IJCSC 0973-7391 Volume 8 • Issue 2 March 2017 - Sept 2017 pp. 160-173 available online at www.csjournals.com

Experimental Evaluation of Quality of Service Parameters over VoIP Network

Amit Chhabra[1], Dr. Amit Verma[2] and Dr. Dheerendra Singh[3]
[1]IKG Punjab Technical University Jalandhar, Punjab,India & Faculty of Engineering, CCET,Chandigarh
[2]Professor, Chandigarh Engineering College, Landran, Punjab, India,
[3]Associate Professor, Chandigarh College of Engineering & Technology,Chandigarh
amitchh80@gmail.com, dramitverma.cu@gmail.com, professordsingh@gmail.com

Abstract: Voice over Internet Protocol (VoIP) is an emerging Internet application with massive potential to be useful dueto alot of advantages it has over previous telephony network. It has been expected that within next few years, a significant percentage of the total communication will be made through VoIP.And in lieu of that, this multimedia application has been chosen by theService providing corridors because of its low cost voice services to the client. There are various factors which affect the quality of this real time service over the wireless network. In this paper, we evaluate the effect of codecs used and the increase in mesh hops in wireless scenario over the quality of voice traffic.The parameters of ITU-T’s E-Model i.e. R-Factor and Mean Opinion Score have been analyzed for different codecs. Different codecs (G.711, G.723.1 and G.729) are selected for evaluating the quality of voice traffic for different number of mesh hops in wireless mesh network. In this paper, Simulation Results for delay modeling of voice traffic over G.711 have been found and are validated using M/D/1 queuing model. The different delays encountered during voice transmission have been analyzed and delay modeling of the voice traffic has been performed over different number of hops. This paper presents an analytical and simulation model for calculating the average one way delay encountered during transmission of voice.Through this research and performance analysis, technicians will be able to make the best selection of codec for providing better services to the customer.

Keywords:-VoIP, Codecs, Queuing Model, M/D/1.

I. INTRODUCTION

In the emerging world of research in wireless communication,Wireless Mesh Network(WMN) provides the most reliable, self-configured, and self-organized way of communication. WMN is architecture through different types of nodes -mesh clients (MCs),mesh routers(MRs), and mesh gateways(MGs)[1]. In the basic architecture of WMN as shown in Figure 1, mesh gateways connect infrastructure network with the internet so that client can access the internet, and mesh routers provide a static mesh backbone which act as an access point. The clients can be stationary or mobile in nature which can communicate with each other through static mesh backbone of mesh access points.Wireless mesh network comprises of such a heterogeneous network structure with both wired and wireless links in which the mesh access points are characterized with minimal mobility[2].

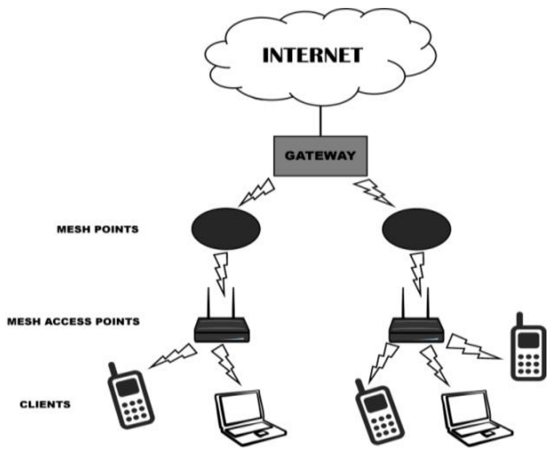

Figure 1. Wireless Mesh Network

WMN can be further classified as Hybrid, Client and Infrastructure WMN. There is a peer to peer connection with each other between the clients in client WMN. The combination of Infrastructure and Client WMNconstitutes Hybrid WMN. The Mesh Routers can be optimized to have Multi Channel Multi Radio interfaces, where each router can have different radio interface each for client and Mesh Gateways [3],[4].The various routing protocols which can be used for communication of traffic in wireless mesh network can be classified into two categories i.e.reactive and Proactive. In proactive routing protocols, the common routing protocols which are broadly used are DSDV, OLSR. Also in Source initiated Reactive Routing protocols AODV, DSR are popular and in Hybrid Routing Protocols, HWMP is commonly used[5],[6].

The remaining sections of the paper arepresented as follows. Section 2 describes about transmission in VoIP network. Section 3 describesVoIP Quality Model followed by detail of different codecs in Section 4. Section 5represents Experimental setup. Section 6represents delay components in a VoIP network, Section 7 represents the proposed queuing model and Section 8 gives the conclusion of this paper.

II. VOIP NETWORK

The main feature which makes VoIP telephony a better choice than traditional telephony is the lower cost.Traditional Telephony requires dedicated circuit between sender and receiver, whether the connection is established or not.VoIP involves the occupancy of the resources only during communication. VoIP system performs resource utilization by allocating bandwidth to some other application when no call is being made. VoIP has revolutionized the complete telephony system operating in the world[7], [8].In a VoIP network, at the Source, the encoder compresses the voice signal into frames.The voice signal frames are transmitted in the form of packets over the network. The RTP/UDP/IP header is added to the payload at the sender side. The header is acquired by thede-packetizer and the voice signal is reconstructed at the receiver side. The jitter bufferminimizes the variation of the arrival time of the packets at the destination as shown in Figure 2.

