Traffic based QoS Specific Adaptive Buffer Allocation in WiMAX Networks

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Abstract— Buffer allocation in WiMAX maximizes the throughput by achieving high link utilization of system and minimizes the power consumption. In this paper, Traffic Based QoS Specific Adaptive Buffer Allocation in WiMAX networks is proposed to increase the system throughput by using an adaptive buffer allocation strategy. The buffer management block initially allocates buffer to the flow requests based on buffer allocation factor which is computed using fuzzy logic. The allocated buffer is verified periodically by buffer reallocation technique. It computes satisfaction factors for real time and non real time flows. Delay and minimum reserved data rate are considered as a metric for real time flows and for non real time flows respectively. Based on estimated satisfaction factor value, flow rate is adjusted for real time traffic using PID controller and additional buffer is allocated for non real time flows. We evident that our proposed scheme performs efficient adaptive buffer allocation, thereby reducing the transmission delay under heavy traffic condition with QoS features conciliation.

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

I. INTRODUCTION
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] which 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 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 SS 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

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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.\[3\]

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 [3, 12].

A. 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, level headers, management messages and so on) and to minimize power consumption while keeping feasible algorithm complexity and ensuring system scalability. IEEE 802.16 standard defined for WiMax does not specify any resource allocation mechanisms or admission control mechanisms. [3]

The resource allocation of OFDMA is in Time Division Duplex (TDD) mode [6]. The new transmission frame with multiple time slots are popped up on every pre-scheduled period. Frame resource is divided into chunks that are composed of a group of subcarriers which has 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 one who have high network costs are not guaranteed in order not to waste the precious bandwidth [4].

Although, a number of scheduling algorithms have been proposed such as Fair Scheduling, Distributed Fair Scheduling, MaxMin Fair Scheduling, Channel State Dependent Round Robin (CSD-RR), Feasible Earliest Due Date (FEDD) and Energy Efficient Scheduling. These algorithms cannot be directly used for WiMAX network due to the specific features of the technology [3].

In our previous paper [13], we have proposed a fuzzy based dynamic buffer management technique in WiMAX 16m network which performs buffer allocation and estimates packet dropping. This technique is operated in the base station (BS). As per application requirements, Base Station estimates the parameters such as flow rate, number of user requests, received signal strength and queue length and values are updated periodically. When a buffer request packet arrives at BS, buffer allocation factor (BAF) is estimated using the fuzzy logic applied over the parameters estimated. The user requests are sorted in the descending order of BAF. This ensures that the flow request with more BAF is admitted prior and rest of the flow requests await in queue. When a new request arrives, first its BAF is tested. If it indicates low, the request packet is dropped. Otherwise, the pending service request packets in the queue are emptied on analyzing their channel condition and the available buffer is allocated for new request. Our previous work did not distinguish the real-time from non-real time service requests and 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. To alleviate the problems, in this paper we proposed to deploy an adaptive buffer allocation technique based on traffic classes in WiMAX networks.

II. RELATED WORKS

Chakchai So-In et al., [3] focus on the management of resource allocation and scheduling in IEEE 802.16e based mobile WiMAX networks. Since mobile WiMAX uses orthogonal frequency division multiple access (OFDMA), the scheduling issues can apply for other OFDMA-based networks as well. Unlike wireless LANs, WiMAX networks incorporate several Quality of Service (QoS) mechanisms at the media access control (MAC) layer for guaranteed services for data, voice, and video.

NararatRuanchajatupon et al., [4] consider 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. They formulate the optimization problem with the objective of minimizing the total cost. This technique does not integrate the heuristic method to serve traffic with various requirements, which are defined by the IEEE 802.16 standard series.
Wafa Ben Hassen et al., [7] have presented a new resource allocation algorithm in downlink Mobile WiMAX Release 2 networks. The study considers three types of service including a real-time Polling Service (rtPS), a non-real-time Polling Service (nrtPS) and a Best Effort (BE) service. Each service type has its own QoS requirements (e.g. radio bandwidth, packet loss ratio, latency delay, etc.) each type of them is stored in a global buffer in order to reduce time processing. The proposed scheme includes three steps which are: radio resource reservation, arriving connections scheduling and adaptive resource allocation. A fourth step is introduced when a threshold for rtPS-class is defined based on the overall system capacity. Our scheduler gives the priority to rtPS service to ensure an adequate resource allocation without discriminating against nrtPS and BE services performances. This technique supports only for single-cell and does not support multi-cell system, mobility management.

