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QoS ANALYSIS OF DIFFERENT SERVICE CLASSES IN WiMAX NETWORK USING VOIP

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Abstract

WiMAX (Worldwide interoperability for micro wave access technology) is a technology based on IEEE 802.16 standards, which provides broadband service over long distances at high speed. A service class is a way of managing traffic in a network by grouping similar types of traffic together and treating each type as a class with its own level of service priority. Various service classes of WiMAX are BE, rtPS, UGS, ertPS and nrtPS. The Voice-Over Internet Protocol (VoIP) technology allows the voice information and the video traffic to pass over Internet Protocol network. Streaming video-content in compressed form passed over the Internet and displayed by the viewer in real time .NOAH is a wireless routing agent that (in similarity to DSR, DSDV, ...) only supports direct communication between wireless nodes or between base stations and mobile nodes in case Mobile IP is used. OPNET simulator is used to analyse the QoS parameters. This simulates scenarios where multi-hop wireless routing is undesired. The QoS parameters are average throughput, average jitter and average delay for analyzes Voice and Video traffic using WiMAX service classes with respect to QoS parameters under different network size. In this paper the QoS of voice in WiMAX is analyzed with the various modulations and coding scheme.

Key words: ***WiMAX, VoIP,QoS, BE, rtPS, UGS, ertPS, nrtPS, OPNET***

I. INTRODUCTION

Many related works were published on analysing different QoS parameters using WiMAX service classes. In [7], Mohamed, Zaki and Elfeki evaluate the performance of different VoIP codecs in WiMAX network with respect to network performance such as MOS (Mean Opinion Score), packet end-to-end delay, jitter and packet delay variation. In [4], the impact of voice codec schemes and statistical distribution for VoIP in WiMAX has been analysed. The simulation results show that better choice of voice codec and statistical distribution have important impact on VoIP performance in WiMAX network. In [3], the paper compares the performance of two different QoS service classes namely UGS and ertPS service classes. Joshi and Jangale [6] focus on analysing the performance of different VoIP codecs using BE service class only in WiMAX network with respect to Qos parameters such as throughput, average delay and jitter. Abid, Raja, Munir, Amjad, Mazhar and Lee [2] analyse the performance of WiMAX network when multimedia contents are transferred using BE and ertPS service classes. Vikram and Gupta [8] analyse the QoS parameters like jitter, throughput, delay, PDR (Packet delivery Ratio) and PLR (Packet Loss Ratio) in WiMAX network using UGS service class. In [9], simulation study was conducted to evaluate the user's QoE (Quality of Experience) when video is streamed from a source to a Mobile Station (MS) via a WiMAX Base Station (BS) in term of the following parameters, namely, the reserved rate at the BS for the video stream, the Modulation and Coding Scheme employed, the distance between BS and MS, and the tolerable end-to-end delay. Zhang, Hu, Le and Nguyen [10], evaluate the transmission performance of multimedia streams, especially SVC (Scalable Video Coding), in mobile WiMAX network by comparing the throughput and the packet delay in different scenarios, and count the frame loss of the received video. The simulation results indicate in terms of frame loss, the number of MS is critical to the performance of video transmission. In the previous work [5], our simulation study was limited on analysing QoS performance for VoIP traffic using UGS, BE and rtPS service classes in term of throughput, jitter and delay. In this paper the QoS of voice in WiMAX is analysed with the various modulation and scheme choose technique. It is found that the better throughput is utilised a changing the modulations and coding scheme.

II. WIMAX TECHNOLOGY

WiMAX is the most recent technology to establish wireless access network. It works as broad band access solutions for wireless communications. WiMAX supports various multimedia applications like VoIP, voice conference and online gaming.

WiMAX technology is a telecommunications technique that offers transmission of wireless data via a number of transmission methods; such as portable or fully mobile Internet access via point to multipoint links.

A. QoS in WiMAX Networks

Mobile WiMAX supports QoS requirements for a wide range of data services and applications by mapping those requirements to unidirectional service flows that are carried over Uplink (UL) or Downlink (DL) connections. QOS is defined as the ability of the network to provide different services for different users with high performance level. It depends on the application and the end user. It can analyze different WiMAX service classes with respect to the QoS parameters such as Jitter, throughput, delay and packet loss.

QoS is the particular concern for the continuous transmission of high-bandwidth video and multimedia information transmitting these kind of content dependably is difficult in public networks using ordinary "best effort" protocols.

