CHAPTER 4

AN IMPROVED CROSS LAYER SCHEDULING ALGORITHM FOR QoS ENHANCEMENT

4.1 INTRODUCTION

The VoIP uses the existing IP infrastructure and provides a flexible and a low cost alternate to normal telephone services. Customers are switching over from traditional communication methods like email and web browsing to feature rich multimedia applications such as VoIP and video conferencing and video streaming which are made available through the new broadband technologies. Service providers are constantly looking for better and cheaper alternative methods for delivering the multimedia applications to users using the broadband wireless network. VoIP over WiMAX has been emerging as one of the key technologies for providing broadband wireless voice service with cost efficient reliability and guaranteed QoS. Many challenges are involved in supporting VoIP over WiMAX network as compared to the traditional wire Digital Subscriber Line (DSL) network. Hence the VoIP over WiMAX implementation needs to address key deployment issues such as the network architecture, system design, network capacity, configurations and QoS provisioning.

The objective of this study is to,

1. Evaluate the performance of VoIP CODEC over the BE scheduling class in WiMAX network.
2. Analyze the capability of a WiMAX network to deliver sufficient QoS to voice and data applications under different network conditions and load.

The simulation model does the following tasks for validating the objectives:

1. WiMAX network creation and deployment
2. Configuring the required applications for streaming video, FTP and VoIP telephony
3. QoS parameters configuration
4. Multiple simulations runs are performed using the NS-2 based simulator for different CODECS under different packetization time, buffer size and with different network load.

4.2 VoIP SYSTEM

VoIP is a transmission technology, which transmits the voice over internet protocol. One of the examples of VoIP applications is Skype. The VoIP is flexible, and provides low cost telephony to customers over the existing IP infrastructure. However, there are still many challenges that need to be addressed, to provide a steady and good quality voice connection over the Internet. In this research work, the performance of different VoIP CODECS over the WiMAX network has been evaluated. The IEEE WiMAX network’s QoS performance metrics, such as jitter, delay and packet loss and throughput have been used to evaluate the performance of the VoIP CODECS. The simulation is performed in the NS-2 simulator, with varying values of packet size, number of calls and jitter buffer sizes. Our results
indicate that varying the jitter buffer size and packetization time affect the quality of voice over the network.

Google talk as a reference VoIP application to evaluate the performance under WiMAX networks has been extensively studied by (Iwan Adhicandra 2010). All these VoIP flows are running over the scheduling class which has the lowest priority among all scheduling classes, because the Google talk application initiates the HTTPs session, and secondly, due to the freely availability and much popularity among the broadband users. The application referred to uses different types of audio CODECS such as G711, G729, G723 etc. to provide good voice quality to users; the CODEC selection depends on the available BW. To evaluate the VoIP CODECS performance in the WiMAX network, we take G711, G726, G728, G729 and G723 with data rates of 64Kbps, 32Kbps or 24Kbps, 16Kbps, 8Kbps and 5.3 Kbps respectively. For those CODECS, the standard voice frame’s duration is used during the whole simulation time.

VoIP application works with sender and receiver components as follows:

**Sender Side**

1. Voice signal is sampled, digitized and encoded by the algorithm.
**Receiver Side**

1. The received data is de-packetized and processed by the payout buffer.
2. The delay incurred in the network has been smoothed.
3. Data is decoded and the voice signal is reconstructed.

**4.2.1 VoIP QoS**

The service quality of a VoIP is to ensure that voice packets are not delayed, lost, or dropped, during transmission over the network, since the VoIP application is delay sensitive and loss sensitive. The general VoIP QoS parameters are packet loss, jitter and packet delay and bit error rate. The higher delay and loss lowers the QoS of VoIP leading to decline in user satisfaction. Delay has the greatest impact on the throughput of the VoIP which can occur for different reasons like network congestion as discussed by Alshakhs & Hasbullah (2010). Reducing the layer interaction or combining the different layer protocols together would help to minimize the transmission delay, which improves the VoIP QoS.

There are many existing approaches designed to improve the performance and quality of wireless networks for real time applications. Past studies and surveys show that the cross layer interaction approach had a greater impact on the QoS requirements of wireless networks. As discussed in earlier chapters, the cross layer design allows the neighboring and non-neighboring layer to exchange information with each other, to attain optimal results.
4.4 PROPOSED CROSS LAYER SCHEDULING ALGORITHM

The cross layer scheduler is modeled as a channel aware scheduler. The main focus of the cross layer design is to provide the best possible end-to-end performance for the applications. The objective is to maximize the total throughput, when satisfying the QoS requirements of different service classes.

