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Analysis of QOS in Multiservice Networks Based on IMS

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ABSTRACT:IP Multimedia Subsystem (IMS) has resulted from the work of the Third Generation Partnership Project (3GPP) toward specifying an all-IP communication service infrastructure. Mainly looking at the needs and requirements of mobile operators, the 3GPP first specified IMS as a service architecture combining the Internet's IP technology and wireless and mobility services of current mobile telephony networks. After that the IMS architecture was extended to include fixed networks as well. By deciding to use session initiation protocol (SIP) as the signaling protocol for session establishment and control in IMS instead of developing its own set of protocols, 3GPP has opened the door toward a tight integration of the mobile, fixed and Internet worlds. Recent reports already indicate that there are more than 200 million subscribers using the IMS technology for telephony services.

KEYWORDS: Quality of Service (QoS), Bandwidth, Session Initiation Protocol (SIP)

I. INTRODUCTION

In this paper we provide a theoretical model that can be used by operators and network designers to determine the effects of introducing IMS to their networks in terms of bandwidth usage for example and the effects of losses and delays on the service quality. This model uses as the input various traffic characteristics such as the number of calls per second and mean holding time and network characteristics, such as 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 [1].

Voice traffic in IP Multimedia Subsystem (IMS) will be served using Internet Protocol (IP) which is called Voice over IP (VoIP). This chapter uses the "E-Model", (ITU-T Rec. G.107 2005), as an optimization tool to select network and voice parameters like coding scheme, packet loss limitations, and link utilization level in IMS Network. The goal is to deliver guaranteed Quality of Service for voice while maximizing the number of users served. This optimization can be used to determine the optimal configuration for a Voice over IP in IMS network.

II. RELATED WORK

With the success of SIP, there have already been a number of studies addressing aspects of performance evaluation and modeling of SIP. Chebbo et al. describe in (Chebbo et al. 2003) [5] modelling tool with which it is possible to estimate the number of required SIP entities for supporting certain traffic. Gurbani et al. present in (Gurbani et al. 2005) [2] a theoretical modelof a SIP server using queuing theory. This model is then used to evaluate the performance of a SIP server in terms of response time and number of served requests. Wu et al. analyze in (Wu et al.2003) the usage of SIP for carrying telephony information in terms of queuing delay and delay variations. In general, these studies aim at investigating the performance of SIP servers in terms of the number of SIP sessions that can be supported by a SIP server or the processing delays at such servers. In contrast, in our work we donot aim at modeling the performance of a SIP server but to investigate the performance ofSIP in terms of the number of messages and amount of time needed by SIP for establishing a session in lossy environments. Fathi et al. (Fathi et al.2006) [3] present a model of SIP in VoIP networks and investigate the effects of mobility on the performance of session establishment using SIP. The used model is however rather simplified and is only applicable to stateless SIP proxies which have no notion of transactions. Alam et al. (Alam et al.2005) discuss different performance model for SIP deployment scenarios in mobile networks. This involves providing models for evaluating the performance of push-to-talk applications or the effects of different mobility concepts. The work does not however provide for a model of how SIP itself deals with losses. Sisalem et al. (Sisalem et al. 2008) [4] provided a theoretical model of the effects

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of losses and delays on the performance of SIP. While that work is providing the basis for our work here, it is rather limited to simple SIP networks as are discussed in IETF.

