Performance Evaluation of Resource Allocations in WiMAX Network

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Abstract: Future generation networks will be characterized by variable and high data rates, high security issues, Quality of Services (QoS), both within a network and between networks of various technologies and service providers. Worldwide Interoperability for Microwave Access (WiMAX) system, based on IEEE 802.16e has gained much attention recently for its capability to support high and variable transmission rates, high security issues and Quality of service (QoS) for different applications. In this paper, we evaluate the performance of resource allocation schemes, used to allocate the resources in WiMAX networks.

Keywords*:* WiMax, Resource allocation, Quality of Service (QoS), IEEE 802.16

1. Introduction

WiMAX is introduced by the Institute of Electrical and Electronic Engineers (IEEE) which is standard designated 802.16d-2004 (used in fixed wireless applications) and 802.16e-2005 (used in mobile wireless applications) to provide a worldwide interoperability for microwave access. Since next generation broadband wireless applications require high data rate, low latency, minimum delay, real-time applications; in short highly demanding QoSs. WiMAX is suitable technology to provide these requirements.

Network reference model (NRM) divides the end-to-end WiMAX network architecture into three logical parts: **1)** Mobile Station (MS), **2)** Access service Network (ASN), **3)** Connectivity Service Network (CSN). ASN performs mobility-related functions, such as handover, location management, and paging within the ASN and Radio resource management (RRM). Functions, IP address allocation to the MS for user sessions and Subscriber billing, are performed by CSN.

Wireless communication is based on orthogonal frequency division multiplexing (OFDM) technology and this enables going towards the 4G mobile in the future. So WiMAX is also called typical OFDMA system where radio resource is partitioned in both time and frequency domains. Basically OFDMA is multiuser OFDM.

OFDM belongs to a family of transmission schemes called *multicarrier modulation*, which is based on the idea of dividing a given high-bit-rate data stream into several parallel lower bit-rate streams and modulating each stream on separate carriers—often called subcarriers, or tones. The high-data-rate systems will generally have $t \geq T_s$, where T_s is symbol duration and t is channel delay spread. Then the ISI becomes very severe in WiMAX systems. OFDM technology is used in high-data-rate applications due to its efficient and flexible management of inter-symbol interference (ISI) in highly dispersive channels i.e. OFDM is an elegant and effective technique for overcoming multipath distortion.

WiMAX systems define two layers: **1)** Physical Layer, **2)** MAC Layer. The purpose of the PHY layer [2] is to reliably deliver information bits from the transmitter to the receiver, using the physical medium. It specifies the frequency band, the modulation scheme, error-correction techniques, synchronization between transmitter and receiver, data rate and the multiplexing techniques. The WiMAX physical layer is based on orthogonal frequency division multiplexing. The Media Access Control (MAC) layer, which resides above the PHY layer, is responsible for controlling and multiplexing various links over the same physical medium. It performs Segment or concatenate the service data units (SDUs) received from higher layers into the MAC PDU (protocol data units), schedule MAC PDUs over the PHY resources, gives support to the higher layers for mobility management, provide security and key management and perform power-saving mode and idle-mode operation.

The WiMAX technology has a very rich set of features like robust security, secure handover mechanisms, multiple antenna technique, adaptive modulation and coding (AMC) and OFDMA to provide multimedia applications. WiMAX defines five services in which all applications are supported. Four classes of QoS were defined in IEEE 8002.16d-2004 (later added in WiMax) standards:

1. Unsolicited Grant Service (UGS)
2. Real-Time Polling Service (rtPS)
3. Non-Real-Time Polling Service (nrtPS)
4. Best Effort (BE)

A fifth one has been added with IEEE 802.16e-2005:

5. Extended real-time Polling Service (ertPS) class.

UGS is designed to support real time data streams to support fixed-size data packets issued at periodic intervals. rtPS supports real-time service consisting of variable bit-rate (VBR) data packets that are issued at periodic intervals. nrtPS supports delay tolerant data streams consisting of variable size data packets for which a minimum data rate is required. Best Effort (BE) is used for best effort traffic which does not guarantee any Quality of service (QoS), like the short length FTP or the email. ertPS is based on the efficiency of both UGS and rtPS.