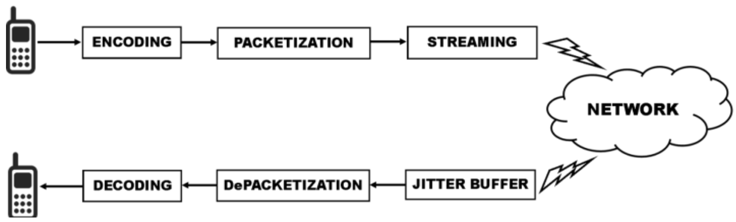

Figure 2. VoIP Network Architecture

III. VOIP QUALITY MODEL

ITU has given various standards which help in evaluating the quality of voice. ITU has recommended both subjective and objective evaluation methods for evaluating the quality of voice[7],[8].One standard defines the quality of voice as tested by various listeners.

The parameter which defines the opinion of these listeners is Mean Opinion Score (MoS).The other standard involvesmeasurement of the QoS Metrics.Objective evaluation methods are machine based; the voice quality is calculated using quantitative distortion parameters between the source and the destination endpoint.

E-Modelis a tool developed by European TelecommunicationStandardsinstitute (ETSI) recommended by ITU G.107for assessingthe quality of voice call[7],[8]. It provides a mathematical algorithmwhich calculates a numerical value called R-Factor. It considers various parameters like the jitter buffer delay, codec delay, compression type of codec and packet loss to represent the numerical figure for voice quality. The Range of the R-factor lies between 0 to 100.The value of R factor can be representedby equation (1).

$$R = R_0 - I_e - I_s - I_d + A_d \qquad (1)$$

Here, R_0 represents the voice quality without any noise, I_e represents the impairment caused by codec, I_s represents the impairment by packet loss, I_d represents the impairment caused by delay and its variation, A_d represents the parameter of advantage that is encountered during voice communication.

The MoS can also be calculated from the R-factor using the following equation[7].

$$MoS = 1 + 0.035R + 7 \times 10^{-6} \times R(R-60)(100-R) \quad (2)$$

Table 1 shows relationbetween MoS andR-factor[8].Both these values can be calculated from each other.R-factor can be used to calculate the subjective evaluation scale i.e. Mean Opinion Score. This paper deals with the Mean Opinion Score (MoS) which is the common subjective voice evaluation method.

Table 1: Relation between R-factor and MoS[8]

Quality Scale	MoS	R-factor
Excellent	4.3-5.0	90-100
Very Good	4.0-4.3	80-90
Good	3.6-4.0	70-80
Fair	3.1-3.6	60-70
Poor	2.6-3.1	50-60
Bad	1.0-2.6	0-50

IV. VOIP CODECS

There are a number of codecs exist in real life, but mainly codecs G.711,G.723.1 and G.729 are mainly utilized for encoding the voice traffic.G.711 is the traditional codec which requires high bandwidth for transmission of voice. Requirement of high bandwidth no doubt makes the voice quality to be optimal but in some application, a much lower bit rate is required for carrying the high amount of traffic within limited capacity. But in some previous years many other new encoding algorithms like G.723.1 and G.729 have been developed which affect the quality metrics of the Voice traffic defined by E-Model. Different research works have been carried for selection of appropriate codec for different networks. Codecs perform according to the environment i.e.it depends upon the network architecture of wired, wireless LAN or WiMax and users density[9].Table 2 shows the different attributes of these codecs.

G.711is thethe traditional codec which is used in Public Switching Telephony Network which does not perform any compression. It uses the technique of Pulse code modulation (PCM) It generates 64 kbps stream, with low CPU Utilization .It generates best audio quality as compared to other codecs due to uncompressing of voice signal. The main limitation of G.711 is that it uses high bandwidth than other codecs[9] .It sends 160 byte payload at bit rate of 64 kbps[9],[10].

Table 2: Codecs Attributes [9]

Attributes	G.711	G.723.1	G.729
Packet Inter arrival time (ms)	20	30	20
Bit rate Kbps	64	5.3	8.0
Payload size (bytes)	80	20	10
IP Packet Size (bytes)	120	60	50

G.723.1 codec belongs to the category of G.723 codec. It transfers the voice signal at bit rate of 5.3 Kbps. This codec utilizes the technique of ACELP i.e. Algebraic code excited linear prediction Algorithm .The other codec

G.723.a works at 6.3 Kbps bit rate. It provides good voice quality in terms of parameters like packet loss and bit errors.

G. 729 codec is a licensed compression technique which is designed to provide good voice call quality without any high consumption of bandwidth[8],[9].Due to compression of signal, It uses less bandwidth than G.711.It follows the procedure on CS-ACELP technique i.e. Conjugate-Structure Algebraic-Code-Excited Linear Prediction algorithm with 8 kbps bit rate.

V. EXPERIMENTAL SETUP

In this research work, simulator QualNet 7.4[11] is used for studying the performance of the scenarios. QualNet can be used in simulating large wired and wireless networks. The simulation was run long enough so that accurate results can be calculated. The simulation was executed 20 times with different seed values. In this, terrain area has been taken as 1500 m*1500 m. Simulation was executed for simulation time of 300 seconds and experimental results are plotted for different codecs (G.711, G.723.1 and G.729). Voice traffic is exchanged between source and destination point. The voice traffic is modeled with the codecs G.711, G.723.1 and G.729.Table 3 shows the different Simulation Parameters.

Table 3: Simulation parameters

Parameters	Value
No. of mesh hops	1,3,5,7,9
Node Position	Random
MAC\PHY	802.11
Area	1500 m*1500 m
Traffic Type	VoIP
Simulation Time	300 sec
Voice Codecs	G.711, G 723.1 and G.729
Traffic Model	Exponential On-OFF
Routing Protocol	AODV
Application layer	VoIP
Transport layer	UDP

Firstly, different Scenarios with hops 1,3,5,7 and 9 are simulated over codec G.711. The Experimental Result in the figure 3 shows that the delay increases exponentially with the increase in hops. Figure 3 shows the impact of change in mesh hops on average one way delay incurred during transmission of voice traffic over mesh hops.