Figure 4.1 Cross layer diagram

The proposed scheduling algorithm modifies the Shuaibu et al (2010) cross layer algorithm, which incorporates the SNR value and the minimum required throughput of the SS in its formulation. The cross layer interaction is shown in Figure 4.1. The SS with the highest priority is selected to transmit the frame. The priority of the SS is calculated, based on the traffic class it belongs to. The cross layer scheduling algorithm will improve the performance of the WiMAX networks. The main goal of the scheduling algorithm is to minimize the packet loss and maximize the throughput.
4.4.1 Scheduling Procedure

1. Define a higher priority queue

2. Schedule the bandwidth request opportunities, which should be scheduled in the next frame

3. Check the deadline for the service flow periodically

4. Check the minimum BW availability

5. Resources should be periodically distributed among the service flows, according to the deadline.

The algorithm is executed at the BS, at the beginning of every frame; thereby, the priority is assigned to each SS. The cross layer algorithm proposed by Najah Abu Ali et al (2009) has the following drawbacks. The scheduler allocates all the available slots to the high priority packets which ultimately reduces the BW utilization. When multiple connections have same priority value the scheduler randomly chooses the packet. The modified cross layer scheduling algorithm overcomes these drawbacks in the following ways, and efficiently manages BW allocation.

1. Required slots are allocated to the higher priority packets and not only to one packet

2. Multiple packets have the same priority, so the one that arrived first has been picked up to decrease the delay.

3. Fragmentation is done for the service types to make use of the available slots, except the eTTPS connection in the WiMAX frame.
The algorithm not only prevents the high priority service class from occupying more BW but also adjusts the amount of scheduling data to reduce the packet loss rate according to the MCS in the following Table without starving the low priority connections.

A base station centrally controls the transmission of packets in both the directions. All packets from the higher layers are classified as service flows in the base station. Each of the service flows are with diversified QoS requirements. A service flow is a unidirectional flow of packets with a set of QoS parameters. The QoS parameters include traffic priority, maximum sustained traffic rate, maximum burst rate, minimum tolerable rate, scheduling type, minimum delay and so on.

All subscriber stations are uniformly distributed around the BS and transmit their packet during the uplink. Each transmitted packet arriving at the BS is given a serial number, service flow identifier, packet size, arrival time and SNR. The scheduler works, as shown in Figure 4.2.

![Scheduler Diagram](image)

**Figure 4.2  Scheduler**

Three different buffers were used, one for each service flow. Each buffer has length t and each packet received in the uplink session is stored in the buffer with a serial number, service flow identification, SNR, arrival time
and packet size. The responsibility of the scheduler is to visit each buffer during the downlink subframe, and to schedule the packets based on the proposed algorithm.

Based on the SNR, the type of modulation can be chosen.

4.4.2 SS Uplink Unit

This unit is responsible for allocating the data uplink slots for connections. These slots are used by the SS to send the data in the uplink direction. The BS wants to know the queue information of connections at each SS. The BS knows the queue information of the connections from the previous frame status. The BS allocates slots to the corresponding CID and they are related to enough number of slots left in the uplink sub frame.

An SS requests the BS for BW for uplink. The BS grants total BW for all the connections belongs to SS. The SS redistributes the sum total of BW among its users according to service class of the connections and its stringent QoS requirements.

4.5 EXPERIMENTAL SETUP

4.5.1 Cross-Layer Interaction Framework

In this section, creation and management of the cross layer message using the NS-MIRACLE framework has been discussed. With these messages we are able to establish communication among modules of the node. They could be a request of some statistics from a layer ("get the channel condition") or a request to perform a particular action in another layer ("turn on radio" or "Start handover"). The cross layer interaction framework has been shown in the Figure 4.3. In the research framework, there are two kinds of messages. These are Synchronous and Asynchronous messages. Both of them are
created from the ClMessage class, but they differ in the way they are propagated within the node.

Asynchronous messages are very similar to an ordinary packet and a copy of the message is sent to each target module/plugIn. Synchronous messages are a bit different, because the same message is sent to all the target modules. This means that the message is temporarily shared by the sender and the receivers and therefore the instruction flow returns directly to the source when all the destinations have been finished.

The two message types acquire the synchronous or asynchronous nature, by the method used to propagate them among the nodes: you have at disposal the sendAsyncClMsg (Up/Down) and the sendSyncClMsg (Up/Down) PlugIn and Module classes methods.
The message which is used in this research which is to be described in detail the synchronous message, often used to retrieve information from different layers. Synchronous means that the query and reply are in the same message, and therefore, the answer is given synchronously by the receiver.

