WiMAX System Design and Evaluation Methodology using the NS-2 Simulator

X. Guo†, R. Rouil‡, C. Soin⋆, S. Parekh§, B. Sikdar† and S. Kalyanaraman†∗
†Department of ECSE, Rensselaer Polytechnic Institute, Troy, NY 12180
‡National Institute of Standards and Technology, Gaithersberg, MD 20899
⋆Department of CSE, Washington University, St. Louis, MO 63130
§Bell Labs, Alcatel-Lucent, Murray Hill, NJ 07974
∗IBM India Research Labs, Bangalore, India

***Abstract*— The development of the IEEE 802.16 mobile broadband wireless access or WiMAX standards promises to usher the next generation of mobile wireless networks. There is considerable ongoing interest in evaluating the performance of such networks with simulations forming the primary tool for conducting such studies. This paper presents the design and implementation of the Medium Access Control (MAC) and Physical layers of WiMAX based on IEEE802.16e standard on the *ns-2* network simulator. Appropriate abstractions and methodologies are carefully chosen to ensure the maximal conformance of simulation to reality while keeping the computational complexity acceptable. Results that validate the accuracy of the simulator as well as those that describe its performance are presented.**

I. Introduction

With the shift towards deployment of Broadband Wireless Access (BWA), the research and development of BWA related technologies has received increasing interest in the industry and academia. Wireless Broadband is a fairly new technology that provides high-speed wireless Internet and data network access over a wide area. According to the IEEE 802.16 standard, broadband refers to "having instantaneous bandwidth greater than around 1 MHz and supporting data rates greater than about 1.5 Mbit/s". This means that Wireless Broadband features speeds roughly equivalent to wired broadband access, such as that of ADSL or a cable modem. Fixed BWA technologies like Local Multipoint Distribution Service (LMDS) and Multichannel Multipoint Distribution Service (MMDS) use a Base Station (BS) to provide broadband wireless access service to discrete Subscriber Stations (SS). The IEEE 802.16e standards allow for both Fixed and Mobile BWA technology.

To provide BWA service, the IEEE 802.16 specification uses a wide range of RF spectrum and WiMAX (worldwide interoperability for microwave access) based systems can function on any frequency below 66GHz. While there is no uniform global licensed spectrum for WiMAX, the WiMAX Forum has published three licensed spectrum profiles: 2.3 GHz, 2.5 GHz and 3.5 GHz. Typically the bandwidth allocation is 20 or 25 MHz (United States) and 28 MHz (Europe) in 10-66GHz band or variable channel bandwidth in the range 1-30MHz in the 2-11GHz band. For Time Division Duplex (TDD) operation, WiMAX profiles define channel sizes of 3.5 MHz, 5 MHz, 7 MHz and 10 MHz. The mobile profiles are 5 MHz, 8.75 MHz and 10 MHz. Since October 2007, the Radiocommunication Sector of the International Telecommunication Union (ITU-R) has decided to include WiMAX technology in the IMT-2000 set of standards. This enables spectrum owners (specifically in the 2.5-2.69 GHz band at this stage) to use Mobile WiMAX equipment in any country that recognizes the IMT-2000.

With the IEEE 802.16 or WiMAX standards gaining popularity, researchers in both the industry and the academia are interested in evaluating its system level performance in a number of settings. While some work has been done on developing analytic models, simulation platforms remain the primary tool for performance evaluation of WiMAX systems. Though some commercial vendors like OPNET have started to include WiMAX modules in their simulation software, open source platforms have not seen similar activities. Our work addresses the development of a methodology for the system level simulation of WiMAX and its implementation in the open source, *ns-2* network simulator. *ns-2* is a discrete event simulator that enjoys widespread popularity in the academia and industry due to its extensibility, group support and plentiful online documentation. Because of its open source policy, *ns-2* comes with a rich library of network topology and traffic generators, applications and protocols at all network layers that are easy to modify [1], [7]. The accuracy of the various transport and routing protocols implemented in *ns-2* as well as its traffic generators have been independently verified.

This paper describes the simulation methodology adopted in our *ns-2* extension for the WiMAX system level simulations. A key part of our work is the introduction of realistic physical layer simulation in the *ns-2* code and the support for Orthogonal Frequency Division Multiple Access (OFDMA) transmissions. In addition, various MAC layer features of the IEEE 802.16 as well as scheduler support have also been added. Details of the physical and MAC layer abstractions made for the simulation purposes and their impact on the accuracy and faithfulness of the results are presented. The paper also presents results to validate the accuracy of the simulator as well as its performance in terms of execution times.

The rest of this paper is organized as follows. Section II presents an overview of the IEEE 802.16 architecture. Section III presents the details of the abstraction and modeling of WiMAX physical layer while Section IV focuses on the design and implementation of the key features of the WiMAX MAC layer. The performance of the simulator in terms of its execution time is described in Section V. Finally, Section VI presents the conclusions and scope for future work.

II. Overview of the IEEE802.16 Architecture

This section presents a brief overview of the architecture of WiMAX systems. The simulation implementation architecture is shown in Figure 1 and is based on [2], [3], [4], [8].

In Figure 1, the service specific Convergence Sublayer (CS) resides on top of the MAC Common Part Sublayer (CPS), via the MAC Service Access Point (SAP). The MAC Management module is mainly used to control the management messages and algorithms behind them, like the downlink MAP (DL-MAP), uplink MAP (UL-MAP), initial ranging, registration etc. The Connection Classifier module is used to manage service flow and Quality of Service (QoS) related issues. Five types of connections (Unsolicited Grant Service (UGS), real-time Polling Service (rtPS), non-real time Polling Service (nrtPS), extended real-time Polling Service (ertPS) and Best Effort (BE)) and their buffers are needed to handle the traffic along with corresponding algorithms to schedule the transmissions. The scheduler module is responsible for the burst allocation in both the downlink and uplink traffic and handle all bandwidth allocation related processes. The Code Division Multiple Access (CDMA) code module focuses on the CDMA ranging related issues. Fragmentation and packing is supported in our architecture. Physical Layer abstraction and simulation methodology are deployed and these are described in detail in Section III.