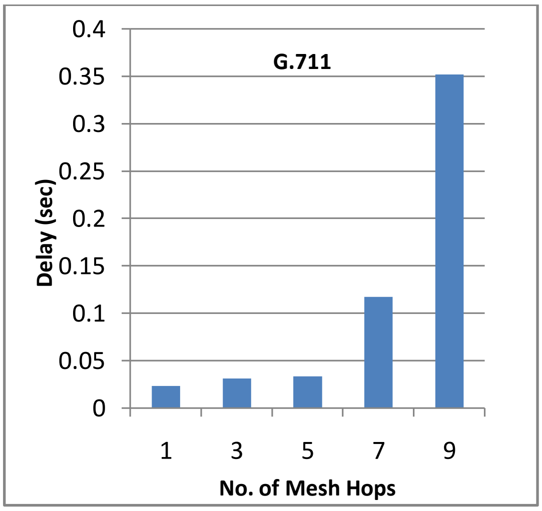

Figure 3.Average one way delay incurred for voice traffic in G.711 at packetization interval of 20ms for different hops

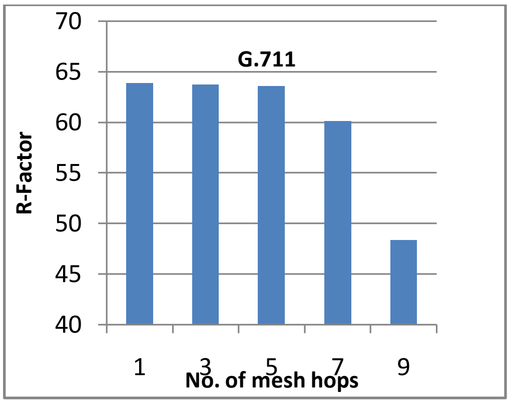

Figure 4. R-Factor of VoIP in G.711 at packetization interval of 20ms for different hops

Similarly, as shown in Figure 4, the R-Factor which has the value between 0 to 100 also decreases in an exponential manner from 63.8 at 1 hop to 48.3 at 9 hops.As represented in the Figure 5, the MoS of Voice traffic which is proportional to the R-Factor also decreases when the number of mesh hops is increased from 1 to 9 during the transmission of voice traffic over codec G.711.

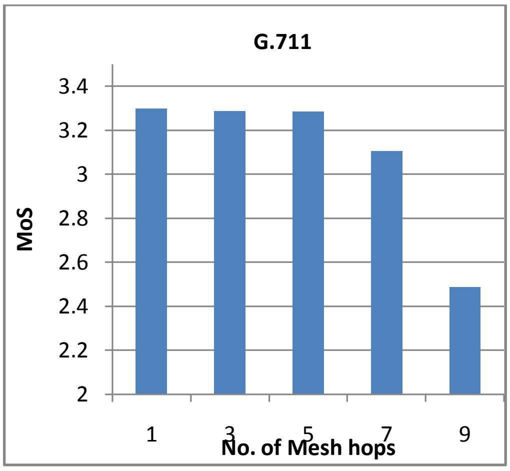

Figure 5.Average MoS of VoIP over G.711 at packetization interval of 20ms for different hops.

Similarly when Codec G.723.1 is used as a compression algorithm for the voice traffic, the delay increases with the increase in hops but not up to the extent as G.711 reaches. Figure 6 shows the increase in delay for the codec G.723.1.

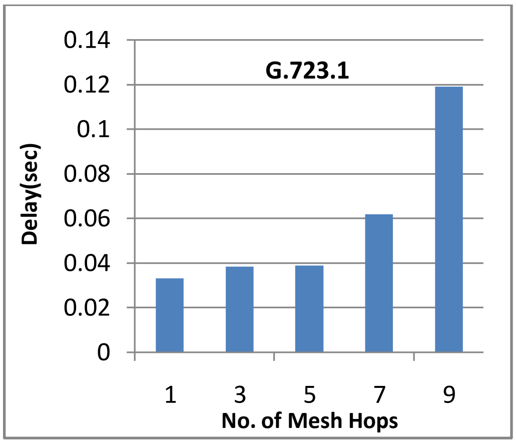

Figure 6. Average one way delay incurred for voice traffic in G.723.1 at packetization interval of 30ms for different hops

Similarly as shown in the figure 7 and 8, the Mean opinion Score and R-Factor decreases with number of hops respectively. Both the values i.e. Mean Opinion Score and R-Factor are correlated with each other and can be calculated from each other. From the calculated value of Mean opinion Score from the Simulator, R-factor can be calculated using equation (2) as mentioned above in the paper.

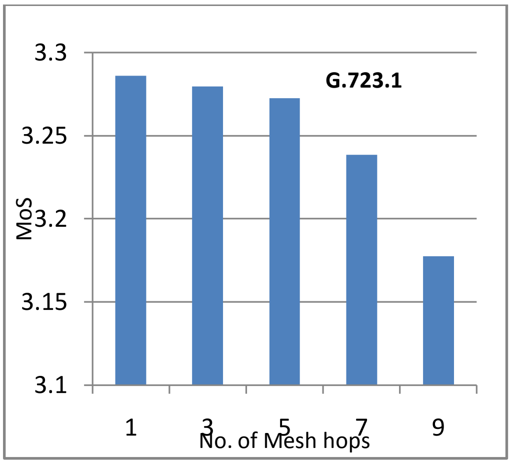

Figure 7Average MoSof VoIP overG.723.1 at packetization interval of 20ms for different hops

R-Factor is correlated with the subjective evaluation method i.e. Mean Opinion Score. After the calculation of the Mean Opinion Score from the simulator, R-Factor can be calculated from the Mean opinion score as represented in the Figure 8.

