Fuzzy based Adaptive Buffer Allocation based on Traffic Classes in WiMAX Networks

D.Karunkuzhali,
Research Scholar, Department of CSE,
Sathyabama University,
Chennai, India

D.C.Tomar
Professor, Department of CSE,
Shree Motilal Kanhaiyalal Fomra Institute of Technology, Chennai, India

ABSTRACT
Buffer allocation in WiMAX maximizes the throughput of system and minimizes the power consumption. Hence, an adaptive buffer allocation strategy is required to increase the goodput of the system. In this paper, we propose an adaptive buffer allocation technique based on traffic classes in WiMAX networks. Initially buffer is allocated to the flow requests based on buffer allocation factor. This factor is estimated using fuzzy logic. The parameters namely number of user requests, flow rate, queue length and received signal strength are taken as inputs. The originally allocated buffer is verified periodically by buffer reallocation technique. It computes two different satisfaction factors for real time and non real time flows. Delay is considered as a metric for real time flows and minimum reserved data rate is for non real time flows. Based on estimated satisfaction factor, flow rate is adjusted for real time traffic using PID controller and additional buffer is allocated for non real time flows. Through simulation results, we show the performance of our proposed technique.

Keywords
Buffer Allocation Factor (BAF), Base Station (BS), Quality of Service (QoS), Subscriber Station (SS), Base Station (BS), Proportional Integral Derivative (PID) Controller.

1. INTRODUCTION
1.1 Worldwide Interoperability for Microwave Access (WiMAX)
Worldwide Interoperability for Microwave Access (WiMAX) defined by IEEE 802.16 standards is designed for long distance broadband multimedia communication. [1]. IEEE 802.16 WiMAX system aims at providing high-speed internet access and multimedia services through wireless medium provides low cost all IP solutions for scalable networks with voice, data and video services. [2]. WiMAX networks incorporate several Quality of Service (QoS) mechanisms at the media access control (MAC) layer for guaranteed services for data, voice, and video.[3]. It relies on OFDMA as an access technique. OFDMA can greatly increase network capacity and maintain connectivity by adjusting modulation and coding rate [4,5].
The architecture of WiMAX network consists of different parts such as**,** BS: Base Station, SS: Subscriber Station, MSS: Mobile Subscriber Station, RAN: Radio Access Network, PSTN: Public Switched Telephone Network [6]. The algorithms are classified according to their channel awareness/unawareness connection admission control (CAC) plays an important role in assuring the QOS requirements and it needs to be designed along with the scheduler. Before joining the network, the subscribers need to have a permission from the BS to transmit data with a QoS agreement. The CAC basically maintains the current system load and QoS parameters for each existing connection. Then, it can make a decision if a new connection should be admitted and if admitted, what QoS the BS can provide. It should be obvious that if the CAC cannot support at least the minimum reserved rate for a new flow, that connection should be rejected. Otherwise, the QoS requirements of the existing flows can be broken. For example, instead of admitting another UGS flow, a BE flow is accepted if there is no way to guarantee the maximum allowable delay.[3]

1.2. Service classes of WiMAX and their Priority
The services classes of WIMAX can be classified as given below
- Real-time Polling Service (rtPS)
- Non-real-time Polling Service (nrtPS) and
- Best Effort (BE) [7]

