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QoS of VoIP over WiMAX Access Networks

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Abstract— **Voice over Internet Protocol is a rapid growing technology which enables the transport of voice data over Internet Protocol based networks. Voice over Internet Protocol has become a possible alternative to Public Switched Telephone Network due to its capabilities to transport voice data in the form of digital Internet Protocol packets over the TCP/IP based Internet. In parallel, a remarkable increase is happening in the deployment of IEEE 802.16 standard based WiMAX networks. This research paper investigates the performance of VoIP traffic over WiMAX networks. The impact of various voice codec schemes and statistical distribution for VoIP over WiMAX has been investigated in detail. Through various simulation experiments under realistic networking scenarios, this study provides an insight into the VoIP performance in the WIMAX networks. Parameters that indicate the Quality of Service such as delay, jitter, packet loss and MOS are analyzed in these scenarios. The simulations results indicate that better choice of voice codecs and statistical distribution have significant impact on VoIP performance in the WiMAX networks.**

Index Terms— **VoIP, WiMAX, ertPS and Video Codecs**

I. INTRODUCTION

A growing trend has been noticed in real time voice communication such as Voice over Internet Protocol (VoIP) in the recent years [1]. VoIP enables users to use Internet or intranet as transmission medium for telephone calls by sending voice data in packets using Internet Protocol (IP) rather than by traditional circuit switched Public Switched Telephone Network (PSTN) [2]. VoIP is based on IP and therefore the transmission technology is essentially digital [3]. Because of the digital transmission system, caller's voice is first digitized and then separated in packets using complex algorithms known as codecs. Different codecs like G.711, G.723 and G.729 are used for encoding and decoding, most of which are defined by International Telecommunication Union-the Telecommunication Division (ITU-T). VoIP can be deployed on any IP based data network such as the Internet, Ethernet, Fiber Optic or wireless such as WiMAX and 3G.

The telecommunication technology has also evolved in the recent years to meet the increasing demands of the network users. In the past few years, the IP network has been extended to use wireless access technologies like 802.11 based Wireless Local Area Network (WLAN) [4] and Third Generation (3G) [5] cellular networks. These networks are already in an excessive demand for real time applications like voice, video and other multimedia related applications. An alternative solution is sought with the success of IEEE 802.16e standard [6] for Mobile Worldwide Interoperability for Microwave Access (WiMAX) [7] in the metropolitan areas. WiMAX is an access technology which provides wireless data transmission in various ways ranging from point-to-point links to mobile cellular access [8]. It is based on IEEE 802.16 standard, which provides wireless broadband access as an alternative to the cable and Digital Subscriber Loop (DSL) [9]. WiMAX provides basic IP connectivity to the users using mobile broadband data access.

The real time applications, such as VoIP, are very sensitive to delays, jitter and packet losses [10], therefore to ensure the consistency and efficiency in real time packets delivery, the network carrying VoIP should be designed and configured properly [11]. WiMAX is wireless broadband technology; therefore, it is susceptible that VoIP performance will be degraded up to some extent. WiMAX has a powerful Quality of Service (QoS) feature, which ensures better quality for interactive and real time audio and video services. Five service types have been proposed and incorporated into the QoS model of WiMAX, *i.e.* Unsolicited Grant Service (UGS), real time Polling Service (rtPS), non real time Polling Service (nrtPS), extended real time Polling Service (ertPS) and Best Effort (BE).

Unsolicited Grant Service (UGS): The UGS algorithm is designed to support real time constant bit rate (CBR) traffic such as VoIP that periodically generates fixed size data packets.

Real time Polling Service (rtPS): The rtPS is designed to support real time traffic such as MPEG video and teleconferencing that periodically generates variable size data packets.

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Non real time Polling Service (nrtPS): The nrtPS is designed to support non-real time application with minimum rate such as FTP. Therefore nrtPS is not suitable for VoIP.

Best Effort (BE): The BE is designed to support data streams which do not require a minimum service-level guarantees such as the web browsing and the file transfers..

