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The Impact of Coupling Signaling Protocols and Codecs Scheme in Achieving VoIP Quality

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Abstract

VoIP is short for voice over Internet protocol and is also known as IP telephony, VoIP is a modern technology that enables us to make voice calls using IP networks. Therefore, VoIP can be achieved on any data network that uses IP, like the Internet, Intranets and Local Area Networks. In traditional network data is always fragmented into many data packets then transmitted independently. As the result packets arrived out of order at the destination ,in e-mail applications and downloading document this disorder is represent no problem since the packets will be reassembled in the correct order when they all has arrived at the destination. VoIP uses different signaling protocols and Coding schemes, choosing the inappropriate coding scheme for one of the signaling protocols leads to poor Quality of Service (QoS). This paper aims to specify the best combination of coding scheme and signaling protocols and Measure the QoS parameters (jitter, end to end delay, and throughput) of transmission protocol.

Keywords: VoIP; Quality of Service (QoS); Codec schemes; SIP.

1. Introduction

VoIP means that calls are transmitted over an IP network such as the Internet instead of Public Switched Telephone Networks (PSTN), in other world transferring voice over the traditional network as well as data.

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The VoIP protocols suite is broken into two Types, control plane protocol and data plane protocol. The control plane is the traffic required to connect and maintain the actual user traffic. It is also responsible for maintaining overall network operation (router to router communications). The data plane (voice) portion of the VoIP protocol is the actual traffic that needs to get from one end to another. VoIP also uses codecs schemes or voice coder-decoder (VOCODERs) to convert analog signal to digital signal and vice versa. Codecs are characterized with different sampling rates. Different codecs employ different compression methods, using different bandwidth and computational requirements. The common codec types used in VoIP networks are 64 kb/s G.711, 8 kb/s G.729 and 5.3/6.3 kb/s G.723.1 [11].

VoIP uses number of protocols in order to manage connection establishment. Signaling protocols are used to set up and tear down calls, carrying information required to locate users. There are several VoIP call signaling protocols. H.323 protocol suite, Session Initiation Protocol (SIP), Media Gateway Control Protocol (MGCP), and Megaco/H.248. H.323 and SIP is peer-to-peer Control-signaling protocols [7]. SIP is the Internet Engineering Task Force's (IETF's) standard for multimedia conferencing over IP. SIP is an ASCII-based, application-layer control protocol that can be used to establish, maintain, and terminate calls between two clients. In the other side, H.323 is a recommendation from the Telecommunication Standardization Sector (ITU-T), H.323 consists of a set of protocols that responsible for encoding, decoding, and packetizing audio and video signals, for call signaling and control, and capability exchange [12].

VoIP uses different signaling protocols and Coding schemes, choosing the inappropriate coding scheme for one of the signaling protocols leads to poor Quality of Service (QoS).

The aim of this paper is specify the best combination of coding scheme and signaling protocols, evaluate the suitable transport protocol (TCP, UDP) that has better performance for VoIP packets transmission the., and finally, to measure the QoS parameters (jitter, end to end delay, throughput) of transmission protocol**.**

Design H.323 and SIP architecture and implement E-model to calculate R factor and MOS is the methodology followed by this research to achieve the desired goals. OPNET Modeler 14.5 has been chosen to emulate the performance of the VoIP networks because it is one of leading environment for network modeling and simulation**.**

2. Methodology

Flow chart in Figure (1) indicates the research methodology, first step is choosing the signaling protocols SIP and H.323 each one has different set of elements and parameters.

The second step selects the codec type (G.711, G.729, and G.732.1), after that define the number of frame per packet and transmission protocols (TCP and UDP) as they have impact in QoS parameters. Then run the simulation to get result and measured the throughput, packet loss, end to end delay and jitter which are all QoS parameters. Afterward calculate MOS and R factor .Finally come out with best combination between signaling protocols and codec schemes that gives better VoIP quality.

3. Result and discussion

The topologies in figure (2) (3) used to experiments different codec scheme and the two signaling protocols, H.323 and SIP, using OPNET simulation in order to get the best couple. The E-model was used to calculate the R factor and MOS. The results are presented in this section.