Jan Beneš et al., [8] deal with RRM (Radio Resources Management) functions that play a very important role in Mobile WiMAX system. These functions are responsible for supplying optimum coverage, ensuring efficient use of physical resources, keeping the desired QoS and providing the maximum planned capacity. This approach of accessing Mobile WiMAX specific MAC and PHY layers and using its associated parameters in simulations to achieve main goal with best results. This technique does not deal with handling of data marked for repeated sending by ARQ.

III. TRAFFIC BASED QoS SPECIFIC ADAPTIVE BUFFER ALLOCATION IN WiMAX

In this paper, Traffic Based QoS Specific Adaptive Buffer Allocation in WiMAX network is proposed. 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, queue length, flow rate 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 and minimum reserved bandwidth allocated are considered as a metric for real time flows and for non real time flows respectively. Two different satisfaction factors are measured for real and non real time flows. By comparing the estimated satisfaction factor value with threshold values, flow rate is adjusted for real time traffic using PID controller and additional buffer is allocated for non real time flows.

A. Estimation of Metrics

Estimation of Queue Length and Flow Rate in the subscribed base station

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

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

The queue length is evaluated by considering the liquid fluid model,

$$Q_{ij}(t) = N_{i0}(t) IR(t) (1-R_p) BW_{ij}(t) \eta_{ij} \quad (1)$$

where $N_{i0}(t)$ is Number of base stations in a network, $IR(t)$ is input traffic flow to the subscribed base station, $N_{ij}(t)$ $IR(t)$ is aggregate downlink flow rate at base station $i$, $R_p$ is average packet error rate (PER) for abstracting the channel quality, $\eta_{ij}$ is average spectral efficiency of the network in bits/s/Hz, $BW_{ij}(t)$ is bandwidth allocated for draining the queue and $BW_{ij}(t) \eta_{ij}$ is 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. It can be viewed as the disturbance to the system. It is denoted as

$$IR(t) = IR_0 + \omega(t) \quad (2)$$

where, $IR_0$ 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 may occur due to the randomness of the packet arrival from the various applications in the network[9,13].

Estimation of Channel Condition

The physical layer constraints such as channel fading, multi-path propagation, scattering, reflection and other climatic effects on the channel reveals the channel condition. This channel condition of the mobile station 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)
\[
\text{RSS} = \frac{P_{tx} \cdot \alpha \cdot \beta \cdot H_{tx} \cdot H_{rx} \cdot \lambda^2}{(4 \cdot \gamma \cdot d)^2 \cdot \delta}
\]  \hspace{1cm} (3)

Where,

- \( P_{tx} \) is transmission power,
- \( \alpha \) is transmitter gain,
- \( \beta \) is receiver gain,
- \( H_{tx} \) is height of the transmitter,
- \( H_{rx} \) is height of the receiver,
- \( \lambda \) is wavelength,
- \( d \) is distance between the transmitter and receiver and
- \( \delta \) is system loss.

From the above computed RSS, the signal to noise ratio (SNR) is computed using Eq: (4)[10]

\[
\text{SNR} = \log_{10} \left( \frac{P_{tx}}{P_{rx}} \right) \text{ dB}
\]  \hspace{1cm} (4)

**Fuzzy based buffer allocation**

Upon receiving Bandwidth Request Packet, the buffers are allocated using buffer allocation factor (BAF) and estimated with the help of fuzzy controller, Figure 1. The fuzzy logic technique involved for buffer allocation is detailed below.

**Fuzzification**: In this step, the crisp inputs are changed into linguistic values. Each of these linguistic values are represented using a fuzzy set and each fuzzy set is related to a membership function used to describe the way by which the crisp input belongs to the set.

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). Triangular membership functions are used for representing each of the fuzzy parameters as it represents minimum and maximum boundary conditions. The membership to each of fuzzy variables is assigned using intuition method and this technique minimizes the computation complexity.

The membership function for these input parameters and output is represented as \( f(\text{UR}) \), \( f(R) \), \( f(Q) \), \( f(\text{RSS}) \) and \( f(\text{BAF}) \).

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

**Flow rate**: The flow rate varies based on the user requirements as per the applications. 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 and 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**: It describes the communication channel quality between nodes. The linguistic values associated with the membership function \( f(\text{RSS}) \) 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 \( f(\text{UR}) \cdot f(R) \cdot f(Q) \cdot f(\text{RSS}) \). Inference mechanism in fuzzy logic is based on fuzzy rules that connect input and output parameters and the membership functions for input and output parameters. Fig 1 shows The fuzzy controller design for buffer allocation and the fuzzy inference system is designed based on 16 rules described in Table 1. In order to
demonstrate the designed fuzzy inference system, only one rule is taken into account to show how the inference engine works and outputs of each rule are each rule are combined for generating fuzzy decision.

