WiMAX (IEEE802.16e) System Design and Evaluation using Network Simulator ns2

by

Xingting Guo

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Biplab Sikdar, Thesis Advisor

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ABSTRACT

Nowadays great efforts are made on simulating wireless network technologies, including IEEE802.11, Mobile Ad hoc networks (MANETs) and sensor networks. However few works have been done to extend the simulation to IEEE802.16e (Mobile Broadband Wireless Access Networks) or WiMAX. WiMAX has gradually become the focus not only of the academia but also of the industry.

In this thesis, we design and implement the Media Access Control (MAC) layer and Physical layer of WiMAX based on IEEE802.16e standard. Appropriate abstractions and methodologies are carefully chosen to ensure the maximal accuracy of the simulation to reality. We use ns as the simulation platform for this thesis largely because of its wide acceptance in the network simulation community and its open design policy which is suitable for modification. In addition, we evaluate the WiMAX system performance and discuss some other valuable technologies which are not included in IEEE802.16e standard.

1. Introduction

With gradual increasing concern of Broadband Wireless Access (BWA), more and more research and development have been put to BWA related technologies. 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 802.16-2004 standard, broadband means "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 Multi-channel Multipoint Distribution Service (MMDS) were applied by using Base Station (BS) to provide broadband wireless access service to discrete Subscriber Stations (SS). Now Fixed and Mobile BWA technology both have been put into IEEE802.16e perspective.

To provide BWA service, IEEE802.16 specification applies a wide range of RF spectrum and WiMAX (worldwide interoperability for microwave access) could function on any frequency below 66GHz. In fact, there is no uniform global licensed spectrum for WiMAX, although 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 or variable channel bandwidth among 1-30MHz in 2-11GHz. For TDD, WiMAX profiles define channel size 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.

In our simulations, we use ns as the simulation platform. Ns is a discrete event network simulator. It is very popular among academia and industry for its extensibility and plentiful online documentation. Ns is an open source software and sponsored by DARPA. Because of its open source policy, it provides a rich library of network topology and traffic generators which are easy to modify. Flexible topology arrangement and proper composition of the composite traffic of ns simulator can fairly accurately simulate the performance of the system. [1, 7]

In this thesis we organize the structure as follows. In Section 2, we overview the simulation system architecture. In Section 3 we present the details of abstraction and modeling of WiMAX physical layer. In Section 4 we mainly focus on design and implementation of key features of WiMAX MAC layer and some other features are briefly described. In section 5, we will discuss in detail our design and implementation of the algorithms for several technologies in Mobile WiMAX, like Adaptive Modulation and Coding, Handover, Automatic Repeat request (ARQ) etc. In section 6, some of the system level simulation experiment results will be presented. In section 7, some conclusions and future works are addressed. Some potential valuable technologies which are not parts of standard are also briefly analyzed.

2. Overview of the IEEE802.16 Architecture

This section mainly presents the architecture of our implementation of WiMAX system. [2, 3, 4, 8]

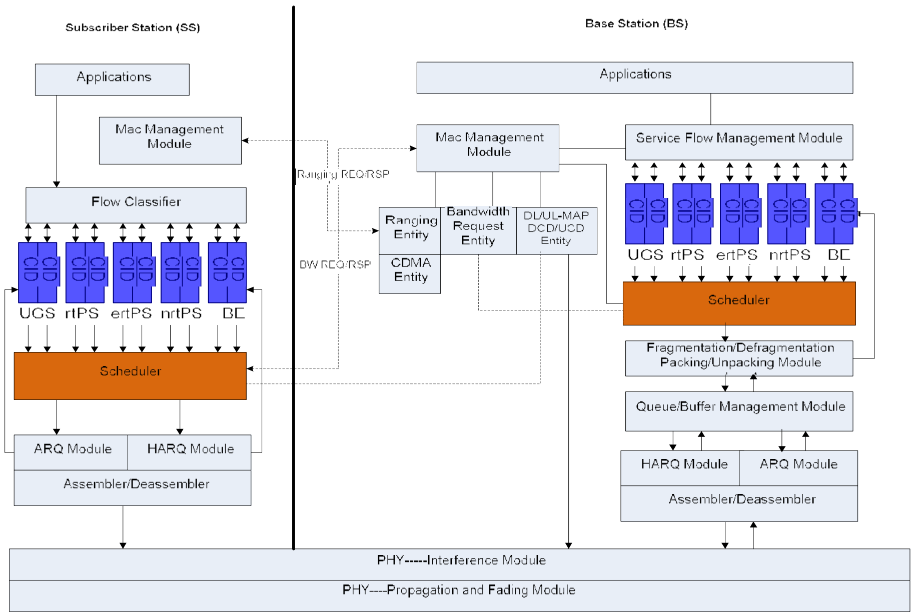


Figure 1: WiMAX Simulation System Architecture

In Figure 1, the service specific CS (Convergence Sublayer) resides on top of the MAC CPS (Common Part Sublayer), via MAC SAP (Service Access Point), service provided by MAC CPS. MAC Management module is mainly used to control the management messages and algorithms behinds them, like DL-MAP, UL-MAP, Initial Ranging, Registration etc. Connection Classifier module is used to manage service flow and QoS related issues. Five types of connections and their buffers are needed ith corresponding algorithms. Scheduler module specializes to schedule the downlink and uplink direction traffic, and handle bandwidth allocation related processes. CDMA code module focus on the CDMA (Code Division Multiple Access) ranging related issues. Fragmentation and Packing is supported in our implementation. Physical Layer abstraction and methodology are deployed and will be described in Section 3 in details.

3. Physical Layer Implementation

3.1 Propagation and Fading Modeling

Radio propagation model is used to predict the received signal power of each packet. By using radio propagation model, ns2 can simulate the effects caused by fading, shadowing, reflection etc. After calculating the received power of each packet and comparing with the threshold, packets will be regarded as corrupted and dropped, or successfully received and sent to MAC layer. Till now in ns2, limited modes of propagation model are supported: free space model, two-ray ground reflection model and shadow model. Free space model simulates the ideal simulation condition in which there is a clear line-of-sight path between transmitter and receiver. 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 path loss model to predict the mean received power at distance d. The second part calculates the variation of received power at a specific distance and it conforms to Gaussian distribution when measured in dB.

Typically there are three basic steps to describe the overall procedure for generating the channel matrix.

1) Describe the environment to simulate where this simulation is **a)** urban micro **b)** suburban macro **c)** urban macro.
2) From the chosen environment, get the parameters
3) From the parameters, generate the channel coefficients.

Roadmap for modeling the channel coefficients are is shown in Figure 2:

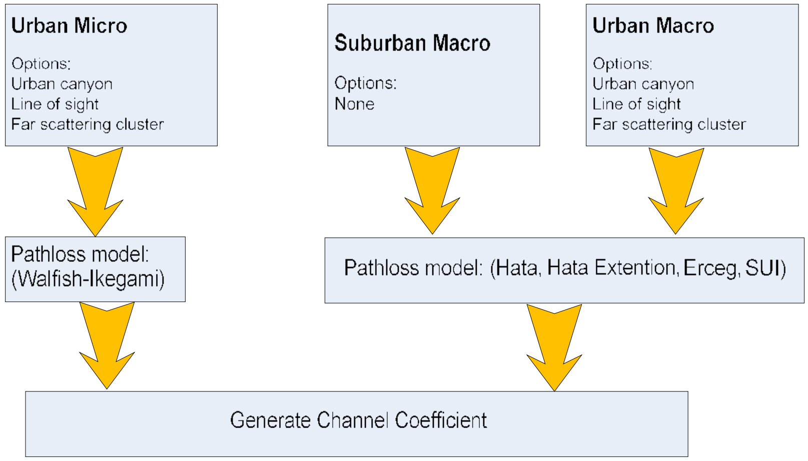


Figure 2: Roadmap of Generating Channel Coefficient

To simulate the propagation which is needed by WiMAX technology, the propagation models provided by ns2 are far from enough. In our implementation, a Cost231 bulk path loss component combined with Clarke-Gans implementation of Rayleigh Fading is used. 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).

2.1.1 Large Scale Fading

Basically large scale fading includes Bulk Path Fading and Log Normal Shadowing. They will be calculated during the simulation because the distance between BS and SS, and at the same time the height of the BS and SS are necessary parameters to calculate the path loss. Several empirical channel models are alternative options to be applied: COST231-Hata-Model, COST-Walfisch-Ikegami-Model, COST 231 Extension to Hata Model, Erceg Model, Stanford University Interim (SUI) Models and ITU Path Loss Models. [12] Among these alternative models, COST231 Extension to Hata model is applied in our simulation to model Bulk Path Fading and Log Normal Shadowing. This model is mainly applied in 1.5-2GHz carrier frequency, 30-300m Base Station height, 1-10m

Mobile Station height and 1-20km distance condition. The formula below is the COST231 Extension to Hata model: [5]

$$P_{L,urban}(d)dB = 46.3 + 33.9\log_{10}(f_c) - 13.82\log_{10}(h_t) - a(h_r) + (44.9 - 6.55\log_{10}(h_t))\log_{10}(d) + C_M$$

(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 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)

C_M is 0dB for suburbs and 3dB for metropolitan areas

3.1.1 Small Scale Fading

In order to model Small scale fading, we need to take Doppler Effects and Fast Fading into account.

3.1.1.1 Doppler Effects and Fast Fading Modeling

In fact, multi-path effects are very complex, so statistical models are used. To transmit information through a channel, we have convolve 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 multiplication and addition operations than in frequency domain (multiplication of X(f) and H(f)). The basic idea for time correlation 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 would weight the taps of the PDP (Power Delay Profile) by time correlated Rayleigh numbers, then take the FFT to obtain H(f).

In real modeling procedure, we generate a vector of Rayleigh numbers (whose length is the number of channel realization) and element -wise multiply it with a Doppler spectrum (using Jakes model) of the same length. The IFFT (Inverse Fast Fourier Transform) of this array is 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 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 points FFT is taken of thess n numbers to give the 1024 channel coefficients. An example ITU model is shown in Table1.

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	

Table 1: ITU Sample

In order to convert the channel impulse response from frequency domain to time domain, we apply FFT to this resulting channel coefficients matrix. We then take a simple FFT of one time domain channel vector calculated above and store the result as a line in a file. The same channel is shown in both the time and frequency domain in Figure 3. [5]

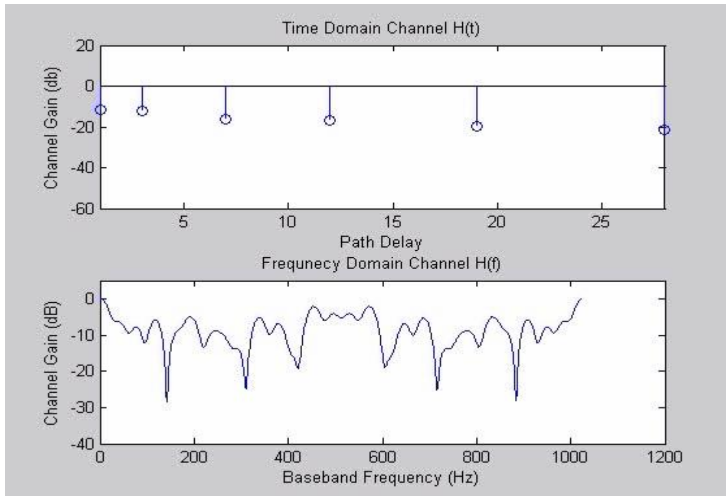


Figure 3: Conversion from Time to Frequency Domain Channel

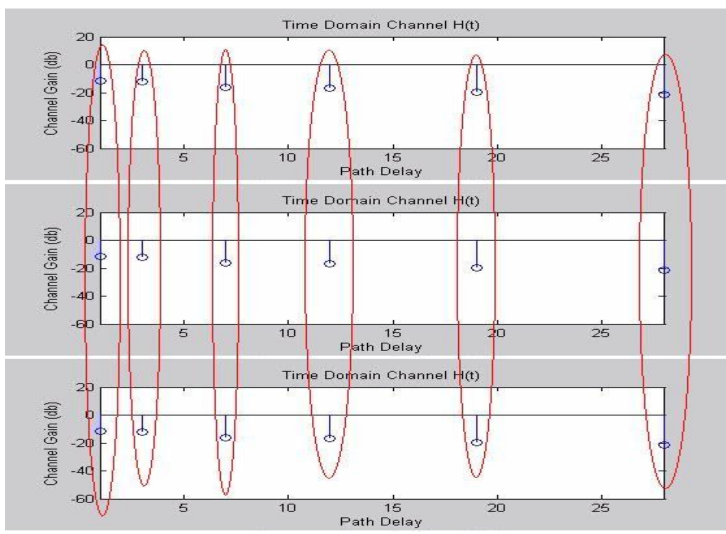


Figure 4: Time Domain Correlation

We then repeat 1024 times the FFT operation to the time domain channel matrix and store each result as one line in a file. Since the time correlation is more clearly visualized in time domain, Figure 4 shows three consecutive snapshots of the channel. The correlation is between the marks in the above 6 circles. In our modeling, we simulate channel with 6 taps

(6 circles in the figure). After all these processes, we have the channel model for the per-tap time correlation of a time varying multi-tap and this is all done before the simulation. An example of a channel realization file is shown in Figure 5

	Subcarrier 1	Subcarrier 2	Subcarrier 3	Subcarrier 4	Subcarrier 5	Subcarrier 6	
Channel 1	0.999952	0.999574	0.998603	0.997050	0.994932	0.992275	• • •
Channel 2	0.999749	0.998962	0.996599	0.992683	0.987248	0.980347	• • •
Channel 3	0.999638	0.998834	0.996341	0.992180	0.986389	0.979020	• • •
Channel 4	0.997511	0.996608	0.993877	0.989342	0.983044	0.975040	• • •
Channel 5	0.998988	0.998225	0.995546	0.990970	0.984536	0.976298	• • •
Channel 6	0.998702	0.998099	0.995523	0.990995	0.984548	0.976235	• • •
Channel 7	0.820728	0.820330	0.818093	0.814030	0.808172	0.800560	• • •
Channel 8	0.935429	0.934921	0.932630	0.928573	0.922780	0.915297	• • •
	•	•	•	•	•	•	•
	•	•	•	•	•	•	•
	•	•	•	•	•	•	•

Figure 5: Example ITU Model Based Channel Realization File

Because we do the above processes offline, huge amount of simulation time can be saved. This is useful because in ns2 we are interested in the receiving power on each sub-carrier.

