Performance Evaluation of Scheduling Services for VoIP in WiMAX Networks

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ABSTRACT
WiMAX is a high speed, wireless broadband access technology for large coverage area. It provides IP based connectivity to the stationary, mobile as well as nomadic users. Voice over Internet Protocol (VoIP) through WiMAX is the most prominent telecommunication service. In this paper, simulative investigations have been done for VoIP in WiMAX network considering scheduling services [1]. Investigations have been done in terms of important Quality of Service parameters like jitter, packet end to end delay and Mean Opinion Score. Unsolicited Grant service has been identified as the most promising scheduling service for VoIP in WiMAX network.

General Terms
WiMAX network, scheduling services.

Keywords
WiMAX, VoIP, OPNET, jitter, MOS.

1. INTRODUCTION
WiMAX is an acronym for Worldwide Interoperability for Microwave Access. WiMAX is based on IEEE 802.16 [2] specification which ensures compatibility and interoperability between broadband wireless access equipments. It can provide broadband wireless access (BWA) upto 30 miles for fixed station and 3 to 10 miles for mobile stations with transmission data rates between 1.5 and 75 Mbps per channel [3]. With current updates, WiMAX can provide data rates up to 1 gigabit/sec for fixed stations. It solves the last mile problem in the metropolitan networks because of its high bandwidth and capacity for long distance transmission [4].

Being an emerging technology, WiMAX supports multimedia applications such as voice over IP (VoIP), voice conference and online gaming. Scheduling services play an important role for insuring guaranteed QoS. Scheduling services provide medium access control functions like data flow control to define how and when devices will receive and transmit in a communication system. The types of services that WiMAX can provide includes guaranteed bandwidth with low delay unsolicited grant service (UGS), random access best effort (BE) service, real time polling service (rtPS), non real time polling service (nrtPS) and extended real time polling service (ertPS) [5]. Therefore, in order to find an effective scheduling service for VoIP over WiMAX system, a study of VoIP quality under different scheduling types is performed.

The rest of the paper is organized as follows: Section II describes a brief overview of related work. Section III discusses about OPNET Modeler, whereas Section IV deals with the simulation setup used in the WiMAX network and QoS parameters for VoIP. Results are discussed in Section V, before we finally conclude in section VI.

2. RELATED WORK
There have been recent studies focusing on the performance evaluation of WiMAX networks.

Grewal et al [6] analysed various QoS provisions for different application traffics. It was concluded that FTP traffic is best served with nrtPS, video traffic with rtPS, email with BE and VoIP with UGS. The effect of Adaptive Modulation Coding (AMC) mechanism on the QoS performance of WiMAX network is also studied.

Adhicandra [7] has investigated the data and voice support in the WiMAX network. Various QoS configurations were used to improve the performance of VoIP over Best Effort (BE) WiMAX. The extended real-time polling service (ertPS) scheduling class that was designed to support variable rate real-time services significantly improves the performance of VoIP over BE WiMAX.

The performance of integrated WiMAX /Wi-Fi network by using different voice codecs has been investigated by Islam, Rashid and Tarique [8]. It was shown that VoIP under GSM Enhanced Full Rate (GSM-EFR) and GSM Full Rate (GSM-FR) codecs achieves desirable speech quality with tolerable delay and jitter. However, G.726 performs poorly in terms of MOS value, delay, jitter, and packet loss.

A lot of information about WiMAX network can be found in literature [9]. The authors have studied the effect of base frequency on the performance of WiMAX and shown the effect of base frequency on MOS, packet end to end delay and throughput with G.711 as voice codec. The study also includes the influence of distance between BS and SS on the parameter path loss. They conclude that the best performance can be achieved at lower base frequencies.

The performance of WiMAX and UMTS (Universal Mobile Telecommunication Systems) for VoIP traffic is evaluated by Jadhav et al. [10]. The performance is evaluated in terms of performance metrics such as MOS, end-to-end delay, jitter and packet delay variation. The results show that WiMAX outscores the UMTS and is the better technology to support VoIP applications.

3. SIMULATION SETUP
To evaluate the performance of the WiMAX network, a scenario was designed in the network simulator OPNET [11] with the assumption that the traffic generated in this network

model is VoIP only. 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 four SSs and one base station (BS) which is connected to the server backbone having only one voice server through an IP backbone. The distance between the SS and BS is set to 1 Kilometers.

