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Analysis of VoIP Traffic in WiMAX Environment

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Abstract— Worldwide Interoperability for Microwave Access (WiMAX) is currently one of the hottest technologies in wireless communication. It is a standard based on the IEEE 802.16 wireless technology that provides a very high throughput broadband connections over long distances. In parallel, Voice Over Internet Protocol (VoIP) is a new technology which provides access to voice communication over internet protocol and hence it is becomes an alternative to public switched telephone networks (PSTN) due to its capability of transmission of voice as packets over IP networks. A lot of research has been done in analyzing the performances of VoIP traffic over WiMAX network. In this paper we review the analysis carried out by several authors for the most common VoIP codec's which are G.711, G.723.1 and G.729 over a WiMAX network using various service classes. The objective is to compare the results for different types of service classes with respect to the QoS parameters such as throughput, average delay and average jitter.
Keywords— WiMAX, VoIP, BE, UGS, rtPS,nrtPS

I. INTRODUCTION

WiMAX, the acronym for Worldwide Interoperability for Microwave Access is a set of technical standards based on IEEE 802.16 standard. It provides wireless connection over long distances at high speed. Unlike DSL (Digital Subscriber Line) or other wired technologies, WiMAX uses radio waves and can provide point-to-multipoint (PMP) broadband wireless access. In parallel, Voice over Internet protocol (VoIP) technology is the biggest revolution in communication technology. It replaces the traditional telephone services like PSTN and offers free, long distance calls over the internet. VoIP are highly delay intolerant and need a high priority transmission. In this paper, we evaluate the performances of the most common VoIP codec's using Unsolicited Grant Service (UGS),Real-Time Polling Service (rtPS) ,Non-Real Time Polling Service (nrtPS) and Best Effort (BE) service classes.NS-2 simulator(or MATLAB) is used to analyze the QoS parameters. Our objective is to analyze different WiMAX service classes with respect to the QoS parameters such as throughput, average delay and average jitter.

II. LITERATURE REVIEW

The simulation study was carried out by [1] to evaluate the performance of VoIP over the WiMAX networks using OPNET modeller. Various parameters such as MOS value, packet end-to-end delays , jitter and voice packets end to end delay were used to estimate the performance of VoIP over WiMAX. Three voice codecs that is G.711, G.723 and G.729 were simulated in order to find the most appropriate voice codec for VoIP traffic over WiMAX network in [1].

In this paper, the performance of VoIP over WiMAX was measured and analysed in terms of crucial parameters like MOS, Jitter, end-to-end delay and packet sent and packet received. Three voice codec's that is G.711, G.723 and G.729 were simulated in order to find the excellent voice codec for VoIP traffic over WiMAX network. The simulation analysis showed that VoIP performed best under the G.711 codec as compared to the other codec's that are G.723 and G.729 [2].

In this article a survey is conducted on all the aspects that have the greatest bearing on voice quality. Beginning with the QoS requirements of voice and methods that can be employed to evaluate the performance of VoIP systems, the survey was continued on voice codec's and header compression techniques. Exhaustive analysis and comparison of different voice codec's and header compression schemes, touching also upon voice activity detection are provided. Then the attention was turned to signaling protocol needed for VoIP. It also shed light to the meaningful issue of security in VoIP networks. In this paper it is concluded that significance been made since the inception of VoIP and Internet telephony will soon be able to provide speech quality at a similar level to the one of public switched telephone network (PSTN) [3].

In this paper the observations that were made are, that under various mobility scenarios that is the variation of packet end-to-end delay and jitter. This work here shows the variation of the above parameter with respect to stationary network in which all the nodes are kept stationary with respect to the mobile network where each node is mobile in nature and has a varying speed. The observation is made that average delay is more in the case of the stationary nodes compared to the mobile nodes in [4].

