Testing Triple Play Services Over Open Source IMS Solution for Various Radio Access Networks

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Abstract* — Nowadays, with the development of next generation networks, new standards for voice and video services have been introduced, and VoIP (*Voice over IP*) has become a very popular protocol. It leads to the convergence of networks which provide voice and data transfer services with enhanced and improved applications for customers. As a key element of the next generation networks, IMS (*IP Multimedia Subsystem*) architecture aims to offer multimedia services via fixed-mobile convergent solutions. In this paper, a work model offering triple play services via a UMTS network and an IMS subsystem is realized. This model is realized through an Open Source software solution for an IMS subsystem, and the success of the triple play service provided by this system is analyzed. Testing of access networks and of different codecs using an Open IMS Core system and a UMTS access network is carried out. The most successful codec is determined and proposed for use while offering triple play services. Further, an optimization of the UMTS radio-access network of a middle size operator is done by planning the optimal number of NodeB's for this software solution and the dimensioning and optimization of the same. The conclusion is that, from the standpoint of offering triple play services to a certain number of users, no significant extensions of the radio network capacity are needed. The resulting system for offering triple play services is functional and optimized for exploitation in an active network with high reliability.

Keywords – UMTS; IMS; Open IMS Core; EDGE; WiFi.

I. INTRODUCTION

Third generation mobile systems (3G) were introduced in accordance with the needs of customers for high speed data transfers, in order to enable a broad spectrum of Internet services. The UMTS (*Universal Mobile Telecommunications System*) presents one of the most frequently used standards for third generation mobile networks [7]. The combination of Internet and mobile networks has resulted in UMTS enabling triple play services, whose primary goal is the transition of all current and future services to IP: data, voice and video.

IMS (*IP Multimedia Subsystem*) represents a standard that defines core and service layer architecture of the fixed and the mobile telecommunication networks for a new generation [8]. The specification of its main elements, protocols and mechanisms is based on the cooperation of leading telecommunication standardization groups: 3GPP, ETSI/TISPAN and IETF. The main technological characteristic of IMS is its transparency towards different access networks. This is why IMS is the main initiator of the evolution of telecommunication networks, leading towards fixed-mobile convergence (FMC) solutions. Users can access the integrated user interface and are offered a unique broad spectrum of services using different devices and access technologies.

In [1], Magedanz, Vingarzan and Harjoc tested the Open IMS Core system [9] when a million users access the same system. However, an analysis or evaluation of the system while providing triple play services in a corresponding real environment of a medium size operator is not presented. Vingarzan and Weik presented in [2], the performance results of the Open IMS Core system in the case of transferring signalization traffic for voice transmission over various access networks. Testing of the video service and choosing appropriate codecs is not observed. In [3], Prokkola, Perala, Hanski and Piri presented the testing of HSDPA and WCDMA networks. The performances of the HSDPA network are not compared with other wireless radio access technologies. The main research objectives presented in this paper are:

- Realization of the IMS test platform based completely on open source technology.
- Realization and testing of triple play services over an IMS platform (voice call, video call and data transfer).
- Comparison of the performance of different access technologies (EDGE, WiFi, UMTS) through an IMS test platform by measuring appropriate parameters.
- Selection of the optimal codec in order to achieve the best quality for voice and video communication.
- Optimization of the UMTS radio access network in terms of predicting the number of users who will initially use the triple play services over UMTS radio access networks and the Open IMS Core system, and finally the optimization of the number of NodeB's needed to meet the users' needs.
- Considering the results published in [1], the design of a potential IMS network for a medium size operator, which would meet the needs of the initial number of users and whose network core would be an Open IMS Core system.

In the next section, the testing of triple play services over various radio access networks (EDGE/WiFi/UMTS) is described, with an analysis of the quality of the voice, video and data services. Thereafter, the optimization of mobile networks is described. The analysis is performed based on data from a medium sized telecom operator with a developed mobile network (GSM/EDGE/UMTS) and a fixed network (PSTN/ADSL), assuming that the number of users of the mobile network is 1.000.000, and 700.000 users for the fixed network. This is followed by the

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dimensioning and optimization of the Open IMS Core system.