III. Physical Layer Implementation

A. Propagation and Fading Modeling

A key contribution of our work is the addition of a realistic physical layer simulation mechanism in the *ns-2* simulation software. In our physical layer implementation, a radio propagation model is used to predict the received signal power of each packet. Using this radio propagation model, the simulator takes into account the effects of fading, shadowing as well as bulk path loss. For each packet, the simulator calculates the received power as well as the interference caused by simultaneous transmissions. The resulting signal to interference and noise ratio (SINR) value is then mapped to the bit error rate (BER) (or the block/packet error rate (BLER/PER)) for the chosen modulation and coding scheme. This error rate is then used to determine if a packet is corrupted or not.

Existing distributions of the *ns-2* software supports only limited modes of the propagation model: free space propagation, two-ray ground reflection model and the shadow model. The free space propagation model simulates the ideal condition in which there is a clear line-of-sight (LOS) path between the transmitter and the receiver. The two-ray ground reflection model considers both the LOS path and a ground reflection path. The shadow model consists of two parts: the first part uses the path loss model to predict the mean received power at distance d while the second part calculates the variation in the received power at a specific distance and it conforms to the Gaussian distribution when measured in dB.

For simulating the propagation environment in order to evaluate the performance of WiMAX systems in realistic environments, the existing propagation models in *ns-2* are far from enough. In our implementation, a Cost 231 bulk path loss component combined with Clarke-Gans implementation of Rayleigh Fading is used for simulating the physical layer. Doppler effects are included to capture mobility effects. Fast fading is included by modeling the channel as a Rayleigh fading channel with multiple taps (as described by select ITU power delay profiles). The following three basic steps are followed to generate the channel matrix:

1) Select the environment to simulate: **(a)** urban micro **(b)** suburban macro **(c)** urban macro.
2) For the chosen environment, get the parameters
3) From the parameters, generate the channel coefficients.

The roadmap for modeling the channel coefficients is shown in Figure 2 and the details of the channel modeling procedure are presented next.

1) Large Scale Fading: Large scale fading includes bulk path fading and log-normal shadowing. These can be calculated during the simulation runtime because only the distance between BS and SS, and the heights of the BS and SS antennas are the necessary parameters to calculate the path loss. Several empirical channel models are alternative options that may be applied: COST 231 Hata model, COST Walfisch Ikegami model, COST 231 Extension to Hata model, Erceg model, Stanford University Interim (SUI) models and ITU Path Loss models [9]. Among these alternative models, the COST 231 Extension to Hata model is applied in our simulations to model bulk path fading and log-normal shadowing. This model is mainly applied in the 1.5-2GHz carrier frequency, 30-300m Base Station height, 1-10m Mobile Station height and 1-20km distance between the BS and SS. The COST 231 Extension to Hata model is given by [5]

$$\begin{aligned} P_{L,urban}(d)dB &= 46.3 + 33.9\log_{10}(f_c) - 13.82\log_{10}(h_t) \\ &\quad - a(h_r) + (44.9 - 6.55\log_{10}(h_t))\log_{10}(d) \\ &\quad + C_M \end{aligned} \tag{1}$$

where $P_{L,urban}$ is the propagation loss, f_c is the carrier frequency (2GHz in this case), h_t is the height of the transmit