Extended real time Polling Service (ertPS): The ertPS algorithm is designed to support real-time applications, such as VoIP with silence suppression, that have variable data rates but require guaranteed data rate and delays.

The aim of this study is to investigate the performance evaluation of VoIP over WiMAX networks in order to identify the most appropriate VoIP codec and statistical distribution in such scenarios. The main objectives of this research are:

- To inspect the importance of VoIP and to identify the factors effecting the VoIP performance
- To identify the best methodology in order to design and model the WiMAX network in order to carry VoIP calls.
- To simulate the WiMAX network, different VoIP codecs and WiMAX service classes in order to investigate and analyze the behavior and performance of the model.

The rest of the paper is organized as follows: Section II briefly discusses related work. Section III deals with the simulation setup used in OPNET [20] for WiMAX. Section IV evaluates and analyzes the simulation results of the VoIP application running over WiMAX. Finally, in Section V we conclude this paper.

II. Related Work

A rapid growth has been noticed in various wireless technologies in recent years. This has resulted in an increase in demand for wireless data services and multimedia application such as VoIP, streaming audio and video [12]. In order to provide good service and to meet the user demands, research has been in progress both in wireless technologies and VoIP network system. VoIP is becoming more and more popular especially after the deployment of WiMAX network in many countries [13]. Different aspects of VoIP over WiMAX have been addressed by researchers. The authors in [14] have investigated the data and voice support in the WiMAX network. The aim of their work was to examine the QoS deployment over WiMAX network and compare the performance obtained using two different WiMAX services classes i.e. UGS and ertPS. The author in [15] has pointed out different factors like delay, jitter and packet losses and discussed how WiMAX network can deal with them. In [16], the authors have considered a fixed WiMAX network in order to evaluate the performance of VoIP.

They have measured the performance of different transmission schemes in term of cumulative goodput, packet rate, sample loss rate and Mean Opinion Score (MOS) using R-score specified by ITU-T. In [17], the authors have proposed a traffic- aware scheduling algorithm for VoIP applications in WiMAX networks. They have studied the performance of their proposed method and compared it with that of some conventional methods. They have discussed the trade-off between delay and bandwidth efficiency and it is shown that using their scheduling methods enhances the efficiency of VoIP over WiMAX. The authors in [18] have discussed different issues related to VoIP and voice quality measurement models. They have outlined a new methodology for developing models for nonintrusive prediction of voice quality. The researchers in [19] have presented a voice quality measurement tool based on the ITU-T E-model. They have tested the tool in some calls generated through the RNP backbone, between two endpoints located at different Brazilian cities.

III. Simulation Setup

To evaluate the performance of VoIP over the WiMAX network, the scenarios were designed and in the network simulator OPNET [20] with the assumption that the only traffic generated in this network model is VoIP. There are only peer-to-peer voice calls throughout the simulation, which means there is no voice conferencing and the subscriber stations (SS) are considered as fixed during the simulation runs. Fig. 1 illustrates the WiMAX network model considered in the simulations. The WiMAX network consists of seven cells and an IP backbone. The cell radius is set to 0.2 Kilometers.

Each cell consists of five Subscriber Stations (SS) and one Base Station (BS). There is a server backbone containing only one Voice Server. The parameters of Base Station (BS) and Subscriber Station (SS) can be seen in the Fig. 2 and Fig. 3.

For the given simulation setup, the following experiments were performed.

Experiment 1: Voice quality is important for VoIP system because of the users' high demands for good quality voice services. In these scenarios, we considered the use of various voices codecs in the same WiMAX network in order to investigate the performance of voice codecs for VoIP. G.711 (64kpbs), G.729 (16kbps) and G.723 (5.8kbps) were considered for this experiment.

