud providers. These works try to solve the problems involved to store sensitive documents in cloud, however, they can not provide all the necessary features to make a secure sharing. The work of Itani et al. [14] didn't provide a sharing scheme and a fault tolerance system. If the privacy service stops, the client can not encrypt or decrypt the files. The propose of Padilha [16] use the technique of homomorphism to modify the privacy parts of sensitive documents through the use of addictive functions, however, the techniques to provide secrecy using total homomorphism are theoretical and lack of pragmatic implementations. Practical implementations of total homomorphism to secrecy systems are still open issues research, therefore, are not applicable in the present circumstances of corporations.

Other line of work is to use Atributed Based Encryption (ABE), where each user receives credentials from the trusted authorities which releases the access to the sensitive documents. Some works as [6], [7], [8], [9], [10], [11] proposes the use of multi-authorities to generate the user private key, avoiding the cloud key escrow. These papers also works with a distributed manner of providing the attributes, supplying the needs of identity confidentiality. However, the solutions that involve ABE have the drawback of revocation. Once the key that encrypts one document is revoked, the system needs to re-encrypt the sensitive documents and then have to re-keying. Another peculiarity in almost all these related works is that they don't support fault tolerance. The proposes are based on multi-authorities just do extract the private keys, nevertheless, they store the files on just one cloud provider. If this cloud

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provider fail for any reason, the user will not have access to his sensitive documents. Another issue due to these facts is that if an attacker can compromise one cryptographic key and have access to the cloud provider, he can obtain all the sensitive data.

Other line of work made by Bessani et al [18] uses concepts that we are going to use: symmetric encryption, Shamir secret sharing and erasure optimal code. Nevertheless the paper does not propose any mechanism to share the necessary keys to guarantee the integrity of the shares. It simply admits that there is a mechanism to share keys, however, this is one of the main challenges in sharing sensitive documents using encryption. Another downside is the read data algorithm that does not check integrity with a public key to verify the hash of the parties. If the cloud provider is a malicious attacker, it can modify the parts and provide false hashes for integrity check, thus compromising the system.

Our work resembles to the Zhou et al [12]. His work proposes the use of a modified IBE to enforce role-based access control, providing the possibility to encrypt a file to a single user or roles. The propose is efficient if it does not have many revocation operations, otherwise it will present high complexity. Zhou work does not solve the key escrow problem. The system administrator can access users keys in the extract operation. Another issue is the single point of failure and if the administrator fails, undertake part of the revocation system.

Based on earlier research, this article identified the main challenges to share sensitive documents in a secure manner. We have proposed an architecture to tackle all these challenges involving the use of IBE to provide the following features: Secure User Revocation, Reliable Integrity Check, Backward and Forward Secrecy, Byzantine Fault Tolerance, Storage Economy, and Efficient Document Sharing.

III. System Architecture

There will be two main types of components in the architecture for the solution. The first component represents public clouds that host the application servers and store sensitive documents. The second component is the end users that possess mechanisms to encrypt and decrypt sensitive data, as well as mechanisms to securely store cryptographic keys used.

The client side is responsible for editing, encrypting and decrypting files. This side also aims to define who are the custodians of sensitive data to be encrypted. These custodians are delimited by access rules specified for each application. The storage servers must be hosted on different cloud providers. The main features that are needed: the distribution of access control through state machine replication and the distributed manner of application servers spread across nodes.

Figure 1 illustrates how the architecture of secrecy works. Briefly, the architecture works as follows: N nodes (cloud providers) will conduct first a system setup, so they can share a private key. Each node shares access control and provides an application to users to share sensitive documents. Users who wish to share confidential documents must encrypt the document locally, perform operations of breaking and encoding then the document is sent and stored in the cloud providers. The document owner must inform who may have access to sensitive data. A user who wishes to obtain a document must authenticate to the cloud providers to obtain the required data to join, decode, and decrypt the document. In this work, it is used an abstraction called Data Block that store data about the sensitive documents. The Data Block contains five items: ID, part of the encrypted symmetric key, signed hash of the encrypted key, part of the encrypted document and signed hash of the encrypted document.

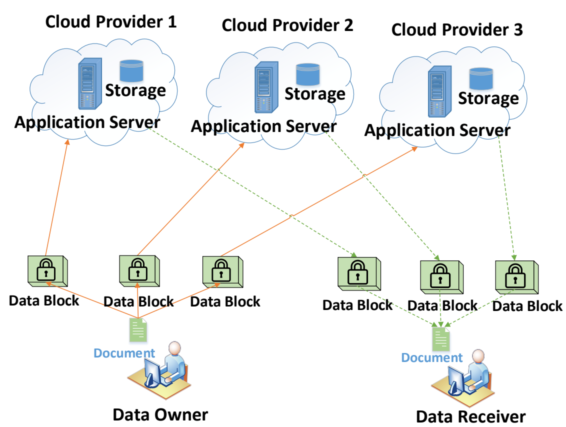

Figure 1. System Model - A Data Owner send N datablocks to cloud providers to share it. The Data Receiver must obtain a pre-defined number of datablocks to recompose the file.

The access control must be implemented in a distributed form. A cloud provider should not impersonate a user to obtain the other parts of the secret. Each cloud provider will posses an access control list and it will make the authentication and authorization of each user. As we consider the cloud as a semi-honest entity, the access control is considered reliable.

We focus on the architecture to provide distributed access control to documents and we will not discuss in details the access control. This proposal has the following assumptions:

- Every cloud provider has an access control;
- There is no concurrency for documents access;
- There is no concurrency for control access;
- The cryptography algorithms are resistant collision;
- The cloud providers are semi-trusted (They will behave correctly before the user requests, but can be curious and see the data stored);

IV. System Model

We use the secret sharing scheme to integrate the confidentiality and the availability. Since all encrypted keys are splitted in N pieces (N is the total number of cloud providers), the user needs a minimum of M parts (M is provided in the setup of the system) to recombine the encrypted key. In fact, we reuse the access control of an application to control which readers are able to access the stored data. We also use the mechanism of information-optimal erasure code [19], enforcing an economy in the cloud providers to store different versions of the same document. Otherwise, the costs would increase by a factor of n

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if it was necessary to replicate it to n clouds. Each share has a reduced size by a factor of $\frac{n}{f+1}$ [20], considering f the number of faulty servers. Here we consider that the minimum number of parts to recover a Shamir secret sharing is directly related to the redundant parts of the information-optimal erasure code. For example, if we consider a total number $N = 4$ of nodes and a minimum number $M = 2$ of nodes to recover some key, the erasure code will consider a total of $T = N - M$ and a redundant number of $R = M$. It will be always necessary to gather at least two parts of the total to recombine the parts.

We use the BF-IBE scheme to encrypt symmetric keys building a specific ID containing the following information: Name of the Document, Group of Custodians, and Document Version. We use this specific ID due a set of characteristics that are necessary to share sensitive documents. The Name of the Document in ID is to generate a different key to each document stored in the cloud. The Group of Custodians is to limit the access control of the document. As we will be reusing the access control system, this group of custodians will be used to authenticate and allow access to sensitive documents. For every different group of custodians, there is a different key. The version number and the others elements has the intention to control the access for different versions of the documents and to guarantee the backward and forward secrecy. The Hess Signature technique [21] is used to sign the parts of encrypted symmetric key and encrypted data to guarantee the integrity. In this work, we use the same key pair to encrypt and sign, facilitating the key sharing and management.

Before users can use the system, it must be made the Private Key Generators (PKGs) setup. The protocol is described in the section II-C and it is responsible for the distributed generation of IBE master secret. After this step, reusing the system access control, the users can share sensitive documents through the use of the following methods: Write Data and Read Data. The Write Data algorithm is data owner responsibility and it encrypts, encode, and send all the encrypted and signed parts to the cloud providers. The Read Data algorithm is executed by the data receivers that wants to visualize the document content.

The Write Data (Figure 2) authenticates the user and asks the access control for the document metadata (line 4). The new document to be stored will have the last version found (line 5) plus one (line 6). A symmetric key is randomly created (line 7) to encrypt the document (line 9). An ID for the document is defined (line 10) and a public key based on this ID is created (line 11) using the BF-IBE technique. The symmetric key is encrypted with the public key (line 12) and then splitted into shares using Shamir secret sharing scheme (line 13). The encrypted document is encoded using an information-optimal erasure code algorithm (line 14), reducing the size of data that will be stored in the cloud providers. A private key is created, based on the ID (line 15) based on the procedure of the Session II-B. For each part of splitted encrypted key and encoded encrypted data, we provide hashes (lines 17 and 18) and then sign it (lines 19 and 20) using the Hess Signature scheme. A data block will be created, gathering all necessary information to store the document (line 21). The data block will be sent to the cloud providers (line 22) and the entry of this storing is sent to the access control (line 25).

The Read Data (Figure 3) first authenticate the user and asks the access control for the document metadata (line 3). The last version is chosen (line 4) and the ID is composed (line 7). A key pair is generated from the ID using BF-IBE (lines 5-6) and the requests for data are started (line 8). The data blocks are requested from clouds (line 9), where each gotten data block is verified about its signatures and hashes using Hess Technique (line 12). After getting the necessary data blocks (line 19), the pieces of the encrypted document are joined using erasure code (line 23), the symmetric key is restored using secret sharing (line 24) and decrypted (line 25), and finally, the original document is decrypted (26).

Algorithm 1 WRITEDATA$(fileName, fileGroup, user)$

1: total $\longleftarrow n$
2: redundant $\longleftarrow m$
3: data_ver $\longleftarrow 0$
4: mt $\longleftarrow queryMetadata(fileName, fileGroup, user)$
5: data_ver $\longleftarrow max(mt[i].ver : 0 \leq i \leq n-1)$
6: new_data_ver $\longleftarrow data\_ver + 1$
7: ks $\longleftarrow generateSymKey()$
8: data $\longleftarrow openFile(fileName)$
9: e_data $\longleftarrow E(data, ks)$
10: id $\longleftarrow fileName + "/" + fileGroup + "/" + new\_data\_ver$
11: pubk_id $\longleftarrow generate\_pub\_key(id)$
12: e_ks $\longleftarrow E(ks, pubk\_id)$
13: enc_ks[0 ..n-1] $\longleftarrow split(e\_ks, total - redundant, total)$
14: enc_data[0 ..n-1] $\longleftarrow encode(e\_data, total - redundant, redundant)$
15: privk_id $\longleftarrow generate\_priv\_key(id)$
16: **for** $(0 \leq i \leq total - 1)$ **do**
17:     data_hash $\longleftarrow H(enc\_data[i])$
18:     ks_hash $\longleftarrow H(enc\_ks[i])$
19:     data_signed_hash $\longleftarrow Sign(data\_hash, privk\_id)$
20:     ks_signed_hash $\longleftarrow Sign(ks\_hash, privk\_id)$
21:     dataBlock $\longleftarrow (id, enc\_ks[i], enc\_data[i], data\_signed\_hash, ks\_signed\_hash)$
22:     ack $\longleftarrow sendStoreMessage(cloud_i, dataBlock)$
23:     **if** $(ack =' ok')$ **then**
24:         dataControlBlock $\longleftarrow (id, user, fileGroup)$
25:         $sendAccessControlMessage(cloud_i, dataControlBlock)$
26:     **end if**
27: **end for**

Figure 2. Write Data Algorithm - Executed by the data owner to encrypted and send the sensitive information to the cloud of clouds.

V. IMPLEMENTATION

We have implemented the protocols in Java and C++. The implementation is divided into three parts: The PKGs Setup, The Distributed Access Control and the Algorithms of Write and Read Data. The PKGs Setup was implemented by Aniket in his work [3] using C++ and modifying the JF-DKG protocol. The work of Aniket uses the pairing-based cryptography library [22]. The necessary communication protocols to make a distributed access control were implemented in java using sockets to send and receive messages. The Write and Read Data algorithms was implemented in C++

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Algorithm 2 READDATA$(fileName, fileGroup, user)$

total $\longleftarrow n$
redundant $\longleftarrow m$
mt $\longleftarrow queryMetadata(fileName, fileGroup, user)$
data_ver $\longleftarrow max(mt[i].ver : 0 \leq i \leq n-1)$
privk_id $\longleftarrow generate\_priv\_key(id)$
pubk_id $\longleftarrow generate\_pub\_key(id)$
id $\longleftarrow fileName + "/" + fileGroup + "/" + data\_ver$
while $(i \leq n-1)$ **do**
  temp_dataBlock $\longleftarrow cloud_i.getDataBlock(id)$
  temp_eks $\longleftarrow temp\_dataBlock.return\_enc\_ks()$
  temp_edata $\longleftarrow temp\_dataBlock.return\_enc\_data()$
  rt $\longleftarrow Verify(temp\_dataBlock.ks\_signed\_hash_i, temp\_dataBlock.data\_signed\_hash_i, temp\_eks, temp\_edata, pubk\_id)$
  if $(rt = true)$ **then**
    $enc\_ks[i] \longleftarrow temp\_eks$
    $enc\_data[i] \longleftarrow temp\_edata$
  else
    return $ERROR$
  end if
  if $(i > redundant - 1)$ **then**
    $Break$
  end if
end while
e_data $\longleftarrow decode(enc\_data, total - redundant, redundant)$
e_ks $\longleftarrow combine(enc\_ks, total - dedundant, total)$
ks $\longleftarrow D(e\_ks, privk\_id)$
return D(e_data, ks)

Figure 3. Read Data Algorithm - Executed by the data receiver to obtain and decrypt the sensitive that from the cloud of clouds.

using the following libraries: pbc library [22] to make the pairing based operations, the gfshare library to make Shamir secret sharing [23], jerasure to encode and decode files using information-optimal erasure code [19] and OpenSSL to make some traditional cryptographic operations [24].

Our implementation was made by modules and it was a proof of concept that could be optimized to achieve better results. We have used the AES algorithm to encrypt files with 128-bit key size. We also have used SHA-1 [25] for cryptographic hashes, BF-IBE [13] to encrypt symmetric keys, Hess Signature to sign the parts of the encrypted keys and data and we used a total of four nodes with a redundant number of two nodes (this number is tied to the PKG Setup of Aniket, which requires a minimum of 3f + 1 nodes, being f + 1 the quorum to recompose the master secret).

VI. ANALYSIS

This section presents quantitative and qualitative analysis about the algorithms and the security of the system. Due to the lack of space in this paper, we can not extend our analysis and make a deeper review of the benefits. We will focus on the main benefits and we will make a superficial analysis about the algorithms and what they solve.

A. Security Analysis

One of the problems pointed out by SP800-144 [26] is the vulnerability of internal attacks and lack of legal support in cases of intrusion due to the geographical location of the servers. To solve these problems, this paper proposes the use of distributed PKGs using the JF-DKG protocol to generate the master key without any of the trusted entities have ultimate control of it. By using this scheme, two parameters are set: t and N where N is the total number of PKGS and t represents the minimum number of parts that must be collected to recover a private key a user. Thus an agent discovering malicious need a total of t shares to recover the secrets, therefore decreasing the chances of success in an attack. Using the protocols of Aniket, we achieve a numerous benefits as: Distributed PKG Setup, Forward Secrecy, Availability of the Public Key, Periodic master-key modification, Secret Share Renewal, Secret Share Recovery, Group Modification and Threshold Modification. All the computation costs of these operations are $O(n^2)$.

To maintain the confidentiality of information, it is recommended to use more parameters to identify the public key of the IBE. One part of the solution is not to bind a key per user and rather bind user groups with documents, thus having a semantic key. This work proposes the use of identifiers containing access rules concatenated with the name of the document and a version of the same. The access rules are checked by the PKGs through access controls that must be done in a distributed and reliable manner . The document name binds the public key to a specific document. The version makes for each modification of this document to have a different public key. Thus, a member who was part of a group and obtained the private key to decrypt a document in a version X does not obtain a consequent deciphering key to a document with version $X + 1$ if he is no longer part of the group. If a user has already obtained the private key, this had access to document content in this version. Therefore, there is no need to reencrypt a document content that was already exposed. If a user has not obtained the private key and get out of a particular group, it will no longer have access to the private key, because the access control check if the member complies with the rules imposed on the classified document.

Using distributed PKGs, secret sharing and erasure codes we can ensure fault tolerance. Thus, this paper proposes protocols based on verifiable shared secret quorum, thus increasing the rigor of the checks widespread parts. The Byzantine fault tolerance and availability are provided with the property that a total of $3f + 1$ PKGs only f may fail and therefore should always consult a total of $f + 1$ responses to verify that the majority of obtained final secrets are equal.

We also use the Hess Signature Scheme to guarantee the integrity of the digest obtained from the encrypted symmetric key and data parties. The use of the same key pair to encrypt the symmetric keys and to sign the digests improve our architecture due to the key granularity and the simple and secure manner to share the IBE keys. The security of the Hess Signature follows from the security of the generic scheme in the random oracle model and is based on the Diffie Hellman Problem in the domain of the used pairing. Using IBE we also achieve chosen ciphertext security (IND-CCA) that is the

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standard acceptable notion of security for a public encryption scheme.

TABLE I. COMPARISON OF DIFFERENT SOLUTIONS AND SOLVED PROBLEMS.

Procedure	**This Work**	Ruj et al. 2011	Zhou et al. 2011	Bessani et al. 2011
Custody of Keys	✓	✓	X	✓
User Revocation	✓	✓	✓	✓
Fault tolerance	✓	✓	X	✓
Reliable Integrity Check	✓	X	X	X

The work proposed here can solve some cloud privacy sharing problems in a simple and safe manner using a set of cloud providers, splitting techniques and as main tool the identity-based encryption. Table I compares the results of the related work to this article.

B. Performance Analysis

With our implementation we could evaluate the performance of the algorithms. We have chosen to evaluate the performance of the algorithms of Write Data and Read Data, mentioned in the Section IV. We did a sequence of simulations using the implementation with different file sizes. We have started with 1Kbyte files to 524288Kbytes (512 MBytes). The tests were executed in a computer with the following characteristics: Processor Intel I3, 4GB RAM with the operational system Linux Ubuntu. We have executed one week each algorithm and evaluated the standard deviation. Among 1 Kbyte and 16384 Kbytes times were unstable, surpassing the 5% standard deviation. However, this is because the size of the files were small. But even files with 1 Kbyte to 16384 Kbytes kept coming times of 300ms. From 16384 Kbytes files, the time began to grow linearly as doubling the size of the file. As shown in the graph of Figure 4, it can be noted that the increase was linear, demonstrating the stability of the algorithms for different file sizes.

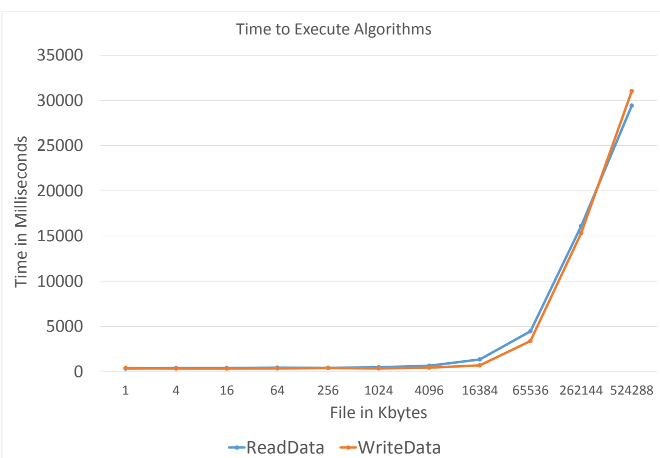

Figure 4. Graph illustrating the performance of Write Data and Read Data Algorithm.

Depending on the application that are using the algorithms, most of the files will be encrypted and sent to the clouds without major performance problems. For example, in applications sharing text files as PDFs with sizes less than 16Mbytes have good performance around 300ms, as in the case of courts of justice (e-justice and e-health). The cases where there is a greater need to share documents, such as medical imaging, where there is a need for high resolutions and video files starting with the 16Mbytes, have satisfactory performance due to file sizes.

VII. CONCLUSION AND FUTURE WORK

This paper proposed an architecture to share sensitive documents in clouds. Because the new trends of computerization of data, large corporations and government entities are increasingly investing resources in cloud computing that has demonstrated economically viable for data storage. These stored documents require encryption support to ensure the confidentiality of sensitive data that should not be accessible to unauthorized third parties.

The proposed model is based on the use of identity-based encryption and provides a secure manner to share sensitive documents between data providers and consumers of information. The main benefits that this work provides are: secure sharing of sensitive documents,reliable integrity check, reducing the custody of cryptographic keys and fault tolerance.

As future work, we suggest to improve the implementation to achieve better performance results and improve the work to provide identity privacy. It is also suggested the validation of protocols in a formal way so that they can be used in practice for large companies and government entities.

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Sensitive Information Protection on Mobile Devices Using General Access Structures

Tomasz Hyla, Jerzy Pejaś, Imed El Fray, Witold Maćków, Włodzimierz Chocianowicz
West Pomeranian University of Technology
Szczecin, Poland
e-mail: {thyla, jpejas, ielfray, wmackow, wchocianowicz}@zut.edu.pl

Marcin Szulga
Unizeto Technolgies SA
Szczecin, Poland
e-mail: marcin.szulga@unizeto.pl

Abstract- **Mobility of users and information is an important feature of information systems that must be considered during design of sensitive information protection mechanisms. This paper introduces the architecture of MobInfoSec system. MobInfoSec is designed to be an information system that allows sharing documents with sensitive information using fine-grained access rules described by general access structures. The system is for users who want to use cryptographic data protection mechanisms to protect sensitive information on mobile devices with a specialized cryptographic module. MobInfoSec will be distributed, modular, and configurable cryptographic access control system to sensitive information that works in a public environment. The system will enable cryptographic protection of sensitive information in accordance with ORCON access control rules. The architecture is designed to be flexible enough, so several business scenarios can be implemented. The paper presents the MobInfoSec system, which the two main goals are to secure mobile information and to release the user from the obligation to monitor any classified information contained in his/her mobile device.**

Keywords-mobile device; sensitive information; access structure; ORCON.

I. INTRODUCTION

Mobility of users and information is an important feature of information systems that must be considered during design of sensitive information protection mechanisms. Mobility of information causes that level of information protection, regardless of its location, must be the same or at least not lower than in other locations. Obtaining such property is not a trivial problem and requires creation of a system that will prevent leakage of access rights to certain information.

Currently, information technology (IT) market does not offer a system that transparently enforces protection of sensitive information collected from various sources and stored on mobile devices. Moreover, the system should prevent its transmission to any third party without the originator consent. The major problem is the disclosure of such information to persons who are not authorized to view this information.

This paper presents the architecture of MobInfoSec system, which will be distributed, modular, and configurable cryptographic access control system to sensitive information. The system will enable cryptographic protection of sensitive information in accordance with Originator Controlled (ORCON) access control rules [1]. A user will be released from the obligation to monitor any information (especially against unauthorized copying). The system will allow building confidence to software and hardware components of popular mobile devices available at the market.

The MobInfoSec design takes into account the European Directive on Electronic Signatures and related standards developed by ETSI and CEN, and most representative of the ISO/IEC directives, recommendations and normative sketches available [2][3].

The paper is organized as follows. Section 2 contains description of the key theoretical elements that are used to build MobInfoSec system, i.e., ORCON access control model and general access. Section 3 contains description of MobInfoSec architecture. The paper ends with summary and conclusions.

II. BACKGROUND

A. Originator Controlled Access Control

The two main goals of MobInfoSec system are to secure mobile information and to release the user from the obligation to monitor any classified information contained in his/her mobile device. In practice, the mobile information should be protected and monitored in accordance with the ORCON requirements [1][4][5][6][7]. In this model, it is assumed that each resource (document) has its owner. A document owner has the authority to manage his/her documents, for example, he/she may have read, write or update right to his/her documents. However, the access control is done by an originator of the document (i.e., the entity which has, on behalf of the document owner, the right to share a document, or entity to which the originator has delegated that right). The owner can determine who can share a document, but the final decision is up to the document originator. It is assumed that any copy of the document must have the same access restrictions as the original document. A user with a read right to a document cannot share or make a copy of a document (i.e., it should be technically impossible).

The implementation of ORCON rules is difficult (and on the IT market practically do not exists any system allowing to manage and secure documents according to ORCON

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model), because in commonly used computers users has a wide access to operating system and memory. When the user has access to memory, he can access decryption key. However, few solutions to that problem exist. One of them, proposed by Yu-Yuan and Lee [1], shows that ORCON requirements might be enforced by a hardware-software mechanism provided by Secret Protection (SP) architecture [5][6], which protects directly Trusted Software Module (TSM). SP architecture consists of a trusted software module, operating at the application layer, and the SP mechanism in the system microprocessor. The authors present a text editor application that contains TSM implementing ORCON. TSM module cannot be bypassed or manipulated by the operating system, because of a direct connection with the SP mechanism. Also, Hoole and Traore [8] present a tool for the exchange of documents according to the ORCON model requirements. However, the solution uses only the software working in the OS environment and because of that, it does not provide complete security, but it is useful from business point of view.

B. *Access structures*

The MobInfoSec system use two main innovative elements to support ORCON model and information mobility requirements: a) a specialized cryptographic module for secrets protection on mobile devices, b) an access policy built over general access structures. The cryptographic module will be the source of trust and will support ORCON requirements. The module will protect directly Trusted Software Module (TSM) compliant with ORCON rules. In turn, the general access structures will allow generating appropriate information protection mechanisms, including group encryption schemes with an arbitrarily pre-defined access structure.

An access structure [9] is a rule that defines who has access to particular assets in IT system. Access structures can be classified into structures with and without threshold [10][11]. Although threshold access structures are frequently used (e.g., the most familiar examples are (*n*, *n*) and (*t*, *n*) secret sharing schemes given by Shamir [12] or by Asmuth-Bloom [13]), the non-threshold structures are more versatile. It is especially visible when the sender of the information defines special decryption rules, which have to be met by the document recipient.

