Channel quality based cross-layer scheduling algorithm in Wimax networks

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Abstract: The objective of the broadband wireless technologies is to ensure the end to end Quality of Service (QoS) for the service classes. Wimax is a revolution in wireless networks, which could support real time multimedia services. In order to provide QoS support and efficient usage of system resources, an intelligent scheduling algorithm is needed. The design of the detailed scheduling algorithm is a major focus for researchers and service providers. In this paper, a channel aware cross-layer scheduling algorithm for Wimax networks has been proposed. This scheme employs the Signal to Noise Ratio (SNR) value, which allocates the bandwidth based on the information about the quality of the channel, and the service requirements of each connection. The proposed algorithm is described in detail, and evaluated with one VOIP codec and real time video traffic, through a series of simulations. The QoS parameters of throughput, packet loss, average delay and average jitter have been measured in simulation.

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1. Introduction

IEEE 802.16 is a series of Wireless Broadband standards authored by the Institute of Electrical and Electronics Engineers (IEEE).The 802.16 family of standards is officially called the Wireless MAN in IEEE: it has been commercialized under the name Wimax" (from "Worldwide Interoperability for Microwave Access").

The main advantage of Wimax over other access network technologies is the long range of coverage area up to 30 miles and more support for QoS at the MAC layer. The standard defines two operational modes for communication. One is the point to multipoint mode (PMP) and the other one is the mesh mode. In the point to multi point mode, subscriber stations (SSs) can communicate with one other and to the Base station (BS). In the case of the PMP mode, the SSs are allowed to communicate only through the BS. Multiple connections are there between the BS and the SS. At the BS, downlink connections have special dedicated buffers, and slots are allotted for grant per connection. In the case of an uplink, slots are allotted per SS and not per connection. The SS has to decide how the UL slots are used. The PMP mode configuration is shown in Figure 1.The BS and SS connect through high speed wireless links. The BS acts as a gateway to the internet.

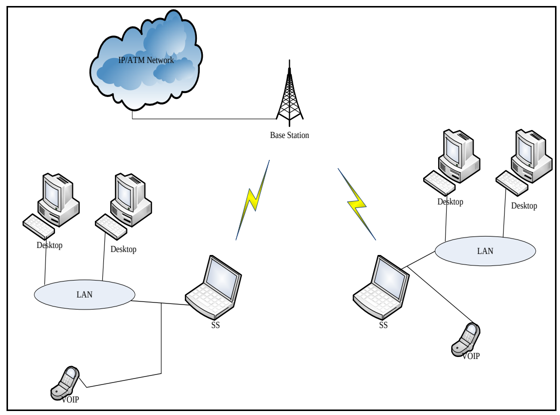

Figure1. Wimax PMP mode communication

The IEEE 802.16 defines layer 1 (physical (PHY)) and layer 2 (Data link or media access control (MAC)) of the open system interconnection (OSI) seven layer network model [13]. The different types of standards for PHY supports are the Single carrier (SC), Single carrier Access (SCA), Orthogonal Frequency Division Multiplexing (OFDM) and Orthogonal Frequency Division Multiple Access (OFDMA). Recent research focuses mainly on the OFDM and OFDMA PHY supports. The Wimax standard defines five different standard QoS classes. Three QoS classes can be used for real time connections. The remaining QoS classes are defined for non real time traffic. The five QoS classes are as follows, according to their priority,

1. UGS (Unsolicited Grant Services): designed to support the Constant bit rate services like voice applications.

2. RTPS (Real Time Data Polling Services): designed to support real time services that generates variable size data packets on a periodic basis like the MPEG but insensitive to delay.
3. ERTPS (Extended Real Time Polling Services): designed to support real time applications with variable data rates which require guaranteed data and delay. Example: Voice Over Internet Protocol (VOIP) with silence suppression.
4. NRTPS (Non Real Time Polling Services): designed to support non real time and delay tolerant services, that require variable size data grant burst types on a regular basis such as File Transfer Protocol (FTP).
5. BE (Best Effort) designed to support data streams that do not require any guarantee in QoS, such as Hyper Text Transfer Protocol (HTTP).