Experiment 2: In this experiment, we set up simulation scenarios which measure different traffic distributions to investigate the impact of traffic arrival distributions on VoIP

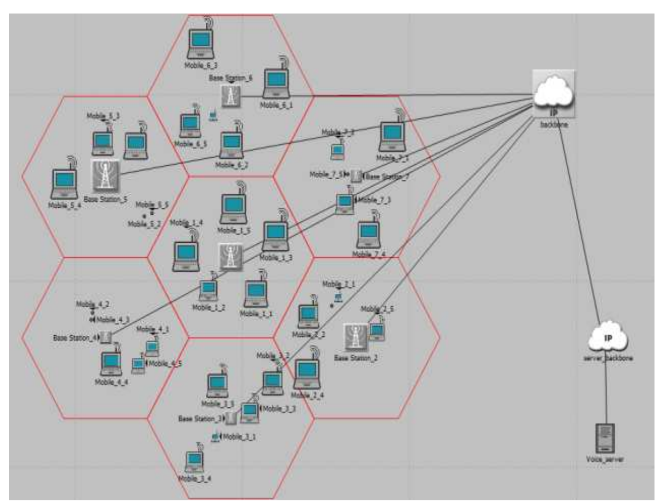

Fig. 1: WiMAX Network Model

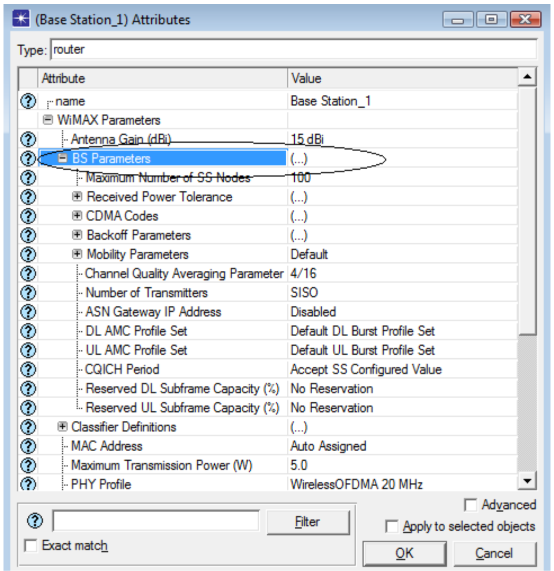

Fig. 2: Base Station (BS) parameters

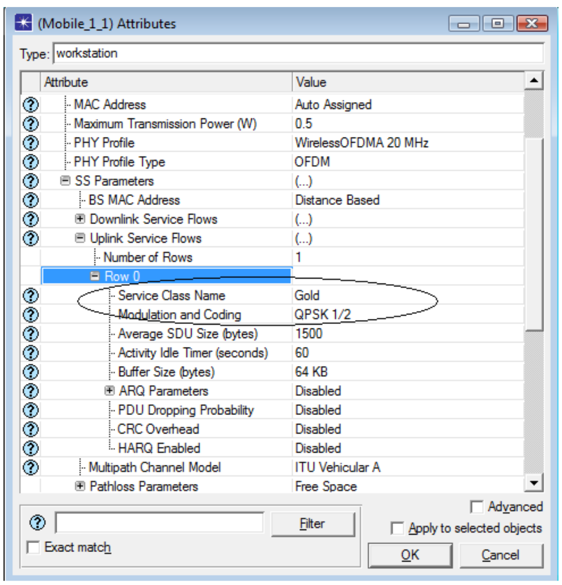

Fig. 3: Subscriber Stations (SS) parameters

performance in a WiMAX network. Different traffic arrival distributions are available in OPNET [20] for voice application *e.g.* the voice calls can be considered for fixed or variable amount of time.

A. Performance Metrics

In this study, we used the following four metrics to evaluate the performance of WiMAX in terms of end-to-end QoS for VoIP.

Mean Opinion Score (MOS): The Mean Opinion Score (MOS), recommended by ITU-T in 1996, is the most widely used subjective measure of voice quality. A MOS value is normally obtained as an average opinion of quality based on asking people to grade the quality of speech signals on the five point scale (Excellent =5; Good=4; Fair=3; Poor=2; Bad=1) under controlled conditions as set out in the ITU-T standard p.800.

Packet End to End Delay: The total voice packet delay is calculated as:

$$D_{e2e} = D_n + D_d + D_c + D_{de} \tag{1}$$

Where D_{e2e} represents the end-to-end delays while D_n , D_d , D_c , D_{de} represent the encoding, decoding, compression and decompression delays, respectively.