C. *General Access Structure*

Assume that $U = \{ u_1, u_2, \ldots u_n \}$ is a set of *n* participants. The set $\Gamma = \{ A \in 2^U$: *a set of shareholders, which are designated to reconstruct the secret*$\}$ is an access structure of *U*, if the secret can be reconstructed by any set $A \in \Gamma$. The access structure $\Gamma_{(t,n)}$ of the threshold scheme (*t*, *n*) is defined as follows:

$$\Gamma_{(t,n)} = \left\{ A \in 2^U : |A| \geq t \right\} \qquad (1)$$

It is easy to notice that in case of the access structure $\Gamma_{(t,n)}$ of the threshold scheme (*t*, *n*), all users have the same privileges and credentials. Simmons [11] generalized a secret threshold sharing scheme (*t*, *n*) and gave the definition of hierarchical (multilevel) and compartmented threshold secret sharing. In an approach, in contrast to the classical threshold secret sharing, trust is not distributed evenly among the members of the participants' set *U*. Multilevel access structures are particularly useful in organizations with a hierarchical structure and compartment access structures might be used in the cases that require the consent of various parties. Both structures are multilateral access structures, which mean that the set of participants is divided into several subsets and all participants belonging to the same subset have an equivalent role.

When it is possible to implement the access structure Γ, we say that the structure is useful. An example of the access structures realization is the approach proposed by Benaloh-Leichter [14]. However, the application of access structures to build a group-oriented decryption scheme is effective only when it is possible to reuse shares being in possession of participants. Solutions to meet this requirement are discussed in the work [4][15]. In this paper (Section III), the MobInfoSec system is based on the approach combining certificate public-key cryptography with general access structure [16][17].

D. *Our contribution*

The main issue related to the ORCON model is of an architectural and implementation nature. In the case of the MobInfoSec system, the architecture supporting the ORCON model is based on the following innovative elements:

- a specialized SP cryptographic module;
- an access policy built over general access structures and assertions confirming permission to access sensitive information;
- a group encryption scheme, in which the decryption operation is preceded by a strong mutual authentication between the SP modules involved in the sensitive information decryption.

The components mentioned above are partially consistent with a solution shown in [5][6]. In contrast to [5][6], MobInfoSec system uses a group encryption scheme based on a general access structure [16][17] and on a single secret stored in each SP module. Access structures might have different topologies (e.g., threshold, hierarchical) and may also change over time. The SP module can be used by its owner only after two conditions are fulfilled:

1. all users belonging to the privileged shareholders set, which plays the combiner role, are authenticated with their SP modules.
2. access rights authorization of the SP owner to sensitive information, which is going to be decrypted, is positive.

III. MOBINFOSEC SYSTEM

This section contains brief description of general MobInfoSec architecture followed by description of its subsystems. The subsystems are divided into three categories: subsystems that work on server-side of the system (at service provider site), subsystems that are used

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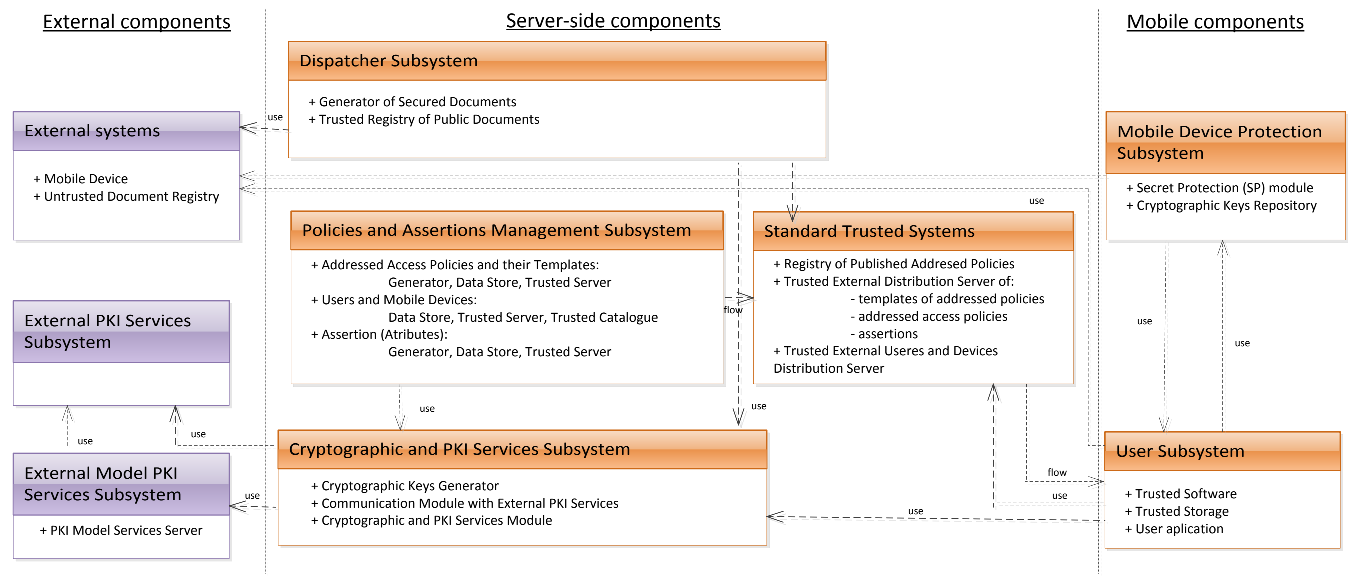

Figure 1. MobInfoSec high-level architecture

by mobile users and external subsystems which include services used by MobInfoSec.

Architecture of MobInfoSec system consists of six logical subsystems connected with three subsystems in external environment (Figure 1). Physical elements of individual subsystems might be shared. Deployment of subsystems into physical elements will depend on the target number of documents that system will manage and on the existing services used by a MobInfoSec provider.

The MobInfoSec consists of the following subsystems:

- User Subsystem;
- Mobile Device Protection Subsystem;
- Dispatcher Subsystem;
- Standard Trusted Systems;
- Policies and Assertions Management Subsystem;
- Cryptographic and Public Key Infrastructure (PKI) Services Subsystems.

A. Server-side components

Policies and Assertions Management Subsystem (PAMS), Standard Trusted Systems (STS), and PKI and Cryptographic Services Subsystem (PCSS) will be implemented on server-side of the system, probably in cloud environment. Moreover, in basic MobInfoSec version Dispatcher Subsystem (DS) will be implemented on the stationary part of the system.

1) Policies and Assertions Management Subsystem

PAMS contains several components that provide key features and can be divided into three categories. The first is related to management of targeted access policies and their templates. It includes generation, storage and distribution. The second group of functions is related to management of users and mobile devices. The third category contains functions related to assertions (attributes) management. This subsystem only distributes the data to Trusted Standard Services and is not available directly for mobile devices. The subsystem stores targeted access policies and targeted access policies templates, users and devices information.

2) Standard Trusted Systems

During decryption process the user system (located on a mobile device) need information, for example, about other devices or other users' assertions. The STS is the source of that information via the trusted components providing the following features:

- distribution of addressed access policies and assertions from PAMS to mobile devices (user systems);
- distribution of trusted devices and trusted user IDs to mobile devices (user systems);
- central users and devices authentication (e.g., MS Active Directory);
- provision of secure access to cryptographic services and PKI services.

3) PKI and Cryptographic Services Subsystem

This subsystem is a service provider for the authentication and encryption schemes. It will be integrated with existing PKI services.

4) Dispatcher subsystem

DS is used to generate targeted access policies and to encrypt documents with sensitive information in accordance with those policies. Generated policies are published in the repository located in STS. An encrypted document linked with a target access policy is published in External Subsystem in an untrusted document registry.

B. Mobile components

User Subsystem (US) and Mobile Device Protection Subsystem (MDPS) are two logical subsystems that are located in Mobile Device.

1) User Subsystem

US contains components that perform authentication and authorization of users and mobile devices and distributes access policies to the mobile devices as well. Finally, US enforce access policy in the case of decryption. Additionally, in US may be located a trusted or untrusted (produced by external suppliers) application that presents the data subjected to access policy. US contains the trusted software configuration data set. The integrity of trusted applications and trusted data sets is protected by MDPS. If a trusted code requires using specific cryptographic keys, it receives them from MDPS through Secret Protection module.

2) *Mobile Device Protection Subsystem*

MDPS contains a dedicated cryptographic module called SP module. The module is a source of trust (at various levels depending on SP type). SP protects directly trusted US components implementing ORCON rules. This protection is possible by controlling the integrity of the code and configuration data. Different possible variants of specialized SP module are considered, including:

- a software SP module;
- a software SP module supported by the device with a built-in crypto processor (e.g., smart card with a crypto processor);
- a software token SP supported by the safety mechanisms built in mobile devices, e.g., laptops with TPM modules, tablets, and smartphones.

SP module should provide: the protection of cryptographic keys; building confidence to software components in the device; the authentication of the mobile device; the protection of information exchange between the device and the network environment where there are other devices of this type.

The whole device is managed by a trusted program module (Trusted Software), which has exclusive access to the functionality provided by SP module and to a reliable data storage module, i.e., TSM module. SP module should provide the following functionalities:

- TPM functionality, i.e., two keys (one for authentication and one for storage), the encryption and decryption keys from the store;
- registers and a code needed to verify the integrity (i.e., the integrity of mobile device hardware, system components and trusted code);
- the code that forces certain behaviour in the case of incorrect integrity verification.

SP module must be unambiguously assigned to the mobile device. Moreover, in its most secure form, a transfer of SP to another device should be impossible. From the business point of view it should be permitted to reset the connection between SP module and mobile device, or even more: the ability to derive trust from one SP module to many devices. This means that one SP module, which is owned by the user, can protect all mobile devices in the local range.

C. *External components*

External PKI Services Subsystem provides PKI services, and External Model PKI Services Subsystem provides PKI services which are not available in External PKI and Cryptographic Services Subsystem and are necessary for the functioning of new algorithms and protocols. External Model PKI Services Subsystem is not a part of MobInfoSec system and belongs to its environment.

External Systems subsystem contains untrusted mobile device that can be an untrusted device vulnerable for attempts that tamper its integrity. The mobile device is a platform that can be used to place dispatcher and user subsystems. Mobile Device Protection Subsystem (containing SP module) is integrated with a mobile device. Another system in external systems is an untrusted document registry**.** The untrusted document registry contains encrypted documents. It might be public http or ftp server or services intended to store files in a cloud, e.g., Dropbox.

IV. CONCLUSION AND FUTURE WORK

The main role of MobInfoSec is to secure documents that contain sensitive information. The system is designed to work in mobile environment. This is especially important when the most of the computer’ users use mobile devices. The architecture is designed to be flexible enough, so several business scenarios can be implemented. The system is intended, for example, for small companies that want to protect they business information, as well as for healthcare organization to allow physicians to work with medical documentation outside the healthcare site.

The most difficult part in the creation of MobInfoSec is the design of Secret Protection module. From the information security point of view, it will better to design a new Secret Protection device from a scratch. However, it could be difficult to connect it with popular mobile devices and development costs would be initially too high. This is the reason why the system is targeted to use security mechanisms available in current devices or in devices that are easy to connect. The most difficult requirement to fulfil is prevention against unauthorized dissemination. This requirement implies that a special trusted application is needed for document viewing. The popular applications can be used, obviously after they are adjusted to MobInfoSec requirements and certified by a local system operator.

MobInfoSec should achieve commercial status in 2015, preceded by a prototype in 2014. Current works are carried out in two areas simultaneously. The first area is related to authentication and authorization issues and the second one to group encryption algorithms.

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IT Service Management in Multiple Actor Network: Service Support from Customer's Point of View

Antti Lahtela
Development and Steering Unit for the Local Register Offices
Regional State Administrative Agency for Eastern Finland
Kuopio, Finland
e-mail: antti.lahtela@avi.fi

Marko Jäntti
School of Computing
University of Eastern Finland
Kuopio, Finland
e-mail: marko.jantti@uef.fi

***Abstract*—Many IT customer organizations see service support and multiple actor network as a big challenge. Multiple actor network may cause several types of problems between different organizations, such as communication gaps, unclearness in roles and responsibilities, contract related challenges, lost incidents and conflicts in understanding the content of agreements between development and continuous services. The research problem of this study is: how IT service support can be improved from an IT customer organization's perspective? The main contribution of this paper is to describe multiple actor network of IT service support. A case study research method with a single case from Finnish public sector was used as a main research method.**

Keywords-IT Service Management; Multiple Actor Network; Public Sector; Enterprise Architecture.

I. INTRODUCTION

The Finnish Public Sector ICT is under massive change. The changes are based on the *Government Programme of Finland* [1], which is an action plan agreed on by the parties represented in the Government and it sets out the main functions of the Government, also for ICT development:

- The development of public on-line services will be placed under the management of a full-bodied actor.
- Clear goals will be set for productivity improvements.
- The interoperability of public information systems will be ensured through the use of open interfaces and standards.
- The implementation and effectiveness of the project entities of the Electronic Services and Democracy Development Programme will be evaluated.
- Productivity in the public sector will be boosted through better utilisation of business intelligence, more compatible information systems, and by bringing together information management data and procurement resources data in public administration.
- Shared use of public administration information will be facilitated.
- All common functions of the State ICT service centers will be brought together.
- To promote interoperability of information systems, open source standards are used in public administration, which determine the compatibility of information content and IT interfaces.
- Enterprise architecture will be employed, utilizing shared information platforms and shared eGovernment platforms and eServices.

A new *Public Sector ICT Strategy* [2], based on the Government Programme, has been published in year 2013 to provide strategic guidelines for the whole IT management of public sector in Finland. As a result of the new strategy, several new services and organizational changes will be established, for example a new government IT service centre in year 2014.

The implementation of the Public Sector ICT Strategy was chaired by the Ministry of Finance of Finland as an open process including people from public administration, business enterprises and non-governmental organizations. The vision in the strategy reaches to 2020, and policy approaches and measure to the end of 2015. The vision of the ICT Strategy has two main points where the main goals are service improvement and public sector cooperation:

- Services and information required by users are available and usable easily and securely.
- Cooperation of public sector organizations, businesses and users at the leading edge of development.

According to the Government Programme and the Public Sector ICT Strategy, Finnish government organizations and officials must improve their own service production as well as bring services on-line for citizens and corporations. The purpose is to develop organizations' operation more efficient and lead customers to use eServices as a primary service. For the service improvement, a large number of organizations are involved in managing, developing and implementing IT services in public sector. This is mainly due the acquisition policy that the Finnish government practices. Usually the multiple actor network consist of following participants (Figure 1):

- End user organization.
- Customer.

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- Service provider
- Third party service provider

The end user organization is usually the one who receives the delivered services that the customer has ordered from the service provider. The service provider delivers the service and can use third party service providers to build and produce the service. Sometimes the end user organization and the customer are the same. The presented network is often called as a *Customer-Provider Model* where the agreements of the delivered service is made between the customer and the service provider. After the service is delivered, continuous services begin (service management).

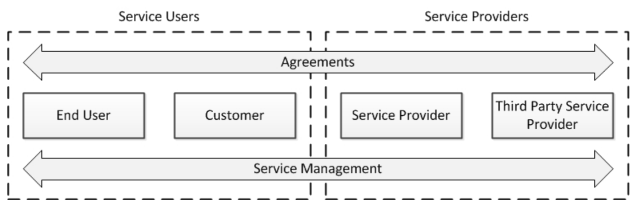

Figure 1. Example of a multiple actor network.

Much has been written about IT service management, help desks and service desk processes. For instance, Caldeira and Brito e Abreu [3] has studied how to use statistical methods to learn from incident classifications. Additionally, Hochstein, Zarnekow and Brenner [4] have provided recommendations how to improve IT management processess based on four case studies. One of the recommendations was to create a company specific IT service management model. Surprisingly, few studies have dealt with customer support in multiple actor network. Jäntti [5] has presented a single point of contact service model as a solution for managing support requests between a customer, a main service provider and a third party service provider.

There are studies that have focused on exploring challenges in customer support. For example, the study of Jäntti, Tanskanen and Kaukola [6] reported incident resolution documentations, incident classification, a lack of knowledge base and the missing knowledge flow as main challenges. Kajko-Mattsson, Ahnlund and Lundberg [7] have presented corrective maintenance maturity model (CM3) where service level management is one of the CM3 modules. The model was tested with four Swedish organizations. Additionally, Lahtela and Jäntti [8] [9] have done research on IT services support both from service provider and from customer point of view. They have studied on challenges and problems in release management process and in service support interface.

Many IT service organizations use *IT Infrastructure Library* (ITIL) to improve IT service management processes. ITIL describes the service lifecycle that contains five core phases: Service Strategy [10], Service Design [11], Service Transition [12], Service Operation [13] and Continual Service Improvement [14]. In the ITIL framework, customer support is divided into several processes, such as incident management, service request management, event management and problem management.

In addition to incident management, there is a process called problem management within IT service management frameworks. While incident management aims to solve incidents as soon as possible, the main objective of problem management is to identify the root cause of the incidents [13]. Thus, problem management is responsible for proactive customer support. Jäntti et al. [15] have studied implementation of problem management from knowledge management point of view. Incident management and service request management are also visible in *Control Objectives for Information and Related Technology* (COBIT) [16]. COBIT is a IT governance framework that describes IT service management processes, metrics, roles, responsibilities and control objectives.

ISO/IEC TS 15504-8:2012 process assessment model [17] can be used to improve the service management process capability and maturity. This is a very useful tool for carrying out process assessments, when the target of the assessment is a single service management process. Compared to *Capability Maturity Model Integration* (CMMI) [18], the assessment can be done in a more light weight way because the assessment can focus even on one process.

The Finnish Government has published several guidelines and instructions on how government officials should manage their IT. *The Public Administration Recommendations* (JHS recommendations) [19] provide information management guidelines for public administration. The JHS recommendations system aims to improve the interoperability of information systems and the compatibility of data in them, to facilitate process development and to make the use of existing data more efficient. The JHS recommendations are approved by the Advisory Committee on Information Management in Public Administration (JUHTA). JUHTA is also the head of the Finnish public sector enterprise architecture where the enterprise architecture work is done by the Public Sector Enterprise Architecture Division. The Division develops and publishes government level enterprise architecture descriptions. For example, the architecture description for eServices [20] is one of the documents that is implemented for government officials to use as a guideline in their own architecture work.

For information security, the Government Information Security Management Board (VAHTI) [21] has established a set of instructions for directing information security measures in central government. Additionally, the board functions as an organ for coordination, cooperation and preparation among government organizations in charge of steering and developing data protection and information security. For open data, eServices and eDemocracy, Finland has organized two separate programs. *The Open Government project* [22] aims to apply for membership in the Open Government Partnership initiative. *The Action Programme on eServices and eDemocracy* [23] develops comprehensive services for citizens, companies and the authorities. The purpose of this program is to enhance service quality and cost-efficiency in the public sector.

A. *Our Contribution*

This case study is a part of the results of Keys to IT Service Management and Effective Transition of Services (KISMET) research project at the University of Eastern Finland, School of Computing, Finland. The main objectives of the KISMET project are to develop, share new ideas and experiences regarding IT service management in the network of organizations,

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and strengthen the research and know-how of IT service management in Eastern Finland. The case study was made in cooperation between KISMET project and Regional State Administrative Agency for Eastern Finland, Development and Steering Unit for the Local Register Offices in year 2013.

Our contribution is to present a case from Finnish public sector where IT service management and IT service support are managed in a multiple actor network. The remainder of the paper is organized as follows. In Section 2, the research methods of this study are described. In Section 3, we present the results of this study. The conclusions are given in Section 4.

II. Research Methods

The research problem of this study is: how IT service support can be improved from an IT customer organization's perspective? The research problem was further divided into following research questions:

- How service level agreements (SLAs) should be designed in order to support IT service management in multiple actor network?
- What kind of communication strategy should be established between a customer and a service provider?
- How service support roles and responsibilities should be defined both from a customer's and a service support provider's point of view?
- How to decrease the number of support requests (incidents and service requests)?
- Which issues must be taken into consideration while creating IT agreements?
- How to deal with organizational changes that affect the service provider model?

A case study research method with a single case was used as a main research method. Our case organization is a small unit that manages IT for another organization. The case organization's main responsibilities include IT agreements, IT development and IT management. The service providers consist from different government IT service providers and third party IT service provider companies.

Our case organization was selected for this study because the researcher had easy access to the research data. Additionally, the case organization was a representative case of Finnish government organization that deals with multiple service providers and acts in different roles (customer and service integrator for another organization).

A. Case Context

Figure 2 describes the case study context and how our case organization is positioned from service, steering and development perspective. The case organization is located inside of another government official in a ministry administration. The main objective of the case organization is to develop, steer and control another government agency that provides services for Finnish citizens. Other main stakeholders in the case context are different ministries and agencies inside ministries' administration. The stakeholders are mainly content and substance developers. Other coordinators are the Finnish legislation, strategy and IT strategy (organizational level strategies are made in cooperation with all the context organizations), enterprise architecture, guidelines and recommendations as well as other architectures that effect our context.

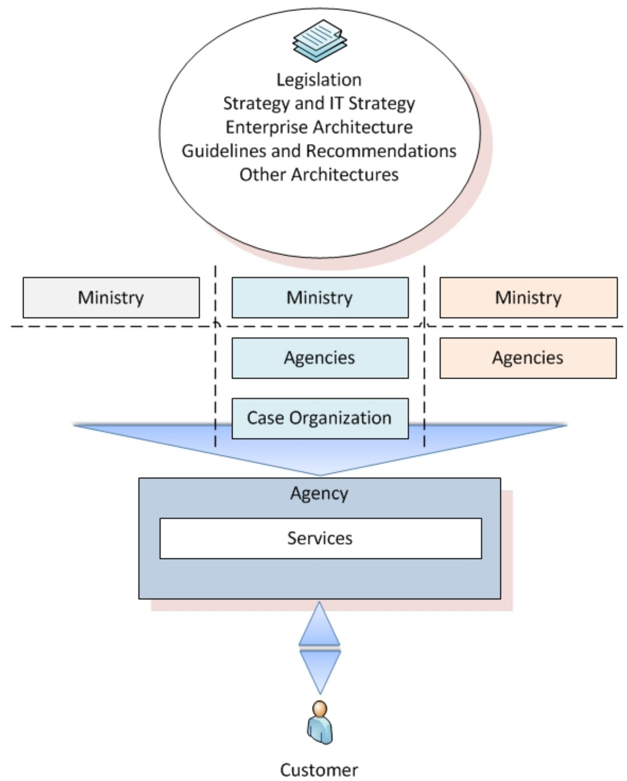

Figure 2. Case context.

Figure 3 shows how IT services are managed in our case. The case organization, where the IT management is situated, executes all the agreements with the service providers from delivered servicers that are provided for the agency and the end users. For example, a individual SLA is made between case organization and every different service provider.

First, incidents and service requests are sent to the internal IT service provider (also a government agency), which will handle them. Secondly, the internal IT service provider can escalate incidents and service requests to a third party service provider depending on the case. Usually, the internal IT service provider deals all the cases concerning to infrastructure services (workstations, networks, usernames, etc.) and the third party service provider deals with software services (coding, major problems in the system, development work, etc.). In some cases, the third party service provider can also be another government agency.

B. Data Collection and Data Analysis

The research data was collected by using the sources of evidence defined by Yin [24]:

- Interviews: customer, end user and service provider interviews.

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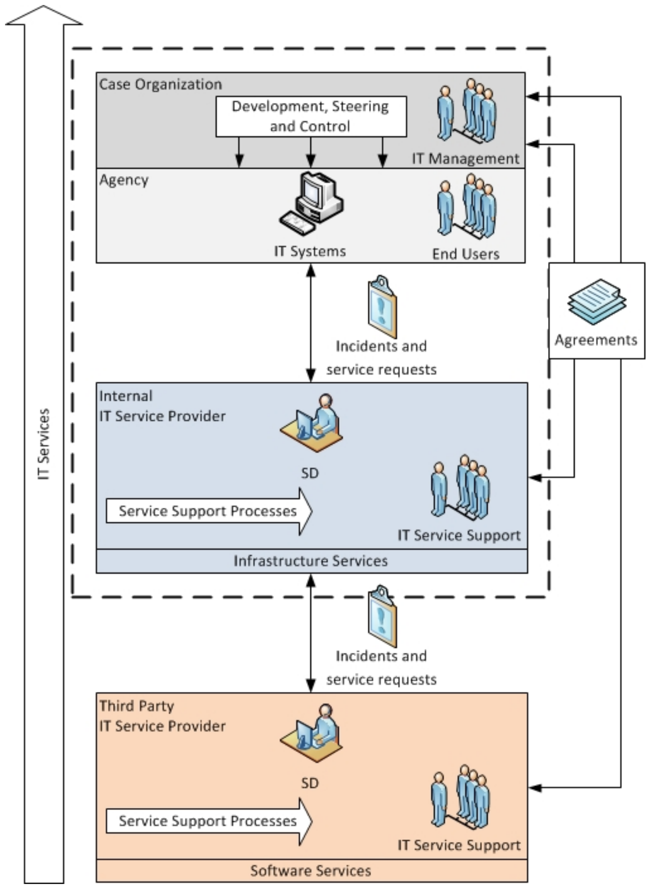

Figure 3. IT service management in our case.

- Documentation: enterprise architecture descriptions, IT agreements, service definition documents, etc.
- Physical artifacts: access to customer's and end user's IT systems.
- Observation: participative observation in the case organization.
- Archives and records: access to document archives of the case organization.

The collected data was analyzed with the KISMET research project team and a member from the case organization. A list of identified challenges was created and each challenge was analyzed for improvement suggestions.

The within-case analysis method was used to analyze the case study material. Analysis of the data was based on the following categories: SLAs, communication strategy, service support roles, proactive service support, IT agreements, and organizational changes.

III. RESULTS

The main contribution of this study was to describe multiple actor network of IT service support and how it can be improved. The following six major challenges and improvement suggestions from an IT customer organization's perspective were identified and presented based on the data analysis of the case:

1) **SLAs**
Challenge: The case organization creates individual SLAs between every service provider. This is a major challenge when tickets are escalated from one service provider to another because it is almost impossible to manage that the service process will be executed according to the agreed service levels.
Improvement Suggestion: All the SLAs should be collected and documented on a single document, which includes an end-to-end service process description with agreed service levels. Thus, it is much easier to monitor service levels and support IT service management in a multiple actor network. SLAs may cover the service level objectives, such as target response times and target resolution times.

2) **Communication Strategy**
Challenge: The communication between all the actors of the network is difficult to manage. There isn't any specified strategy on how to perform communication (how, when, where and to who) between participants. Usually, the communication is done in a hurry without any proper plan. During the study, we observed that the communication related to incident resolutions was not systematic. This was a problem especially during the diagnosis and solving of high priority incidents and holiday times.
Improvement Suggestion: Correct information for all the participants in a multiple actor network needs a good communication strategy with clear escalation points (can be implemented, e.g., into Excel file). The strategy should include clear definitions on how to perform communication with various actors (e.g., application/service, contact person name, email, phone, holiday times and work responsibilities). Managing the feedback, related to services, should be part of the communication strategy. In IT service management, managing the feedback is part of the Business Relationship Management [11].