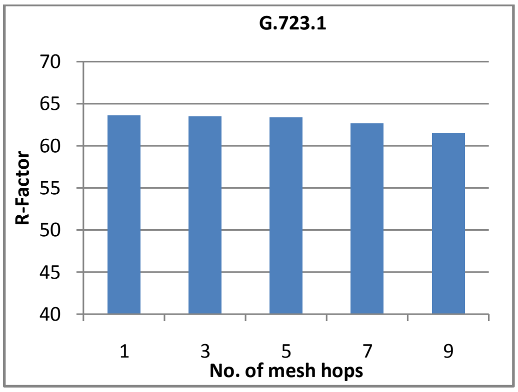

Figure 8.R-Factor of VoIP over G.723.1 with packetization interval of 30ms for different hops

Further, while transferring the voice traffic over G.729 codec, it performs intermediate between G.711 and G.723.1.

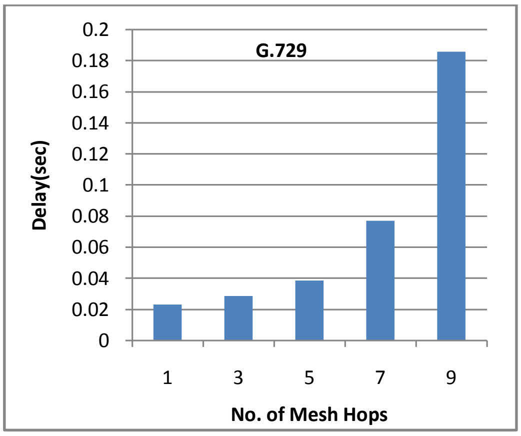

Figure 9. Average one way delay incurred for voice traffic in G.729 at packetization interval of 20ms for different hops

The voice traffic while encoding with G.729 generate delay which is between the delays generated by other two codecs at the respective mesh hops.As shown in Figure 9, mouth to ear delay does not show any major change upto 5^{th} hop, but from 7^{th} hop, there is major change in delay. Earlier, different QoS metrics have been analyzed using different hybrid codecs. While comparing the performance of these codecs, G.729 codec has shown lesser delay and delay variation than G.711 codec due to the reason that G.729 is already compressed, because of which less processing time is required.

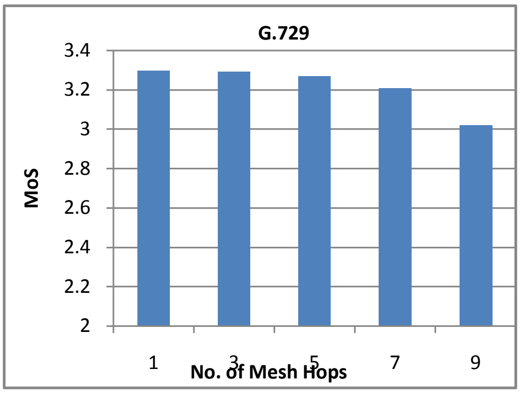

Figure 10 Average MoS of VoIP overG.729 at packetization interval of 20ms for different hops

The MoS as shown in Figure 10 and R-Factor in Figure 11 which are proportional to each other, also show the decrease in their value from 3.3 to 3.0 and from 63 to 58 respectively.

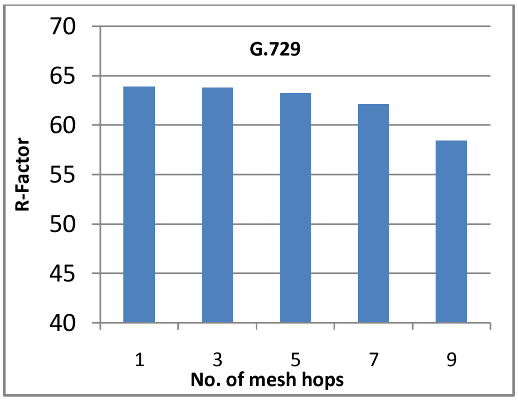

Figure 11R-Factor of VoIP over G.729 with packetization interval of 20ms for different hops

While making comparison between different compression algorithms, as G.711 is having high bit rate (64 Kbps), voice payloads will be higher in case of G.711. Due to its high payload size, the chunk of voice signal lost will be high. So it causes more average delay as shown in the Figure 12, it can be seen that G.711 makes more delay to the voice traffic with increase in mesh hops.

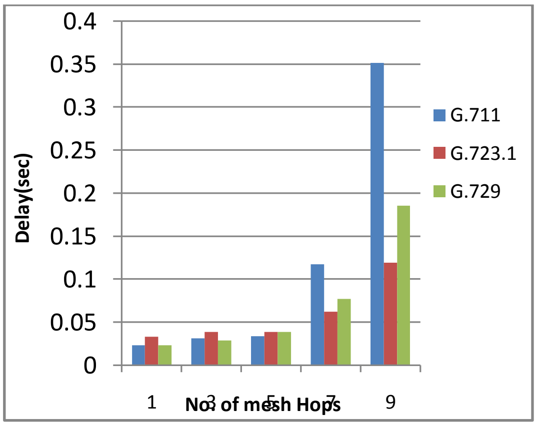

Figure 12. Average Delay incurred using different Codecs

Figure 12 indicates the End-to-End delay for the voice codec over G.711 is the highest when the mesh hops reaches to 9 hops. Initially, as the bit rate of G.711 is high i.e. 64 Kbps, the quality of the voice over G.711 will be in a better

