Quality of Service (QoS) in Wireless Network, VOIP Simulation Environment

Ashish Dubey
M.Tech Scholar, Department of Computer Sci and Engineering, Gurgaon Institute of Technology and Management, Bilaspur, Gurgaon, India
er.ashish.dubey@gmail.com

Abstract- A Quality of Service (QoS) is determines if a wireless technology can successfully deliver high value services such as voice and video. The PHY and MAC layers of 802.16 are described in detail with regarding their QoS. The relations and interactions of these QoS mechanisms are described to give an understanding them. The UGS service flow can handles the traffic generated by VOIP calls in the most optimum way. Wireless networking has the potential for use in broadband Internet services, video and audio streaming, and as an alternative to the PSTN for voice service. The 802.16 has to handle the Requirements of very-high-data-rate applications, such as voice over IP (VoIP) and video or audio streaming, as well as low-data-rate applications, such as web surfing, and handle extremely bursty traffic over the Internet. Voice IP (VoIP) over wireless local area network (WLAN), what are the performance limitations in establishing VoIP calls over WLAN networks we performed some simulations. Simulation is a powerful tool for analysis and improvement of networking technologies, and many simulation packages are available.

Keywords- QoS, VOIP, IEEE 802.16 Standards.

I. INTRODUCTION
The WiMAX forum and IEEE 802.16 subcommittee are both involved in the development of open standards based broadband wireless networks. The IEEE 802.16 subcommittee is purely a technical body that defines the 802.16 family of broadband wireless radio interface standards. IEEE 802.16 defines the layer 1 (physical, also referred as PHY) and layer 2 (data link or Media Access Control – MAC) of the (Open Systems Interconnection) OSI seven layer network model. It does not define standardized network architecture beyond the base station. The Standardized network architecture is necessary to ensure inter-working between equipment from different vendors and inter-working between networks of different operators.

II. LITERATURE REVIEW
The IEEE 802.16 MAC layer performs the standard Medium Access Control (MAC) layer function of providing a medium-independent interface to the physical (PHY) layer. WiMAX systems are based on Orthogonal Frequency Division Multiple Access (OFDMA). Scalable OFDMA (OFDMA) is introduced in the IEEE 802.16e amendment to support scalable channel bandwidths. The MAC protocol is connection-oriented. All data transmissions take place in the context of connections.

III. MECHANISMS
The 802.16 standard includes several QoS mechanisms at the PHY layer, such as Time Division Duplex(TDD), Frequency Division Duplex (FDD) and Orthogonal Frequency Division Multiplexing (OFDM). Each can help in providing QoS.TDD can dynamically allocate uplink and downlink bandwidth, depending on their requirements. This is illustrated in Figure. Each 802.16 TDD frame is one downlink subframe and one uplink subframe, separated by a guard slot. 802.16 adaptively allocates the number of slots for each, depending on their bandwidth needs.

IV. SERVICES CLASSIFICATION
The main feature of 802.16 QoS provisioning, and what distinguishes it from its competitors (i.e. 802.11 and 3G), is that it associates each packet with a service flow. 802.16 is a connection-oriented MAC.
Each connection is assigned a unique Connection ID (CID) and a Service Flow ID (SFID) with an associated service class. The upper part of the MAC maps data into a QoS service class.

802.16 provides four scheduling service QoS Service Description

<table>
<thead>
<tr>
<th>QoS Service</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unsolicited Grant</td>
<td>Supports CBR Services</td>
</tr>
<tr>
<td>Service(UGS)</td>
<td></td>
</tr>
<tr>
<td>Real Time Polling</td>
<td>Supports real time services with variable size data on a periodic basis</td>
</tr>
<tr>
<td>Service(rtps)</td>
<td></td>
</tr>
<tr>
<td>Non Real Time</td>
<td>Support non-real time services with require variable size data burst on a</td>
</tr>
<tr>
<td>Services(nrtps)</td>
<td>regular basis</td>
</tr>
<tr>
<td>Best Effort(BE)</td>
<td>For application that do not require QoS</td>
</tr>
</tbody>
</table>