UGS: The UGS traffic is designed to support real time applications, which generate fixed size packets at periodic intervals. The UGS connection uses the unsolicited grant bandwidth request mechanism. It never requests bandwidth. It requires a periodic bandwidth without any polling or contention. The UGS grant size is calculated by the BS based on the minimum reserved traffic rate.

rtPS: The key QoS parameter of the rtPS connections are the minimum reserved traffic rate, and maximum latency. The packet size is not fixed for rtPS, it requires to notify the BS for its current bandwidth requirements. The rtPS needs periodic unicast polling from the BS.

nrtPS and BE: Unlike the UGS and rtPS, the nrtPS and BE connections request bandwidth by either responding to broadcast polls from the BS or piggybacking request from an outgoing PDU.

ertPS: The goal of the ertPS is to combine the features of the UGS and rtPS. In this connection mode, the BS continues to grant the required amount of bandwidth, corresponding to the maximum sustained traffic rate of the connection until it explicitly requires a change in the polling size.

The QoS depends upon a number of implementation details like scheduling, buffer management and traffic shaping. The responsibility of scheduling and BW management is to allocate the resources efficiently based on the QoS requirement of the service classes. The QoS provision in Wimax requires a complete scheduling mechanism, which is not defined in the standard. The scheduling mechanisms have to provide a guarantee to the bandwidth required by SS as well as a wireless link usage. The goal of designing a scheduler is to minimize power consumption and a bit error rate (BER) and to maximize the total throughput. Wired networks scheduling algorithms are unfit for wireless networks, due to location dependency and burst channel errors. Thus, the scheduling algorithm should take Wimax QoS classes and service requirements into consideration.

In this work, we apply a cross layer design approach to design a cross layer scheduling algorithm which considers channel quality as a feedback parameter for downlink scheduling. While ensuring all the QoS requirements the CL scheduler will try to provide throughput enhancement, by reducing the packet loss rate and average delay.

2. Related work

Borin and Fonseca proposed a standard compliant scheduling solution for uplink traffic in IEEE 802.16 networks [5], but wireless channel characteristics are not considered in this solution. Many other scheduling mechanisms have been proposed in [6] and [7]. But none of them is able to support the QoS requirements of the five types of service flow defined by the IEEE 802.16e standard. To provide guaranteed latency requirements to real time applications, the scheduling mechanism proposed in [8] uses a history of packet delays to classify packets in four classes and the scheduler gives a higher priority to packets destined to users whose instant channel conditions are better. Iera et al [9] proposed a scheme in which packets can be blocked when the user channel conditions are not satisfactory. In [2], the authors proposed a two stage cross-layer QoS support framework with a scheduling algorithm. That scheduler provided the latency guarantee, and a mechanism to avoid starvation, but failed to provide the maximum rate guarantee.

It has been proved that the scheduling algorithm that considered the wireless link performs better than the algorithm that does not consider thewireless link [15]. The scheduling algorithm proposed in [13] was capable of scheduling all the service flow types considering the nature of the wireless link, delay and buffer size. Schedulers can use different metrics to estimate the channel condition. In [9], the channel quality is measured as SNR while in [8], [11] and [12], it is estimated according to the instantaneous transmission rate. Fluid Fair Queuing (FFQ) is a well-known algorithm which provides fairness among the packets through the shared link [4]. In [4], the author classified the uplink schedulers as the Weighted Round Robin (WRR), Earliest Dead line First (EDF) and Weighted Fair Queuing (WFQ). Down link schedulers are classified into Proportional Fairness (PF), Adaptive Proportional Fairness (APF), Integrated Cross-Layer (ICL) and Round Robin (RR).

In [15], the authors emphasise the MAC scheduling architecture for IEEE 802.16 wireless networks in both the uplink and downlink directions to broadcast the frame. Further they used the WFQ as the uplink as well as the downlink scheduling algorithm, for improving delay and throughput. There is no separate scheduling policy for Unsolicited Grant Services (UGS). EDF is appropriate for real time data Polling Services (rtPS) [16] and WFQ for non-real time Polling Services (nrtPS). The remaining bandwidth is split for all BE connections. This work has not considered extended real time Polling Services (ertPS). In [17], the authors proposed scheduling algorithms for voice activity. Only Uplink scheduling is taken into consideration. The results are not based on real frame values, and ertPS has not been taken into account. Cross layer communications would be needed to inform the MAC layer about the transitions. Yang et al [18] had considered only the real time video traffic. They have not considered the OFDMA scheduler and the traffic classes. Even though there are numerous of works based on scheduling in single hop networks, these algorithms cannot be applied for multihop relay scenarios. The TCP aware uplink scheduling algorithm focuses on the allocation of bandwidth higher than the actual sending rate of the connection.

