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Analysis of VoIP traffic in WiMAX using NS2 simulator

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***Abstract*— Worldwide Interoperability for Microwave Access (WiMAX) is a wireless broadband technology, which supports point to multi-point (PMP) broadband wireless access. It is a fixed and mobile wireless access technology based on the IEEE 802.16 standards. WiMAX technology is expected to meet the needs of a large variety of users from those in developed nations wanting to install a new high speed data network very cheaply without the cost and time required to install a wired network, to those in rural areas needing fast access where wired solutions may not be viable because of the distances and costs involved. It can be used for mobile applications, proving high speed data to users on the move. Voice over Internet Protocol (VoIP), called Internet Protocol (IP) Telephony, Internet telephony or Digital Phone. It utilizes the IP network (Internet or intranets) for telephone conversations. Unlike traditional TCP/IP services, VoIP is highly delay intolerant and hence it needs a high priority transmission.The performance of different VoIP codecs over the best effort service flow for WiMAX network is analyzed in this paper. The parameters considered for the evaluation of network are throughput, average delay & jitter. The simulation is done using network simulator 2 (NS2) by varying number of nodes and results are presented in graphical form.**

***Index Terms*—BE, NS2, VOIP, WiMAX.**

I. Introduction

The IEEE 802.16 standard forms the basis of Worldwide Interoperability for Microwave Access (WiMAX). It was developed by the WiMAX Forum with the objective of provide high speed data transfers over the air. The WiMAX Forum is an industry-led, not-for-profit organization that certifies and promotes the compatibility and interoperability of broadband wireless products based upon IEEE Standard 802.16 [1]. WiMAX has its origin in the computer industry and is an alternative to Third Generation Partnership Project (3GPP) and technologies like High Speed Packet Access (HSPA) and Long Term Evolution (LTE).

WiMAX network consists of two main parts: Base station (WiMAX tower) and subscriber station (WiMAX receiver). As shown in Fig.1,when a user sends data from a subscriber device to a base station then that base station broadcast the wireless signal into channel which is called uplink and base station transmit the same user or another is called downlink [2].

IEEE 802.16 standard supports two modes for sharing wireless medium [3]:

1) Point to multipoint (PMP)

In this mode, a base station (BS) serves number of subscriber stations (SSs) in the same antenna sector in broadcasting manner. Therefore all SSs receive same transmission from BS.

2) Mesh

In mesh mode, communication can occur directly among SSs. Traffic can be routed through SSs.

There are two types of WiMAX that are available:

1) 802.16d (802.16-2004)

Its range is up to 50 km and it is Fixed WiMAX which is based on OFDM PHY.

2) 802.16e (802.16-2005)

Its range is up to 15 km which is Mobile WiMAX based on OFDMA PHY.

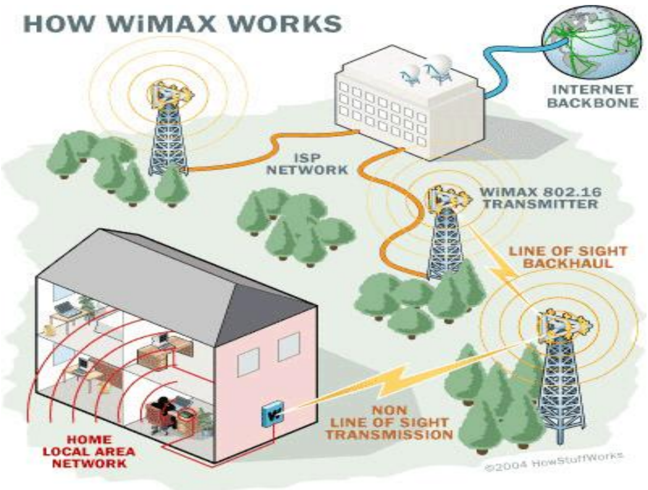

Fig.1 Working of WiMAX

The two types of WiMAX are used for different applications and the implementation of each has been optimized to suit its particular application

VoIP is a technology that is in great demand these days. VoIP sends digitized voice across computer networks. Its interactive nature makes it very appealing for users and today it is one of the most dominant technologies for communication. VoIP over WiMAX has been emerging as an infrastructure to provide broadband wireless voice service with cost efficiency reliability and guaranteed quality of service. WiMAX provides basic IP connectivity. Mobile devices capable of using WiMAX network will need to support voice calling over the internet protocol.

