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IMS Based IP Telephony Over MPLSNetwork

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Abstract— Due to the fast growth of the internet, the present trend people request more applications over internet such as video calling and flash. VoIP (Voice over Internet Protocol) is an emerging technology for voice communication which deploys many techniques to produce a high-quality service. Signaling is one of the most prominent key of Voice over IP; the second one is most promised approaches in this case are SIP and IMS. This paper will compare the call setup delay of SIP and IMS , Measure the QoS of a Network and compare Different voice codec's using different parameters using a powerful network simulation tool (OPNET Modeler) which enables network simulation by employing various protocols.

Keywords*-VoIP, IMS , SIP, G.711, G.723, G.729, GSM OPNET, QoS*

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I. INTRODUCTION

The IP Multimedia Subsystem (IMS), specified by the 3rd Generation Partnership Project (3GPP) as part of Next Generation Networks (NGN), is used to reduce operational cost and provide converged services to the users. The main protocol used for session control procedures is Session Initiation Protocol (SIP) Using Voice over IP (VoIP) .compared to SIP , IMS can offer better data transmission services to real-time applications which benefits not only the IMS clients but also the IMS network provider.

This paper focuses on the IMS based IP Telephony over MPLS using SIP as a signaling protocol and there are three issues : the first and foremost issue is deploying a SIP and IMS Networks, The second issue is regarding the call setup delay of SIP is more compared to IMS, The third issue is measuring QOS Of a network and are compared with different common voice codec's such as G.711, G.723,G.729,GSM using different parameters such as Mean Opinion Score, Traffic send, Traffic received, Packet delay variation, Packet end -to -end delay, Jitter.

II. RELATED WORK

The SIP protocol is designed to be a part of the overall IETF multimedia data and control architecture and is used in conjunction with several other IETF protocols such as SDP,RTSP, SAP.. It is a signaling protocol that handles the setup, modification and teardown of multimedia sessions.SIP is a client-server protocol, VoIP calls using SIP are originated by a client and terminated by a server.

Multi-protocol Label Switching (MPLS) is a new switching architecture for IP core network. It provides end-to-end resource availability for a large no of sessions. The approach with MPLS is to attach a label to the packet. The content of the label is specified according to an FEC, determined at the point of ingress to the network..

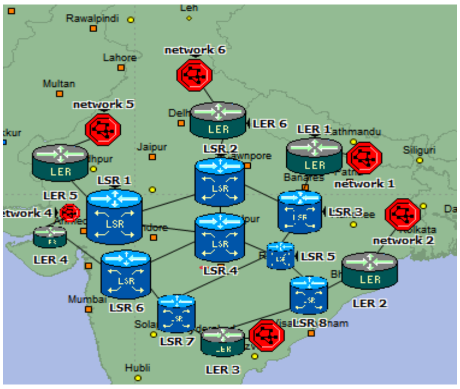

Fig.1: SIPOver MPLS Network

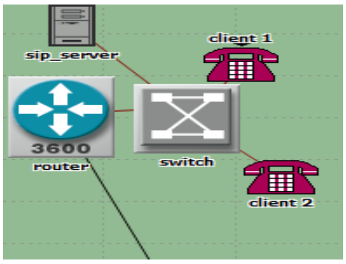

Fig.2 : Inside a SIP Network

Figure1 shows that the scenario contains LSR's, LER's and networks. In Figure2 it describes that each sub-network attached to the LER have SIP clients and server, a switch and a gateway. To enable MPLS switching, we set up dynamic label switched paths (dynamic LSPs) between all the LERs. The SIP clients in all sub-networks are configured to call each other randomly according to the properties of profile, routing and application attributes in Figure[3,4,5]. In Figure6, the SIP call setup delay of Client during 16minutes simulation is shown. We can see that the SIP call setup delay shows in between 0.45milliseconds and 0.55milliseconds.