3. IEEE 802.16 Scheduling Architecture

The basic IEEE 802.16 communication architecture [21] includes the Base station and multiple subscriber stations (SS). Both the base station and subscriber stations are immobile when a client wants to connect the SS to a mobile station. The Base station acts as a central entity, which transfers all the data from the subscriber stations in the point-to-multi point architecture. Two or more subscribers are not allowed to communicate directly. The BS and SS architecture are connected through wireless links. Communication occurs in two directions: that from the BS to the SS is called the downlink and that from the SS to the BS is called the uplink. During the downlink, the BS broadcasts data to all the subscribers and the subscriber selects the packets destined for it. The uplink channel is shared between all multiple SSs, while the downlink channel is used only by the BS. Figure 2 depicts the basic architecture of the IEEE 802.16.

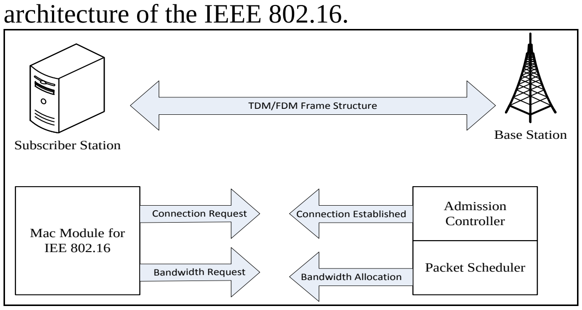

Figure2. IEEE 802.16 network architecture

If the SS wants to open a connection to the BS, it first sends a request. Upon receiving the message the BS performs admission control based on the requested traffic, QoS specification, and available resources. Once the connection is established, the SS may obtain the particular bandwidth by sending a class specific request. The BS then aggregates all the requests and allocates the bandwidth to each connection or SS, through an appropriate scheduling scheme.

In order to ensure slotted channel sharing, i.e that the slots are allocated by the BS to various SS in one uplink frame, Time Division multiplexing (TDD) or Frequency Division multiplexing (FDD) is used. This slot allocation information is broadcast by the BS through the uplink map message (UL-MAP) at the beginning of each frame. The UL-MAP contains an information element which includes the transmission opportunities and the time slots in which the SS can transmit during the uplink subframe.

4. Scheduling Algorithms

The IEEE 802.16 MAC layer adopts a connection oriented architecture, in which a connection must be established before data communication. Each connection is assigned a unique identifier (connection IDI) and it is associated with a service flow which defines the desired QoS level of the connection. In a standard scheduling framework, the data packets arriving at the BS are classified into connections, which are then classified into service flows. Packets of the same service flow are placed in a queue, and then further classified based on their service priorities of the connection. For packets in multiple queues with different service requirements, a packet scheduler is employed to decide the service order of the packets from the queues. If properly designed a scheduling algorithm may provide the desired service guarantees.

The scheduler should consider the following important parameters;

1. The traffic service type
2. The set of QoS requirements of the connections
3. The capacity of the bandwidth for data transmission
4. The bandwidth requirements from the connections
5. Waiting time of the bandwidth request in the system

The ideal scheduler should be able to make optimum use of the available bandwidth to reduce traffic delays and satisfy the QoS requirements to the best extent, so as to reduce the packets drop rate and sustain the QoS support.

Wimax schedulers can be classified into two main categories, channel unaware schedulers where the channels are assumed to be error free, and channel aware schedulers where the channel state information is taken into consideration while scheduling the packet. Channel unaware schedulers are further classified into homogeneous and hybrid schedulers. Hybrid schedulers combine more than one scheduler to satisfy the QoS requirements of the multiple service class traffic in Wimax networks.