The rest of the paper is organized as follows. In section II, we present literature survey of different service flow classes supported by IEEE 802.16 standard and focus on VOIP codecs. In section III, simulation details and some important configuration parameters are explained. Section IV consists of simulation results and result analysis. Finally, section V concludes the paper, with outline of future work.

II. LITERATURE SURVEY

A. IEEE 802.16 service flow classes

The protocol layers architecture of WiMAX defines only two layers, the physical layer and the Medium Access Control (MAC) layer [4]. The IEEE 802.16 MAC is connection oriented and is made up of three sublayers

- Convergence sublayer,
- Common part sublayer
- The security sublayer.

Connections are unidirectional and they are referenced with 16-bit CIDs (connection IDentifiers). A service flow is a MAC transport service that provides unidirectional transport of packets on the uplink or on the downlink. A service flow is identified by a 32-bit SFID (Service Flow IDentifier).

Present days several performance studies are done on WiMAX. A threshold-based priority algorithm to improve the delay jitter of the UGS and rtPS classes is proposed in [5] is based on adjusting the threshold imposed on the nrtPS queue. How different factors such as load and mobility affect the performance of WiMAX in a single cell environment has been studied in [6].This analysis is done using Qualnet simulator. Analysis of WiMAX in PMP mode is performed in [7] by applying different schedulers at base station. A Quality of Service (QoS) mechanism for WiMAX delay in PMP mode of IEEE 802.16 is also proposed along with the comparison. WiMAX QoS mechanisms in detail are introduced and categorized in [8]. The purpose of the study in [9] was to design a WiMAX system configuration to optimize network capacity as well as application delays.

Fixed and Mobile WiMAX supports four and five scheduling classes respectively [10]:

a) UGS (Unsolicited Grant Service)

The UGS class is designed to support real-time uplink service flows that transport fixed-size data packets on a periodic basis, such as T1/E1 and VoIP without silence suppression. The service offers fixed-size grants on a real-time periodic basis, which eliminate the overhead and latency of SS requests and assure that grants are available to meet the flow's real-time needs.

b) rtPS (real-time Polling Service)

The rtPS class is designed to support real-time uplink service flows that transport variable-size data packets on a periodic basis, like MPEG video. The service offers real-time, periodic request opportunities, which meet the flow's real-time needs and allow the terminal to specify the size of the desired grant. Therefore this service flow requires more request overhead than UGS, but it supports variable grant sizes.

c) ertPS (extended real-time Polling Service)

Extended rtPS is a scheduling mechanism available only for Mobile WiMAX. It builds on the efficiency of both UGS and rtPS. The base station (BS) provides unicast grants in an unsolicited manner, thus saving the latency of a bandwidth request message. UGS allocations are fixed in size, whereas ertPS allocations are dynamic. The BS may provide periodic uplink allocations that may be used for requesting the bandwidth as well as for data transfer. There is a default allocation size that can be changed by the terminal.

d) nrtPS (non-real-time Polling Service)

The nrtPS offers unicast polls on a regular basis, which assures that the uplink service flow receives request opportunities even during network congestion. The BS typically polls nrtPS connections on an interval on the order of one second or less.

e) BE (Best Effort)

The best effort service provides no QoS guarantees. The service is suitable for data streams for which no minimum service level is required and therefore may be handled on a space-available basis. Unlike UGS and rtPS scheduling services, nrtPS and BE are designed for applications that do not have any specific delay requirement.

TABLE I
QOS SPECIFICATIONS

Service flow class	QoS specifications
UGS	Minimum Reserved Traffic Rate, Maximum Latency Tolerated Jitter.
rtPS	Minimum Reserved Traffic Rate, Maximum Sustained Traffic Rate, Maximum Latency and Unsolicited Polling Interval.
ertPS	Maximum Sustained Traffic Rate, The Minimum Reserved Traffic Rate, The Maximum Latency Unsolicited Grant Interval.
nrtPS	Minimum Reserved Traffic Rate, Maximum Sustained Traffic Rate Traffic Priority.
BE	Traffic priority Maximum sustained rate

QoS determines if a wireless technology can successfully deliver high value services such as voice and video. The service flow defines the QoS parameters for the packets that are exchanged on the connection. Each service flow class is associated with corresponding QoS parameters set. These parameters are listed in Table I. Support for QoS is a fundamental part of the WiMAX MAC-layer design.

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B. VoIP Codecs

VoIP utilizes the IP network (Internet or intranets) for telephone conversations.