Measurement of QoS is essential for any Wimax network / broadband wireless communication. In order to ensure that a user- centric broadband experience becomes a reality, the broadband wireless access networks must meet a number of Quality of Service (QoS) parameters, including guaranteed low delay, jitter ,throughput and packet loss. Today in broadband wireless access (BWA) the perception is that as adoption enhances , so does the need for guaranteeing a good QoS. The issue of QoS, therefore, has become a critical area of concern for suppliers of broadband wireless access (BWA) equipment and their customers too. This paper helps in analysing various essential Wimax QoS parameters which are critical in determining the performance of a Wimax network. A very low value of delay ,packet loss and jitter is achieved , whereas a very high average value of throughput and packet delivery is obtained using AODV protocol. Therefore in this paper it helps us in understanding these critical QoS parameters which helps in improving performance for a given Wimax network[5].

In this paper the performances of the High quality video traffic over a WiMAX network using various service classes has been observed. To analyze the QoS parameters, the WiMAX module developed based on famous network simulator NS-3 is used.Different parameters that determine QoS of real life usage scenarios and traffic flows of applications is analyzed. The target is to compare different types of service classes with respect to the QoS parameters, such as, throughput, packet loss, average delay and average jitter[6].

In the normal process, the voice connection has a load of 96 Kbps which results in the throughput of 64Kbps. This discrepancy between throughput and load is causing unacceptable delays for voice traffic. To solve this problem, we have used the Improve Voice scenario. We re-sized the UGS service class from 64Kbps to 96Kbps data rate. The throughput is more drawn out in time than the load, suggesting some queuing. The Unsolicited Grant Service (UGS) connection was sized for 64Kbps, but it receives 96Kbps in load. The difference is made up by overhead between the application layer and the MAC layer as described in[7].

In this paper, the BE, rtPS and UGS service classes using different VoIP codecs have been simulated and analysed in terms of average jitter,throughput and average delay.The rtPS service class comes out to be better than BE service class for average jitter. Otherwise, all the service classes over VoIP codecs under consideration work efficiently when it comes to less than six nodes.In conclusion, it is observed that UGS service class has the best performance parameters serving VoIP. Indeed, UGS service class is designed to handle real-time service flows that generate fixed size packets at regular interval, which is the case for VoIP[8].

III. WIMAX NETWORK ARCHITECTURE

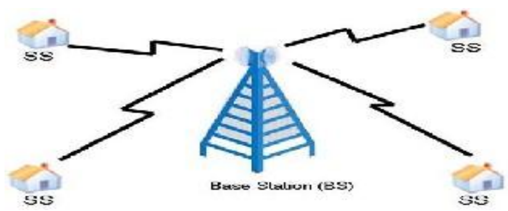


Fig shows the WiMAX network architecture.

The WiMAX network architecture can provide multiple levels of QoS over its classification, queuing, control signalling mechanisms, scheduling, modulation, and routing. It's is a combination of base station (BS) and subscribers station (SS). Above Figure shows the WiMAX network architecture.

IV. QOS SERVICE CLASSES IN WIMAX

WiMAX gives network operators the opportunity to provide the best of services to differentiate their offerings and attract a large range of subscribers. It features a variety of flow types that can be used to optimize performance for voice,data and video.Offering Voice,data,video convergence makes sense for enterprises and service providers alike. For example, effective voice over IP (VoIP) communications needs QoS features that can quickly identify voice traffic and prioritize it to assure high quality audio and service level.

The IEEE802.16d WiMAX standard offers four classifications for the enhancement of traffic:
(1) Unsolicited Grant Service (UGS)
(2) Real-Time Polling Service (rtPS)
(3) Non-Real Time Polling Service (nrtPS) and
(4) Best Effort (BE).
Each of this service class is intended for specific application as described below:

Unsolicited Grant Service (UGS):

UGS is basically intended for Constant-Bit-Rate (CBR) services such as VoIP, which means that achieving low jitter and low latency is very important. At the same time, low percentage of packet drops is feasible. UGS flows are configured to send fixed size packets at recurring intervals with as little jitter and latency as possible.