1. New Optimization Scheme for Resource Allocation in OFDMA based WiMAX Systems

Proportional Fair (PF) [1] optimization does not consider user's priority and treats every user equally and unbiased way. This unbiased instantaneous Proportional Fair Optimization is based on instantaneous computation of proportional fair metric (PF) and the resultant optimization does not consider the time diversity and priority of the users. Hence PF is not the most optimized solution for delay-tolerant applications, like nrtps, BE class traffic in IEEE 802.16. Therefore this kind of subcarrier allocation may deprive the higher priority user with bad channel condition to maintain its QoS. If the deprived user is a high priority customer, then the allocation is very much unacceptable from service provider as well as user perspective. PF in OFDMA system maximized the sum of logarithmic mean user rates. If the total available bandwidth B is partitioned into N equal narrowband OFDMA subcarriers then R_{kt} is the achieved data rate of the k^{th} user at allocation epoch t and R_{kt} is expressed as,

$$R_{kt} = \frac{B}{N} \sum_{n}^{N} \rho_{knt} f(h_{knt})$$

$$\sum_{n=1}^{N} \sum_{k=1}^{K} \rho_{knt} = N$$

Where $\boldsymbol{\rho}_{\mathbf{knt}}$ is the subcarrier assignment matrix at allocation epoch t and $R_{knt} = f(h_{knt})$ be the instantaneous achievable rate for k^{th} user when n^{th} subcarrier is allocated at t^{th} time instant.

The proportional fairness index at t^{th} instant is given by:

$$PF\,|_t = \max \sum_{n=1}^{N} \sum_{k=1}^{K} \ln R_{kt}$$

To overcome drawback of this algorithm biased proportional fair optimization has to be introduced that is called Priority indexed long-term (PILTPF) **[1]** resource allocation algorithm. This Algorithm dynamically allocates the OFDMA resources to the users to meet their QoS requirement, which is dependent on user's derived priority profile. PILTPF has two parts and operates sequentially. First part estimates the user's priority (priority index PI) based on its current QoS class, available buffer–size, delay limit, Signal to Noise Ratio (SNR), and minimum data rate requirement and the next one is subcarrier allocation. Higher the priority of the user, higher the magnitude of PI and more is the chance to get the subcarrier allocated even in bad channel condition and high data rate. Subcarrier allocation part assigns the appropriate subcarrier to the user to optimize the system performance. According to this algorithm, the highest priority must be allocated by the best of the subcarriers. PILTPF algorithm exploits the time diversity as well as considers the priority of the user by calculating priority index (PI) based on Proportional Fair (PF) metric as follows:

$$PI_k = \frac{Q_k \times \gamma_k}{(bs_{kMAX} - bs_k |_t) \times (\delta_{MAX} - \delta |_t)}$$

Where Q_k is quality of service,γ_k is user data rate and also bs_{kMAX} is max buffer size, bs_k is used buffer size, δ_{MAX} is max delay limit and $\delta |_t$ is elapsed delay for k^{th} user at current allocation instant t.

The PILTF algorithm is described as follows:

Step 1: Set initial mean achievable data rate as $E(R_{kt}) |_{t=1}^{T} = \varepsilon$ and t=0, where ε is a small number.

Step 2: Find PI_k for all users.

Step 3: Calculate the data rate achieved as per $R_{kt} = \sum_{n=1}^{N} \rho_{knt} \times R_{knt}$ for all the users at the t^{th} instant.

Step 4: Find the mean data rate achieved by the k^{th} user at t^{th} instant.

Step 5: if $E(R_{kt}) |_{t=1}^{T} \geq \gamma_k$, find next best k, else continue.

Step 6: $t+T_f$ and go to step 2.

Under the defined simulation parameters, simulation results shows that PILTF algorithm provide very high data rate compare to PF algorithm because it considers both the priority of the user and the time diversity gain and also it results in better performance both in terms of throughput and QoS guarantee.

