Ontology for IP Telephony Networks

Ilija Basicevic, Miroslav Popovic, and Nenad Cetic
Department of Computer Engineering
Faculty of Technical Sciences
Novi Sad, Serbia
Email: ilibas@uns.ac.rs, miroslav.popovic@rt-rk.com, nenad.cetic@rt-rk.com

***Abstract*—Ontologies have already been used in computer science research in different fields. In this paper, we apply the concept to the area of IP telephony. The paper presents ontology for development of software for IP telephony networks. It is based on the analysis of SIP and H.323 protocols, as dominant signaling protocols in VoIP telephony of today. The ontology we developed covers both static aspects (structure of the network) and dynamic aspects (most importantly telephony sessions, but also other types of associations). The structure of the telephony network is modeled with classes that represent different types of nodes in the network: end user terminals and various types of infrastructure nodes. The dynamic aspects of network operation are modeled with a set of classes, the most important being those that represent different associations that are established and terminated in the network. The realized ontology can be used in development of frameworks for telephony applications, and for specification of common data format used by cooperating telephony applications.**

Keywords-Ontologies; Network communications; Internet Applications; VoIP; IP telephony; SIP; H.323

I. Introduction

Ontologies have already entered and claimed an important place in several fields of computing science. In this article, we present application of ontology in the field of IP telephony. The ontology is focused primarily on signaling aspect, but covers all aspects of operation of IP telephony networks. Computer telephony is a result of merging of computing and telephony domain. Traditionally the two domains were separated, and telephony applications restricted to circuit switched networks, e.g. Public Switched Telephone Network (PSTN). With the intensive development of Internet in the last three decades, it became logical that IP network (although a packet switched network) could be used for personal communications with acceptable quality, which gave birth to Voice over IP (VoIP). Today VoIP is used in great extent - for example majority of new installed Private Branch Exchange (PBX) lines are VoIP. In most of the published papers dealing with application of ontologies in telephony domain, only segments or certain details of used ontologies are presented. This paper provides a systematic overview of the ontology we developed. The overview begins with the development process. The method that has been applied in development of the ontology is described. Next, two important class hierarchies that embody structure and dynamics of IP telephone networks are presented, visually and in text. Also, the paper tries to position the developed ontology in the context of existing work in this area.

Section II describes shortly the method applied in development of the ontology. In Section III are presented classes dealing with the structure of the telephony network, which is a very important static aspect of the network functionality. The prevailing formats of protocol messages in IP telephony systems are shortly described in section IV. Section V presents classes that are responsible for dynamic aspect of the operation of telephone network. In the section VI are shortly presented some published papers on the use of ontologies in this application domain. Section VII contains an analysis on possible applications of the ontology, including the one already realized. Section VIII contains concluding remarks.

II. The development of IP telephony ontology

The ontology that is presented in this paper has been developed using a combination of deductive and inductive approaches. In the conception phase of development, the deductive approach has been followed. As in other fields, published standards can be used as one of the starting points in definition of ontologies. Number of telephony standards is quite large. Each of many telephony standards in a way establishes its own taxonomy and usually defines most important terms. Thus the existing standards: most notably Session Initiation Protocol (SIP) [1], and in less extent H.323 [2] influenced very much the conception and elaboration phases in the development of the ontology. The importance we gave to SIP is due to its dominant place in the market.

The analysis of standards and telephony applications resulted in an informal description of ontology, and has been an input for the elaboration phase. The result of elaboration phase has been expressed in Unified Modeling Language (UML) and Java. The ontology has been the basis for the development of framework [3]. By building the framework we tested and further developed the ontology (see Fig. 1). In this manner we combined the deductive and inductive approach. This approach is similar to Forward-Lockstep Build-Test model of Helix-Spindle [4].