WRR, WFQ, EDF and Strict priority (SP) are the few examples of homogeneous scheduling algorithms. According to the research, none of the homogeneous scheduling algorithm provides the QoS requirement of the Wimax networks. So, researchers have attempted to hybridise the algorithms to get a satisfactory QoS level. Cross-layer scheduling is one of the algorithms in the channel aware scheduling algorithm.

5. Proposed Cross-Layer Scheduling Algorithm

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 the different service classes. The proposed scheduling algorithm modifies the cross-layer algorithm, which incorporates the SNR value and the minimum required throughput of the SS in its formulation. The SS with the highest priority is selected to transmit in the frame. The priority of the SS is calculated, based on the traffic class it belongs to.

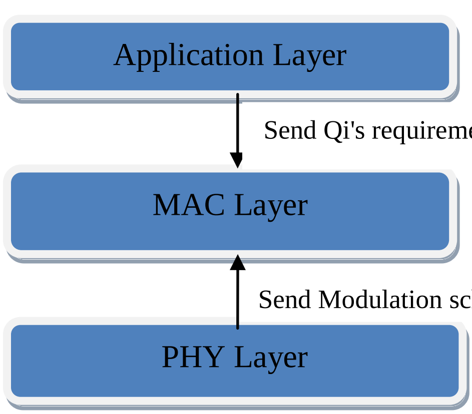

Figure3. Cross Layer Diagram

Algorithm:

1. Define higher priority queue
2. Schedule the Bandwidth request opportunities, which should be scheduled in the next frame
3. Periodically check the deadline for the service flow
4. Do check the minimum bandwidth 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 in [3] implies three drawbacks. The modified cross-layer scheduling algorithm overcomes those drawbacks in the following ways and efficiently manages the bandwidth 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 service types to make use of the available slots, except the ertPS connection in the Wimax frame.

Based on the SNR, the type of modulation can be chosen from Table 1.

Table 1. MCS and receiver SNR

S/N	Modulation	Coding rate	SNR(dB)
1	QPSK	½	5.0
		¾	8.0
2	16-QAM	½	10.5
		¾	14.0
3	64-QAM	½	16.0
		2/3	18.0
		3/4	20.0

Four different buffers were used, one for each for one service flow. Each buffer has length t and each packet received in the uplink session is stored in the buffer with the 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.

6. Proposed Cross-Layer Scheduling Algorithm

6.1 Simulation Platform

The scheduler proposed in this paper was implemented in the IEEE 802.16 module in the network simulator (NS-2) simulator. 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 Object Tool

Command Language (OTCL) interpreter for easy user interface, has input models, written in Tool Command Language (TCL) scripts. A base station and a subscriber station can be set up as a node in ns2.When the number of nodes increases, the amount of packets received and sent also increases. For a single node configuration the simulation would run fairly. But as the number of nodes increases, the packet traffic will increase.

6.2 Simulation Parameters

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

Table 2. Main parameters used in simulation

Parameters	Values
Frequency band	6 Mhz
Duplexing	OFDM
Propagation model	Two way
Antenna	Omnidirectional
Frame duration	20ms
Downlink bandwidth	7 Mbps
Uplink bandwidth	10 Mbps
Simulation time	60s
Coverage radius	1600-1800m

The simulation parameters settings are shown in Table 2. The Base station receives all transmitted the packets from the subscriber stations; assigns the packet serial number, packet service flow identification and arrival time, and stores the packet in the appropriate buffer of the service flow. Each transmitted packet has its own estimated SNR value as shown in Table 2. 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 numbers 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 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, it is considered to be lost since there is no memory to hold it. Once the packet is scheduled, it should be removed from the buffers and the memory is considered empty to store the next packet. The uplink duration is 4.5ms and the downlink duration is 5.3ms.

6.3 Simulation Results

The quality of service of Wimax has been analyzed by considering various real time and non real time cases. Since Wimax provides voice over ip connectivity it will support the voice calling over internet protocol. Thus the VOIP is the first application considered. The availability of a wide range of support of data pipes another common application these days is viewing videos over internet. So video streaming is analyzed as well.

