Deploying IP Multimedia Subsystem (IMS) Services in Future Mobile Networks

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Abstract—The Third Generation Partnership Project (3GPP) has proposed the Long Term Evolution and System Architecture Evolution (LTE/SAE) as the next stage of technologies aimed at overcoming the limitation in the existing 2G and 3G networks while driving mobile networks towards 4G standardisation. Although a new architecture has been proposed for the LTE/SAE, however, some technologies from the existing 2G and 3G frameworks are reused to allow for smooth transition and backward compatibility. One of such technologies is the service delivery framework known as the IP Multimedia Subsystem (IMS). In this paper, we investigate and report our early-trials findings on the integration of the IMS and the LTE/SAE architecture specifically looking at key parameters such end-to-end call quality, QoS parameters, IP connectivity and session management. This was achieved by firstly critically analysing key 3GPP and non-3GPP approaches of deploying telephony services over LTE/SAE while making various recommendations based on a wide range of existing research work in the field. In order to demonstrate the recommendations suggested, a high level prototype of the Evolved Packet Core (EPC) prototype was modeled and developed. The EPC model was then integrated with the FOKUS Open IMS Core and deployed on our testbed while carrying out various Quality of Service (QoS) related experimental tests. Our results shows that even though an EPC controlled IMS session generates more packets for initial signaling, it will in the long-run lead to a reduction in bandwidth consumption by the client due to the fragmentation of SIP messages. This in return makes available additional bandwidth for the integration of additional service such as presence and high-resolution video to existing voice services.

Keywords: LTE/SAE, EPC, IMS, VoLGA

I. INTRODUCTION

Over the recent years, 3G technologies have provided huge benefits, applications and a large number of services from multimedia services to mobile broadband innovations. These innovations are continually needed in wireless voice, data and multimedia technologies in order to provide better quality of service (QoS) and higher capacity to mobile end-users. The Third Generation Partnership Project (3GPP) has proposed the Long Term Evolution and System Architecture Evolution (LTE/SAE) as the next stage of technologies aimed at overcoming the limitations in the existing 2G and 3G networks while driving mobile networks towards 4G standardization [1]. The LTE/SAE is an all-IP converged network integrating both fixed and mobile networks based on 3GPP standards. In the LTE/SAE architecture, voice, data and multimedia services can be easily integrated and also communicate over the single core network known as the Evolved Packet Core (EPC) Network. Although a new architecture has been proposed for the LTE/SAE, however, some technologies from the existing 2G and 3G frameworks are re-used to allow for smooth transition and backward compatibility. One of such technologies is the service delivery framework known as the IP Multimedia Subsystem (IMS). The IMS was first initially defined by the 3GPP and 3GPP2 with the objective of integrating and delivering voice, data, multimedia and other mobile network services over an IP-based infrastructure [2]. This will enable operators offer new and innovative services via IMS that in turn attract more subscribers and maintain their existing infrastructure. Although most of the mobile operators do not fully support VoIP and IMS telephony services, however, various options that are later described in this paper have been proposed based on a wide range of concepts and technologies. Since there are various debates, exploring the various ways of deploying telephony services over LTE, it is very important and useful that most of these options be explored experimentally in a controlled environment in order to explore various performance related characteristics of the various implementations.

In order to explore these options and demonstrate the recommendations suggested, a high level prototype of the Evolved Packet Core (EPC) was modeled and developed within our existing Advanced Next Generation Network (ANGN) testbed to allow for various performance based analysis of telephony services in future mobile networks. Our results show that QoS parameters that are centrally controlled by the EPC can further improve the amount of bandwidth consumption in mobile networks as compared to using best effort and retransmission concepts of the public Internet. We also provide a performance based evaluation of key parameters such as end-to-end call quality and session management in environments controlled by the EPC versus best effort concepts in which there is wide various based on the various scenarios as discussed later on in the paper. Section II provides a high level description of various approaches and proposals aimed at delivering telephony services in future mobile networks. Section III focuses on our efforts in extending our transnational ANGN testbed to support telephony and multimedia services for future mobile networks especially within the UK-India project which is sponsored by both UK and India governments. In Section IV the outcomes and analysis of the experiments carried out are further discussed. Finally in Section V we
provide our conclusions and discuss the future direction of our work.

II. RELATED WORK

The Evolved Packet System (EPS) is the 3GPP standard for the convergence of fixed and wireless networks and was first introduced in Release 8 by 3GPP. Together with LTE, it mainly provides huge services for mobile broadband networks. As the EPS is aimed at being an all-IP infrastructure, all the services that will be supported must be able to meet up with both end-users and mobile operator’s expectations. Although most of the services to be deployed over the EPS have already been deployed successfully over the Internet, however, there is a rising concern for the level of quality of telephony services which are currently deployed as Voice over IP (VoIP) over the Internet [3-5]. This is because telephony services in both 2G and 3G networks have normally been Circuit Switched (CS) as this is more reliable because end users are provided with dedicated resources. However the 3GPP EPS architecture does not natively support a CS domain [6], and so if the EPS will be deployed in the future the new solution must be examined to handling voice services. There are currently various proposals and solutions aimed either providing CS enabled voice services in LTE or increasing the level of quality in VoIP services over LTE. In this section we discuss some of these approaches and solutions while also providing overall comparison of the different approaches.