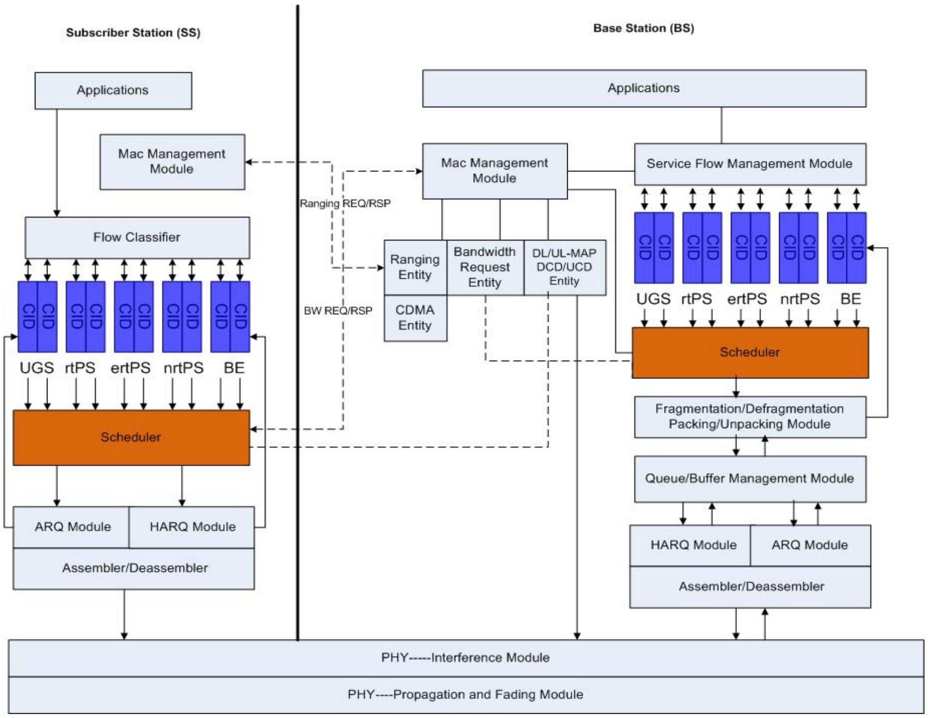

Fig. 1. WiMAX Simulation Implementation Architecture

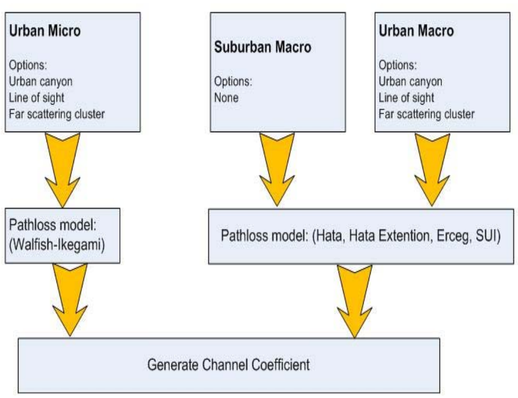

Fig. 2. Physical Channel Coefficient Modeling RoadMap

antenna (can vary from 30m to 300m), h_r is the height of the receive antenna (can vary from 1m to 10m), d is the distance between the transmitter and receiver (can vary from 1km to 20km) and C_M is 0dB for suburbs and 3dB for metropolitan areas.

2) Small Scale Fading: In order to model small scale fading, we need to take Doppler effects and fast fading into account. Since multi-path effects are very complex, statistical models are used in our simulation. To transmit information through a channel, we have convolve the signal $X[n]$ with the channel impulse response $h[n]$. But to work in time domain (the convolution of $x[n]*h[n]$) will require many more multiplication and addition operations than if one were to work in the frequency domain (multiplication of $X(f)H(f)$). Given the resources available at a typical user of the *ns-2* simulator, it is too computationally intensive to work in time domain and we thus operate in the frequency domain. The basic idea behind our time-correlated channel modeling is that we assume the channel coherence time to be 5ms. Therefore we should see a different but related channel every 5ms. To model the time correlation, we weigh the taps of the power delay profile (PDP) by time correlated Rayleigh variable, then take the fast Fourier transform (FFT) to obtain H(f).

In the actual simulation procedure, we generate a vector of Rayleigh distributed numbers (whose length is equal to the number of channel realizations required for the simulation) that are element-wise multiplied with a Doppler spectrum (using Jakes model) of the same length. The inverse FFT (IFFT) of this array is then taken to give a correlated time domain sequence of the appropriate length. This operation is repeated n times, where n is the number of taps in the ITU-PDP model (the ITU-PDP are shown in Figure 3). The n numbers of each array are put at appropriate sample points and scaled by the tap powers given by the ITU model. In our simulation, we provide 1024 channels for use. Therefore a 1024 pt FFT is taken of this n numbers to give the

Power Delay Profile		Pedestrian-A		Vehicular-A		Pedestrian-B	
Number of Paths		4		6		6	
Power of the each path (dB)	Path Delay (ns)	0	0	0	0	0	0
		-9.7	110	-1	300	-0.9	200
		-19.2	190	-9	700	-4.9	800
		-22.8	410	-10	110	-8	1200
				-15	1700	-7.8	2300
				-20	2500	-23.9	3700
Speed (km/h)		3		60		3	

Fig. 3. ITU Power Delay Profiles

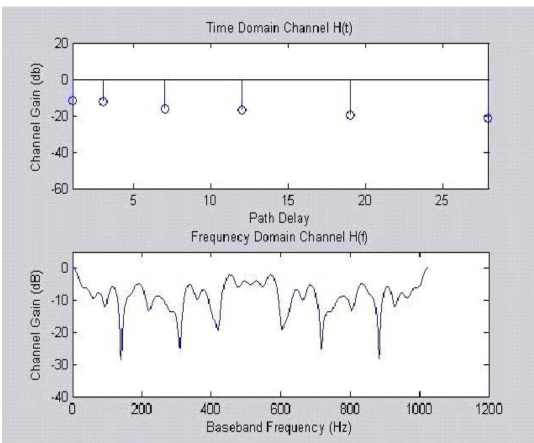

Fig. 4. Conversion from Time Domain to Frequency Domain Channel

1024 channel coefficients. An example realization of the ITU Pedestrian A model is shown in Figures 4 and 5.

In order to convert the channel impulse response from frequency domain to time domain, we apply FFT to this resulting channel coefficients matrix. A simple FFT is taken of each time domain channel vector calculated above and the result is stored as a line in a file. The same channel is shown in both the time and frequency domain in Figure 4. This FFT operation is then repeated 1024 times to get the entire channel realization. Since the time correlation is easier to visualize in time domain, Figure 5 shows three consecutive snapshots of the channel. The correlation is between the marks in the circles. For these figures, we simulated a channel with 6 taps (6 circles in the figure).

After this process, we have a channel model for the per-tap time correlation of a time varying multi-tap channel and this is all done before the simulation begins. The channel realization file is read by the simulator at the beginning of the simulation. Each node starts from a random position in the file and uses successive values in the file to model its channel state in successive frames. An example of a channel realization file is shown in Figure 6. Since the channel generation is done

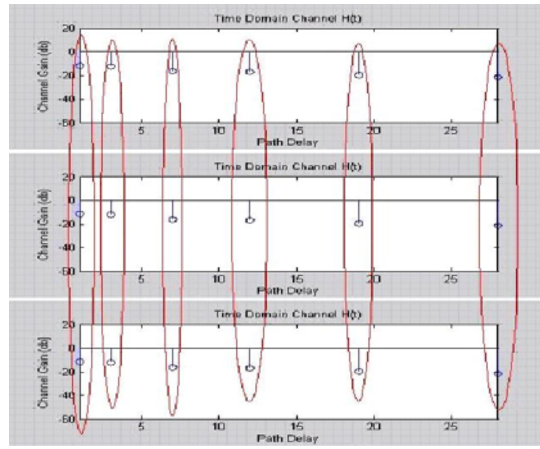

Fig. 5. Time Domain Correlation

	Subcarrier 1	Subcarrier 2	Subcarrier 3	Subcarrier 4	Subcarrier 5	Subcarrier 6			
Channel1	0.999952	0.999574	0.998603	0.997050	0.994932	0.992275	.	.	.
Channel2	0.999749	0.998962	0.996599	0.992683	0.987248	0.980347	.	.	.
Channel3	0.999638	0.998834	0.996341	0.992180	0.986389	0.979020	.	.	.
Channel4	0.997511	0.996608	0.993877	0.989342	0.983044	0.975040	.	.	.
Channel5	0.998988	0.998225	0.995546	0.990970	0.984536	0.976298	.	.	.
Channel6	0.998702	0.998099	0.995523	0.990995	0.984548	0.976235	.	.	.
Channel7	0.820728	0.820330	0.818093	0.814030	0.808172	0.800560	.	.	.
Channel8	0.935429	0.934921	0.932630	0.928573	0.922780	0.915297	.	.	.
	.	.	.	.	.	.	.		
	.	.	.	.	.	.		.	
	.	.	.	.	.	.			.