Copyright (c) IARIA, 2013. ISBN: 978-1-61208-262-2

III. Structure of telephony network

Structure of telephony network is in general case a graph consisting of nodes and links. Identification property of any network node is its network address. There are several types of nodes. The most important classification is into end points, directly accessed by end users who use the services of network, and infrastructure points, "invisible" to end-users. For that reason class NetPoint has the following subclasses: EndPoint, RoutingPoint, LocServPoint and ConferencePoint (see Fig. 2). ConferencePoint presents infrastructure nodes that support conferencing feature (conference servers and media mixers). The majority of infrastructure nodes in a typical network are responsible for routing. Those are modeled by RoutingPoint. Examples of RoutingPoint are SIP [1] proxy and H.323 [2] gatekeeper. Essential property of a routing point is the administrative policy database and a routing table. Routing relies on address mapping which is the responsibility of LocServPoint (location service or registrar in SIP terminology). Network is administratively separated into segments called domains in SIP and zones in H.323. Usually, there is one location service and one routing point per telephony domain. Location service keeps mappings for endpoints in that domain. Essential property of a location service is the table of address mappings that map telephony addresses to network layer addresses. Fig. 3 presents a typical case of IP telephony communication.

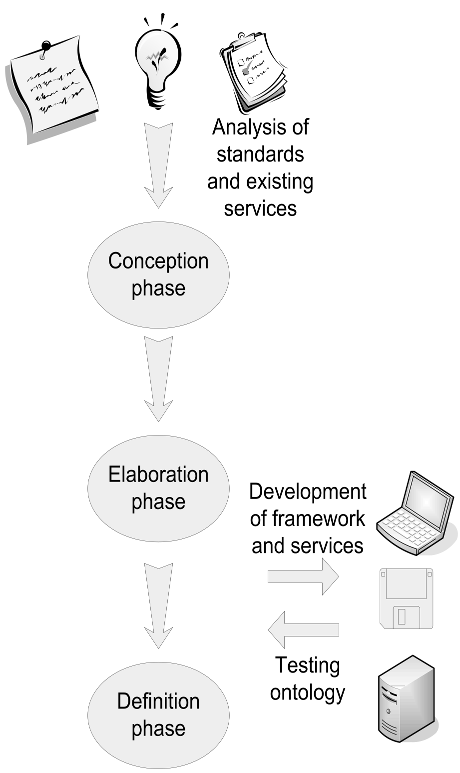

Figure 1. The development process

It is usually referred to as SIP trapezoid. There are two EndPoint instances (SIP user agents) shown in Fig. 3, one in each domain. In both domains there is RoutingPoint and LocServPoint - in the SIP case, the two (SIP proxy and SIP registrar) are often collocated.

An important class of telephony applications is the class of network side applications [5]- applications with logic residing at infrastructure nodes. For that reason ontology contains another class, subclassed from RoutingPoint RoutingPointExt. In SIP networks, network side applications are often built using back-to-back user agents which is a network element containing SIP client and server side user agent stack, controlled by one application. Essential property of endpoint is its configuration and the list of supported capabilities. The important property of EndPoint class is the active user currently logged in at the end point, modeled by the EndUser class. The characteristics of telephony services are such that there is only zero or one user active at any endpoint at any moment of time. The identification property of this class is the end user address, specified using the address scheme supported by telephony protocol of the network in a case. Examples are sip and sips address schemes, supported by SIP protocol or E.164 directory numbers [6], supported by H.323. While functions of endpoint are obvious - it is a user interface to end users that supports required functions (login/logout, authentication, call control, input/output of multimedia stream), the functions of infrastructure elements are more sophisticated and include:

- routing of messages
- enforcement of administrative policies
- bandwidth control
- location service
- authorization and authentication.

It can be seen that authentication is a support function required at all network nodes, both end points and routing points.

IV. Protocol Messages

All telephony operations are carried out as exchange of messages. Rigid property of ProtocolMessage is the message syntax. H.323 uses ASN.1 [7] while SIP uses UTF.8. H.225 [8] messages are based on Q.931 [9] syntax. The procedural view of telephony provides us with a list of most important operations including location service, capabilities query, session-related operations (establishment, modification, termination) and authentication. Each of listed operations assumes parameters. In case of session establishment and session modification, that parameter is proposed session description (in SIP case, it is presented in Session Description Protocol (SDP) [10] notation and in H.323 it is part of OpenLogicalChannel H.245 message [11]). In case of location query, this parameter is user address. But, modern protocols allow specification of many parameters of secondary importance - for instance, ringing tone that is used

Copyright (c) IARIA, 2013. ISBN: 978-1-61208-262-2

to signal new call can be specified in session establishment in SIP.