The experiment was conducted with VOIP and video traffic with the proposed algorithm, with three different service flows. The vital QoS parameters of throughput, packet loss, and average delay were calculated for three different kinds of service flows, with varied number of SSs. To analyze the QoS in Wimax networks, the first VOIP application is considered. For each of the scenarios, the simulation time is 40s. The following simulation results are obtained based on the average of 10 independent simulations, presented in 95% confidence intervals. The VOIP traffic is a constant bit rate flow .The size of the packet and packet rate are defined by the VOIP codec scheme which is used .The codes are defined by the ITU. 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.

In the simulation run we considered 2 scenarios.

1. VOIP codec is set to G.711 with the CL algorithm and measure critical QoS parameters
2. Real time Video traffic considered with the CL algorithm and measure critical QoS parameters

6.3.1 Scenario 1-Comparative results for all service flows for VOIP traffic in the CL algorithm

Throughput:

The data collected from all three service flows for throughput are presented in a single chart. Since the UGS traffic has less packet loss, the throughput is high. The throughput of rtPS and BE are very similar. The UGS service flow is designed with a constant bit rate traffic, in which the periodic bandwidth is allocated by the BS to the SS. As we can see from Figure 4, the graph shows the better throughput of all the three service flows, for the cross-layer scheduling algorithm.

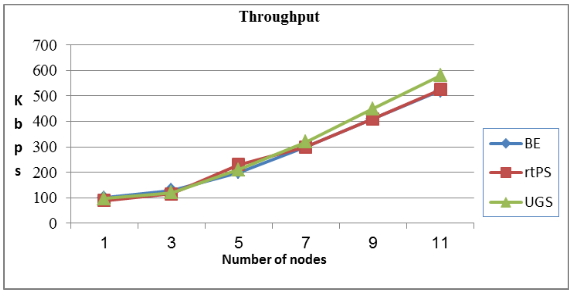

Figure 4. Throughput with the number of nodes all service flow.

Packet loss:

Packet loss is the sum of all the packets which do not reach the destination, over the sum of packets which leaves 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 value difference is the data that was lost. The ratio of the total data sent to the total data lost, gives the packet loss.

The comparative packet loss percent variation is shown in Figure 5.Since the UGS traffic supports real time traffic; it has a very low packet loss. This is one of the expected behaviors. In the case of rtPS, the SS was allocated with a fixed bandwidth and it transmits the data packets in a specific slot. The bandwidth is not allotted to the rtPS service flow on a regular basis. So the packet loss is comparatively low with the BE service flow.

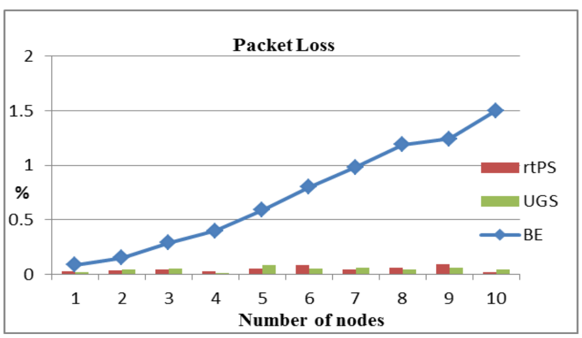

Figure 5. Packet loss with the number of nodes for all three service flow

Average jitter:

Jitter is one of the vital parameters to quantify the performance of the VOIP service. Figure 6 shows the average jitter for all the 3 service flows. BE has the highest jitter value, whereas the UGS has a lesser jitter value. It is proved that the jitter does not vary when the number of nodes increases.

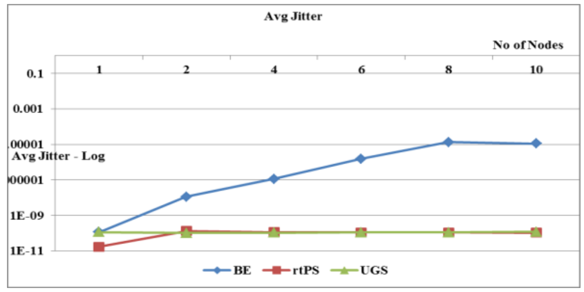

Figure 6. Average jitter with the number of nodes for all three service flow

Average delay:

The time taken by the packets to start from the source and reach the destination and traverse back to source is the delay produced by the packet. The source which causes the delay can be propagation delay, network delay, source delay, or destination delay.