Jitter: Jitter is calculated as the signed maximum difference in one way delay of the packets over a particular time interval [21]. Generally, jitter is defined as the absolute value of delay difference between selected packets.

Packet loss: Packet loss is another factor that can degrade the performance of VoIP. The packet loss can occur if packets are lost during the transmission or if the packets arrive too late to be useable by the receiving application.

IV. SIMULATION RESULTS AND ANALYSIS

In this section we compared the performance of VoIP in WiMAX through extensive simulations. Following are the results of the experiments as presented in the Section 3.

A. Experiment 1 Results

The deployed WiMAX network consists of seven cells having five SS placed in each cell in the coverage area of BS. The VoIP traffic is based on Constant Bit Rate (CBR) flows with the packet size and packet rate is governed by the specific codec used. Three codecs schemes are used in this experiment *i.e.,* G.711, G723 (5.3k) and G.729 (8k).

Mean Opinion Score (MOS): Figure 4 plots the average MOS values for three selected voice encoders *i.e.* G.711, G.723 and G.729.

The major observation of this experiment is that G.711 has highest MOS value of 3.7. This shows that G.711 provides good speech quality as compared to the other two codecs schemes. The reason for low MOS values in the case of G.723 and G.729 can be the jitter factor in comparison to G.711 codec scheme.

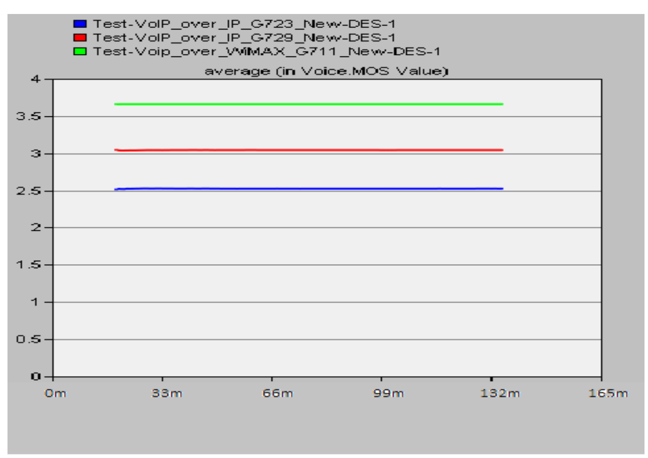

Fig. 4: Mean Opinion Score (MOS)

Jitter: Fig. 5 shows the comparative results of voice jitter for the codecs that are used in this experiment. It can be seen from the figure that the G.729 codec scheme has large value of jitter variation of 25 ms (-0.0000025) and therefore yields highest curve among the codec scheme. The negative value of jitter means that the time difference between the packets at the destination is less than that at the source. The voice jitter value for G.711 is 0ms which shows that there is no jitter or delay variation between the VoIP packets. The voice jitter for G.723 is almost identical to G.711, with a value of 1ms which shows that there is a slight delay variation in the of VoIP packets.

Packet end-to- end delay: Packet end-to-end delay is one of the most important performance metric in VoIP. Figure 6 shows the average packet end-to-end delay by considering the G.711, G.723 and G.729 codec schemes.

As can be seen in the Fig. 6, the results show that G.723 codec scheme yields the highest voice packet end-to-end delay averaging around 113ms, which is still an acceptable value. Other two codec schemes *i.e.,* G.711 and G.729 yields the lowest voice packet end-to-end delay averaging around 70ms.