Codecs are used to convert an analog voice signal to digitally encoded version. Codecs vary in the sound quality, the bandwidth required, the computational requirements, etc [11]. Each service, program, phone, gateway, etc typically supports several different codecs.

a) G.711

G.711 is a codec that was introduced in 1972 for use in digital telephony such as integrated services digital network (ISDN), T.1 and E.1 links. Its formal name is Pulse code modulation (PCM) of voice frequencies. G.711 uses a logarithmic compression and squeezes each 16-bit sample to 8 bits. Thus it achieves a compression ratio of 1:2. The resulting bit rate is 64 Kbit/s for one direction. So a call consumes 128 kbps with some overhead for packet headers. This codec can be used freely in VoIP applications as there are no licensing fees. It works best in local area networks (LAN) where a lot of bandwidth is available. The codec has two variants: A-Law is being used in Europe and in international telephone links while µ-Law is used in the U.S.A. and Japan. Both are logarithmic, but A-law was specifically designed to be simpler for a computer to process. G.711 µ-law tends to give more resolution to higher range signals while G.711 A-law provides more quantization levels at lower signal levels.

b) G.729

G.729 is a codec that has low bandwidth requirements but provides good audio quality. The codec encodes audio in frames and each frame is 10 milliseconds long. Its Mean Opinion Scores (MOS) is 4. With the sampling frequency of 8 kHz, the 10 ms frame contains 80 audio samples. This codec algorithm encodes each frame to 10 bytes, so the resulting bit rate is 8 kbps for one direction. G.729 compresses digital voice in packets of 10 milliseconds duration. It is officially described as Coding of speech at 8 kbit/s using conjugate-structure algebraic code-excited linear prediction (CS-ACELP). G.729 is a licensed codec and is mostly used in VoIP applications where bandwidth conservation is required.

c) G.723.1

G.723 is a standard speech codec and is an extension of G.721.It provides voice quality covering 300 Hz to 3400 Hz using Adaptive Differential Pulse Code Modulation (ADPCM) to 24 and 40 Kbit/s for digital circuit multiplication equipment (DCME) applications. There are two variants of G.723.1. They both operate on audio frames of 30 milliseconds (i.e. 240 samples), but the algorithms differ. The bit rate of the first variant is 6.4 Kbit/s and the MOS is 3.9. The bit rate of the second variant is 5.3 Kbit/s and MOS is 3.7. G.723.1 is a licensed codec.

d) GSM 06.10

GSM 06.10 which is also known as GSM Full Rate, is a codec designed by the European Telecommunications Standards Institute for use in the GSM mobile networks. This variant of the GSM codec can be freely used in open source VoIP applications. The codec operates on audio frames 20 milliseconds long (i.e. 160 samples) and it compresses each frame to 33 bytes, so the resulting bitrate is 13 Kbit/s. precisely, each encoded frame is exactly 32 and 1/2 byte. Therefore 4 bits are unused in each frame. The codec's M.O.S. is 3.7.

e) Speex

Speex is an open source patent-free codec designed by the Xiph.org Foundation. It is designed to work with sampling rates of 8 kHz, 16 kHz, and 32 kHz and it can compress audio signal to bitrates between 2 and 44 Kbit/s. The most usual choice is the 8 kHz (narrow band) variant for its use in VoIP telephony.

f) iLBC

Internet Low Bit Rate Codec (iLBC) is a free codec. The two frame options available are 20 ms or 30 ms and resulting bit rate is 15.2 Kbit/s and 13. 33 Kbit/s, respectively.

III. SIMULATION DETAILS

The Network Simulator 2 (NS-2) is an open-source and powerful event driven simulation tool for the simulation of packet-switched networks. NS2 is an object oriented simulator written in OTcl and C++ languages. OTcl acts as the frontend (user interface) and C++ acts as the backend, running the actual simulation [12][13]. In this study, we explore the same methodology as in [14]. However, we evaluate the performance of various VoIP codecs with different performance metrics by using NS-2 instead of QualNet simulator.

Simulation scenario for this study is created such that there are multiple SSs in the range of a base station and the base station is connected to the core network. In NS-2, Packet tracing records the detail of packet flow during a simulation. It can be classified into a text-based packet tracing and a NAM packet tracing. Text-based packet tracing records the detail of packets passing through network checkpoints (e.g., nodes and queues) in a trace file.

We have used NS-2 version 2.31 [15] with WiMAX module patch release version 2.6 [16]. The TCL scripts were developed such that the input parameters can be varied. In NS-2, the core network is represented by a sink node which can only accept the incoming packets. Number of mobile nodes in the simulation and type of VOIP codec were passed in as input parameters while running the simulation. The VOIP codecs are varied as G.711, G.723, and G.729.For each of the codes, number of mobile nodes with the VOIP traffic was varied from 2, 4, and 6. Some of the parameters used in

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simulation are mentioned in table II.