To send a cross layer message the NS MIRACLE inherits a reference class named ClMessage. The ClMessage class is shown in the Figure 4.4. Extending these classes, customized methods can be created to manage our own messages. In this example, an application layer has been considered, and the objective is to collect some information stored at layer 3.

4.5.2 Simulation of VoIP Model

As shown in the following Figure 4.5, the simulation model is based on a PMP model, with multiple SSs connected to a centralized BS node. The nodes are implemented, using the MIRACLE framework, and the
simulation data has been analyzed by utilizing the tracing module in the MIRACLE framework.

Figure 4.5 VoIP MIRACLE framework

4.5.3 Simulation Platform

The cross layer scheduler proposed in this study, was implemented in the IEEE 802.16 module in the NS-2. The NS-2 is a widely used tool for the simulation of packet switched networks. It gives huge support for the simulation of TCP routing and MAC protocols, over wired and wireless networks. The network elements in the NS-2 simulator are developed as classes in an object oriented manner. It has an OTCL interpreter for easy user interface and has input models which are written in TCL scripts. A BS and a SS can be set up as a node in NS-2. When the number of nodes increases the amount of packets received and sent increases. For a single node
configuration the simulation would run fairly. But as the number of nodes increases the packet traffic will also thicken.

4.5.4 **Simulation Parameters**

The simulated network uses a PMP topology with a centralized BS and the SS. The distance between the SS and BS ranges from 1600 to 1800 meters. In our simulation, for sending the BW request from all SSs, unicast polling is used. Here, the Grant Per Subscriber Station (GPSS) BW allocation scheme is used. In the simulation the number of calls generated by SSs is varied, and is randomly generated.

Each transmitted packet has its own estimated SNR value as shown in Table 4.1.

**Table 4.1 MCS and receiver SNR**

<table>
<thead>
<tr>
<th>S.No</th>
<th>Modulation</th>
<th>Coding rate</th>
<th>SNR(dB)</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Quadrature Pulse Shift Keying (QPSK)</td>
<td>1/2</td>
<td>5.0</td>
</tr>
<tr>
<td></td>
<td></td>
<td>3/4</td>
<td>8.0</td>
</tr>
<tr>
<td>2</td>
<td>16-Quadrature Amplitude Modulation (QAM)</td>
<td>1/2</td>
<td>10.5</td>
</tr>
<tr>
<td></td>
<td></td>
<td>3/4</td>
<td>14.0</td>
</tr>
<tr>
<td>3</td>
<td>64-QAM</td>
<td>1/2</td>
<td>16.0</td>
</tr>
<tr>
<td></td>
<td></td>
<td>2/3</td>
<td>18.0</td>
</tr>
<tr>
<td></td>
<td></td>
<td>3/4</td>
<td>20.0</td>
</tr>
</tbody>
</table>
The simulation parameters settings are shown in Table 4.2. The base station receives all transmitted packets from the subscriber stations; assigns a packet serial number, packet service flow identification and arrival time, and stores the packet in an appropriate buffer of the service flow.

Table 4.2  VoIP G.711 simulation parameters

<table>
<thead>
<tr>
<th>Parameters</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Physical layer</td>
<td>Wireless MAN-OFDM,TDD</td>
</tr>
<tr>
<td>No of OFDM symbols and sub channels</td>
<td>19,32</td>
</tr>
<tr>
<td>Bandwidth and frame duration</td>
<td>10 MHz and 5 ms</td>
</tr>
<tr>
<td>Minimum resource allocation unit(slot)</td>
<td>2 OFDM symbols in time, 1 subchannel in frequency</td>
</tr>
<tr>
<td>Max PDU size</td>
<td>2048 byte</td>
</tr>
</tbody>
</table>

The BS schedules the packets based on the cross layer scheduling algorithm, during the downlink session. According to the values of the packet size and SNR value, the required number of slots are allotted for each of the packets. If the required number of slots on the current frame is not enough to schedule the current packet, then the packet is lost. The buffers are used for handling the different service flows. Each buffer can store 250 packets at a time. If the buffer is full and there is a packet in the queue the packet is considered to be lost since there is no memory to hold it. Once the packet is scheduled, it should be removed from the buffer, and the memory is considered to be empty to store the next packet. The uplink duration is 4.5 ms and the downlink duration is 5.3 ms.
4.6 SIMULATION RESULTS

The experiment was conducted with the proposed algorithm with three different service flows. The vital QoS parameters of throughput, packet loss and average delay was calculated for three different kinds of service flow, with varied number of SSs. To analyze the QoS in Wimax networks, VoIP application is considered. For each of the scenarios, the simulation time is 40s. The following simulation results are obtained based on an average of 10 independent simulations presented in 95% confidence intervals. Comparative results for service flows in VoIP (G.711 CODEC) traffic using the cross layer algorithm has been presented in the following sections.