A. Circuit Switched (CS) Fallback

The CS Fallback concept is proposed by 3GPP and defined in the 3GPP TS 23. Its primary aim is to extend the existing GSM/UMTS network by providing the telephony-services and solutions for the LTE networks [7]. GSM/UMTS components such as Mobile Switching Centers (MSCs), Operation Support Systems (OSSs), CS service platforms and billing systems are re-used and enabled to meet the requirements of voice services for LTE. In CS Fallback option, new network elements do not need to be added and also upgraded to the existing network nodes. The SGS interface based on the Gs interface is configured in new MSCs for CS Fallback option. However, CS fallback may experience a high level of delay during the fallback to the GSM/UMTS network [8]. Moreover, this delay can be further increased if the mobile terminal is expected to execute the location update within the GSM/UMTS network before originating or answering a call. Another disadvantage of the CS fallback is that while the CS fallback supports both concurrent voice and data with Dual Transfer Mode (DTM) but actually the 2G networks might not be able to handle simultaneously voice and data connections.

B. Voice over LTE Generic Access (VoLGA)

VoLGA [9] provides voice services for LTE access by using the existing GSM/UMTS voice core networks. Initially, this solution based on 3GPP TS 23.879 option 2 and then further development by the VoLGA Forum. The proposed approach is a new dedicated Internetworking Function (IWF), also known as VoLGA Access Network Controller (VANC). The major advantage of VoLGA is that it enables mobile operators to start offering the voice service with the LTE access by using the existing GSM/UMTS core network without any more requirements or upgrades. VoLGA proposes to deliver the same stable and reliable CS service used in the 2G/3G network while also providing excellent vertical handovers between GSM/UMTS and LTE without any disruption. Similar to CS fallback, VoLGA also provides a complete CS service transparency between the GSM/UMTS and the LTE. However, in VoLGA the voice traffic load is handled by the LTE, not GSM/UMTS access network. Data and voice services are supported simultaneously by VoLGA without any call setup delay because the mobile terminal is always located within the LTE domain. Conversely, the major disadvantage of VoLGA is that its standards have not been accepted by 3GPP. Another disadvantage of VoLGA is that it introduces some new network elements such as the VANC, AAA server and security gateway that add more complexity to its architecture. Finally, Volga also requires a modification of the mobile and has no support for IMS services which are viewed as a differentiator for LTE.

C. Voice over IMS (VoIMS)

VoIMS is seen as the native solution for delivering telephony and multimedia services over LTE [1][2][10]. It uses the IMS call control defined in 3GPP TS 23.228 for LTE voice-service access. As discussed earlier on, the IMS provides an integration of multimedia and data services such as voice, presence, video, gaming and instant messaging by using a wide range of application servers. Furthermore, VoIMS is expected to be widely deployed, ensuring coverage for LTE users whether at home or roaming. The major benefit of VoIMS is that it is globally accepted as the solution for support voice service over LTE. VoIMS is based on the LTE/IMS network that offers a flat all-IP network with full operational cost saving while allowing the mobile operators to offer new revenue-generating applications and services. Since VoIMS has also been implemented within some UMTS networks, it will also provide an excellent ability to handover voice and data simultaneously because data and voice are handled over a single packet switched domain. On the other hand, the major disadvantage of VoIMS is that it requires a large investment in the LTE network due to the mobile operator may need to deploy the IMS core components such as CSCF, TAS and IP-SM-GW [6]. The HSS also is upgraded to support IP-SM-GW. With VoIMS implemented with SRVCC and ICS, some components like SCC, SRVCC ASs and SRVCC also need to be upgraded to the MME and E-UTRAN [8]. For the GSM/UMTS networks, the implementation of SRVCC and ICS requires the MGCF deployment and all MSCs upgrade in order to support the Sv interface; the HSS also needs to upgrade to support IP-SM-GW.