II. TESTING OF THE IMS SYSTEM

The parameters observed for the testing of every access network are: packet loss, delay, jitter, MOS (*Mean Opinion Score*) and R factor. The jitter diagram and the value of the MOS factor, when testing the voice and video service, will be shown for each access network. All of these measurements will be done for the GSM FR (GSM Full Rate) codec. The presumption is that a unloaded Open IMS Core system is used. The Open IMS Core is installed on the PC with a 1024/128 Kbit/s access link, and the conversation participants are stationary.

Finally, these parameters will be tested for a UMTS access network using an iLBC (*Internet Low Bitrate Codec*) codec, and it will be determined if and what kind of progress was achieved by introducing this codec.

In the case of data and instant messaging transfer, it can generally be concluded that there were no problems in the realization of these services through any of the used radio access technologies.

A. Testing through the EDGE radio access network

The test results of the triple play services for the EDGE radio access will be listed below, with an analysis of the measured parameters.

Figure 1 shows the block diagram of the triple play testing scenario through the EDGE radio access network and the Open IMS Core system.

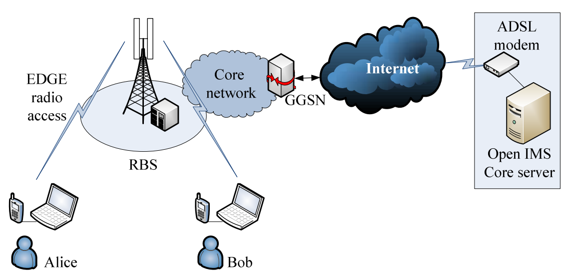

Figure 1. Testing of triple play services through the EDGE radio access network

The test results of the voice service through the EDGE radio access network and the Open IMS Core system can be interpreted as follows:

- The average value of the MOS parameter was 3.3, resulting in a very good quality of the voice communication.
- Variations in the packet delay were highly emphasized during the voice communication, as shown in Figure 2. This refers especially to the jitter increase in the interval between the 22nd and the 25th second of the voice session. Average jitter was 23.42 ms, and is marked by squares in the picture. According to [10], where the suggested maximum value for jitter in the VoIP network is 30 ms, it can be concluded that this network provides a good quality of the tested service.

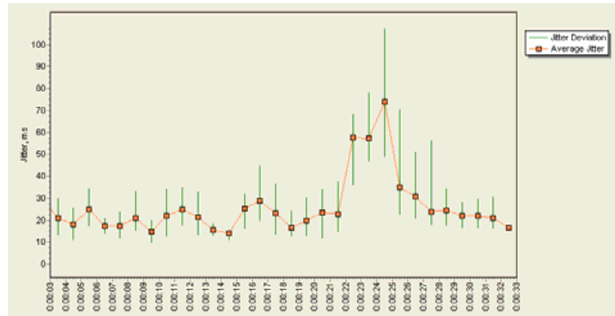

Figure 2. Measured jitter through the EDGE radio access network in the case of voice communication

While testing the voice and video service together, a large decrease in value of the MOS parameter during the session was noted. In the beginning of the communication, this parameter had the value of 3.7, then immediately decreased to 1 (which is basically an unusable call). It retains this value until the end of the call.

It can be seen from Figure 3 that the jitter had higher values during the first 9 seconds of the video session. Average jitter was 66 ms which is a very poor result for the tested type of service.

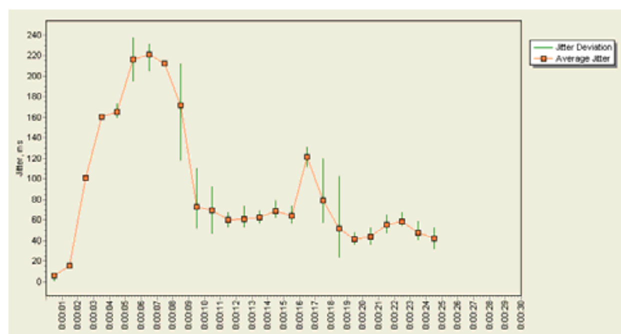

Figure 3. Measured jitter through the EDGE radio access network in the case of voice and video communication

From the measurements performed on the test system it can be summarized that video call services through an EDGE radio access network and an Open IMS Core system cannot be provided, since the introduction of a video call degrades the performances of the voice call to a level where it becomes impossible to use. This conclusion was to be expected given the practical transmission rate characteristic for EDGE technology.

