Abstract—As IMS becomes more available to academia and industry, the requirements for IMS clients are growing faster than ever. From all IMS components, IMS clients are vital components for the success of overall services provided by the IMS. Most of ongoing research is focusing on adding more services on clients side, none has been done for voice over Internet protocol (VoIP) quality adaptation so far.

Since the end user is the only one to perceive the quality of services provided by the IMS network, this paper reports a testbed that demonstrates the concept of adaptation of VoIP system based on an open IMS core and open source mobile terminal.

Index Terms—IMS, 3GPP, SIP, QoS, PQoS, AMR

I. INTRODUCTION

The IP Multimedia subsystem (IMS) is an overlay system that is serving the convergence of mobile, wireless and fixed broadband data networks into a common network architecture where all types of data communications are hosted in all IP environments using the session initiation protocol (SIP) [1] protocols infrastructure.

As recent public trials have shown, IMS technology still suffers a number of confining factors, amongst them is perceived quality of service (PQoS). The existing IMS infrastructure does not provide any PQoS aware management mechanism within its service provision control system.

It is expected that the success of multimedia services within the IMS infrastructure will depend on how end users perceive the quality of the services provided. Therefore, novel IMS compatible user centric network management solutions that employ cross layer adaptive techniques are inevitable. These techniques will be deployed into the existing IMS architecture in order to complement it with the objectives to

1) compensate for network impairments,
2) perform content dependent optimization of the encoding and/or streaming parameters, and to
3) improve the end user experience/satisfaction by maximizing the delivered PQoS level.

IMS is logically divided into two main communication domains, one for data traffic, i.e., real time protocol packets consisting of audio, video and data and the second one is for SIP signalling traffic.

During an ongoing session or even before a session has been established, SIP UPDATE method [2] can be used to clients to update parameters of a session (such as the set of media streams and their codecs). SIP UPDATE method has no impact on the state of an existing dialog.

This paper describes a testbed with which two communicating IMS clients perform VoIP quality adaptation under the open IMS core, this is the initial step towards novel IMS compatible user centric network management solutions.

The motivation of this research stems from the fact that current 3G mobile handsets do not have adaptation mechanisms, as a result, when there is a network congestion, handsets have no mechanisms to adopt sending bit rates to help release network congestions. With built-in adaptation mechanism, 3G mobile handsets become more network friendly.

The rest of the paper is outlined as follows, Section II describes the Android IMS client. Section III presents the open IMS core, whereby Section IV depicts the overall built testbed. Section V discusses VoIP quality adaptation in which SIP signalling, voice quality monitoring and adaptation mechanism are outlined. Section VI presents experimental results. Section VII concludes the paper and gives future direction of the ongoing research.

II. ANDROID IMS CLIENT

Android platform [3] has been chosen as an IMS client because its future has shown to be very promising for UMTS access networks. Android is an open handset alliance, a group of more than 30 technology and mobile companies. To help developers to develop new applications, the alliance has offered the android software development kit. The Android emulator screenshot is depicted in Fig. 1.

The Android platform is an open software stack for mobile devices including an operating system, middleware and key applications. Developers have full access to the application framework APIs used by the core applications. The application architecture is designed to simplify the reuse of components; any application can publish its capabilities and any other application may then make use of those capabilities (subject to security constraints enforced by the framework). This same mechanism allows components to be replaced by the user.

The overall Android architecture is illustrated in Fig. 2 [3]. The testbed uses SIPDROID, the basic SIP client application built on the MjSip by HSC. SIPDROID and MjSip have both been released under GPL. SIPDROID has been modified to support basic IMS signalling flow and installed as a package in the Android emulator.
As of today, the Android emulator does not support audio capture, hence the real time transport protocol (RTP) part of the modified SIPDROID has only been emulated. The Android platform has been chosen in this research because it provides a platform to test adaptation mechanism.

III. OPEN IMS CORE

The open IMS core is an implementation of IMS Call Session Control Functions (CSCFs, i.e., P-CSCF, S-CSCF and I-CSCF) and a Home Subscriber Server (HSS), which together form the core elements of all IMS architectures as specified today within 3GPP, 3GPP2 and ETSI TISPAN. All components are based upon open source software and are used to exchange SIP messages, register users and setup/terminate multimedia sessions. The open IMS core has formed the heart of the open IMS playground at FOKUS (c.f., Fig. 3) [4].

The CSCF serves as a centralized routing engine, policy manager and policy enforcement point to facilitate the delivery of multiple real time applications. It is application aware and uses dynamic session information to manage network resources, and to provide advance allocation of these resources depending on the application and user context. The CSCF can act as any of the following:

- **Proxy CSCF (P-CSCF).** This is the first contact point within the IMS for the subscriber. It accepts requests and serves them internally or forwards them.
- **Interrogating CSCF (I-CSCF).** This is the contact point within an operator’s network for all connections destined for a user of that network, or for a roaming user currently located within that network’s service area. There may be multiple I-CSCFs within an operator’s network.
- **Serving CSCF (S-CSCF).** This is responsible for identifying the user’s service privileges, selecting access to the home network application server, and providing access to that server.
- **Home Subscriber Server (HSS).** The HSS maintains a database containing unique service profiles for end users. Service profiles contain service and preference information, such as current registration information (IP address), roaming information, telephony services (call forwarding information), IM service information (buddies list), voice main box options (greeting message), etc.

