A QoS Oriented Analysis of ertPS and UGS flows in voice application over WIMAX

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Abstract- Worldwide interoperability for microwave access (WIMAX) is a IEEE 802.16 wireless communication standard that provides high performance, high throughput and wide coverage area. In addition to quality of service (QoS) support, the IEEE 802.16 standard provides data rate up to 100Mbps and covers area up to 70km. The VOIP through WIMAX is the most prominent service and growing rapidly in the world of telecommunication. Recent studies concentrated on QoS scheduling services and performance related metrics such as throughput, packet end-to-end delay, and jitter. These metrics will be discussed in this paper in addition to a comparison between the Unsolicited Grant Service (UGS) and extended real time Polling Service (ertPS) for voice application. The analysis utilizes the OPNET 14.5 network simulator tool. It is found that the effective scheduling algorithm can provide high service standards to support the QoS required by different type of traffics in addition to different type of users.

Keywords: IEEE802.16, WIMAX, MOS, Qos, Throughput, VoIP, UGS, ertPS.

I. INTRODUCTION

In recent years the evolution of wireless communication systems and networks has sped up at a rare pace and become a vital a part of fashionable life vogue necessities [1]. The demand of high speed information transfer with top quality is taken into account the leading issue for technology evolution as in WIMAX and still growing. The WIMAX may be a commonplace for wireless communications that supports higher range of users with higher information rates, coverage, and accessibility. This technology relies on IEEE 802.16 standards and therefore the most relevant versions ar the mounted WIMAX supported IEEE 802.16d, and therefore the mobile WIMAX outlined by IEEE 802.16e [2]. Moving towards the fourth generation communication networks the integrated networks are used. Similarly, the VOIP is predicted to be a coffee value communication medium

IEEE 802.16 QOS

A basic WIMAX network includes a base station (BS) and a mobile subscriber stations (MSSs). The BS schedules the traffic flow and therefore the communication between BS and Selective Service is bifacial, downlink channel (BS to SS) is in broadcast mode, and therefore the transmission channel (SS to BS) is shared by varied SSs[3]. The IEEE 802.16 commonplace supports two kind of duplex mode; these are the Time Division Duplex (TDD) and therefore the Frequency Division Duplex (FDD).

The Time Division Duplex (TDD) frame consists of downlink (DL) and transmission sub frames. The period and range of sub frame slots are determined by the scheduler of BS. The downlink (DL) sub frame has DL map contains data concerning the period of sub frames. It additionally contains details on which period slot belongs to a specific SS. The downlink channel and transmission map (UL map) consists of (information element IE) which has transmission opportunities [4].
A. MAC-Layer overview
The WIMAX MAC layer provides an interface between the upper transport layers and therefore the physical layer. The layer takes service data unit MSDU packets from the higher layer and organizes them into Mac protocol data units (MPDUs) for transmission and sent transmission. Its style includes a convergence sub layer which will interface with a range of higher-layer protocols, cherish ATM, TDM, Ethernet, IP, and any future protocol. Additionally to providing mapping to and from higher layers, the metal supports MSDU header suppression to scale back the upper layer overheads [5].

B. Quality of Service (QoS)
Supporting the QoS may be a basic part of designing WIMAX MAC-layer. WIMAX borrows a number of the essential ideas behind the designing of QoS from the data Over Cable Service Interface Specification (DOCSIS) cable electronic equipment commonplace. In the architecture of MAC connection the traffic between BSs and SSs transmitted within the context of connection. Each connection is defined within the (MPDU) by a connection ID that additionally provides a mapping to a service flow ID (SFID). The SFID is a crucial idea within the Mac layer that provides a mapping to the QoS parameters for a specific information entity [1]. To support several applications over WIMAX, the IEEE 802.16 defines five QoS categories for WIMAX:

1. Unsolicited Grant Service (UGS)
   This service class is built to support real-time streams consisting of fixed-size data packets issued at periodic intervals. The BS provides data grant with fixed-size at periodic intervals, just like the case in E1 and VOIP while no silence suppression [3].

2. Real time Polling Service (rtPS)
   This service class is created to support real-time streams consisting of data packets with variable size that are issued at periodic intervals. The BS provides periodic unicast uplink (UL) request opportunities, just like the case in MPEG video transmission [3].

3. Extended Real-Time Polling Service (ertPS)
   This class is appropriate for variable rate of real time applications that have rate and delays necessities, just like the case in VOIP while no silence suppression. The IEEE 802.16e commonplace indicates that ertPS is made upon the potency of each UGS and rtPS. [3].

4. Non-Real-Time Polling Service (nrtPS)
   nrtPS support delay tolerant data streams consisting of data packets with variable size that a minimum rate is needed, just like the case in FTP traffic. The BS provides unicast transmission request polls on an everyday basis, that guarantees that the service flow receives request opportunities even throughout network congestion [3].

5. Best Effort (BE):
   BE class support data streams that no minimum service guarantees are needed, just like the case in HTTP traffic. The BS doesn't have any unicast transmission request polling obligation for BE Selective Service. Therefore, a protracted amount will run while not sending any BE packets [3].