Fig. 6. Example ITU Model Based Channel Realization File

offline, huge amount of simulation time is saved.

B. Details of OFDMA Channel Implementation

To implement the channel modeled above in *ns-2*, a class called prop_OFDMA was created. It is a derived class of the base class propagation. When an OFDMA propagation model is instantiated, the selected ITU model based frequency domain channels are all read into memory from the proper file. In the first frame of the simulation, a random channel is assigned to each SS. Every time the frame timer expires (every 5ms, the coherence time of the channel), the index of the current channel that each SS is using is incremented.

When we receive a packet, we first call the bulk path loss module (COST 231) which returns the received power after path loss. We then compute the multipath fading loss by extracting the subcarrier information over which the packet was transmitted. We assume that if the total power of the packet is P, the power on each sub-carrier is P/N, where N is the total number of sub-carriers over which it was transmitted. We then scale the powers at each sub-carrier with the channel gains that we had generated off-line and do Exponential Effective SIR Mapping (EESM) to get the received signal power. If there are more nodes than available channels, multiple nodes might share the same channel. To enable the propagation model one has to configure in the `tcl` script the propagation model as OFDMA and also specify the ITU model.

C. Interference Modeling

1) Interference Abstraction: Radio interference arises when two or more sources transmit simultaneously over the same frequency band. Radio interference is almost inevitable after cell sectorization and frequency reuse, and will cause the system to be interference limited. Typically, from SS point of view, the uplink and downlink interference is frequency and time selective. It is not accurate to model the channel to be an AWGN (additive White Gaussian Noise) process with flat spectrum. In order to simulate realistic channel conditions, both slow fading and fast time-frequency selective fading are necessary components to be considered [6].

For system level simulation, challenges for the simulator comes from simulating channels that are frequency selective where signals are corrupted by interference signal and thermal noise. A good way is to quantitatively measure the reception quality by estimating the BER/BLER/PER rate for each SS. It is hard and impractical to try to simulate the instantaneous interference which is too complicated and computational intensive. We thus have the following mechanism for calculating the interference.

2) Interference Calculation Implementation: The basic idea behind our implementation of interference calculation is to use a mapping of a series of inputs from the system simulation to a performance metric which is not dependent on the channel or link.

a) SINR Consideration: Usually the performance metric is the SINR in AWGN condition and we shall refer to this performance metric as "effective SINR" or "AWGN equivalent SNR". There are many simple technologies to simulate the average performance. However they are not enough to be applied in OFDMA systems [6]. Observing the average SINR at the receiver is not an adequate way to simulate the system performance because of the following reasons:

1) FEC block bits are spread between sub-carriers
2) Due to frequency selectivity of the desired signal, each sub-carrier observes different SNR
3) Decoder behavior depends on the SINR fluctuations between FEC block bits, and not only on the average SINR
4) Bursts that observe different channel and interference characteristics will display different BER/BLER results even though they may have the same average SINR

Several methods have been proposed in literature to convert from a vector of SINR to an AWGN-equivalent SNR: Quasi-static method, Convex method, Shannon method and EESM. From these four methods, we use the EESM method in our simulation methodology.

b) SINR, EESM and BLER calculation: In order to apply SINR, EESM and calculate the BLER in our simulations, the following steps need to be applied:

1) Calculate the instantaneous SINR for each subcarriers: $SINR_i$. Typically the per subcarrier SINR is computed according to the location of the BS and SS and channel gain to calculate the signal power, as well as the radio interference.
2) Calculate the instantaneous effective SINR by using [9], [10]
$$SINR_{effective} = -\beta \ln \left(\frac{1}{N} \sum_{1}^{N} e^{-\frac{SINR_i}{\beta}} \right) \qquad (2)$$
3) Compute the BER. Based on the MCS and $SINR_{effective}$, by looking up the pre-calculated table, we can get the instantaneous BER.

Note that this methodology is based the assumption that the actual PER for the instantaneous OFDM channel state $SINR_i$ can be approximated from the BER for a basic AWGN channel with the effective SINR for the specific modulation and coding scheme (MCS).

c) Actual Implementation: In our implementation, when a packet is received, the received power on the sub-carriers it is transmitted on is computed at each node. Then the packet is recorded and a timer starts (time is the duration of the packet) and its power is added to an "Interference" array (termed Intf-1) as interference on each sub-carrier. Now when we receive the 1st bit of the packet at the MAC layer, it is marked to be either a "Signal Packet" (a packet destined for the current receiver, contributing to the signal power or S of the SINR) or an "Interference Packet" (a packet not destined for the current receiver, contributing to the interference power or the interference in SINR). If it is a "Signal Packet", the current Intf-1 array is retrieved and the power of signal packet is deducted from this array (since we had added the power of this packet to the array) and a packet timer is started that expires when the packet is received completely. This Interference array is the interference caused due to the packets currently undergoing transmissions when we receive our 1st bit and the interference power can be calculated from it. Since we have to get the signal packet power, we need to initialize another Interference array (termed Intf-2) to add the power of the packets received which are actually not destined for this receiver. When we receive our signal packet completely, we check this Intf-2 array and add it to Intf-1 array at the sub-carrier level and do an EESM to get the effective SINR value. The effective SINR lookup table is queried to get the error probability. If the error probability is higher than the configured threshold, the packet is dropped. Otherwise it is sent to upper protocol layer. Figure 7 shows how the interference power is calculated in our implementation.