Rule 1: If (UR=low, R=high, Q=high and RSS=high) Then BAF=high End if

Defuzzification: It is the method by which a crisp value is extracted from a fuzzy set as illustration value. During fuzzy decision-making, the centroid of area technique is taken into account for defuzzification. The Defuzzifier is based on Equation (5).

\[ F_{\text{priority}} = \left[ \frac{\sum_{\text{all rules}} z_i \cdot \eta(z_i)}{\sum_{\text{all rules}} \eta(z_i)} \right] \]  

where \( F_{\text{priority}} \) = Degree of decision making, 
\( z_i \) = fuzzy variable 
\( \eta(z_i) \) = membership function.

The output of the fuzzy priority function is modified to the crisp value based on the above defuzzification method.

Algorithm for Buffer Allocation

Step 1: BS estimates the parameters such as number of user requests, queue length, received signal strength and flow rate and updates the values periodically

Step 2: When the buffer request packet arrives, BAF value is estimated as per step 3.

Step 3: BAF estimation done by taking the estimated parameters in step 1 as input using fuzzy logic technique and generates BAF as output

Step 4: The requests are sorted in the descending order of BAF

If BAF = high
Then
Allocate the Buffer for the user request
Buffer Allocated = (Buffer Requested by User*BAF)
Remaining flow requests waits in a pending queue.

End if

<table>
<thead>
<tr>
<th>Table I: Fuzzy Rules</th>
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<tbody>
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<td>User Requests</td>
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<td>Low</td>
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D. 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. Two different metrics are used to compute the SF 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,

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

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,

$$D_t = \frac{QS_N}{m} T_{OFDM} * f$$

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 $QS_N$, if $m$, $T_{OFDM}$ and $f$ are constants. The estimated $RSF(t)$ value is compared with two predefined thresholds namely $\text{minRSF}$ and $\text{maxRSF}$. If the satisfaction factor is less than $\text{minRSF}$, then the source adjusts the flow rate using PID controller.

### 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. This controller attempts to minimize the error by adjusting the process control inputs.

The PID controller error calculation algorithm involves three separate constant parameters, the proportional $P$; the integral $I$ and derivative values $D$. $P$, $I$ and $D$ values can be interpreted in terms of time: $P$ depends on the present error rate, $I$ depends on the accumulation of past errors rate and $D$ is a prediction of future errors, based on the current rate of change in the flow rate. The weighted sum of these three actions is used to adjust the flow rate of real time service [11].

Here, the flow rate is controlled considering buffer occupancy of the scheduler. Statistically, the buffer occupancy is given as,

$$BO(t+1) = BO(t) + S_{R}(t) - D_{R}(t)$$

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. By receiving the new data rate value, the source adjusts the transmission rate for upcoming data packets.

The general equation given by the PID controller for flow rate adjustment is given below,

$$NR(t) = D_{R}(t) - x(BO(t) - BO(0)) - \sum_{n=1}^{k} y_{n}(NR(t-n) - z(BO(t) - BO(t-1)))$$

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-2. Further, we define a parameter $\Delta_{B}$ where, $\Delta_{B}$ is the buffer factor that holds residual buffer of flow requests.

When $RSF(t)$ exceeds $\text{maxRSF}$, the BS taken back the excessive buffer ($\Delta_{B}$) and allocates them to the flow requests that needs additional buffer.

### Algorithm for Rate adjustment for Real time flow

1. Let $BAF$ be the buffer allocation factor for the User request
2. Let $\text{minRSF}$ and $\text{maxRSF}$ are the minimum real time satisfaction factor and the maximum real time satisfaction factor
3. If $RSF(t) > \text{maxRSF}$, Then
   - additional buffer $\Delta_{B}$ is estimated and BS can reserve the excessive buffer $\Delta_{B}$
4. Else If ($\text{minRSF} < RSF(t) < \text{maxRSF}$) Then
   - The real time flow request does not need any reallocation
5. Else if ($RSF(t) < \text{minRSF}$) Then
   - The real time flow request require reallocation of additional buffer from $\Delta_{B}$
   - Packet flow rate to the destination is adjusted by the source as per equation (9)
For non real time services, the SF can be computed by considering minimum reserved rate. Let $R_{\text{min}}$ be the minimum reserved rate. The average transmission rate at $t$ is given as,

$$\varsigma_i(t) \left( 1 - \frac{1}{W_S} \right) + \frac{r_i(t)}{W_S} \quad (10)$$

Here, $W_S$ is the window size where we compute the average transmission rate and the throughput factor will be,

$$\text{NSF} = \frac{R_{\text{min}}}{\varsigma_i(t)} \quad (11)$$

Where, NSF is the satisfaction factor of non real time flows. The average transmission rate must be greater than $R_{\text{min}}$.