3.2 Details of OFDMA channel implementation in NS2

To implement the channel modeled above in NS2, 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 MS. Every time the frame timer expires (every 5ms, the coherence time of the channel), the index of the current channel that each MSS is using is incremented.

When we receive a packet, we first call the bulk path loss module (COST231) which returns the received power after bulks path loss. We then compute the multipath fading loss. We extract 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 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 the propagation model as OFDMA and also specify the ITU model. [5]

3.3 Interference Modeling

3.3.1 Interference Abstraction

Basically radio interference comes from same frequency and idiosyncratic frequency source. Especially after cell sectorization and frequency reuse, radio interference is almost inevitable 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 the realistic channel condition, both slow fading and fast time-frequency selective fading are necessary components to be considered. [6]

For system level simulation, challenges for 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 (Bit Error Rate/Block Error Rate/Packet Error Rate) rate for each SS. It is hard and impractical to try to simulate the instantaneous interference which is too complicated and computational intensive.

3.3.2 Interference Implementation

An idea of interference implementation is to map a series of inputs from the system simulation to a performance matrix which is not dependent with the channel or link.

3.3.2.1 SINR consideration

Usually the performance matric is the SINR (signal-to-interference-plus-noise-ratio) in AWGN condition and we shall refer to this performance matric by name “effective SINR” or “AWGN equivalent SNR”. There are many simple technologies to simulate the average performance. But they are not enough to be applied in OFDMA system s.[6] Observing the average SINR at the receiver is not an adequate way to simulate the system performance because of the following reasons:

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

Several methods are used to convert from a vector of SINR to an AWGN -equivalent SNR: Quasi-static method, Convex method, Shannon method and EESM (Exponential Effective SIR Mapping). From these four methods, we select EESM to do mapping work.

3.3.2.2 SINR, EESM and BLER calculation

In order to apply SINR, EESM and calculate the BLER in our simulation, 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 the below formula: [6, 9, 10, 11]

$$SINR_{effective} = -\beta \times \ln\left(\frac{1}{N}\sum_{1}^{N} e^{-\frac{SINR_i}{\beta}}\right) \quad \textbf{(2)}$$

where:

N is number of subcarriers

β is an adjustment factor that depends on the FEC (Forward Error Correction) type.

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 on 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 MCS.

3.3.2.3 Actually Implementation

In our implementation, when a packet is received, the received power on the sub-carriers it comes is computed. Then the packet is recorded and a timer starts (time is the duration of the packet) and its power is added to an Intf-1 array as interference on each sub-carrier. Now when we receive the 1st bit of the packet at the MAC, it is decided to be either an "Signal Packet" (a packet dedicated to the current receiver, contributing to the signal power or S of the SINR) or an "Interference Packet" (a packet not dedicated to the current receiver, contributing to the interference power or the interference in SIR). If it is "Signal Packet", the current Intf-1 array is obtained. The power of Signal packet is deducted from

Intf-1 (since we had added the power of this packet to Intf-1) and a packet timer is started to receive the packet completely. This Intf-1 array is the interference caused due to the packets currently undergoing transmissions when we receive our 1 st bit. Then we calculate the interference. Since we have get the signal packet power, we need to initialize another Intf-2 array to add the power of the packets received which are actually not dedicated to 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 looking up table is queried to get the error probability. If error probability is higher than the configured threshold, it is dropped. Otherwise it is sent to upper layer protocol layer. The Figure 6 shows the rational of our implementation.

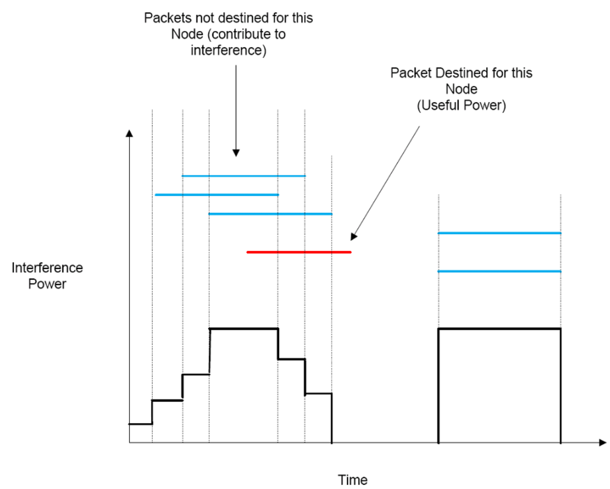


Figure 6: Snapshot of Interference

3.3.2.4 Experimental Result

Figure 7 shows the AWGN BLER curve obtained by modeling link-level performance by using a system level simulation. Within this simulation, over 150 blocks are applied using different modulation and coding schemes. This waterfall shaped curve shows that with the increased in SNR, the BER drops accordingly. At the same time, the more advanced the modulation coding scheme, the more the energy needed to keep the same level of BLER. [6]

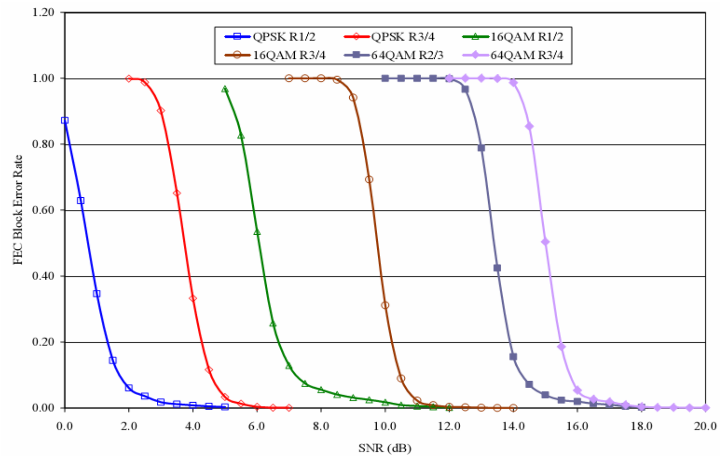


Figure 7: SNR-BLER Curve in an AWGN channel

4. MAC Layer Supplementation

4.1 Scheduling

Scheduling is one of key modules of the MAC layer and it will, to a large extent, determine the system performance of WiMAX system, BS and SS. Scheduling module is used to handle data transport on a connection. Each connection associates with a specific scheduling service. Scheduling service is represented by a series of QoS parameters which quantify its behavior. Four kinds of scheduling types are supported in our implementation: UGS (Unsolicited Grant Service), rtPS (real-time Polling Service, nrtPS (non real-time Polling Service) and BE (Best Effort). [2, 3, 4] In IEEE802.16, the BS controls the bandwidth and pinpoints transmission time for every registered SS. Figure 8 shows the components of the scheduler for BS and SS.

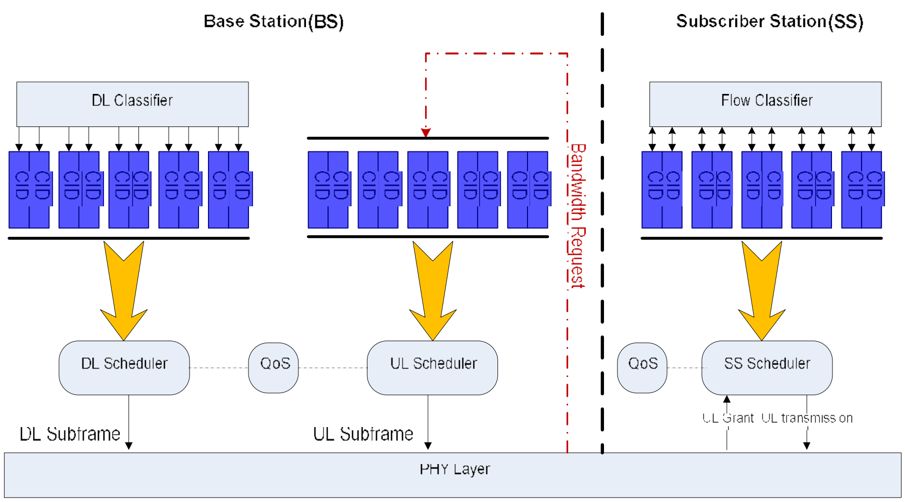


Figure 8: Scheduling Flow Chart for BS and SS

The scheduler in BS is in charge of all resource allocation 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 the BS goes over all the bandwidth requests from all the 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 of sub-channels, horizontal or vertical stripping and Interval usage code (IUC). In our current implementation, the minimum granularity of bandwidth allocation is slot (1 sub-channel by 3 OFDMA symbols). It would waste some bandwidth especially when packets are small.

4.1.1 Scheduler Policy

For BS, Round Robin algorithm is used for allocating bandwidth in the order of basic, primary, secondary and data connections. For allocation toward data connections, different policies will be applied for different type of services. UGS and rtPS type of data connections have higher priority over nrtPS and BE ones. After meeting the bandwidth requirement for UGS connections, if there is any leftover bandwidth, they would be fairly allocated to all remaining rtPS, nrtPS and BE connections.

For UGS, if any data is present, two policies would be used. First,

$$N_{slot} = \frac{\left\lceil \dfrac{SDU\_size}{UGS\_period} \right\rceil}{Slot\_Capacity} \qquad \textbf{(3)}$$

where

SDU_size is the size of the data packet. Usually we set it to 700 bytes.

UGS_period is the period of the UGS transmission. For example, 1 means every one frame.

Slot_Capacity is the transmission capacity of a slot. It relates to the modulation and coding scheme. For example, in UL with QPSK 1/2 modulation scheme, the Slot_Capacity will be 8 bytes (48*2*1/2*1/8).

Second, the scheduler will allocate full SDU_size in every frame.

For rtPS, the policy for bandwidth allocation is:

$$N_{slot} = ceil\{\frac{(Minimum\_reserved\_bandwith(M_{bps}) \times Frame\_size)}{Slot\_Capacity}\} \quad \textbf{(4)}$$

where:

$Minimum\_reserved\_bandwith(M_{bps})$ is the minimum reserved bandwidth for rtPS service. One can set it to zero for simplicity.

$Frame\_size$ is the WiMAX physical frame duration time. In our simulation, w e set it to 5ms, i.e. 200 frame per second.

Slot_Capacity is the same as the one in Equation 3.

For nrtPS and BE, leftover bandwidth will fairly be allocated to every connection even if the requested bandwidth is far more than the actually allocated o ne.

On the other end, in SS, besides splitting the allocated bandwidth to its variable connections, the scheduler of SS is also in charge of sending bandwidth requests to BS. The policy used by SS is similar to the one applied by BS when allocating bandwidth. UGS and rtPS do not need to send bandwidth request s. But nrtPS and BE do need to. In our implementation, two mechanism s are supported to send bandwidth request : bandwidth and piggyback.

Without piggyback, given the transmission opportunity, S S 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 because the bandwidth request is aggregation for now. With piggyback, when it comes the same situation, a bandwidth request will be piggybacked to BS.