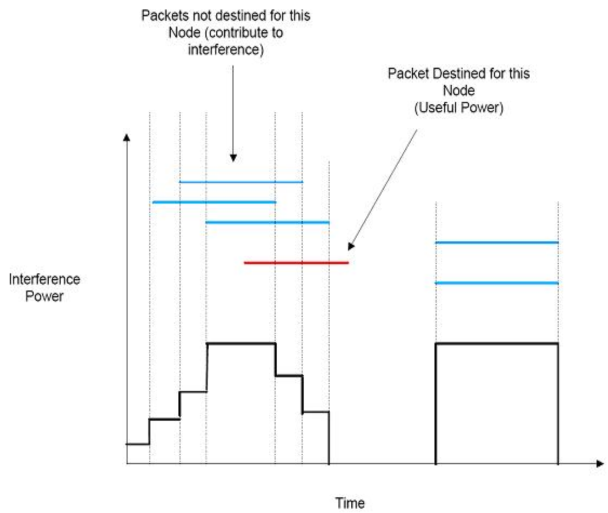

Fig. 7. Interference power calculation

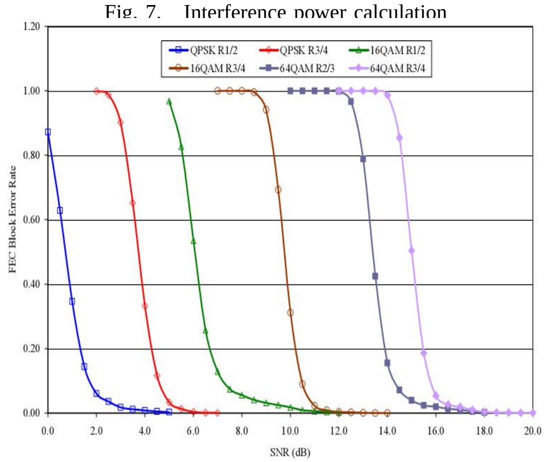

Fig. 8. SNR-BLER curve in an AWGN channel

d) Experimental Results: Figure 8 shows the AWGN BLER curve obtained by modeling the link-level performance using our system level simulator. In this simulation, over 150 blocks are transmitted are using different modulation and coding schemes. This waterfall shaped curve shows that as the SNR increases, the BER drops accordingly. At the same time, the more advanced the modulation and coding scheme, the higher the energy needed to maintain the same level of BLER.

IV. MAC Layer Implementation

In this section we describe the various MAC layer additions that were made to the *ns-2* code.

A. Scheduling

Scheduling is one of the key modules of the IEEE 802.16 MAC layer and it will, to a large extent, determine the performance of WiMAX systems. The scheduling module is

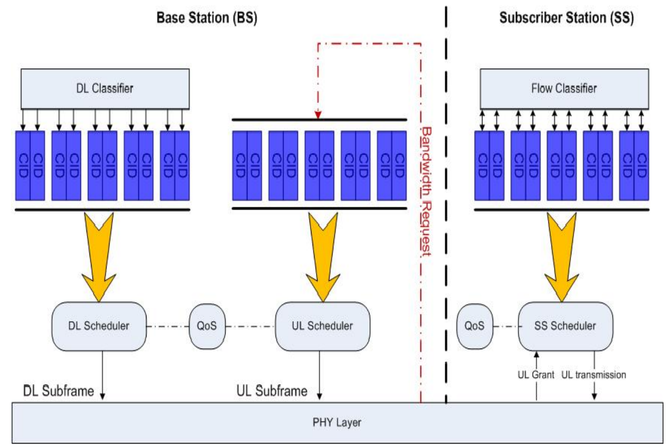

Fig. 9. Scheduling Flow Chart for BS and SS

used to handle data transport on a connection. Each connection associates with a specific scheduling service. A scheduling service is represented by a series of QoS parameters which quantify its behavior. Four connection types are supported in our implementation: UGS, rtPS, nrtPS and BE. In IEEE 802.16, the BS controls the bandwidth allocation and pinpoints the transmission time for every registered SS. Figure 9 shows the components of the scheduler for BS and SS [2][3][4].

The scheduler in the BS is in charge of all resource allocations for every associated SS. On the other end, the scheduler in SS is mainly responsible for dividing the allocated bandwidth to all its connections. The scheduler in a BS goes over the bandwidth requests from all its connections and calculates the DL-MAP for DL traffic and UL-MAP for UL traffic. Six main items are imbued into the DL-MAP and UL-MAP messages: number of OFDMA symbols, offset of OFDMA symbols, number of sub-channels, offset for the sub-channels, horizontal or vertical stripping and Interval Usage Code (IUC). In our current implementation, the minimum granularity of bandwidth allocation is a slot (1 sub-channel by 3 OFDMA symbols). This would waste some bandwidth especially when packets are small.

1) Scheduler Policy: At the BS, the Round Robin algorithm is used for allocating bandwidth in the order of basic, primary, secondary and data connections. When the BS makes the allocations for the data connections, different policies are applied to different types of services. UGS and rtPS data connections have higher priority over nrtPS and BE ones. After the bandwidth requirements for UGS connections are met, if there is any leftover bandwidth, it would be fairly allocated to all remaining rtPS, nrtPS and BE connections, in that order.