B. Testing through the WiFi radio access network

The test results of the triple play services for a WiFi radio access network along with an analysis of the measured parameters will be presented below. Figure 4 shows the block diagram of the test scenario through a WiFi radio access network.

For the voice communication tested through the WiFi network, the MOS parameter had the value of 3.7, and was approximately stable during the entire period of the voice session, which is an excellent result for this type of codec.

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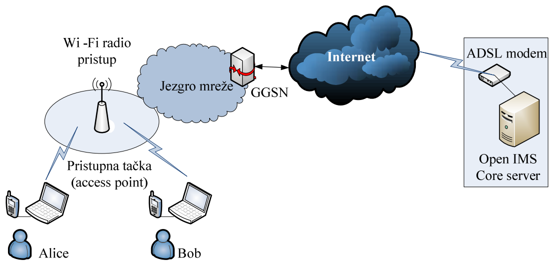

Figure 4. Testing of triple play services through a WiFi radio access network

Variations in packet delay were not highly expressed (as was the case for the EDGE access network) during the voice communication, as shown in Figure 5. Average jitter was 3.98 ms, which is a very good result for this type of service.

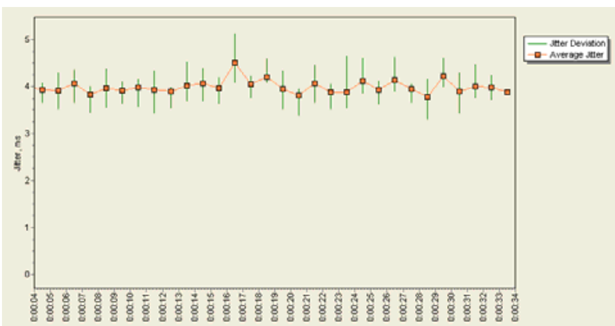

Figure 5. Measured jitter through the WiFi radio access network in the case of voice communication

While transferring voice and video communication through the WiFi network, the MOS parameter was 2.1, which means that the introduction of the video service significantly degraded the quality of the transmitted voice.

It can be concluded from Figure 6 that the jitter values during the call session were within acceptable limits. There was only one larger increase of these values in the 19th second of the call duration. Average jitter was 20.58 ms, which is an acceptable value for this type of service.

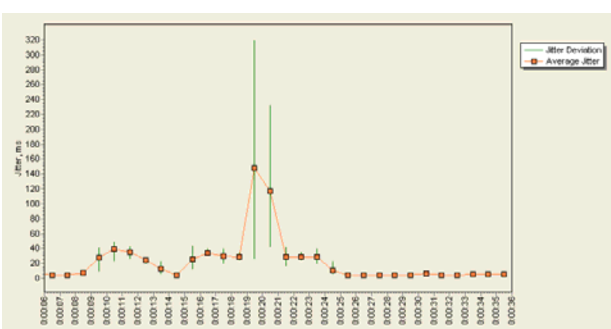

Figure 6. Measured jitter through the WiFi radio access network in the case of voice and video communication

It can be concluded that, when transmitting voice and video services over a WiFi access network and an Open IMS Core system, the transmission performance is significantly degraded when compared to the transmission of a voice service alone. The WiFi technology can be used for the transmission of video services, but a poorer quality can be expected for the enduser.

C. *Testing through a UMTS radio access network*

In this section, the test results of the triple play services for the UMTS radio access will be shown, with an analysis of the measured parameters.

Figure 7 shows a block diagram of the test scenario through the UMTS/HSxPA radio access network.

The test results of the voice service through the UMTS radio access network and Open IMS Core system can be interpreted as follows:

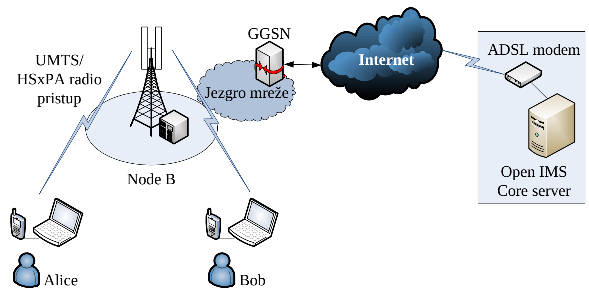

Figure 7. Testing of triple play services through UMTS radio access network

- MOS parameter of the voice service tested through the UMTS network was 3.7, which is, as we know, an excellent result. It is important to emphasize that this parameter was stable throughout the entire duration of the session.
- Variations in packet delay were not so emphasized during the voice communication, as is shown in Figure 8. Average jitter was 4.86 ms, which completely satisfies the requirements listed in [10].