2. Dynamic resource allocation in mobile WiMAX using Particle Swarm Optimization techniques

A lot of methods have been presented in literature on how to allocate resources which are generally classified as either dynamic resource allocation or fixed resource allocation. In fixed resource allocation, the resources are allocated to users when admitted throughout the call duration. In dynamic resource allocation, the allocation of the resources is dynamically changed as the channels change. Therefore allocating radio resources dynamically in wireless network is highly complex and non-linear. It becomes even more complex when the wireless network is designed for heterogeneous traffics with different quality of service (QoS) requirement like WiMAX network. In dynamic resource allocation, there are two optimization techniques that exist in literature, one minimizes the total transmit power subject to users data rate and the other maximizes the user data rate within constraint to total transmit power. Allocation of sub-carriers to minimize the total transmit power with constraint on users' data rate, this technique is called margin adaptive. When each user data rate is maximized with constraint to total transmits power, which is called rate adaptive. Here a system is modelled in which m $(1 \leq m \leq M)$ users communicating with base station using N subcarriers. Each user m, feedbacks the channel condition to the allocation block, in which subcarriers, bits and power are assigned to each user. Then each channel is modelled with six independent Rayleigh multipath fading, each with amplitude A and initial phase shift of C. It is assumed that all users experience independent fading. Particle swarm optimization [2] is a bio-inspired optimization technique, where particles are randomly deployed in search space with n dimensions. Here this algorithm considered 2-dimensional space x, y correspond to I, Q of the channel gain. The search space is from -2 to 2 in both dimensions. The channel gain of each user m $(1 \leq m \leq M)$ on subcarrier n $(1 \leq n \leq N)$ is given by $\Phi_{m,n}$ with AWGN. The parameter fitness function is thus given by; $f = f(I,Q)$ and $d_{m,n}$, where $d_{m,n}$ is binary function. $d_{m,n}$ is 1when subcarrier n is allocated to m and zero otherwise. This function forms the un-assigned channel gain of a subcarrier. This is stored in form of matrix with dimensions M x N. the fitness function of m^{th} user $f_m(n)$ is the n^{th} subcarrier gain obtained from gain profile as $f_m(n) = \Phi_{m,n} d_{m,n}$.

The general PSO algorithm is given as follows:

Initialize the particles across the search space for each user,

Calculate the no. of subcarriers required

Loop while not optimized or less than no. of iteration

For each particle in the search space

if (local gain > global gain)

Replace the global gain with local gain

Calculate the local gain

Calculate new position and velocity

Move to the new position

end if

end loop

return the global channel gain.

end for

The system is optimized when the PSO converged or the number of iterations reached its maximum defined value. *Pbest* is the local channel gain found by any particle while *Gbest* is the global channel gain found by all the particles, which has not been assigned to any user. Then we compare the three algorithms (Linear Algorithm, Root-Finding Algorithm and PSO approach) as variation in no. of users under the above simulation parameters. Results show that for small no. user capacities of Linear, Root-Finding and PSO approach are very close to each other but capacity of Linear approach is always greater than the both. But the average CPU time is always less than for the PSO approach.