V. Dynamic characteristics of the network

A. Associations

Speaking about dynamic relationships that appear during the operation of telephony network, one of the most important classes in this segment is Association. This class is identified with addresses of participating sides and identification. We consider the association of two sides first. Association is a subclass of feature class (Feature). Feature is any function provided by the network to end users or a primitive function that goes unnoticed by end users, serving as a building block for complex functions (services) that end users invoke. One rigid property of Feature is OperationInterface. Subclasses of Association class are: Session – for telephony session, RegAssociation - for registration association, PubSubAssociation – for publish/subscribe association, and Conference (see Fig. 4). Session class models peer-to-peer telephony communication. The notion of telephony session implies existence of media (audio/video) stream that typically conveys speech or a video phone call, thus a rigid property of the Session class is SessionDescription which includes negotiated characteristics of audio/video stream and some other parameters. Registration association is the prerequisite of location service. In case of RegAssociation, rigid property is address mapping. Publish/subscribe associations are used for event publishing and are building blocks for different types of complex services. Rigid property of subscribe/publish association is the specific item of interest, in other words, the topic of information that is published. Conference is a feature that requires existence of infrastructure support (media mixers). Conference server is node which contains signaling sessions to participating nodes and media mixer contains media sessions to participating nodes. The two do not have to be collocated. Conference is a subclass of Association that presents association of $n > 1$ points. Rigid property of this class is the list of participants. In its early form, SIP dialog was about multimedia session only but, with the introduction of publish/subscribe mechanism, the notion of dialog has been extended. At this point, we do not recognize the need to incorporate such a composite class in ontology because all operations can be described using the existing association classes.