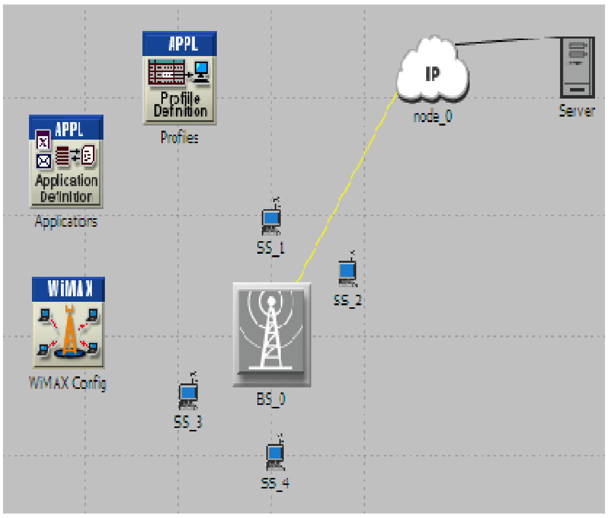

Fig.1 WiMAX Network Model

The WiMAX parameters used in the network are listed in the Table 1 & Table 2.

Table1. Network Design Parameters

WiMAX Physical Profile	Wireless OFDMA 20MHz
Bandwidth	20MHz
FFT	512
Duplexing Technique	TDD
Efficiency Mode	Physical Layer Enabled
BS Transmission Power	10W
SS Transmission Power	0.5W
Modulation & coding	16 QAM 3/4
Multipath Channel Model	ITU Pedestrian B
Path loss Model	Suburban Fixed (Erceg)
Terrain Type	Terrain B
Application	Voice over IP
Voice codec	G.711

Table2. Traffic Characteristics

Match Property	IP ToS
Match Condition	Equals
Match Value	Interactive Voice

3.1 Path loss Model

WiMAX has a powerful Quality of Service (QoS) feature, which ensures better quality for interactive and real time audio and video services. Five scheduling services were defined into the QoS model of WiMAX are,

3.1.1 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.

3.1.2 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.

3.1.3 Non real time Polling Service (nrtPS)

The nrtPS is designed to support non-real time application with minimum rate such as FTP. This WiMAX QoS class is used for services where a guaranteed bit rate is required but the latency is not critical.

3.1.4 Best Effort (BE)

The BE is designed to support data streams which do not require a minimum service-level guarantee. This WiMAX QoS is used for Internet services such as email and web browsing.

3.1.5 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.

3.2 Performance Metrics

IP telephony/ VoIP is something that is growing up very fast. The QoS for this is measured by performance metrics such as Mean Opinion Score (MOS), end-to-end delay, and jitter.

3.2.1 Mean Opinion Score (MOS)

MOS provides a numerical measure of the quality of human speech in voice telecommunications, with value ranging from 1 to 5 where 1 is the worst quality and 5 is the best quality. It can also listed in Table3.

Table3. Network Design Parameters

Quality Scale	Score	Listening Effort Scale
Excellent	5	No effort required
Good	4	No appreciable effort required
Fair	3	Moderate effort required
Poor	2	Considerable effort required
Bad	1	No meaning understood with effort

3.2.2 Jitter

Jitter is technically the measure of the variability over time of the latency across a network i.e. the variation in arrival time of consecutive packets. If two consecutive packets leave the source node with time stamps t1 & t2 and are played back at the destination node at time t3 & t4, then:

Jitter = (t4 - t3) - (t2 - t1)

Negative jitter indicates that the time difference between the packets at the destination node was less than that at the source node.

3.2.3 Packet End to End Delay

Packet End to End delay or latency is characterized as the amount of time it takes for speech to exit the speaker's mouth and reach the listener's ear. The total voice packet delay is calculated as:

De2e = Dn + De + Dd + Dc + Dde

Where De2e represents the end-to-end delays while Dn, De, Dd, Dc, Dde represent the network, encoding, decoding, compression and decompression delays, respectively.