4.1.2 Bandwidth Request and Allocation

4.1.2.1 Contention Resolution

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

4.1.2.2 CDMA Based Contention

CDMA based bandwidth request and Initial-ranging are mandatory when OFDMA mode is built in WiMAX physical layer. [2, 3, 4]

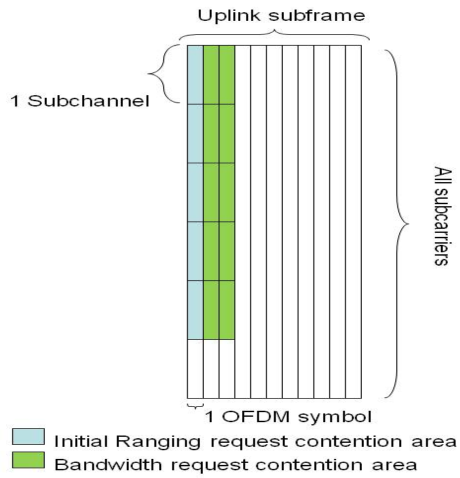


Figure 9: CDMA Contention Frame Structure

CDMA contention structure is shown in Figure 9. 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:

CDMA schemes	UIUC	Tx Opportunity Size
CDMA_INIT_REQ	UIUC_INITIAL_RANGING	2symbol×6 subchannels
CDMA_BW_REQ	UIUC_REQ_REGION_FULL	1symbol×6 subchannels

Table 2: CDMA Scheme Properties

On receiving the CDMA_INIT_REQ or CDMA_BW_REQ, BS scheduler will firstly do the collision detection. If there is a collision of CDMA code and/or transmission opportunity, BS will drop this request. Otherwise, BS scheduler will allocate ranging equest opportunity for CDMA_INIT_REQ or CDMA_BW_REQ. Once the subscriber gets the opportunity, it will randomly choose a CDMA code(1~256) and include it into the CDMA initial ranging request. On receiving CDMA initial ranging request or CDMA bandwidth request, BS scheduler uses CDMA Allocation IE to indicate the UL transmission opportunity and CDMA code for the requesting subscriber. Once 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.

4.1.3 Experiment of Results

In our experiment, Best Efforts (BE) type of QoS is chosen to show the scheduling system performance.

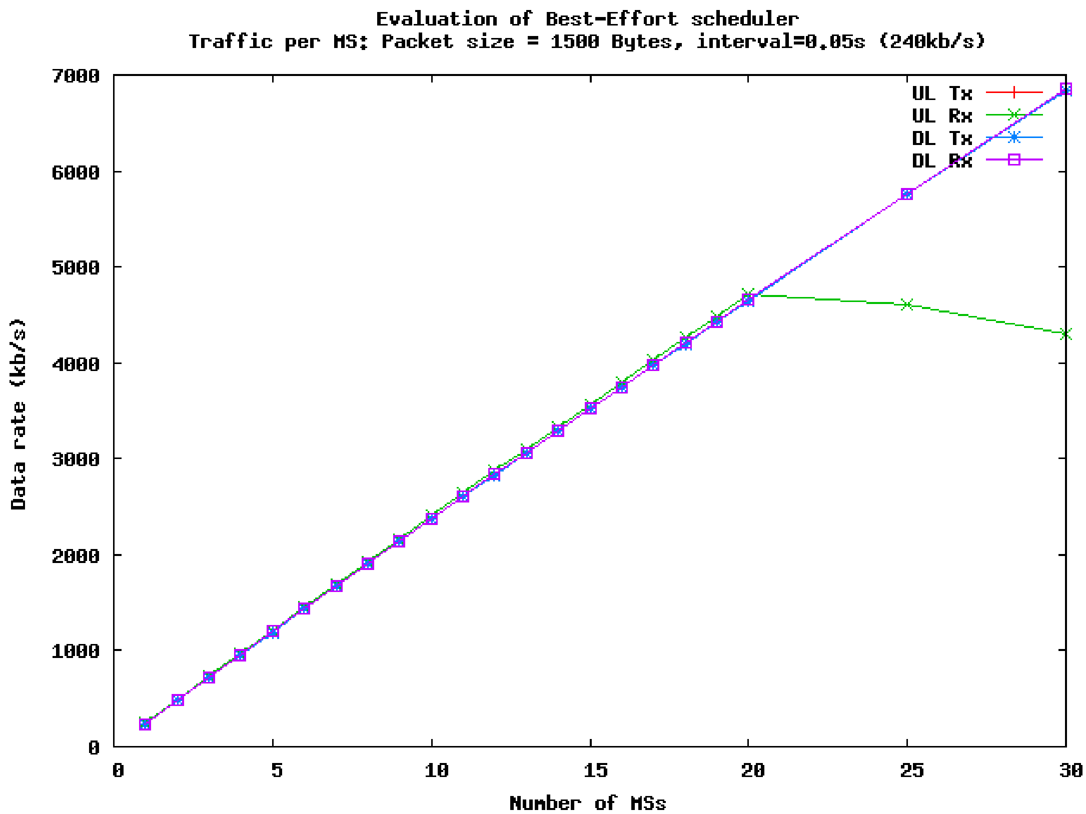


Figure 10: Evaluation of Best Effort Scheduler -Throughput

The results shown in Figure 10 show the data rate for MS as the number of MSs increases. When the number of MS increases, the system throughput increases linearly. When the number of MSs increases above 20, the UL Rx (Receiver) scheduling keeps the data rate at a relative constant value, since the system performance in UL Rx is saturated with the increasing number of MSs.

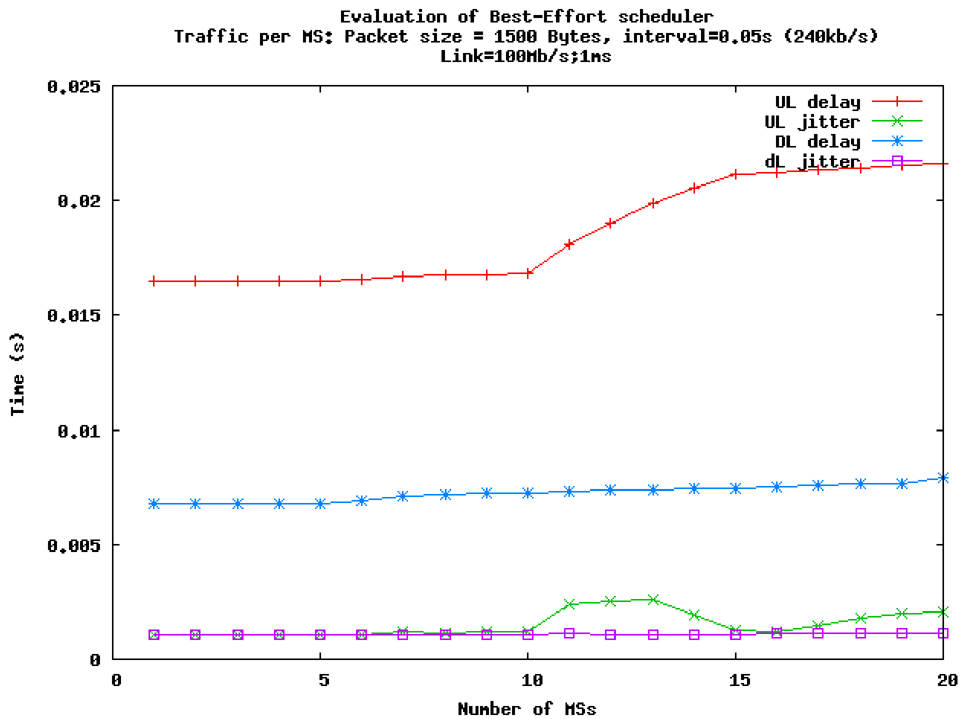


Figure 11: Evaluation of Best Effort Scheduler-Delay

Figure 11 shows the delay and jitter in uplink and downlink with the increasing number of MSs. The uplink direct traffic has higher delay than the downlink direction, because the BS scheduler can allocate transmission resource in current downlink subframe and transmit the traffic immediately. Therefore the delay in downlink is relative low. In uplink direction, the MS has to send bandwidth request to BS and asks for certain amount of transmission resource. In the next frame, BS might allocate the transmission resource. As a result, the delay in uplink is relative high. Our simulation proves this idea.

4.2 Fast Feedback Channel

4.2.1 General Introduction

The fast feedback channel is used by the MS to provide the corresponding information which are demanded by the BS. The scheduler module of BS could use these feedback information to provide better service. Normally fast feedback channel is also called CQI (Channel Quality Indiation) channel. The CQI channels are used to transmit relevant channel-state information including: Physical CINR, effective CINR, MIMO mode selection and frequency selective sub-channel selection. With TDD implementations, link adaptation can also take advantage of channel reciprocity to provide a more accurate measurement of channel condition. HARQ combined together with CQICH and MCS can provide robust link adaptation in mobile environments at vehicle speed within 120km/h.

4.2.2 System Overview

Fast Feedback channel mechanism can be widely used in WiMAX systems. Adaptive Modulation and Coding (AMC) mechanism could be implemented based on the Fast Feedback channel. The Base Station could send request to its Subscriber Stations and ask them to send up their measurement of CINR/SNR/RSSI of DL channel. According to this information the BS could relative accurately and runtime know the radio quality of the connection with each SS and therefore use appropriate and suitable modulation and channel coding methods to achieve the maximum throughput and most robust adaptive DL channel. The BS also could apply Hybrid Automatic Retransmission reQuest (HARQ) based on Fast Feedback channel mechanism. As for the Downlink traffic, SSs need to send ACK/NACK for each HARQ block toward the BS by using the Fast Feedback channel. By this way, the reception of each HARQ block will be confirmed. Multi Input and Multi

Output (MIMO) technique could also be constructed on the Fast Feedback channel mechanism. By using the Fast Feedback channel, the S Ss could send the selected MIMO mode type to the BS. There are some other techniques which could be implemented over the Fast Feedback channel. For example, the MSs could send Indication Flag Feedback via Fast Feedback channel to transmit a Feedback header or Bandwidth Request header without the need to perform bandwidth request ranging. By this way, MSs could expedite the bandwidth request procedure.

Fast Feedback channel mechanism tightly relates to the other techniques in WiMAX system. The Figure 12 shows, on some extents, the relationship between the Fast Feedback channel and other techniques.

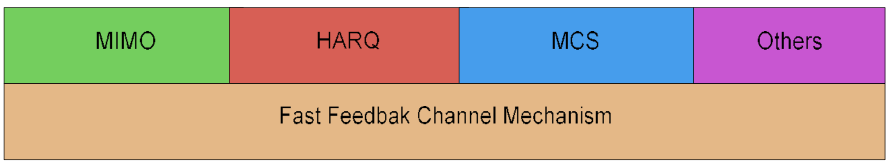


Figure 12: Fast Feedback Relationship with other s

4.2.3 Class Description and Design

4.2.3.1 Module overview

The Figure 13 shows the relationship between the Fast Feedback mechanism and the others from UML (Unified Modeling Language) point of view.

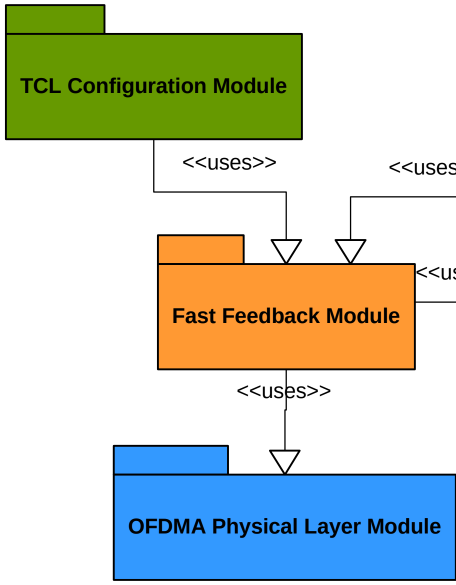


Figure 13: Fast Feedback Module Relationship

4.2.3.2 Class overview

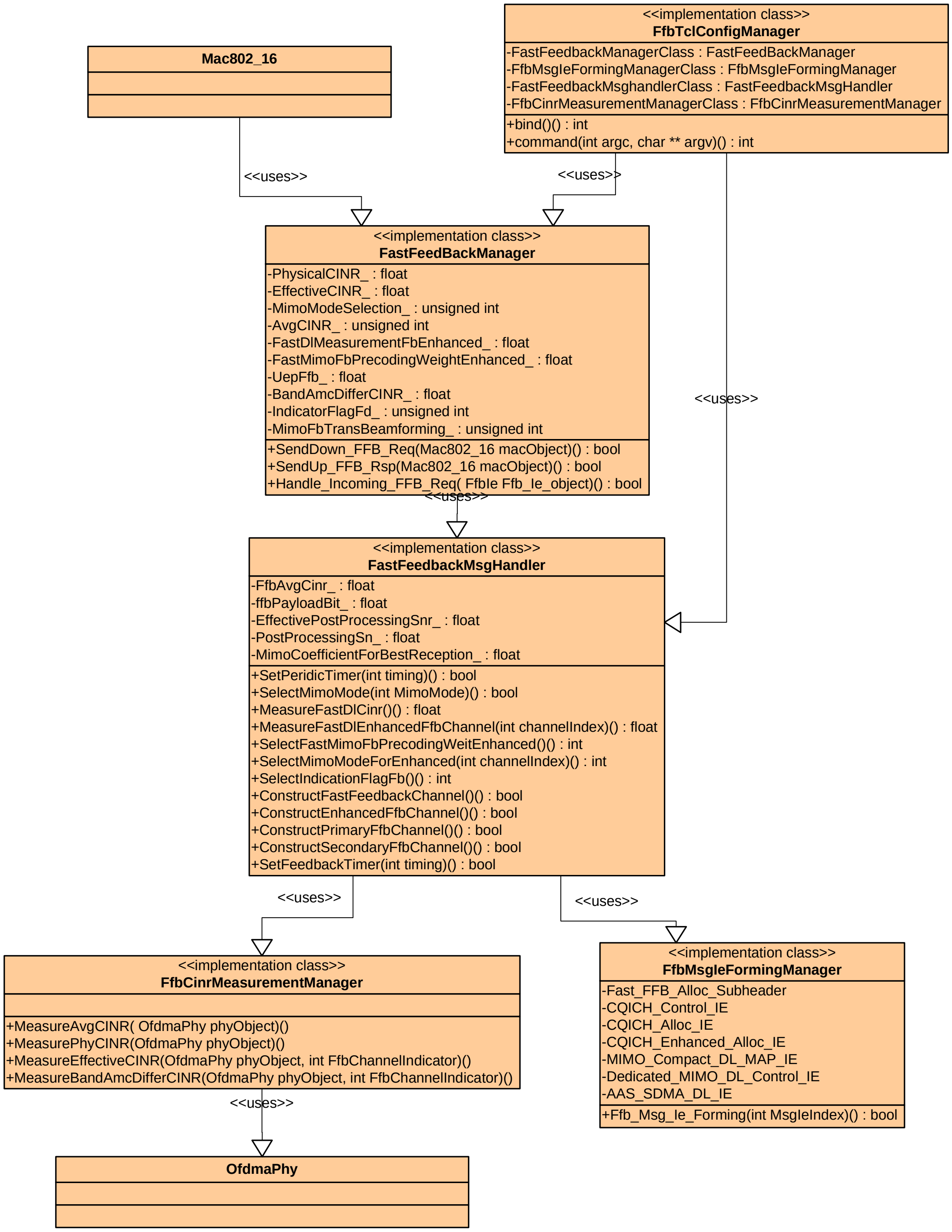


Figure 14: Fast Feedback Channel Class Overview

As shown in Figure 14, there are five major classes including

FastFeedbackManager class, MsgIEFormerManager class, FastFeedbackHandler class, TclConfigManager class and CinrMeasureManager class.