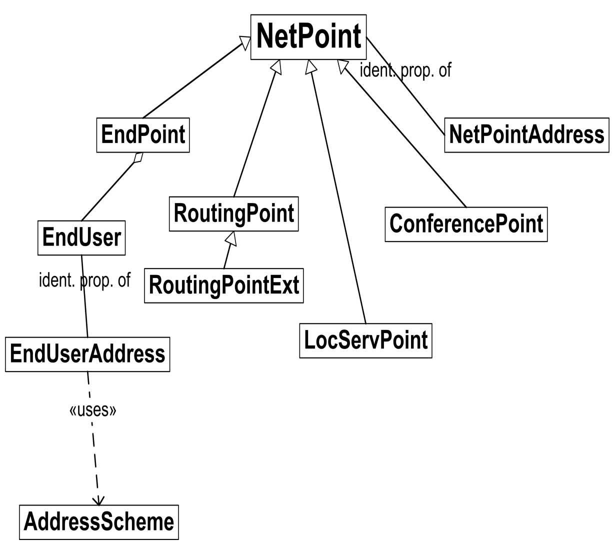

Figure 2. The NetPoint class and related classes

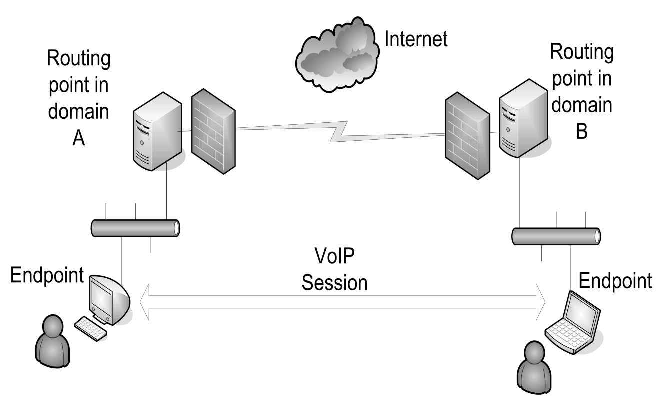

Figure 3. Typical IP Telephony communication path

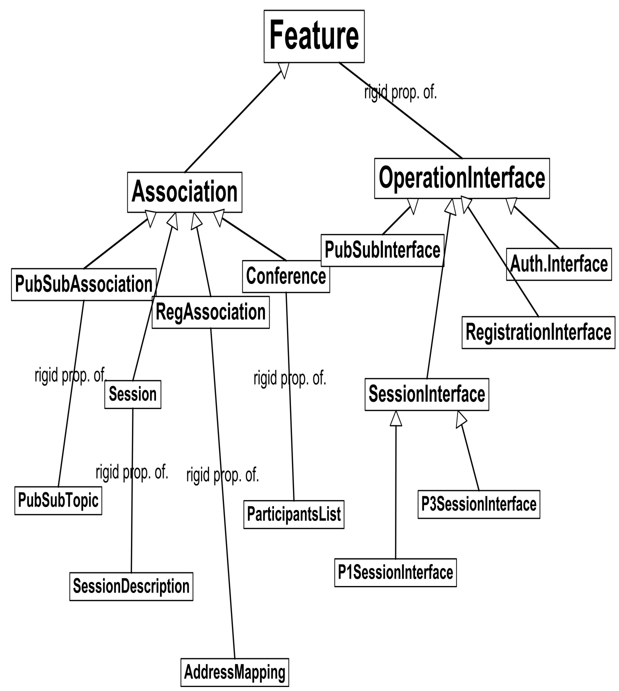

Figure 4. The Feature class and related classes

B. Interfaces

The telephony operations are provided by interfaces of our ontology. For each subclass of Association, there is

Copyright (c) IARIA, 2013. ISBN: 978-1-61208-262-2

analogous interface, providing telephony operations typical for that class. The following are subclasses of OperationInterface: SessionInterface, PubSubInterface, RegistrationInterface and AuthenticationInterface. There are two instances of SessionInterface: P1SessionInterface and P3SessionInterface, one being for the first party, and the other for the third party call control of the telephony session, which is used in complex telephony applications - call centers etc.

Several assumptions are used. Terminate is a two-step transaction and Establish is done in three steps, as it is the case in three way handshake that is used in SIP protocol. The session negotiation supports exchange of three messages both in SDP [10] based offer-answer model and in H.245 [11] Open Logical Channel sequence - in the case of bidirectional channel. (In the case of two unidirectional channels H.245 assumes exchange of four messages in session negotiation.) Also, session modification is in our ontology realized in three steps reusing some of the messages of Establish transaction. This is again in accordance with SIP protocol and its re-INVITE mechanism. On the other hand, a protocol has been proposed where instead of two-step and three-step transactions, each message represents a transaction by itself [12]. Essential methods of PubSubInterface are: Subscribe, Accept and Notify. Subscribe/Publish association finds its use in many complex services. For example, in SIP environment, it is used for dissemination of presence and registration information, for publication of the result of transfer operation, and for third party control. In SIP protocol, the session transfer operation is complex feature that uses basic service session and publish/subscribe association as building blocks. For de-registration, we use SIP convention: register message with specific values of parameters (expiration time set to 0). For that reason, there is no explicit Deregister method in the interface.

C. Authentication

Session establishment as well as registration may require authentication. In our ontology, we assume stateless challenge-based solution as in SIP protocol. Authentication interface contains the following methods: AuthenticationRequired, AuthenticationRequest and AuthenticationResponse. However, the last two are not transferred as separate messages but as message parameters in messages like Establish or Register. For example, endpoint sends Establish with no user credentials. Routing point checks the administrative policy rules, determines that authentication is required and sends AuthenticationRequired back with the challenge included. Endpoint sends Establish again, this time with user credentials and computed response to the received challenge. This solution implies existence of some kind of public key infrastructure.

VI. Related Work

A discussion on the use of ontologies in telecom domain can be found in [21]. The work presented by Geneiatakis et al. [14] is an example of the use of ontology for VoIP security. It is the case of target-centric intrusion detection. Data model based on this ontology would provide a framework for cooperation of elements of distributed IDS system protecting the domain of telephony network. Also, cooperation of different IDS systems protecting different domains of telephony network can be based on the model. The ontology presented in this paper can also be used in security and, if extended with classes related to specific details of telephony protocols, it would cover wider range of detected attacks since it comprises a holistic view of IP telephony network.

Campbell et al. [15] presents specialization of APPEL policy language for call control in the scope of the Advanced Call Control Enhancing Network Technologies (ACCENT) project. APPEL encapsulates generic aspects of a policy language, but some specialized aspects related to call control are included too. In [15], parts of call control ontology used in ACCENT project, related to trigger, condition parameter and action classes are presented. Campbell et al. [15] presents segment of the call control ontology at the level of details, which complements the high level overview given in this paper. The Call Processing Language (CPL) scripts are used in many SIP based services. Devlic [16] presents context ontology for enhancement of CPL possibilities and the architecture of context aware VoIP system. Another system that uses context information and is developed using ontology is presented in [22]. The context related classes are not covered in the ontology presented in this paper. It is a potential topic for further work on the ontology.