III.PROJECT SETUP

The simulation is done using OPNET simulation tool IT Guru Academic Edition 9.1 for VoIP in IMS network using SIP Protocol. The network consists of IP-Telephones (VoIP or IMS Clients) connected to the Internet by routers which act as IP gateway, the network is managed by the SIP proxy server (act as P-CSCF) which uses the SIP protocol to establish the voice calls (VoIP) on the IMS network as shown in figure 3. The links between the routers and the Internet are T1 with link speed 1.544 Mbps and the links between the dialer, dialed, Proxy Server and the routers are 1000 Base-x. The idea is to configure the network with a certain parameters and run the simulation then getting from the tool the result values which used in E-Model equations to measure the Quality of service Factor R. The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R). The cases considered are:

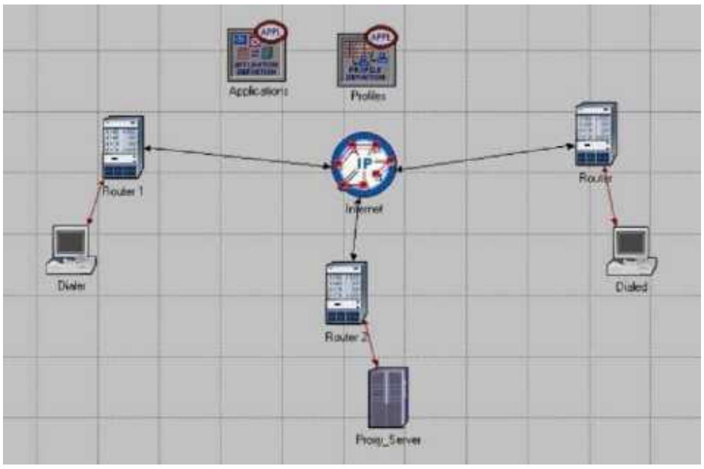

Fig.1. Network Topology

1. Find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level.
2. Find the optimal voice coder and the optimal packet loss level given link bandwidth and background link utilization.
3. Find the optimal voice coder and the optimal background link utilization level given link bandwidth and packet loss level.

Table 1. Codec Parameters

Codec Information	Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Number of Voice Frames per Packet	Packets Per Second (PPS)	Bandwidth with RTP/UDPTP/ PPP Header	MTU Size (Bytes)
G.711 (64 Kbps)	160 Bytes	20 ms	1	50	82.8 Kbps	207 Bytes
G.729 (8 Kbps)	20 Bytes	20 ms	1	50	26.8 Kbps	67 Bytes
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	1	34	18.9Kbps	71 Bytes

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IV. RESULTS

The results are divided into three general cases. For all cases, the aim is to maximize the number of calls that can be carried on a link while maintaining a minimum voice quality level (R > 70).If two combinations produce the same number of calls, the highest R value will be considered the best selection.

Case 1 - Optimizing for coder selection

The goal of this case is to find the optimal voice coder given link bandwidth, packet loss level, and background link utilization level. OPNET is configured by these parameters which are according to the ITU-T G.107 (ITU-T Rec. G.107 2005).

Table 2 shows the main differences between the different codecs G.711,G.729 and G.723.1 with respect to the coding type, coder bit rate, frame length , number of voice frames per packet and finally the Ie for each coder in case of no packet loss.

Table 2. Codec Parameters for case1-1 (ITU-T Rec. G.107 2005) & (ITU-T Rec. G.113 2007)

Standard	Type	Codec Bit Rate (kbps)	Voice Frame Length (ms)	Look ahead (ms)	Frame length (ms) Packet Length	Number of Voice Frames per Packet	I_e№ PL
G.711	PCM	64	0.125	0	20	1	0
G.729	CS- ACELP	8	10	5	20	2	11
G.723.1	MP- MLQ	6.3	30	7	30	1	15

Table 3. Codec Parameters for case1-2 (ITU-T Rec. G.107 2005) & (ITU-T Rec. G.113 2007)

Codec Information	Bandwidth Calculations					
Codec & Bit Rate (Kbps)	Voice Payload Size (Bytes)	Voice Payload Size (ms)	Number of Voice Frames per Packet	Packets Per Second (PPS)	Bandwidth with RTPUPPIP PPP Header	MTU Size (Bytes)
G.711 (64 Kbps)	160 Bytes	20 ms	1	50	82.8 Kbps	207 Bytes
G.729 (8 Kbps)	20 Bytes	20 ms	2	50	34.8 Kbps	87 Bytes
G.723.1 (6.3 Kbps)	24 Bytes	30 ms	1	34	18.9 Kbps	71 Bytes