3. A Novel Algorithm for Efficient Paging in Mobile WiMAX

Since most of the time, mobile stations (MSs) are powered on in mobile WiMAX networks but are not in active call sessions. To use these durations as battery conserving opportunities, *idle mode, location update, and paging* operations are specified in IEEE 802.16e standard. So mobile WiMAX networks use idle mode operation to conserve battery power when a mobile station (MS) is not engaged in active call sessions. When MS enters into Idle mode (low-power mode), the MS relinquishes all of its connections and states associated with the base station (BS) it was last registered with. While in idle mode the MS periodically listens to the radio transmissions for paging messages, in a deterministic fashion. The period for which the MS listens to paging messages is known as “paging listen interval” (PLI) and the period for which the MS powers off its radio interface is known as the “paging unavailable interval” (PUI). One paging unavailable interval and one paging listening interval constitute a *paging cycle* (PAGING_CYCLE). Therefore, once in every PAGING_CYCLE interval the idle-mode MS wakes up and listens for paging messages. When traffic arrives for the idle-mode MS the network performs paging to locate the MS and to bring it back to active mode. The performance of paging mechanism in IEEE 802.16e based mobile WiMAX networks can be specified by two parameters: *paging signalling overhead* and *paging latency*. **Paging signaling overhead** is defined as the number of bits per second used for paging one idle-mode MS. **Paging latency** is defined as the time delay between the initiation of paging operation by the network and the completion of MS’s response to the paging operation. The objective of this algorithm [3] is to minimize paging signalling overhead while ensuring that paging latency is bounded by a maximum desired value, called paging latency upper limit. Therefore, this algorithm achieves a good trade-off between paging latency and paging signalling overhead.

In this algorithm, idle-mode MSs are grouped into different sets, referred to as paging sets, in such a way that the paging information of these MSs can be combined into one *mobile paging advertisement* (MOB-PAG-ADV) message. This reduces the overhead associated with paging operation.

Paging Algorithm:

This algorithm decides the maximum number of MSs who’s paging information can be combined in one MOB-PAG-ADV message. Once this number is determined, the idle-mode MSs can be grouped into different sets with each set containing the maximum number of MSs. These sets are referred as paging sets. We assume that the numerical values of PAGING_CYCLE and the PLI for each MS are already known to the PC. Moreover, we consider that the numerical values of PAGING_CYCLE and PLI are same for all idle-mode MSs.

The expression for *p* is given by:

$$p = 1 - e^{-\lambda(PAGING\_\mathrm{CYCLE})}$$

Where p is probability that an idle-mode MS is paged in a paging cycle and **λ** is call arrival rate (number of calls per second) of each idle mode MS.

The number of idle-mode MSs of a particular paging set that are paged simultaneously in a cell in one paging cycle is:

$$\alpha = \frac{L \times p}{K}$$

Where L is maximum cardinality of a paging set, K is number of cells in a paging group PG.

T: The upper limit for paging latency, i.e., the paging latency should be limited to T

R: maximum number of idle-mode MSs in a cell that can perform initial ranging simultaneously such that paging latency does not exceed its upper limit T

To ensure that the paging latency is limited to T, α should be equal to *R*.

Using above two equations, the expression for L:

$$L \leq \frac{K \times R}{p} = \frac{K \times R}{1 - e^{-\lambda(PAGING\_CYCLE)}}$$

Therefore, the number of paging sets, *m*, to accommodate N number of idle-mode MSs in a PG is

$$m = \left\lceil \frac{N}{L} \right\rceil$$

The expression for *N* is given by

$$N = \rho(K\pi r^2)\omega n$$

Where ρ is the population density of the area where a WiMAX network is deployed, *r* is the radius of one WiMAX cell, ω is the fraction of population using WiMAX network, and η is the fraction of WiMAX users in idle mode. The paging sets are as {S1, S2, S3, ...Sm}. Once, *m* and *L* are determined the PC assigns idle mode users to individual sets.

4. A Cross-layer queue management algorithm in 802.16 wireless networks

A lot of factors can impact on the performance of wireless networks due to the characteristic of wireless link. These factors includes: limited bandwidth, increased delay, congestion, channel loss and mobility, etc. Congestion is regarded as one of the main factor for degradation the performance in wireless networks, those exists in 802.16 wireless networks too. There will be packet loss in wire line networks as well as in wireless networks due to congestion. In recent research, queuing delay is considered as one parameter of congestion detection is usually used to replace packet loss. The congestion detection becomes efficient by queuing delay method compare to packet losses method. To control congestion better an active queue management (AQM) algorithm named WDBQ (Wireless Delay-Based Queue) is developed according to the characteristic of 802.16 wireless networks.