Three service flows average delay variation is comparatively shown in Figure 7.The delay for the UGS service flow and the rtPS service flow are close to each other, which is shown in the figure as an overlapped line. The BE service flow has the highest delay when compared to the other 2 traffics.

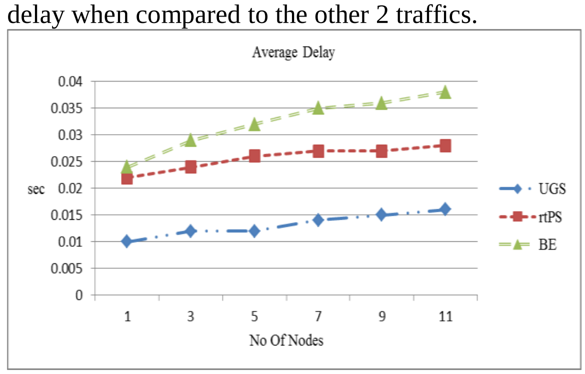

Figure 7. Average delay with the number of nodes for all three service flow

6.3.2 Scenario 2-Comparative results for all the service flows for real time video traffic in the CL algorithm

Video streaming is a variable bit rate traffic. For the simulation analysis the H.263 stream used. Along with video streaming, theH.263 format which is used is video-conferencing and video-telephony applications. The performance analysis is done, using the BE, rtPS and UGS service flows for the video traffic. The parameters of throughput, packet loss, average jitter and average delay are analyzed. The QoS parameters are observed for each service flow, as the number of nodes with the video traffic increases.

Throughput:

Figure 8 shows the combined throughput variation for video traffic over all the three service flows. For the BE service flow,as the number of nodes increases, the throughput increases gradually.

The sample video streams at 64kbps.In the case of real time polling service flow, when the number of nodes with streaming video traffic increases, the throughput increases. The value of throughput for the 64kbps video sample with one node is close to 63.5 kbps. For 10 nodes it is about 600 kbps. This value is lower than expected. The reason for the lower value of throughput is the higher packet loss incurred at the higher nodes. The overall throughput is marginally higher in the case of the UGS service flow. For 10 nodes, the rtPS throughput is the lowest.

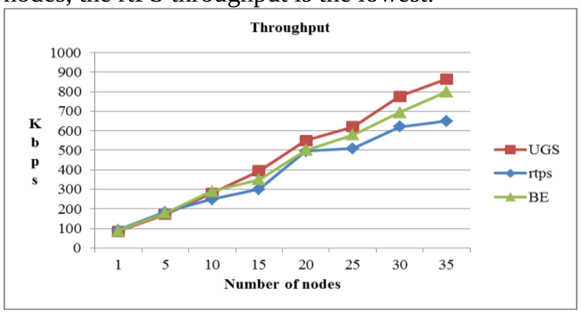

Figure 8.Combined throughput variation of all three service flows

Packet loss:

The following figure shows the overall packet loss for all the three service flows. The rtPS service flow has a high packet loss rate as the number of nodes increases. The bandwidth request mechanism for the rtPS service flow is an overhead procedure, every time the SS has to request for the bandwidth from the base station. As shown in Figure 9,there is a very high level of packet loss at the higher number of nodes. Thus, it has a request overhead as compared to the UGS service flow. But the rtPS is more efficient for a service that generates variable-size data packets.

A very low rate of packet loss is observed with video traffic as the UGS service flow increases. This is because in the UGS service flow, the BS allocates the bandwidth to the MS to send fixed sized packets at a fixed interval. The bandwidth is already allocated and this reduces the packet loss effectively. The packet loss percentage goes up, as well, in streaming video over BE service flow increases. As the number of packets being transmitted goes up, it increases the number of nodes generating the traffic.