For a UGS connection, if any data is present, two policies would be used. First, the number of slots is calculated according to

$$N_{slot}^{UGS} = \frac{\lceil \frac{SDU\_Size}{UGS\_Period} \rceil}{Slot\_Capacity} \quad (3)$$

where SDU_Size is the size of the data packet, UGS_period is the periodicity with which UGS transmission from the flow are scheduled in each frame (for example, 1 means every frame) and $Slot\_Capacity$ is the transmission capacity of a slot which depends on the transmission direction and the modulation and coding scheme. For example, in UL with QPSK $1/2$ modulation scheme, $Slot\_Capacity$ is 6 bytes $(48 \times 2 \times 1/2 \times 1/8)$. Second, the scheduler will allocate bandwidth for the full SDU_{size} in every frame. For rtPS, the minimum number of slots for bandwidth allocation is:

$$N_{slot}^{rtPS} = \left\lceil \frac{Min\_Resv\_BW Frame\_Size}{Slot\_Capacity} \right\rceil \quad (4)$$

where Min_Resv_BW is the minimum reserved bandwidth for rtPS service (in Mbps) and $Frame\_size$ is the WiMAX physical frame duration time. Additional slots may be allocated to rtPS flows if available. For nrtPS and BE, the leftover bandwidth is fairly allocated to every connection even if the requested bandwidth is far more than the actual allocated one.

On the other end, the scheduler at a SS, besides splitting the allocated bandwidth to its various connections, is also in charge of sending bandwidth requests to the BS. The policy used by a SS is similar to the one applied by a BS when allocating bandwidth. UGS and rtPS do not need to send bandwidth requests but nrtPS and BE do need to. In our implementation, two mechanisms are supported to send bandwidth request: contention based and piggybacking. Without piggybacking, given the transmission opportunity, the SS scheduler calculates the necessary bandwidth to see if all the data traffic could be fit into it. If not, a bandwidth request will be issued to ask for more bandwidth. With piggybacking, a bandwidth request is simply piggybacked to the BS whenever additional bandwidth is required.

2) Bandwidth Request and Allocation: The slots allocated by a BS for Initial Ranging request and Bandwidth request are subject to collision in uplink direction [2], [3], [4]. Our implementation supports a truncated exponential backoff scheme for contention resolution. The Uplink Channel Descriptor (UCD) message is broadcasted by the BS with the contention window size inside. During the network entry procedure, a SS will perform initial ranging and adjust its transmission power. Also it will randomly pick up a backoff factor within the contention window range and generate a Ranging Request for transmitting. Then SS then decreases the counter when it finds a new contention slot. When the counter becomes zero, the SS will transmit the Ranging Request to the BS.

a) CDMA Based Contention: CDMA based bandwidth request and initial-ranging are mandatory when the OFDMA mode is used in the WiMAX physical layer. The CDMA contention structure is shown in Figure 10. CDMA_INIT_REQ and CDMA_BW_REQ are supported in our current release based on CDMA contention mechanism. Specifics for these two schemes are as follows.

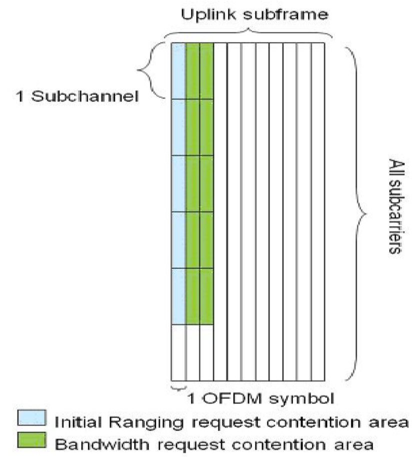

Fig. 10. CDMA Contention Frame Structure

TABLE I

CDMA SCHEME PROPERTIES

CDMA Scheme	CDMA_INIT_REQ CDMA_BW_REQ
UIUC	UIUC_INITIAL_RANGING UIUC_REQ_REGION_FULL
Tx Opportunity Size	2symbol x 6subchannels 1symbol x 6subchannels

On receiving the CDMA_INIT_REQ or CDMA_BW_REQ, the BS scheduler will first do collision detection. If there is a collision of CDMA codes and/or transmission opportunity, the BS will drop this request. Otherwise, the BS scheduler will allocate ranging request opportunity for the CDMA_INIT_REQ or CDMA_BW_REQ messages. Once a subscriber gets the opportunity, it will randomly choose a CDMA code (between 1-256) and include it in the CDMA initial ranging request. On receiving the CDMA initial ranging request or CDMA bandwidth request, the BS scheduler uses CDMA Allocation Information Element (IE) to indicate the UL transmission opportunity and CDMA code for the requested subscriber. Once the SS receives the CDMA Allocation IE, it will check if the CDMA code is the same as what it sent. If the code is the same, SS will transmit the data by using this transmission opportunity.

B. ARQ Support

The Automatic Repeat reQuest (ARQ) mechanism is a part of the MAC layer that may be enabled on a per-connection basis. The per-connection ARQ is specified and negotiated during connection creation [2], [3], [4].