Real-Time Polling Service (rtPS):

The Real-Time Polling Service (rtPS) on the other hand is designed to support real-time service flows that generate variable size data packets on a periodic basis, such as MPEG video. The service offers real-time,unicast request opportunities, periodic, which meet the flow's real-time needs and allow the Subscriber Station (SS) to specify the size of the desired grant. A major drawback to using this QoS approach is the impact on the overall throughput.

Non-Real-Time Polling Service (nrtPS):

This service class is intended to support non-real-time service flows that require minimum data rate, variable size data , such as file transfer protocol (FTP). This ensures that the service flow receives requests even during network congestion.This is accomplished by offering unicast polls on a regular basis.

Best Effort (BE):

The BE service is intended to support data streams that don't require minimum guaranteed rate, and could be handled on best available basis. Unicast polling requests are not guaranteed in this case, requiring contention requests to be used. BE packets may therefore take a long time to transmit when network is highly congested.

V. VOIP TECHNOLOGY

VoIP Transport System

VoIP uses a combination of protocols for delivering phone data over networks. Various signalling protocols are used like SIP and H.323.These can be regarded as the enabler protocols for voice over IP (VoIP) services. VoIP communications require these signalling systems to setup, control, initiate a session and facilitate real-time data transfer in order to provide clear communications. SIP and H.323 works in conjunction with the Real Time Transport Protocol (RTP) and the User Datagram Protocol (UDP) to transfer the voice stream. Voice data is put in data packets using the RTP protocol. The RTP packets, enclosed inside the UDP packets, are then transferred to the receiver.

VoIP Codec's

RTP and UDP protocols are the logical choice to carry voice when transport control protocol (TCP) favours reliability over timeliness. Voice signals are digitally encoded. This means that each voice signal is converted from digital to analog and back. The analog signal is firstly sampled based on a sampling rate of 8 KHz, 8 bits per sample is the most frequently cases. Next, the output is encoded according to many factors: the compression rate and the framing time or the frames length. Finally, one or more of these frames are encapsulated into an RTP/UDP/IP packet for transmission over the network. All these practices are achieved by one of various audio codec's, each of which vary in the sound quality, the bandwidth required, the computational requirements, encoding algorithm and coding delay. They are as discussed below:

• G.711 is the default standard for all vendors and manufacturers and provides very low processor requirements. This standard digitizes voice into 64 Kbps and does not compress the voice. It performs excellent in local networks where we have ample amount of available bandwidth.

• G.729 is supported by many vendors for compressed voice operating at data rate of 8 Kbps. Excellent bandwidth utilization and Error tolerant with quality just below that of G.711.

• G.723.1 was once the recommended compression standard. It operates at data rate 6.3Kbps and 5.3 Kbps. High compression with high quality audio. Although this standard decreases bandwidth consumption, voice is much poorer than with G.729 and is not very popular for VoIP traffic.

VI. PERFORMANCE PARAMETERS

Throughput:

Throughput is the amount of number of packets effectively transferred in a network, in other words throughout is data transfer rate that are delivered to all terminals in a network. It is measured in terms of packets per second or per time slot.

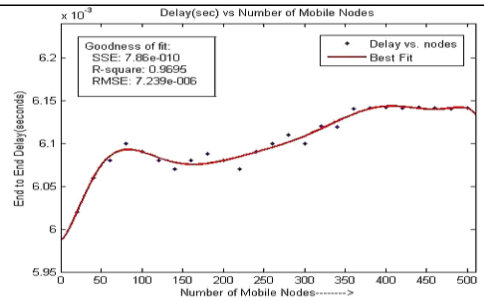
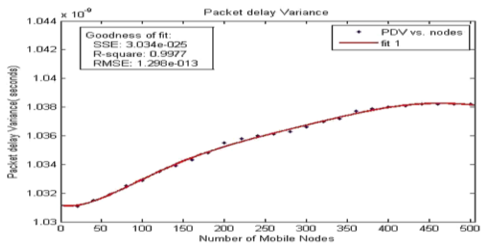
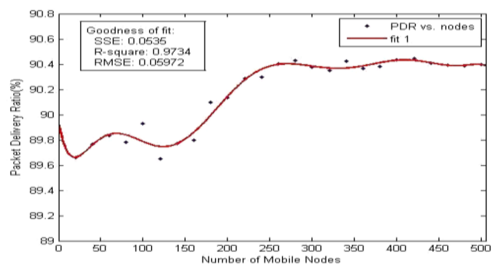
Average Delay:

Delay or latency represents the time taken by a bit of data to reach from source to destination across the network. The main sources of delay can be categorized into: propagation delay, source processing delay, Queuing delay, transmission delay and destination processing delay. Hence, end to end delay which is a measure of elapsed time taken during modulation of the signal and the time taken by the packets to reach from source to destination.

Jitter or Delay variation:

Jitter can be observed as the end-to-end delay variation between two consecutive packets. The value of jitter is calculated from the end to end delay. Jitter reveals the variations in latency in the network caused by congestion, route changes, queuing, etc. It determines the performance of network and indicates how much consistence and stable the network is. The various analysis that are been made by different authors pertaining to performance parameters are as shown below:

Table 1. Performance Analysis of QoS Parameter for WiMAX Networks by Vikram Mehta and Dr.Neena Gupta

Performance parameter and their respective Results	Graphs
Delay: The graph drawn in figure shows that as the number of node increases, the delay increases up to a certain point due to high network traffic but then it becomes constant to up to an average value of 6.15 ms.	
Packet Delay Variance (Jitter): The figure shows the graph between PDV and number of mobile nodes. Here it shows that a very low value of jitter. The value of jitter increases with increase in number of nodes.	
Packet delivery Ratio(PDR): It is the measure of successful delivery of packets.The Graph plotted between PDR and number of mobile nodes is as shown in Fig. With increase in number of mobile nodes, the value of PDR increases but then it actually becomes constant.	

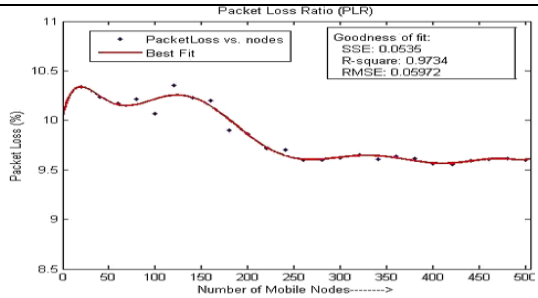
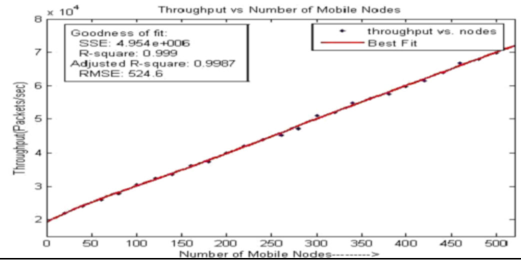
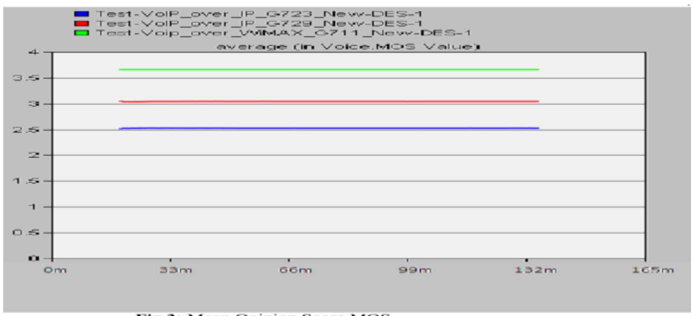
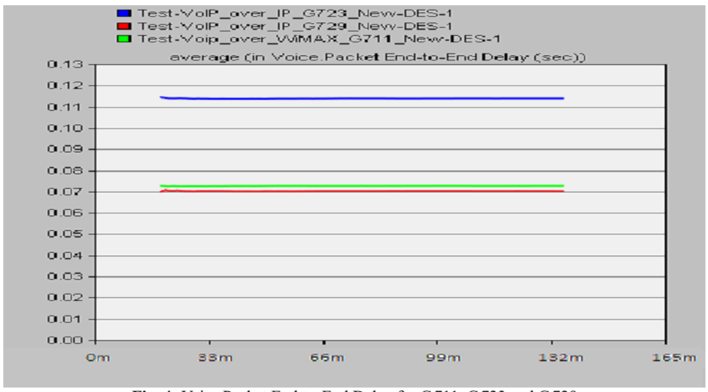
Packet Loss Ratio(PLR): It is actually the measure of number of packets undelivered or lost in the network. The graph is drawn between PLR and number of mobile nodes. It shows that if we increasing the number of mobile nodes, then the value of PLR decreases but it eventually becomes constant after 250 mobile nodes.	
Throughput: The graph between Throughput Vs number of mobile nodes is shown in fig It depicts that as the number of mobile nodes increases, throughput increases as well.	