QoS Service Description

- Unsolicited Grant Service (UGS) Supports CBR Services
- Real Time Polling Service (rtps)
- Non Real Time Services (nrtps)
- Best Effort (BE) For application that do not require QoS

VI. NETWORK ARCHITECTURE

The WiMAX End-to-End Network Systems Architecture document defines the WiMAX Network Reference Model (NRM). It is a logical representation of the network architecture. The NRM identifies functional entities and reference points over which interoperability is achieved. The architecture has been developed with the objective of providing unified support of functionality needed in a range of network deployment models and usage scenarios.

VII. SIMULATION SETUP

The simulation setup would reflect the actual deployment of the WiMAX network. It is based on the network reference model. Figure shows the setup that will be used. There are multiple SS’s in the range of a base station. The base station is connected to the core network. The focus of analysis will be the connection between the subscriber and the base station. Various types of traffic can be set up from the subscriber station to real life scenarios.

VIII. SIMULATION DETAILS

Different approaches were considered in the beginning of the work. He works in event-driven simulation in C++. The simulator core is written in C++. It has an OTcl (Object Tool Command Language) interpreter shell as the user interface and allows input models written as Tc1 (Tool Command Language) scripts to be executed. Perl script used in this simulation.

IX. QoS PARAMETERS
QoS provisioning encompasses providing Quality of Service to the end user in terms of several generic parameters. In the analysis, the throughput, average delay, average jitter and packet loss were considered.

1. Throughput

Throughput is a measure of the date rate (bits per second) generated by the application. Equation 1 shows the calculation for throughput $TP$, where $\text{PacketSize}_i$ is the packet size of the $i$th packet reaching the destination, $\text{PacketStart}_0$ is the time when the first packet left the source and $\text{PacketArrival}_n$ is the time when the last packet arrived.

$$TP = \frac{\sum_i \text{PacketSize}_i}{\text{PacketArrival}_n - \text{PacketStart}_0}$$

Equation 1 Throughput Calculation

2. Average Delay or latency

Delay or latency would be time taken by the packets to transverse from the source to the destination. Equation 2 show the calculation for Average Delay, where $\text{PacketArrival}_i$ is the time when packet “$i$” reaches the destination and $\text{PacketStart}_i$ is the time when packet “$i$” leaves the source. “$n$” is the total number of packets.

$$\text{Average Delay} = \frac{\sum_i \text{PacketArrival}_i - \text{PacketStart}_i}{n}$$

Equation 2- Average Delay

3. Jitter or Delay variation

Equation 3 shows the steps for calculation of average jitter. It is the average of the absolute difference in the time it took for successive packets to reach the destination.

$$\text{Average Jitter} = \frac{\sum (\text{PacketArrival}_{i+1} - \text{PacketArrival}_i) - (\text{PacketArrival}_i - \text{PacketArrival}_{i-1})}{n-1}$$

Equation 3- Average Jitter

4. Packet loss or corruption rate

Packet loss affects the perceived quality of the application. Several causes of packet loss or corruption would be bit errors in an erroneous wireless network or insufficient buffers due to network congestion when the channel becomes overloaded.

$$\text{Packet Loss} = \frac{\sum \text{LostPacketSize}_i}{\sum \text{PacketSize}_i}$$

Equation 4- Packet Loss

X. VoIP TRAFFIC

We use a 64 Kbps half-duplex UDP flow. To generate a 64 Kbps flow, 64 bytes packets were sent every 8 milliseconds.