Table 1: classifies different service classes defined in WIMAX and its description and QoS parameters.
Table 1. QoS service classed in WIMAX

<table>
<thead>
<tr>
<th>Service</th>
<th>QOS Parameters</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>UGS</td>
<td>Maximum sustained rate, maximum latency tolerance</td>
<td>Designed to support real time constant bit rate (CBR) traffic such as VoIP that periodically generates fixed size data packets.</td>
</tr>
<tr>
<td>RTPS</td>
<td>Maximum reserved rate, maximum sustained rate, maximum latency tolerance, traffic priority.</td>
<td>Designed to support MPEG video &amp; Teleconferencing that periodically Generates variable size data packets.</td>
</tr>
<tr>
<td>NRTPS</td>
<td>Minimum reserved rate, maximum sustained rate, traffic priority.</td>
<td>Designed to support non real time application with minimum rate such as ftp.</td>
</tr>
<tr>
<td>BE</td>
<td>Maximum sustained rate, traffic priority.</td>
<td>Designed to support data stream which do Not require minimum service level guarantee. This QOS is used for internet service such as email and web browsing.</td>
</tr>
<tr>
<td>ERTPS</td>
<td>Minimum reserved rate, maximum sustained rate, maximum latency tolerance</td>
<td>Designed to support real time application, such as VoIP with silence suppression that have variable data rate.</td>
</tr>
</tbody>
</table>

C. MOS
The MOS is a technique used to check the work of codecs which compress audio and video files. The MOS of a particular codec is the standard mark given by a panel of auditors listening to various recorded samples. This will range from 1 (unacceptable) to 5 (excellent). The MOS values depend on several factors, not only the network parameters such as delay and packet loss but also on codec used [7]

Table 2 shows relationship between MOS and quality of speech

<table>
<thead>
<tr>
<th>Quality of speech</th>
<th>MOS</th>
</tr>
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<tbody>
<tr>
<td>Excellent</td>
<td>5</td>
</tr>
<tr>
<td>Good</td>
<td>4</td>
</tr>
<tr>
<td>Fair</td>
<td>3</td>
</tr>
<tr>
<td>Poor</td>
<td>2</td>
</tr>
<tr>
<td>Bad</td>
<td>1</td>
</tr>
</tbody>
</table>

Throughput - Technically defined as measurement of amount of data carried by the link and it arrive without error and loss
Packet delay variation - variation in delay of packets across Internet paths (difference in delay)
Packet end to end delay - The time taken by voice to be transmitted from source to destination is called packet end-to-end delay.
II. METHODOLOGY

To emulate the WIMAX networks and investigate the QoS parameters and classes, two scenarios are selected for WIMAX networks. One scenario is chosen to represent the ertPS configuration and the other to represent the UGS configuration. For both cases, the application running on these networks is assumed to be voice over IP (VOIP) in addition to other applications. The emphasis will be on the voice application. Fig.1. shows the network scenarios simulated by the OPNET 14.5 network simulator tool.

III. RESULTS AND ANALYSIS

Fig.2. shows the average packet end-to-end delay for the UGS and ertPS configurations. It can be noticed that the UGS (shown in red) shows a better delay performance when compared to the ertPS (shown in blue). This can be attributed to the UGS being designated to support constant bit rate (CBR) traffic and the voice application periodically generates fixed size performance data packets, hence leading to better delay.
Fig. 3. shows the packet delay variation (Jitter). It can be noted that the UGS shows an enhanced jitter performance when compared to the ertPS. This can be attributed to variable data rate supported by ertPS technique, which results in high jitter related for ertPS traffic when compared to UGS traffic.

![Fig (3): the average packet delay variation](image)

Fig. 4. shows the average UL packet dropped for the UGS and ertPS. The UGS shows lower packet drop when compared to the ertPS. This can be attributed to packet scheduling in the uplink direction at the BS which is more defying than the scheduling at the downlink direction. This can get complicated when all of the QoS parameters defined by the standard are taken into consideration, in addition to no direct access to the connection queues. The uplink scheduler is related to the bandwidth requests that are sent by the SSs in order to retain information on each connection status. Hence, the bandwidth requests can suffer from delays that can be generated by the contention mechanism. The UGS reserves the bandwidth during setup time, poll-me bit for unicast polling, no bandwidth stealing, and it provides fixed bandwidth allocation on periodic basis. Once the connectivity is established, no more requests are needed.

![Fig (4): average UL packet dropped](image)

Fig. 5. shows the point-to-point throughput of the UGS and the ertPS from server to BS and vice versa. The UGS shows an improved throughput than the ertPS due to Static allocation type, grant equal to...
MST (Maximum Sustained Traffic) Rate for UGS traffic flows. As for the ertPS the BS has dynamic allocation and it needs to poll mobile subscriber to know whether silent period has been ended or it is continuing.

![Graph](image)

**Fig (5): average point-to-point throughput**

Fig. 6. shows the congestion with queuing delay in the UGS and ertPS. The UGS has very small queuing time when compared to the ertPS. For mobile has an existing uplink unsolicited grant service (UGS) connections, there is special request mechanism exists for mobile used UGS connection to request extra bandwidth.

![Graph](image)

**Fig (6): average point-to-point queuing delay**

The slip indicator (SI) bit which used to indicate that the transmit queue buffer (for the UGS connection) is overflowing and it needs more bandwidth to relieve the queue. Then the BS may grant a small additional bandwidth up to 1% to the mobile.

Fig. 7. shows the mean opinion score (MOS) where the UGS is performing better than ertPS service flows in terms of voice applications as shown in the figure below.
CONCLUSIONS
We proposed that the mobile WiMAX can be used to provide voice over-IP services and wireless connectivity in addition to high speed internet activities. The network simulator OPNET 14.5 was employed to run simulations for scenarios utilizing the UGS and ertPS techniques. The qualities of average packet end-to-end delay, packet delay variation (Jitter), average UL packet dropped, point-to-point throughput, the congestion with queuing delay, and mean opinion score (MOS) were simulated and results were presented and compared between UGS and ertPS. From the above results we find that the UGS has better performance than ertPS in addition to the end to end delay as shown in the above results because of constant bit rate (CBR).

REFERENCES
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