D. Overall Comparison

Although all the approaches have their respective advantages and drawbacks, it is important to note that operators will need to thoroughly analyse the current state of their networks before committing to any of the approaches. At the moment the 3GPP is fully in support of the VoIMS approach, which seems as the right approach for a new operator without any existing infrastructure but maybe of be of high risk for an operator that has invested heavily in the exiting 2G and 3G infrastructures. This is because the network implementation of CS Fallback requires the least effort as the mobile operator only upgrades MSCs that serves the LTE network area while ensuring that the LTE network area overlaps the GSM/UMTS
However, the network implementation of VoLGA is more complicated as new network elements (VANC, AAA server and security gateway) need to be added. This will require a lot for planning and integration. The network implementation of VoIMS is the most complicated of all three as it needs a new service delivery platform as new functions need to be deployed while interacting with existing infrastructures, if any do exist. Core components in CS platforms such as the HLR needs to be upgraded to IMS compatible HSS while various gateways and back-to-back user agents (B2BUA) for media and control translational services also need to be added to the network. However, the VoIMS is future safe and ideal for new operators without any existing CS networks. Most importantly the 3GPP is fully in support of the VoIMS solution over the VoLGA alternative.

In terms of performance, in the CS fallback option, a call is greatly affected by the call setup delay, especially as the quality of LTE to GSM/UMTS voice handover is not guaranteed. In addition, advanced blended services are not provided in CS fallback without DTM and especially voice and data services are not supported simultaneously. On the other hand, in the VoLGA option, a voice call do not need to fallback to GSM/UMTS network so it eliminates the call setup delay comparing with CS fallback option. By using LTE access, VoLGA supports both voice and data services simultaneously and provides the LTE operational benefits of a flat IP-architecture. Moreover, since VoLGA uses the CS-based handover mechanism of legacy MSCs so the voice handovers are performed smoothly between the LTE and GSM/UMTS networks. However, since the IMS is not used in VoLGA, it cannot fully provide the benefits of an all-IP network.

However, the VoIMS option can also be seen as a long-term solution because if mobile operators choose to firstly deploy VoLGA or CS fallback, they may eventually upgrade to the VoIMS solution to gain the fully benefits of LTE networks. VoIMS does not only supports voice, data and multimedia services simultaneously but also delivers the full operational cost savings of using a flat and all-IP network. If the mobile operator plans to implement VolMs with UMTS, VoIMS supports excellent PS-based handovers and hand-backs between the LTE and UMTS [6].

Finally, as there are really no definite answers to which is the best approach, however, it is evident that all solutions will definitely converge to the VoIMS approach as it promises the deliver all the advantages of the Internet and it services to the mobile domain. This has motivated us to further extend our Advanced Next Generation Networks (ANGN) Testbed to support an integrated VolMS and LTE/SAE architecture. A high-level description of the testbed is provided in the next section.

III. THE ADVANCED NEXT GENERATION NETWORKS (ANGN) TESTBED

The Advanced Next Generation Networks (ANGN) testbed at the University of Surrey is an evolving testbed that has been dedicated to the research and development for future converged networks, protocols and services. It has been developed using a layered approach as shown in Figure 1. This layered approach allows for the development of concepts or protocols using a single layer or cutting across different layers. The layers are mainly divided into the access technologies, the IP core and the service domain. More details of each individual layers and the
testbed as a whole can be found in our publication describing the testbed [11].

The ANGN testbed is currently involved in a wide range of international projects, however, the UK-India project is responsible for adding the VoIMS functionality over a live LTE access network. This is motivated by the need to put in place a support infrastructure to facilitate the research and development of Next Generation Networks, Systems and Services in both countries. This is important to allow for early trials, adaptations and implementations of various ideas, concepts and technologies developed within the consortium and by industrial partners more widely. In order to develop the testbed, a number of universities and companies in the UK and India agreed to collaborate under the Theme 10 section of the UK-India EPSRC-DST project. The project has now been running for a year now and this paper provides some of the results from our early trials.

IV. EXPERIMENTAL TESTS AND ANALYSIS

As a follow-up to our discussion in Section II that establishes VoIMS as the future of telephony services, we then put together various scenarios in our lab to carry out early trials and experimental tests and analysis to investigate key parameters that are needed to make VoIMS fully functional in LTE/SAE environments. In order to achieve this we had to integrate a wide range of technologies. The ANGN was first extended to support cloud-computing environments by deploying VMware vSphere, allowing for easy deployment and management of resources. The cloud environment allowed for the provision of virtual severs and clients on demand without the need for additional physical resources. The Open IMS Core implementation from FOKUS Institute at Berlin was then deployed in the cloud environment to provide IMS core network functionality. Finally, the IMS Client developed at the University of Cape Town also known as the UCT IMS client was then deployed in the cloud environment to emulate various end-user configurations. In order to simulate the effects of the EPC, a Linux virtual machine with traffic shaping capabilities was then deployed. IPtables, an open source tool in Linux, was used to enforce the QoS parameters passed by the IMS clients by initiating various network profiles while carrying some sort of deep packet inspection. Although the high-level EPC developed for this paper does not demonstrate other properties of the EPC such as mobility and security as we still in early stages of the UK-India project, however, at the moment it is sufficient enough to evaluate the QoS parameters which within the scope of this paper.