Tetlow et al. [17] presents system architecture of an ontology based application server. Sources that provide input to the system are various metadata files (including application server configuration files). This information is parsed into semantic metadata - metadata in terms of the applied ontology. The semantic metadata and the ontology are inputs to the inference engine that is embedded in the application server. The engine is used either by core services or by tools which provide GUI interface. Akhmanov et al. [18] presents an ontology based system for prevention of Spam over Internet Telephony (SPIT) attacks in SIP networks. Anti-SPIT security policy is expressed as a set of rules, coded in SWRL language. Each rule contains two segments: conditions and actions. Applicable actions are: Allow, Block, Check Further. The rules are incorporated into the ontology model. An example of a rule that checks PRIORITY and FROM headers in a SIP message is given, but the details of the ontology are not.

Copyright (c) IARIA, 2013. ISBN: 978-1-61208-262-2

VII. Applications of the Ontology

Presented ontology is used in the development of telephony framework [3]. The framework supports the following features:

- first-party call control operations,
- third-party call control operations,
- instant messaging,
- presence services,
- monitoring of the state of logged-in users and active sessions,
- network side applications.

Besides the classes described in this paper, the framework includes several other important classes. An object of SignalingDevice class interfaces the underlying session layer protocol. P3Server and P3Session classes implement third party call control. AppLEObserver is a parent of classes that provide support for monitoring, in the framework are used EndUserObserver and SessionObserver classes. Monitoring is based on publish-subscribe relationship which maps directly to SIP event notification mechanism. An object of FeatureMng class is responsible for feature interaction control (in collaboration with the classes that implement active features, and inherit the generic Feature class).

Certain design decisions regarding feature management are further explained in [13]. The session transfer and session redirection features are used as examples. Also in the paper are described two simple taxonomies of call control features. The first one groups features into primitive, derivative and composite. Basic call with interface for first party call control is a primitive feature. An example of a composite feature is conference. The relationship between composite and primitive feature is similar to aggregation ('has' relationship in Unified Modeling Language). In call processing derivative features can have read access to basic call data, but no feature has full (read and write) access to other feature's data. The second taxonomy distinguishes symmetric and asymmetric features. The former are implemented as peer-to-peer, and the later as client-server modules. Basic call is a symmetric feature, while session transfer and redirect are asymmetric features.

The ontology could be used in a framework as Multimedia Ontology-Driven Architecture (MODA) [18]. It would complement the ontologies used in [18] (ITU multimedia services ontology, IETF protocols ontology, software implementations ontology etc.). In network security, this ontology could be used for building data models, as in [14]. In that case the ontology should be extended with the work in [20]. In SIP environment, complex services are built as software applications residing on top of a SIP stack. Such ontology could be used to establish common data format for cooperation of different VoIP applications.

VIII. Conclusion and Future Work

The paper systematically presents important classes of a new ontology for IP telephony networks - by giving an overview of important aspects of network functionality and corresponding classes, while in most of the published papers only segments of used ontologies are presented. The ontology outlined in this paper is built around several key classes that are roots of class hierarchies. Those classes embody the most important qualities of the telephony network. The most important static aspect is the structure of telephony network. The corresponding class hierarchy starts with the class NetPoint. The dynamics of the telephone network are best described with appearance, and duration of telephony associations. The corresponding class hierarchy starts with the class Association. The most obvious association is the telephony session, or call, but there are also other important associations, which are not so obvious. An example is the publish/subscribe association, which is a building block for many complex services. Class Feature represents a generic telephony service.

Application of this ontology is possible both in design of telephony software (as in [3], [13]), and in data formats used for interoperability of different telephony applications and/or security applications in telephony systems. The support for context-based applications and potential application of ontology for multimedia systems beyond telephony are elements for future work.

Acknowledgment

This work was partially supported by the Ministry of Education and Science of the Republic of Serbia under the project III 44009-2.