FastFeedbackManager class is the interface to all the invoker classes which relate to fast feedback service. FastFeedbackManager class could be multiple initialized. The Base Stations 802_16 MAC objects might need the instance of this class to demand the fast feedback from SSs which belong to itself. On the other end, the 802.16 MAC object of SS may need to initialize this class to ask for support to send demanded information via fast feedback channel.

FfbMsgIEFormingManager class mainly shoulders the responsibility to form the IEs which are needed by BS and sent to SSs. The IEs include Fast Feedback Channel Allocation Subheader, CQICH_Control_IE, CQICH_Alloc_IE, CQI_Enhanced_Alloc_IE, MIMO Compact DL_MAP_IE, Dedicated MIMO_DL_Control_IE and AAS_SDMA_DL_IE etc.

FastFeedbackMsgHandler class is the core class to handle different conditions. It needs to send down the fast feedback messages from the BS to the SSs. It also needs to parse the incoming fast feedback message, get the appropriate information from the message and begin function as the requirement of the messages. It still needs to send up the required information via fast feedback channel (by fast feedback channel index). A globally unique resource: fast feedback channel index will be maintained by this class and updated in runtime. All the instances of this class could and only could share a same bunch of fast feedback channel index.

FfbTclConfigManager class is the TCL interface class. It is used to supply the mechanism on TCL side to give the access right to TCL space and could easily set the value of the member attributes and methods in necessary classes.

The FfbCinrMeasurementManager class is mainly used to measure the CINR information which is needed by the BS, including physical CINR and effective CINR etc. Under different conditions, the ways to get the CINR value are quite different.

There are some direct relationship between FastFeedbackManager class and other classes like wirelessPhy or OfdmaPhy class. Also FastFeedbackManager class may get the necessary information from the physical layer and report to the BS in time.

5. Algorithm Design and Evaluation

5.1 Adaptive Modulation and Coding Scheme

WiMAX system uses adaptive modulation and coding scheme to deal with the fluctuation in the channel. In reality, the wireless channel might undergo large scale fading as well as small scale fading including multi-path fading, frequency selective fading etc. Usually small scale fading effects the channel condition in a short time period. Adaptive Modulation and Coding allows dynamic selection of modulation type and coding efficiency depending on conditions in every subchannel and required data transmission speed for both transmission directions. When the channel condition is of a high quality, WiMAX systems will select a higher modulation and coding scheme and can give higher throughput to the system, such as 64QAM with ¾ convolutional, turbo or LDPC coding. When the channel condition deteriorates, a lower modulation and coding scheme will be chosen, for example, QPSK with ½ convolutional coding.

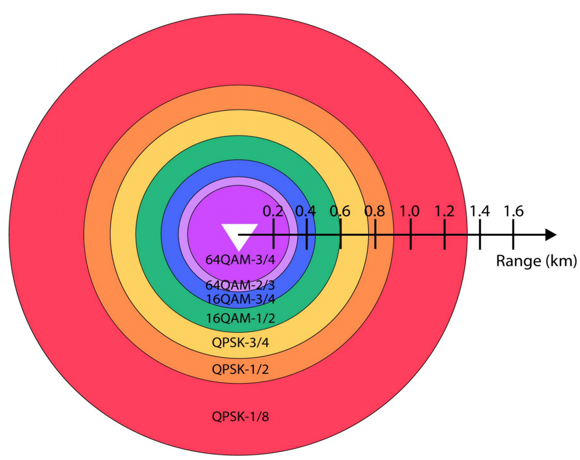


Figure 15: Modulation and Distance

AMC allows WiMAX system improve modulation plan of signal. In case wireless connection is of high quality, higher modulation and coding scheme is being used and will give the system higher throughput. In case the signal is weak, WiMAX system can change to a lower modulation and coding scheme to maintain the quality and stability of Connection.

5.1.1 System Design

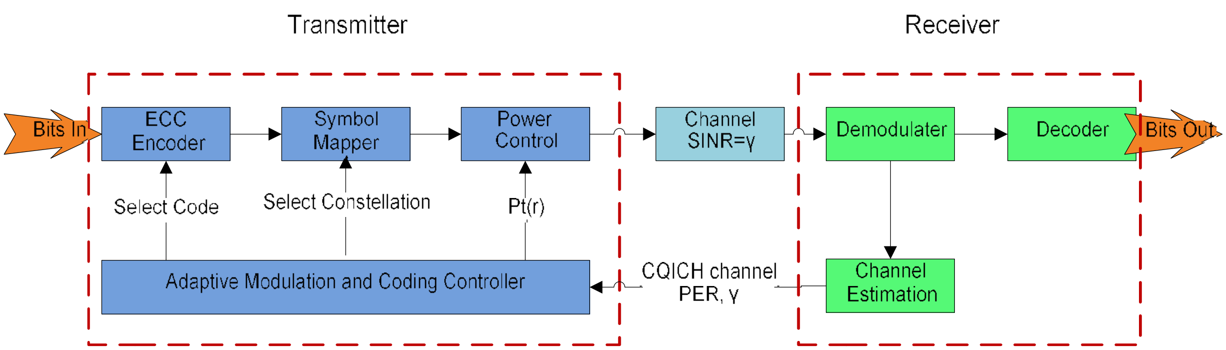


Figure 16: Function Block Diagram for Adaptive Modulation Coding Algorithm

Figure 16 shows the AMC system in our simulations. ECC encoder is mainly specialized in coding scheme selection, like convolutional coding, turbo coding, LDPC coding etc. Symbol Mapper is used to choose the modulation. Generally there are 52 configurations of modulation orders and coding types. But most of the WiMAX implementations only offer a fraction of them and the modulation and coding schemes are referred to as burst profiles.

On receiver side, the SINR of the downlink channel is measured and this measurement is fed back to the transmitter. On transmitter side, once it receives the channel quality feedback from CQICH, it will compare the downlink channel quality with the predefined thresholds and determine if it needs to change the modulation and coding scheme.

5.1.2 Positive and Passive AMC

AMC in WiMAX is desired in the IEEE802.16-2004 and 2005 standards. It is made differently in the donwlink and in the uplink. I will try to giv e a brief explanation.

Donwlink: The BS does not know the ideal coding scheme to use in the DL, so the SS must inform the BS about the required DIUC (Downlink Interval Usage Code). The SS continuously measures RSSI, CINR and data loss in the downlink, so it can decide whether current coding/modulation is correct or not. If it needs a change, either to a more robust profile or a more efficient one, it sends DBPC -REQ or RNG-REQ with the required DIUC. The BS then answers back with a DBPC-RSP or RNG-RSP with the new allowed modulation in the DL (DIUC).

Uplink: The BS continuously monitors the signal quality of all SS s connected to it. It does it by measuring the quality of demodulated symbols and ensures that all SSs are allocated data in the UL path using a timer. If the quality of the signal drops, the BS can decide to change the uplink modulation scheme for that SS, but it does not send any specific message to the SS. The BS will just use the new more robust DIUC in the next burst it allocates to that SS, indicated in the correspondent ULMAP. The same applies when the quality of the signal improves.

The above scheme is the positive AMC method. In this scheme, BS and SSs will trigger channel measurement at the other end and get the feedback. Then the corresponding modulation and coding scheme will be adopted.

In passive AMC method, take BS for example, the BS will roughly estimate the modulation and coding scheme for downlink by measuring the uplink channel. It is also called Open Loop. The advantage of this method is that it does not need the feedback from the other end of channel and therefore save radio resource and processing capacity of the equipment. The disadvantage of this method is also obvious. It can only roughly estimate the downlink channel condition. The idea behind this method is that it assumes that the channel condition will not change within a frame time period (5ms). But in reality, it does change especially in a metropolitan condition where multi -path fading, frequency selective fading will greatly change with the mobility of MS.

5.1.3 Important Properties

In AMC, the key challenge is to efficiently control three quantities at once: transmission power, modulation (constellation) and coding rate. Theoretically by applying larger transmission power, more advanced modulation and coding, we should get higher system throughput. But in practice, we might need to well tune the algorithm, based on many factors. They are:

1) BLER and received SINR

In AMC, the transmitter needs to know the instantaneo us channel SINR. From the channel SINR, transmitter will apply the suitable modulation and coding scheme. In practice, we not only use the channel SINR, but also the BLER which is the final word on whether the data rate (modulation coding) should be increa sed or not. In fact, higher SINR does not always lead to lower BLER, since in order to get the same BLER/BER, higher SINR will be needed for more advanced modulation coding scheme s.

In our simulations, the downlink AMC is more challenging than the uplink AMC. In downlink direction, BS will be in charge of all the resource alloc ation and traffic transmission. BS will allocate the data transmission region to each SS and transmit data in these regions to SSs. But the point is that in the current allocation, if the transmission region location in the OFDMA Symbol-Subchannel 2D range is different from the previous one, the feedback of the previous channel SINR will be useless. The more challenge thing is that when we apply PUSC (Partial Usage of Subchannels), the subcarrier to subchannel mapping is pseudo-random. The current mapping is definitely different from the previous one. This makes the channel SINR feedback in vain. In our simulations, we limit the permutation method to Band-AMC when we apply Adaptive Modulation and Coding. At the same time, the scheduler of BS will not change the location of the transmission region for each SS.

2) Automatic Repeat Request

ARQ allows rapid retransmission, and Hybrid-ARQ will generally increase the BLER although the channel SINR might not good if Incremental Redundancy is used in boosting the unsuccessfully transmitted packets.

3) Adaptive modulation in OFDMA

When deploying Adaptive Modulation Coding scheme in OFDMA physical layer, each user will have a block of sucarriers, each having a different modulation and coding scheme. Because of the limited battery capacity of the Mobile Stations, only increasing the transmission power when SINR is low or decrease transmission power when SINR is high will exhaust the battery soon. Adaptive Modulation Coding scheme will be another approach to maintain BLER without increasing the Tx power. It will save Tx power and therefore prolong the sustainability of the battery.