A. Experimental Scenarios

In order to investigate the impact of the different QoS parameters on the whole architecture, all traffic within the system was monitored using wireshark while listening to designated IP addresses and ports numbers. In order to simulate real network environments, wireshark was also used to inject real-life VoIP traffic into the EPC by using packet replay techniques. All client signaling and media traffic were also passed via the EPC as shown in Figure 2, while packets with QoS parameters where given strict high priority classes.

B. Experimental Tests and Analysis

All tests were carried out over 12 times and the results provided in this section based on averages. From the results, we noticed that a significant amount of SIP messages were exchanged in all six scenarios. However, this is not really reflected in the call setup times, which ranges between 7.6 seconds in the non-QoS scenario to 11.7 seconds in the fully mandatory QoS scenarios. This is expected as more SIP request and replies are required to ensure that the expected level of QoS from the clients are met. Please note that the high call setup times recorded in this experiment are due to the fact that the testbed was not optimised to mobile network standards as a typical setup time in mobile networks is about 30 milliseconds. However, Scenario 3 (Optional-Optional) and 5 (Mandatory-Optional) displays a significant drop in call setup time as compared to the other scenarios. This is due to the fact that in scenario 3, there are no changes in the amount of SIP messages as compared to scenario 2. However in scenario 5 this is not the case as there are more SIP messages.

![Testbed Setup with QoS Guarantees](Image)

**EXPERIMENTAL QoS MATRIX**

<table>
<thead>
<tr>
<th>Scenario</th>
<th>Alice QoS</th>
<th>Bob QoS</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>None</td>
<td>None</td>
</tr>
<tr>
<td>2</td>
<td>Optional</td>
<td>None</td>
</tr>
<tr>
<td>3</td>
<td>Optional</td>
<td>Optional</td>
</tr>
<tr>
<td>4</td>
<td>Mandatory</td>
<td>None</td>
</tr>
<tr>
<td>5</td>
<td>Mandatory</td>
<td>Optional</td>
</tr>
<tr>
<td>6</td>
<td>Mandatory</td>
<td>Mandatory</td>
</tr>
</tbody>
</table>

![Call Setup Times](Image)

In order to understand the drop in call-setup time in scenario 5, the wireshark traces where deeply analysed and it
showed that since one of the clients opted for optional QoS, the sequence reverted back to that of scenario 3. This allowed for the call setup times to drop dramatically, however additional messages where still sent by the client requiring mandatory QoS. This in return led to an increase in the amount of SIP messages generated. Although there was an increase in the number of SIP messages this further led to a drop in bandwidth in Scenario 5 as shown in Figure 4.

![Total Bandwidth Usage](image1)

**Figure 4. Total Bandwidth Usage**

![Number of Packets Generated](image2)

**Figure 5. Number of Packets Generated**

Figure 4 also shows that Scenarios 2 and 4 also experienced a drop in bandwidth usage even though they generated additional SIP messages. This is due to the fact that the provisional messages used for enforcing QoS allow for the segmentation of the SIP messages, which in return reduces the size of the SIP messages and the total bandwidth used by the client in the long run. Figure 5 also shows how the amount of messages increases significantly, doubling from 17 packets in scenario 3 to about 45 packets in scenario 5. Although the EPC promises the availability of high bandwidth which may make this insignificant, the fact is that in the long run, this will provide some sort of saving allowing for additional bandwidth to allow voice to be easily integrated with additional services such as video and presence that are more bandwidth demanding.

V. CONCLUSION AND FUTURE WORK

In this paper, the integration challenges between the IMS and the LTE/SAE have been explored and reviewed. Three options for providing voice services over LTE were introduced: CS Fallback, Voice over LTE Generic Access (VoLGA) and Voice over IMS (VoIMS). Each option has their respective advantages and disadvantages, with no clear “winner”. Depending on different factors, mobile operators must consider to choose an option or a combination of options that is the best suited for their network. Another the main challenge faced in voice-over-LTE is the enforcement of the quality of service for telephony services. Our existing ANGN testbed has been extended in order to evaluate various scenarios and option that used by the EPC to enforce QoS. The experimental tests carried out observed the impact QoS parameters have on the IMS SIP signaling load when two subscribers negotiate an IMS session. The observations were implemented with six different setup procedures with no QoS requirements, with optional QoS requirements and with mandatory QoS requirements. The results of the experiments indicated that the amount of data sent by the clients and the core network increases in which some QoS parameters are present between 30% and 70% but the long run reduces the amount of bandwidth consumed by client which can then be passed as saving to either the client or the operator. Consequently, it seems obvious to affirm that the EPC is very important since it provides higher bandwidth, lower latency comparing the current core networks. As a follow up to these results, our next step to start the integration of our extended testbed with the advanced LTE access network in IITM, India as part of the UK-India project.

REFERENCES