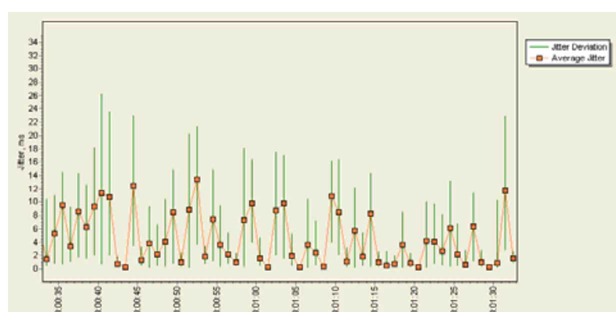

Figure 8. Measured jitter through the UMTS radio access network in the case of voice communication

The MOS parameter for the transfer of voice and video communication over the UMTS network was 1, which means that the introduction of the video signal notably degraded the quality of the voice transmitted; therefore, it cannot be used for those purposes.

From Figure 9 it can be concluded that jitter is emphasized at the beginning of the call, but later on during the communication, it decreases to a slightly lower level. Average jitter was 48.29 ms, which is an insufficient value for this type of service.

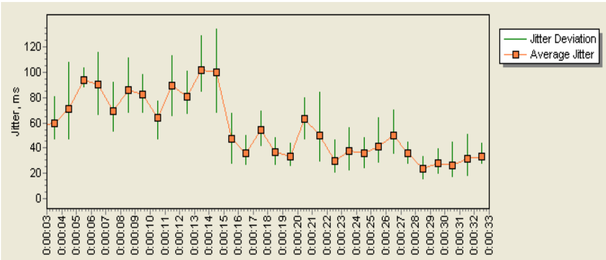

Figure 9. Measured jitter through the UMTS radio access network in the case of voice and video communication

It can be concluded that the transmission of a voice and video service through the UMTS access network and the Open IMS Core system significantly degraded the transmission performance, when compared to the transmission of voice service alone. Therefore, the UMTS technology cannot be used for the transmission of a voice and video service at the same time and under these conditions.

D. Testing through theUMTS radio access network – iLBC codec

In order to optimize and provide high quality triple play services, the analysis of parameters measured when using the iLBC codec in the network will be given below. This codec is primarily intended for the coding of speech in IP communications. The main advantage of this codec is an acceptable degradation of speech quality if it comes to packet loss or packet delay.

When testing the voice service using the iLBC codec through the UMTS access network and the Open IMS Core system, the results can be interpreted as follows:

- MOS parameter for the voice service tested through the UMTS network had an average value of 4.1, which is an excellent call quality. It also took stable values throughout the entire duration of the session.
- Variations in packet delay were not as emphasized during the voice communication, shown in Figure 10. Average jitter was 6.27 ms.

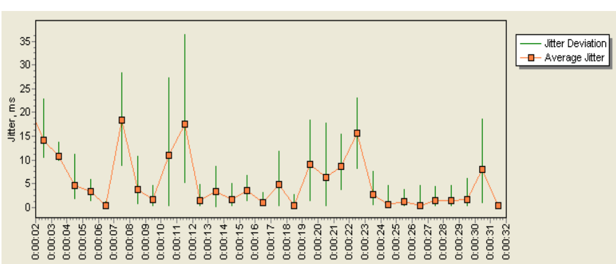

Figure 10. Measured jitter through the UMTS radio access network in the case of voice communication using iLBC codec

When testing the voice service together with the video signal through the UMTS access network and the Open IMS Core system, the results can be interpreted as follows:

- MOS parameter for voice and video communication through the UMTS access network had an average value of 3.9, and was stable throughout the entire session.
- Variations in packet delay were not as emphasized during the communication, except at the beginning of the call, as shown in Figure 11. Average jitter was 19.57 ms.