4. RESULTS AND DISCUSSION

Voice Quality is important for VoIP system because of user's high demand for good quality voice services. Performance of VoIP in WiMAX is compared in terms of scheduling services UGS, rtPS, nrtPS, ertPS and BE & produces following results.

4.1 Packet end-to- end delay

VoIP in WiMAX network would be transmitted to destination as a collection of packets where each one might follow different routes thus arrive at the destination with different delays. Figure 2 shows the average packet end-to-end delay for the different service flows.

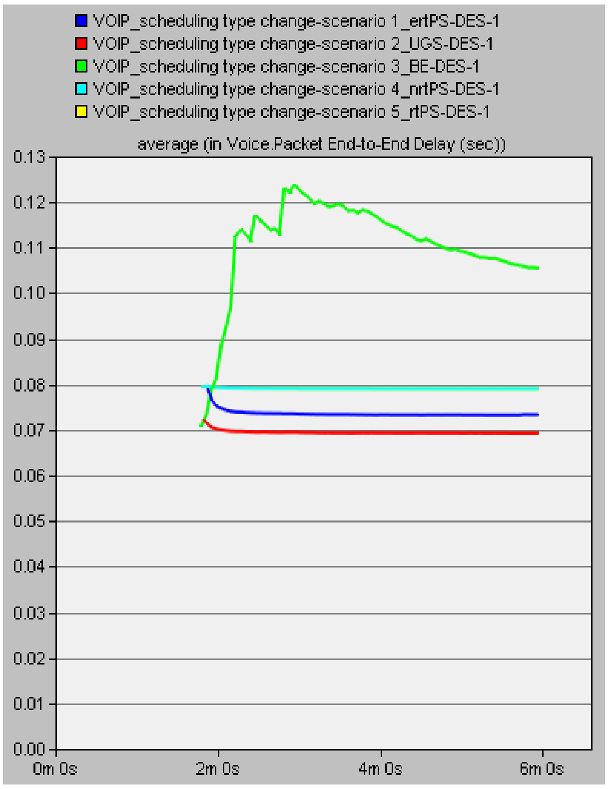

Fig.2 Packet End to End Delay (sec)

UGS service flow has the lowest delay for WiMAX. UGS is designed to support real time services that generate fixed size data packets on a periodic basis. VoIP application does exactly that. UGS offers fixed size grants on a real time periodic basis and does not need the SS to explicitly request bandwidth. The SS is already granted a fixed bandwidth and can transmit the data packets in a specific slot during the uplink transmission.

For BE service flow, connections are never polled and receive resources through contention. That is why, BE service flow exhibits maximum delay.

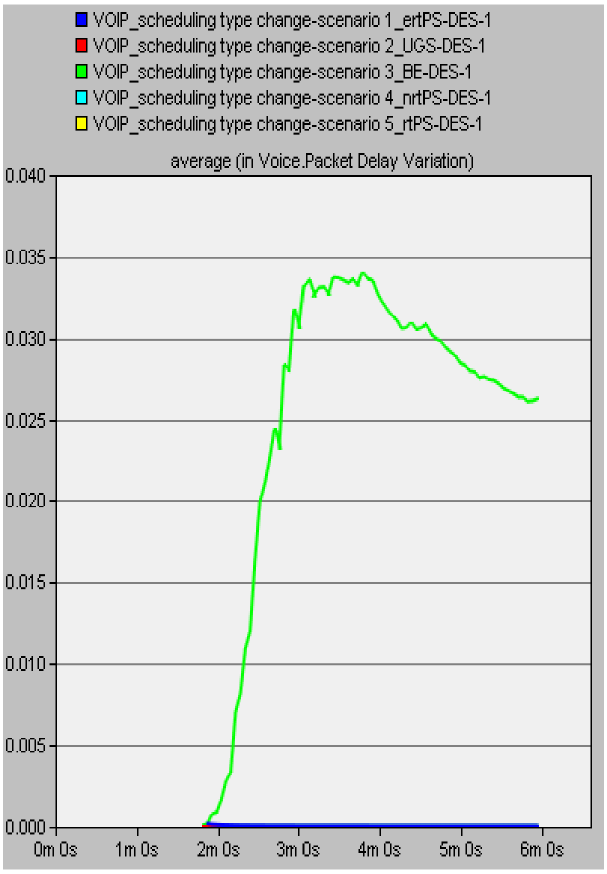

Fig.3 packet delay variations

As packet end to end delay is maximum for BE service flow, only this type of service flow shows packet delay variations.