Table 3 also shows other differences between voice codes G.711, G.729 and G.723.1 with respect to the bandwidth calculations like voice payload size, number of packets per second and the bandwidth required after adding the headers of other protocols. For Case 1 with a link speed of 1.544 Mbps, The simulation was run for 2 hours and 4 hours and in all cases G.723.1 gave the max. Number of calls with R value more than 70, so G.723.1 was selected as the optimum Coder. G.711 gave the max. Quality of service (Highest R value) but the lowest number of calls, G.729 gave middle number of calls between G.711 and G.723 and also middle R value. As shown in figure 5. Figure 4 shows the average packet end to end delay for different codecs and figure 6 shows the number of connected calls for different coders.

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Table 4. Opnet results for case 1

PL ratio	Codec	Delay (T_A) (ms)	I_D	I_e	R	MOS	Calls
0%	G.711	115	2.76	0	90.44	4.33	28
	G.729a	89	0	11	82.776	4.02	30
	G.723.1	97	0	15	78.2	3.86	39

The results of this case are shown in Figure 6 not surprising, as G.723.1 is a more efficient but lower quality of voice.

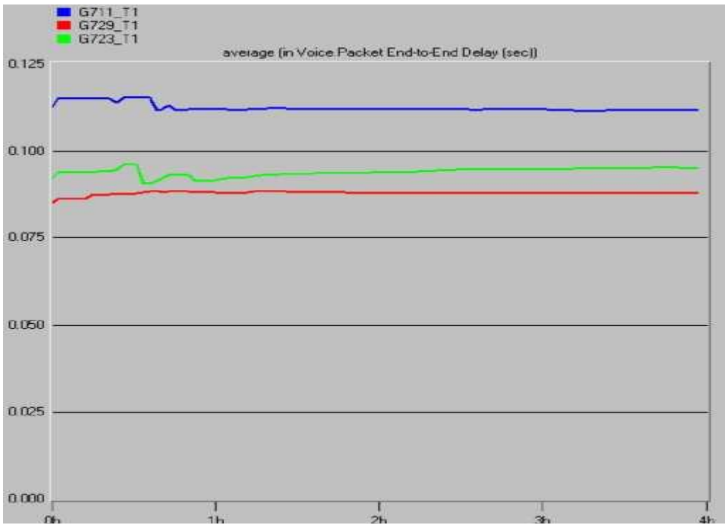

Fig.2. Average Packet End to End Delay

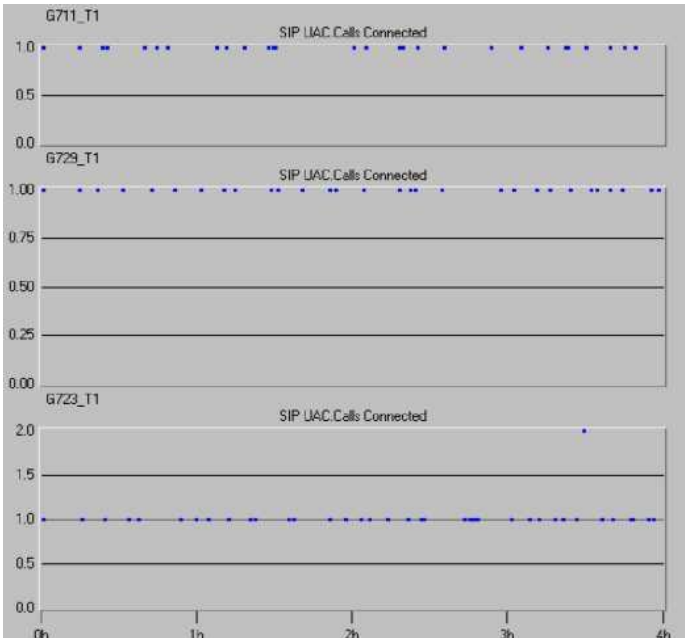

Fig.3. Number of Connected Calls for different codecs

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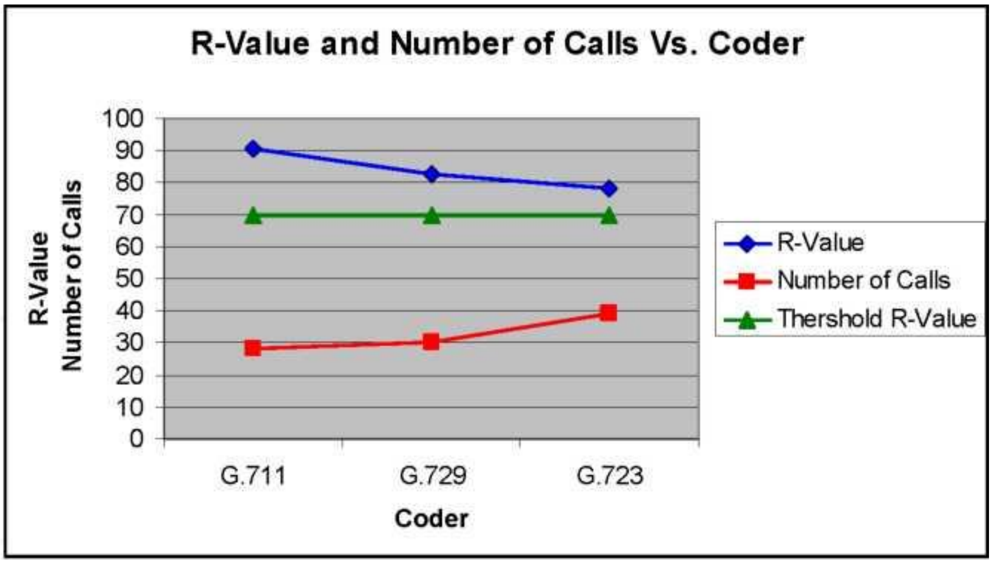

Fig.4. R Value, Number of Calls vs. Coder - case (1)

V. CONCLUSION

IP Multimedia Subsystem (IMS) is very important due to the critical role it plays in the Next Generation Network (NGN) of the Fixed and Mobile Networks.