3) **Service Support Roles**
Challenge: Who is responsible of the service? Usually, this should not be a problem, but environment of different actors can make it very challenging. It can be hard to be responsible from a part of the service if the whole service is developed in cooperation between many providers. From the customer's point of view, it can be demanding if the end user and the customer are different agencies. Additionally, from the perspective of the internal service provider, which is always the first line support and acts as a service desk, it is difficult to coordinate services that are provided by a third party service provider. Following roles were identified regarding customer support: service managers, customer service managers, developers, process managers (incident, problem, change and release managers), help desk and service desk workers, specialists, end users, customers, project managers and service owners.
Improvement Suggestion: The whole network of services and its participants has to be described and

documented. Previous studies [5] have used UML sequence diagrams to model complex interactions between actors of customer support. In order to document the roles and responsibilities, a service based matrix can be used to show the different roles and tasks for all participants. An example of this table is given in Figure 4 to show what kind of attributes the table can contain. The table is presented in organizational level.

4) **Proactive Service Support**
Challenge: The number of service desk cases increases rapidly, because end users and the customer report every single defect or incident to the service desk. There isn't any specific proactive service support for the end users or the customer where they could solve minor problem by themselves. This leads to unnecessary tickets and extra work.
Improvement Suggestion: To promote proactive support, the service provider organization could implement a knowledge base that contains instructions, guidelines and workarounds to known errors. The service provider must instruct end users and the customer to provide required details on incidents and service requests, such as printer numbers and names. For simple tasks, such as lost passwords and new users, the service provider could give permission for the customer to handle these situations or automate the process of generating new passwords. Because of every contact to the service desk (incident or service request) leads to opening a support request to the service provider's IT service management system, one should identify the types of unnecessary requests, such as information requests on the progress of incident resolutions.

5) **IT Agreements**
Challenge: IT agreements are usually a stack of multiple different documents without clear management. It is complicated to get a general overview on what has been agreed about the service as well as between development and continuous services. Additionally, one customer might have dozens of agreements, which may cause difficulties (delay in processing) in classifying the incidents and service requests in the service desk.
Improvement Suggestion: IT agreements management process should be established in order to improve the management and to clarify agreements between development and continuous services. This requires communication between several process managers, such as service level management, sales management and supplier management. Our simple model for managing IT agreements is presented in Figure 5.

6) **Organizational Changes**
Challenge: Finnish public sector ICT is under constant change. The Government Program, various productivity programs and the Public Sector ICT Strategy are driving organizational changes for government officials to save costs and to increase effectiveness. This has a big influence for the public sector, which struggles to improve and deliver services for the citizens.
Improvement Suggestion: Every organizational change has to go through the change management process. Changes, that have major impacts, should be considered as major changes that need more careful planning and much more effort than minor changes. For the end user, organizational change should go smoothly without any massive impact to the service. Correct communication between the participants supports this.

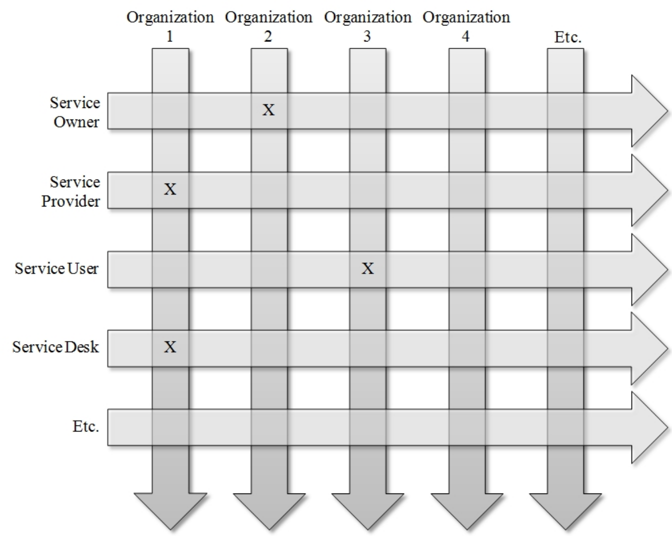

Figure 4. Matrix for roles and responsibilities in organizational level.

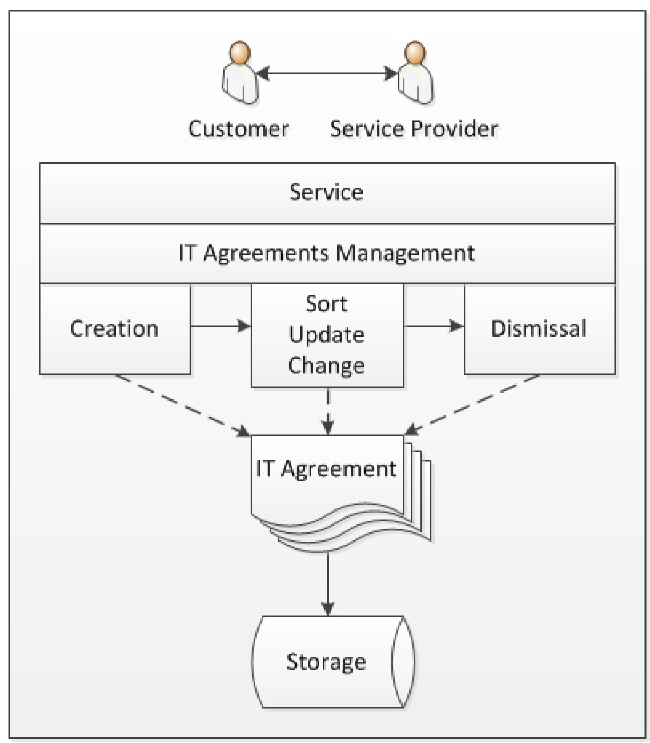

Figure 5. IT agreements management process.

One major improvement suggestion, for developing operation in a multiple actor network, could be the use of the *Enterprise Architecture* (EA). EA is a method that helps to describe the operational environment and to manage the development of services as a whole [25]. The method can help getting the big picture on how IT service management and IT service support is established and to find potential environment and operational problems. The Finnish public sector uses a EA framework called *JHS 179* [19] as a recommendation for describing their present state and future environment from operation, information, information system and technology point of views.

IV. Conclusion

This paper and the included case study aimed to answer on the research problem: how IT service support can be improved from an IT customer organization's perspective? The problem was divided into six different research questions. The case itself was build between a Finnish government agency and its service providers. We used a case study research method with a single case as a main research method. The within-case analysis method was used to analyze the case study data.

The main contribution of this study was to describe a multiple actor network of IT service support, present found challenges and bring improvement suggestions to develop operation in the network of customers, end-users and service providers. The following six major challenges and improvement suggestions from an IT customer organization's perspective were identified and presented based on the data analysis of the case: SLAs, Communication Strategy, Service Support Roles, Proactive Service Support, IT Agreements and Organizational Changes. The next task is to implement the presented improvement suggestions and evaluate the results.

We also introduced a recommendation to use EA as a development tool in multiple actor networks. Further work could focus on studying how EA and IT service management are related and how EA is used to improve IT service support.

Acknowledgment

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Using Reference Traces for Validation of Communication in Embedded Systems

Falk Langer, Erik Oswald
Fraunhofer Institute for Embedded Systems and Communication Technologies ESK
Munich, Germany
Email:{falk.langer,erik.oswald}@esk.fraunhofer.de

***Abstract*—This paper addresses the problem of evaluating the communication behavior of embedded systems. An important problem is missing, wrong or incomplete specification for the interaction in the distributed system. In this paper, a new approach for evaluating the communication behavior based on reference traces is introduced. The benefit of the approach is that it works automatically, with low additional effort and without using any specification. The introduced methodology uses algorithms from the field of machine learning to extract behavior models out of a reference trace. With the presented algorithm, the complexity of the learning problem can be reduced significantly by identifying parallel execution paths. The efficiency of the proposed algorithm is evaluated with real vehicle network data. At this data the self-learning algorithm covers up to 69% of the behavior from the presented trace.**

Keywords—embedded system validation, testing procedures,network trace analysis, self-learning test methods

I. INTRODUCTION

This paper focuses on test and validation of the communication behavior from embedded systems. In systems with highly distributed functionality like it can be found in modern car's electronics, the communication behavior is an important aspect on system validation. At a cars development cycle, it is important to analyze the network traffic in a fully assembled car. Even if all single electronic control units are tested exhaustively, a significant portion of remaining bugs resulting in errors or malfunction is lately found at real driving tests. Because network traffic represents the internal behavior of a distributed system, its analysis can help to detect possible bugs earlier and faster. But especially on system level test it is not easy to rate about the correctness of communication at the network.

The most important problem of ensuring the correct interaction at system level is missing, wrong or incomplete specification of the interaction of functions in the distributed system (compare [1] and [2]). There are many works of research in progress that tries to improve the process of creating system specification, with the goal of building better test cases for validating the communication on system level. Nevertheless it is still an extensive process to get sufficient test models.

In this paper, a new approach for evaluating the communication behavior automatically, with low additional effort and without using any specification will be presented. The goal is to detect problems early, best before detectable errors or malfunctioning occurs. The proposed approach shall help to detect these remaining bugs faster without a significant increasing of testing effort.

This paper is structured as follows. Section II gives a short overview of the state of the art and the gaps that the proposed approach addresses. Section III describes the expected usage, benefit and outcome of the approach. Section IV provides the technical background of the learning problem. In Section V, the methodology for identifying parallel execution paths in traces is discussed an evaluated and Section VI offers an optimization. The paper closes with section VII that presents the conclusion and future work.

II. OVERVIEW AND RELATED WORK

The car's network can be seen as a closed but distributed system. The network behavior mostly depends on sensors and actors and its input or output, which results from different environment or user interaction. Therefore, in the most cases it is only possible to observe the communication behavior. Because of the nature of a closed system, it is not possible to stimulate a behavior on network level and evaluate the response. To rate about the correctness of network communication it is necessary to build more or less passive observer models. To build such models it is important to have a detailed description or specification of the communication protocols between the applications. In difference to well-known protocols like TCP/IP, this is a kind of *meta* communication protocols because they are mostly not noticed as a protocol. In [3], *meta* states that a communicating systems can internally take place, are described and it is pointed out that this meta states are often the cause of malfunction because they are mostly not known.

Basically, it did not surprise that one of the main causes of malfunctions detected at system level test is wrong, incomplete or missing specification (compare [1] and [2]). Therefore, the focus of most research projects working on testing network behavior is to enhance the specification. The key aspects in research are requirement engineering and its interaction with test methods. For this reason a popular approach is to use additional description languages to describe the systems behavior more accurate and build better test cases([4, 5]). Another approach for getting better specification is the automatic update of specification from already developed systems ([6, 7]). This shall help to get the specification up-to-date and provides the tester an overview about yet not specified behavior. Obviously, this approach stands in contradiction to top down software engineering methods like the V-Model ([8]), which is very popular in embedded systems development. Nevertheless incomplete specification is an unavoidable problem in software engineering and because of this reason nearly all methods that help to close this gap, will enhance software quality.

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In this background, the focus of the proposed method within this paper, is the analysis of network communication without the need of specification as it is described in [9]. The communication is recorded within a trace which can be analyzed offline. Therefore, a trace represents all data observed in the network within a given time. The proposed method basically uses reference traces as replacement for missing specification. A reference trace represents the allowed behavior or the possible states a system can take place at the surrounding influences provided to the system at recording time. If the reference trace represents most of the possible behavior of the system, it could be interpreted as the normal or norm behavior. This comes close to the idea to use examples as specification like it is described in [10], but differs in the kind the specification is represented.

III. EXPECTED OUTCOME

The goal of this work is to construct a method that allows a qualitative comparison between the reference trace and newly recorded traces with respect to the represented system behavior. The essential outcome of the proposed procedure is the awareness, that the newly recorded network trace represents a new system behavior, which is not represented within the reference trace. If such a behavior is recognized, the method outputs a trigger or some equivalent information to the tester. At this point two potential expectations about the tested network behavior can be made: 1) A newly implemented or just jet not observed behavior was found, or 2) A bug in in communication behavior is detected. Just at this point a system expert has to decide if the proposed method detects case 1) or 2). Surely it is not possible to detect bugs, which are already within the reference trace included, but if no other tests detect these bugs und these bugs did not lead to malfunction, it is not sure if it is a bug or just unspecified behavior.

The described scenario has some analogy to regression tests. But at system test level, regression tests are usually not interpreted or executed on network level. On network level it is only possible to observe some kind of internal system reaction as consequence to external test stimuli. The internal behavior represented within a network trace is hard to interpret. As mentioned before, this is mostly done by using passive reference models ([11, 12]). Because these models are hard to build, in many cases only search of negative examples is done on the network trace. This is mostly a search of error codes or bad sequences, which are known from previous bugs.

With the proposed method a kind of reference model shall be extracted from the reference trace. In comparison to manually build reference models this method comes for free and can be applied without any specification. Therefore, the proposed method shall help to improve the evaluation of network communication at system tests.

IV. THE LEARNING PROBLEM

This section describes the algorithmic foundations and the basic functioning of the proposed self-learning trace analyzing approach. The goal of the approach is the qualitative evaluation of network traces, with the focus on interpreting the sequence of observed events. It was pointed out above that such sequences can be potentially described by protocol automata which are finite state machines. This leads to the basic assumption that a network trace can be described by one or more finite state machines. According to the intention of learning reference models, it is only needed to accept the trace and not to generate it. So, one can use the definition of a 5-tupel acceptor automaton for describing the network trace:

$$A := (\Sigma, S, s_0, \sigma, F) \tag{1}$$

Where: Σ is the input alphabet consisting from events $\in \Sigma$, S a finite set of non-empty-states, s_0 is the initial state with $s_0 \in S$, σ as state transition function with : $\sigma : \Sigma \times S \mapsto S$ and F the set of final states $F \subseteq S$.

It can be pointed out, that a state in A is represented by a sequence of events $\varphi \rightarrow S$ with $\varphi = e_1, e_2 \dots e_n$ and $e \in \Sigma$. This means that the a learned reference model must predict for any given sequence $\varphi = e_1, e_2 \dots e_n$ the next event e_{n+1}. This can be repeated in an unlimited manner that $n \rightarrow \infty$, which means that a sequence is potentially endless.

Another important expectation about the network behavior results from the paradigm of parallelism in distributed systems ([13]). This results in the expectation that there exist several independent automata A_i with disjoint input alphabets. A trace would then be observable by an automata A^*, which is a product of all automata A_i

$$A^* = A_0 \times A_1 \times A_2 \times \dots A_i \tag{2}$$

with $\Sigma^* = \uplus \Sigma_i$ (non overlapping alphabets)

These assumptions describe a basically system hypothesis for the network trace. With this hypothesis it should be possible to describe the learning problem, which is the first step to find applicable learning algorithms. If this hypothesis is correct the network trace would consist of several sub traces describing the execution path of a single automaton A_i.

For learning sequences even if they are infinite long, a lot of algorithms can be used. For example neural networks ([14,15]), Markov chains ([16, 17]) and Angluin Style automata learning ([18]) algorithms are usable. It was shown that the fundamental problem by applying these learning algorithms, is the parallelism resulting from (2). This leads to an exponential growing of complexity with the number of parallel executed automata. In ([18] this effect was shown by using a CAN trace from a cars powertrain.

With these results it can be pointed out that the major problem for learning behavior or reference models, is the identification of parallel execution paths within the reference trace. If it is possible to extract group off events, where each group belongs to an independent executed automata, the complexity of the learning problem can be reduced exponential.

V. IDENTIFYING PARALLISM

The grouping of events can be seen as a clustering problem. Clustering means to identifying groups of elements with most similar properties. But what are the properties of events belonging to the same execution path? It is most likely the property that these events are generated by the same automaton. But this property is not directly measurable. Because of that, the properties of an event need to be checked if they give an advice to that not measurable property.

There are only two properties of an event that are somehow usable for this problem. The first one is the recording time of the event and the second one is the information theoretic probability of an event. For clustering the usage of these two properties is not straight forward. In the following, the usage of the recording time within clustering algorithms will be introduced. The proposed method basically interprets the timestamp by applying them to a behavior model and extract new features that are more suitable for clustering methods. This kind of process is called feature extraction or feature construction [19].

A. Feature construction based on timestamps

The absolute value of the timestamp of an event is not usable for clustering because a dedicated event can occur several times in a trace. But for clustering it is important to calculate a distance between two different events. Additionally the calculated distance must be related to the problem of identifying groups of events that belonging to the same execution graphs. To interpret the timestamps in that way, a hypothesis of the system is needed, that explains the values of the timestamps. For this reason the definition of acceptor automata (1) needs to be extended to deal with time. Such automata are called timed automata and are invented by [20]. Timed automata extend the classical automata with a finite set of clocks and the transition function with are enriched with guards that control the time when a transition is allowed to be executed. A timed automaton $\mathrm{A_t}$ can be defined as follows:

$$A_t := (\Sigma, S, s_0, X, \phi, F) \tag{3}$$

$\Sigma, \mathrm{S}, s_0, \mathrm{F}$ are equivalent to (), X is a finite set of clocks, and ϕ replaces the transition function σ. ϕ: Σ x S x G(X) $\mapsto$ S were G(X) describes the timing guard.

This definition is a very general description of timed behavior. To use a model of timed automata for feature construction it is necessary to made some simplification or assumption of a typical use. For this reason it is argued that every transition will take place in in a fixed time. Even if there is more flexible timing behavior allowed or thinkable, it could be expected that in an implementation in most cases a fixed timing values will be used.

With this simplification it is possible to argue about the expected timing behavior of a dedicated event. For a continuously executed automaton where the transitions add a fixed delay between emitted events, it can be expected that the delay in a fixed loop is constant. If the automata have different ways for an execution loop, different but countable delays between the emitting of the same event can be expected. For example consider the automaton in Fig. 1, there the time period $\mathrm{t_P}$ between two occurrences of event **a** can be $\mathrm{t_P} = \mathrm{t_b} + \mathrm{t_e}$ or $\mathrm{t_P} = \mathrm{t_b} + (\mathrm{n}+1)\mathrm{t_e} + \mathrm{n}(\mathrm{t_c} + \mathrm{t_d})$ and for event **b** it can be $\mathrm{t_P} = \mathrm{t_e} + \mathrm{t_a}$ or $\mathrm{t_P} = \mathrm{t_a} + (\mathrm{n}+1)\mathrm{t_e} + \mathrm{n}(\mathrm{t_c} + \mathrm{t_d})$. For this example one can see that there would be a good chance that there are time periods of event **a** and **b** that are in the range of $\mathrm{t_P} = (\mathrm{n}+1)\mathrm{t_e} + \mathrm{n}(\mathrm{t_c} + \mathrm{t_d})$. That means that the automaton takes sometimes the lower path through event **c** → **d** → **e** before it enters the upper path **a** → **b**.

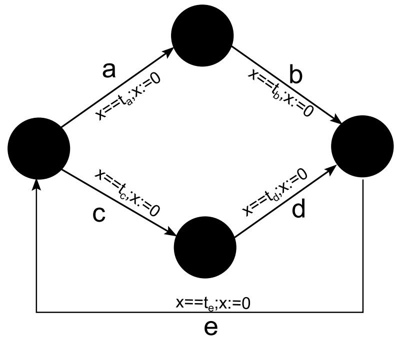

Fig. 1. An example of a looped automaton

If this assumptions are correct, it is supposable that events that belonging to the same automaton having same or similar frequency components. These frequency components should be better usable to calculate a distance between two different events. This distance is later usable for clustering reasons.

B. Calculating the frequency components

A dedicated event can occur several times within a trace. Because of this, an event is located in time. It describes more or less a signal function with different frequencies (see Fig. 2). To get the main frequency components out of the signal curve the most common algorithm to use is the Fourier transformation.

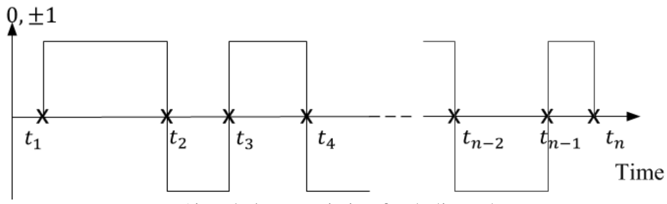

Fig. 2. Signal characteristic of a dedicated event

After applying the Fourier transformation to the signal curve, the result looks like the chart in Fig. 3. Basically after applying the Matlab function "*findpeaks*" several times, until less than 10 frequency components are left, the results looks like in Fig. 4. In the next steps these frequency components are used for calculating a distance between different events.

C. Distance metrics

For clustering a simple density based hierarchical clustering method is used. To get good clustering results the usage of the right distance metric is essential. For this reason four different metrics are tested within the evaluation.

The well-known distance metrics Euclidian, Manhattan and the Hamming distances are used for a first analysis. The definitions of this metrics are as follows:

- Euclidian distance

$$d_E(x,y) = \sqrt{\sum_{i=1}^{max_{freq}} (freq_{x,i} - freq_{y,i})^2} \tag{4}$$

- Manhattan distance

$$d_M(x,y) = \sum_{i=1}^{max_{freq}} |freq_{x,i} - freq_{y,i}| \qquad (5)$$

- Hamming distance

$$d_H = \sum_{freq_{x,i} - freq_{y,i} > \varepsilon} 1 \,;\, i = 1, \ldots, max_{freq} \qquad (6)$$

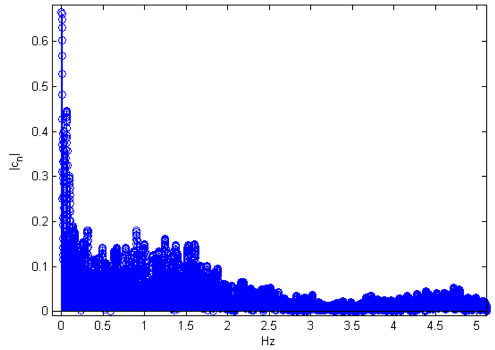

Fig. 3. Unfiltered frequency spectrum

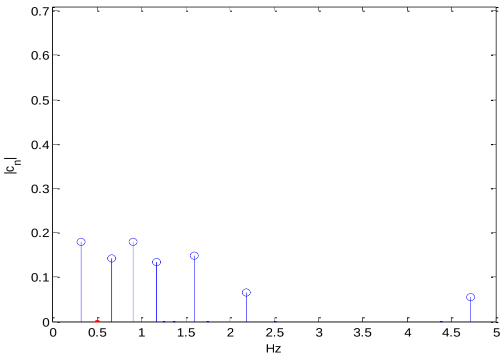

Fig. 4. Filtered frequency spectrum with only 7 main frequencies

It was mentioned before, that if two different events has one equal frequency component the probability that this two events belonging to the same automaton is very high. Because of this, a new distance metric is introduced and will be called in the next *frequency distance*.

This new metric is derived from the hamming distance. The difference to the hamming distance is that it does not care about the ordering of the points that are to be compared. Because for the proposed problem mainly the equality of frequency components is important, not it's ordering.

- Definition of frequency distance

$$d_{freq}(x,y) = max_{freq} - sum\_eqal\_freq_{x,y} \qquad (7)$$

With $\mathrm{sum\_eqal\_freq}_{x,y} = \sum_{|\mathrm{freq}_{x,i} - \mathrm{freq}_{y,j}| > \varepsilon} 1$ and $\mathrm{i,j} = 1, \ldots, \mathrm{max}_{\mathrm{freq}}$

D. Evaluation Criteria

To interpret the clustering results in the manner of identifying parallel execution graphs within a trace, it is important have an evaluation procedure. This helps to rate if the events in one group are belonging to the same execution graph.

As mentioned before the assumption of clustering is, that if events belonging to the same execution graph, they should be generated by the same protocol automaton. It was also shown, that if an automaton is constructed from events that belonging to different independent automata, the resulting automaton has a significant greater complexity. This is because of the exponential growing of possible states, if two or more independent automata are merged to one automaton. This behavior is called the product of automata (2).

If it would be possible to build an automaton, which describes the behavior of a group of events, the complexity of the build automaton should correlate to the quality of the clustering results. The fewer the complexity of the automata, the better the clustering results are.

A good starting point for such a rating mechanism would be the L* algorithm introduced by [21]. This algorithm constructs the smallest automaton describing a given set of sequences. It was shown in [18] that this algorithm can be applied to the given problem were the sequences are located within a trace. On the first glance one would calculate the complexity from the structure of the inferred automaton. But [18] has furthermore shown, that if L* can learn an automaton with a tested sequence length larger than five events within a short time, the resulting automaton will be less complex.

With this background, the evaluation of the clustered event groups is done by applying the L* algorithm. If the algorithm learns an automaton, which can be tested successfully with a test length greater than five events, within a time frame of three minutes, a good event group is argued. A good event group means its events are not generated by independent automata and therefore, the describing automaton of the event group is minimal.

E. Example Application

For proofing the described method of learning behavior models from network traces by identifying parallel execution paths an example with real network data is provided. The network data are recorded at a cars powertrain CAN network within real road tests. The example application takes a reference trace of approximately 13 min driving time.

From this reference trace all discrete events are extracted. A discrete event is represented by discrete signals. That means no measurement values like speed, temperature or similar continues information are used. Discrete events are most likely internal states like the engine status, discrete input values like the position of light or blinker levers and internal protocol values of interfaces. Because on CAN usually most information are sent periodically, duplicated send events are explicitly filtered.

After the preprocessing the CAN-trace, approximately 7500 different events are detected. Afterwards events are deleted that occur less than two times. From the resulting 7170 different events the frequency components were calculated. The overall quantity of events in the reference trace is about 2.4 10^6 events.

The count of events containing to one cluster was limited to 127 events, because the implementation of the L* can only handle that amount of events per automaton. This is not necessary a limitation because in [18] it was shown that the performance of L* drops rapidly with more than 100 events.

F. Discussion of results from sample application

In TABLE I, the count of resulting clusters by applying the different distance metrics are shown. It can be seen that for the given example, a range of 400 to 700 clusters per metric are identified.