QoS requirements become very important in WiMAX technology to guarantee their performance, in the presence of various types of connections, such as current calls, new calls and the handoff connection.

a) Throughput is measure of number of packets successfully delivered in a network. It is measured in terms of packets/second. The throughput value should be high or low it affects every service class defined in WiMAX.
b) Jitter is the variation in the delay time of packets arriving at their destination. VoIP packets sent at regular intervals from the sender to the receiver, but network latency of interval between packets can vary at the destination.
c) Delay or latency could be defined as the time taken by the packets to reach from source to destination. The main sources of delay can be categorized into: propagation delay, processing of source delay, network delay and destination delay. Here calculated 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.

B. WiMAX Network Architecture

WiMAX operates in infrastructure mode. It consists of a base station which sends data to clients, receiver's requests and forwards to the network provider. It can provide various levels of QoS in teams of queuing, scheduling, classification control signaling mechanisms, and routing**.**

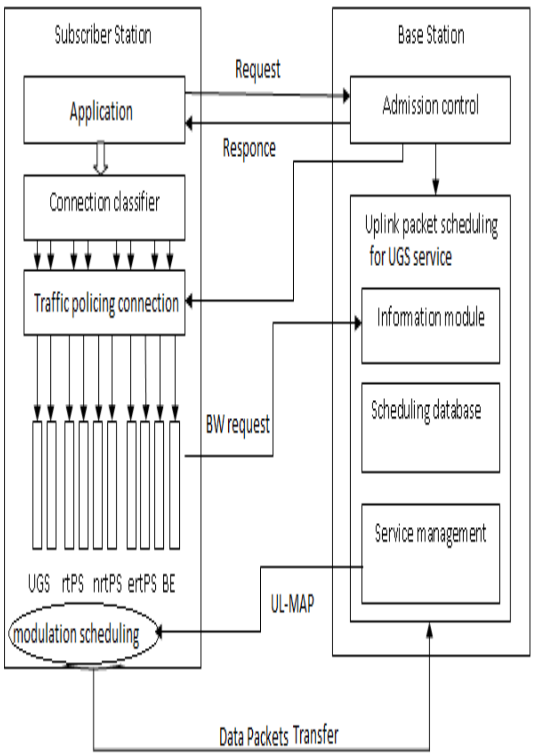

Fig.1. WiMAX network architecture

S .No	Service	Description	Qos Parameters
1	UGS	Support real-time data streams consisting of fixed-size data packets issued at periodic intervals. EX: T1/E1 and Voice over IP without silence suppression	Maximum Sustained Rate , Maximum Delay, Tolerance Jitter Tolerance
2	rtPS	Support real-time data streams consisting of variable-sized data packets that are issued at periodic intervals. EX: moving pictures experts group (MPEG) video.	Traffic priority, Maximum Delay tolerance, Maximum reserved rate
3	ertPS	Support real-time applications. EX: VoIP with silence suppression that have variable data rates but require guaranteed data rate and delays.	Minimum Reserved Rate, Maximum Sustained Rate, Jitter Tolerance ,Traffic Priority
4	nrtPS	Support delay-tolerant data streams consisting of variable-sized data packets for which a minimum data rate is required.	Traffic priority, Maximum reserved rate , Maximum sustained rate
5	BE	The BE is designed to support data streams which do not require a minimum service-level guarantees. EX: The web browsing and the file transfers.	Maximum Sustained Rate, Traffic Priority

Table. 1. IEEE 802.16 Service Classes

C. Different Service Classes in WiMAX

To meet all the different QoS requirements such as jitter, throughput, delay, load, WiMAX utilizes different scheduling mechanisms to allocate downlink and uplink transmission opportunities for the different PDUs. Based on different types of application, WiMAX standard defines five service classes like BE rtPS, nrtPS, UGS and ertPS. The details of the service classes supported by WiMAX are:

III. VOIP TECHNOLOGY

Voice over Internet Protocol is called as Voice over IP or VoIP technology. It has a major impact on the telecommunications of industry. VoIP technology provides advantages for the user and client, allowing calls to be made more cheaply, as well as enabling data and voice data to be carried out over the same network efficiently.

Until recently voice traffic was carried using a circuit switched approach. Here a dedicated by circuit switched to provide a call for a user. Now include new data and Internet style technology used for VoIP, packet data and Internet Protocol (IP) is used to enable a much more efficient use of the available capacity. The most widely used VoIP Codec is

G711, although there are a variety of others codec that are used with varying data rates and providing different levels of voice quality.