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IP Interface Addressing Mode	Auto addressed
LDP Discovery End Time	250
IP Dynamic Routing Protocol	OSPF
LDP Discovery Start Time	100
LSP Signaling protocol	RSVP
RSVP Sim Efficiency	Enabled
OSPF Sim Efficiency	Enabled
OSPF Stop Time	260

Fig.3 : Routing Protocol Attributes

Silence Length(seconds)	Default
Talk Spurt Length (seconds)	Default
Symbolic Destination Name	Voice Destination
Encoder Scheme	G.729
Voice Frames Per Packet	10
Type Of Service	Interactive Voice (6)
RSVP Parameters	(...)
Traffic Mix(%)	All Discrete
Signaling	SIP

Fig.4: Application Attributes

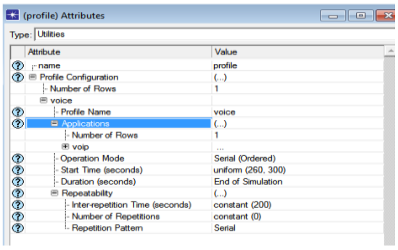

Fig.5: Profile Attributes

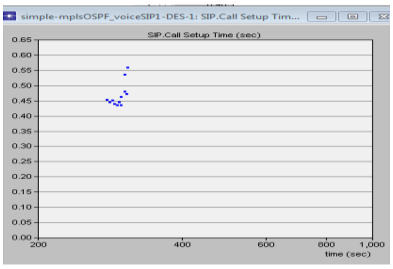

Fig.6 : Call Setup Delay

III. Proposed Work

Since the call setup delay problem existed in SIP based IP telephony over MPLS network we have introduced an IMS based IP telephony over MPLS network, in Figure7we will deploy the IMS Network.we will be setting the properties of IMS servers and In Figure8 the call setup delay is measured and compared by using both SIP and IMS .

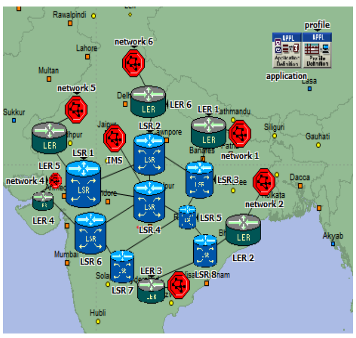

Fig.7: Network Deployment of our OPNET Simulation Scenario

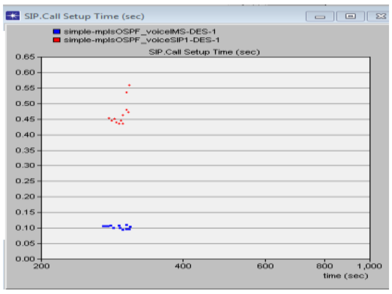

Fig.8: comparison of call setup delay

Table.1: Call setup set up delay values

Call setup delay	*SIP*	*IMS*
	0.48ms	0.10ms

Fig.9: Table for call setup delay

Fig.10: Configuring QoS in the Network

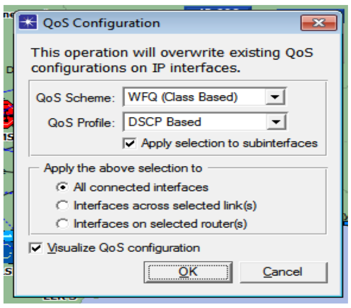

Fig.11:QoS configuration

In figure [10,11] it describes how the QoS properties have been set and the DSCP marking values of the network here we will be using AF33 for high traffic priority in order to know the measure the QoS of the network and the class based weighted fair queuing (CBWFQ) mechanism is used define traffic classes based on match criteria including protocols, access control lists (ACLs), and input interfaces. Packets satisfying the match criteria for a class constitute the traffic for that class. A queue is reserved for each class, and traffic belonging to a class is directed to the queue for that class. . In figure[12,13] traffic is added to the networks by setting QoS properties to one network and the other IMS network does not have QoS and

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further we will compare the results of the network in Figure14 that the QoS is good for the congested QoS IMS Network.