IEEE 802.16 defines five QOS service classes: Unsolicited Grant Scheme (UGS), Extended Real Time Polling Service (ertPS), Real Time Polling Service (rtPS), Non Real Time Polling Service (nrtPS) and Best Effort Service (BE). Each of these has its own QOS parameters such as the way to request bandwidth, minimum throughput requirement and delay/jitter constraints.
UGS: This service class provides a fixed periodic bandwidth allocation. Once the connection is setup, there is no need to send any other requests. This service is designed for constant bit rate (CBR) real-time traffic such as E1/T1 circuit emulation. The main QOS parameters are maximum sustained rate (MST), maximum latency and tolerated jitter (the maximum delay variation).
ertPS: This service is designed to support VoIP with silence suppression. No traffic is sent during silent periods. ertPS service is similar to UGS in that the BS allocates the maximum sustained rate in active mode, but no bandwidth is allocated during the silent period. There is a need to have the BS poll the MS during the silent period to determine if the silent period has ended. The QOS parameters are the same as those in UGS.
rtPS: This service class is for variable bit rate (VBR) real-time traffic such as MPEG compressed video. Unlike UGS, rtPS bandwidth requirements vary and so the BS needs to regularly poll each MS to determine what allocations need to be made. The QOS parameters are similar to the UGS but minimum reserved traffic rate and maximum sustained traffic rate need to be specified separately. For UGS and ertPS services, these two parameters are the same, if present.
nrtPS: This service class is for non-real-time VBR traffic with no delay guarantee. Only minimum rate is guaranteed. File Transfer Protocol (FTP) traffic is an example of applications using this service class.

BE: Most of data traffic falls into this category. This service class guarantees neither delay nor throughput. The bandwidth will be granted to the MS if and only if there is a left-over bandwidth from other classes. In practice most implementations allow specifying minimum reserved traffic rate and maximum sustained traffic rate even for this class [3]

1.3 Resource Allocation and Issues

The resource allocation among the users is not only to achieve QOS but also to maximize goodput (throughput after overheads such as preamble, management messages, level headers, and so on) and to minimize power consumption while keeping feasible algorithm complexity and ensuring system scalability. IEEE 802.16 standard does not specify any resource allocation mechanisms or admission control mechanisms. [3]

The resource allocation of OFDMA in Time Division Duplex (TDD) mode in which the new transmission frame with multiple time slots is popped up on every pre-specified period. Frame resource is divided into chunks that are composed of a group of subcarriers with equal and constant time duration. The centralized resource allocation scheme aims to provide the guaranteed service to users by converting the required service into the network cost. The users whose network costs are too high are not guaranteed in order not to waste the precious bandwidth [4].

In our previous paper[13], we have proposed a fuzzy based dynamic buffer management technique in WiMAX 16m network which performs buffer allocation and packet dropping. This technique operates in the base station (BS). As per application requirements, BS estimates the parameters such as number of user requests, flow rate, queue length and received signal strength and updates periodically. When a buffer request packet arrives, buffer allocation factor (BAF) is estimated using the fuzzy logic applied over the parameters estimated in the BS. The user requests are sorted in the descending order of BAF. This reveals that the flow request with more BAF is admitted and rest of the flow requests await in queue. When a new request arrives, its BAF is tested. If the value is low, the request packet is dropped. Otherwise, the pending request packet in the queue is emptied on analyzing their channel condition and buffer is allocated for new request.
The drawbacks identified are: (1) The proposed technique does not distinguish between real-time and non-real time service requests. (2) As the QoS requirements of these two types of service classes are different, the buffer allocation factor should be checked and re-allocated according to the obtained QoS levels.

3. ADAPTIVE BUFFER ALLOCATION TECHNIQUE BASED ON TRAFFIC CLASSES IN WIMAX

3.1 Overview

In this paper, we propose an adaptive buffer allocation technique based on traffic classes in WiMAX networks. Initially, when a flow request reaches the base station (BS), it estimates the buffer allocation factor using fuzzy logic. The parameters such as number of user requests, flow rate, queue length and received signal strength are taken as inputs. Based on buffer allocation factor, the BS allocates buffer to the flow requests. The originally allocated buffer is periodically verified by the BS using buffer reallocation technique. In this, the BS estimates two different metrics for real time and non real time flows. Delay is considered for real time flows and minimum reserved data rate is taken for non real time flows. Two different satisfaction factors are measured for real and non real time flows. By comparing with threshold values, data rate is adjusted for real time flows and additional buffer is allocated for non real time flows.