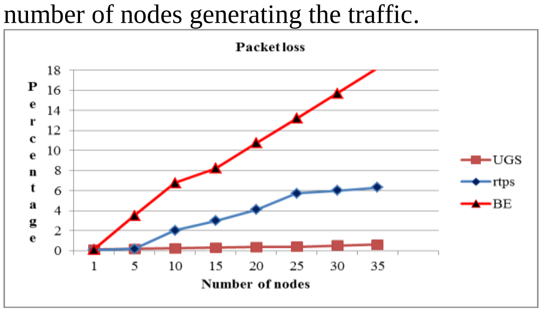

Figure 9.Combined packet loss variation of all three service flows

Average jitter:

The value of jitter is very low for a single node and it increases as the number of nodes with streaming video traffic increases. Figure 10 shows the variation in the average jitter for all the three service flows. The average jitter increases as the number of nodes increases. For the UGS service flow, the average jitter value varies within a very small range. It does not show any relation to the number of nodes.

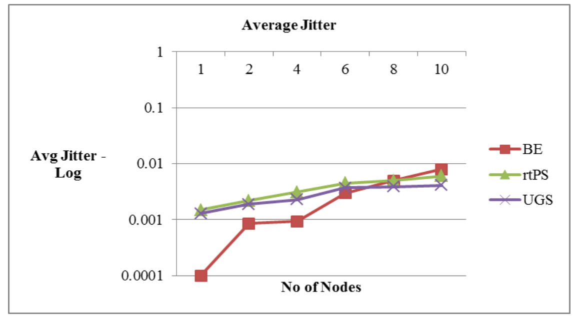

Figure 10.Combined average jitter variation of all three service flows

Average delay:

Figure 11 displays the variation in the average delay as the number of nodes with all the three service flows, traffic increases. Again, similar to the average jitter, the variation in delay as the number of nodes increases is very small. As seen from the figure, the delay steadily increases as the number of nodes increases.

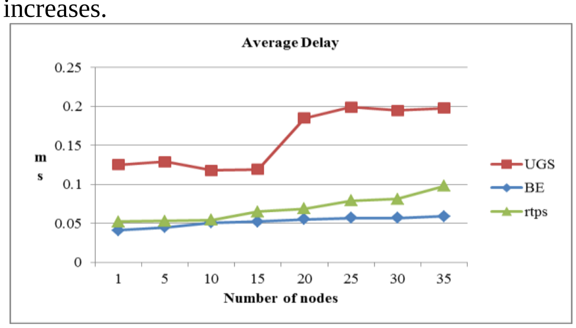

Figure 11.Combined average delay variation of all three service flows

Table 3. Comparative analysis for service classes with 2 different traffics

		Service Class		
App. Type	Parameters	rtPS	BE	UGS
Video	Throughput	388	435	468
	Delay	0.068	0.052	0.15
	Packet loss	3.425	9.55	0.353
VOIP	Throughput	279	277	296
	Delay	0.025	0.032	0.013
	Packet loss	0.0477	0.7229	0.0472

Table 3 shows the analysis of service class in 2 kinds of traffics. The UGS service flows perform better in terms of packet loss and delay. The network resources are not utilized effectively when the UGS service flow is used for streaming video traffic. The rtPS service flow performs better than the BE service flow in terms of delay and packet loss. The average delay value for the rtPS is low for video traffic. Thus the rtPS service flow appears to be most optimized for streaming video traffic than UGS service flow. But in case of VOIP the packet loss and delay for the rtPS and the BE is more than UGS service flow. Thus the UGS is suitable for VOIP traffic.

8. Conclusion

In this paper we addressed the problem of a crucial scheduling strategy which takes the channel condition as a feedback for better bandwidth usage for IEEE 802.16 wireless networks. In this work, the static IEEE 802.16 network is considered for study.To validate the proposed algorithm a Wimax simulation platform based on NS-2 has been implemented. The simulation results have verified that our proposed scheduling algorithm is capable of enhancing the performance of Wimax networks. The performance improvement of the proposed scheme is illustrated through the simulation results. The proposed algorithm not only meets all the QoS requirements of the service classes but also provides higher throughput, low delay and packet loss rate, while promising fairness to all the other service classes. Currently, we have worked on the G.711 VOIP codec scheme, real time video traffic and three service classes along with the proposed scheduling scheme. In future work, subscriber mobility will be considered and more codec schemes for VOIP will be taken up for more real-time operating environment.

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8/4/2013