3.1.1End to End Delay

End to end delay, or mouth to ear delay, is combination of network delay, encoding delay, decoding delay, compression delay and decompression delay. Table (1) (2) shows the end to end delay for both SIP and H.323 operate different codecs G.711, G.729 and G.723. When compared these values with H.323 which is 0.18, 0.22ms and 0.25 for G.711, G.729, and G.723.1 respectively, it is clear that SIP and G.711, when number of frame per packet is one and transport layer protocol is UDP, gives better result in term of delay than other combination. Packet end to end delay for voice codec G.723.1 is highest in the two protocols 0.29 ad 0.34 in SIP, H.323 respectively. This is because, G.723.1 uses coding rate of 5.3Kbps or 6.3Kbps which results in the formation of packets of smaller size and larger count. As the number of packets increases in the network, the congestion in the network increases. Congestion directly affects the network packet delay and thus results in increasing packet end to end delay. Also the numbers of frame per packet impact on end to end delay, when the numbers of frame per packet increase the delay also increase.

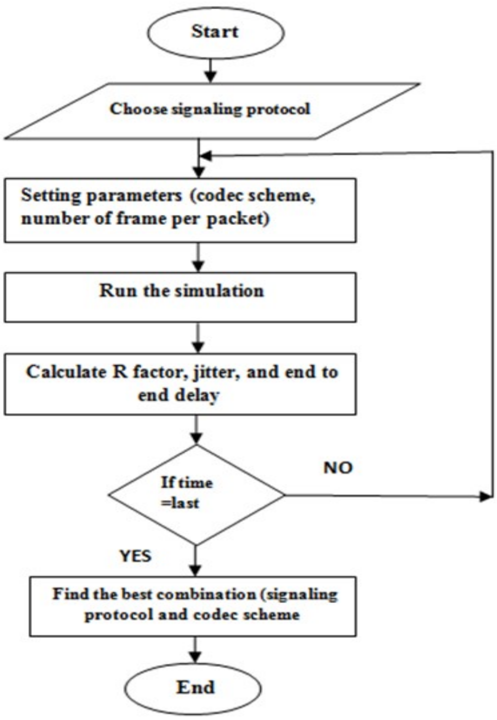

Figure 1: Methodology Flow Chart

In SIP TCP is 0.24 and UDP is 0.16 this indicates TCP has highest delay in both two protocols (SIP and H.323) and in all codecs scheme than UDP.

3.1.2Jitter

Each packet of voice information takes a different amount of time to go from one end of the network to the other. This variation is called jitter. The jitter value should not exceed 20 to 50 milliseconds. Table (3) (4) show the jitter values, all the jitter values are in the acceptable area.