IJCSC 0973-7391 Volume 8 • Issue 2 March 2017 - Sept 2017 pp. 160-173 available online at www.csjournals.com

condition. But as the number of hops increases, the quality of voice in terms of Average delay varies with the number of hops. When distance between endpoints increases, packets sent at the source are not received at the final destination. When the voice payload is higher, loss of packets in terms of information will be high. The results which are shown in Figure 12 are due to the large packet size (160 bytes) and transfer rate of G.711.The codec suffers highest delay than G.723.1 and G.729 due to its largest packet size of 160 bytes. The values of average delay in case of G.723.1 and G.729 codecs is less than codec G.711 due to the reason that the G.723.1 and G.729 are already compressed, so less processing time is required for these codecs when the communication between sender and receiver is started. As shown in Figure 13 and Figure 14, G.723.1 and G.729 are having acceptable value of MoS and R-factor when hops reaches to 9, but the value of Mean opinion Score and R-Factor decreases below 2.6 and 50 respectively when codec G.711 is used, which is very poor quality of voice according to the ITU recommended standards.

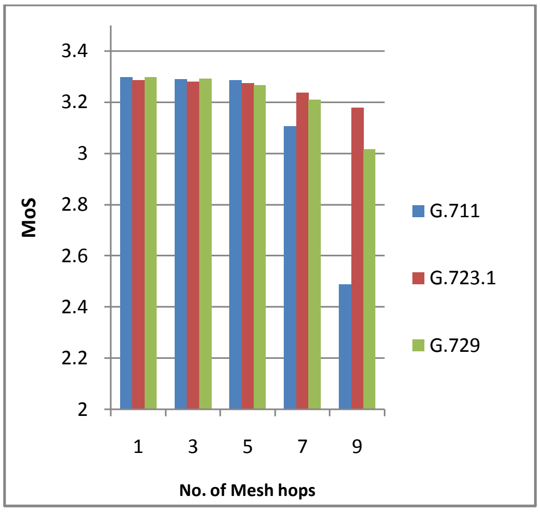

Figure 13. Average MoS incurred using different codecs

R-Factor and Mean Opinion Score represent quality for a voice call, which is again proportional to the delay and packet loss. Initially, all codecs behave in almost similar manner in case of different QoS metrics, but with increase in hops, MoS and R-Factor for G.711 shows minimum value when mesh hops are increased up to 9, it is due to the fact that at 9th hop the delay and packet loss for G.711 is at maximum value, which will cause the quality of voice to go down. The simulation results clearly indicate that correct selection of voice codec is the most important factor for VoIP services over mesh networks. During the transmission of audio traffic with hops up to 5, all the codecs provides almost same quality of voice (MoS value, delay and R-factor) in a scenario. But as the hops increases in the scenario, the parameters of the voice quality also changes for every codec.In the first experiment using G.711, VoIP traffic was generated after every 20 ms of the packetization interval. Since, the packet size of G.711 is larger; it has higher possibility of suffering from packet drop as compared to G 723.1 thus having higher delay in comparison of G 723.1 and G.729.

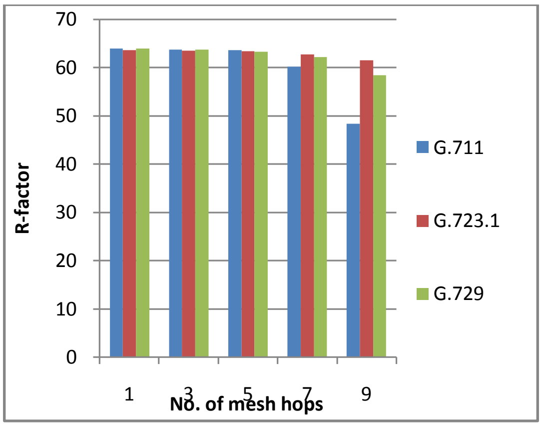

Figure 14. R-Factor incurred for voice traffic using different codecs

So, with the increase in the hops in the network, the delay in G.711 will also increase exponentially and is more than other codecs.Due to lowest bit rate and small size of packet in case of codecs G.723.1 and G.729, the traffic gains small delay.

VI. DELAY COMPONENTS IN A VOIP NETWORK

The total delay incurred from source to destination for a voice signal comprises of 3 key components of delay i.e. delay at source (encoding and packetization delay), delay in network (propagation, queuing and transmission delay) and delay at destination (jitter buffer and depacketization delay)[12].As discussed in different papers, Propagation Delay, Packetization Delay, Network Delay and transmission Delay, etc.constitutethe overall average delay.The paper[12] has defined various delay components and has formulated a mathematical model for average one way delay.

Encoding delay: It is basically the time taken by coder to encode the voice signal.

Packetization Delay (D_{pack}): It refers to the amount of time to build a packet from multiple frames of voice signal. It depends upon the number of frame blocks to be transmitted in one packet. The packatezition refers to accumulating voice signal into a packet which ranges approximately from 10 to 30 ms.Equation (3) represents the packetization delay[12]:

$$D_{pack}(ms) = \frac{PS(bytes)*8(bits/byte)*1000\,(ms/\sec)}{CBw(bps)} \qquad (3)$$

Where $D_{pack}(ms)$ refers to packetization interval or the amount of voice which is digitally encapsulated in each IP packet, $PS(bytes)$ is the Payload size of voice datagram, and $CBw(bps)$ refers to the codec bit rate.

Transmission Delay (D_{trans}): The time for transmission of all the bits of frame over physical media. The transmission delay depends upon the link interface speed. More will be speed of link interface, lesser will be the serialization delay or transmission delay.It refers to the ratio of the size of Frame in bits to the link speed[12],[13].The equation(4) represents the formula for calculation of transmission delay.

$$D_{trans}(ms) = \frac{(PS(bytes))*8(bits/byte)}{LS(bps)} \qquad (4)$$

Where PS=Payload size, LS=link speed.