In this paper we provide a theoretical model that can be used by operators and network designers to determine the effects of introducing IMS to their networks in term of bandwidth usage needed to establish IMS session. The inputs of this model are the required number of Calls or Sessions per Second, Network losses, SIP Messages size, Number of Network hops and number of ringing times. The output of this model is the bandwidth needed to insert IMS in the network.Voice traffic in IMS will be served using Internet protocol (IP) which is called Voice over IP (VoIP). This chapter uses the "E-Model" developed by ITU-T as design tool to select network and voice parameters like coding scheme, packet loss limitations, and link utilization level in IMS Network.The objective function for all cases is to maximize the number of calls that can be active on a link while maintaining a minimum level of voice quality (R> 70).The cases considered are:

- Find voice coder given link bandwidth, packet loss level, and link utilization level.
- Find voice coder and packet loss level given link bandwidth and background link utilization.
- Find voice coder and background link utilization level given link bandwidth and packet loss level

OPNET and MATLAB are the optimization tool that is used in this chapter.

In case 1, we found that G.723.1 is the optimized coder as it gives the maximum number of calls keeping its R factor more than 70. The quality of speech is generally higher with G.729A and G.711. But G.729A and G.711 uses more bandwidth than G.723.1. In Case 2, both G.729A and G.723.1 were sensitive to changes in packet loss, but G.711 was not as sensitive. In Case 3, voice quality was not sensitive to changes in the link load until the link load grew above approximately 94%.

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