The results from the evaluation of the identified clusters with the L* algorithm are shown in TABLE II. As result the highest percentage of the identified clusters that can be inferred to valid automata with L*, offers the frequency distance with 73%. The most dedicated events describe by inferred automata was reached with the Manhattan distance with 46%. Even the coverage of the trace is best with Manhattan distance.

These results could be rated as success. In [18] a hand sorted list of events that describe less than 30% of the trace were used to learn an automaton. The learning process for this set of events was even not successful. With the usage of the Manhattan distance it is possible to describe about 49% of the behavior of the trace without any prior knowledge about dependencies of events.

TABLE I. CLUSTERING RESULTS BEFORE EVALUATION

Distance metric	Results from different distance metrics	
	Count of clusters	***Events per cluster***
Frequency	694	9,7
Euclidian	422	16,4
Manhattan	469	14,7
Hamming	570	12,6
Sum	2155	12,9

Within an additionally executed test, with randomly created clusters, the learning rate of successfully inferred automata was less than 4%. This leads to the expectation that the proposed clustering methodology gets a significant improvement for learnability of network traces with L* algorithm.

VI. IMPROVING TRACE COVERAGE

The presented clustering method extracts non overlapping events groups from a given set of events. From this event groups only these are usable, that lead to a learnable automaton by L*. For describing the system behavior represented within a trace, the description would be that better the more parts of the trace are considered to be used.

TABLE II. CLUSTERING RESULTS AFTER EVALUATION WITH L*

	Results from different distance metrics			
	Frequency	***Euclidian***	***Manhattan***	***Hamming***
Cluster successfully learned automaton (percentage of found clusters)	506 (73 %)	254 (60 %)	295 (61 %)	316 (55 %)
Ratio of successful clustered single events (abseolute count)	37 % (2,689)	40 % (2,883)	46 % (3,298)	37 % (2,676)
Ratio of event quantity (trace covery) (abseolute count of events)	27 % (678,820)	36 % (917,442)	49 % (1,259,452)	32 % (805,294)

Additionally, there is no strict requirement that the groups of events need not to overlap with each other. It would be quite the contrary if there are overlapping groups usable. Because it could be suggested, that clustering did not lead to a perfect separation in the sense of (2), the clusters will most likely describe only parts of independent automata. If there are overlapping event groups the merging of this group will lead to an automaton with describes a more complex but no parallel execution graphs.

This consideration leads to the conclusion that all identified clusters shall be used for describing the systems behavior. For that reason the results from the former clustering results are merged. The merge in that case is basically done by interpreting the learning results of L* from all clustering approaches. Like it is shown in Fig. 5, the different event groups resulting from clustering are overlapping. For the merge it should not be necessary to build new event groups and infer new automata. Instead the different sets are analyzed and the union set of all identified groups is calculated.

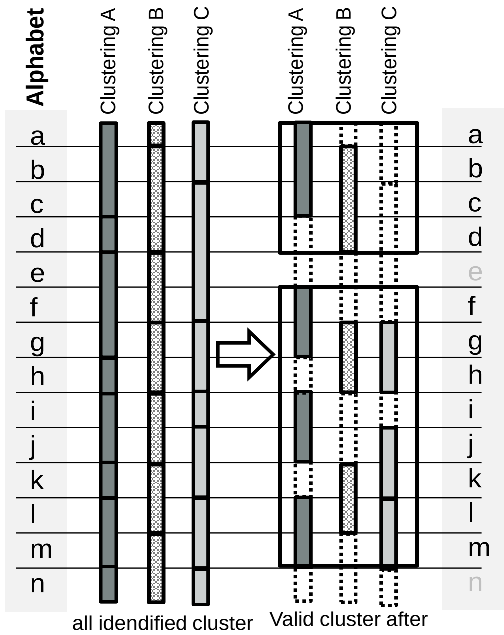

Fig. 5. Combining different clustering results

TABLE III. COVERAGE WITH MERGED CLUSTERS

	Results from combining the clustering results from Frequency, Euclidian, Manhattan and Hamming distance
Union set of dedicated events coverd (abseloute count)	69 % (4,981)
Union set of trace covered (abseloute count)	65 % (1,664,373)

A. Discussion of results

One can see in TABLE III. that the union set of all covered events in relation to the total amount of events increases from 46% at Hamming distance to 69% for the union set of all clustering results. Also the coverage of the trace increases from 49% to 65%. As a result of combination of different clustering approaches the behavior of one single event will be described with more than one automaton. The mapping of events to automata is over-determined. One dedicated event is now mapped to 2.3 automata at average.

VII. CONCLUSION AND FUTURE WORK

In this paper, a method for learning behavior models from network traces is proposed. These behavior models are the foundation for using reference traces to validate network communication in distributed embedded systems.

To enable learning algorithm the problem of high complexity within a network trace must be solved. Based on the assumption that a trace contains several parallel and independent activities, a system hypothesis was formulated. With the help of this hypothesis a technique for clustering and an evaluation method is derived and evaluated.

This technique basically uses the timestamps of the events and generates frequency components of each dedicated event. It was shown that clustering, based on these frequency components, produces sufficient results. The inferred automata from the L* algorithm helps to evaluate the clustering results and provide at the same time the behavior models to compare other network traces with the reference trace.

The next steps in feature work will be an estimation of the false positive and false negative rate, when the learned models from L* are used to compare the reference trace with other network traces. Additionally the concept of merging several clustering approaches promises good results and needs to be improved. For that reasons it would be a good idea to try some completely different clustering technics, that the union set and so and the coverage of events will be increased more efficient.

Based on a real world example it was shown that it is possible to separate a CAN trace in different sub-traces in the manner that these sub-traces contain independent execution graphs. The proposed methodology covers 69% of the events from the example trace. That is in comparison to previous work a significant improvement. With respect to the huge amount of 7,500 dedicated events and a trace length of 2.4 million events of the example trace, this is high amount of successfully interpreted data. With this results a step ahead to establish an unsupervised self-learning approach for validating network communication in scenarios were no specification is available.

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Towards to an Agent-Oriented Meta-Model for Modeling Vehicular Systems with Multi-Configuration Ability

Mohamed Garoui
PRINCE Research Unit, ENSI,
Manouba, Tunisia
garouimohamed2010@gmail.com

Belhassen Mazigh
Department of Computer Sciences
FSM, Monastir, Tunisia
belhassen.mazigh@gmail.com

Béchir El Ayeb
PRINCE Research Unit
FSM Monastir, Tunisia
ayeb_b@yahoo.com

Abderrafiaa Koukam
IRTES-SET, EA 7274, UTBM,
F-90010 Belfort cedex, France
abder.koukam@utbm.fr

***Abstract*— Agent technology is a software paradigm that permits to implement large and complex distributed applications. In order to assist the development of multi-agent systems, agent-oriented methodologies (AOM) have been created in the last years to support modeling more and more complex applications in many different domains. By defining in a non-ambiguous way concepts used in a specific domain, Meta modeling may represent a step towards such interoperability. In the transportation domain, this paper proposes an agent-oriented meta-model that provides rigorous concepts for conducting transportation system problem modeling. The aim is to allow analysts to produce a transportation system model that precisely captures the knowledge of an organization so that an agent-oriented requirements specification of the system-to-be and its operational corporate environment can be derived from it. To this end, we extend and adapt an existing meta-model, Extended Gaia, to build a meta-model and an adequate model for transportation problems. Our new agent-oriented meta-model aims to allow the analyst to model and specify any transportation system as a multi-agent system. In this paper, we aim to propose an Agent-Oriented meta-model adequate to any Transportation Systems with multi-configuration ability problem.**

Keywords-*Agent technology; Transport domain; Meta-model; multi-agent system.*

I. INTRODUCTION

The purpose of the Agent-Oriented Software Engineering is the creation of a path towards integration and interoperability of methodological approaches for Multi-Agent Systems (MAS) development. This involves the definition of a common framework for MAS specification, which includes the identification of a minimum set of concepts and methods that can be agreed in the different approaches. The tool for defining this framework is meta-modeling. The principle of meta-modeling has been already used in other fields of software engineering, for instance, in the specification of Unified Modeling Language (UML) [1] by Object Management Group (OMG) [1], to describe the elements of the language, their constraints and relationships.

In platooning systems Research, such as [2] and [3], each vehicle determines its own position and orientation only from its perceptions of the surrounded environment. In this context, the reactive multi-agent paradigm is well adapted to specify and analyze this system. The interest of those approaches results from their adaptability, simplicity and robustness. In this case, platoon configuration can be considered as the result of the self-organization of a Reactive Multi-Agent System (RMAS). A platoon multi-agent system can then be defined as a set of agents, each one corresponding to a vehicle. Two agent roles can be distinguished: leader and follower agents. The Leader agent interacts only with its environment (road, obstacles, etc.).

Our problem here is that when we model the vehicular system, we need an agents-oriented meta-model that gives us a set of basic concepts. These concepts are necessary to model the entire of transport system problem in different ***environment*** (Urban, Agricultural, and Military) and with various ***navigation policies*** and its ***behavior***. In addition, as soon as we obtain the system model, it will be easy to implement our multi-agent system by using agent oriented programming.

In this paper, our contribution is to provide an agent-oriented meta-model adequate to transportation system problem, which allowed us to model the vehicular system in their navigation environment. Our proposed meta-model has been built by *adopting* and *extending* the existing Extended Gaia meta-model [4] and thus, we define two levels of models inspiring from PASSI meta-model [5]. This seems to us coherent with the most accepted definition of meta-model: a meta-model is a “model of a model”, and it provides an explicit representation of the constructs and relationships needed to build specific models within a domain of interest. This proposition arises by remarking that in the field of transport doesn’t occur any Agent oriented meta-model to clearly specify and analyze any transport system in the form of multi-agent systems.

We choose to use the Extended Gaia meta-model as it is well adapted to organizational structures such as *teams*, *congregations*, and *coalitions* which are used in clustering and collaborative missions of the platoon entities. Furthermore, the proposed approach must take into account, in their meta-model, the concept of environment and different social structures associated with different application areas (Urban, Agricultural, Military), as indicated in Table I. Extended Gaia specifies the notion of the environment by Environment concept. The abstraction of the

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environment specifies the set of entities and resources of a multi-agent system can interact with, limiting interactions using the authorized shares.

TABLE I. SOCIAL STRUCTURE ACCORDING TO THE APPLICATION AREAS

Application Area	Suitable Social Structure
Urban	Congregations, Coalition
Agricole	Congregations, Teams, Coalition
Military	Teams, Congregations, Coalition

The Extended Gaia meta-model adds some organizational based concepts. The organization itself is represented with an entity, which models a specific structure (or topology). The organizational rules are considered responsibilities of the organization. They include safety rules (time-independent global invariants that the organization must respect) and a liveness rules (that define how the dynamics of the organization should evolve over time).

This paper is structured as follow: in Section II, we present a state of the art about the existing agent-oriented meta-model used for modeling and specify multi-agent systems. Section III presents our Proposed Agent-oriented Meta-model for transportation systems. Then, Sections IV illustrates our Agent-oriented Meta-model with an application of urban public transportation systems. Finally, Section V concludes by giving a list of possible future works.

II. STATE OF THE ART

Many agent-oriented meta-model have been proposed for modeling of multi-agent system. The first version of the Gaia methodology, which modeled agents from the object-oriented point of view, was revisited 3 years later by the same authors in order to represent a MAS as an organized society of individuals [6, 7].

Agents play social roles (or responsibilities) and interact with others according to protocols determined by their roles. With that approach, the overall system behavior is understood in terms of both micro- and macro-levels. The former explains how agents act according to their roles, and the latter explains the pattern of behavior of those agents. These constraints are labeled organization rules and organization structures, respectively.

A central element of the meta-model of Gaia is the agent entity, which can play one or more roles. A role is a specific behavior to be played by an agent (or kind of agents), defined in terms of permissions, responsibilities, activities and interactions with other roles. When playing a role, an agent updates its behavior in terms of services that can be activated according to some specific pre- and post-conditions. In addition, a role is decomposed in several protocols when agents need to communicate some data. The environment abstraction specifies all the entities and resources a multi-agent system may interact with, restricting the interactions by means of the permitted actions.

The Extended Gaia meta-model adds some organizational based concepts. The organization itself is represented with an entity, which models a specific structure (or topology). The organizational rules are considered responsibilities of the organization. They include safety rules (time-independent global invariants that the organization must respect) and liveness rules (that define how the dynamics of the organization should evolve over time). Given the aggregation association defined from agents with respect to organizations rules and from organizations with respect to organization structures, Gaia permits to design a hierarchical non-overlapping structure of agents with a limited depth. From the organizational point of view, agents form teams as they belong to a unique organization, they can explicitly communicate with other agents within the same organization by means of collaborations, and organizations can communicate between them by means of interactions. If inter-organization communication is omitted, coalitions and congregations may also be modeled.

Process for Agent Societies Specification and Implementation (PASSI) [8] is an iterative-incremental process for designing multi-agent systems starting from functional requirements that adopts largely diffused standards like UML (as the modeling language, although extended to fit the needs of agents design) and FIPA (as the agent platform). PASSI covers all the phases from requirements analysis to coding and testing with a specific attention for the automation of as many activities as possible with the support of PASSI Toolkit (PTK), a specifically conceived design tool.

The PASSI MAS meta-model [8] is organized in three different domains: the Problem Domain (where requirements are captured), the Agency Domain that represents the transition from problem-related concepts to the corresponding agent solution (that is a logical abstraction), and the Solution Domain (where the implemented system will be deployed).

The Problem Domain deals with the user’s problem in terms of scenarios, requirements, ontology, and resources; scenarios describe a sequence of interactions among actors and the system. Requirements are represented with conventional use case diagrams. The system operating environment is depicted in terms of concepts (categories of the domain), actions (performed in the domain and effecting the status of concepts) and predicates (asserting something about a portion of the domain elements), the environment also includes resources that can be accessed by agents.

The Agency Domain includes the agent that is the real centre of this part of the model; each PASSI agent is responsible for accomplishing some functionalities descending from the requirements of the Problem Domain. Each agent during its life can play some roles; these are portions of the agent social behavior characterized by some specificity such as a goal, or providing a functionality/service and in so doing it can also access some resources. The Service component represents the service provided by a role in terms of a set of functionalities (including pre- and post-conditions as well as many other details mostly coming from the Ontology Web Language (OWL) specifications), and can be required by other agents to reach their goals. Agents could use portions of behavior (called tasks) or communications to actuate the roles aims.

Agent-oriented Software Process for Engineering Complex Systems (ASPECS) [9] provides a holonic perspective to design MAS. Considering that complex systems typically exhibit a hierarchical configuration, on the contrary to other methodologies, it uses holons instead of atomic entities. Holons, which are agents recursively composed by other agents, permit to design systems with different granularities until the requested tasks are manageable by individual entities.

Being one of the most recent methodologies, it takes the experience gained from previous approaches, such as PASSI and RIO [10], as the base to define the meta-model and the methodology.

The goal of the proposed meta-model is to gather the advantages of organizational approaches as well as those of the holonic vision in the modeling of complex systems. A three layer meta-model, with each level referring to a different aspect of the agent model, is proposed: The *Problem domain* covers the organizational description of the problem. An organization is composed by roles which interact within scenarios while executing role plans. Roles achieve organizational goals by means of their capacities (i.e., what a behavior is able to do). The organizational context is defined by means of ontology. This meta-model layer is used mainly during the analysis and design phases. The *Agency domain* defines agent-related concepts and details the holonic structure as a result of the refinement of the elements defined in the Problem domain. Each holon is an autonomous entity with collective goals and may be composed by other holons. Holonic groups define how members of the holon are organized and how they interact in order to achieve collective goals. At the finest granularity level, holons are composed by groups and their roles are played by agents, which achieve individual goals. A rich communication between agent roles (which are instances of organizational roles) is also supported, specifying communicative acts, knowledge exchange formalized by means of the organizational ontology, and protocols specifying sequences of messages.

III. OUR PROPOSED META-MODEL: PLATOONING META-MODEL

The UML is based on the four-level meta-modeling architecture. Each successive level is labeled from M_3 to M_0 and are usually named meta-meta-model, meta-model, class diagram, and object diagram respectively. A diagram at the M_i-level is an instance of a diagram at the M_{i+1}-level. Therefore, an object diagram (an M_0-level diagram) is an instance of some class diagram (an M_1-level diagram), and this class diagram is an instance of a meta-model (an M_2-level diagram). The M_3-level diagram is used to define the structure of a meta-model, and the Meta Object Facility (MOF) belongs to this level. The UML meta-model belongs to the M_2-level.

After studying the Extended Gaia meta-model, we observe how this explicit and useful models of the social aspect of agents. Although, it was not designed for open systems, and it provides little support for scalability and simplicity to allows improvements to facilitate with its relative concepts. It models both the macro and micro aspects of the multi-agent system. Gaia believes that a system can be regarded as a company or an organization of agents.

In this section, we try to solve our contributions mentioned from the start. It manifests itself to extend and adapt an existing meta-model to build a meta-model and an adequate model for transport problems.

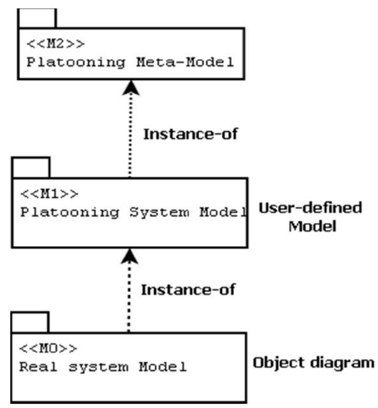

Figure 1. Model instantiation checking

In Fig. 2, the classes presented in black color are the base classes of Extended Gaia meta-model. For against, the blue classes are the classes added to the existing meta-model to be adapted to platooning applications and then help us to implement our own methodology for modeling and dependability analysis. Table II presents the definition of the added new concepts.

TABLE II. DEFINITION OF THE ADDED NEW CONCEPTS

Concept	Definition
Functional Requirement	A function that the software has to exhibit or the behavior of the system in terms of interactions perceived by the user
Non-Functional Requirement	A constraint on the solution. Non-functional requirements are sometimes known as constraints or quality requirements
AgentModel	Abstract description of a formal model which gives an abstract view about ***the agent behavior***.
OrganizationModel	Abstract description of a formal model which gives an abstract view about ***the organization behavior***.

The concept *Functional Requirement* is a function that the software has to exhibit or the behavior of the system in terms of interactions perceived by the user. This concept allowed us to identify our system requirements. The *Non-Functional Requirement* concept provides a constraint on the solution. Non-functional requirements are sometimes known as constraints or quality requirements. *AgentModel* concept gives an abstract view about *the Agent behavior*. *OrganizationModel* gives an abstract view about ***the*** *organization behavior*. Behavior is described by formal state-based models [11].

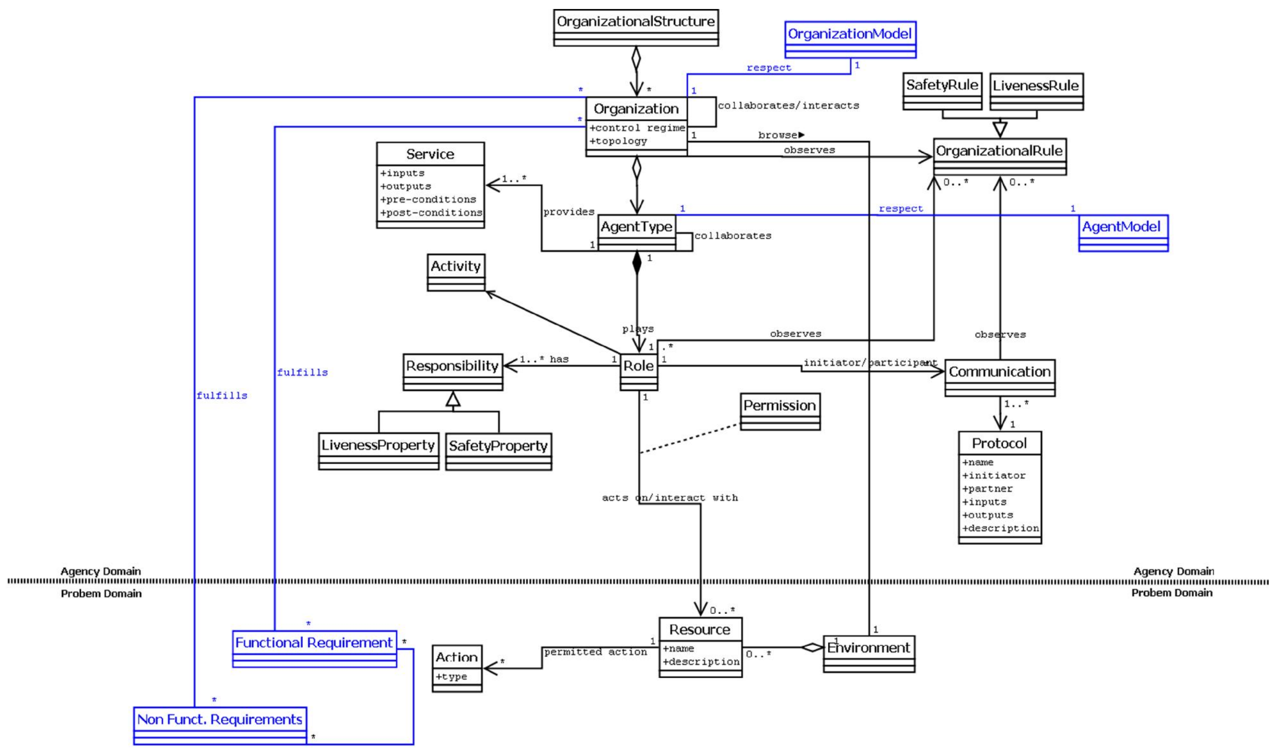

Figure 2. Our proposed Meta-model: Platooning Meta-model.

By inspiring from PASSI [8] and ASPECS meta-models [9], we tried to organize our meta-model in two areas: Problem Domain and Agent Domain. Problem Domain involves elements (Fig. 2) are used to capture the requirements problem and perform initial analysis. Agent Domain includes elements (Fig. 2) are used to define an agent-oriented solution to the problem described in the previous step.

After this, we pass to M_1-level (Fig. 1) describes the *Platooning System Model* (see Fig. 3) which constitutes of instance of the concepts of M_2-level model. This model includes all the basic concepts and necessary for us to model any type of application to platooning with their bodies, interaction, environment, their geometric configuration and formal models associated with each component platoon. The table below provides the concepts related to platooning System Model and their relationship with the concepts of the M_2-level model.

Table III gives an idea about the basic concepts of *Platooning System Model* (Fig. 3) which is instances of our meta-model that shown in the Fig. 2. The *Platoon* concept represents the main element in our model which is an instance of meta-concept *Organization*. Any Platoon is modeled as a set the *Entity*. There are two kinds of entities: Leader and Follower, which are modeled by the two concepts *Leader* and *Follower*. The tow concepts *Entity_Model* and *Platoon_Model* are used to describe the behavior of entity and platoon in the environment. The concept *Area* model the environment notion. In our transportation problem, there are three types: Urban, Agricultural, and Military. The concepts *Parameters*, *Entity_Parameters* and *System_Parameters* provided a general idea about the parameters of the entities and of the system. These parameters are necessary and useful for Dependability Evaluation in our future work.

TABLE III. THE RELATED CONCEPTS TO PLATOONING SYSTEM MODEL

Concept	Instance of
Platoon	Organization
Structure	OrganizationalStructure
Geo_Configuration	OrganizationalRule
Navigation_Policy	OrganizationalRule
Interaction	Communication
Entity	AgentType
Leader	AgentType
Follower	AgentType
Parameters	OrganizationalRule
Entity_Parameters	--
System_Parameters	--
Model	--
Entity_Model	AgentModel
Platoon_Model	OrganizationModel
Area	Environment

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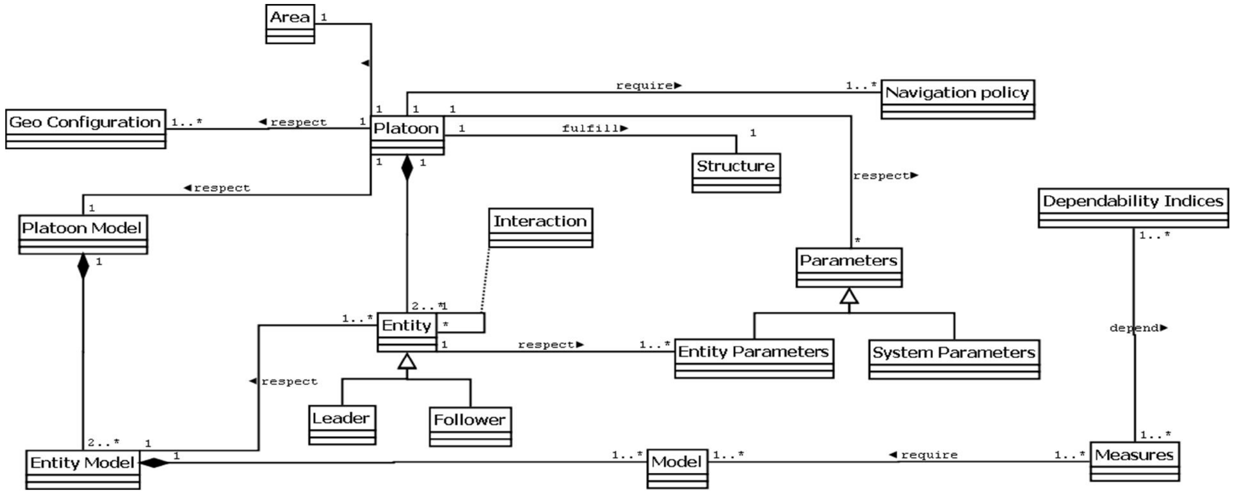

Figure 3. Platooning System model

IV. A CASE STUDY: URBAN PUBLIC TRANSPORTATION SYSTEM

According to the agent-oriented meta-model, we try to specify a transport applications in urban environment. The convoy adopts a *line* configuration (see Fig. 4a) Longitudinal gap (Inter-distance) between vehicle 0 meter and 2 meters in lateral gap. For these scenarios, the convoy will have a fixed number of vehicles between two and three and will move on a track with a radius of curvature ranging from 15 m to infinity. The train moves at a maximum speed of 50 km/h with an acceleration of 1 m/s^2 and a deceleration of -3 m/s^2 on a maximum distance of 1000 meters. From these settings, two scenarios are proposed. The first is to evolve a convoy of vehicles with fixed-line configuration. During the movement, the convoy can change its geometric configuration from Line configuration to Echelon configuration (see Fig. 4).