Queuingdelay (D_{queue}): When packets being sent is more than the packets being processed at the receiving endpoint, causes the queuing delay to occur. It mainly occurs at the intermediate routers through which voice traffic is transferred. This lengthy period of queuing delay is a variable delay which can cause decrease in the voice quality. It is the main component in the delay which can make voice quality to an unacceptable state[12],[13].

Propagation delay(D_{prop}): The time taken by packets to transverse through telecommunication infrastructure. It depends on the number and condition of routing equipment's in network. It depends upon the number of hops, voice signal has to transverse.Propagation delay[13] can be represented by equation(5).

$$D_{prop}(ms) = \frac{D*1000(ms)}{V} \qquad (5)$$

Where D (m) is the distance and V (ms^{-1}) is the velocity of the light in optical fibre communication.

Playout Buffer Delay (D_{jitter}): To compensate the jitter in arrival rate of the voice packets, a playout buffer has to be introduced at receiving point. Unfortunately, it adds to the total delay. Setting size of the jitter buffer as very low or high can make negative effect on the quality of VoIP. There are also different methods which can optimize the quality of VoIP. Packet Aggregation is one of the method which can improve the quality of VoIP but can also lead to increase in aggregation delay[15],[16].

VII. QUEUING DELAY MODEL OF VOIP TRAFFIC

Inprevious research work [17],mainly all the delay components, the method of their generation and their mathematical modeling has been explained.One more method of delay modeling has been analyzed using M/D/1 and M/D/2 technique and it has beenvalidated with real time emulation[18].Various queuing techniques i.e.,FIFO queuing, Weighted-Fair queuing and Priority queuing are analyzed on various parameters.While comparing weighted fair queuing shows better results than othertechniques [19].Also, in the previous research, mathematical model has been proposed based on the use of nonlinear differential equation. In this, different queuing systems i.e.M/D/1, M/M/1and M/E/1 with queue utilizations are analyzed for different packet sizes [20].

In the research work, an analytical model M/D/1[21]has been proposed for estimating the closed form expression for average delay incurred during transmission of voice traffic from source to the destination using codec G.711.

a)Theoretical analysis

In this, following assumptions have been made in wireless scenario:

Arrival Rate (λ) represents the arrival rate

Service time ($T_{service}$ **)** is identically distributed.

There is **one server.**

Number of wireless mesh hops (k): Hops between sender and reciever are varied from 1 to 9.

We consider wireless mesh network with Line topology as shown in the Figure 15.

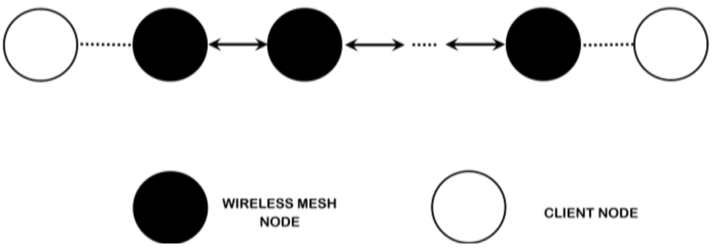

Figure. 15: Wireless mesh network with line topology

b)Establishment of Queuing Model

This paper uses M/D/1 queuing model [21],[22] for studying the behavior of voice traffic in a wireless mesh network.Total waiting time encountered by the packets ina network will be addition of the waiting times encountered by voice packets at all the hops along the network [23]. In equation (6), T_i represents the waiting time of voice packets at ith hop. T_{wait} describes the total waiting time.

$$T_{wait} = T_1 + T_2 + \text{----} T_n \qquad (6)$$

Now, correlating the value of arrival and processing of voice packets with M\D\1 model, here $T_{service}$ represents the service time of the gateway or the destination client in a wireless mesh network.For anExponential arrival and deterministic service distribution[21], equation (7) represents the waiting time as:

$$T_{wait} = \frac{\lambda (T_{service})^2}{2(1-\lambda T_{service})} \qquad (7)$$

Where, T_{wait} is the expected waiting time at the buffered queue.Applying the concept of Exponential arrival and deterministic service distribution ona Wireless Mesh Network with a linear topology as shown in the Figure 16 having k meshhops [24], the net arrival rate from source to destination will keep on increasing fromλ to$k\lambda$and the net arrival rate at the gateway destination will become $k\lambda$ .i.e. the Eq. (7) can be changed to Eq. (8).

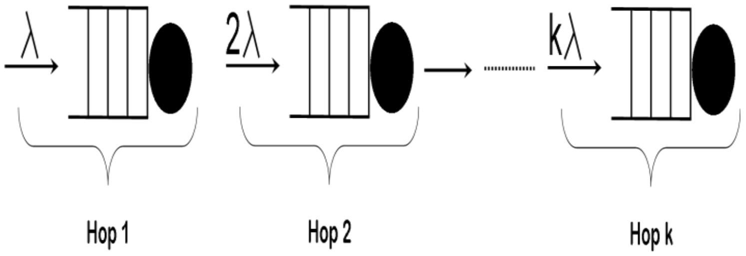

Figure 16.Arrival and Service distribution of voice traffic with k mesh hops.

$$T_{wait} = \frac{k\lambda (T_{service})^2}{2(1-k\lambda T_{service})} \qquad (8)$$

Now, the total average packet Queuing delay (D_{queue})is the sum of Servicetime ($T_{service}$) at the Gateway andWaiting Time(T_{wait}) inQueue. So, the Queuing delay (D_{queue}) can be represented in equation (9) as:

$$D_{queue} = T_{service} + T_{wait} \qquad (9)$$

Now, substituting the values of equation (8) in equation (9) will result in equation (10)

$$D_{queue} = T_{service} + \frac{k\lambda (T_{service})^2}{2(1-k\lambda T_{service})} \qquad (10)$$

While plotting the equation (10) in Figure 17, shows the increase in queuing delay with the increase in hops.Number of k mesh hops is varied from 1 to 9.In the Figure 17, the queuing delay always increases with increase in mesh hops.