For the CODEC scheme G.711, the number of nodes with the VoIP traffic is varied as 1, 3, 5, 7, 9 and 11. The experiment is repeated only for the following service flows defined by IEEE 802.16e standards BE, rtPS and UGS.

To present a comparison between different service flows, the data collected from all the three service flows are presented in the same chart. This will clearly indicate the behavior of different service flows under same traffic for different QoS parameters. In Figure 4.6, the horizontal axis represents the number of subscriber stations while the vertical axis shows the throughput values.

4.6.1 Throughput

A graph is prepared using the number of subscriber stations and the throughput in Kbps. The data collected from all three service flows for throughput are presented in a single chart, as shown in Figure 4.6. Since the UGS traffic has less packet loss, the throughput is high. The throughputs of rtPS and BE are the same.
Since the UGS service flow is designed with a constant bit rate traffic in which the periodic BW is allocated by the BS to the SS, the throughput is high.

4.6.2 Packet Loss

Packet loss is the sum of all the packets which do not reach the destination, over the sum of packets which leave the destination. To calculate the packet loss, first the sum of the received packet rates is calculated. Then the sum of the packet size of the sent packets is calculated. The difference in value is the data that has been lost. The ratio of the total data sent to the total data lost, gives the packet loss. A graph is drawn using the number of subscriber stations and the packet loss in percentage.
The comparative packet loss percent variation is shown in Figure 4.7. Since the UGS traffic supports real time traffic, it is designed to generate fixed size data packets on a regular interval basis. The VoIP traffic does exactly that. It has a very low packet loss. This is one of the known behaviors. In the case of the rtPS, the SS allocates a fixed BW, and it transmits the data packets in a specific slot. The SS need not request BW explicitly. The BW is not allotted to the rtPS service flow on a regular basis. So the packet loss is comparatively low with the BE service flow.

4.6.3 Average Jitter

Jitter is one of the vital parameters to quantify the performance of the VoIP service. Figure 4.8 shows the average jitter for all the 3 service flows. A graph is plotted between the number of subscriber stations and the average jitter in logarithmic scale. BE has the highest jitter value, whereas the UGS has a lesser jitter value.
Figure 4.8  G.711 CODEC average jitter for all service flows

It is proved that the jitter does not vary when the number of nodes increases. Jitter is one of the main characteristics which determine the quality of voice received at the receiver side.

4.6.4 Average delay

The time taken by the packets to start from the source, and reach the destination and traverse back to the source is the delay caused by the packet. The source which causes the delay can be propagation delay, network delay, source delay, or destination delay. A graph is drawn using the number of subscriber stations and the average delay in seconds.
The three service flows’ average delay variation is comparatively shown in Figure 4.9. The delay for the UGS service flow and the rtPS service flow are close to each other. The BE service flow has the highest delay when compared to the other 2 service flows.

4.6.5 Comparative Results Analysis

The proposed algorithm not only meets all the QoS requirements of the service classes but also provides higher throughput, low delay and low packet loss rate, while promises fairness among all the other service class.

Table 4.3 Comparative performance analysis among the service classes

<table>
<thead>
<tr>
<th>Algorithm</th>
<th>UGS</th>
<th>rtPS</th>
<th>BE</th>
</tr>
</thead>
<tbody>
<tr>
<td>Throughput (Mbps)</td>
<td>292.1250</td>
<td>278.500000</td>
<td>276.66670000</td>
</tr>
<tr>
<td>Packet loss (%)</td>
<td>0.0472</td>
<td>0.047760</td>
<td>0.72290000</td>
</tr>
<tr>
<td>Average delay (ms)</td>
<td>0.0315</td>
<td>0.051333</td>
<td>0.032333333</td>
</tr>
</tbody>
</table>
From Table 4.3, it is proved that UGS service flow has higher throughput, lowest delay and lowest packet loss. This makes UGS service class the most suitable service flow for VoIP traffic.

4.7 SUMMARY

In this chapter we presented a cross layer approach to packet scheduling in fixed WiMAX networks. Simulation observations showed that the UGS service flow has the highest throughput, least packet loss and lowest average jitter. The results indicate that the UGS traffic is best suited for VoIP traffic. UGS service flow can handle fixed size packets which are generated at a regular interval. The simulation has been performed using VoIP CODEC scheme G.711 based application. The simulation showed a fair scheduling among the service flows because scheduling algorithm considers wireless link channel condition in the scheduling decision.