![Fig 2: The process of PID controller](image)

Where S is Source node, $S_R(t)$ is Transmission rate of source, $D_s(t)$ is Transmission rate of destination, BO(t) is Buffer occupancy length and $N_R(t)$ is Newly estimated length.

**Algorithm for Rate adjustment for Non-Real time flow**

Let $\text{minNSF}$ and $\text{maxNSF}$ be the minimum and maximum non real-time satisfaction factor.

If ($\text{NSF} > \text{maxNSF}$) Then

- additional buffer $\Delta_B$ is estimated.
- The BS can reserve the excessive buffer $\Delta_B$

Else If ($\text{minNSF} > \text{NSF} < \text{maxNSF}$) Then

- The flow request does not require any buffer reallocation

Else if ($\text{NSF} < \text{minNSF}$) Then

- The flow request needs buffer reallocation
- Tries to allocate additional buffer from $\Delta_B$

End if

End if

### IV. Simulation Results

**A. Simulation Model and Parameters**

Network simulator (NS2) [10] is used to evaluate performance of the proposed scheme Priority Based Adaptive Buffer Maintenance (PBABM) and this scheme is implemented over IEEE 802.16 MAC protocol. For the simulation, clients (SS) and the base station (BS) are deployed in a 1000 meter x 1000 meter region for 50 seconds simulation time and all the nodes taken for simulation have the same transmission range of 500 meters. In the simulation, the CBR traffic is used. There are 8 downlink traffic flows from BS to SS. The simulation settings and parameters are summarized in table 2.

**B. Performance Metrics and Results**

The proposed PBABM scheme is compared with the Novel Cross Layer Scheduling Algorithm NCS [5] scheme. The performance evaluation is done based on the following metrics:

- **Aggregated Bandwidth**: The received bandwidth (in Mb/s) is measured for CBR traffic of all flows
- **Bandwidth Utilization**: For each flow, the utilization is measured as the ratio of bandwidth received of each flow to the available channel bandwidth in the network.
- **Average end-to-end delay**: The average of overall surviving data packets from the sources to the destinations is measured.

The performance results are presented as below. 

*Effect of varying the Rate of flow*
In our experiment we vary the rate as 250, 500, 750 and 1000 Kb.

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

<table>
<thead>
<tr>
<th>TABLE II: SIMULATION SETTINGS</th>
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<tbody>
<tr>
<td>Area Size</td>
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<tr>
<td>Mac</td>
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<tr>
<td>Clients</td>
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<td>Radio Range</td>
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<td>Simulation Time</td>
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<td>Routing Protocol</td>
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<td>Traffic Source</td>
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<td>Physical Layer</td>
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<td>Channel Error Rate</td>
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<td>Packet Size</td>
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<tr>
<td>Frame Duration</td>
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<td>Transmission Rate</td>
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<td>No. of Flows</td>
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</table>

From figure 3, we can see that the received bandwidth of our proposed PBABM is higher than the existing NCS technique. From figure 4, we can see that the delay of our proposed PBABM is higher than the existing NCS technique. From figure 5, we can see that the packet drop of proposed PBABM is less than the existing NCS technique. From figure 6, we can see that the utilization of our proposed PBABM is higher than the existing NCS technique.
(ii) For Video Traffic (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 delay of our proposed PBABM is higher than the existing NCS technique. From figure 9, we can see that the packet drop of proposed PBABM is less than the existing NCS technique. From figure 10, we can see that the utilization of our proposed PBABM is higher than the existing NCS technique.

V. CONCLUSIONS

In this paper, Traffic Based QoS Specific Adaptive Buffer Allocation in WiMAX networks is proposed to increase the system throughput by using an adaptive buffer allocation strategy. The buffer management block initially allocates buffer to the flow requests based on buffer allocation factor which is computed using fuzzy logic. The allocated buffer is verified periodically by buffer reallocation technique. It computes satisfaction factors for real time and non real time flows. Delay and minimum reserved data rate are considered as a metric for real time flows and for non real time flows respectively. Based on estimated satisfaction factor value, flow rate is adjusted for real time traffic using PID controller and additional buffer is allocated for non real time flows. We evident that our proposed scheme performs efficient adaptive buffer allocation, thereby reducing the transmission delay under heavy traffic condition with QoS features conciliation.

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