5.1.4 Experiment of Results

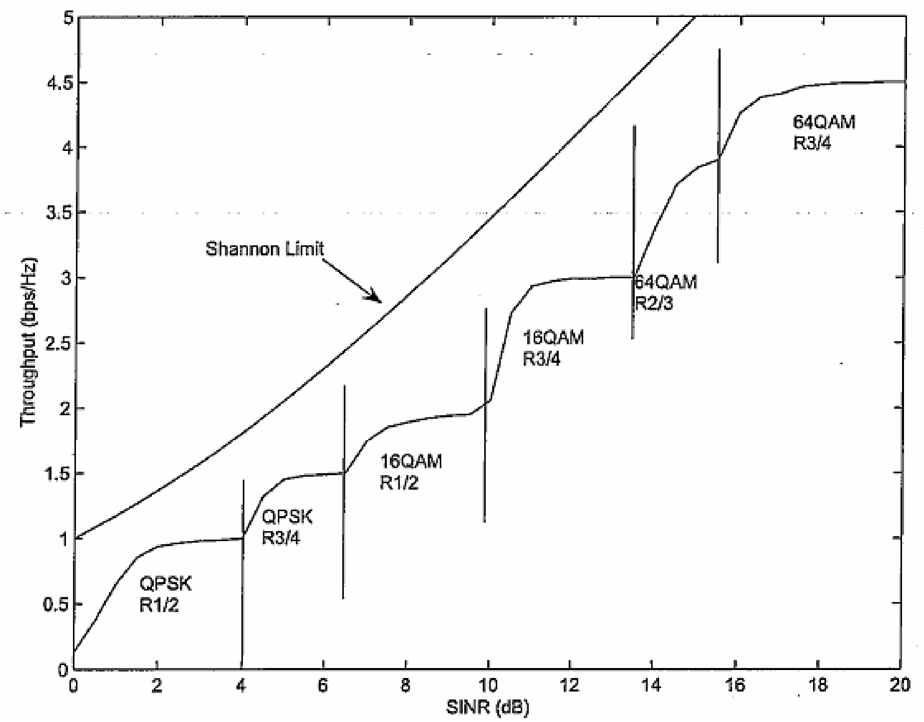


Figure 17: Throughput versus SINR

By using the Adaptive Modulation Coding mechanism, we can boost the system throughput as closely as possible to the Shannon Limit. In the experiment shown in Figure 17, the lowest throughput is obtained by using QPSK with ½ Turbo coding. The highest throughput is obtain by applying 64QAM rwith ¾ Turbo coding. The throughput is defined as:

$$T = (1 - BLER) r \log_2(M) bps / Hz \tag{9}$$

where BLER is the block error rate, r is the coding rate which is usually less than 1 and M is the number of points in the constellation. From this formula, we can see the throughput can be higher if a bigger M or r is used. Theoretically it is true, but it is not the case in reality. By applying more advanced modulation, the M will increase. However the more advanced the modulation scheme that is used, the better the radio condition it will need.

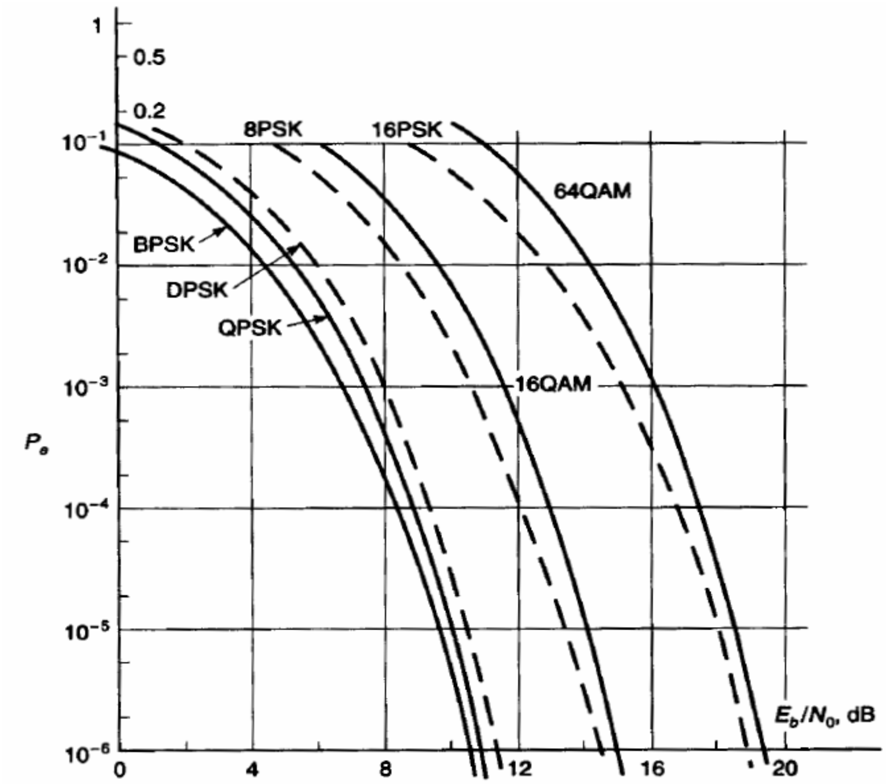


Figure 18: BER and Eb/No for different Modulation

As shown in Figure 18, in order to achieve the same level of BER, BPSK needs about 10dB Eb/No. On the other hand, 64QAM needs about 19dB Eb/No. That is to say, the radio condition must be good or we need to apply higher transmission power. But the thing is that with the increasing of Tx power, the interference between S Ss will increase greatly.

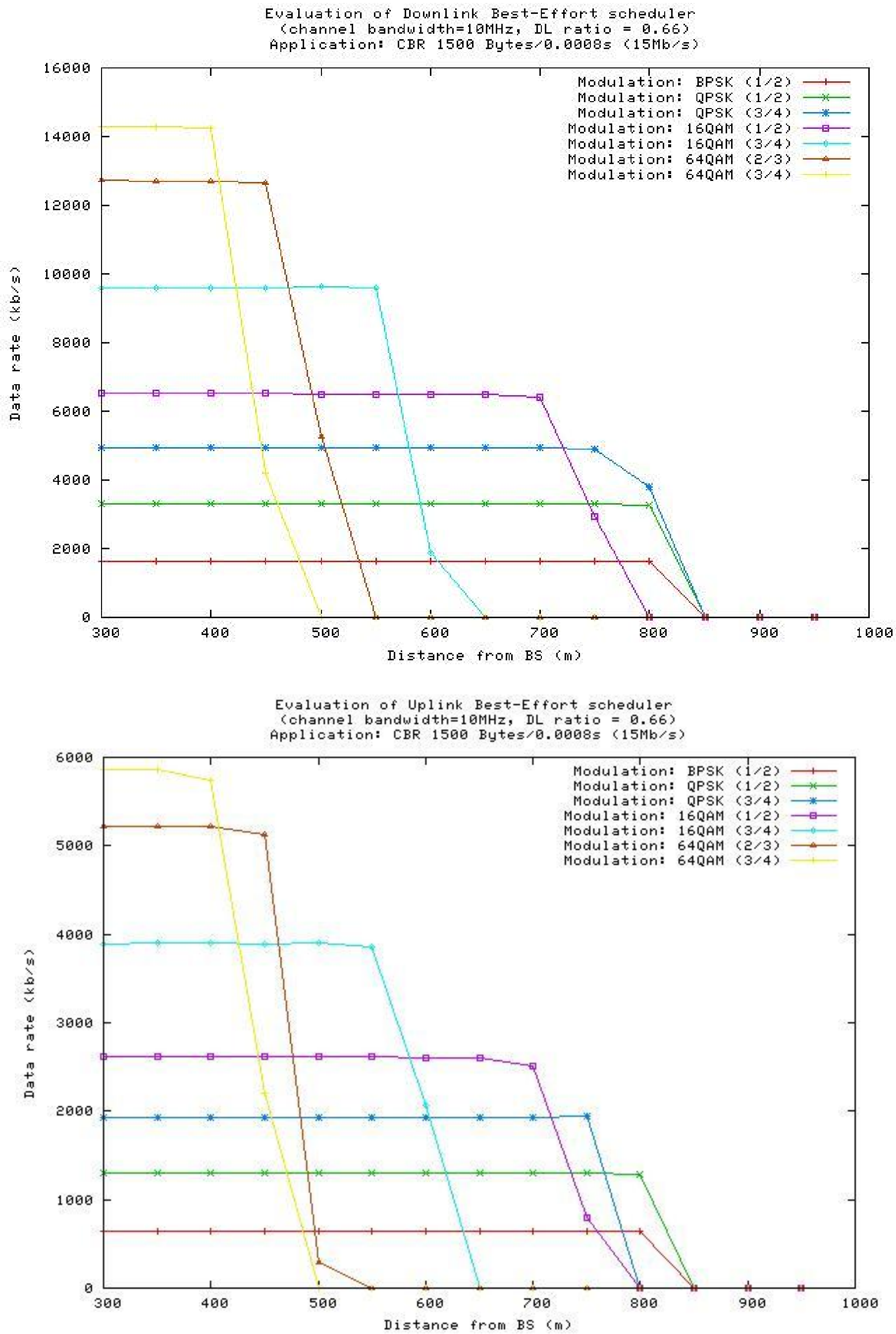


Figure 19: Evaluation the Modulation and Distance in UL and DL without AMC

Figure 19 shows the system throughput in downlink and uplink directions as the distance between BS and MS increases. When the distance increases, the BLER is increased greatly. Therefore at a certain distance for each modulation and coding scheme, the system throughput will drop to zero.

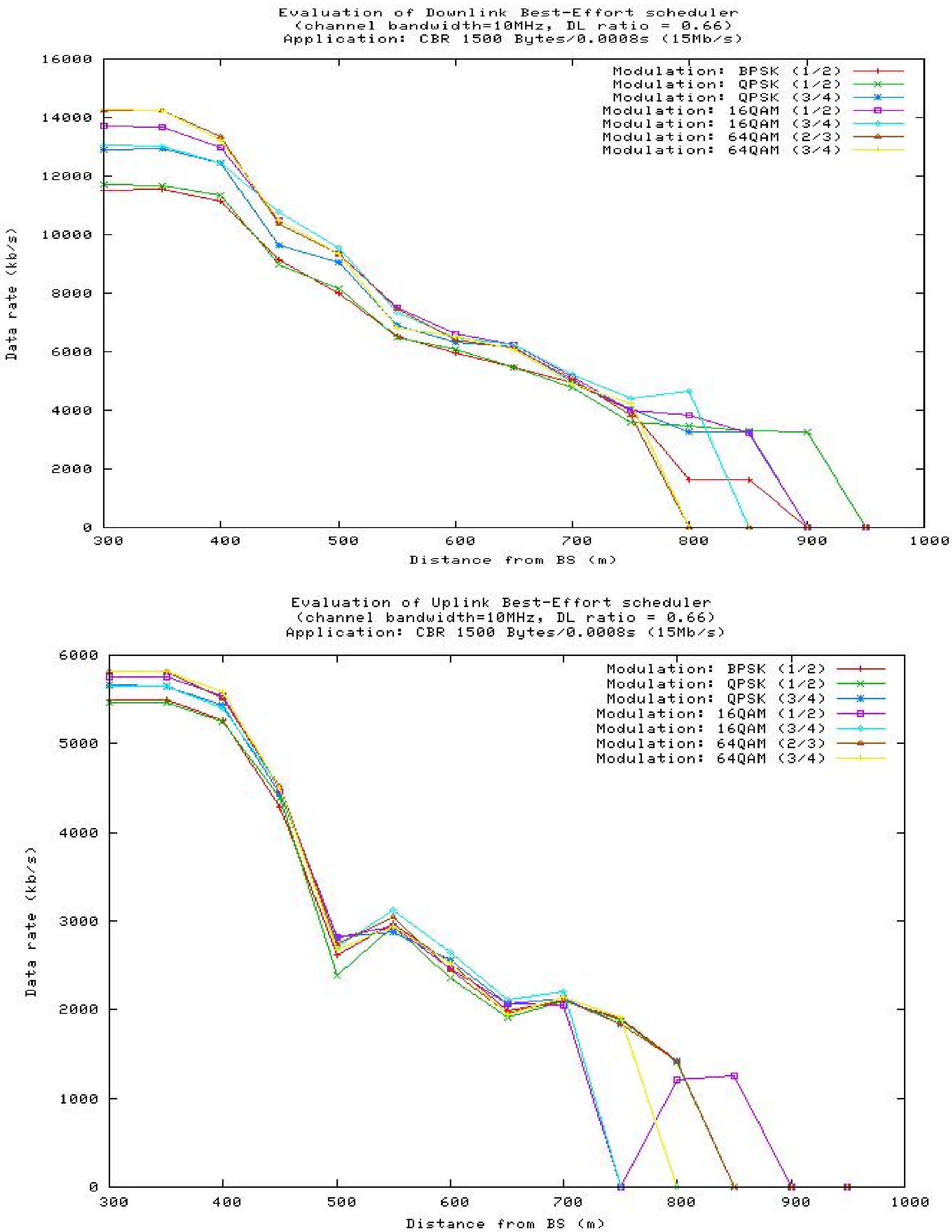


Figure 20: Evaluation the Modulation and Distance in UL and DL with AMC-Constant Rate