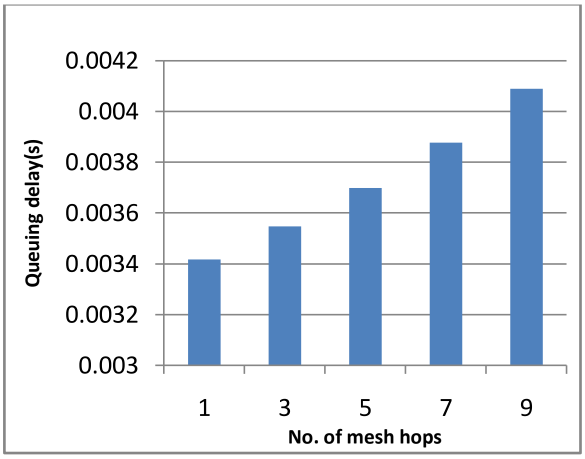

Figure 17.Queuing Delay increaseswith increase in mesh hops at 100 ms of packetization interval.

c)Proposed Model for calculation of Analytical Average Delay for Voice Traffic

Following is the proposed model for calculating analytical average delay for Voice Traffic

Assuming k hops between the endpoints, λ *as the arrival rate,* $T_{service}$ *represents the service time.*

1. *Define a Network with k hops with Source and Destination with the Position Specification*

2. *Define initial values of arrival rate* (λ) *and service rate* (μ) *for voice packets with RTP/UDP/ IP Header in form of packets per second (pps).*

3. *Define Queuing delay for voice traffic over wireless mesh network with k hops by assuming network as M\D\1 model. From the equation (10), queuing delay can be represented as:*

$$D_{queue} = T_{service} + \frac{k\lambda(T_{service})^2}{2(1-k\lambda T_{service})}$$

4. *For all mesh hops, define Propagation delay* (D_{prop}) *and Transmission delay* (D_{trans})

Define D_{pt} *as the sum of Propagation delay* (D_{prop}) *and Transmission delay* (D_{trans}).

Initially the value of $D_{pt} = 0$;

For(each mesh hop $\leq k$*)*

{

$$D_{pt} = D_{pt} + D_{prop} + D_{trans} \quad (11)$$

}

The value of propagation delay (D_{prop}) *can be analyzed from the parameters of the Simulation.*

Transmission delay (D_{trans}) *in the VoIP network can be calculated from the given formula*

$$D_{trans}(ms) = \frac{(PS(bytes)) * 8(bits/byte)}{LS(bps)}$$

Where PS = Payload Size, LS = Link speed

5. *Total Average delay* (D_{total}) *will be addition of all delay incurred at sender side, network side and at destination. It can be represented by equation (12)*

$$D_{total} = D_{pack} + D_{queue} + D_{jitter} \quad (12)$$

The overall average delay is calculated from source to destination in equation (12) is the sum of D_{pack} which has been taken as 100ms for the codec G.711;here packets are aggregated to be transferred at 100ms rather than by default packetization interval. D_{queue} can be obtained from the equation (10).During the voice transmission, no jitter buffer has been taken i.e. D_{jitter} has not been considered.Substituting all the values in the equation (12), the analytical average delay can be calculated. The Figure 18 represents the Validation of the theoretical results i.e. the simulation and analytical results closely match with each other.

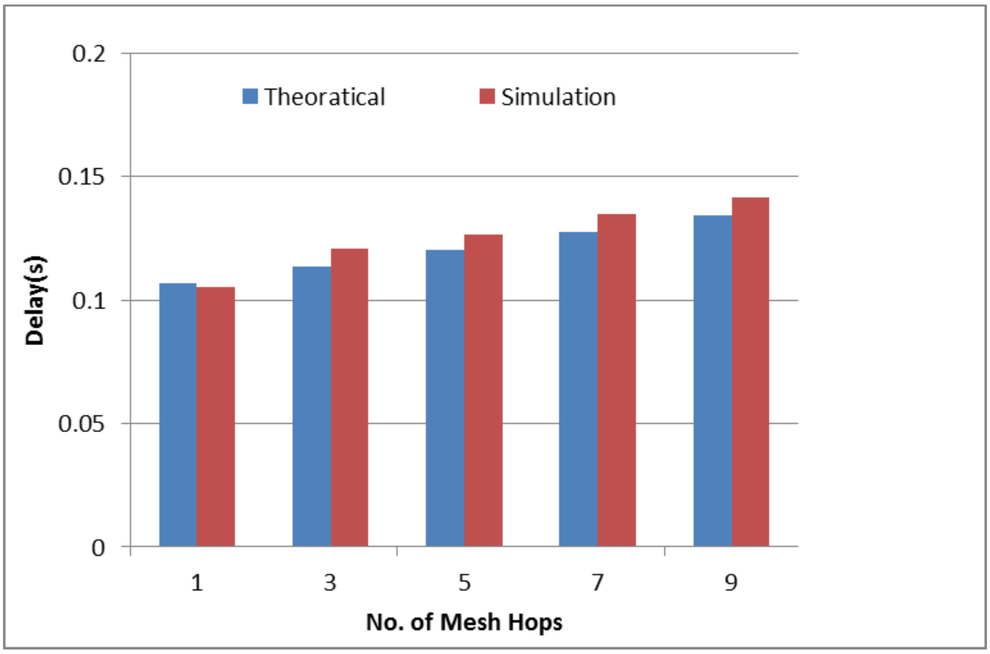

Figure 18.Validation ofaverage delay results of VoIP traffic with packetization interval of 100 ms

VIII. CONCLUSION

The paper focuses on different QoS parameters of VoIP affected by change in the number of mesh hops over different codecs. The results represented in the paper showbehaviorof different audio codecsin wireless mesh scenario.According to the obtained simulation results,MoS and R-factor parametersof VoIP decreases with increase of scaleof number of hops in a network.At the end, wireless mesh network is modeled as a queuing model, in which client at the source can be assumed as generator of voice traffic and gateway or the client at the destination can be modeled as server. The results of queuing delay and the overall average one way delay is calculated by modeling the complete network as M\D\1 model. Both the results i.e.analytical delay and simulation delay results are calculated and matched with each other to determine the validity of the result.