3.2 Estimation of Metrics

3.2.1 Estimation of Queue Length and Flow Rate

Let Q (t) be the queue length of aggregated traffic flow of service type j, (j [1, 2] for direct and relay cooperation transmission modes respectively) at base station i (i {1, 2,….., M}) at time t.
The vector of the queue status of all base stations is given as

$$Q = \{Q_{11} (t), Q_{12} (t) \ldots, Q_{M2} (t)\}]^T$$

By considering the liquid fluid model, the queue length is evaluated as follows.

$$Q_{ij} (t) = N_{ij} (t)\, IR(t) - (1-R_{ij})\, BW_{ij}(t)\, \eta_{ij} \quad (1)$$

$$\forall \ i \ \{1,2,\ldots M\}, j \in \{1, 2\}$$

where
$N_{ij}(t)$ = Number of base stations
$IR(t)$ = input traffic flow to the subscribed base station.
$N_{ij}(t)\, IR(t)$ = aggregate downlink flow rate at base station i
R_{ij} = average packet error rate (PER) for abstracting the channel quality.
η_{ij} = average spectral efficiency (in bits/s/Hz)
$BW_{ij}(t)$ = bandwidth assigned for draining the queue
$BW_{ij}(t)\, \eta_{ij}$ = queue depletion rate
The initial state of the queue $Q_{ij}(0)$ represents the initial size of the backlogged data of the queue. There is a possibility that IR(t) may get fluctuated over time depending on the source behavior and can be viewed as the disturbance to the system. In general it is denoted as

$$IR(t) = IR_n + \omega(t) \quad (2)$$

where, IR_n is a normal value of the input rate and $\omega(t)$ is a disturbance which can be either stochastic (e.g. white noise Gaussian process) or deterministic (e.g. impulse traffic load).
This disturbance can occur due to the randomness of the packet arrival from the applications. [9]

3.2.2 Estimation of Channel Condition

The physical layer constraints such as channel fading, multi-path propagation, reflection, scattering and other climatic effects on the channel reveals the channel condition. This channel condition can be estimated based on the received signal strength (RSS) and signal to noise ratio (SNR) at the receiver.
The received signal strength (RSS) is estimated using Friis equation which is shown in Eq: (3)

$$RSS = \frac{P_{tx} * \alpha * \beta * H_{tx} * H_{rx} * \gamma^2}{(4 * \gamma * d)^2 * \delta} \quad (3)$$

Where,
P_{tx} = transmission power, α = transmitter gain, β =receiver gain, H_{tx} =height of the transmitter, H_{rx} =height of the receiver, γ =wavelength, d=distance between the transmitter and receiver and δ =system loss
From the above computed RSS, the signal to noise ratio (SNR) is computed using Eq: (4)

$$SNR = \log_{10} (P_{tx}) - \log_{10} (P_{rx}) \text{ dB} \quad (4)\ [10]$$

3.3 Fuzzy based buffer allocation

Upon receiving BRP, the buffers are allocated using buffer allocation factor (BAF). It is estimated with the help of fuzzy controller. The steps involved in the fuzzy logic technique are detailed below.

1) Fuzzification
2) Inference with rule base
3) Defuzzification

Fuzzification:
Our technique considers four input parameters for Fuzzification such as number of user requests (UR), flow rate (R), queue length (Q) and received signal strength (RSS). Based on the input parameters and inference engine, the output obtained is the buffer allocation factor (BAF). Each of the fuzzy parameters is represented using triangular membership function as it represents minimum and maximum and maximum boundary conditions. The membership to each of fuzzy variables is assigned using intuition method. This technique minimizes the computation complexity.
The membership function for these input parameters and output is represented as f (UR), f(R), f (Q), f (RSS) and f (BAF).

Number of user requests: Based on the count of user request, linguistic values associated with the membership function f (UR) are low, and high. The low UR is preferred for buffer allocation.