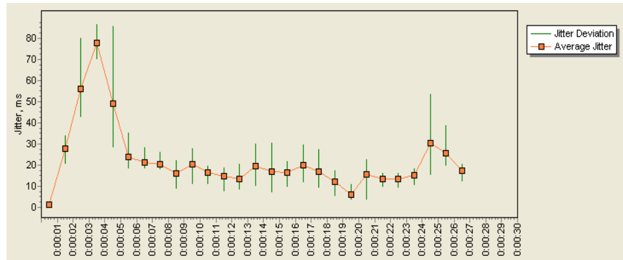

Figure 11. Measured jitter through the UMTS radio access network in the case of voice and video communication using iLBC codec

E. Conclusions

After obtaining the results of the transfer of the triple play service via the EDGE, UMTS and WiFi radio access networks, it can be concluded that the EDGE radio access network can be used for voice service, but the quality of the call is degraded completely when transferring the voice together with the video signal, and it cannot be used to offer these services to the end-users. As opposed to the EDGE network, the WiFi access network achieved an excellent performance for voice calls, however, for video calls there occurs a decrease in the quality of the call (MOS has the value 2), and it can be concluded that the WiFi technology, despite the significantly better performance, cannot be used for the transfer of triple play services over an Open IMS Core system, due to the relatively poor values of the MOS parameter. On the other hand, UMTS has also achieved an excellent performance for the voice call, while the video call has introduced a complete decrease in the quality of the service. It can also be concluded that UMTS is only to be used for voice and data services while it does not provide satisfactory parameters for a video service.

An improvement of the quality of voice and video communication through the Open IMS Core system and the UMTS radio access network using the iLBC codec is achieved in the final stage. When the iLBC codec is used to test the quality of triple play services of a voice communication, excellent performances of the quality of voice and video communication through the UMTS radio access network and the Open IMS Core system are achieved, which was to be expected taking into consideration the characteristics of this codec.

III. OPTIMIZATION OF THE RADIO ACCESS NETWORK

The fundamental factor for defining the number of users who will share network resources is the traffic produced by each one of them. The traffic per user depends on the call frequency and average call duration.

One approach to the optimization of the UMTS radio access network for triple play users will be described below. Conditions, input data, presumptions and methodology for an optimal number of UMTS network NodeB's will be listed, taken from an average telecom operator.

Input data and presumptions are listed as follows:

- The operator has 1.000.000 mobile users.
- It is assumed that 20% of the mobile users use the UMTS service.
- It is assumed that 30% of the UMTS users use the triple play service.
- It is assumed that 10% of the triple play users access the system during busy hour.
- It is assumed that the RAB traffic class is used for the triple play service.
- Ideal propagation conditions are assumed.

RAB (Radio Access Bearer) represents a logical relation between the core network and user equipment. It is used to enable a connection for the UMTS service across the UTRAN (*UMTS Terrestrial Radio Access Network*) network. The RAB connections are realized as Radio Bearer connections between the RNC (*Radio Network Controller*) and the core network.

Triple play service users are considered to use the RAB interactive class of traffic and a certain service subscription package with a certain amount of traffic included. It is important to mention that (in order to define the busy hour) for packet traffic, it is assumed that 10% of the daily traffic is used during busy hour and the remaining 90% of data traffic is used during the rest of the day.

The traffic measured in kB/busy hour is calculated by dividing the amount of traffic from the subscription package into 30 days, taking into consideration ten percent of the obtained average daily traffic.

Considering the above calculation, and after the analysis of the average HSPA user, for the 1GB data traffic included in the monthly subscription, the traffic during busy hour is 28633 kbit.

The relation between uplink and downlink traffic is 10%, meaning that an average user realizes 318,19 kB of downlink traffic and 31,81 kB of uplink traffic during busy hour.

The configuration of every base station/cell will be designed to support HSDPA up to 14,4 Mbps and EUL up to 1,4 Mbps, using only one frequency bearer, because of the theoretical restriction of the code tree resources (HSDPA requires 15 out of the 16 available codes). This can only be realized with the assumption that there is no other (R99) traffic (otherwise the use of two bearers is necessary).