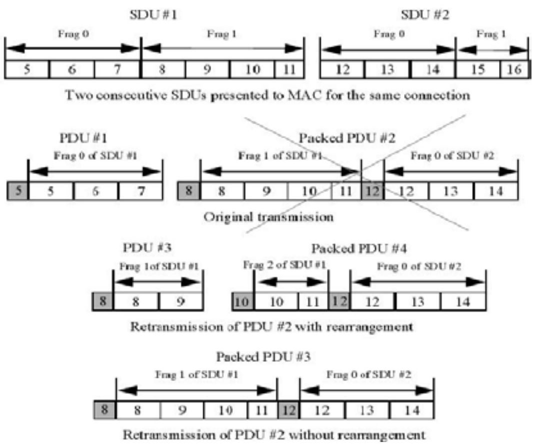

Fig. 11. Block usage examples for ARQ with and without rearrangement

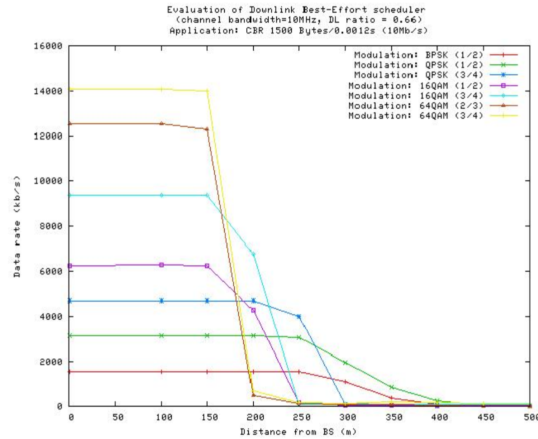

Fig. 12. Throughput as a function of distance with ARQ disabled

C. ARQ with Fragmentation and Packing

This section describes our implementation of the ARQ mechanism. When ARQ is enabled, fragmentation and packing would also be add-on parts for ARQ. Figure 11 shows the fragmentation and packing of ARQ blocks with and without rearrangement. Once fragmentation is enabled, it always occurs at the ARQ boundaries, i.e. the fragmentation block will always consists of multiple number of ARQ blocks. If packing is enabled, the packed Protocol Data Unit (PDU) may consist of ARQ blocks from different MAC Service Data Units (SDUs). On some occasions, because of re-transmission and limited transmission slot size, rearrangement of the ARQ blocks in the former packed ARQ PDU would be necessary. In our implementation, the minimum granularity of re-arrangement of packed SDU is one ARQ block in order to achieve maximum usage ratio of one slot. Our experiments show that this re-arrangement mechanism greatly enhanced the efficiency of the transmission especially when the workload of the system is high.

D. ARQ Transmitter and Receiver Logic

At the transmitter side, MAC SDUs will be divided into ARQ blocks with configurable length. After adding necessary Packing Sub-header (PSH) and/or Fragmentation Sub-header (FSH), these ARQ blocks will be put into the transmission queue and wait for scheduling and transmission. A sliding window scheme is applied in the transmission and reception logic. Once an ARQ block is acknowledged within a certain time period, it will be removed from the transmission queue. Otherwise it will be moved into the re-transmission queue. From scheduling point of view, the re-transmission queue has higher priority over the transmission queue and therefore ARQ blocks in the re-transmission queue will be scheduled earlier than those in the transmission queue. If the times of re-transmission of a certain ARQ block exceed a configured value, this ARQ block will be dropped. On the receiver side, a receiver will acknowledge successfully received ARQ block(s) by sending an ACK back. Once out-order ARQ blocks are received, the receiver will send back ACKs with the corresponding Block Sequence Number (BSN). Because of the limitation imposed by the sliding window size, the receiver will not be overwhelmed due to lack of capacity to handle the incoming traffic. If all the ARQ blocks for an original MAC SDU are present, the reassembly module of the receiver will reconstruct this MAC SDU and send it to the upper protocol layer.

E. Experimental Results

Our experimental results show that the ARQ mechanism can greatly improve the successful packet reception rate. Figure 12 shows the successful packet reception rate as the distance between the BS and the MS increases, for various modulation and coding schemes, with ARQ disabled. Figure 13 shows the same results but with ARQ enabled. We note that the addition of an ARQ mechanism significantly increases the achieved data rates.

As the SS moves further away from the BS, because of path loss, fading, Doppler effect etc., the rate of successful reception of data traffic will drop dramatically if ARQ is not enabled. But with ARQ enabled, the corrupted data traffic will be re-transmitted. Figure 13 shows that the rate does not drop that dramatically. Instead it stays flat over a large range of distance with different modulation and coding schemes.

F. Cost of ARQ

Although ARQ can improve the successful reception rate of data traffic, it is achieved at the cost of re-transmission time and extra bandwidth. At the same time, the fragmentation, packing, reassembly and acknowledgment processes would impose an extra cost or burden on the system resources of both the BS and SS. For SS, because of its limited battery capacity, the extra system resource requirement and burden brought by ARQ would shorten the lifetime of the SS.

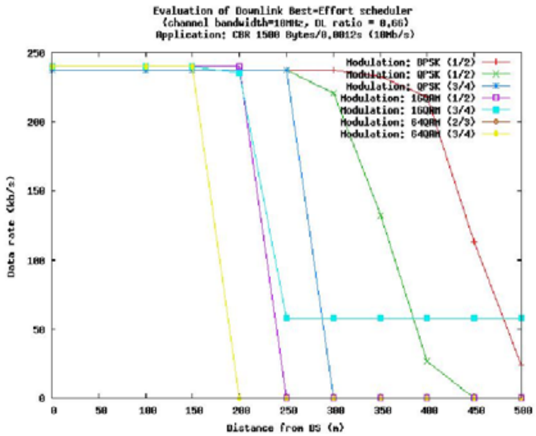

Fig. 13. Throughput as a function of distance with ARQ enabled

Fig. 14. Simulation time as a function of the number of nodes in the simulation with ARQ disabled.

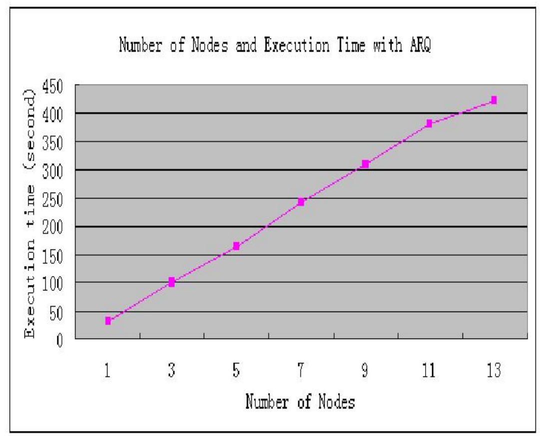

Fig. 15. Simulation time as a function of the number of nodes in the simulation with ARQ enabled.

G. Miscellaneous features

Besides the features described above, several other features are also supported in our simulation. These include: Network entry, Service Flow and QoS related technologies, scanning and Handover, configurable modulation, Time Division Duplexing etc. [2], [3], [4].