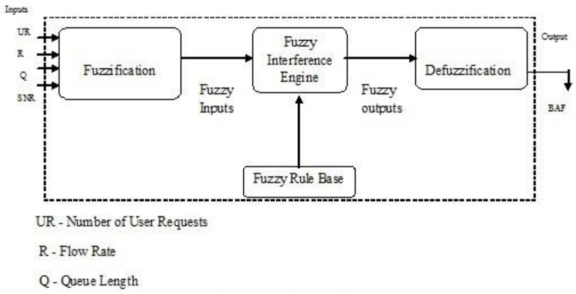

Fig 1 : Fuzzy Controller for Buffer Allocation

Flow rate: The flow rate varies based on the user requirements as per the applications. They considered since they provide the required buffers. The variation level in the rate of flow is represented by using linguistic values related to the membership function f(R) such as low and high. The high R is preferred for buffer allocation

Queue length: The queue length is measured based on the number of tasks in each queue i.e. it gives the measure for buffer availability. The linguistic values associated with the membership function f (Q) are low and high. The higher Q is preferred for buffer allocation

Received Signal Strength: The received signal strength describes the communication quality among the two nodes. The linguistic values associated with the membership function f (RSS) are low and high. The higher RSS is preferred for buffer allocation

Buffer Allocation Factor: Output of four input linguistic value is buffer allocation factor. The allocation factor is represented by linguistic values associated with membership values such as low, medium and high.
The fuzzy buffer allocation scheme forms a fuzzy set of dimension f (UR)* f(R)*f (Q)* f (RSS).

To alleviate the problems described in our previous paper, in this paper, we propose to deploy an adaptive buffer allocation technique based on traffic classes in WiMAX networks.

3.4 Traffic Based Buffer Reallocation Technique

Once the BS allocates the buffer to the requests based on BAF, it periodically measures satisfaction factor (SF) of the flows. The value of SF is calculated by the BS, to know whether the allocated buffers are sufficient to process the requests. If so the BS continues the same allocation, else, it reallocates the buffer to the request or it minimizes the traffic flow.
For real time services, the SF is calculated in terms of delay and it is measured using minimum reserved data rate for non real time services. The SF is computed using two different metrics to satisfy the QoS of real time and non real time flows.
In order to estimate the SF for real time flows, we calculate delay by considering queue size that is not allocated at current frame. Then the average queue size at time t of flow F is given as,

$$QS_N = \begin{cases} 0, & \text{if } C=M; \\ \dfrac{\sum_{i=1}^{C-M} F_i}{C-M} & \text{if } C > M \\ & \text{if } C>M; \end{cases} \qquad (6)$$

Where, C denotes number of flows and M represents maximum number of flows that are connected at time t. From the estimation of QS_N, we can predict the delay using the following equation,

$$D_t = \frac{QS_N}{m} * T_{OFDM} * f \qquad (7)$$

Here, D_t is the predicted delay at time t, T_{OFDM} is the OFDM symbol time and f denotes frame duration. The predicted delay D_t will be directly proportional to QSN, if m, T_{OFDM} and f are constants. The estimated RSF(t) is compared with two predefined thresholds namely minRSF and maxRSF. If the satisfaction factor is less than minRSF, then the source adjusts the flow rate using PID controller.

3.4.1 Rate Adjustment using Proportional Integral Derivative (PID) Controller

A proportional–integral–derivative controller (PID controller) is a generic control loop feedback mechanism (controller) widely used in industrial control systems – a PID is the most commonly used feedback controller. A PID controller calculates an "error" value as the difference between a measured process variable and a desired set point. The controller attempts to minimize the error by adjusting the process control inputs.
The PID controller calculation (algorithm) involves three separate constant parameters, and is accordingly sometimes called three-term control: the proportional, the integral and derivative values, denoted P, I, and D. Heuristically, these values can be interpreted in terms of time: P depends on the present error, I on the accumulation of past errors, and D is a prediction of future errors, based on current rate of change. The weighted sum of these three actions is used to adjust the process via a control element such as the position of a control valve, or the power supplied to a heating element. [11]. We utilize PID controller to adjust the flow rate of real time service. Here, the flow rate is controlled considering buffer occupancy of the scheduler. Statistically, the buffer occupancy is given as,

$$BO\,(t+1) = BO(t) + S_R(t) - D_R(t) \qquad (8)$$

Where, $BO\,(t+1)$ is the buffer occupancy at time t+1, $S_R(t)$ is the traffic rate at the source node and $D_R(t)$ is the traffic rate at the destination.