Table 2.Enhanced Quality of Service in Worldwide Interoperability for Microwave Access Network by Gihad mohammed and Amin Mustafa

Performance parameter and their respective Results	Graphs
MOS: It defines the quality of voice. It is observed that, the MOS significantly increases (above 3.5), indicating improvement of the call quality of VoIP streams and according to this the voice over IP in WIMAX for G.711codec is the best.	 **Fig 3:** Mean Opinion Score MOS
Packet End to End delay: This parameter gives the total voice packet delay. The mean end to end delay is shown for the network. Research have confirm that packet delay of 100ms doesn't do any problem, but if the delay grow up to 150ms the voice signal is damage . The service providers have to guarantee that the delay happen is equal or less than 100ms .and according to figure the delay in G.729 and G.723 gives best result .	 **Fig. 4:** Voice Packet End-to-End Delay for G.711, G.723 and G.729

The last two figure shows Average in voice packet End to End Delay and average in voice jitter according three distances 50,100,200 km and at 50 km low jitter and delay.

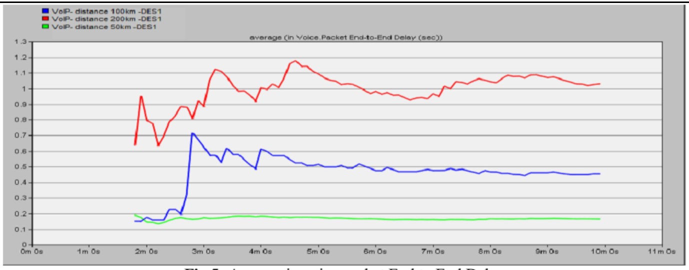


Fig 5: Average in voice packet End to End Delay

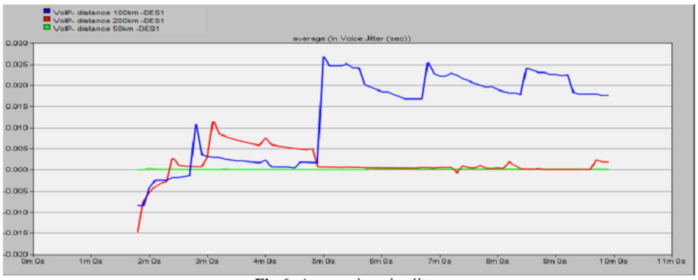


Fig 6: Average in voice jitter

Table 3. Performance Analysis of VoIP Traffic in WiMAX using various Service Classes by Tarik ANOUARI , Abdelkrim HAQIQ