Since R99 traffic is still expected, it is optimal to include dynamical code allocation, i.e. a minimum of 10 codes per cell are allocated to the HSDPA, and the remaining five codes will be available when the R99 traffic is low.

If 10 codes per cell are allocated to the HSDPA, it can be concluded that the maximum traffic for that cell will be 7,2 Mb/s for downlink and 1,4 Mb/s for uplink traffic.

Considering the above assumption, that 10% of the triple play users will be accessing the network during busy hour, a total number of 600 active triple play users is obtained for busy hour.

For one cell it can be calculated:

- 7.2 Mbps/318,19 kB = 3 user/cell.

Therefore, it can be concluded that it is necessary to plan 200 cells/NodeB in the UTRAN network, so as to serve the total number of triple play users accessing the system during busy hour.

IV. DIMENSIONIG AND OPTIMISATION OF THE OPEN IMS CORE SYSTEM

In order to carry out the dimensioning and optimization of the Open IMS Core system, besides the number of UMTS users of the triple play service who will access the system during busy hour, it is necessary to define the number of fixed network users who will use the triple play service.

Input data and assumptions for this case are listed as follows:

- The operator has 700.000 users.
- It is assumed that 30% of the users use Internet service.
- It is assumed that 20% of the Internet users use triple play service.
- It is assumed that 10% of the triple play users access the network simultaneously during busy hour.

Using these assumptions a total number of 42.000 users who access the Open IMS Core system is obtained. Taking into account the assumption that 10% of the users use the triple play service at the same time, a total number of 4.200 users is obtained.

Together with the UMTS users, a number of 10.200 users accessing the Open IMS Core system at the same time is obtained.

Taking into account that the computer used for testing the Open IMS Core system has a poor performance, referring to the processor speed, the throughput of the network interface, memory, etc., this machine could not be used for an IMS core network of a real operator.

Such a workstation can, according to [1], simultaneously serve 10000 calls. This would satisfy the requirements listed above.

However, more users are expected to use the triple play service in the future, which will result in the need for proper dimensioning and optimization of the Open IMS Core system.

Scalability and redundancy are the next important parameters which should be observed. In order to achieve a scalable system, expandable in accordance with the future growth of the number of triple play users, and to improve the performances of the system for future users, two parallel Open IMS Core systems can be installed.

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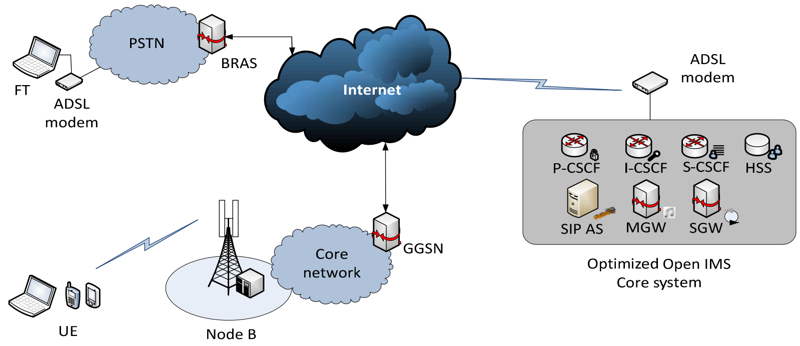

Figure 12. Optimized Open IMS Core system

This is the horizontal scalability principle, where a capacity-expandable platform is achieved by the multiplication of servers (CSCF's and FHoSS database).

From the viewpoint of system redundancy, it is possible to provide database redundancy, computer processor redundancy and geographical redundancy.

V. CHALLENGES FOR MOBILE OPERATORS – OTT (OVER-THE-TOP)

We are witnessing a large increase in mobile data traffic and a significant utilization of the available bandwidth of networks, currently happening worldwide. This growth receives great support from the diverse offerings of IP-enabled intelligent devices. To adjust and to earn additional revenue, mobile operators must optimize their networks. This optimization refers to the installation of new multimedia platforms in the core network (IP RAN aggregate node, IP core network). The new platforms would, in order to identify the used OTT services and to implement certain rules and repayments, enable the following functions:

- Service-aware charge.
- Access control.
- Policy control.
- Content filtering.
- Quality of service (QoS).
- Application detection and control.
- Optimization of data traffic.
- Security.