After calculating the buffer occupancy value, the PID controller is deployed at the scheduler. The new data rate is estimated by the PID controller and the value is send back to the source node. On receiving the new data rate value, the source adjusts the transmission rate for upcoming data packets.
The generic equation given by the PID controller for rate adjustment is given below,
NR (t) =

$$D_R(t) - x(BO(t) - BO(0)) - \sum_{n=1}^{\mu_k} y_n (NR(t-n) - z(BO(t) - BO(t-1))) \qquad (9)$$

In the above equation, x, y_n and z are the proportional, integral and derivative general control components of the PID controller. These components will be set according to the stability of the network [12]. The process of PID based rate adjustment is picturized in figure-7.

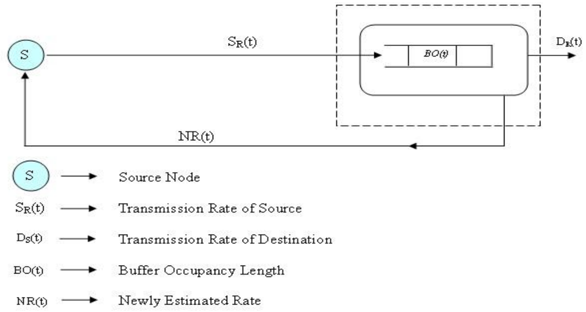

Fig 2: The process of PID Controller

Algorithm-1

Let BAF be the buffer allocation factor
Let minRSF be the minimum real time satisfaction factor
Let maxRSF be the maximum real time satisfaction factor
1. If RSF(t) > maxRSF , Then
    1.1 additional buffer ΔB is estimated.
    1.2 The BS can reserve the excessive buffer ΔB
2. Else If (minRSF > RSF (t) < maxRSF) Then
    2.1 The flow request does not need any reallocation
3. Else if (RSF (t) < minRSF) Then
    3.1 The flow request require reallocation
    3.2 Tries to allocate additional buffer from ΔB
    3.3 Packet flow rate is adjusted by the source as per equation (9)
End if

Algorithm-2

Let minNSF be the minimum non real-time satisfaction factor
Let maxNSF be the maximum non real-time satisfaction factor
1. If (NSF > maxNSF) Then
    1.1 additional buffer Δ_B is estimated.
    1.2 The BS can reserve the excessive buffer Δ_B
2. Else If (minNSF > NSF(t) < maxNSF) Then
    2.1 The flow request does not require any reallocation
3. Else if (NSF < minNSF) Then
    3.1 The flow request needs reallocation
    3.2 Tries to allocate additional buffer from Δ_B
End if

Further, we define a parameter $\Delta_{B,}$ where, Δ_B is the buffer factor that holds residual buffer of flow requests.
When RSF (t) exceeds maxRSF, the BS taken back the excessive buffer (Δ_B) and allocates them to the flow requests that needs additional buffer.
For non real time services, the SF can be computed by considering minimum reserved rate. Let R_{min} be the minimum reserved rate. The average transmission rate must be greater than R_{min}. The average transmission rate at t is given as,

$$\varsigma_i(t)\,(1 - 1/W_S) + r_i(t)/W_S \qquad (10)$$

4. SIMULATION RESULTS

4.1. Simulation Model and Parameters

Network simulator (NS2) [10] is used evaluate performance of the proposed Priority Based Adaptive Buffer Maintenance (PBABM) scheme. The proposed scheme is implemented over IEEE 802.16 MAC protocol. The simulation settings and parameters are summarized in table 1.

Table 1: Simulation Settings

Area Size	1000 X 1000
Mac	802.16
Clients	10
Radio Range	500m
Simulation Time	50 sec
Routing Protocol	DSDV
Traffic Source	CBR
Physical Layer	OFDM
Channel Error Rate	0.01
Area Size	1000 X 1000
Mac	802.16
Clients	10
Radio Range	500m

4.2 Performance Metrics

We compare our proposed PBABM scheme with the NCS [5] scheme. We mainly evaluate the performance according to the following metrics:
Aggregated Bandwidth: We measure the received bandwidth (in Mb/s) for CBR traffic of all flows **Bandwidth Utilization**: For each flow, we measure the utilization as the ratio of bandwidth received of each flow to the available channel bandwidth. The performance results are presented in the next section.