The traffic is started after some time to allow the mobile node to complete the registration because after the simulation starts each node goes through basic registration procedure to get associated with the base station.

The trace files obtained are used for further analysis. Simulation results can be analyzed by running AWK script or PERL script on the trace file to obtain values of different parameters like throughput, average delay, and average jitter. These analysis results are represented graphically to compare following parameters for various number of nodes.

TABLE II
SIMULATION PARAMETERS

Channel type	Wireless channel
Radio propagation model	Propagation/OFDMA
Network interface type	Phy/WirelessPhy/OFDMA
MAC type	Mac/802_16/BS
Routing protocol	NOAH
Antenna model	Antenna/OmniAntenna
Link layer type	LL

a) Throughput

Throughput is a measure of the date rate (bits per second) generated by the application and is equal to the total data transferred divided by the total time it took for the transfer. Theoretical value of throughput for fixed WiMAX is approximately 75 Mbps per channel (in a 20 MHz channel using 64QAM ¾ code rate) and its practical value is around 45 Mbps/channel.

b) Average Delay or latency

Delay is the time taken by the packets to transverse from the source to the destination. The main sources of delay can be further categorized as source-processing delay, propagation delay, network delay and destination processing delay [17]. Average Delay measures the average one way latency observed between transmitting an event and receiving it at each sink node.

c) Jitter or Delay variation

Jitter is defined as a statistical variance of the RTP data packet inter-arrival time. It is the variation in the time between packets arriving. In the Real Time Protocol, jitter is measured in timestamp units. For example, if you transmit audio sampled at the usual 8000 Hertz, the unit is 1/8000 of a second. Delay variation is the variation in the delay introduced by the components along the communication path. Jitter is commonly used as an indicator of consistency and stability of a network.

IV. SIMULATION RESULTS

Table III shows values of throughput for different types of VoIP codecs and Fig. 2 shows the graph of throughput against number of mobile nodes for each codec. From the graph, it is observed that the throughput steadily increases as the number of nodes increases.

TABLE III
THROUGHPUT

No. of Nodes	G.711	G.723	G.729
2	125.54	25.18	45.65
4	220.31	46.51	80.11
6	357.79	69.63	129.56

This kind of behavior is observed because when the number of nodes increases, the number of packets being transmitted also increases. G.711 codec has the highest throughput as a result of highest packet size. The next higher throughput is for G.729. The G723 codec has the lowest throughput.

TABLE IV
AVERAGE JITTER

No. of Nodes	G.711	G.723	G.729
2	0.0000280511761838571	0.0000000041106273204	0.0000280511761838571
4	0.00002307936990258500	0.0000000067609654823	0.0000063913088737 3232
6	0.00005893413528256280	0.0000000082569284516	0.00001086508500230500

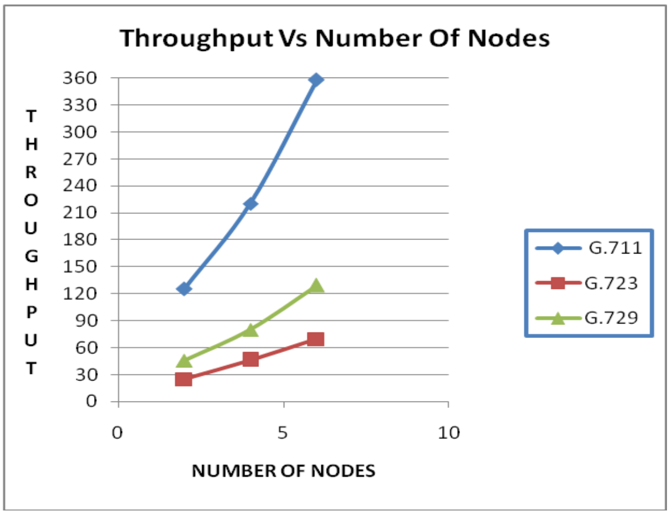

Fig.2 Throughput for various VoIP codecs

Table IV contains value of average jitter for various numbers of mobile nodes. Fig.3 shows variations of average jitter for each type of VoIP codec with increasing number of mobile nodes.

Average jitter increases linearly with increasing number of nodes in case of G.711. In case of G.729 average jitter increase steadily but its highest value for 6 mobile nodes is much less than that of the highest value for G.711. Increasing jitter shows that, the packets arrive at varying delay at the receiver. This reflects adversely in the perceived quality. Each packet contains a voice sample sent from the sender. It the packets are not received with the same delay at the receiver, the voice quality appears to be degraded.