V. Simulator Performance

In this section we evaluate the time it takes to run WiMAX system level simulations using our *ns-2* based simulation software. The results reported were for simulations conducted on a machine with a Intel Xeon CPU with a 3.00 GHz processor running Red Hat Enterprise Linux AS Release 4. The machine had a memory of 4GB and the front side bus operated at 1066 MHz.

In the simulated scenario, each node has one BE flow generating data at a rate of 240kbps and the simulated time is 10 seconds. For the case where ARQ is disabled, Figure 14 shows the simulation execution time as a function of the number of nodes in the network. In our simulations, the execution time increases super-linearly with the number of nodes.

Next we consider the simulator's operation when the ARQ mechanism is enabled. Figure 15 shows the simulation execution time as a function of the number of nodes in the network for this case. For these results we used 512 byte ARQ blocks, 512 bytes ARQ receiving window, 10% data loss rate and ARQ retransmission time of 0.005 seconds. We again observe that the execution time increases super-linearly with the number of nodes in the simulation. We also note that with ARQ enabled, each MAC SDU will be divided into ARQ blocks. The smaller the size of the ARQ block, the longer the simulation time will be.

VI. Conclusion and Discussion

In this paper we presented the design methodology and implementational details for system level simulations of WiMAX systems. An overview of WiMAX features that we have integrated in the network simulator *ns-2* was provided. Our implementation can not only address the issue of accurate link level simulations but also simulate the WiMAX system level performance. Flexible configuration design makes the WiMAX simulation environment more controllable and the results more accurate. Configurable DL-UL ratio, frequency bandwidth, rtg, ttg makes the physical layer properties more realistic. Series of timers and corresponding management messages organically assemble the WiMAX simulation processes. Reasonable buffer and queue management schemes accurately simulate both low and heavy load conditions in WiMAX system.

We are currently working on incorporating more advanced features like Multiple Input Multiple Output (MIMO), sectorization, Adaptive Modulation and Coding (AMC) schemes and Hybrid Automatic Repeat (HARQ) mechanisms. MIMO with Space-Time Coding (STC) implementation can help us quantitatively characterize the improvement that MIMO technology may bring to WiMAX system performance. HARQ and AMC with Channel Quality Indication Channel (CQICH) mechanism facilitate additional degrees of control mechanism to enhance the system performance. Sectorization can measure the mobility related properties of mobile WiMAX systems.

In addition to the above features which are defined in the IEEE 802.16 standard, other useful technologies exist that are not included in the standard but could be valuable supplements to WiMAX system. These include:

1) *Power Balancing Mechanism:* A BS could provide transmission power indication to each SS in order to minimize the near-far effect.
2) *Enhanced Cell Reselection:* This mechanism could improve the interoperability between WiMAX and other wireless communication system like 2G, 3G networks etc.
3) *Radio Admission Control:* It can help a BS to know the current cell load and make reasonable decisions when new SS(s) want to register into it.
4) *Traffic Management:* It can help a BS to quantitatively know its CPU, buffer and queue consumption. Based on this information, a BS can send increase or decrease traffic indications to some SS(s) so that the BS will not be overwhelmed or waste bandwidth.
5) *Radio Overload Protection:*. It can help a BS to control its transmission power within a reasonable range so that its transmission electronic components would not be damaged.

Inclusion of these features in the simulator would allow for further investigation of the performance of WiMAX systems.

REFERENCES

[1] ns Network Simulator. http://www.isi.edu/nsnam/ns/.

[2] IEEE 802.16 Working Group. 802.16-2004, ”802.16 IEEE Standard for local and metropolitan area networks, Part16: Air Interface for Fixed Broadband Wireless Access Systems”. October 2004.

[3] IEEE 802.16 Working Group. IEEE Std.802.16e ”802.16 IEEE Standard for local and metropolitan area networks, Part16: Air Interface for Fixed Broadband Wireless Access SystemsAmendment2: Physical and Medium Access Control Layers for Combined Fixed and Mobile Operation in Licensed Bands, December 2005.

[4] IEEE 802.16 Working Group. 802.16Rev2/D3-2008, ”802.16 IEEE Standard for local and metropolitan area networks, Part16: Air Interface for Fixed Broadband Wireless Access Systems”, February 2008.

[5] WiMAX Forum, ”WiMAX Network Simulator: The Network Simulator NS-2 MAC+PHY Add-On for WiMAX (IEEE 802.16)”, July 2008.

[6] WiMAX Forum, ”WiMAX System Evaluation Methodology”, August 2007.

[7] M.Yuksel, B.Sikdar, K.S.Vastola and B.Szymanski. Workload Generation for ns simulations of wide area networks and the Internet. Proc. of Communication Networks and Distributed Systems Modeling and Simulation Conference, pp 93-98, 2000.

[8] A.Bestetti, G. Giambene, S. Hadzic. ”WiMAX: MAC Layer Performance Assessments”, Wireless Pervasive Computing, 2008. ISWPC 2008. 3rd International Symposium on, 490-494, July 2008.

[9] John G. Proakis (1995). Digital Communications, 3rd edition, Singapore: McGraw-Hill Book Co, 767-768.

[10] Bernard Sklar, ”Rayleigh Fading Channels in Mobile Digital Communication Systems Part I: Characterization”. IEEE Communications Magazine 35 (7): 90-100, July 1997.

[11] Raj Jain, ”WiMAX System Level Modeling Methodology: A Tutorial”. http://www.cs.wustl.edu/ jain/wimax/, October 2007.

[12] FrankChee-DaTsai , Jenhui Chen , Chiang-Wei Chang, Wei-Jen Lien, Chih-Hsin Hung, and Jui-Hsiang Sum, ”The design and implementation of WiMAX module for ns-2 simulator”, ACM International Conference Proceeding Series; Vol. 202, 2006.