References

[1] J. Rosenberg, H. Schulzrinne, G. Camarillo, A. Johnston, J. Peterson, and R. Sparks, "SIP: Session Initiation Protocol", RFC 3261, Internet Engineering Task Force, 2002.

[2] ITU-T, "Visual Telephone Systems and Equipment for Local Area Networks Which Provide A Non-Guaranteed Quality Of Service", H.323, 1996.

[3] I. Basicevic, "Object-Oriented Framework for Development of Telephony Applications", International Conference on Digital Telecommunications, Colmar, France, 2009, pp.11–14.

[4] R. Kishore, H. Zhang, and R. Ramesh,, "A Helix-Spindle Model for Ontological Engineering", Communications of ACM, vol. 47, No. 2, 2004, pp. 69-75.

[5] I. Basicevic and M. Popovic, "Use of SIP in the Development of Telecom Services - A Case Study", The Journal of the Institute of Telecommunications Professionals, vol.2, Part 3, 2008, pp. 55-65.

[6] ITU-T, "The International Public Telecommunication Numbering plan", E.164, 2005.

[7] ITU-T, "Information technology - Abstract Syntax Notation One (ASN.1): Specification of basic notation", ITU-T Rec. X.680 —ISO/IEC 8824-1, 2002.

[8] ITU-T, "Call Signaling Protocols and Media Stream Packetization for Packet-Based Multimedia Communication Systems", H.225.0, 1998.

Copyright (c) IARIA, 2013.

[9] ITU-T, “ISDN User-Network Interface Layer 3 Specification for Basic Call Control”, Q.931, 1998.
[10] M. Handley, V. Jacobson, and C. Perkins, “SDP: Session Description Protocol”, RFC 4566, Internet Engineering Task Force, 2006.
[11] ITU-T, “Control Protocol for Multimedia Communication”, H.245, 1998.
[12] P. Zave and E. Cheung, “Compositional Control of IP Media”, International Conference On Emerging Networking Experiments And Technologies, 2006.
[13] I. Basicevic, “Modular Design of Call Control Layer in Telephony Software”, International Journal Of Computer Science Issues, vol. 8, Issue 1, 2011.
[14] D. Geneiatakis and C. Lambrinoudakis, “An Ontology Description for SIP Security Flaws”, Computer Communications, vol. 30, 6, 2007, pp. 1367-1374.
[15] G. A. Campbell and K. J. Turner, “Ontologies to support Call Control Policies”, Proc. 3rd. Advanced International Conference on Telecommunications, 2007, pp. 5.1–5.6.
[16] A. Devlic, “Extending CPL with context ontology”, IMAC Workshop, Espoo, Finland, 2006
[17] P. Tetlow, J. Pan, D. Oberle, E. Wallace, M. Uschold, and E. Kendall, “Ontology Driven Architectures and Potential Uses of the Semantic Web in Systems and Software Engineering”, World Wide Web Consortium (W3C) working draft, 2006.
[18] S. A. Akhmanov, A. S. Chirkin, K. N. Drabovich, A. I. Kovrigin, D. Stelios, and G. Dimitris, “An Ontology-Driven antiSPIT Architecture”, Lecture Notes of the Institute for Computer Sciences, Social Informatics and Telecommunications Engineering, vol. 26, 2010, pp. 189-198.
[19] E. Exposito, J. Gomez-Montalvo, and M. Lamolle, “Multimedia Ontology-Driven Architecture for Multimedia Systems”, Second International Conference on Information, Process, and Knowledge Management, eKNOW ’10, 2010, pp.7–12.
[20] VOIPSA, “VoIP Security and Privacy Threat Taxonomy”, Public Release 1.0, http://www.voipsa.org/Activities/taxonomy.php, accessed 05.2.2013.
[21] J. Frankowski, P. Rubach, and E. Szczekocka, “Collaborative Ontology Development in Real Telecom Environment”, 1st International Working Conference on Business Process and Services Computing, 2007, pp. 40-53.
[22] P. Gutheim, “An Ontology-Based Context Inference Service for Mobile Applications in Next-Generation Networks”, IEEE Communications Magazine, vol. 49, no. 1, 2011, pp. 60-66.

Copyright (c) IARIA, 2013. ISBN: 978-1-61208-262-2