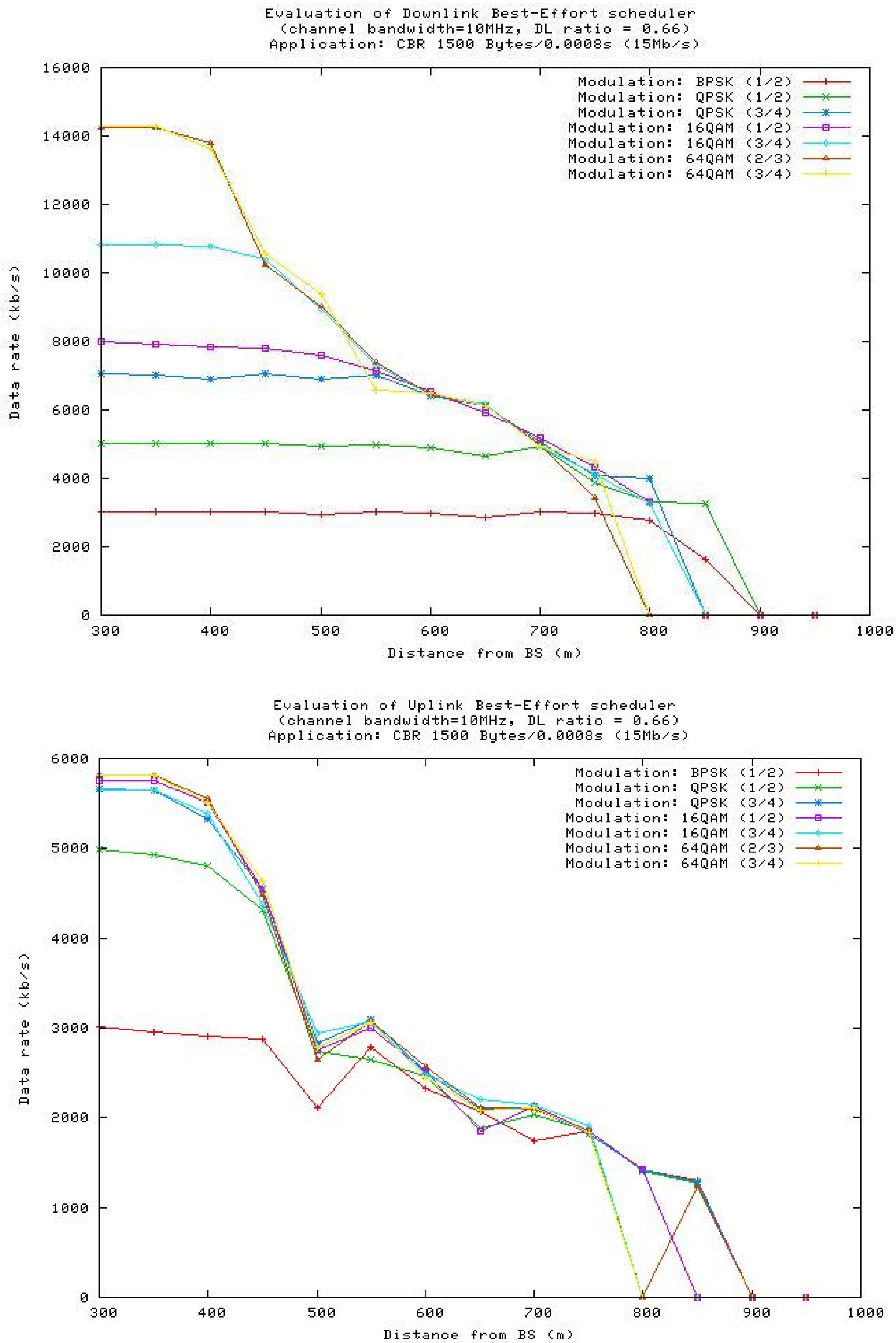


Figure 21: Evaluation the Modulation and Distance in UL and DL with AMC -Variable Rate

In the above simulations, there is one BS and one MS. The MS keeps on moving against the BS while transmitting data. In Figure 20, the data rate is the set to 15Mbps. While in Figure 21, the data transmission rate is set to the maximum value which each specific modulation coding scheme can sustain. As the distance between BS and MS increases, the SINR will degrade and thereby the BLER will increase. In order to maintain a certain level of BLER, the Adaptive Modulation and Coding mechanism will automatically configure the modulation coding scheme to a more robust one, say from 64QAM 3/4 to 64QAM ½ or from 64QAM ½ to 16QAM 2/3. As a result, the BLER will become better and be above a certain level. But all these positive credits are obtained at the cost of proportional decreased of system throughput. Vice versa, if the BLER is below a certain threshold, the adaptive modulation coding mechanism will also automatically choose more advanced modulation coding scheme in order to increase the system throughput without taking up more system resources.

From the above simulation results, we can see that as the distance changes, different modulation and coding scheme shows different properties against the radio interference. The more advanced the modulation coding scheme, the higher the throughput but the shorter the distance between BS and MS that will be needed.

5.1.5 AMC Algorithm Logic

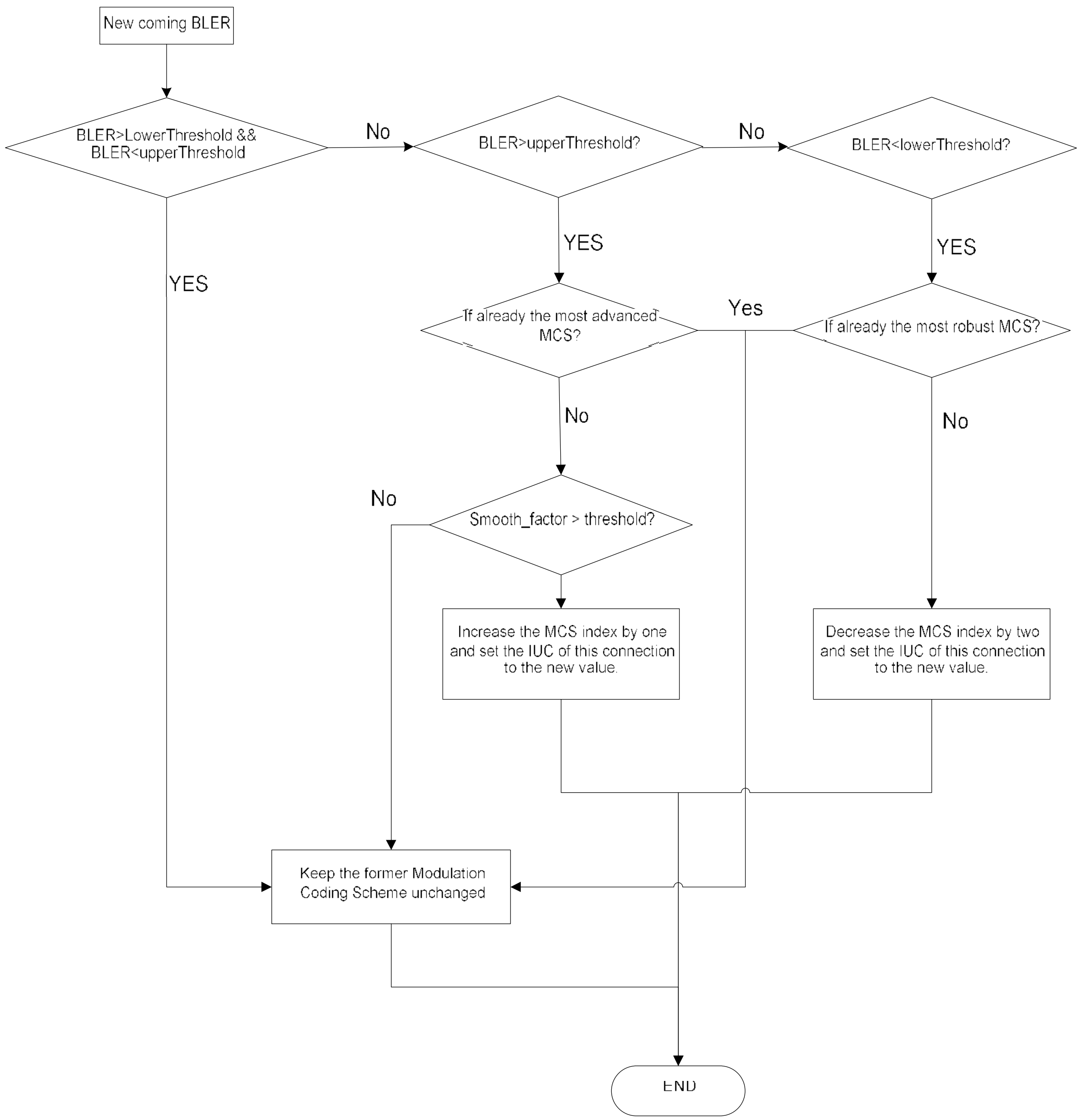


Figure 22: Adaptive Modulation Coding Mechanism Algorithm Logic Basic

Our current AMC algorithm to control the MCS is a basic one: when BLER is lower than a threshold, increase MCS index by one. While the BLER is greater than a threshold, decrease MCS index by two. Since the radio channel will degrade not in a linear form, we need to decrease the MCS index quickly. Here the decrease by two each time is a tradeoff between maximizing system throughput and low level of BLER. In order to get higher system throughput, a higher MCS index is needed. Once the channel is under fading, a lower MCS index is needed to ensure a low level of BLER. Too fast speed of decreasing of MCS index will obviously underutilize the system bandwidth or radio channel.

In this algorithm, a smooth factor mechanism is introduced. It is used to counteract the Ping-Pong effect. Under stable stage (the BLER is in an acceptable range), without the anti Ping-Pong mechanism, the algorithm will try to increase MCS index. And soon the BLER gets worse and therefore the AMC needs to roll back. This Ping-Pong effect will exhaust more quickly the battery of the MS. At the same time, it will increase the BLER and decrease the system throughput. In our algorithm, the smooth factor will increase by one each time there is a new Fast Feedback report. Once the smooth factor is over a threshold, algorithm can increase MCS index. Otherwise it keeps the MCS index unchanged.

One important idea about the smooth factor is that it does not take effect if the algorithm wants to decrease MCS index. Because the attempt of deceasing of MCS index results from the worse BLER, it is very urgent to be done.

The smooth factor is set to upper_threshold for the downlink and lower_threshold for the uplink. After validation, the upper_threshold is recommended to 4 and lower_threshold to 2. These are empirical values and are tradeoffs between the BLER and system throughput.

5.1.6 AMC Algorithm Efficiency Analysis

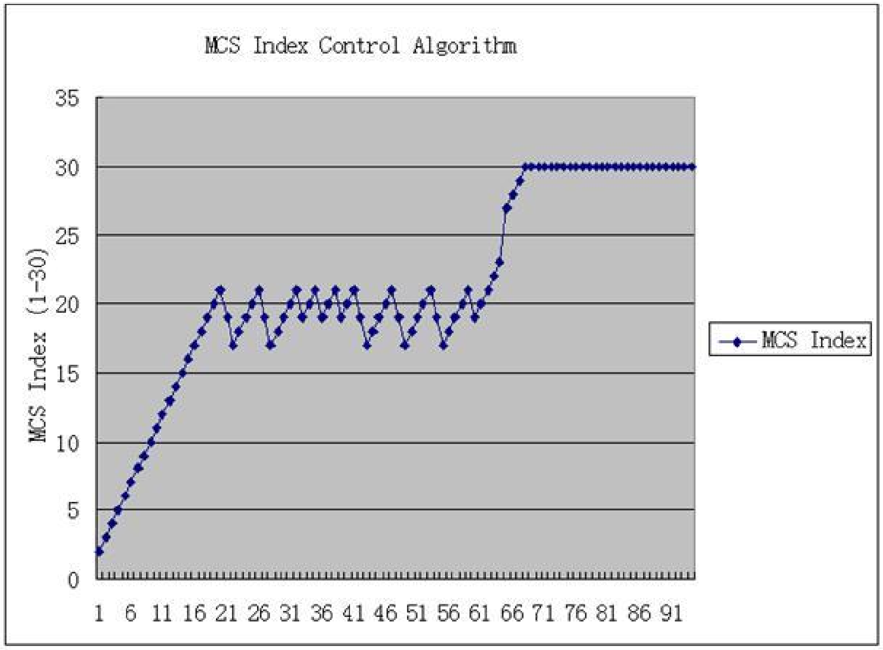


Figure 23: Adaptive Modulation Coding Mechanism Algorithm Efficiency

Figure 23 shows the trace of AMC algorithm to control the MCS index. In our AMC algorithm, we implement a basic scheme as mentioned in the previous section. When the BLER is below the lowerThreshold, the MCS index will be increased by one. When the BLER is above the upperThreshold, the MCS index will be decrease d by two. The motivation of this algorithm to decrease faster than increase is to ensure a n acceptable BLER and suitable system throughput. Since usually the radio fading is not linear, we need to decrease a little bit quickly. But it is not good to decrease too fast since it might underutilize the system bandwidth.

In fact, the algorithm to control the MCS index can be optimized and be more efficient. A simple idea to improve it is that we can borrow idea from TCP congestion control mechanism. We can have two similar phases: slow start and stable phase. In the slow start phase, the MCS will be increased exponenti ally so that to get close to the equilibrium more quickly. On entering the stable phase, if there is a need to decrease the MCS index, we set value to half of the former one and use AIMD (Additive Increasing Multiplicative Decreasing). Technically speaking, this method will be more efficient than the current basic algorithm.

In addition, the current approaches are all post-processing ones, i.e. changing the MCS index according the reported BLER. The fact is that the data in the last frame might be corrupted and therefore waste bandwidth. A more promising idea is to predict the possible fading and change the MCS index before the fading happens. In this way, the system throughput can be greatly enhanced.

5.2 ARQ support

The ARQ (Automatic Repeat reQuest) mechanism is one part of the MAC layer and it may be enabled on a per-connection basis. The per-connection ARQ shall be specified and negotiated during connection creation. In our implementation, ARQ feature could be easily enabled and disabled. [2, 3, 4]

5.2.1 ARQ with Fragmentation and Packing

When ARQ is enabled, fragmentation and packing scheme would also be add-on parts for ARQ. Figure 24 shows the fragmentation and packing of ARQ blocks with and without rearrangement.