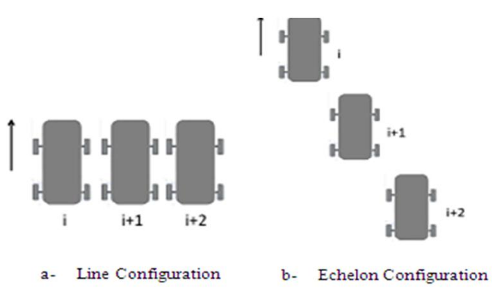

Figure 4. Urban Public Transportation Configuration

In Fig. 5, we present the object diagram which is an instantiation of the Platooning System Model (Fig. 3). The object diagram in the data modeling language UML used to represent instances of classes, that is to say objects. As the class diagram, it expresses the relationship between objects, but also the state of objects, thereby expressing execution contexts. In this sense, this pattern is less general than the class diagram. Object diagrams are used to show the state of object instances before and after the interaction, i.e., it is a photograph at a specific time and attributes existing object. It is used in the exploratory phase.

The object diagram of our study is a set of objects that have the attributes that characterize the system. The object *Convoy_Urban* is an instance of the concept *Platoon* which is it's an instance of *Organization* concept of the metamodel. The convoy adapt a line configuration and Line to Level navigation policy therefore we find an instance of *Geo_Configuration* named *Line_Configuration* and an instance of *Navigation policy* named *Line_to_Level*.

Our transportation system is constitutes of three intelligent vehicles: one Leader and two follower, thus the object diagram contains two items: *V_Leader* instance of the concept *Leader* with cardinality equal to 1 and *V_Follower* instance of *Follower* concept with cardinality equal to 2.

Our system has parameters regrouped in the two tables IV and V. These parameters are divided into two kinds: *Vehicle_Parameters* and *Convoy_Urban_Parameters*, which represent convoy entities parameters and the parameters of the overall system, respectively. They are used for dependability evaluation in our future works. Transportation system behavior is modeled by *Convoy_Urban_Formal_Model* object.

TABLE IV. VEHICLES PARAMETERS

Parameters	Values
Max speed	50 km/h
Acceleration/ deceleration	1 m/s^2/-3 m/s^2
Weight	500 kg

TABLE V. CONVOY PARAMETERS

Parameters	Values
Vehicle Number	3
Configuration	"Line"
lateral gap	2 meters
longitudinal gap	0

The behavior is described by state-based models which are used in system dependability evaluation. This model and her parameters are used in our future work to the dependability evaluation.

V. CONCLUSION AND FUTURE WORKS

In this paper, we have proposed an Agent-Oriented meta-model adequate to any Transportation Systems with multi-configuration ability problem. The aim is to allow analysts to produce a transportation system model that precisely captures the knowledge and the behavior of an organization so that an agent-oriented requirements specification of the system-to-be. The most in our Agent-oriented meta-model against to others allows analysts to specify the system structure and behaviors because there is no meta-models specified twos. We illustrated our meta-model on urban public transportation system. We have tried to model our system as multi-agent system based on our proposed meta-model.

Future works will be devoted to several key points aimed at improving the proposed solution. On the one hand, we will work to provide a generic model for a methodology and will be suitable for all platonning application with different scenario in different transportation field. This model is used for the Dependability evaluation.

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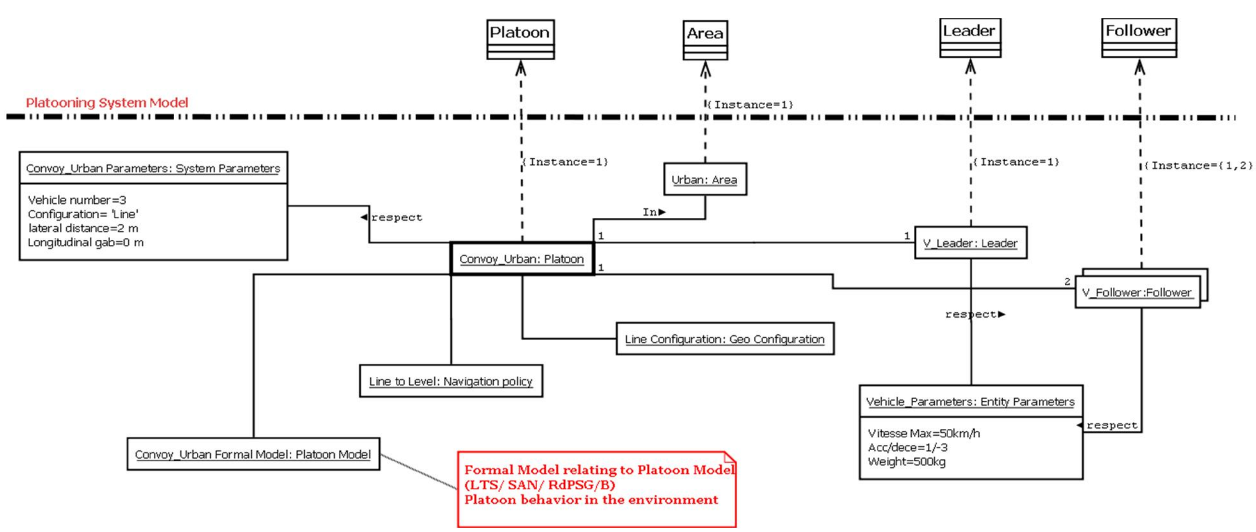

Figure 5. Object diagram relating to urban public transportation systems

A Semantic Platform Infrastructure for Requirements Traceability and System Assessment

Parastoo Delgoshaei
Department of Civil and Environmental Engineering
University of Maryland
College Park, MD 20742, USA
Email: parastoo@umd.edu

Mark A. Austin
Department of Civil and Environmental Engineering
University of Maryland
College Park, MD 20742, USA
Email: austin@isr.umd.edu

Daniel A. Veronica
Energy and Environment Division
National Institute of Standards and Technology (NIST)
Gaithersburg, MD 20899, USA
Email: daniel.veronica@nist.gov

***Abstract*—This work-in-progress paper describes a new approach to requirements traceability and system assessment through the use of semantic platforms. The platform infrastructure corresponds to an integration of traceability mechanisms with reusable domain-specific ontologies, and associated sets of mathematical/logical rules for design rule checking. Engineering system components are modeled as instances of the domain ontologies with values for their properties filled in. We expect that when the proposed infrastructure is fully developed it will enhance systems engineering practice in several ways. First, by filling the gap between system requirements and system models, semantic platforms will lower validation costs by allowing for rule checking early in design. Second, semantic platforms will support performance assessment during the system operation. Our medium-term research and development objective is semantic platform infrastructures capable of supporting the design, simulation, verification, and management of engineering systems having mixtures of discrete and continuous behavior.**

Keywords-Systems Engineering; Semantic Modeling; Design Platform; Requirements; Rule Checking.

I. Introduction

Under financial sponsorship of the National Institute for Standards and Technology in Gaithersburg, Maryland, the authors are working on the design and implementation of procedures and software for the model-based systems engineering, integration, and performance-assessment of cyberphysical systems (CPS). The distinguishing feature of CPS is a coupling of physical and cyber systems, with the cyber affecting the physical and vice versa. Present-day design procedures are inadequate for the design of modern CPS systems. Among the many challenges that CPS presents, design space exploration and trade studies are difficult to conduct because decision variables span parametric, logical, and dependency relationship types. Components are often required to serve multiple functions – as such, cause-and-effect mechanisms are no longer localized and obvious. System relationships can reach laterally across systems hierarchies and/or intertwined network structures.

II. Project Objectives

With the objective of addressing the the long-term systems engineering challenges that CPSs present, this paper describes a new approach to requirements traceability and system assessment through the use of semantic platforms. As we will soon see in Sections IV and V, the proposed platform infrastructure corresponds to an integration of traceability mechanisms with reusable domain-specific ontologies, and associated sets of mathematical and logical rules for design rule checking. Engineering system components are modeled as instances of the domain ontologies with values for their properties filled in.

A key element of our long-term research objective is development of knowledge – methodologies, algorithms, software – required to design and build scalable platform infrastructures, with the latter supporting the simulation and design, verification, and management of multidisciplinary (e.g., mechanical, electrical, software) sensor-enabled engineering systems having mixtures of discrete and continuous component-level behavior. A series of progressively capable software prototypes will be built. Each iteration of development [3], [4] will employ a combination of software design patterns (e.g., networks of model, view, controllers), software libraries and languages for semantic applications development (e.g., OWL, Jena) [7], [13] and executable statecharts, finite element procedures for the computation of behaviors over continuous physical domains (e.g., fluid flow in a pipe network), and customized visualization for the requirements, ontology and engineering models.

III. Research Methodology

Model-based systems engineering development is an approach to systems-level development in which the focus and primary artifacts of development are models, as opposed to documents. Our research methodology is driven by a need to achieve high levels of productivity in system development. We believe this can be achieved through the use of high-level visual abstractions coupled with lower-level (mathematical) abstractions suitable for formal systems analysis. The high-level abstractions provide a "big picture" summary of the system under development and highlight the major components, their connectivity, and performance. The lower-level abstractions are suitable for formal systems analysis – for example, verification of component interface compatibilities and/or assessment of system performance through the use of simulation methods.

As engineering systems become increasingly complex the

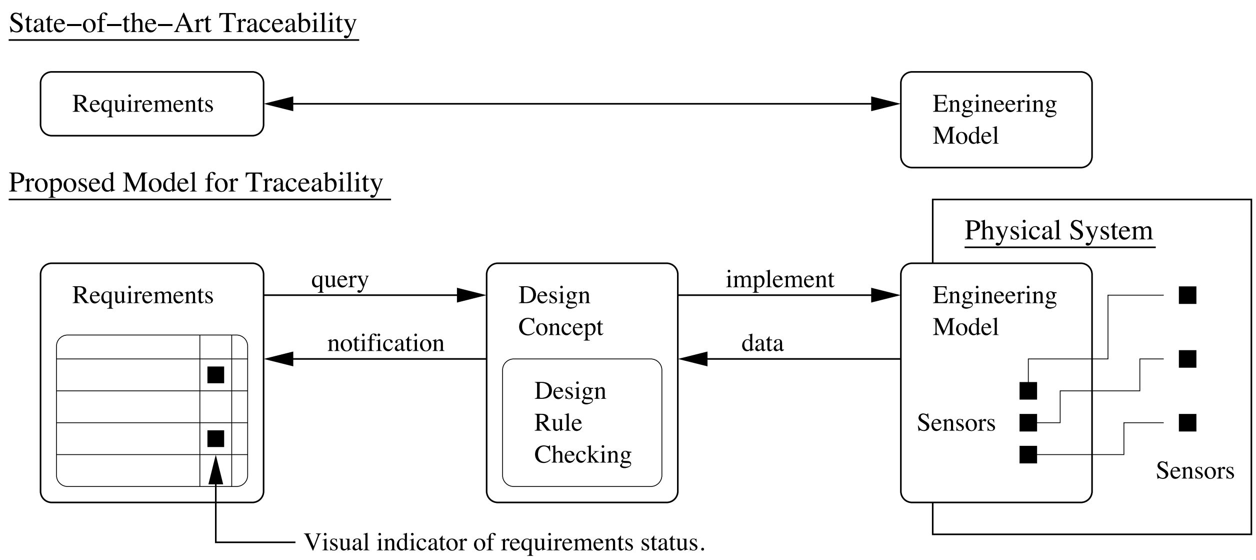

Figure 1. Schematics for: (top) state-of-the-art traceability, and (bottom) proposed model for ontology-enabled traceability for systems design and management.

need for automation arises. A key element of required capability is an ability to identify and manage requirements during the early phases of the system design process, where errors are cheapest and easiest to correct. We believe that methodologies for strategic approaches to design will employ semantic descriptions of application domains, and use ontologies and rule-based reasoning to enable validation of requirements, automated synthesis of potentially good design solutions, and communication (or mappings) among multiple disciplines [1], [2], [11]. Semantic Web-based technologies can play a central role in the design of tools to support these design methodologies. Present-day systems engineering methodologies and tools, including those associated with SysML [6] are not designed to handle projects in this way.

IV. Semantic Platform Infrastructure

The systems architecture for state-of-the-art requirements traceability and the proposed platform model is shown in the upper and lower sections of Figure 1. In state-of-the-art traceability mechanisms design requirements are connected directly to design solutions (i.e., objects in the engineering model). Our contention is that an alternative and potentially better approach is to satisfy a requirement by asking the basic question: What design concept (or group of design concepts) should I apply to satisfy a requirement? Design solutions are the instantiation/implementation of these concepts.

The proposed architecture is a platform because it contains collections of domain-specific ontologies and design rules that will be reusable across applications. In the lower half of Figure 1, the textual requirements, ontology, and engineering models provide distinct views of a design: (1) Requirements are a statement of "what is required." (2) Engineering models are a statement of "how the required functionality and performance might be achieved," and (3) Ontologies are a statement of "concepts justifying a tentative design solution." During design, mathematical and logical rules are derived from textual requirements which, in turn, are connected to elements in an engineering model. Evaluation of requirements can be include checks for satisfaction of system functionality and performance, as well as identification of conflicts in requirements themselves.

A key benefit of our approach is that design rule checking can be applied at the earliest stage possible – as long as sufficient data is available for the evaluation of rules, rule checking can commence; the textual requirements and engineering models need not be complete. During the system operation, key questions to be answered are: What other concepts are involved when a change occurs in sensing model? What requirement(s) might be violated when those concepts are involved in the change? To understand the inevitable conflicts and opportunities to conduct trade space studies, it is important to be able to trace back and understand cause-and-effect relationships between changes at system-component level and their affect on stakeholder requirements.

V. Work in Progress

Our testbed application area and driver for this work is performance-based modeling and design of energy-efficient building environments.

To this end, we are currently working toward the platform infrastructure implied by Figures 2 and 3. Figure 2 pulls together the different pieces of the proposed architecture shown in Figure 1. On the left-hand side the textual requirements are defined in terms of mathematical and logical rule expressions. From a CPS viewpoint, modern buildings contain networks for the arrangement of spaces throughout the building, for the fixed circulatory systems (e.g., power and hvac), for the dynamic circulatory systems (e.g., air and water flows), and for wired and wireless communications. Predictions of dynamic behavior will correspond to the solution of nonlinear differential algebraic equations (e.g., for water, air, and thermal flow) coupled to discrete equations (e.g., resulting from cyber decisions). While there is a desire for each network to operate as independently as possible, in practice the need for new forms of functionality will drive components from different network types to connect in a variety of ways. Within the

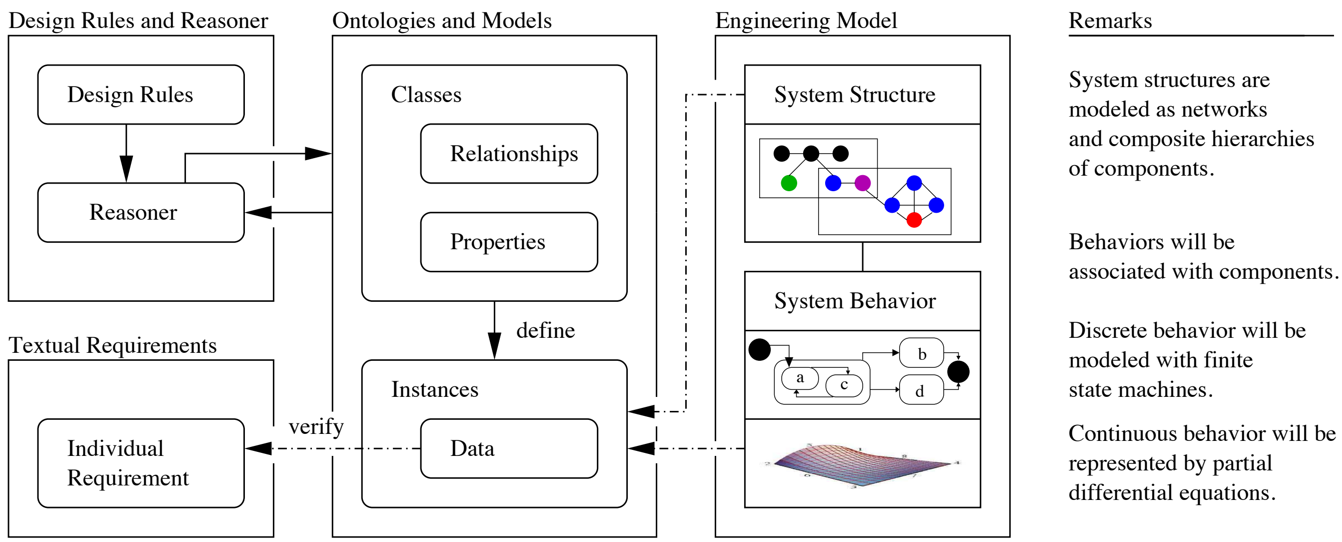

Figure 2. Framework for implementation of ontology-enabled traceability and design assessment.

building simulation community state-of-the-art dynamic simulation is defined by Modelica, and steady-state simulation by DOE-2 and eQuest. Building systems are modeled as networks of connected components, mathematical equations describe component-level behaviors, and models of system-level behavior are defined by graphs of equations synthesized from descriptions of component behavior and their connectivity.

Topic 1. Modeling and Reasoning with Ontologies. The upper section of Figure 1 shows state-of-the-art traceability mechanisms with requirements being connected directly to solutions. The proposed platform provides a base to implement requirements as rules and related parts of the engineering model as classes of the ontology. The lower section of Figure 1 shows the essential details of our three-part framework – textual requirements, ontology model and engineering model – for the implementation of ontology-enabled traceability mechanisms and rule-based design assessment.

Textual requirements are connected to the ontology model, and logical and mathematical design rules, and from there to the engineering model. Ontology models encompass the design concepts (ontology classes) in a domain, as well the relationships among them. Classes are qualified with properties (c.f., attributes in classes) to represent the consequence of constraint and design rule evaluations. Examples of valid relationships are: containment, composition, uses, and ”is Kind of”. These classes are placeholders for the data extracted from the engineering model. Individuals are the object counterpart of classes, with data and object property relationships leading the to RDF graph infrastructure. Each instance of an individual holds a specific set of values obtained the engineering model.

Rules serve the purpose of constraining the system operation and/or system design. They provide the mechanisms for early design validation, and ensuring the intended behavior is achieved at all the times during system operation. We are currently working with reasoners provided in the Jena API. A reasoner works with the RDF graph infrastructure and sets of user-defined rules to evaluate and further refine the RDF graph. Rule engines are triggered in response to any changes to the ontological model. This process assures that the model is consistent with respect to the existing rules. Traceability from ontologies to requirements is captured via implementation of the listeners that are notified as a result of change in the semantic model.

In a departure from past work, we are exploring the feasibility of creating built-in functions to capture and evaluate performance criteria, i.e., energy efficiency ratio of the HVAC system. A second potential use of built-in functions is as an interface to packages that provide system improvements through optimization and performance related queries. We note that a rule-based approach to problem solving is particularly beneficial when the application logic is dynamic (i.e., where a change in a policy needs to be immediately reflected throughout the application) and rules are imposed on the system by external entities [10], [12]. Both of these conditions apply to the design and management of engineering systems.

Topic 2. Modeling Engineering Systems with Components and Finite Elements. Modern buildings contain a variety of intertwined networks for the arrangement of architectural spaces and circulatory systems. They can also be thought of as a hierarchical arrangement of spaces – for example, buildings have floors, floor contain rooms, rooms contain furniture, and so forth. Unfortunately, the time-history analysis and control of building system performance is complicated by the need to model combinations of discrete and continuous behaviors. To address these concerns, we are exploring the feasibility of modeling system structures as networks and composite hierarchies of components, and implementing software prototypes through use of the composite design pattern. Behaviors will be associated with components. Components will be organized into component hierarchies – for example, in order to sense the water level in a tank, sensor components will be positioned inside water tank components. Discrete behaviors (e.g., the operation of hvac machinery) will be modeled with executable statecharts. Continuous behaviors will be represented by partial differential equations.

Our current strategy is to compute solutions to continuous

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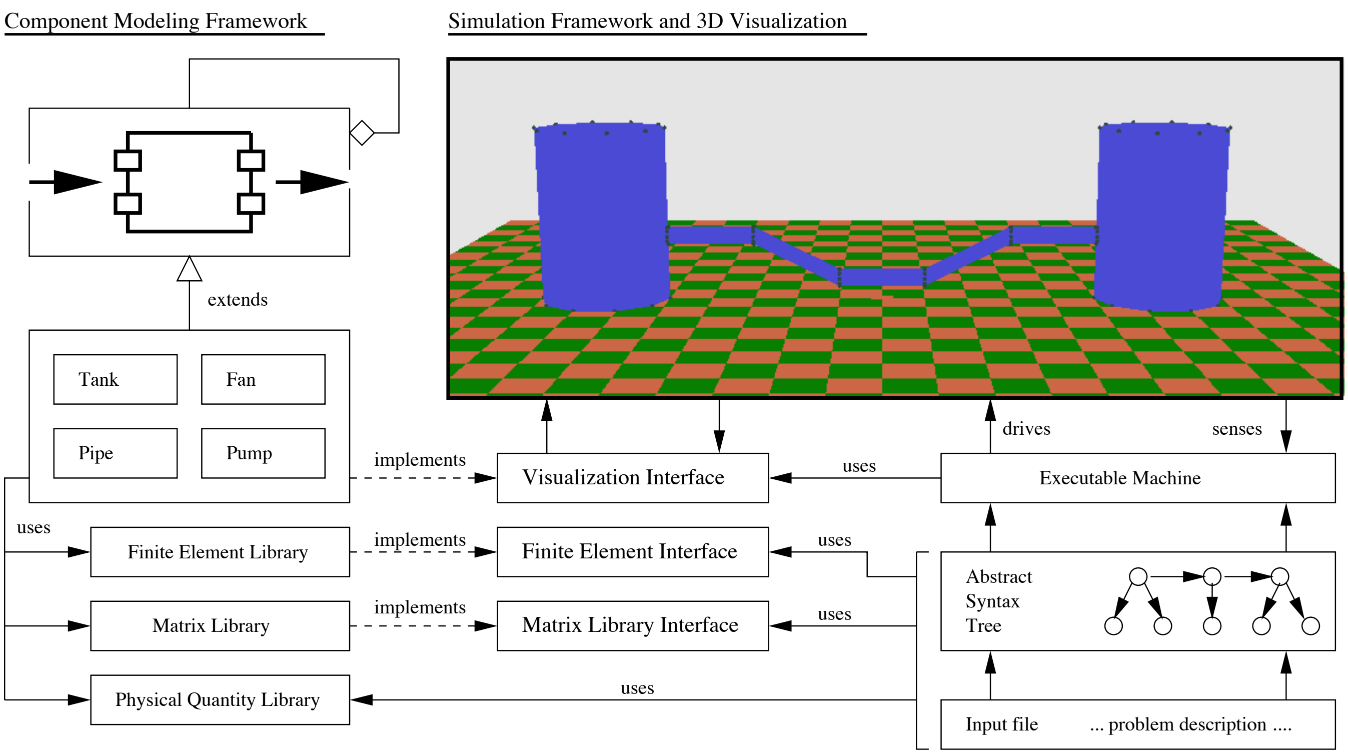

Figure 3. Architecture for modeling hvac systems as networks of connected components, and using finite element solution procedures for computing and visualizing time-history behavior.

behavior with the finite element method, although this certainly isn't the only way it could be done. In a departure from established approaches to engineering systems modeling, our goal is to design a single software architecture that can support this range of organization and behavioral abstractions. The key to making this work is software interfaces designed to support a multitude of system viewpoints – a visualize interface for 2D- and 3D- visualization, and a finite element interface for the description of element-level behaviors cast in a matrix format, a communications interface for sensor to controller communication. Since a Java object can implement and arbitrary number of interfaces, there is reason to believe this might actually work!

Topic 3. Scripting Language Support for Systems Integration. To support the organization and integration of physical components and computation of discrete and continuous behaviors, we are in the process of designing and implementing a scripting language where physical units are embedded within the basic data types, physical quantities and matrices of physical quantities, and scripting language constructs for controlling branching and looping control. For example, the fragment of code:

```
Force    = [ 2 N, 3 N, 4 N ];
Distance = [ 1 m; 2 m; 3 m ];
Work     = Force*Distance;
```

is a simple calculation for the work done by a force moving through a prescribed distance. The output is as follows:

```
Matrix: Force
row/col                 1             2             3
        units           N             N             N
   1          2.00000e+00   3.00000e+00   4.00000e+00

Matrix: Distance
row/col                 1
        units           m
   1          1.00000e+00
   2          2.00000e+00
   3          3.00000e+00

Matrix: Work
row/col                 1
        units         Jou
   1          2.00000e+01
```

The language supports the representation of differential equations in their discrete form, and solution via numerical integration techniques. For example, if the transient flow of fluid between the two tanks shown in Figure 3 is defined by:

$$\left[\frac{dU(t)}{dt}\right] + \left[\frac{f_1}{2D}\right] U(t)|U(t)| = \left[\frac{g}{L}\right] [H_1(t) - H_2(t)], \quad (1)$$

then the script:

```
velFluid  = pRoughness/(4.0*pRadius)*velOld*Abs(velOld)*dt;
velUpdate = g/pLength*( h01Old - h02Old )*dt;
velNew    = velOld + velUpdate - velFluid;
```

shows the essential details of computing the fluid velocity update with Euler integration. During the executable phases of simulation (right-hand side of Figure 3), the runtime interpreter checks for dimensional consistency of terms in statements before proceeding with their evaluation.

VI. Related Work

Our program of research balances the need to build upon the work of others when it makes sense, while also moving boundaries to create new functionality and knowledge. Energy-efficient buildings are highly complex multidisciplinary entities. Part of the challenge we face stems from the difficulty in using a multitude of ontologies to adequately describe system structure, system behavior, and dependency relationships among domains. For example, an HVAC ontology contains concepts like fan, controller, damper, pipe and the relationships between them. Spatial ontologies contain concepts of room, zone and spatial information about the building. While each ontology will have its own distinct rule sets, the domains are loosely connected via common concepts and classes. This leads to the need for rules that are part cross-cutting and part domain-specific (e.g., If sensor measurement drops below 60 in room 1, then open the valve up to 80%).