IV. THE TESTBED

Fig 4 depicts the overall testbed built to perform the VoIP quality adaptation with the open IMS core. The SHUNRA\Storm emulator lies between the Android emulators to emulate the packet loss rate as network impairment. SHUNRA\Storm [5] emulates a multi point WAN in a laboratory conditions. It consists of hardware (emulator) unit and software. Single workstations or entire local area networks are physically connected to one or more SHUNRA\Storm emulators in the laboratory.

The SHUNRA\Storm emulator unit is placed between
the two Android emulator clients to emulate the network conditions during a voice session. In this paper, the SHUNRA\Strom emulates packet loss in random mode.

The UCT IPTV application and VLC streaming servers are added for future research on video and audio streaming PQoS adaptation prediction models. The function of each component in the testbed has been described in detail in the previous sections.

V. VOIP QUALITY ADAPTATION

VoIP is the technology that allows IP networks to be used for voice and video service. VoIP leads to solutions at more or less every layer of an IP network from voice applications to low level quality measurements like packet loss and delay that keep those applications running with acceptable quality. This paper only investigates the voice service, the video part will be dealt in the later stages.

Voice quality prediction model is embedded into the Android terminal, this model is responsible for monitoring the quality of the voice session in real time. The voice quality model used in this paper is the one that was proposed by [6], which measures the mean opinion score (MOS) value of a conversational VoIP session noninteractively.

Adaptive multi-rate (AMR) codec was chosen as an example to develop PQoS models. AMR speech codec was developed by ETSI and has been standardized for GSM. It has been chosen by 3GPP as a mandatory codec and is widely used in current 3G mobile handsets. AMR is a multi-mode codec with eight modes (AMR475 to AMR122) with bit rates between 4.75 Kb/s to 12.2 Kb/s. In current 3G mobile handsets, a fixed bit rate AMR codec (e.g., 12.2 Kb/s) is normally used. Android platform uses AMR and its first mobile handset T-Mobile G1 has been in Market UK since the end of October 2008.

In this paper, for demonstration purposes, only two AMR modes are used for adaptation, the AMR122 and AMR475, packet loss is taken as a network impairment and the callee is the only one monitoring the voice quality.

A. SIP signalling

Assuming users have already registered to the IMS and wanting to have voice session, the caller sends a SIP INVITE message [1] with an offer to use AMR122 to the IMS which then sends the INVITE message to the callee to join the session. The callee sends a RINGING message to the IMS which sends it to the caller. The callee then sends an OK message to the IMS which sends it to the caller. The caller then replies with an ACK signal which is sent to the callee. The session is then setup and both users communicate using RTP with AMR122. These actions are shown in Fig. 5 as use cases.

B. Update

If the voice quality drops below a predetermined MOS value threshold for a predetermined duration, the callee sends an alarm using the instant message (IM) requesting for a change of AMR mode from AMR122 to AMR475 to the caller. The caller will then send UPDATE METHOD request with an offer of AMR475 to the callee who will instantly send an OK to the caller and the RTP session will be running under AMR475. The callee will still be monitoring the voice quality, if the quality continues to drop, there will be no change since AMR475 is the lowest mode. If the quality of the voice has gone up above a predetermined MOS value threshold for a predetermined duration, the callee will send the IM instructing the caller for a change of AMR475 to AMR122 mode. The use cases for these actions are depicted in Fig. 6.
Actor: End user

Preconditions:
- Registration with the IMS network
- Voice session is setup

Postconditions:
- Improvement/degradation of quality
- Continuous monitoring of quality
- Change of AMR mode

The SIP UPDATE message flow for an early media negotiation is shown in Fig. 7. The SIP messages flow is enumerated as follows,

1) The caller sends an initial INVITE which contains an offer of AMR122.
2) The IMS forwards the INVITE to the callee.
3) The callee generates a 180 response which is an answer to the offer.
4) The IMS forwards 180 response to the caller.
5) With the completion of an offer/answer exchange, the session is established, although the dialog is still in the early state.
6) The caller generates a PRACK to acknowledge the 180.
7) The IMS forwards the PRACK to the callee.
8) The PRACK is answered with a 200 OK by the callee.
9) The IMS forwards the 200 OK PRACK to the caller.
10) When the dialog in progress, the callee finds the voice degradation, the MOS value drops below the predetermined value for a predetermined duration, the callee sends an alarm as a request (UPDATE(AMR475)).
11) The IMS forwards the (UPDATE(AMR475)) to the caller.
12) The caller answer the offer with 200 response to the UPDATE (AMR475).
13) The IMS forwards the 200 response to the callee.
14) The callee sends 200 INVITE.
15) The IMS forwards 200 INVITE to the caller.
16) The caller sends an ACK.
17) The IMS forwards the ACK to the callee.