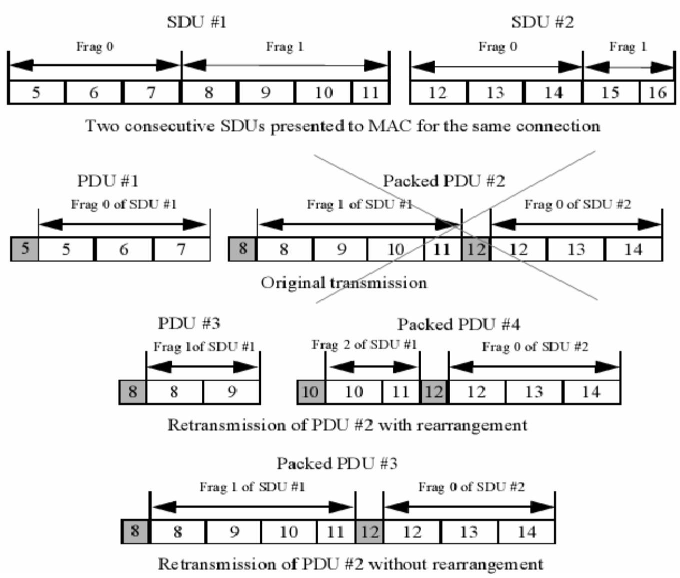


Figure 24: Block usage examples for ARQ with and without rearrangement

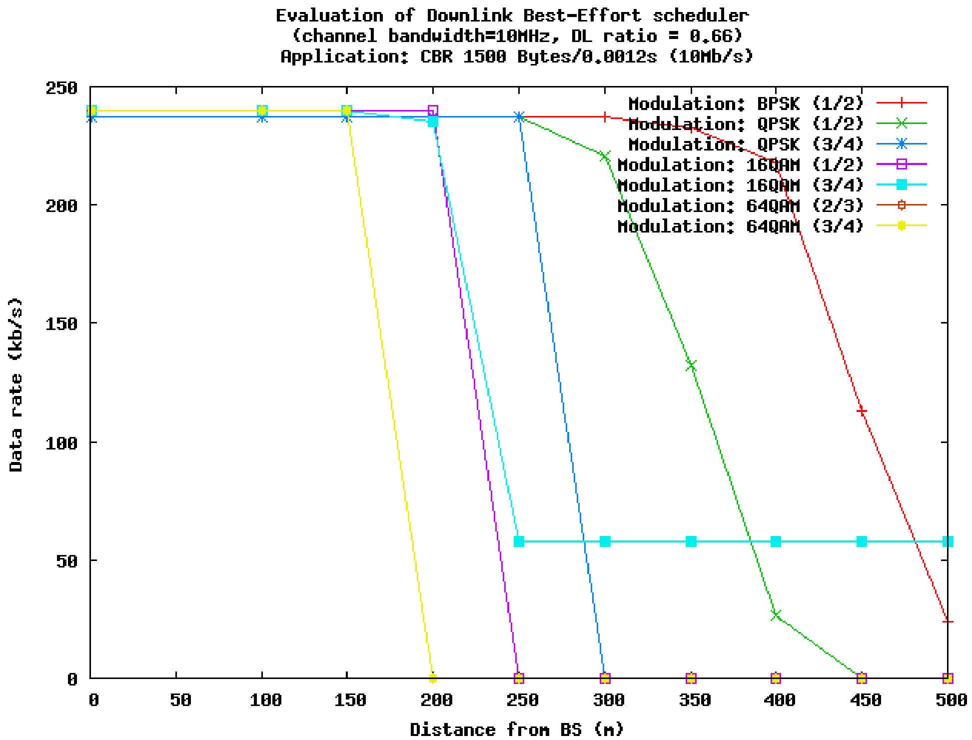
Once fragmentation is enabled, it always occurs at the ARQ boundaries, i.e. the fragmentation block will always consists of multiple number of ARQ blocks. If packing is enabled, the packed PDU would probably consist of ARQ blocks from different MAC SDUs. On some occasions, because of re-transmission and limited transmission slot size, rearrangement of the ARQ blocks in the former packed ARQ PDU would be a necessary mechanism. 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 system is high.

5.2.2 ARQ transmission and receiving logic

In 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 transmission queue, and wait for scheduling and transmission. Sliding window scheme is applied in transmission and reception logic. Once one 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, re-transmission queue has higher priority over transmission queue and therefore ARQ blocks in re-transmission queue will be scheduled earlier than those in transmission queue. If the times of re-transmission of a certain ARQ block exceed a configured value, this ARQ block will be dropped. Bandwidth requests are supported in two ways: bandwidth request and piggyback. In receiver side, it will acknowledge successfully received ARQ block(s) by sending ACK back. Once out-of-order ARQ blocks come, the receiver will send ACK with corresponding BSN (Block Sequence Number) back. Because of the limitation of sliding window size, the receiver will not be overwhelmed because of lack of capacity to handle the incoming traffic. If all the ARQ blocks for an original MAC SDU are present, the reassembly module of thereceiver will reconstruct this MAC SDU and send it to the upper protocol layer.

5.2.3 Experimental Results

Our experiments show that the ARQ mechanism can greatly improve the successful reception rate of traffic. Figure 25 shows that with the increase of distance between BS and MS, as well as with different modulation scheme, the successful reception rate of data traffic changes differently with ARQ disabled on the left and enabled on the right.



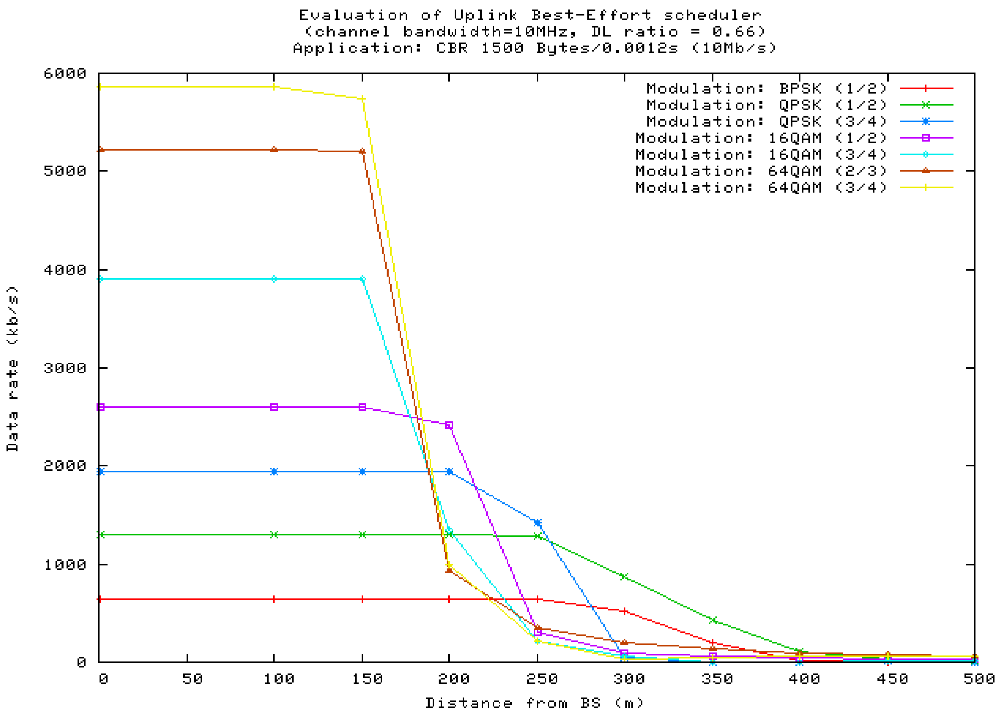


Figure 25: Modulation Throughput Curve with Variable Distance with/without ARQ enabled

When BS moves further 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, corrupted data traffic will be re-transmitted. The diagram in Figure 25 shows that the rate drops not that dramatically. Instead it will keep flat most of the time with different modulation schemes.

5.2.4 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 acknowledgement would cost extra system resource and bring burden to both Base Station and Mobile Station. For SS, because of its limited battery capacity, the extra system resource and burden brought by ARQ would shorten the sustainability of the SS.

5.3 Handover Scheme

For implementing a mobile network, the handover mechanism must be defined to maintain uninterrupted user communication session during his/her movement from one location to another. Handover mechanism handles SS switching from one BS to another. Different handover techniques have been developed. In general, they can be divided into soft handover, hard handover and Macro Diversity Handover.

5.3.1 Soft Handover

The principle of Soft Handover (SHO) is make-before-break. That is to say, the Mobile Station (MS) will keep an active/monitor set in which there are a set of neighboring BS. In case of channel SINR dropping below a certain threshold, BS or MS can initialize the SHO by measuring the channel SINR of neighboring BSs and select the best radio quality BS and proceed to SHO.

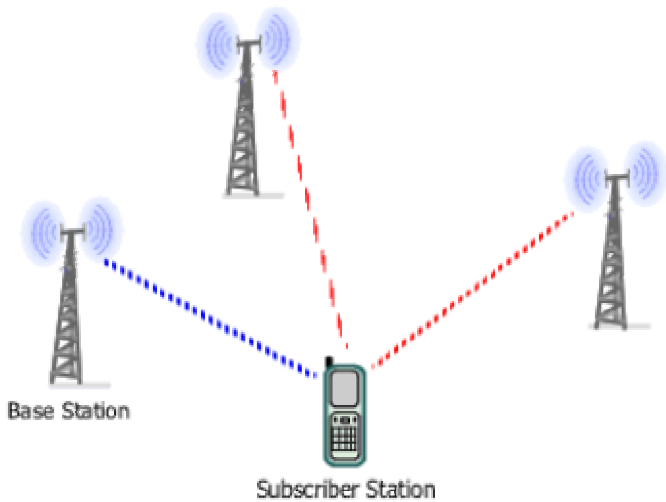


Figure 26: SHO scenario

As shown in Figure 26, as the ss moves away from the original BS, the channel SINR might begin to drop. When the SS moves to the edge of the cell, the channel SINR of neighboring BSs becomes better than the original channel. The SS will initialize the SHO and switch the data traffic seamlessly from the original BS to the target BS.

This technique is suitable to handle voice and other latency-sensitive services such as Internet multiplayer games and video conferences. When used for delivering data traffic (such as web browsing and e-mail), soft handoff will result in lower spectral efficiency because this type of traffic is bursty and does not require continuous handover from one BS to another.

5.3.2 Hard Handover

Opposite to the SHO, the principle of Hard Handover (HHO) is break-before-make. Mobile WiMAX has been designed from the outset as a broadband technology capable of delivering triple play services (voice, data, video). However, a typical Mobile WiMAX network is supposedly dominated by delay-tolerant data traffic. Voice in Mobile WiMAX is packetized (what is called VoIP) and treated as other types of IP packets except it is prioritized. Hard handoff is therefore used in Mobile WiMAX. In case of HHO, the connection should be broken before SSs switch to another BS and setup connection again. Compared to SHO, HHO is more bandwidth efficient than SHO. But it might cause longer transmission delay and large data loss. Therefore a comprehensive network optimization scheme might be needed to deploy the wireless network and let the HHO delay below 50ms which is an acceptable threshold.

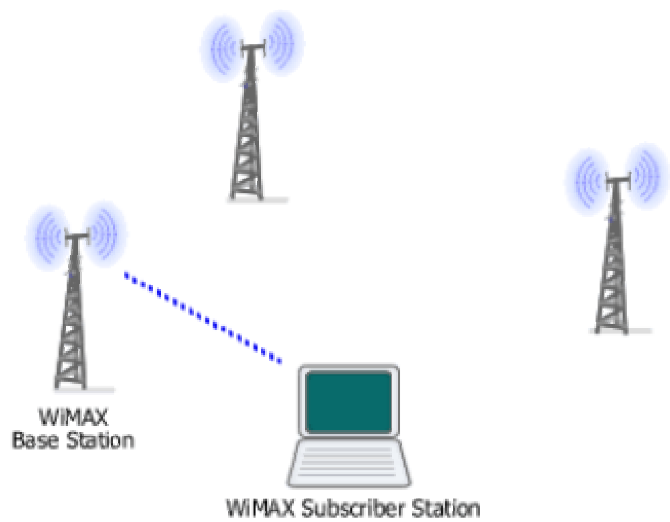


Figure 27: Hard Handover scenario

5.3.3 Macro Diversity Handover

In Macro Diversity Handover (MDHO), a set (called diversity set) of BSs will be regarded as single virtual BS. MS could communicate with any BS(s) in this set and there will be no handover if MSs setup a new connection with BS in this set. Once a MS finds a BS which has better channel SINR and this BS does not belong to the diversity set, a MDHO might be initialized. MDHO greatly reduces the possible handover and therefore cuts down the traffic transmission delay. But it increases the complexity of the implementation of BS as well as MS.

