THE VOIP QUALITY OF SERVICE OVER IMS

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Abstract

IP Multimedia Subsystem (IMS) has resulted from the work of the 3GPP) toward specifying an all-IP communication service infrastructure. This thesis uses the E-Model developed by ITU-T and for select network and voice parameters like coding scheme, packet loss limitations, and link utilization level in IMS network. Is finded the optimal voice coder given link bandwidth

IMS subsystem is an system that is serving the convergence of mobile, wireless and fixed broadband data networks into a network architecture. Used IP environments using the session

initiation protocol (SIP). IMS is logically divided into two main communication domains, one for data traffic and data and the second one is for SIP signaling traffic. Real time protocol packets are consisting of audio, video. To measure of quality is difficult subjective factor. The reasons for low quality voice transmission are due to degrading parameters like delay, packet delay variation, codec related impairments like speech compression, echo and packet loss.

There are models were developed to measure the VoIP QoS a quality rating correlated to the subjective Mean Opinion Score (MOS score) which represents the QoS for Voice calls. Many of the developed models for measuring VoIP quality of service are inappropriate for smaller net-works. They take too much process resource or contain very complicated test algorithms. One of the best models used for measuring VoIP quality of service is the E-model, which is a parameter-based model.

This thesis used estimate to determine the effects of introducing IMS to networks in terms of bandwidth usage on the service quality. As the input traffic characteristics - the number of calls per second and mean holding time and network characteristics, such as a losses and propagation delays. The output of the model provides details on the bandwidth needed for successfully establishing a session when using SIP over UDP in IMS networks. Voice traffic in IP Multimedia Subsystem (IMS) will be served using Voice over IP (VoIP). Thus used E-Model as an optimization to select voice parameters like coding scheme and link utilization level in IMS Network. The goal is to de-liver guaranteed Quality of Service for voice while maximizing the number of users served. This can be used to determine the optimal configuration for a Voice over IP in IMS network. SIP used over various transport protocols such as UDP, TCP or SCTP. To enable the reliable transmission messages even when used over UDP, SIP supports application level retransmission mechanisms. In case no response was received for a sent request then after a timeout the request is re-transmitted. Losses due to overloaded servers or lossy links would cause delays in the session establishment and reduce the perceived service quality.

Bandwidth calculation for IMS session establishment and modeling IMS session establishment for bandwidth calculation for IMS session and estimation set up of IMS session (see figure1). Figure1 show that the calls traverse SIP proxies. Each link of the depicted network has a loss rate of and has a propagation delay. The simulation is done using OPNET for VoIP in IMS network using SIP Protocol. MATLAB is used in this thesis. Getted the result values use in E-Model equations to estimate the quality of service factor R.

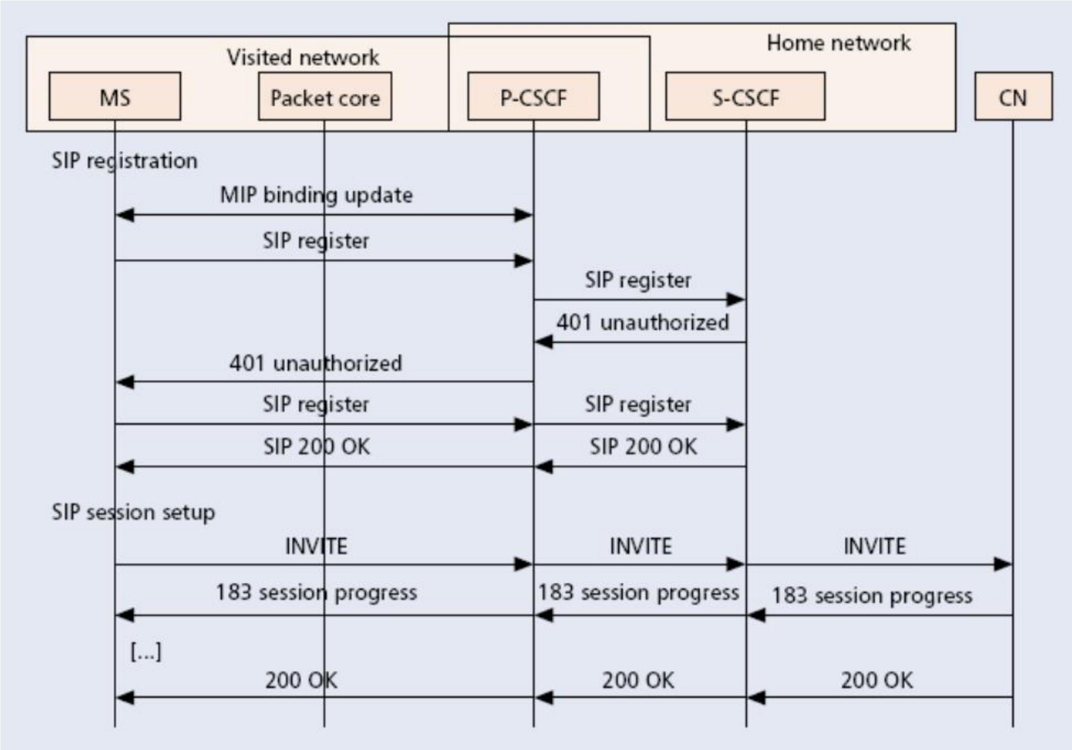

Fig. SIP registration and session establishment in the IMS

Maximized the of voice quality R (table1).

R	90<R < 100	80< R < 90	70< R < 80	60 < R < 70	50< R < 60	R < 50
user satisfaction	Very satis-fied	Satisfied	Some users dissatisfied	Many users dissatisfied	Nearly all users dissat-isfied	Not recom-mended
MOS	4.3-5.0	4.0-4.3	3.6-4.0	3.1-3.6	2.6-3.1	< 2.6

Voice coder selection is done given link bandwidth, packet loss. Adaptive Multi-Rate Wideband (AMR-WB) codec with bit rate 23.85 (kbps) provides improved speech quality as a result of the wider speech bandwidth 50 - 7000 Hz , this comes at the cost of additional processing. The AMR-WB codec has a 16 kHz sampling rate and the coding is performed in blocks of 20ms. There are two frequency bands that are used: 50-6400 Hz and 6400-7000 Hz. We found AMR-WB codec that is the optimal coder as it gives the maximum number of calls keeping its R factor more than 80.

Conclusion

In this thesis use provide a estimate that can be used to determine the effects of introducing IMS with determinate of bandwidth and the effects of losses and delays on the service quality. The output of provides details on the bandwidth needed for successfully establishing a session when using SIP over UDP in IMS networks.

References:

1.3GPP LTE-Advanced. Available from: Access http://www.3gpp.org/technologies/keywords-acronyms/97-lte-advanced links.html. Accessed 28 May 2015.

2.23.228 T 2009 IP Multimedia Subsystem (IMS) – Stage 2 (Release 9), Technical specification group core network and terminals, 3rd Generation Partnership Project.