The Figure 13 presents these multimedia platforms.

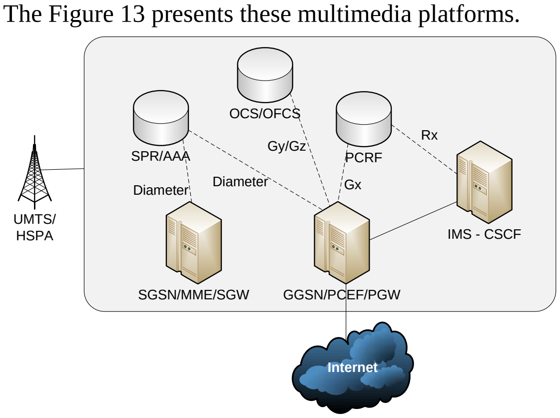

Figure 13. Multimedia platform

Through the implementation of the mentioned platforms, mobile operators get the opportunity to develop new models of revenue sharing with OTT providers. Extra revenue could be achieved by introducing new services.

VI. CONCLUSION

After obtaining the measuring results of the transfer of triple play services across EDGE, WiFi and UMTS access networks, it can be concluded that when introducing a video signal to a voice session over WiFi and UMTS access networks it shows a poor performance, which implies a significantly decreased quality of service. For the EDGE access network this service is completely degraded and is impossible to use.

For the voice transmission, very good performances are obtained for each of these three access technologies.

Further, it is experimented with using different codecs for the transmission over UMTS in order to improve the quality of service for this access network. The best results are obtained using the iLBC codec, which achieved great performances for triple play services.

From the viewpoint of data and instant messages transfer, it can generally be concluded that these sessions were successfully completed for each technology.

Further, the optimization of the number of UMTS radio access network's NodeB's is completed. This was done for a middle size operator, and the conclusion is that in order to offer triple play services to a certain number of users, no additional extensions of the radio access network are needed.

The increase in the number of users using advanced terminals and requests for the triple play service, results in an increase of the network load, which will result in an additional optimization of the network in the future.

Furthermore, a dimensioning of the IMS core for middle-sized fixed operators and mobile operators is done. It can be concluded that the Open IMS Core as a commercial solution would be a functional IMS system, adjusted to customer's requests, and with a high capacity, so user penetration could be satisfied for a long period of time.

Additional research could include testing of the WiFi network for different codecs, understanding jitter behaviour for characteristical measurements, and

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measuring the same parameters over a longer measuring period.

REFERENCES

[1] T. Magedanz, D. Vingarzan, and B. Harjoc, “Generating Realistic NGN load with SIPNuke on Inexpensive Hardware – The One Million Demonstration” Fraunhofer FOKUS, Berlin, 2008.

[2] D. Vingarzan and P. Weik, “End-to-end Performance of the IP Multimedia Subsystem over Various Wireless Networks”, Wireless Communications and Networking Conference, WCNC IEEE, 2006.

[3] J. Prokkola, P.H.J. Perala, M. Hanski, and E. Piri, “3G/HSPA Performance in Live Networks from the End User Perspective”, IEEE International Conference on Communications, 2009.

[4] ITU-T Recommendation P.800, “Methods for subjective determination of transmission quality”.

[5] ITU-T Recommendation G.107, “The E-model, a computational model for use in transmission planning”.

[6] F. Yao and L. Zhang, “OpenIMS and Interoperability with Asterisk/Sip Express VOIP Enterprise Solutions”, Agder University College, Faculty and Communication Technology, Grimstad, Norway, May, 2007.

[7] 3rd Generation Partnership Project, Technical Specification Group Services and System Aspects, “3rd Generation mobile system Release 1999 Specifications”, 3GPP TS 21.101, last access date: 11.02.2011.

[8] 3rd Generation Partnership Project, Technical Specification Group Services and System Aspects, “IP Multimedia Subsystem (IMS), Stage 2, (Release 8)”, 3GPP TS 23.228 V8.5.0 (2008-06), last access date: 11.02.2011.

[9] http://www.openimscore.org/, last access date: 11.02.2011.

[10] T. Szigeti and C. Hattingh, “End-to-End QoS Network Design”, Cisco Press, 2004.