An important facet of our work is use of Semantic Web technologies as both system models and mechanisms to derive system behavior. While the vast majority of Semantic Web literature has used ontologies to define system structure alone, this is slowly changing. Derler and co-workers explain, for example, how ontologies along with hybrid system modeling and simulation, concurrent models of computation can help us better address the challenges of modeling cyber-physical systems (CPSs). These challenges emerge from the inherited heterogeneity, concurrency, and sensitivity to timing of such systems. Domainspecific ontologies are used to strengthen modularity, and to combine model of system functionality with system architecture [5]. The Building Service Performance project proposes use of ontologies and rules sets to enhance modularity, and perform cross-domain information exchange and representation [9]. Koelle and Strijland are investigating the design and implementation of a software tool to support semantic-driven architecture with application of rules for security assurance of large systems in air navigation [8].

VII. Conclusion

The proposed semantic platform infrastructure will enhance systems engineering practice by lowering validations costs (through rule checking early in design), and providing support for performance assessment during the system operation. We envision cyber-physical systems having behaviors that are both distributed and concurrent, and defined by mixtures of local- and global- rule-based control. Further research is need to understand how the RDF graph structure and supporting software infrastructure can be extended in ways that make it more closely mimic object-oriented practices, specifically through increased use of generalization and access modifiers.

For the time-history behavior modeling and control of energy-efficient buildings, the finite element method is attractive because problem solutions can be formulated from first principles of engineering such as momentum balance. Solution procedures will be robust, scalable, and extensible to energy-balance calculations. Our plans are to design a family of component model interfaces (left-hand side of Figure 3), extend them for the implementation of a build components library (e.g., tanks, pipes, etc) and where needed, participate in finite element analysis, actuation, and control. The scripting language will act as the glue for systems integration and specification of simulation and visualization procedures.

Acknowledgment

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Improving Change Management Systems and Processes

Marko Jäntti, Juha Niskala
School of Computing
University of Eastern Finland
P.O.B 1627, 70211 Kuopio, Finland
Email: marko.jantti@uef.fi

***Abstract*—IT service change management aims to ensure that new services and changes to services will be deliverable and manageable at the agreed cost and service quality. Change management is an attractive research topic due to its complicated nature (broad scope, a large number of interfaces, many instances). Our research exploits the best practices of IT Infrastructure Library (ITIL) framework. The research problem of this study is: Which factors should be taken into consideration while implementing IT service change management systems and processes? The main contribution of this paper is to present results of a case study with a Finnish IT service provider organization that provides ICT and medical technology services in Finland. Our results consist of identified functional and data requirements for a change management module within an IT service management system as well as practical implications for change management process implementation. The case study was carried out as a part of a research project that focused on improving Service Transition processes.**

Keywords*—*IT service; change management; change; process

I. Introduction

IT service change management aims to ensure that new services and changes to services will be deliverable and manageable at the agreed cost and service quality. According to the IT Infrastructure Library Glossary [1], an IT service change can be understood as" *addition, modification or removal of anything that could have an effect on IT Services and the scope should include all IT services, configuration items, processes, documentation etc.*"

There are four key challenges related to IT service change management. First, change management occurs in many organizational levels. Top management has to deal with totally different types of changes (typically strategic changes incl. changes to a service portfolio and supplier changes) compared to the changes carried out by operative management (changes to service components, technologies and tools). Second, even in operative change management, there are several types of changes that need to be processed through different change models. Third, IT service organizations often seem to lack the comprehensive understanding what is a change, how does it differ from daily operative activities, service requests and error correction work. Fourth, the interface between release management and change management often remains unclear. Release management activities are performed in order to implement the change. After the the release management process has delivered a release that can contain one or several changes, change management is responsible for organizing the Post Implementation Review meeting. The challenges may exist in relating changes to releases.

In our study, the goal is to improve change management based on the process improvement frameworks that support IT service change management. The most widely used best practice framework for IT service management is IT Infrastructure Library (ITIL) that describes the service lifecycle with five stages: Service Strategy [2], Service Design [3], Service Transition [4], Service Operation [5] and Continual Service Improvement [6]. In addition to ITIL, there is a well known IT governance framework COBIT that defines control objectives for the change management process [7]:

- Develop and implement a process to consistently record, assess and prioritise change requests.
- Assess impact and prioritise changes based on business needs.
- Assure that any emergency and critical change follows the approved process.
- Authorise changes.
- Manage and disseminate relevant information regarding changes.

There are several international standards that IT service provider organizations may use to benchmark their processes. ISO/IEC 20000-1:2010 Part 1: Service management system requirements [8] and ISO/IEC 20000-2:2011 Part 2: Guidance on the application of service management systems [9] are descriptive standards that define requirements for IT Service Change Management. In the ISO/IEC 20000, change management is visible in several areas, such as in:

- Planning and implementing new or changed services (*All service changes should be reflected in Change Management records.*)
- Service reviews (*The service provider and customer(s) should hold service reviews, at least annually and before and after major changes.*)
- Change management (*The change management processes and procedures should ensure that changes have a clearly defined and documented scope; only changes that provide business benefit are approved, changes are scheduled based on priority and risk; changes to configurations can be verified during*

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change implementation; and the time to implement changes is monitored and improved where required.)

The recently published ISO/IEC TS 15504-8:2012 process assessment model [10] provides a systematic framework to measure change management capability.

Much has been written about change management from traditional software engineering and IT management perspective. The study of Sauvé et al. [11] provides a formal method to automatically assign priorities to changes by exploiting a Business-Driven IT Management (BDIM) approach and state that risks associated with changes can be calculated and changes prioritized in an automatic fashion. The most interesting part of their study is the description of the service level agreement (SLA) clauses related to changes (in their case, any security-related RFC costs 1000 USD/hour until the change is implemented). We are also interested in our research to explore more deeply the change-related SLAs.

Hagen and Kemper [12] have proposed an algorithm for the automated detection of conflicting IT change plans. The detection was done based on an object-oriented Configuration Management Database. Machado et al. [13] have investigated how change activities can be grouped together forming atomic groups of activities by using Business Process Execution Language (BPEL). Release management is an important subprocess of change management. Release management has been examined by several studies. Jäntti and Sihvonen [14] have examined the patch and release management activities in Finnish IT service provider organizations. Van Der Hoek and Wolf [15] have explored requirements for release management. Jansen and Bringkemper [16] have presented common misconceptions about product software release management. Furthermore, Jokela and Jäntti [17] have identified challenges in release management process by using product portfolio management perspective in energy domain.

The research gap can be identified in change management from IT service management perspective. Significantly more academic studies are needed to investigate how IT service provider organizations carry out change management activities, measure the performance of change management, and assign roles and define responsibilities to perform the process.

This study was conducted during Keys to IT Service Management and Effective Transition of Services (KISMET) research project where one of the industrial partners was interested in change management process improvement. The **research problem** of this paper is: Which factors should be taken into consideration while implementing IT service change management systems and processes? The main contribution of this study is to

- show functional and data requirements for change management systems,
- present observations from a change management process improvement pilot, and
- provide lessons learnt from change management process improvement.

The results of this study can be used to improve change management systems and processes of IT service provider organizations. The remainder of the paper is organized as follows. In Section 2, the research methods of this study are described. In Section 3, we present the results of the study. Section 4 aims at analysing the findings of the study. The conclusions are given in Section 5.

II. Research Problem & Methodology

The research problem of this study is: Which factors should be taken into consideration while implementing IT service change management systems and processes? The research problem was divided into the following research questions:

- Which data fields should be included in the Request For Change (RFC)/change record?
- What are the benefits of change management process?
- How change management activities are carried out in the case organization?
- Which roles are related to change management?
- Which interfaces does the change management have with other ITSM processes?

A. *The Case Organization and Data Collection Methods*

Our case organization ITSM Ltd (we use a fictive name to maintain the anonymity of the case) is an IT service provider company that provides ICT and medical technology services for municipalities and and public health care sector. Our case organization has around 220 employees. The case organization uses ISO/IEC 20000 IT service management standard as a management framework for IT services and service management processes. This case study focused on IT service change management. The topic of study was selected by the case organization's representative.

The following data collection methods/sources were used during the study:

- Documentation (change management process description)
- Archives (change request records, service request records)
- Interviews/discussions (change managers, development manager, product/service managers)
- Participative observation (observation period in the case organization, 2 weeks; a change management workshop for product managers, 2 hours)
- Physical artefacts (access to the organization's intranet, demonstrations of a change management tool)

The research team established a case study datastore to manage the material received from the case organization and the material created by researchers. The research team focused on analyzing the documentation and comparing it to IT service management best practices rather than finding errors. The deviations that the team found were reported to the case organization's representatives. According to Yin's [18] case study principles, we established a chain of evidence between data sources and findings. We also created a case study

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datastore for the team's internal use. The datastore proved out to be a valuable tool when we created the case study report to the case organization.

A combination of action research and case study research methods was used as a main research method. It is especially important that action research involves a team that includes both researchers and subjects as co-participants in the enquiry and change experiences on the chosen topic [19].

Baskerville [19] defines action research as a two-stage process. In the diagnostic stage (Stage 1), the researcher and the subjects of the research perform a collaborative analysis of the social situation. In our study, collaborative analysis refers to joint meetings with the case organization's change management specialists, change managers, and a development manager. In these joint meetings, the current state and the target state of change management was discussed. In the therapeutic stage (Stage 2), collaborative change experiments are performed. This stage focuses on introducing changes and studying the effects of changes.

Action research suits well to situations where the goal is to improve working practices and find solutions to existing business problems. The research team did not work only as an external observer but also created material for the change management process. Action research methods were supplemented by case study methods. According to Yin [18], a case study is "an empirical inquiry that investigates a contemporary phenomenon within its real-life context". We used an exploratory case study with a single case design.

B. Data Analysis

The case study data was collected and analyzed by three researchers using a within case analysis technique [20] that focuses on analyzing each case stand-alone before making any comparisons. Research findings were validated with the case organization's change manager and development manager and discussed in the workshop with product managers that played change manager roles. The case study report was produced after the case study findings and was delivered to the case organization's contact persons. This paper documents only part of the findings and research questions that were captured in the case study report.

III. IT Service Change Management - Study Findings

We used KISMET (Keys to IT Service Management Excellence Technique) model as a process improvement tool. The model consists of the following seven phases: Create a process improvement infrastructure, Perform a process assessment, Plan process improvement actions, Improve / implement the process based on ITSM practices, Deploy and introduce the process, Evaluate process improvement and Continuous process improvement. The KISMET model provided researchers a roadmap for the process improvement work.

In this section, we provide a summary of the case study results. We shall start with the functional and data requirements for a change management system.

A. Functional requirements for a change management system

During the study, the functional requirements were identified to improve the existing change management system. The case organization had an old ticketing system that had been created (exceptionally) within the organization. We collected requirements from discussions with a change manager and development manager and edited the comments to match ITSM terminology. The following list captures the functional requirements that we identified:

- The system should enable easy reporting of changes by any employee.
- The system should enable separating development ideas from service requests and incidents.
- The system should enable categorization of changes (major, minor, emergency).
- The system should provide appropriate reports to the change management actors (e.g., number of implemented and failed changes by type and time period; throughput times for changes by change type).
- The system should enable linking incidents, problems and service requests to change records.
- The system should be able to maintain the change history (change log).
- The system should have a relationship to the Configuration Management System to enable collecting information on faulty/vulnerable/resource consuming configuration items.
- The system should be able to support Change Advisory Board (CAB) meetings and provide CAB members actions for reviewing, accepting or rejecting change requests.

B. Data requirements for a change management system

The following data requirements were identified for the change management system. Data requirements were captured by analyzing the specifications of existing ITSM system and comparing it to other ITSM tools and ITSM best practices.

- Unique identifier
- Date of creation
- Change requestor's information
- The target of the change
- Trigger of the change
- Title of the change request
- Description of the change (what do you want to change, what kind of user groups are affected by the change)
- Service Area
- Reason for change
- Estimated impact of the change
- Change category (standard, normal, emergency, major)

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- Schedule, required resources and costs
- Change priority (low, normal, urgent)
- Change effects (affecting a single user, department, or entire organization)
- Recovery Plan
- Impact assessment (resources, capacity, costs and benefits)
- CAB decisions and recommendations
- Accepted/rejected date
- Change status
- The authorization date
- Estimated completion date
- Due date of change (Business opportunity window)
- The actual date of change introduction
- PIR meeting
- PIR evaluation data
- Closure code (success, canceled, abandoned, postponed, implementation)

C. Roles related to the change management process

We observed that change management roles were well visible in the change management process description. The change manager was also responsible for maintaining the process description document. The following roles were identified based on the change management process description:

- Change manager (monitors change requests and implementation of changes, ensures that the process fills its outcomes and purpose, ensures that the process fills the requirements of ISO/IEC 20000, performs continual service improvement)
- Change proposer (a customer, service desk or a specialist identifies a need for a change)
- Change processor (monitors change requests and queues, records change request details in the ticket, if the change proposer has not done it, carries out a preliminary assessment and classifies the change request, calls the CAB, is responsible for implementing the change and Post Implementation Review(PIR))
- Problem manager (responsible for emergency changes during office hours, calls the CAB meeting)
- Application Team duty officer (responsible for emergency changes outside office hours, calls the CAB meeting)
- CAB, Change Advisory Board (evaluates the effects of the change, makes decisions on changes)
- Change Implementor (implements the change by utilizing release management and configuration management, records change history)

D. Improving the change management process

During the study, the research team provided the case organization with a list of suggestions how improve the change management process. Before we present improvements, we have to note several positive findings on the organization's IT service management. First, processes were in good shape. Each process had been clearly and consistently documented and given a responsible person (process manager). Service management employees seemed to be well committed to continuous learning and improvement. We witnessed open information sharing on current challenges (ITSM tools and processes) and that they had been trained how to identify process problems and learn from failures. Some of employees reported that getting a training certificate or an ITSM standard should not slow down the learning and that there is always something to improve. Employees also indicated that management is well committed on improving service quality.

Improvement suggestions/challenges were related to the following parts of change management (the source of an issue is presented in parentheses: Documentation (Do), Archives (Ar), Interviews/Discussions (InD), Participative observation (Po), Physical artefacts (Pa)).

- Post Implementation Reviews for changes (Pa, InD, Ar)
- Increasing awareness of change management practices by organizing Process Manager forums (InD)
- Evaluation and estimation of the changes (Po: Change management workshop)
- Information sharing on change management through a centralized datastore (InD).
- Defining interfaces between change management and other ITSM processes as well as IT project management (Do, Po, InD)
- Defining the change management metrics (Do, Pa, InD)

In almost all of our process improvement case studies, we have observed a need for clearly defined interfaces between IT service management processes. This case revealed the need to define the interface also between change management and project management. This interface is very interesting from the following reasons: First, implementing a change as a project allows the service provider organization to better charge the customer for the work required to achieve objectives of the change. Second, because the size and the complexity of the changes varies, there is a need for guidelines and models that help the organization decide how the change shall be completed. The change can be completed in several ways: as a project, as a normal change or as a hybrid model that is coordinated by a business relationship manager until there are project resources available.

Figure 1 shows the interface between project management and IT service management in the context of non-standard change.

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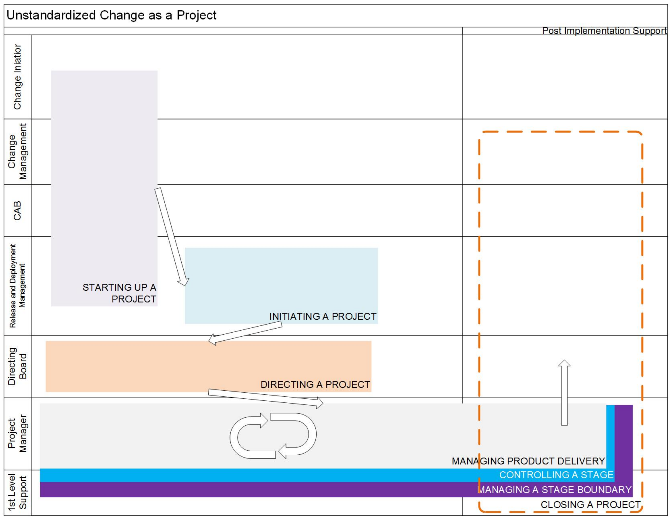

Fig. 1. Interface between project management and IT service management

PIR Meeting – Failed Change Form	
Why did the change implementation fail?	
Was a change back-out plan used? Yes / No	Successful reversal? Yes / No
If a change back-out plan was not used, describe why?	

Potential corrective actions:

Task nro.	Description	Duration	Responsible person/group	Change Effect

Lessons learnt (How a failure could have been prevented?)

Fig. 2. The draft version of the PIR form

IV. Analysis

The data analysis was carried out by using within case analysis technique that focuses on creating a rich view on the case organization's behavior before making any conclusions. Each data source was analyzed from the viewpoint of research questions. In this section, we provide the analysis of findings in the form of lessons learnt. The following lessons learnt were derived from the study.

Lesson 1: Conduct Post Implementation Reviews for changes. Post Implementation Review group should meet after the change implementation and evaluate the success of change (causes, goals, improvement suggestions) and perform a detailed investigation for failed/unsuccessful changes. During the study, we created several templates to assist change management tasks. Figure 2 shows the Post Implementation Review form that should be filled in the PIR meeting in case of a failed change.

Lesson 2: Increase communication on change management practices by organizing Process Manager forums. Interviews showed that process managers need discussion forums to discuss on process practices and make them more unified. One of the interviewees reported: *The challenge is to get daily work and processes together.* In addition to CAB meetings, one could organize change management meetings where change managers and specialists that are related to managing changes could share experiences on change management practices and tools.

Lesson 3: Change evaluation. The change management workshop highlighted the importance and complexity of change evaluation. A participant of the change management workshop stated: *The most challenging part in change management is to estimate the scope and the effects of changes. The evaluation of change is equal to risk estimation.* The change management system should support the decision making, for example, showing the dependencies between service components and modules. Here, the configuration management database plays an essential role.

Lesson 4: Identification of changes. An interview with one of the process managers revealed that employees need guidelines how to identify changes/requests for change and how to separate them from other requests, such as service requests. Interviewees also addressed the need for practical level ITSM training.

Lesson 5: Centralized place for change management guidelines. According to our observations, information on the change management was stored in separate locations, which made it difficult to get an overview how process is executed and monitored. A solution could be to establish an intranet site for change management guidelines and reports. Additionally, we suggested that communication channels to change management office should be clearly visible for employees, such as in the form of an email address (cmo@company.com). The change management office should also communicate its service offering for the organization's employees, for example, which change management training is available.

Lesson 6: Define change management interfaces

During the study, we found that the case organization's change management process had relationships with configuration management, service request management, business relationship management, service level management, release and deployment management, IT financial management, security management, and availability and continuity management. The interface with project management had been mentioned in the change management process description but required clarification. Thus, we provided a visual description of the interface.

Lesson 7: Measuring the performance of change management

A typical challenge in IT service management is how to measure the performance of ITSM processes. A good starting point is to identify critical success factors, key performance indicators and metrics. For example,

- Critical success factor: Service quality and protection of the service.
- Key performance indicator: Percentual decrease in failed changes.
- Metrics: Number of failed changes.

We observed that most of the change management metrics were documented in form of KPIs: Decrease in number of defects due to unauthorized changes, decrease in number of failed changes, decrease in SLA breaches due to changes, decrease in number of urgent changes, better pre-estimation of changes. Our suggestion was to define metrics that the process and tools support. For example, if one would like to use 'decrease in number of failed changes' as a KPI, one has to know exactly the number of failed changes. Additionally, there must be clear rules and criteria what is a failed change. This leads to defining limits and allowed tolerances, for example, a change becomes a failed change when a change implementation exceeds the estimated budget with more than 10 per cent. Our findings support some of the earlier studies in change management such as the study of Sauvé et al. [11] that indicated the need of careful planning of change-related SLAs. In our study, we also identified that it is challenging to create SLAs for error corrections that usually go through problem management process. While change management is often discussed in the contect of organizational changes and software change management, this paper focused on innovative aspect of change management: IT service change management which is important topic for any IT service company. While designing a change management system, it is not enough to focus just on internal requirements (functions and data). One should also take into account the external metrics, such as usability, availability and user satisfaction.

V. Conclusion

The research problem of this study is: Which factors should be taken into consideration while implementing IT service change management systems and processes? By using case study and action research methods, we performed a study in an IT service provider organization and focused on exploring IT service change management from a tool and process perspective. As key results, we presented functional and data requirements for a change management system, benefits that the organization has received from improving change management, and roles and responsibilities within change management.

The key improvement ideas we identified were related to organizing frequent Post Implementation Reviews both for successful and failed changes, classification of changes, defining change management interfaces both with service management processes and project management.

This following limitations are related to our study. First, data were collected from manager level representatives during a relatively short time period (2 weeks). Although we managed to get a rich set of material (5 interviews, informal discussions, 15 documents, 7 observation periods) time should be spent on the data collection and analysis to increase the amount of observation and interviews. Data collection and analysis of employee level information could have provided richer viewpoints to the research topic. Second, we used a single case design in this study. We agree that this causes generalization difficulties for our results. However, case studies and action research studies do not have to lead statistical conclusions. Instead, they can be used to improve the theory. Here, we focused on contributing to IT service management and change management theory. Further research could explore in more detail the interfaces between IT service management processes and IT project management.

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A Branch-and-Cut Algorithm to Solve the Container Storage Problem

Ndèye Fatma Ndiaye
University of Le Havre
Le Havre, France
e-mail: farlou@live.fr

Adnan Yassine
Superior institute of logistic studies
Le Havre, France
e-mail: adnan.yassine@univ-lehavre.fr

Ibrahima Diarrassouba
University Institute of Technology
Le Havre, France
e-mail: diarrasi@univ-lehavre.fr

Abstract— **We study the Container Storage Problem in port terminal (CSP), which consists to effectively manage the storage space so as to increase the productivity of port. When a ship arrives, the inbound containers are unloaded by Quay Cranes (QC) and then placed on quays. So, they are collected by Straddle Carriers (SC). Each is able to carry one container at a time, and store it in its storage location. In order to reduce the waiting times of ships, we propose a mathematical model which minimizes the total distance traveled by SC between quays and container yards. In this paper, we take into account additional constraints which are not considered previously. We also propose an effective branch-and-cut algorithm (BC-CSP), which is an optimal resolution method. Performed simulations prove the effectiveness of our algorithm.**

Keywords-Container Storage Problem; complexity; branch-and-bound; CPLEX.

I. INTRODUCTION

In a seaport, the container terminal manages all actions concerning containers. Generally, three types of containers are distinguished: outbound, inbound, and transshipment containers. All these containers are temporarily stacked in the container yard, before leaving the port. Outbound containers are brought by External Trucks (ET), also picked up by SC which store them in their storage location, and then loaded onto vessels. Inbound containers are unloaded from vessels by QC, transported to their storage locations by SC, and then recuperated later by ET. Nowadays, the competition between ports is very high. Therefore, each of them tries to improve continuously the quality of its service in order to attract more customers. The most important criteria to measure service level include the waiting time of ET which collect inbound containers. In fact, when an ET arrives at port and claims a specific container, it waits during all the time required to retrieve it. If the desired container is under others, it may be necessary to move these containers in first. This kind of movements, named reshuffles, are unproductive and require so much time. Therefore, it is very important to optimally store containers so as to avoid them. Another important criterion to measure the quality of service is the time required to unload ships. The importance of this factor is justified by the fact that it is more beneficial for both the port and the customers to shorten the stay of vessels. First, it is better for the port to quickly free the berths in order to allocate them to others incoming vessels. Second, generally, ship-owners rent vessels; therefore, they tend to minimize the berthing durations in order to reduce rental costs. These two issues are addressed in this paper. We consider a modern container terminal, which uses SC instead of Internal Trucks (IT). The advantage of a SC is that it is able to lift and to store a container itself; therefore it is not necessary to use Yard Cranes (YC). A storage yard is composed of several blocks. In order to allow good circulation of SC, each block is composed of bays which are separated by small spaces. In every bay, there are stacks wherein containers are stored. A stack must have a height inferior or equal to the limit fixed by the port authorities. Fig. 1 shows an example of block wherein circulate straddle carriers.

Fig. 1 Straddle carriers circulating in a containers yard

We propose in this paper an effective method to solve the container storage problem. For this, we propose a new mathematical model which determines the optimum storage plan and minimizes the time required to transport containers between quays and storage areas. For the numerical solution

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of the model, we propose an effective branch and cut algorithm (BC-CSP), which is an exact method.

The remainder of the paper is organized as follows: a literature review is given in the second part. A detailed description of the addressed problem is exposed in the third part. The complexity of the problem is discussed in the fourth section, while the proposed mathematical model is explained in the fifth section. The branch-and-cut algorithm is itemized in the sixth section; the numerical results are presented in the seventh section. Our conclusion is given in the eighth section.

II. LITERATURE REVIEW

There are many papers addressing the storage of outbound containers than inbound containers. However, there are some papers which consider both simultaneously. Zhang et al. [2] considered in addition to these two categories, those which are in transition, that means containers which are unloaded from some vessels and are waiting for being loaded onto others. They used the rolling-horizon approach [2] to solve the storage space allocation problem. For each planning horizon, they solved the problem in two steps formulated as mathematical programs. In the first step, they determined the total number of containers assigned to each block at a period so that the workloads of loading and unloading of each vessel are balanced. Then, in the second step they determined the number of containers associated to every vessel in order to minimize the total distance traveled to transport these containers from quays to the storage blocks. Bazzazi et al. [3] proposed a genetic algorithm to solve an extended version of the Storage Space Allocation Problem (SSAP). It consists to allocate temporarily locations to the inbound and outbound containers in the storage yard according to their types (regular, empty, and refrigerated). They aimed to balance the workloads between blocks with the goal to minimize the time required to store or to retrieve containers. Park and Seo [4] dealt the Planar Storage Location Assignment Problem (PSLAP), in which only planar movements are allowed. The purpose of PSLAP is to store inbound and outbound containers so as to minimize the number of moving obstructive objects. The authors made a mathematical formulation of PSLAP and proposed a genetic algorithm to solve it. Lee et al. [5] combined the truck scheduling and the storage allocation problems. They considered inbound and outbound containers, and tend to minimize the weighted sum of total delay of requests and the total travel time of yard trucks. For numerical resolution, they proposed a hybrid insertion algorithm. Kozan and Preston [6] developed an iterative search algorithm using a transfer model and an assignment model. At first, the algorithm determined cyclically the optimum storage locations for inbound and outbound containers, and secondly, it found the corresponding handling schedule. They solved the problem using a genetic algorithm, a tabou search algorithm and a hybrid algorithm.