4.3 Results

Effect of varying the Traffic Flows

In order to measure the impact of buffer allocation on the traffic flows, we vary the CBR downlink traffic flows from 2 to 8.

(i) For CBR Traffic (Non-real time)

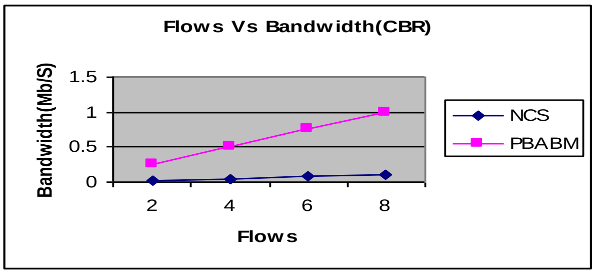

Fig 3: Flows Vs Bandwidth

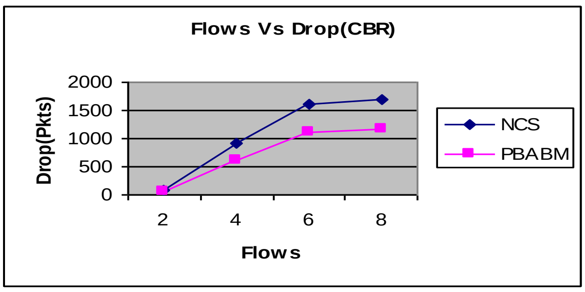

Fig 4: Flows Vs Drop

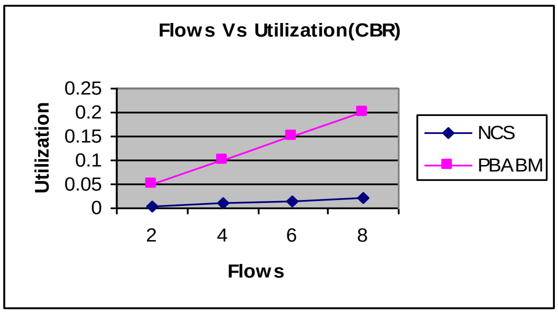

Fig 5: Flows Vs Utilization

From fig 3, we can see that the received bandwidth of our proposed PBABM is higher than the existing NCS technique. From fig 4, we can see that the packet drop of proposed PBABM is less than the existing NCS technique. From fig 5, we can see that the utilization of our proposed PBABM is higher than the existing NCS technique.

(ii) For Video (for real-time)

From figure 7, we can see that the received bandwidth of our proposed PBABM is higher than the existing NCS technique. From figure 8, we can see that the packet drop of proposed PBABM is less than the existing NCS technique. From figure 9, we can see that the utilization of our proposed PBABM is higher than the existing NCS technique.