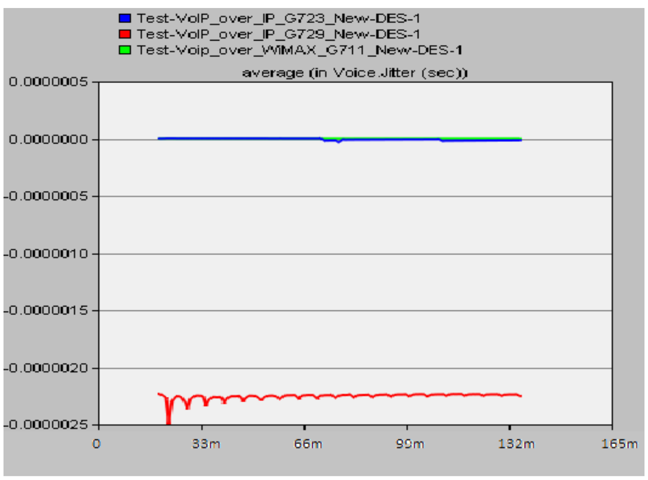

Fig. 5: Voice Jitter for G.711, G.723 and G.729

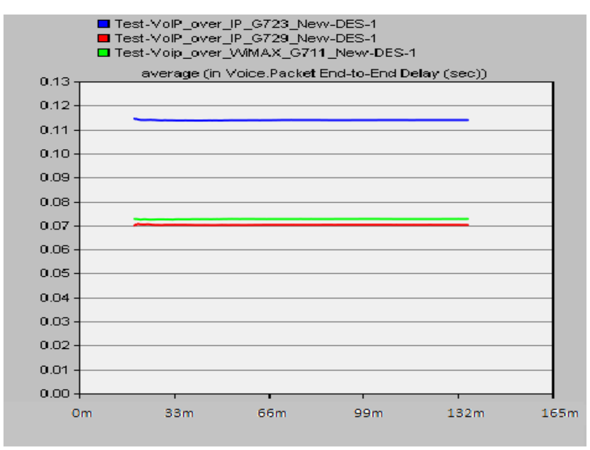

Fig. 6: Voice Packet End-to-End Delay for G.711, G.723 and G.729

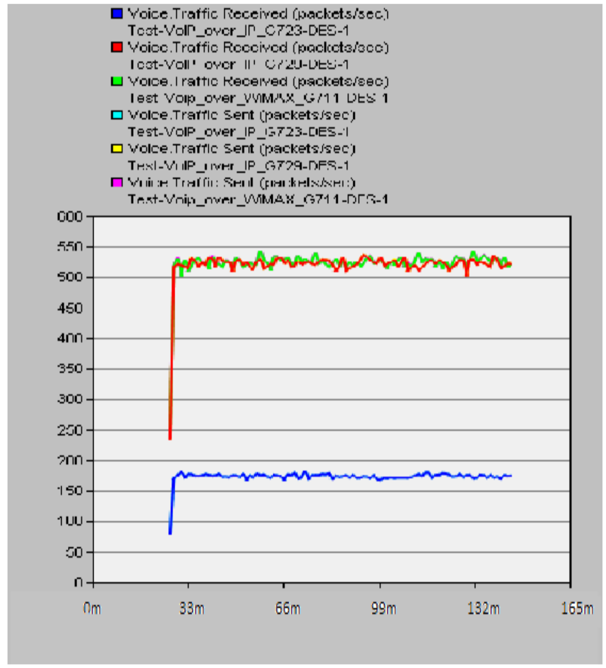

Fig. 7: Packets sent and received by codec schemes

Table 1: Summary of Experimental Results (voice codec)

Codec Scheme	Voice Jitter	MOS Value	Voice Packet End-to-End Delay	Voice Traffic Sent (packets/sec)	Voice Traffic Received (packet/sec)
G.711	0ms	3.7	70.2 ms	516	516
G.723	0ms	2.5	70.0 ms	173	173
G.729	25ms	3.0	113 ms	520	520

These simulation results indicate that G.711 and G.729 can provide better VoIP services in terms of end-to-end packet delays.

Packets sent and received: Fig. 7 shows packets send and received by the three voice codec schemes.

It can be seen that there is no packet drop in all of the three codec schemes. The simulation results clearly indicate that G.711 codec is the most appropriate voice codec scheme for VoIP services over WiMAX network, as it provides the best quality of voice in terms of MOS value, end-to-end delays and jitters. The main findings of experiment 1 are listed in Table 1.