REFERENCES

1. Akyildiz IF, Wang X. A survey on wireless mesh networks. IEEE Communications magazine. 2005 Sep;43(9):S23-30.
2. Benyamina D, Hafid A, Gendreau M. Wireless mesh networks design—A survey. IEEE Communications surveys & tutorials. 2012 May;14(2):299-310.
3. Uludag S, Imboden T, Akkaya K. A taxonomy and evaluation for developing 802.11-based wireless mesh network testbeds. International Journal of Communication Systems. 2012 Aug 1;25(8):963-90.

4. Jahanshahi M, Dehghan M, Meybodi MR. On channel assignment and multicast routing in multi–channel multi–radio wireless mesh networks. International Journal of Ad Hoc and Ubiquitous Computing. 2013 Jan 1;12(4):225-44.
5. Piechowiak M, Zwierzykowski P, Owczarek P, Wasłowicz M. Comparative analysis of routing protocols for wireless mesh networks. InCommunication Systems, Networks and Digital Signal Processing (CSNDSP), 2016 10th International Symposium on 2016 Jul 20 (pp. 1-5). IEEE.
6. Daas A, Mofleh K, Jabr E, Hamad S., Comparison between AODV and DSDV routing protocols in mobile Ad-hoc Network (MANET). InInformationTechnology:Towards New Smart World (NSITNSW), 2015 5th National Symposium on 2015 Feb 17 (pp. 1-5). IEEE.
7. Goode B. Voice over internet protocol (VoIP). Proceedings of the IEEE. 2002 Sep;90(9):1495-517.
8. Singh HP, Singh S, Singh J, Khan SA. VoIP: State of art for global connectivity—A critical review. Journal of Network and Computer Applications. 2014 Jan 31;37:365-79.
9. Ali MA, Rashid I, Khan AA. Selection of VoIP CODECs for Different Networks based on QoS Analysis. International Journal of Computer Applications. 2013 Jan 1;84(5).
10. Labyad Y, Moughit M, Haqiq A. Performance analysis and comparative study of voice over IP using hybrid codec. InComplex Systems (ICCS), 2012 International Conference on 2012 Nov 5 (pp. 1-6). IEEE.
11. QualNet Network Simulator [Online]. Available online at: http://web.scalable-networks.com/qualnet-network-simulator-software (accessed: 2017)
12. Kovacik M, Baronak I, Janata V, Kyrbashov B. Evaluation and Investigation of the Delay in VoIP Networks. Radioengineering. 2011.
13. Kovac A, Halas M. Analysis of influence of network performance parameters on VoIP call quality. Knowledge in Telecommunication Technologies and Optics. 2010:26-30.
14. Meeran MT, Annus P, Le Moullec Y. The current state of voice over internet protocol in wireless mesh networks. InAdvances in Computing, Communications and Informatics (ICACCI), 2016 International Conference on 2016 Sep 21 (pp. 2567-2575). IEEE.
15. Di Stasi G, Karlsson J, Avallone S, Canonico R, Kassler A, Brunstrom A. Combining multi-path forwarding and packet aggregation for improved network performance in wireless mesh networks. Computer Networks. 2014 May 8;64:26-37.
16. Jung JY, Kang HS, Lee JR. Performance evaluation of packet aggregation scheme for VoIP service in wireless multi-hop network. Ad Hoc Networks.Elsevier, 2013 May 31;11(3):1037-45.
17. Gupta V, Dharmaraja S, Arunachalam V. Stochastic modeling for delay analysis of a VoIP network. Annals of Operations Research. 2015 Oct 1;233(1):171-80.
18. Voznak M, Halas M. Delay variation model with RTP flows behavior in accordance with M/D/1 Kendall's notation. Advances in Electrical and Electronic Engineering. 2010 Jan 1;8(5):124.
19. Rashed MM, Kabir M. A comparative study of different queuing techniques in VOIP, video conferencing and file transfer. Daffodil international university journal of science and technology. 2010;5(1):37-47.
20. Lebedenko T, Ievdokymenko M, Ali SA. Research of Influence Flow Characteristics to Network Routers Queues Utilization,1st International Conference on Advanced Information and Communication Technologies,2015.
21. Zukerman M. Introduction to queueing theory and stochastic teletraffic models. arXiv preprint arXiv:1307.2968. 2013 Jul 11.
22. Khodadadi A. A Comparison between M/M/1 and M/D/1 using models of queuing delay. International Journal of Biotechnology and Allied Fields. 2016;4(2):215-20.
23. Verma PK, Wang L. Voice Over IP Networks: Quality of Service, Pricing and Security. Springer Science & Business Media; 2011 Jan 12.
24. Wu X, Liu J, Chen G. Analysis of bottleneck delay and throughput in wireless mesh networks. InMobileAdhoc and Sensor Systems (MASS), 2006 IEEE International Conference on 2006 Oct (pp. 765-770). IEEE.