IV. VIDEO STREAMING TECHNOLOGY

Video Streaming is a principle used essentially for sending content in near real time, A client media player can play the data (such as a movie) before the entire file has been transmitted.

V. MODULATIONS AND CODING SCHEME

They are 14 types of modulations and coding scheme are available like QPSK 1/2, QPSK 3/4, 16-QAM 1/2, 16- QAM 3/4, 64-QAM ½, 64- QAM 2/3, 64 –QAM 3/4, QPSK1/2, repetition 2, QPSK 3/4, repetition 2, QPSK 1/2, repetition 4, QPSK 3/4, repetition 4, QPSK 1/2, repetition 6, QPSK 3/4, repetition 6.

VI. SIMULATION ENVIRONMENT

A. Simulation Model

In this work, we analyze the performance of VoIP and Video traffic using WiMAX service classes. The OPNET Technologies is a software business that provides performance analysis for computer networks and applications. OPNET stands for Optimized Network Engineering Tools. OPNET simulator is a tool to simulate the behavior and performance of any type of network. This simulator makes possible working with OSI model, from 7 layers. Our simulation consists of creating a number of subscriber stations and connecting them to a base station. A sink node is created and attached to the base station to accept incoming packets.

The OPNET simulator was used in WiMAX network. Parameters as VoIP traffic, Video streaming, number of mobile nodes and WiMAX service classes was passed while running the simulation scenario. For each service classes under consideration, number of mobile is varied from 2, 4, 6, 8, and 10. The main parameters used in our simulation are listed in table 2.

B. Simulation Parameters

Table.2. Simulation parameters

Parameter	**Value**
Network interface type	Phy/Wireless Phy/OFDMA
Propagation model type	Propagation/ OFDMA
Medium Access Control type	Mac/802_16/Base Station
Routing protocol	NOAH
Antenna model	Antenna/Omni Antenna
Link layer type	Logical Link layer
Frame size (msec)	5 (msec)
Duplex scheme	TDD
Packet Rate	4 packet/s
Modulation and Coding Scheme Technique	QPSK 16
Simulation time	30s

C. The Performance Parameters

Our simulation focuses on analyzing the main QoS parameters for WiMAX Network, namely average throughput, average jitter and average delay. Our simulation focuses on the service class UGS. In the paper the UGS is proved to high throughput for voice than the service classes. This paper focuses on different modulations and coding scheme for UGS.

VII. SIMULATION RESULT AND ANALYSIS

Throughput, delay and jitter of rtPS, ertPS, nrtPS UGS and BE service classes using G.711 VoIP codec and H.263 video stream format are analyzed and compared. The figure.2 shows the variations of throughput for VoIP traffic respectively under various WiMAX service classes.

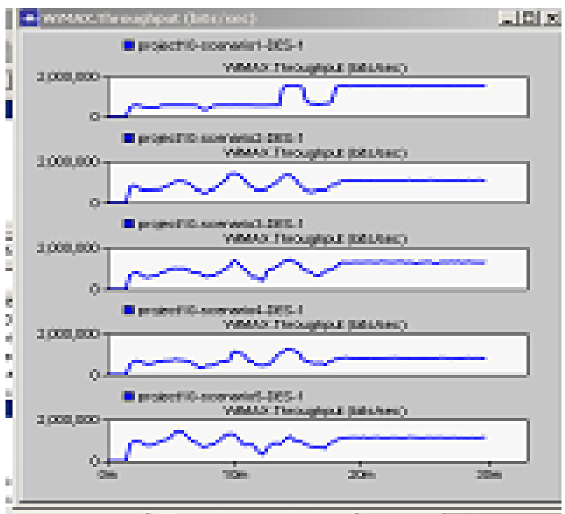

Fig.2. Estimation of Throughput report

The voice parameter jitter is figure.3 shows bellow the rtPS, ertPS, nrtPS, UGS and BE different WiMAX service classes.

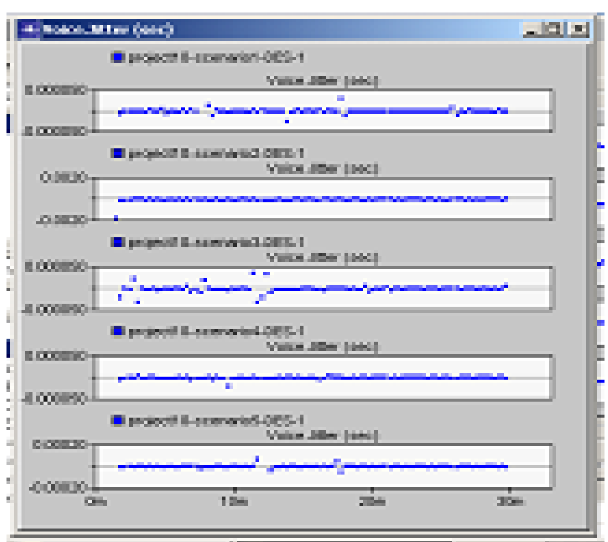

Fig.3. Estimation of jitter and average jitter

On the figure 4 given rtPS, ertPS, nrtps, BE and UGS service classes delay for VoIP traffic.