<table>
<thead>
<tr>
<th>Name</th>
<th>Throughput in kbps</th>
<th>Payload in bytes</th>
<th>Packet Rate in packets/sec</th>
</tr>
</thead>
<tbody>
<tr>
<td>G.711</td>
<td>64</td>
<td>160</td>
<td>50</td>
</tr>
<tr>
<td>G.729</td>
<td>8</td>
<td>20</td>
<td>50</td>
</tr>
<tr>
<td>G.729,2</td>
<td>8</td>
<td>30</td>
<td>33</td>
</tr>
<tr>
<td>G.723</td>
<td>6.3</td>
<td>30</td>
<td>26</td>
</tr>
<tr>
<td>G.723,2</td>
<td>5.3</td>
<td>30</td>
<td>22</td>
</tr>
</tbody>
</table>

Table Details for VOIP Codecs

XI. VOIP traffic over Best Effort Service Flow

In the first case the VOIP traffic is setup over Best Effort (BE) service flow. Fig shows the variation of throughput, percentage packet loss, average jitter, and average delay as the number of nodes increase from 2 to 10.

Each node is transmitting 160 byte packets at the rate of 50 packets per second as per the G.711 specifications For nodes, the value reaches around 519 kbps. This is because of the packet loss. The variation of percent packet loss.
When number of nodes increase and the number of packets transferred increases. These include data packets as well as control packets that are exchanged between the SS and BS.

Fig shows the average jitter for BE traffic with G.711 codec. The average jitter is seen to increase steadily. Jitter is the measure of the variability over time of the latency across a network. The packets are not received with the same delay at the receiver, the voice quality appears to be degraded.

**XII. VOIP APPLICATIONS**

VoIP uses a number of protocols which ensure that voice communication is appropriately established between parties, and that voice is transmitted with a quality close to that we are accustomed to in the PSTN. It uses signaling protocols such as the Session Initiation Protocol (SIP) and H.323. The R-factor is related in a non-linear fashion to the MOS through the following equation:

\[
MOS = 1 + 0.035 \times R + 7 \times 10^{-6} \times R \times (R - 60) \times (100 - R)
\]

The relationship of the R-factor values to the MOS and the typical categorization of the R-factor values are presented in Table. It can be seen that connections with R-factors of less than 60 are expected to provide poor quality, whereas R-factors of 80 and above provide high quality.

<table>
<thead>
<tr>
<th>R-factor</th>
<th>Quality of Voice Rating</th>
<th>MOS</th>
</tr>
</thead>
<tbody>
<tr>
<td>0 &lt; R &lt; 100</td>
<td>Best</td>
<td>4.34 – 4.50</td>
</tr>
<tr>
<td>80 &lt; R &lt; 90</td>
<td>High</td>
<td>4.03 – 4.34</td>
</tr>
<tr>
<td>70 &lt; R &lt; 80</td>
<td>Medium</td>
<td>3.60 – 4.03</td>
</tr>
<tr>
<td>60 &lt; R &lt; 70</td>
<td>Low</td>
<td>3.10 – 3.60</td>
</tr>
<tr>
<td>50 &lt; R &lt; 60</td>
<td>Poor</td>
<td>2.58 – 3.00</td>
</tr>
</tbody>
</table>

**XIII. CONCLUSION**

The IEEE 802.16/WiMAX network architecture was presented and the MAC layer features that enable end-to-end QoS mechanism in the network were discussed. Various service flows that are supported in WiMAX were discussed in details. Related literature survey was presented in detail to provide background to the reader. VOIP traffic and video streaming traffic was analyzed using a simulation based on network simulator, ns-2. The effect of different service flows on QoS parameters like throughput, packet loss, average jitter and average delay was studied. In general, it was observed that the UGS service flow has the least overhead in terms of bandwidth request and it is the highest in RTPS service flow.

**XIV. FUTURE SCOPE**

VOIP traffic and video streaming were the two applications considered in the current analysis. Further analysis could be done for other applications including, video telephony which combines video traffic and VOIP traffic, File Transfer Protocol (FTP) traffic etc. The WiMAX module used in the analysis did not support nrtPS and ertPS service flows that are defined by the IEEE 802.16 standards. ertPS service flow is designed for applications which generate variable rate traffic which are delay dependent. An example of such traffic is VOIP with silence suppression. In this case, the VOIP application is not required to send packets during silent periods. The capability to stop sending packets during silent periods is known as "Silence Suppression" or VAD (Voice Activity Detection). Variable rate and non real time applications such as FTP are supported by the nrtPS service flow.
REFERENCES