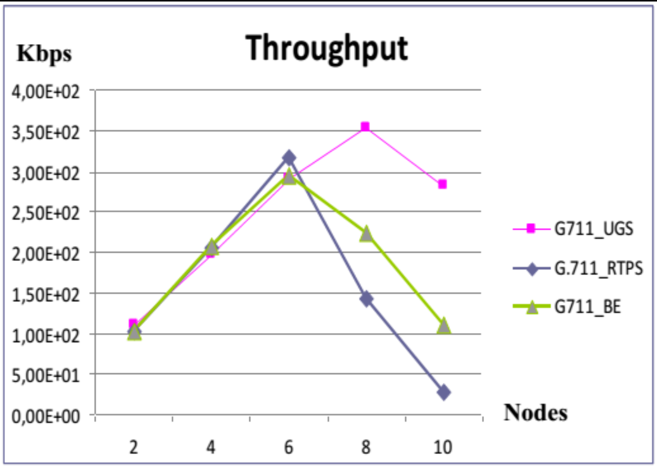
<table>
<tr><th>Performance parameter and their respective Results</th><th>Graphs</th></tr>
<tr><td>Throughput: The graphs show throughput against number of mobile nodes for each codec under various service classes. It is observed that the average throughput increases steadily when the number of nodes increases before reaching six nodes, then it begins to go down. Among the three graphs, from the sixth node, the throughput of the rtPS class decreases faster than the other service classes and has finally the lowest throughput. Throughputs of BE and UGS traffic are very similar except for the G.711 codec for which the UGS service class performed better than BE.</td><td>


Fig 3(a): Throughput for G.711 Codec under various service classes

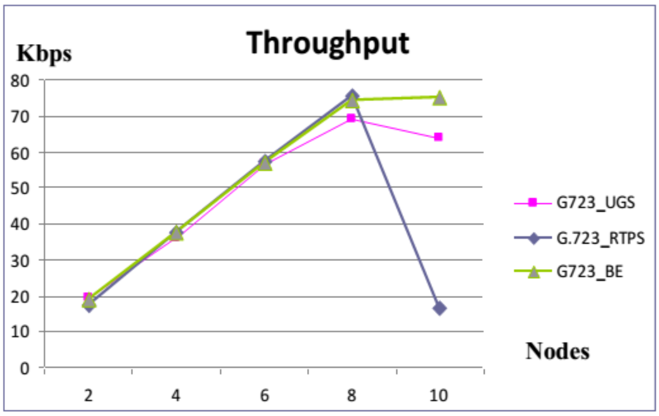


Fig 3(b): Throughput for G.723 Codec under various service classes

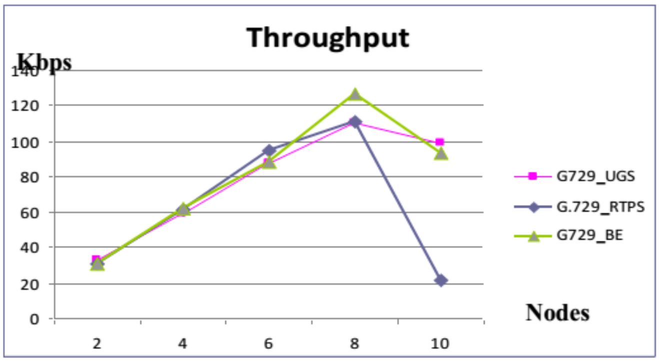


Fig 3(c): Throughput for G.729 Codec under various service classes
</td></tr>
</table>

Average Jitter: BE service class has the highest jitter.
Average jitter of all service classes under simulation increases starting from the sixth node, except for the rtPS class which
decreases from the eight node.
In case of the UGS class, the average jitter does not vary as much as the number of nodes increases. In addition to that, the
value is very small.