B. Experiment 2 Results

In the first experiment, VoIP calls were generated constantly after every 5 seconds and the duration of each VoIP call is kept 300 seconds. In real WiMAX networks, it is unlikely that call generation will occur with a fixed arrival distribution. The assumption, as considered in the experiment 1 for the duration of a fixed duration calls, cannot be considered in realistic scenarios. In this scenario, different statistical distributions were used to model the behavior of real VoIP traffic such as constant, exponential and poisson.

Connection Delay: Fig. 8 shows the WiMAX connection delay for constant, exponential and poisson statistical distributions.

As can be seen in the Figure 8, the WiMAX delay has the smallest value around 0.5 ms for all traffic arrival distributions. Therefore, the delay parameter can be ignored as all these values are less than 1ms.

Packet Delay Variation: Fig. 9 shows the comparison graph of voice packet delay variation for the three traffic distributions.

Packet delay variation is an important parameter and has a crucial role in the degradation of the network performance and affects the user-perceptual quality. Higher packet delay variation results in congestion of the packets, which can result in the network overhead.

Fig. 9 shows that exponential distribution experiences the smallest delay variation of 0.12 ms, which can be tolerated because of buffering and jitter compensation within the voice decoder. This shows that exponential distribution provides the stable Quality of Service (QoS) for the VoIP services. On the

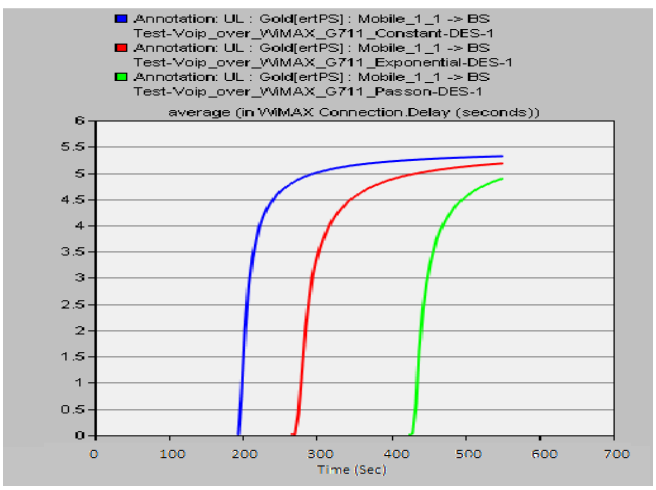

Fig. 8: Delay for three traffic distributions

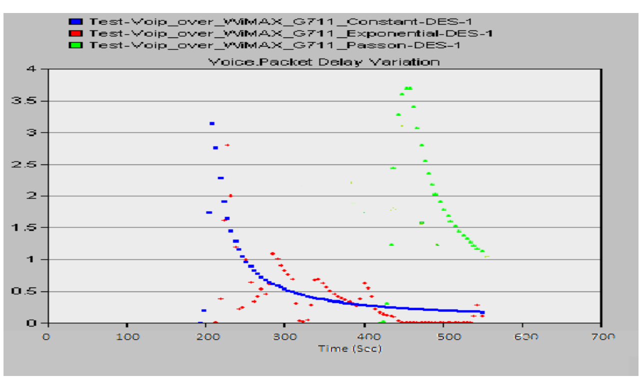

Fig. 9: Delay Variation for three traffic distributions

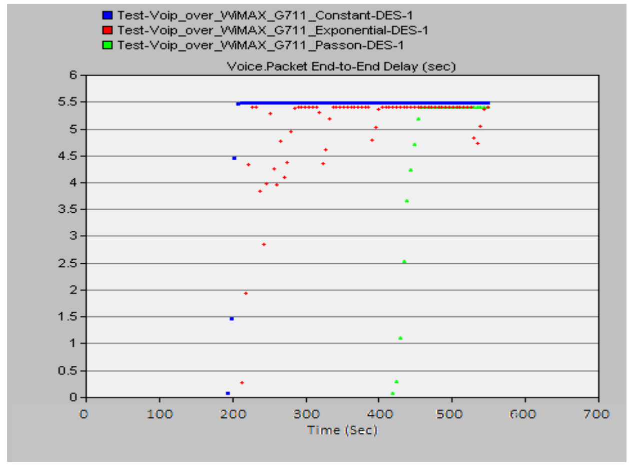

Fig. 10: Packet end-to-end delay for three traffic distributions

other hand, Poisson distribution experiences the highest delay variation of 1.13 ms which has significant effect on the voice performance in the WiMAX network.