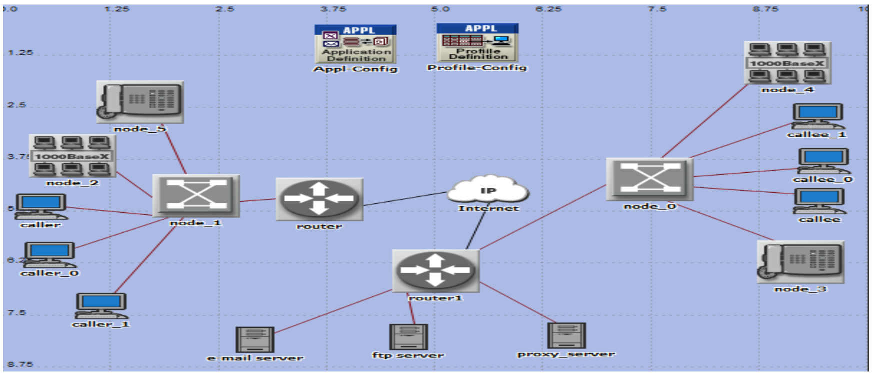

Figure 2: SIP Architecture Implemented in OPNET

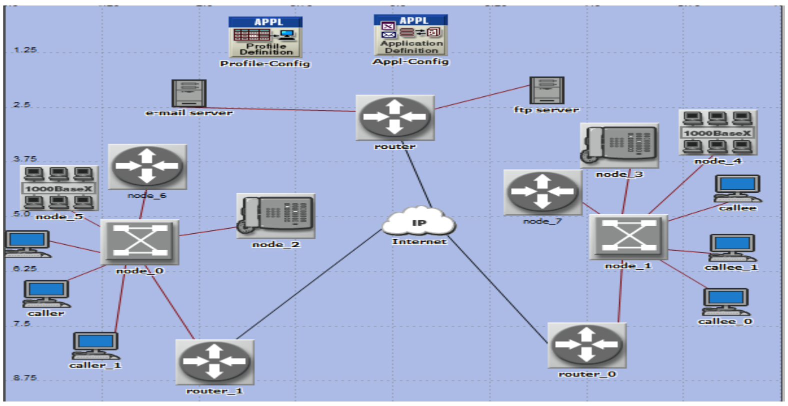

Figure 3: H.323 Architecture Implemented in OPNET

Table 1: Result Comparisons End to End Delay (H.323 protocol)

Codec Scheme	**Transport layer protocol**	**One Frame (sec)**	**Three Frames (sec)**
G.711	TCP	0.22	0.17
	UDP	0.18	0.20
G.729A	TCP	0.24	0.21
	UDP	0.22	0.22
G.723.1	TCP	0.26	**0.35**
	UDP	0.25	0.34

Table 2: Result Comparisons End to End Delay (SIP protocol)

Codec scheme	**Transport layer protocol**	**One Frame (sec)**	**Three Frames (sec)**
G.711	TCP	0.24	0.24
	UDP	0.16	0.22
G.729A	TCP	0.17	0.20
	UDP	0.16	0.18
G.723.1	TCP	0.25	0.35
	UDP	0.22	0.29

Table 3: Result Comparisons Jitter Results (H.323 protocol)

Codec scheme	**Transport Layer protocol**	**One Frame**	**Three Frames**
G.711	TCP	0.000090	0.000002
	UDP	0.000061	0.00018
G.729A	TCP	0.00010	0.000070
	UDP	0.000035	0.000061
G.723.1	TCP	0.00007	0.00007
	UDP	0.00008	0.00008

Table 4: Result Comparisons Jitter Results (SIP protocol)

Codec scheme	**Transport layer protocol**	**One Frame (sec)**	**Three Frames (sec)**
G.711	TCP	0.00015	0.00028
	UDP	0.000005	0.00020
G.729A	TCP	0.000044	0.00013
	UDP	0.000045	0.00000
G.723.1	TCP	0.00036	0.00020
	UDP	0.00040	0.00012

3.1.3Throughput

Throughput is the average amount of data that is successfully delivered over a communication link. Figure (4) (5) shows the throughput delivered rate for TCP and UDP, TCP has larger throuput in all scenarios. TCP is 185 packets in H.32 and 79 in SIP which mean that H.323 has larger throuput than SIP. In both protocol SIP and H.323 there are seems to be loss in performance after period of time.

3.2Mathematical Analysis

The calculation of the R factor and MOS are implemented and listed in Table (5) (6).

The results shows that codec G.711 and SIP is 89.1, 4.3 for R factor, MOS respectively, and in case of H.323 is 87.8, 4.28 for R factor, MOS respectively; whereas, codec G.723.1 has the lowest MOS value in H.32 is 3.24 and SIP is 3.52. It means that codec G.711 has higher quality, compared with the other two codecs. As the bit rate for G.723.1, G. 729 and G.711 is 5.3 Kbps, 8Kbps and 64 Kbps respectively .The delay for G .729 and G.711 is 30 milliseconds, 10 milliseconds, and 0.75 milliseconds. Based on the bit rate and delay for three codec, the MOS value rating from the simulation results makes sense because higher compression rate makes shorter delay which leads to higher voice quality. We observe that the faster the bit rate and shorter the delay of a codec, the better the quality, MOS, of VoIP.