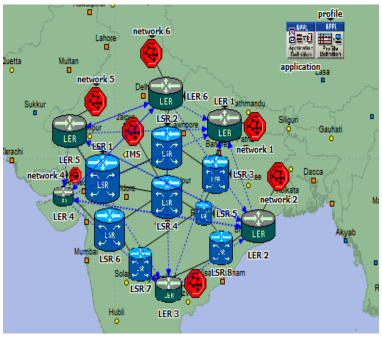

Fig.12 : Congested IMS based Network

Fig.13:Congested QoS IMS based network

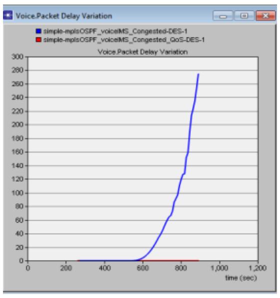

Fig.14 comparison of congested IMS verses Congested QoS IMS Network

IV. RESULTS AND ANALYSIS

The comparative analysis of different voice codec's of different parameters using IMS Network while keeping the simulation environment and attributes same, is discussed in this section. The performance of each codec is evaluated in the network models depending on the delay.

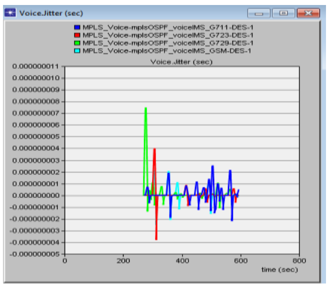

Fig.15: comparison of jitter through different voice codec's

In figure15 jitter is high for G.729 voice codec i.e. 7nsand it is best suitable for GSM as it is less i.e. 1ns

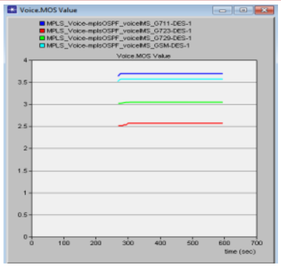

Fig.16: comparison of mean opinion score through different voice codec's

In figure16 the MOS value is worst for G.723 i.e. 2.6 and it is best suitable for G.711 as the MOS value is 3.7 while compared to other voice codec's .

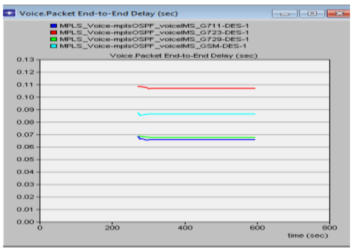

Fig.17: comparison of packet end-to-end delay through different voice codec's

In figure17 packet end-to-end delay is high for G.723 i.e. the end-to-end delay is 110µsand it is best suitable for G.711 as the end-to-end delay is 60µs.

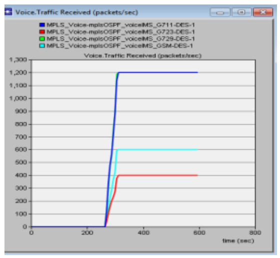

Fig.18: comparison of traffic sent and received through different voice codec's

In figure18 the traffic sent/received for G.729 and G.711 is almost same i.e. the no of packets that are sent /received is 1200 packets and it is good as the no of packets that are sent/ received are more it means the traffic is very high and it is worst for G.723 as the no of packets sent/received is 400 packets.

In figure19 it describes the simulation results of different parameters of different voice codec's.

Table2 : comparison of different voice codec's using different parameters

	G.711	*G.723*	*G.729*	*GSM*
jitter	3.5ns	4ns	7ns	1ns
End-to-end delay	60µs	110µs	70µs	67µs
Mean opinion score	3.8	2.6	3.2	3.7
Traffic sent/received	1200 packets	400 packets	1200 packets	600 packets

Fig.19 Different performance metrics for different voice codec's

V. CONCLUSION

Performance of various VoIP codec's in different networks is analyzed using the OPNET Modeler. A variety of simulations are carried out to get the most effective and efficient results. On the basis of results attained, conclusion for the selection of VoIP codec's in different performance metrics have measured . Depending on the results it is concluded that IMS network performs well irrespective of the VoIP codec being used i.eG.711and call setup delay has been reduced and QoS is Measured. Voice codec can be selected for VoIP communications for SIP network, G.729A codec is the most effective one. In IMS network the best VoIP quality is given while using G711. The quality of G.723.1 codec is observed as it is a low quality codec.

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