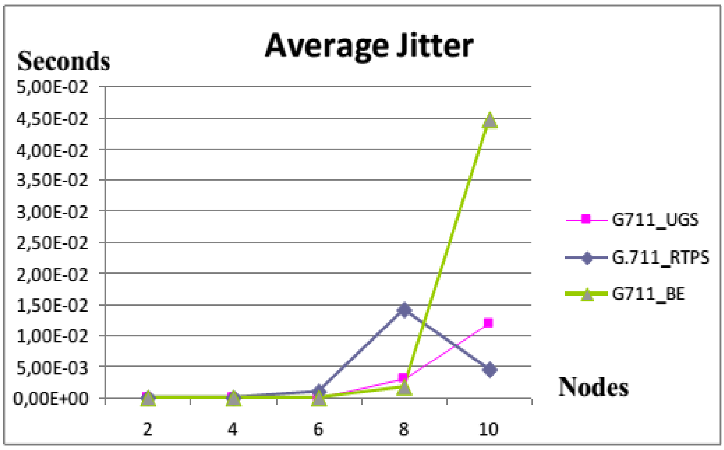


Fig 4(a): Average Jitter for G.711 Codec under various service classes

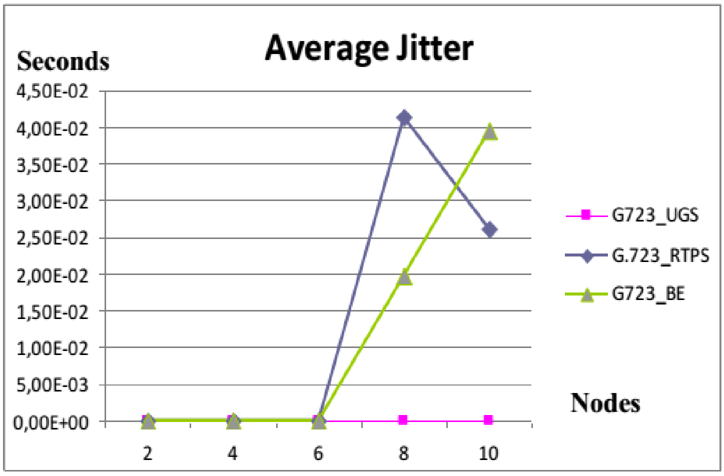


Fig 4(b): Average Jitter for G.723 Codec under various service classes

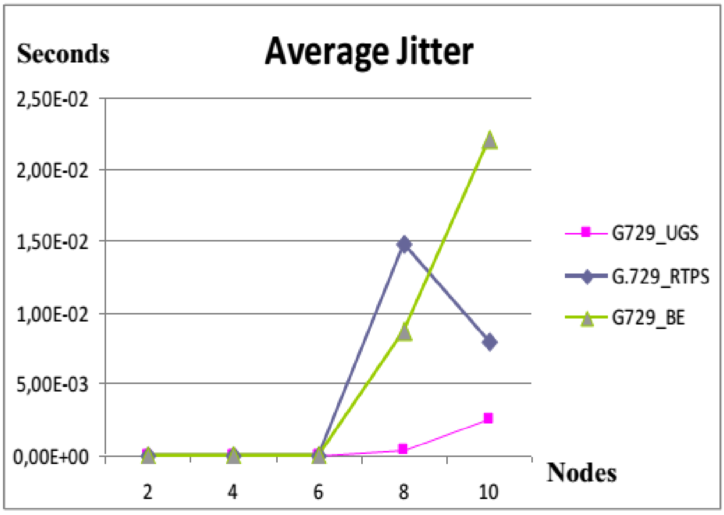


Fig 4(c): Average Jitter for G.729 Codec under various service classes

Average delay: The figure shows the average delay variation of the three service classes. The delay values of BE and rtPS traffic vary similarly with increasing nodes and keep still higher compared with UGS traffic.
From node 2 to 6, average delay is low for the three VoIP codec's, starting from the sixth node, the average delay of rtPS and BE traffic increases sharply. Whereas, the UGS traffic keeps insignificant in comparison to BE and rtPS.