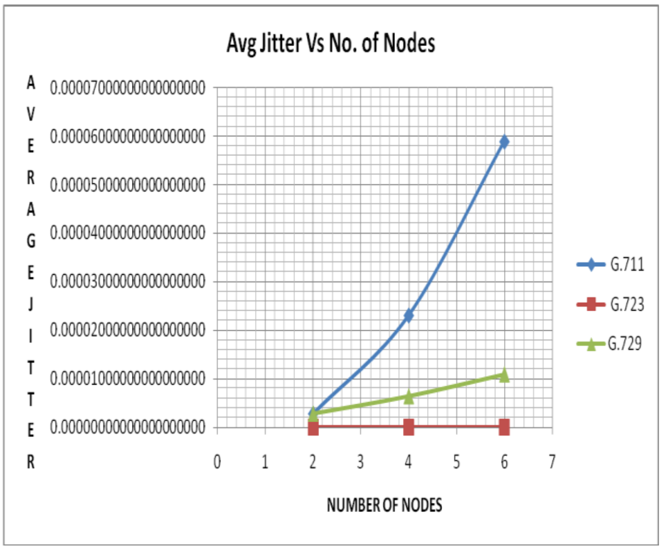

Fig. 3 Average jitter for various VoIP codecs

In case of G.723, jitter is very low and is almost zero for all values of number of nodes.

The value of average delay for each type of codec is mentioned in table V. Fig.4 shows variation of average delay versus number of mobile nodes for different VOIP codecs. It is observed from the graph that delay variation is very high in case of G.711 and the value of average delay is also highest among three codecs under consideration.

TABLE V
AVERAGE DELAY

No. of Nodes	G.711	G.723	G.729
2	0.00034476079197837500	0.00034335820552921300	0.00034476079197837500
4	0.00035491651033132300	0.00034335848100772000	0.00034655414516996100
6	0.00037295546610429200	0.00034335876802218900	0.00034879159978281400

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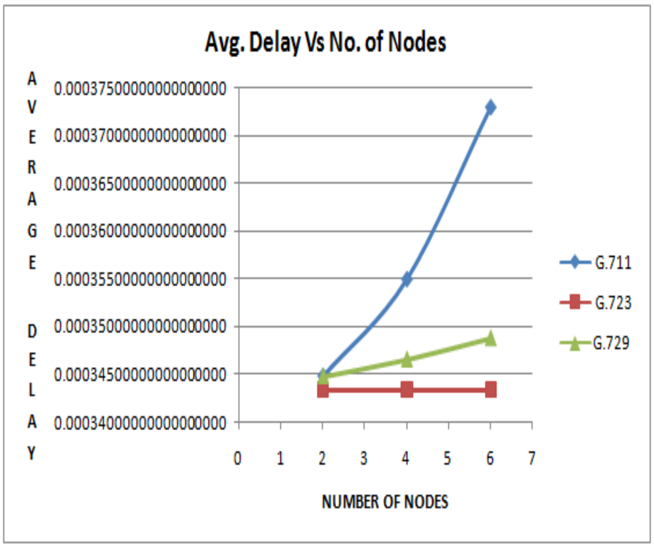

Fig. 4 Average Delay for various VoIP codecs

Average delay increases slowly in case of G.729, whereas in case of G.723, average delay is lowest and is almost constant for increasing number of nodes. When number of nodes increases from 4 to 6, average delay for G.711 increases sharply from 0.000355 to 0.000373. When the number of mobile nodes increases, the BS downlink scheduler is not able to schedule each VoIP packet before the next one is generated from the same application. Hence, the delay variation increases sharply.

I. CONCLUSION

We have assessed best efforts WiMAX service flow for different VOIP codecs by considering various performance parameters like throughput, average delay, and jitter. It can be concluded that G.723 is best codec amongst these three codecs for VOIP application in WiMAX for best efforts service flow. VoIP traffic generates fixed size packets at fixed intervals. Even though the throughput of G.723 is low as compare to the value of throughput in case of G.729, and G.711, average delay and average jitter is almost zero for G.723. The trace files which are generated after running TCL scripts for each of the VOIP codecs are very large in size and consume lot of memory space. The desired information can be extracted by using PERL scripts.

The same experiment can be carried out for different type of service flows in WiMAX. In future we are planning to analyze the performance of remaining VoIP codecs for different types of service flows which are supported by fixed WiMAX.

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