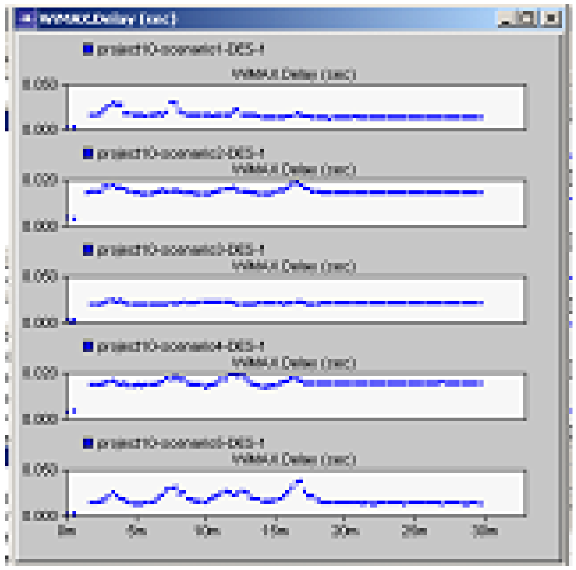

Fig.4. Estimation of delay report

From the 2, 3 and 4 figures, it's observed that UGS service class has the better throughput, lowest average jitter and lowest delay. UGS service class appears to perform better in terms of throughput when analysed the various service classes. However UGS service class has the bandwidth already attributed to transmit data on a periodic basis, even when there is no data being sent. So the network resources are not effectively exploited with UGS service class for VoIP traffic.

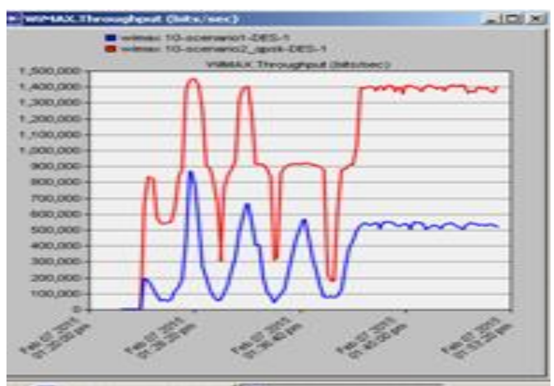

Fig.5. Estimation of modulation and coding scheme

The above figure.5 shows that for different modulations and coding, the throughput of WiMAX different service classes.

VIII. CONCLUSION

In this paper, performances of different service classes such as BE, UGS, rtPS, ertPS and nrtPS service classes have been analyzed using VoIP traffic in terms of throughput, average jitter and average delay. For VoIP traffic, rtPS and ertPS perform better than nrtPS and BE service classes, UGS perform better than all the other service classes in term of jitter, delay and throughput. It can observe that UGS service class has the best performance parameters serving VoIP. UGS service class is dedicated to handle real-time service flows. The frames are generated in fixed sizes at regular interval. The bandwidth can be periodically requested in the rtPS and ertPS service classes instead of fixed. It is concluded that selecting different modulations and coding scheme for the service class UGS provide different throughput. This performance analysis can be improved on selecting appropriate modulation and coding scheme, UGS is will provide better throughput.

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BIOGRAPHIES

Mrs. SairaBanu received the B.E degree in Computer Science and Engineering from the Madras University in 2003.She completed the Master’s in Computer Science and Engineering and received the degree from Anna University in 2006. She is working as Assistant Professor senior Grade, in the department of Computer Science and Engineering in B.S. Abdur Rahman University. She is perusing her research in improving the quality of VOIP Application.

Muthulakshmi. G received the B.E. degree in the Department of Computer Science Engineering from IFET Engineering College, Villupuram, in 2013. She is currently studying toward the M.Tech degree in the Department of Computer Science Engineering in B.S. Abdur Rahman University. Her current research interests in WiMAX enhancement of VOIP application.