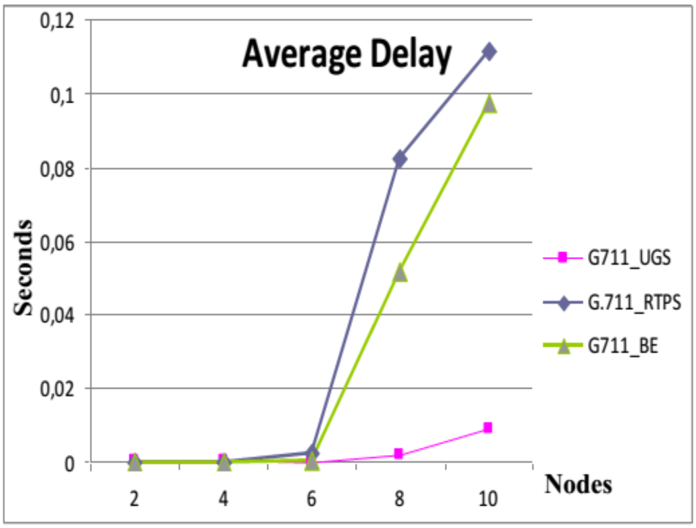


Fig 5(a): Average Delay for G.711 Codec under various service classes

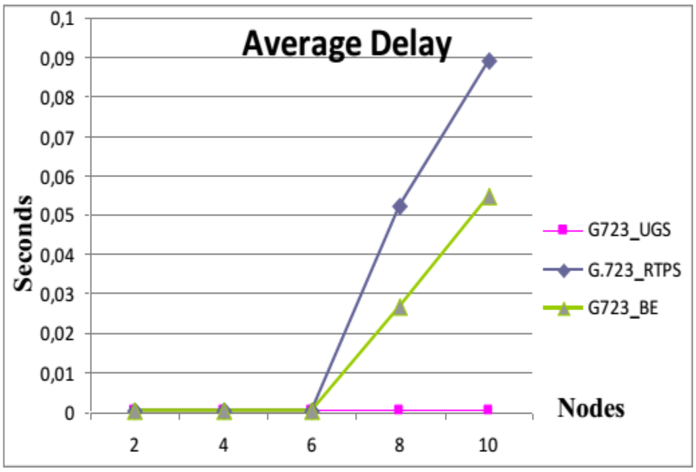


Fig 5(b): Average Delay for G.723 Codec under various service classes

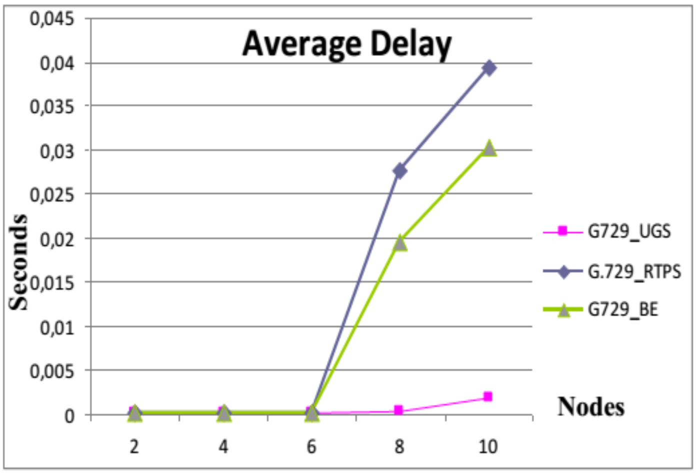


Fig 5(c): Average Delay for G.729 Codec under various service classes

VII. CONCLUSIONS

The results obtained by various authors are presented in this paper. From the above tabulation we conclude that this paper helps in analysing various essential WiMAX QoS parameters which are critical in determining the performance of a WiMAX network. A very low value of delay ,packet loss and jitter is achieved , whereas a very high average value of throughput and packet delivery is obtained[5].From next observation we analyse the parameters such as MOS,packet end- to-end delay, average in voice packet and voice jitter, where in G.711 codec is the best in terms of MOS and G.723 codec is best in terms of packet end-to-end delay[12].Lastly, from [8] we conclude that in this paper, the BE, rtPS and UGS service classes using different VoIP codec's have been simulated and analyzed in terms of average jitter, throughput and average delay. The rtPS service class comes out to be better than BE service class for average jitter. Otherwise, all the service classes over VoIP codec's under consideration work efficiently when it comes to less than six nodes. In conclusion, it is observed that UGS service class has the best performance parameters serving VoIP. Indeed, UGS service class is designed to handle real-time service flows that generate fixed size packets at regular interval, which is the case for VoIP. In future we will simulate the performance parameters of various service classes of WiMAX network using VoIP traffic in MATLAB and NS-2.

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