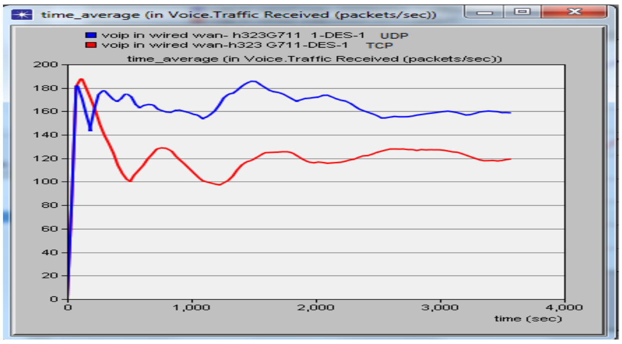

Figure 4: Packet Receive in H.323 (UDP and TCP)

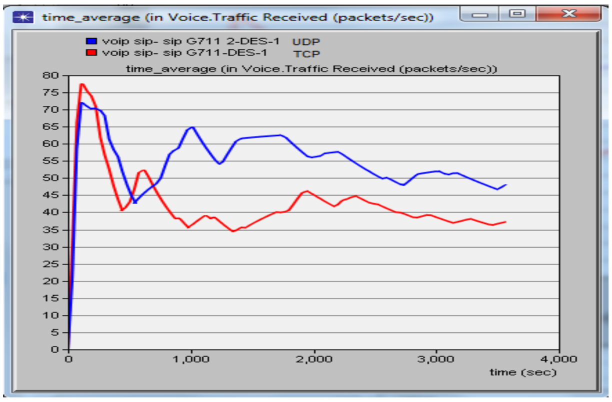

Figure 5: Packet Receive in SIP (UDP and TCP)

Table 5: Result Comparisons R factor and MOS (H.323 protocol)

Transport layer protocol	**Codec**	**R Factors 3 Frames**	**MOS 3 Frames**	**R Factors One Frame**	**MOS 1 Frame**
TCP	G.711	88.6	4.3	83.3	4.14
	G.729A	73.6	3.76	69.4	3.58
	G.723.1	**52.0**	**2.68**	62.8	3.24
UDP	G.711	**89.5**	**4.33**	87.8	4.28
	G.729A	74.8	3.81	72.3	3.7
	G.723. 1	55.3	2.85	62.8	3.24

Table 6: Result Comparisons R factor and MOS (SIP protocol)

Transport layer protocol	**Codec**	**R Factors Three Frames**	**MOS Three Frames**	**R Factors One Frame**	**MOS One Frame**
TCP	G.711	80.6	4.04	80.6	4.04
	G.729A	74.8	3.81	78.1	3.95
	G.723.1	**52**	**2.68**	64.2	3.31
UDP	G.711	**83.3**	**4.14**	89.1	4.32
	G.729A	76.8	3.90	77.7	3.93
	G.723.1	58.9	3.04	68.3	3.52

4. Conclusion

The research study the performance of the most popular signaling protocols used in VoIP, these protocols are , SIP and H.323 with codec schemes G.711, G.729A and G.723.1. OPNET is used to simulate the SIP and H.323 architectures to evaluate the QoS parameters (end to end delay and jitter), also investigate the impact of the number of frame per packet

The average voice jitter variation in case of codec G.723.1 is higher than the other two codecs. The jitter variation in case of G.711 lies between two other audio codecs and audio codec G.729A gives better results than audio codecs G.711 and G.723.1 respectively. So there is a high increase in jitter as audio codecs G.711 and G.723.1 are added to the network. This increase in voice jitter makes the voice difficult to understand as arriving packets at different time. All these codec schemes yield acceptable voice jitters .The use audio codec G.729A will make the jitter less and best performance of VoIP application in both signaling protocols. Furthermore, the study concludes that the end to end delay is increase as number of frames in one packet increase.

The E-model is used to calculate the R factor and MOS; accordingly the most appropriate couple of codec and signaling protocols is G.711 and SIP as offers suitable jitter, best minimum delay and best call quality

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