Concerning inbound containers, most of papers deal with the management of reshuffles. Sauri and Martin [7] proposed three different strategies to store inbound containers. The purpose of their work is to determine the best strategy which minimizes re-handles in an import container yard. For this, they developed a mathematical model based on probabilistic distribution functions to evaluate the number of reshuffles. Kim and Kim [8] considered a segregation strategy to store inbound containers. This method does not allow placing newly arriving containers over those which arrived earlier. Therefore, storage spaces are allocated to each vessel so as to minimize the number of reshuffles expected during the loading operations. Jinxin et al. [9] proposed an integer programming model, which addressed the trucks scheduling and the storage of inbound containers. They minimized at the same time the number of congestions, the waiting time of trucks, and the unloading time of containers. The authors designed a genetic algorithm to solve the model, and another heuristic algorithm which gave them best results. Yu and Qi [10] treated the storage problem of inbounds containers in a modern automatic container terminal. They aimed to minimize reshuffles in two steps. For this, they first resolved the block space allocation problem for newly arriving inbound containers, and then after the retrieving of some containers they tackled the re-marshaling processes in order to re-organize the block space allocation. They suggested three mathematical models of storage containers, the first is a non-segregation model, the second is a single-period segregation model, and the last is a multiple-period segregation model. They conceived a convex cost network flow algorithm for the first and the second models, and a dynamic programming for the third. They found that the re-marshaling problem is NP-hard and then designed a heuristic algorithm to solve it. We considered in [1] a container terminal wherein reshuffles are not allowed. We proposed a new mathematical model to allocate storage spaces to inbound containers in such a way that no reshuffles will be necessary to retrieve them later. We designed a hybrid algorithm including genetic algorithm and simulated annealing to solve it.

In most container terminals, the departure times of inbound containers are unknown. K. H. Kim and K. Y. Kim considered in [11] a container terminal in which there is limited free time storage for inbound containers, beyond which customers have to pay storage costs per time unity. The authors proposed a mathematical model to find the optimal price schedule.

Papers dealing with the storage problem of outbound containers have generally different goals. Preston and Kozan [12] proposed a Container Location Model (CLM) to store outbound containers in such a way that the time service of container ships is minimal. They designed a genetic algorithm for the numerical resolution. Kim et al. [13] developed a dynamic programming model to determine the storage locations for outbound containers according to their weight. They minimized the number of relocations expected during the ships loading. They also made a decision tree using the set of optimal solutions to support real-time decisions. Chen and Lu [14] addressed in two steps the storage space allocation problem of outbound containers. In the first step, they used a mixed integer programming model to calculate the number of yard bays and the number of locations in each of them. So, in the second step, they determined for each container the exact location where it must be stored. Woo and Kim [15] proposed a method to allocate storage spaces to groups of outbound containers. They reserved for each group of containers having the same attributes, a collection of adjacent stacks. At the end, the authors proposed a method to determine the necessary size of the storage space expected for all the outbound containers. Kim and Park [16] gave two linear mathematical models to store outbound containers. In the first, they considered a direct transfer, and so, in the second, they dealt with an indirect transfer system. They designed two heuristic algorithms to solve these models. The one is based on the duration-of-stay of containers, and then, they used the sub-gradient optimization technique in the other.

Among the few papers dealing only transshipment containers include that of Nishimura et al. [17]. They developed an optimization model to store temporarily transshipment containers in the storage yard, and proposed a heuristic based on lagrangian relaxation method for the numerical resolution.

III. CONTEXT

When a container ship arrives at port, QCs unload containers and place them on quays. So, they are picked up by SCs, which carry and store them in the container yard. The firsts containers which are placed on quays are the first picked up. In order to avoid congestion on quays, which could increase the time required to unload ships, we minimize the total distance traveled by SCs between quays and the container yard. In this study, we consider the following five main hypotheses:
(1) Reshuffles are not allowed,
(2) In each stack, containers are arranged according to:
  (2.a) the same order that they are unloaded from ships,
  (2.b) and the descending order of their departure time,
(3) In a stack, all containers have same dimensions,
(4)We take into account containers which are already present in the storage areas before the start of the new storage period,
(5) We respect the maximum capacity of each stack.

Excepted (2.a), these hypotheses are considered in [1].

IV. COMPLEXITY OF THE PROBLEM

In this section, we study the complexity of the CSP. In particular, we show that it is equivalent to the Bounded Coloring Problem (BCP); therefore, is NP-hard in the general case.

A. Some reminders about the BCP

Let us begin by recalling some concepts and definitions that will be useful for the following.

1) Preliminary notions: Let *G(V,E)* be an undirected graph, *V* is the set of vertices and *E* is the set of edges.

G is a *comparability graph* if and only if there are a sequence of vertices $v_1,...,v_n$ of *V* such that for each *(p, q, r)* checking *1 < p < q < r < n,* if $(v_p, v_q) \in E$ and $(v_q, v_r) \in E$ then $(v_p, v_r) \in E$.

A *co-comparability* graph is the complement of a comparability graph.

An undirected graph *G = (V,E)* is a permutation graph if and only if there are a sequence of vertices $v_1,...,v_n$ of *V* and a permutation *σ* of the vertices such that for all *i* and *j* satisfying $1 \le i \le j \le n$, $(v_i, v_j) \in E$ if and only if *σ(i)≥σ(j).*

Theorem 1: A graph *G* is a permutation graph if and only if *G* and its complement are comparability graphs [18].

2) The bounded coloring problem: Given an undirected graph *G = (V,E)*, a set of s colors $l_1,...,l_s$, an integer *H* and a vector that gives the weight of assigning a color l_i to a vertex of the graph. The bounded coloring problem with minimum weight consists to determine a minimum weight coloring of *G* using at most *s* colors in such a way that a color is assigned to at most *H* vertices.

Theorem 2: [19] The bounded coloring problem with minimum weight is NP-hard for the class of permutation graphs for all $H \ge 6$.

B. Equivalence between the CSP and the BCP

We show that the CSP is NP-hard. For this, we introduce an undirected graph *G(N,O,T) = (V,E)* constructed from an instance of the CSP, where *N* is the set of containers and *O* and *T* are vectors, which give respectively the unloading order and the departure times of each container. The graph *G* is constructed as follows. A vertex of the graph corresponds to a container. To simplify notation, the index *k* is used to denote both a container and the vertex of the graph which corresponds to it. There is an edge between two vertices *k* and *k'* if and only if $O_k < O_{k'}$ and $T_k < T_{k'}$. We have the following lemma.

Lemma 1: The graph *G(N,O,T)* obtained from a instance of CSP is a permutation graph.

Proof: To prove that the graph *G(N,O,T)* is a permutation graph, it suffices to show that it is a comparability graph as well as its complement (see Theorem 1).

First, we show that *G(N,O,T)* is a comparability graph. The vertices are ordered according to the same order that the unloading of the corresponding containers from ships. If two containers *k* and *k'* are unloaded from ships at the same time (that is to say if $O_k = O_{k'}$), then the vertices *k* and *k'* are ordered in the ascending order of their departure times. If

$O_k = O_{k'}$ and $T_k = T_{k'}$ then the vertices are ordered in the lexicographical order. Without loss of generality, we consider that the vertices are numbered in the order that is previously determined. Now, consider any three vertices *k, k'* and *k''* of the graph such that $k < k' < k'', (k,k') \in E$ and $(k',k'') \in E.$ We will prove that necessarily $(k,k'') \in E$. As $(k,k') \in E$ and $(k',k'') \in E$, we have $O_k < O_{k'}$ and $T_k < T_{k'}$, and we have also $O_{k'} < O_{k''}$ and $T_{k'} < T_{k''}$. We thus obtain that $O_k < O_{k'} < O_{k''}$ and $T_k < T_{k'} < T_{k''}$, which implies that the graph *G(N,O,T)* has an edge between vertices *k* and *k''*. So *G(N,O,T)* is a comparability graph.

Now, we will prove that the complement of *G(N,O,T)*, denoted $\overline{G}(N,O,T)$ is also a comparability graph. First, note that there is an edge between two vertices *k* and *k'* of $\overline{G}(N,O,T)$ if and only if there is no edge in *G(N,O,T)* between *k* and *k'* in other words $O_k < O_{k'}$ and $T_k > T_{k'}$. The vertices of $\overline{G}$ are ordered in the same order as those of G. As before, for any three vertices *k, k'*, and *k''* of the graph $\overline{G}(N,O,T)$ such that $k < k' < k'', (k,k') \in E$ and $(k',k'') \in E,$ we have $O_k < O_{k'} < O_{k''}$ and $T_k > T_{k'} > T_{k''}$. So, $O_k < O_{k''}$ and $T_k > T_{k''}$, and then there is an edge between *k* and *k''* in $\overline{G}(N,O,T)$. Therefore, $\overline{G}(N,O,T)$ is also a comparability graph.

Now, it is easy to see that a solution of the container storage problem is a solution of the corresponding bounded coloring problem. In fact, a similar result is given in [20]. Consider an instance *ICSP = (N,O,T,* N_p *,H,r,R,d)* of the CSP and the graph *G(N,O,T)* associated. Now, consider an H-coloring of *G(N,O,T)* that has s colors. Each color of the bounded coloring problem is matched to a stack of the CSP. Indeed, as all vertices having the same color form a stable set, in other words they are not connected by any edge, therefore any two containers corresponding to two vertices of this stable set satisfy these two inequalities $O_k < O_{k'}$ and $T_k \geq T_{k'}$. The unloading order as well as the departure times of containers corresponding to the vertices of a stable set are compatible; thereby, they can be stored in a same stack if it has enough empty slots. In addition, there are at most *H* vertices in this stable set. So, the number of containers assigned to the corresponding stack is inferior or equal to *H*. Therefore, an H-coloring corresponds to a valid assignment for the CSP. Similarly, it is easy to see that a solution of the CSP is a solution of the H-bounded coloring problem in the graph *G(N,O,T)*. We have the following lemma.

Lemma 2: Let *ICSP = (N,O,T,* N_p *,H,r,R,d)* an instance of the container storage problem. The CSP has a solution for this instance if and only if the bounded H-coloring problem on the graph *G(N,O,T)* has a solution.

We now give the main result of this section.

Theorem 3: The container storage problem is equivalent to the bounded coloring problem with minimum weight.

Proof: To establish this result, we prove that an instance of the CSP is equivalent to an instance of the BCP and vice versa. Let *ICSP = (N,O,T,* N_p *,H,r,R,d)* an instance of the storage container problem and *G(N,O,T)* the permutation graph associated. Consider *IBCP = (G(N,O,T),H,* N_p *,d)* an instance defined on the graph *G(N,O,T)*, where *Np* is the number of colors, *H* is the bound, and *d* a matrix containing the weights. According to the Lemma 2 a solution of the CSP is a solution of the BCP, and similarly a stack of CSP corresponds to a color of BCP and vice versa. It follows then that the cost d_p^k of assigning a container $k \in N$ to the stack $p \in N_p$, is the same as the assignment of the vertex *k* to the color corresponding to the stack *p*. So, the cost of H-coloring in the graph *G(N,O,T)* is the same as the cost of the solution of the corresponding CSP and vice versa. Therefore, we can find the optimal solution of the CSP if and only if we find the optimal solution of BCP.

According to the Theorem 2 the bounded coloring problem is NP-hard for the class of permutation graphs if $H \geq 6$. It therefore follows from Theorem 3 that the CSP is NP-hard if $H \geq 6$.

Corollary 1: The container storage problem is NP-hard if the maximum capacity of every stack is superior or equal to six.

V. MATHEMATICAL MODELING

In the mathematical model, we use the following indices:

p: stack,
k: container.

The data of the problem are:

N : number of containers,
N_p : number of stacks,
c_p : number of empty slots in the stack p,
r_p : type of container which can be placed in the stack *p*,
t_p : departure time of the container which was on the top of the stack p at the begin of the new storage period.

R_k : type of the container k,

T_k : departure time of the container k,

O_k : unloading order of the container k from ships.

d_p^k : traveled distance to transport the container k from quay to stack p,

G(V,E): a graph, where *V* is the set of vertices and *E* the set of edges. Every vertex represents a container, and |V| = N. There is an edge between two vertices v_k and $v_{k'}$ if and only if $T_k < T_{k'}$ and $O_k < O_{k'}$, this means that container *k* and *k'* can't be assigned to a same stack.

The decision variables are defined as follows:

$$x_p^k = \begin{cases} 1 & \text{if container } k \text{ is assigned to stack } p \\ 0 & \text{otherwise} \end{cases}$$

We propose the following mathematical model:

$$Minimize \sum_{k=1}^{N} \sum_{p=1}^{N_p} d_p^k x_p^k \quad (1)$$

$$\sum_{p=1}^{N_p} x_p^k = 1, \quad \forall k = 1,..., N \quad (2)$$

$$x_p^k + x_p^{k'} \le 1, \quad \forall (k,k') \in E, p = 1,..., N_p \quad (3)$$

$$\sum_{k=1}^{N} x_p^k \le c_p, \quad \forall p = 1,..., N_p \quad (4)$$

$$\sum_{1 \le k \le N, T_k > t_p or R_k \ne r_p} x_p^k = 0, \quad \forall p = 1,..., N_p \quad (5)$$

$$x_p^k \in \{0,1\} \quad \forall p = 1,..., N_p, k = 1,..., N \quad (6)$$

The objective function (1) minimizes the total distance traveled between ships and the container yard. Constraints (2) require that each container is assigned to a single stack. Constraints (3) ensure that the containers of each stack are arranged following the same order that they were unloaded from ships, and the decreasing order of their departure times. Constraints (4) enforce the stack capacity. Constraints (5) secure the compatibility between containers and stacks.

Let k a vertex of the graph ($1 \le k \le N$), *N(k)* the set of its neighbors, *p'* a stack ($1 \le p' \le N_p$). Constraints (3) lead to the following neighborhood inequality as in [21].

$$\sum_{k' \in N(k)} x_{p'}^{k'} + |N(k)| x_{p'}^{k} \le |N(k)| \quad (7)$$

Proposition 1: For an integer solution, the inequalities (3) and (7) are equivalents.

Proof: It suffices to prove that (7) implies (3), because the reverse is highlighted by the definition of (7).

Since $x_{p'}^k$ is a binary variable, then it can be equal to either 0 or 1.

- If $x_{p'}^k = 1$ then $\sum_{k' \in N(k)} x_{p'}^{k'} = 0$. Therefore, for all *k'* neighbor of *k*, we have $x_{p'}^{k'} = 0$. Thereby $x_{p'}^k + x_{p'}^{k'} = 1, \forall k' \in N(k)$.
- If $x_{p'}^k = 0$ then $\sum_{k' \in N(k)} x_{p'}^{k'} \le |N(k)|$ which means that for all *k'* belonging to *N(k)*, $x_{p'}^{k'}$ can be equal to either 0 or 1. Thus, $x_{p'}^k + x_{p'}^{k'} \le 1, \forall k' \in N(k)$.

VI. BRANCH-AND-CUT ALGORITHM

The branch-and-cut is an improvement of the branch-and-bound, which is an exact resolution method. Each of these two methods uses a search tree to explore the solution space. To do this, the search space is divided into smaller subsets, each representing a node of the search tree. So, the problem is solved by considering one by one all subsets. This strategy is called *divide and conquer*.

To build the search tree, we first create the root node; it corresponds to the released problem. Other nodes of the tree are obtained by making connections.

In the branch-and-cut, unlike the branch-and-bound, at each node of the search tree, some constraints called *valid inequalities* are added to the released problem so as to improve the solution.

A. Relaxation of the problem

After the relaxation of integrity constraints (6), we find that the total number of constraints of the mathematical model remains great. Therefore, since the adjacency constraints (3) are equivalent to the neighborhood inequalities (7), so we delete them from the model knowing that the admissibility of solutions will be ensured by the gradual addition of valid inequalities along the branch-and-cut algorithm.

B. Preprocessing

The number of variables increases depending on the number of stacks (N_p) and containers *(N)*. In the case where all stacks were empty at the beginning of the storage period, we can reduce the number of variables. We consider that all containers are equidistant to the stacks. Knowing that, generally, $N_p \geq N$, we only use the N stacks which are more near to quays. This allows to significantly reduce the number of variables and to speed up the computation.

C. Upper bound

To find an upper bound, we solve the bounded vertex coloring problem on the graph defined in section V. Each color corresponds to a stack. We use a heuristic algorithm which colors vertices one by one following the descending order of their number of uncolored neighbors. For each vertex, it chooses among the admissible colors the one that fits to the nearest stack. The eligible colors are those not assigned to a vertex which is a neighbor of the considering vertex, and correspond to the stacks which are not full. Whenever a vertex is colored, the number of empty slot of the stack corresponding to the used color is reduced.

D. Branchings rules

We use the classical branching rule. At each node of the search tree, we create two branches by rounding the largest fractional variable. Let x_p^k this variable. We put $x_p^k = 0$ in a branch; it means that container k will not be assigned to stack p in this branch. Then, in the other branch, we put $x_p^k = 1$, which means that container k will inevitably be assigned to stack p in this branch.

E. Separation method

At each node of the search tree, before creating branches, we use a simple heuristic algorithm to look for neighborhood inequalities which are violated. To do this, we treat one by one all variable which is superior to 0.5 in the optimal solution of the current node. Let x_p^k one of these variables and S an integer initialized to zero. We calculate the number $|N(k)|$ of neighbors of the vertex k. Then, we add to S the value of x_p^k multiplied by $|N(k)|$. And we seek all variables $x_{p'}^{k'}$ such that k and k' are neighbors and $p=p'$, and we add the sum of their values to S. If S > $|N(k)|$ then there is a violated inequality therefore we add to the sub-problem a constraint to avoid this.

F. Description of the algorithm

1: We begin by solving the problem using a heuristic algorithm to find an upper bound named BS.
2: Then, we create the root node of the search tree which represents the released problem.
3: We solve the sub-problem using the CPLEX solver.
4: Then, we seek all neighborhood inequalities that are not satisfied by the solution of the current node, and then we add them to the released problem.
5: We solve the problem again using the CPLEX solver.
6: If the solution is integer and inferior to BS then we update BS.
7: If the solution is fractional and inferior to BS then we do:
- 7.a: Perform connections,
- 7.b: Choose an unexploited node,
- 7.c: Go back to 3.

VII. NUMERICAL RESULTS

In this section, we present the numerical results of our branch-and-cut algorithm. For the implementation, we use SCIP, which is a framework allowing a total control of the solution process. The experiments were performed using a computer DELL PRECISION T3500 with an Intel Xeon 5 GHz processor.

To test the effectiveness of our algorithm, we naturally compare it to CPLEX version 12.5. Performed tests on several instances prove that BC-CSP is very fast and it is able to solve large instances which can not be solved by CPLEX because requiring a lot of memory.

In Table I, we note the execution times of BC-CSP and of CPLEX for various instances.

— means that the execution is interrupted because it lasted more than 3 hours.

--- means that the computer memory is insufficient to resolve this instance.

TABLE I. NUMERICAL RESULTS

N	N_p	BC-CSP	CPLEX
100	500	0 sec	2 min 58 sec
150	500	1 sec	15 min 45 sec
100	700	0 sec	4 min 11 sec
100	1500	0 sec	11 min 16 sec
50	200	0 sec	2 sec
200	200	3 sec	14 min
80	100	0 sec	1 min 5 sec
90	100	0 sec	1 min 29 sec
100	100	0 sec	2 min 11 sec
150	200	1 sec	53 min 50 sec
100	3500	0 sec	---
100	3500	0 sec	---
200	3500	3 sec	---
300	3500	14 sec	---
400	3500	41 sec	---
500	3500	1 min 36 sec	---
600	3500	6 min 13 sec	---
700	3500	5 min 57 sec	---
800	3500	9 min 49 sec	---
900	3500	15 min 35 sec	---
1000	3500	1 h 41 min 26 sec	---
1100	3500	1 h 43 min 47 sec	---
1200	3500	2 h 5 min 4 sec	---
1300	3500	2 h 21 min 20 sec	---
1400	3500	—	---

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In Table I, we remark that, in the most cases, the resolution of a great instance requires more time than the resolution of a small instance. However, in some cases, we observe the reverse. This phenomenon can be justified by the influence of the values of parameters like the departure times, the unloading order, etc. In fact, in some cases the search tree can have too lot of nodes; therefore its exploration may require more time. But, even with these instances, our branch-and-cut algorithm is faster than CPLEX.

The mathematical model of our container storage problem has too many variables, especially when there are a lot of empty stacks in the terminal. Therefore, the elimination of the farthest stacks reduces the size of the problem and improves the resolution. Fig. 2 shows that preprocessing reduces the execution times.

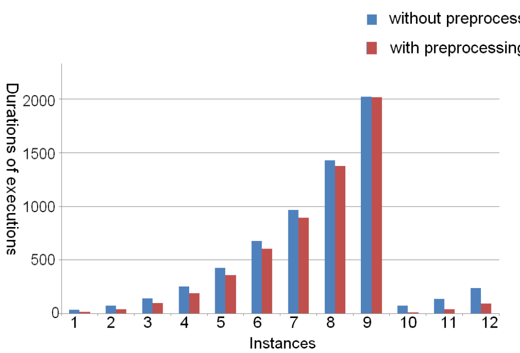

Fig. 2 Comparisons of execution times

As can be seen in Fig. 2, the preprocessing is more efficient when the number of stacks is superior to the number of containers.

VIII. Conclusion and Future Work

In this paper, we studied the container storage problem. We widely improve the work that we did in [1] by considering additional constrains in order to avoid reshuffles at quays. We take into account the order in which containers are unloaded from vessels, and we minimize the total distance traveled by SC between quays and container yards in order to shorten the berthing times of ships. The major contribution of this paper is the effective branch-and-cut algorithm, which is very fast and is able to solve great instances. This is an exact resolution method, unlike the hybrid algorithm proposed in [1], which has an average percentage deviation equal to 10.22%. It may be possible to improve our branch-and-cut algorithm; therefore we prospect to design more effective branching rules and separation methods. We also plan to adapt our approach to container terminals which use modern equipments, such as automatic guided vehicles.

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Towards Interoperability to the Implementation of RESTful Web Services: A Model Driven Approach

Nírondes Tavares and Samyr Vale
Department of Computer Science
Federal University of Maranhão
São Luis – MA – Brazil
nirondes@gmail.com, samyr@deinf.ufma.br

***Abstract* — Nowadays, a issue that has been gaining relevance is the RESTful Web services technology. RESTful has been more prominent than SOAP due to the interoperability gained by the Web support. Nevertheless, the implementation tier of the RESTful Web Services can be developed with various programming languages and with different specifications, frameworks, and plugins. This diversity of ways to implement RESTful Web services reduces the interoperability proposed by this technology and prevents reuse. In this paper, we present a model-driven approach to implement RESTful Web Services. By the use of concerns separation, modeling on different abstraction levels and by the use of transformation rules we solve the interoperability lack.**

Keywords-Web service; RESTful; modeling; transformation; MDE;

I. INTRODUCTION

The development of the Web has transformed the exchange of organizational information. The information that was previously accessed and presented only via browser could, with the advent of Web services, also be accessed by other ways. This allows that besides humans, computer programs could also make use of this information [1].

Among the existing Web service architectures, the Representational State Transfer (REST) [2] has been gaining space especially for the lightweight and convenience in the use and dissemination of information.

RESTful Web services aims to be interoperable, because a client, that uses a specific platform, can establish communication with a server that uses a different platform. But, the interoperability feature is just at the communication level, because both parts communicate through the Hypertext Transfer Protocol (HTTP) layer, which is the standard for the Internet communication. However, RESTful Web services, as well as SOAP ones, need to be implemented with a specific programming language.

Today, there exist numerous of programming languages (e.g., Java, C++, C #, etc) that support the development of such services, and within these languages, there is still a significant amount of different specifications, Application Programming Interfaces (APIs), frameworks, and plugins. Face of this variety of ways, the problem of interoperability arises in the implementation tier of RESTful Web services. To be interoperable, the programming languages must conform to a certain degree of compatibility with the others [3].

Model Driven Development has been applied in different issues as an approach which provides, by concerns separation, the development of business logic independent of technologies and programming languages. Model driven approaches, such as Model Driven Engineering (MDE) and Model Driven Architecture (MDA) provide interoperability by metamodeling and transformation techniques.

MDE uses the model as the main artifact and the transformation as the principal activity in system modeling and development.

In [4], we have presented the use of MDE to the development of the syntactic and the semantic description of Semantic RESTful Web Services. In this paper, we propose a MDE-based approach to solve the problem of the lack of interoperability in the Application tier of the RESTful Web Services technology. Thus, we present the WSSR metamodel proposed for the implementation of interoperable RESTful Web Services. We also present a target platform metamodel and the transformation rules to the implementation of the RESTful Web Service in the chosen target language.

This paper is organized as follows. Section 2 presents an overview of the main technologies used in the proposed solution. Section 3 presents our model-driven approach for implementing RESTful Web services. Section 4 presents a case study with a sample implementation of a RESTful Web service. Finally, Section 5 concludes the paper.

II. TECHNOLOGICAL CONTEXT

A. RESTful Web Services

Web Services can be defined as a way to let applications exchange information with Web servers [5]. In this interaction, the information exchanged may be contained inside documents written in a machine-readable format. The most used formats are Extensible Markup Language (XML) and JavaScript Object Notation (JSON) [6].

Currently, the main architecture of Web services is the Remote Procedure Call/Simple Object Access Protocol (RPC/SOAP), which is a World Wide Web Consortium (W3C) standard for the information exchange between Web services. The RPC/SOAP architecture is XML based, which ensures interoperability regardless of the technology used by the parts [7]. This is the central advantage of this architecture

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front of its predecessors, like Distributed Component Model (DCOM), Common Object Request Broker Architecture (CORBA) and Java Remove Method Invocation (RMI). Its predecessors were developed in specific technologies which difficult the communication between parts of services [5]. The RPC/SOAP, must be encapsulated within a standard format package named as SOAP envelope and must define a contract containing the communication rules. This contract is is defined in the Web Services Description Language (WSDL) document [8], which performs, among other functions, the syntactic description of the Web service elements. However, the RPC/SOAP architecture also has drawbacks. The excessive use of envelopes difficult the traffic and bring the addition of unnecessary computations, low performance and poor scalability [9].