This algorithm [5] detects congestion by using packet queuing delay, and then control network congestion by setting a congestion detection threshold. Congestion detection threshold can be decided by round trip time (RTT) of flows passing by base station.

$$Th = \begin{cases} \frac{1}{n}\sum_{i=1}^{n} k_i, & n \leq m \\ \frac{1}{m}\sum_{i=1}^{m} j_i, & n > m \end{cases}$$

where *n* is current total flow numbers passing by the base station; k_i is current packet round trip time estimate of flow *i*; *m* is the number of sample flow chose for calculating *Th*, which can be set according to the available buffer size in base station; j_i (i = 1, 2, ... , n) is the round trip time estimate of sample flow *i* chose for calculating *Th*.

Let *l* denote the number of flow that can be kept in flow status information table (FSIT), then

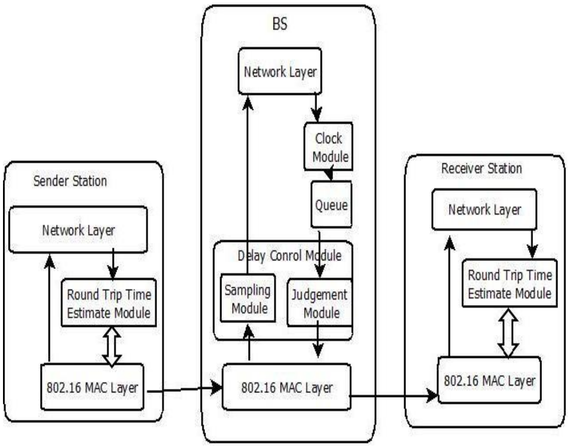
$$l = \begin{cases} [m, n], \mathrm{m} < \mathrm{n} \\ n, m \geq n \end{cases}$$

Base station chooses *m* flows (or *n* flows, when $n \leq m$) as sample flows to calculate *Th*. Here round trip time includes packet round transmission time and queuing delay. As the network status changes, the congestion detection threshold should be updated according to the change of network status that if the network status is unchanged, the *Th* will keep original value. If *T* denotes the cycle of the threshold updating and we should update threshold in every cycle. If a packet queuing delay exceeds the threshold, the packet will be marked, if not the packet will be sent directly. And marking information feedback to sender station for retransmission.

Algorithm Implementation:

The algorithm includes a cross layer interface module named round trip time estimate module (RTTEM) in sender station and receive station. The module in sender station needs to estimate round trip time and fill current timestamp in the message for measuring round trip time. And in receive station the module insert the timestamp in the packet from the sender station to the packet which will be transmitted to the sender.

Base station includes a cross-layer interface module DCM.DCM is composed of Sampling Module, Judgment Module and Clock Module. Sampling Module collects the information of the status and round trip time of the flows passing by base station and sends these to the Judgment Module. Clock Module attaches a timestamp for every packet transmitted from the upper layer in order to calculate queuing delay. Judgment Module calculates threshold.



The algorithm needs to add three messages for sending timestamp to receive station from sender station to get the sample of round trip time and transmitting the estimate of round trip time to the base station. These three messages are round trip time transmit message (RTTS), round trip time request message (RTTQ) and round trip time respond message (RTTR). RTTS message transmits the estimate of round trip time to base station to calculate congestion threshold. RTTQ message transmits the timestamp from the sender station to the receive station to request for round trip time. And RTTR message is for receive station to respond RTTQ and send back it to sender station for getting sampled round trip time. Then RTTEM in sender station calculates the estimate. Then the sender station transmits the estimate to base station by RTTS message. By setting the values of simulation parameters (l, m and n) WDBQ algorithm is compared with different algorithms then simulation results show that GOODPUT is always greater than the all other algorithms while LOSS RATE with network status change is always less than the others.

5. Conclusion

In this paper we have studied different resource allocation and management schemes used in WiMAX networks. So the real time Quality of services (QoS) to the respective users can not be provided unless the system resources, like bandwidth and transmitter power which are limited, are intelligently used and properly optimized. In all the schemes performance of the WiMAX network has been improved in terms of data rate, overall capacity, throughput etc.

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