In this scenario neither the 200 OK to the INVITE, nor the ACK, will contain SDP.

The SIP UPDATE message flow for an active media session is shown in Fig. 8 and is outlined below for the callee requesting a change of AMR122 to AMR475 during degradation of VoIP quality,

1) The callee sends the IM requesting for AMR475.
2) The IMS sends the IM to the caller.
3) The caller send an Ok to the IMS.
4) The IMS forwards the Ok to the callee.
5) The caller send the UPDATE message with the offer of AMR475.
6) The IMS forwards the offer to the callee.
7) The callee sends Ok.
8) The IMS send Ok to the caller.

B. VoIP quality monitoring

An alarm caused by voice quality degradation is generated by the QoS probes module (c.f., Fig. 9) within an Android emulator and sent to another Android emulator as a SIP UPDATE method via the IMS when the quality falls below a predetermined threshold.

C. Adaptation Mechanism

The callee of the ongoing VoIP session monitors the PQoS using the model proposed in [6]. The MOS values are used for monitoring the PQoS and average packet loss (in percentage) over a period of time (1 second in this paper) is taken as a network impairment. The SHUNRA\Storm emulates the random packet loss. At the start of the session the AMR122 mode is used, this mode has the highest MOS value when average packet loss is zero. Once the PQoS starts to drop to or below the predetermined MOS value (3.0 MOS score in this paper) for a predetermined duration (1 second in this paper), the callee sends an alarm using the IM with the request to lower the AMR mode to AMR475, this mode has the lowest quality regardless of the packet loss rate. The caller will then change the AMR mode to AMR475.

The callee will still be monitoring the PQoS while the session is still going on, if for a predetermined duration (1 second in this paper), and if the current session is in AMR475
mode, and the callee finds that the average packet loss is zero, then the callee will send the IM to the caller to request a higher AMR mode.

If the current session is in AMR475 mode and the MOS value goes up to or more than 3.2 at the predetermined duration of 1 second, then the callee will send the IM to the caller to request a higher AMR mode of AMR122.

This changes of AMR mode between higher and lower constitutes a simple adaptation mechanism and goes on until the end of the VoIP session. The MOS value against packet loss rate for AMR122 and AMR475 are depicted in Fig. 10.

Lingfen and Ifeachor [6] showed how to derive conversational MOS model from end to end packet loss and delay based on the combined ITU-PESQ and e-model structure. The Equation 1 was used to calculate the MOS values [7].

\[
MOS = \begin{cases} 
1, & \text{for } R \leq 0 \\
1 + 0.035R + R(R - 60) & \text{for } 0 < R < 100 \\
4.5, & \text{for } R \geq 100 
\end{cases}
\]

(1)

where

\[
R = 3.026MOS^3 - 25.314MOS^2 + 87.06MOS - 57.336
\]

(2)

is the rating factor.

\(I_e\) is the equipment impairment from packet loss and codec and \(R\) can be converted to \(I_e\) as in (3).

\[
I_e = R_0 - R
\]

(3)

where \(R_0 = 93.2\) by default [7].

After some manipulation and curve fittings methods, \(I_e\) is derived as seen in (4).

\[
I_e = a \cdot \ln(1 + b \rho) + c
\]

(4)

where, \(\rho\) is the packet loss rate in percentage and the \(a, b\) and \(c\) are constant parameters shown in [6] for different AMR modes.

VI. EXPERIMENTAL RESULTS

The experimental results have shown that, there is a gain in PQoS if there is VoIP quality adaptation mechanism Fig.11. It can be seen that, if the current session is in AMR122 (AMR(H) in the figure) mode and the voice quality starts to drop below a predetermined MOS value for a predetermined duration, the adaptation mechanism is triggered and hence a gain in voice quality is achieved, the graph also shows how the change of AMR475 to AMR122 mode takes place if there is no packet loss for a predetermine duration or when the MOS value is at least 3.2.

The MOS value for VoIP quality without adaptation is depicted in Fig. 12. The monitored MOS values drops down without a gain when there is no adaptation mechanism in place, this contrary to what happened when there is adaptation mechanism taking place.
VoIP Quality Adaptation

AMR (H)

AMR (L)

Fig. 11. MOS versus packet loss rate (%)

VoIP Quality Without Adaptation

AMR (H)

AMR (L)

Fig. 12. MOS versus packet loss rate (%)

VII. CONCLUSIONS AND FUTURE WORK

This paper has built an open IMS core testbed with VoIP quality adaptation using Android platform as IMS clients, AMR codec with two modes (i.e., AMR122 and AMR475) for adaptation and SHUNRA\Storm to emulate packet loss rate as network impairment. The experimental results have shown the importance of VoIP quality adaptation mechanism during an IMS VoIP session. Future work will entail a fully working Android platform with added modules in IMS for monitoring and adaptation of video and voice IMS session, multiple network impairments will also be considered in the future.

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