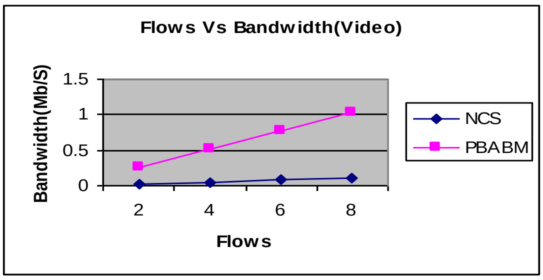

Fig 7: Flows Vs Bandwidth

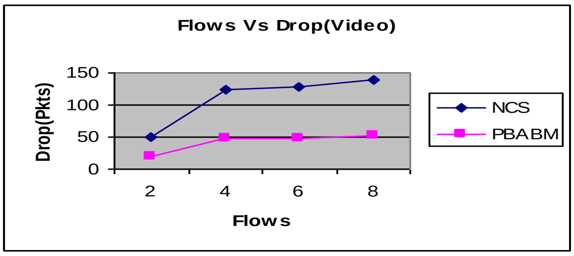

Fig 8: Flows Vs Drop

5. CONCLUSION

In this paper, we have proposed an Fuzzy based adaptive buffer allocation technique based on traffic classes in WiMAX networks. Initially buffer is allocated to the flow requests based on buffer allocation factor. This factor is estimated using fuzzy logic. The parameters such as number of user requests, flow rate, queue length and received signal strength are taken as inputs. The originally allocated buffer is verified periodically by buffer reallocation technique. It computes two different satisfaction factors for real time and non real time flows. Delay is considered as a metric for real time flows and minimum reserved data rate is for non real time flows. Based on estimated satisfaction factor, flow rate is adjusted for real time traffic using PID controller and additional buffer is allocated for non real time flows. Through simulation results, we have shown the performance of our proposed technique.

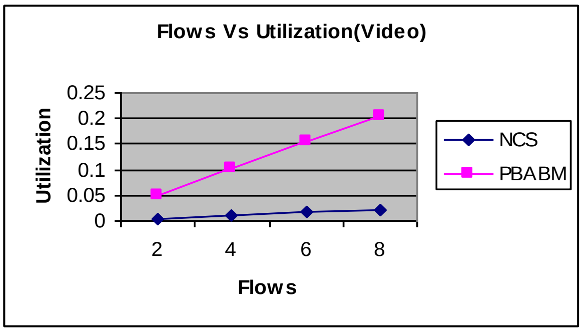

Fig 9: Flows Vs Utilization

6. REFERENCES

[1] D. David Neels Pon Kumar, K. Murugesan, "Performance Analysis of Neural Networks Based Priority Scheduler for WiMAX under Bursty Traffic Conditions", 2012

[2] S.Geetha, R.Jayaparvathy, S.Anand, "Dynamic Weight based Fair Resource Allocation in IEEE 802.16 Broadband Wireless Access Networks", May 2012

[3] Chakchai So-In, Raj Jain, and Abdel Karim Al-Tamimi, "Resource Allocation in IEEE 802.16 Mobile WiMAX", 2010.

[4] Nararat Ruangchaijatupon, Yusheng Ji, "Resource Allocation for Guaranteed Service in OFDMA based Systems", 2009

[5] Ronak Farhadi, Vahid Tabataba Vakili, Shahriar Shirvani Moghaddam, "A Novel Cross Layer Scheduling Algorithm for OFDMA-Based WiMAX Networks", 2011.

[6] Chung-Wei Lin, Yu-Cheng Chen, Ai-Chun Pang, "A New Resource Allocation Scheme for IEEE 802.16-based Networks", 2005

[7] W.B.HASSEN and Meriem AFIF, "A Novel Adaptive Resource Allocation Scheme With QOS Support in Mobile WiMAX Release 2 Wirlss Netwrks" June Edition, 2011.

[8] Jan Beneš, "MOBILE WIMAX RADIO RESOURCES MANAGEMENT SIMULATION" 2009

[9] Z. Kun, D. Niyato, and P. Wang, "Dynamic service selection and bandwidth allocation in IEEE 802.16m mobile relay networks," IEEE Journal on Selected Areas in Communications.

[10] S. J. K. J.Kumar and T. Purusothaman, " An Enhanced Scheduling Scheme for QoS Guarantee using Channel State Information in WiMAX Networks", European Journal of Scientific Research, pp. 285-292, Vol.64, No.2, 2011.

[11] http://en.wikipedia.org/wiki/PID_controller

[12] Hanwu Wang and Weijia Jia, "Effective Traffic Control in IEEE 802.16j WiMAX Networks", IEEE 18th Intr. Workshop on Quality of Service (IWQoS), pp- 16-18, 2010

[13] D. Karunkuzhali and D.C.Tomar, "Fuzzy Based Dynamic Buffer Management in Wimax 16m Networks", Journal of Computer Science, Volume 8, Issue 12, Pages 1946-1956, 2012