4.2 Mean Opinion Score

Mean Opinion Score is one of the most important performances metric in VoIP. Figure 4 shows the comparative results of average MOS values for the service flows used in the scenario.

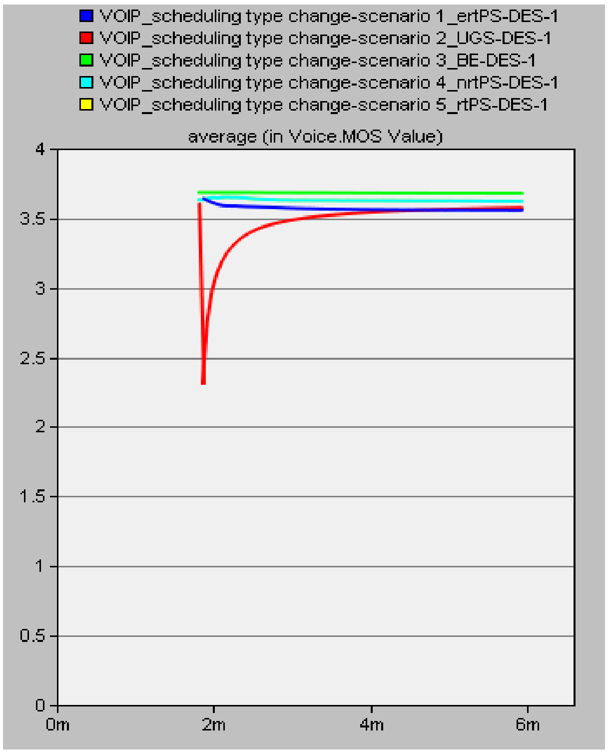

Fig.4 Average Mean Opinion Score

Mean Opinion Score value is app 3.5 for all the service flows and does not show much change.

4.3 Jitter

Fig. 5 shows the comparative results of voice jitter for the service flows that are used in this experiment.

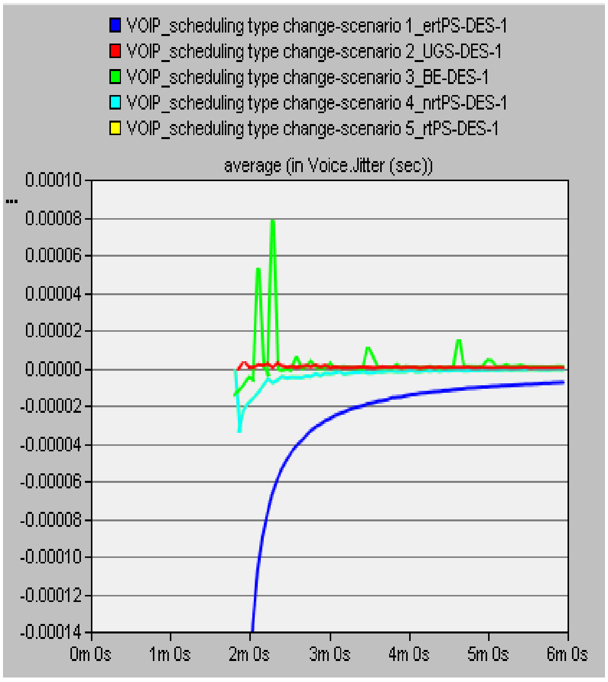

Fig.5 Relative Average Jitter (sec)

Average jitter for UGS service flow has smallest value. UGS service flow is designed for constant bit rate traffic which generates fixed size packets on periodic basis and allocates bandwidth. It proves efficient for VoIP applications.

5. CONCLUSION

In this paper, the performance of VoIP over WiMAX networks is studied in terms of prominent QoS parameters such as jitter, MOS and packet end to end delay. Different scheduling services have been taken for comparative simulative analysis. It has been found from the simulation results that UGS service flow has lowest average jitter, least packet end to end delay and satisfactory MOS value. This makes it best suited for VoIP traffic through WiMAX networks.

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