In this context, the REST architecture is gaining importance, especially in the era we live in, the Web 2.0 [6][9]-[12]. This architecture was created by Roy Fields [2], one of the HTTP protocol creators. The main advantage of the REST architecture is the fact that communication occurs directly on the HTTP layer, without encapsulation need or use of envelopes, and it uses the basic elements of the protocol, like verbs and status codes. REST architecture focuses on resources, not on procedure calls or services, and it is an interesting approach for applications where the focus on interoperability is more important than the formal contract between parts.

Richardson and Ruby [13] have created the term RESTful to describe Web services that follow the REST paradigm and also created the Resource-Oriented Architecture (ROA) to define the architecture that faithfully follow the concepts and properties, defined by [2].

A resource is any real-world entity exposed on the internet and accessible by a Uniform Resource Identifier (URI). The URI is responsible for distinguish a resource from other. The resource can be a text, image, or even a device. The representation is the state of a resource at a given moment, in other words, representations are some data that represents a resource and are serialized to a machine-readable document like XML or JSON [12]. The representations are stateless, which means that each transaction does not keep information related to the previous transaction. This issue is solved using the concept of connectivity, which means that each representation has a link to the subsequent representation. The resources must provide a uniform interface for its handling, in other words, must always be accessed through the same URI, changing only the HTTP verbs. The most used verbs are POST, GET, PUT and DELETE, respectively associated to Create, Read, Update and Delete operations (also known as CRUD operations).

The syntactic description of RESTful Web service is optional and there still no consensus about what language should be standard [11][14]. The Web Application Description Language (WADL) is gaining notoriety despite the new 2.0 version of WSDL, which can also be used to syntactically describe this kind of service. According to Richardson et al. [13], WADL is “the most simple and elegant solution” to solve this problem.

B. *RESTful Implementation*

RESTful provides a multi-tier Web architecture composed by the Client, Application and Data tiers. The independence of these tiers provides flexibility to these dynamic Web applications. Divers programming languages, frameworks and styles can be used for developing each tier [13]. In addition, the Services can be developed in a specific language, such as Java, Python, PHP, Perl or Ruby and be consumed by an application written in a different language.

The Application tier provides a Presentation layer, a Business Logic layer and a Database Connector layer. The Business Logic intermediates between Presentation and Database Connector. Data is provided to the Presentation layer in the structure of objects. For example, in the Rails technology, services send and accept representations of active objects. These services map URIs to Rails controllers, Rails controllers to resources and resources to active objects. Despite the diversity of technologies available to develop RESTful Web Services, few of them are interoperable.

The Java platform has been the most powerful, flexible and user friendly platform for implementing the RESTful Web Services Application tier [15]. The development of RESTful Web services with Java is possible since 2008, when a new specification known as JAX-RS (The Java API for RESTful Web Services) [16] was defined to facilitate the implementation of such services. JAX-RS is based on metadata grammar of JDK 5, supporting the standardization of the RESTful services implementation. Today, JAX-RS is part of JavaEE 6 [17].

However, there exist several frameworks and APIs to implement RESTful services in Java as RESTEasy, Restlet, Struts2, Grails, Axis2, Certia4, sqlREST, REST-art [15].

The variety of programming languages also prevents reuse of the Business Logic implementation and reduces the interoperability due to the different formats and specifications used to implement the services.

C. *Model Driven Approach*

Model-Driven Engineering (MDE) is a software engineering approach that has gained significance in recent years. In MDE, the model is the central figure in the development of applications. The source-code is automatically generated since the application of transformation rules on pre-defined models [1].

By definition, the models are abstract entities that represent various aspects presented in the software, such as structure, behavior and graphical user interface [18]. MDE has two main approaches: MDA from the Object Management Group (OMG) and Eclipse Modeling Framework (EMF) from the Eclipse Foundation. The EMF provides a development framework which supports the MDA-based approach. We have been using concepts and resources of both approaches.

Figure 1 shows how the MDA approach involves the use of modeling languages, abstraction levels and independence of platform and programming languages. The M0 layer represents the real-world objects. The M1 layer is a model representation of the previous layer, represented by a modeling language. The models in M1 layer are defined using concepts described by metamodels in M2 layer. Each metamodel of M2 layer determines how expressive models can be. Analogously, metamodels are defined using concepts described by meta-metamodels in the M3 layer [19].

The Unified Modeling Language (UML) and Enterprise Distributed Object Computing (EDOC) are examples of modeling languages. At the M2 level UML and EDOC are defined by their metamodels which represent the elements of the structure of the modeling language. At the M3 level, we can use the OMG's Meta Object Facility (MOF) language or the Ecore language defined by the EMF, as depicted by the Figure 1.

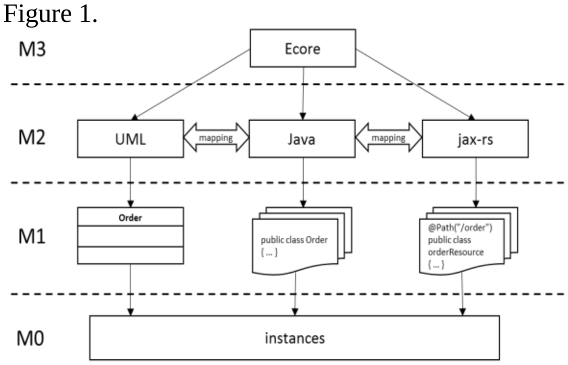

Figure 1. MDA's Abstraction Levels

In MDA, transformations are performed from source models to target models according to mappings, which are created by the identification of semantic correspondences between elements present in both models. The transformations rules are defined based on these mappings, executed by a transformation engine (e.g., the EMF engine) and written using transformation languages, such as Atlas Transformation Language (ATL) or OMG's Query View and Transform Language (QVT).

The ATL language (chosen in this study) provides a simple Object Constraint Language (OCL) based declarative language that facilitates the definition of transformation rules and it is available in a toolkit format to be used together with Eclipse [20].

The Eclipse Modeling Framework is an integration tool that uses class diagrams to represent metamodels and supports creation, storing, changing and opening model instances in XML Metadata Interchange (XMI) format. EMF unifies three important technologies: Java, XML and Unified Modeling Language (UML) with the Eclipse Integrated Development Environment (IDE) [19].

The main advantages of the MDA approach are portability, interoperability, reusability and technology independence, acting on the architectural concepts of separation between specification and implementation of software [21]. Thus, software engineers no longer need to worry about details of implementation language, focusing on the business rules and minimizing the occurrence of errors. The development of models containing only the business logic independent of technological details (platform, programming languages, and architectures) makes the software more portable. These business models can be mapped to many platforms only by the creation of new transformation rules [1].

III. MODEL DRIVEN RESTFUL WEB SERVICES

Fokaefs et al. [22] discuss the interoperability issue raised when two services using different architectures, like RPC/SOAP and REST, need to exchange information. The proposed solution was a metamodel, that abstracts architecture details and focuses only on the service elements.

We have discussed in [4] the problem of interoperability between syntactic and semantic description of RESTful Semantic Web services against the various existing languages. We have presented a model-driven approach, specifically on the creation of a metamodel and transformation rules in order to generate automatically the required documents that make the description of such services independently of the chosen language.

We have presented a metamodel named as RESTful Semantic Web Service (WSSR), which abstracts information present in the RESTful Web services and, to exemplify, we defined transformation rules that generated the syntactic description in WADL and the semantic description in Ontology Web Language for Services (OWL-S).

This work presented a solution of interoperability between syntactic and semantic description. However, it was not addressed the problem of interoperability also present in the Application tier of RESTful Web services, due to the fact of the wide variety of ways to implement these applications.

RPC/SOAP and RESTful Web services implemented in different frameworks and by different languages and formats are not interoperable and have many restrictions in their designs.

Some efforts have been made to design interoperable Web Services and model driven approaches have been applied as a solution to this problem. Some works can be found in the literature applying MDE-based techniques for developing RPC/SOAP Web Services. By the separation of concerns and by the development on different abstraction levels, Web Services metamodels can represent different tiers independent of technologies and programming languages then mapped to a target platform.

In this paper, we present an approach for the model driven development of the RESTful Web Service Application tier.

Figure 2 shows the architecture of our approach. The WSSR metamodel can be mapped to a semantic description language (A), to a syntactic description language (B) and to an implementation language (C). The (A) and (B) mappings were presented in [4] and in this paper, we discuss (C).

Through the solution proposed in this paper, it is possible to define the implementation logic of a RESTful Web Service independent of programming language, then by mapping rules and transformation process target different platforms without rewriting of the service implementation logic.

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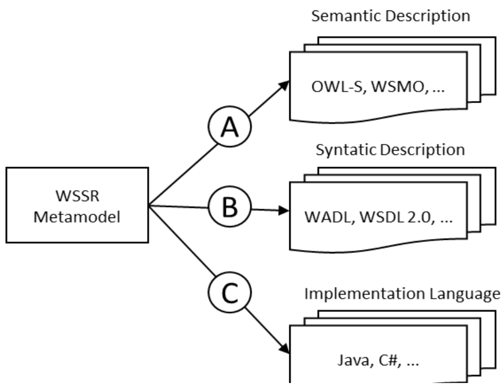

Figure 2. Possible transformations from WSSR metamodel

We provide the WSSR metamodel, which is responsible for defining the service business logic in the Application tier, abstracting platform (technologies, programming languages, etc) details. Then, a specific technology metamodel must be defined as the target platform chosen by the programmer to implement the service. Further, the identification of correspondences, named as mapping operation, will be defined between both metamodels. The mapping operation provides the correspondences needed to describe the transformation rules. The transformation rules, written in a transformation language, define which elements from a source metamodel will be transformed in which elements of the target metamodel. The transformation rules are applied in the model level, i.e., in instances of services. A source model must be conforms to the WSSR and the target model will be generated by the transformation engine, and it conforms to its target metamodel.

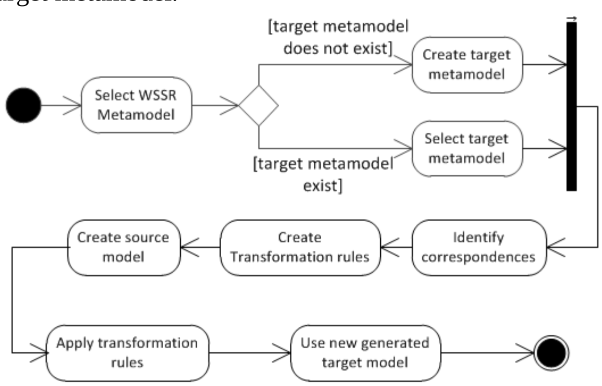

Figure 3. UML Activity Diagram of the Approach

Figure 4 illustrates, using a UML class diagram, the identification of semantic correspondences (mapping) of a fragment of our WSSR metamodel with the JAX-RS metamodel. The JAX-RS [16] specification was chosen because it is the standard specification for developing RESTful Web services in Java. This mapping allows the creation of the new transformation rules that will result in the implementation of the service in a fast and automated way.

In this figure, it is possible to see on the left side the WSSR metamodel with its main classes. The classes destined to do the semantic description were suppressed, because it is not the focus of this paper. It is important to note that the same WSSR metamodel – without any inclusion – can also be mapped to any language that implements RESTful Web services. The main class of the metamodel, named as *RESTService*, represents a RESTful service and contains the attributes necessary for the identification of such services, like URI, name, description, etc. The RESTful service has resources (*Resource class*), that are source of representations (*Representation class*). The resources are accessed by their methods (*Method class*) though HTTP methods (*HTTPMethod attribute*), which can have the values predefined in the *HTTPMethods* enumeration. The RESTful service is based on the paradigm of HTTP request and response, here represented by the *Request class and Response class*. The responses may be presented as representation format, and the requests are accessed through parameters (*Parameter class)*. These parameters may previously have established values, which are the options (*Option class*).

On the right side, can be seen the JAX-RS metamodel, which is an abstraction of the JAX-RS specification. This specification is present in the JavaEE and it is responsible for implementing RESTful Web services. The JAX-RS uses some Java language elements, such as packages, classes, methods and parameters, respectively represented by the *JPackage*, *JClass*, *JMethod* and *JParameter* classes. The *GET*, *POST*, *PUT* and *DELETE* classes are specializations of the *JMethod* class and inform which HTTP methods are related. The *JClass*, *JMethod* and *JParameter* classes are associated with the *JValue* class to combine Java annotations with themselves.

Between the two metamodels, each arrow identified by a circle represents the identification of correspondence between elements present in both WSSR and JAX-RS metamodel. The *R2C* arrow means that each *Resource class* corresponds to a *JClass class*. The *P2P* arrow means that each *Parameter class* corresponds to a *JParameter class*. Each *Method class* corresponds to a distinct class in the JAX-RS metamodel, depending on the *HTTPMethod* associated with it. If the *HTTPMethod* is GET, so the correspondence is between *Method class* and *GET class*, corresponding to the element M2Ge. Analogously, when the *HTTPMethod* is POST, PUT and DELETE, the *Method class* will be respectively associated with *POST*, *PUT* and *DELETE* classes, corresponding to elements M2Po, M2Pu and M2De.

This paper aims to generate an automated source-code generator related to the implementation of a RESTful Web service in any programming language that implements such services. To accomplish this, the WSSR metamodel must be mapped, by the correspondence identification, to the chosen programming language. Therefore, we chose the Java language with JAX-RS specification to exemplify the implementation.

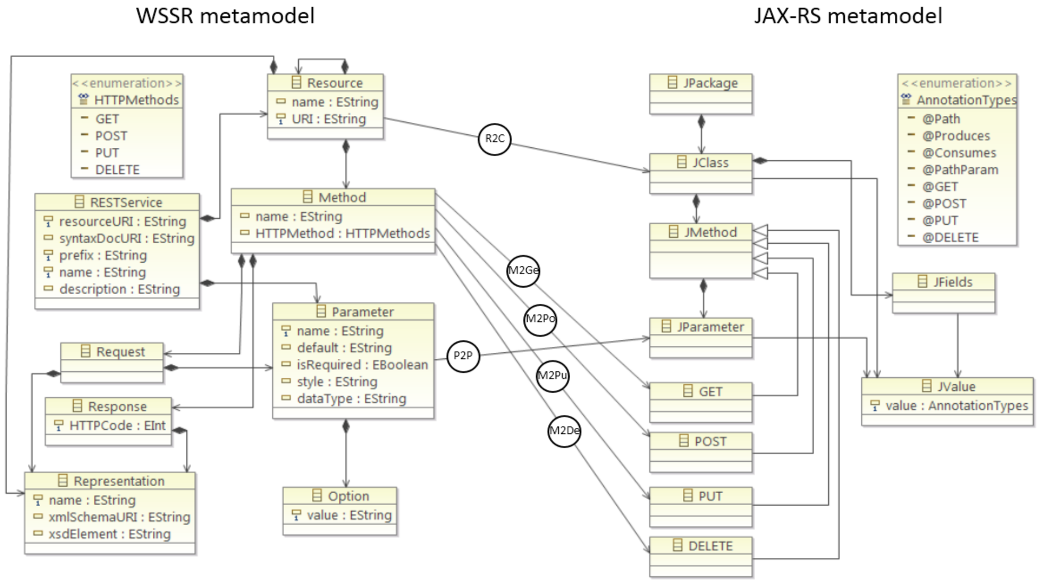

Figure 4. Correspondences between WSSR and JAX-RS metamodel elements

Figure 5 shows the transformation rules that were created using the elements that have the correspondences identified. The transformation rules where written using the ATL language and will transform the source model (conforms to the WSSR metamodel) to a target model (conforms to the JAX-RS metamodel). The target model will be the source code, i.e., the implementation of the RESTful Web service.

As can be seen, between lines 1 and 6, the code snippet refers to the creation of the resources. The code does an iteration over each method of the resource, calling a helper that corresponds to the creation of the methods between the lines 8 and 60. In this code snippet, also can be seen in lines 10, 30, 38 and 51 a condition that, according to the *HTTPMethod* associated with the method in the source model, will create a Java method with its corresponded annotation, such as the path and parameter that belong to this method.

```
01 helper context METAMODEL!Resource
02 def: toString(): String=
03   '@Path("/' + self.name + '")   ' +
04   'public class ' + self.name + 'Resource {   ' +
05   self.methods->iterate(i; acc: String='' | acc + i.Methods(self)) +
06 '}';
07
08 helper context METAMODEL!Method
09 def :Methods(resource: METAMODEL!Resource) : String =
10   if self.HTTPMethod.toString()='GET' then
11     '@GET' +
12     '@Produces("text/xml")' +
13     'public ArrayList<'+resource.name+'> get'+ resource.name +
14     'List('+self.methodRequests->iterate(i; acc : String = '' | acc +
15     i.requestQueryParameters->iterate(j; acc : String = '' | acc +
16     '@QueryParam("' + j.name + '") ' +
17     thisModule.convertDataType(j.dataType) +
18     resource.name + j.name + ', ')) + ') {  } ' +
19     '@Path("{' + thisModule.defaultParameter(self, resource) +
20     ' //return code }")' +
21     '@GET' +
22     '@Produces("text/xml")' +
23     'public ' + resource.name + ' get' + resource.name + '(' +
24     '@ParamPath("{' +
25     thisModule.defaultParameter(self, resource) + '}") ' +
26     thisModule.defaultDataType(self) +
27     thisModule.defaultParameter(self, resource) + ')
28   { //return code }'  else '' endif +
29
30   if self.HTTPMethod.toString()='POST' then
31     '@POST' + '   ' +
32     '@Consumes("text/xml")' +
33     '@Produces("text/xml")' +
34     'public ' + resource.name + ' post' + resource.name +
35     '(' + resource.name + ' ' + resource.name + ') ' +
36   '{ //return code }' else '' endif +
37
38   if self.HTTPMethod.toString()='PUT' then
39     '@Path("{'+thisModule.defaultParameter(self,resource)+'}")'+
40     '@PUT   ' +
41     '@Consumes("text/xml")' +
42     '@Produces("text/xml")' +
43     'public ' + resource.name + ' put' + resource.name + '(' +
44     '@ParamPath("{' +
45     thisModule.defaultParameter(self, resource) + '}") ' +
46     thisModule.defaultDataType(self) +
47     thisModule.defaultParameter(self, resource) + ', ' +
48     resource.name + ' ' + resource.name + ') ' +
49   '{ //return code }' else '' endif +
50
51   if self.HTTPMethod.toString()='DELETE' then
52     '@Path("{'+thisModule.defaultParameter(self,resource)+'}")'+
53     '@DELETE' + '   ' +
54     'public void del' + resource.name + '(' +
55     '@ParamPath("{' +
56     thisModule.defaultParameter(self, resource) + '}") ' +
57     thisModule.defaultDataType(self) + '
58     thisModule.defaultParameter(self, resource) + ', '
59     resource.name + ' ' + resource.name + ') '  +
60   '{ //return code }' else '' endif;
```

Figure 5. ATL snippet code of WSSR to JAX-RS transformation

By the specification of the correspondences between the source and target languages, the same code can be mapped generating different RESTful Web services implementations.

IV. CASE STUDY

As a case study, this paper proposes the implementation of a RESTful Web service that performs a simple product purchase order. The service must enable the order to be created, read, updated, and deleted, according to the concepts of the REST architecture described in the technological context of this paper.

To apply a model-driven approach, aiming to the implementation of the service, a model must be created conforms to the WSSR metamodel, previously presented. The model of the purchase order can be seen as an instance of the metamodel WSSR.

The model is depicted in Figure 6. Note that the model does not make use of all elements contained in the WSSR metamodel. Elements related to server Response (*Response class*) and the representation of the resource (*Representation class*), referred to the HTTP return code on the operations and the payload corresponding to representation sent and received from the server are not covered in this paper.

Thus, between lines 2 and 26 can be seen a RESTful Web service named *Order Service*, which have a resource named as *order*. Between lines 3 and 12 is the *GET method* and parameters that compose an application (code, product and quantity). The methods are responsible to perform operations on this resource. The *POST method* (line 13) refers to an operation of creating a new item. Between the Lines 14 and 19 can be found the *POST method*, which accepts the code parameter related to the order code that will be updated. Finally, between the lines 20 and 26 is depicted the *DELETE method*, which excludes the item passed by parameter.

```
01 <wssr:RESTService  name="Order Service"
02  <resources name="order" URI="order">
03   <methods name="Order-GET">
04      <methodRequests>
05         <requestQueryParameters name="code" default="true"
06         isRequired="true" style="query" dataType="xsd:int"/>
07         <requestQueryParameters name="product"
08         isRequired="true" style="query" dataType="xsd:string"/>
09         <requestQueryParameters name="quantity"
10         isRequired="true" style="query" dataType="xsd:int"/>
11    </methodRequests>
12   </methods>
13   <methods name="Order-POST" HTTPMethod="POST"/>
14   <methods name="Order-PUT"  HTTPMethod="PUT">
15     <methodRequests>
16      <requestQueryParameters name="code" default="true"
17      isRequired="true" style="query" dataType="xsd:int"/>
18    </methodRequests>
19   </methods>
20   <methods name="Order-DELETE" HTTPMethod="DELETE">
21     <methodRequests>
22        <requestQueryParameters name="code" default="true"
23        isRequired="true" style="query" dataType="xsd:int"/>
24     </methodRequests>
25   </methods>
26  </resources>
```

Figure 6. A sample model conforms WSSR metamodel

The execution of the transformation rules, created in the previous chapter, results in a source code written in Java and using the JAX-RS specification, as shown in Figure 7.

The figure shows the *orderResource* class containing methods and parameters appropriately associated with the Java annotations that will transform these elements in resources, methods and parameters of a RESTful Web service.

```
@Path("/order")
public class orderResource {
 @GET
 @Produces("text/xml")
 public ArrayList<order> getOrderList(
    @QueryParam("code") int orderCode,
    @QueryParam("product") String orderProduct,
    @QueryParam("quantity") int orderQuantity)
 { //return code }

 @Path("{order_code}")
 @GET
 @Produces("text/xml")
 public order getorder(
    @ParamPath("{order_code}") int orderCode)
 { //return code }

 @POST
 @Consumes("text/xml")
 @Produces("text/xml")
 public order postOrder(Order order)
 { //return code }

 @Path("{order_code}")
 @PUT
 @Consumes("text/xml")
 @Produces("text/xml")
 public order putOrder(
 @ParamPath("{order_code}") int orderCode, Order order)
 { //return code }

 @Path("{order_code}")
 @DELETE
 public void delOrder(
 @ParamPath("{order_code}") int orderCode, Order order)
 { //return code }
}
```

Figure 7. ATL snippet code of WSSR to JAX-RS transformation

Figure 8 shows a use case example developed through our approach, which is implemented in the Eclipse EMF. EMF has been the most used framework for the development of model driven approaches. It provides plugins for defining models, transformation language APIs, transformation engines and different modeling languages. In part (A) of the figure, we present the implementation of the source model, which conforms to the WSSR metamodel, previously defined in the EMF. The transformation rules are defined in the ATL language, as shown in part (B). By the interpretation of the transformation rules, the ATL transformation engine will generate, in a semi-automatic way, the target code of the RESTful Web Service in the Java language, as shown in part (C). The parts (A), (B) and (C) represent the implementation of the source code (fragment) depicted in the Figures 5, 6 and 7. The same source model can generate different services in different languages providing the interoperability proposed.

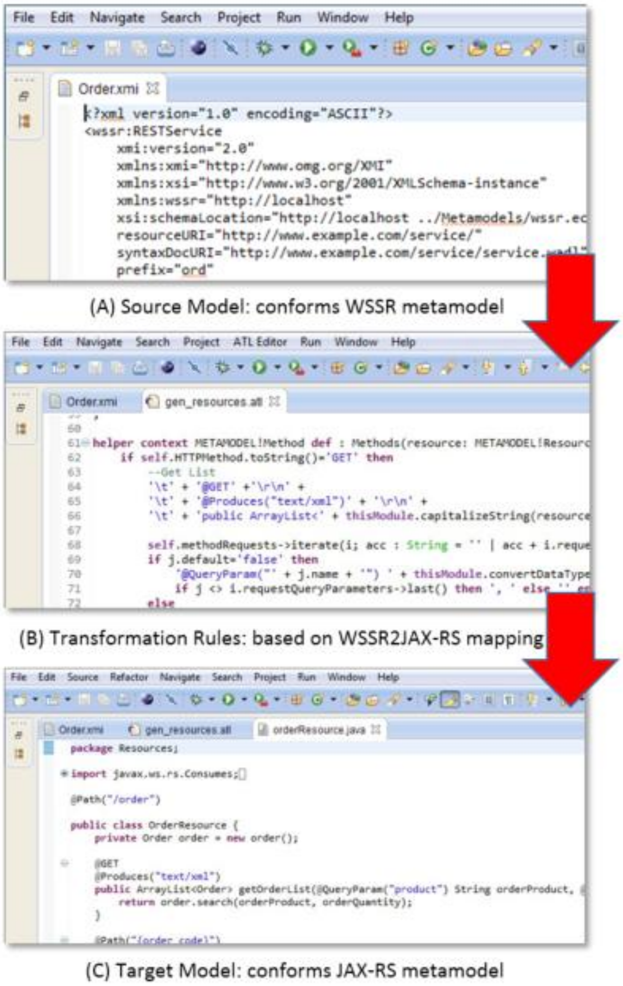

Figure 8. Development of the approach in EMF/Eclipse environment

V. CONCLUSION AND FUTURE WORK

The lack of interoperability between RESTful Web services have been discussed at the architectural level, syntactic description and semantic description. In this paper, we have discussed this issue at the implementation level. We have proposed a model-driven approach to solve the lack of interoperability of the Application tier of the REST architecture. We have presented a metamodel, mapping specifications and the transformation rules targeting the Java language, but the same approach can target any language that implements RESTful Web services. Comparing to the prior research, this approach provides some benefits beyond interoperability such as agile development, standardization, reuse and focus on the business rules. As future work, the WSSR metamodel may be mapped to others languages or specifications that are also used to implement RESTful Web services and generate the correspondent source code.

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