End to End Delay: Fig. 10 depicts the packets end-to-end delay for the three different traffic distributions.

It can be seen in the Figure 10 that all the three traffic distributions yield almost the same values. However, comparing to the maximum values shown in the Table 2, the exponential distribution has the lowest voice packet end-to-end delay (5.41ms). This is only slightly lower than Poisson traffic distribution (5.45ms) and constant traffic distribution (5.47ms).

Traffic Sent and Received: Fig. 11 shows traffic sent and received by the three distributions considered in the simulation study. The number of packet sent during the poisson distribution (21466 packets) and exponential traffic distribution (192767 packets) are less as compared to constant distribution (472775 packets) and the default constant traffic distribution sent the most voice packets during the simulation as shown in the Fig. 11. The main findings of this scenario are listed in the Table 2.

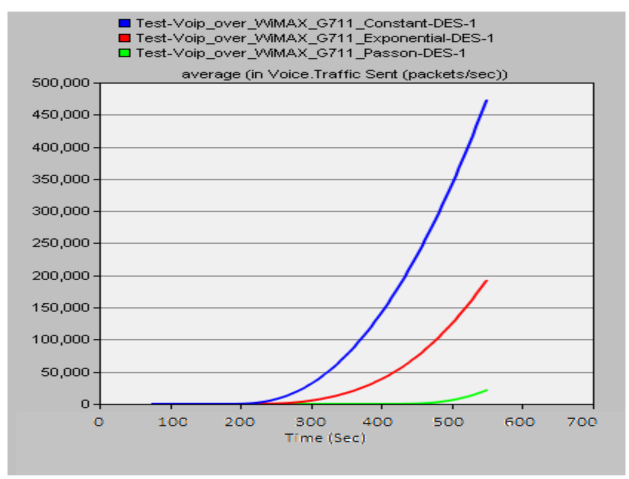

Fig. 11: Traffic sent (packets/second) by three traffic distributions

Table 2: Summary of traffic arrival distribution results

Traffic Distribution	Voice Jitter	Packet End-to-End Delay	WiMAX Delay	Voice Traffic Sent (packets/second)	Voice Traffic Received (packets/second)
Exponential	0.12 ms	5.41 ms	0.5 ms	192767	192767
Passion	1.13 ms	5.45 ms	0.4 ms	21466	21466
Constant	0.17 ms	5.47 ms	0.5 ms	472775	472775

The overall results in this scenario indicate that the VoIP services perform well under exponential traffic distribution. However, considering the voice packets sent from the constant traffic distribution are nearly twice as that of the exponential traffic distribution, but the results under each distribution are very similar. Based on the results it is concluded that the traffic arrival distributions have little impact on the VoIP overall performance.

V. Conclusion

WiMAX is a new emerging wireless broadband access technology which supports a variety of real-time services. One of the well known examples of the real time applications is VoIP. VoIP telephony has become a strong competitor to existing telephone networks. In this paper, the performance of VoIP over WiMAX was measured and analyzed in terms of crucial parameters like Jitter, MOS, end-to-end delay and packet sent and received.

The simulation study was carried out to evaluate the performance of VoIP over the WiMAX networks. Different parameters such as jitter, MOS value, packet end-to-end delays and packets sent and received were used to measure the performance of VoIP over WiMAX. Three voice codecs *i.e.* G.711, G.723 and G.729 were simulated in order to find the most appropriate voice codec for VoIP over WiMAX network. The simulation results showed that VoIP performed best under the G.711 codec as compared to the G.723 and G.729 codecs. The research findings also show that VoIP applications can perform better under the exponential traffic distribution.

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