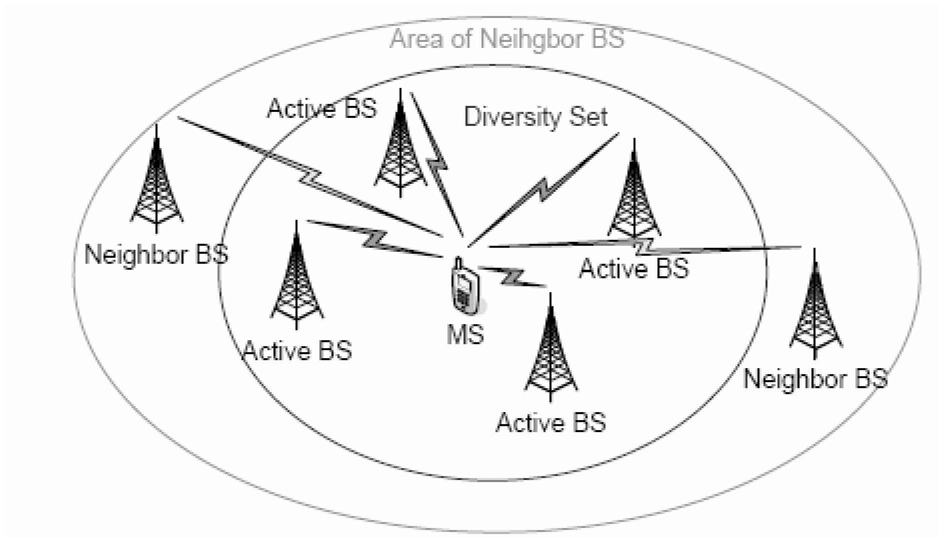


Figure 28: MDHO Handover scenario

5.4 Experiment of Results

In our simulations, we only support HHO which is the mandatory option of mobile IEEE802.16e standard. In our experiments, two BSs are deployed and one MS will move from BS to another and after a few seconds it will move back to its original BS. Therefore two HHOs will be occurring during the movement.

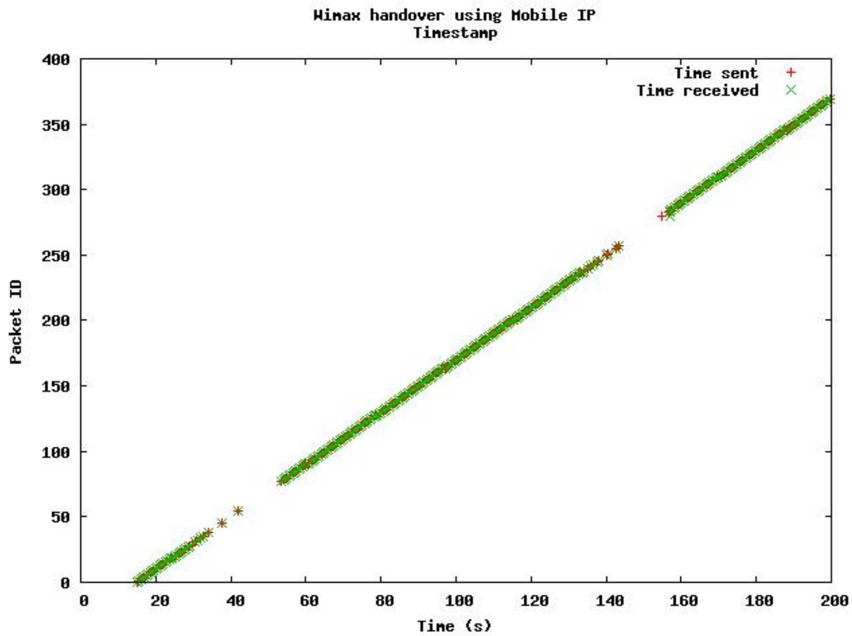


Figure 29: Throughput and Transmission Time during WiMAX Handover

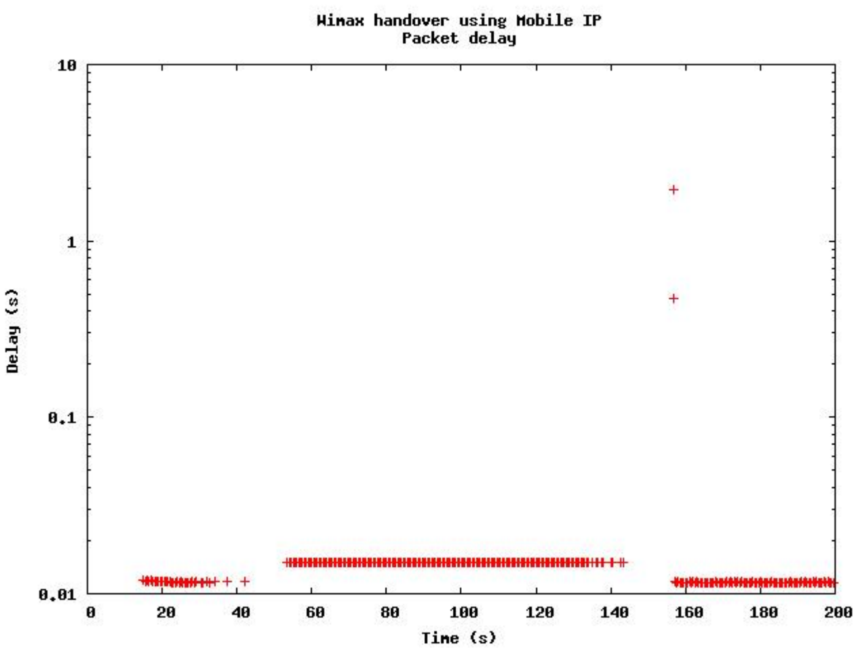


Figure 30: Transmission Delay during the WiMAX Handover

From the Figure 29, we can see that the first HHO takes place at about 40s and the second HHO is at about 150s. During the HHO, some of packets are dropped because in

HHO the connection between BS and MS will be break first and a few time periods later a new connection will be established. The data traffic will then be transmitted again.

From Figure 30, during the HHO, there is no delay point in the figure because there is no data traffic at this time period. One thing to note is the figure is that the delay of transmission when the MS switches to the target BS is a little bit larger than its original BS. The reason of this phenomena is that the traffic coming from Core Network (CN) still does not switch to the target BS yet. The data traffic still goes through the original BS to the target BS and then comes to the MS. Therefore the traffic delay is a little higher than that in original BS.

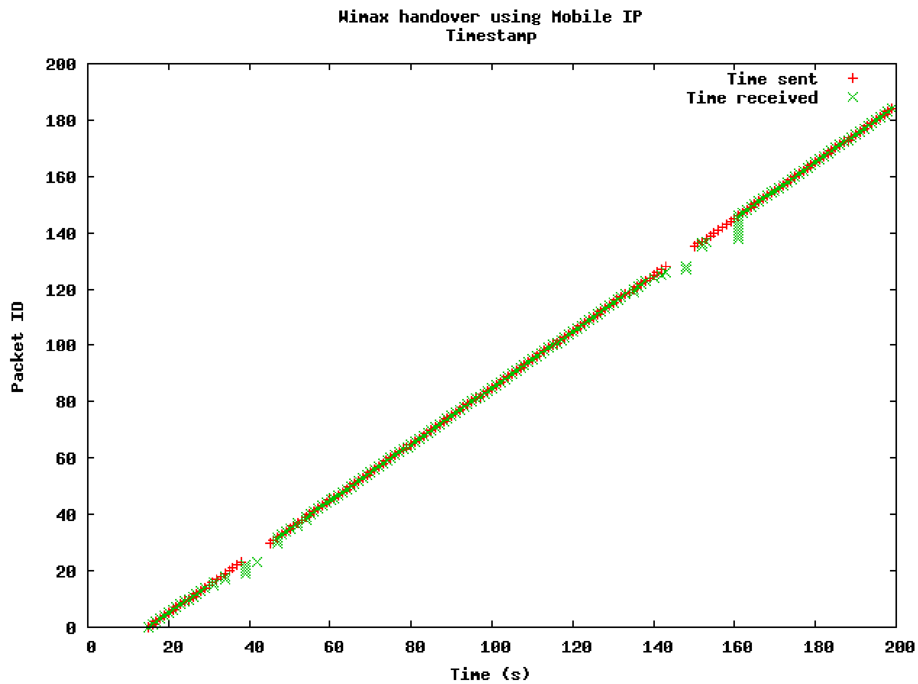


Figure 31: Throughput and Transmission Time during WiMAX Handover with ARQ

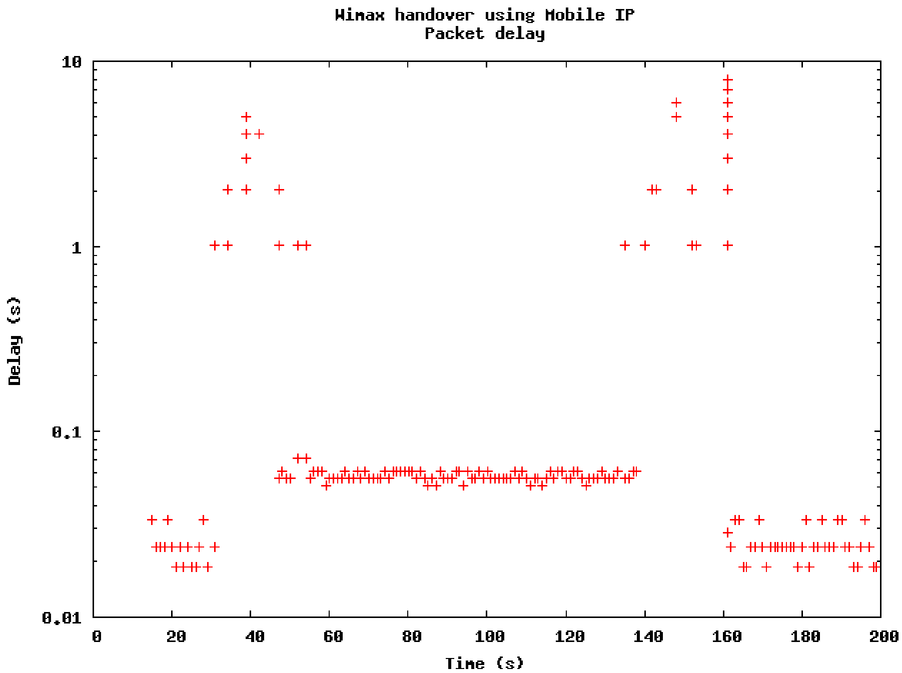


Figure 32: Transmission Delay during the WiMAX Handover with ARQ

Figure 31 and Figure 32 show the handover scenario with ARQ enabled. Basically they looks very similar to the figures for handover scenario without ARQ. The transmission delay of WiMAX handover with ARQ enabled is larger than that without ARQ because in case of ARQ enabled scenarios, there is a certain data loss rate (0.0-1.0). In our simulation, we manually configure the data loss rate to 50%. Since there is some data loss and therefore retransmissions, the handover delay will certainly be higher than that without ARQ, as seen from the above figures.

5.5 Miscellaneous Features

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

6. System Simulation Results

In our simulator, the simulation time will increase almost exponentially with the number of nodes. Given the data traffic bandwidth of 240kbps, ARQ disabled and data traffic persistent time 10 seconds, Figure 33 shows that with the increasing in the number nodes, the simulation time increases almost exponentially.

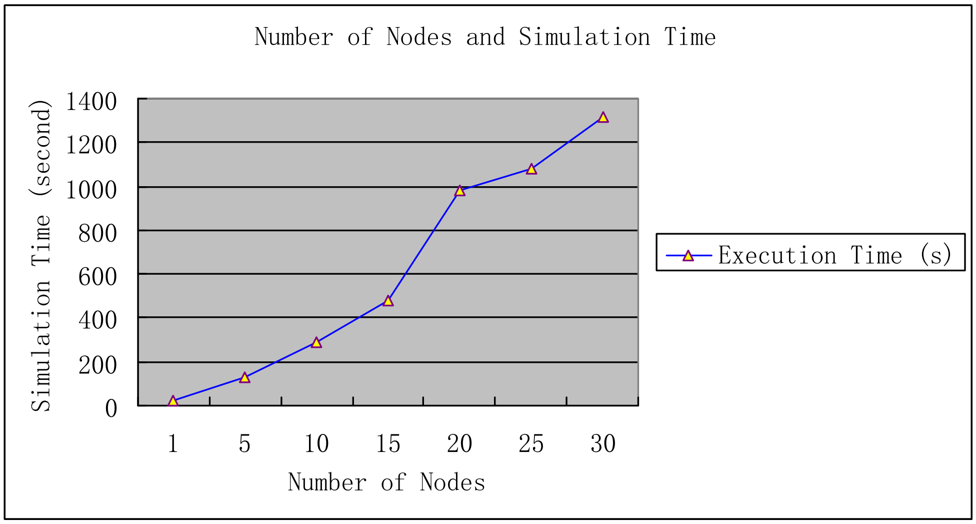


Figure 33: Number of Node and Simulation Time diagram w/o ARQ

With 512 ARQ block size, 512 ARQ receiving window size, 10% data loss rate and data traffic persistence time of 10 seconds and one node, Figure 34 shows that given the number of nodes and with ARQ disabled, with the increase of data traffic bandwidth for each node, the simulation time also increases exponentially.

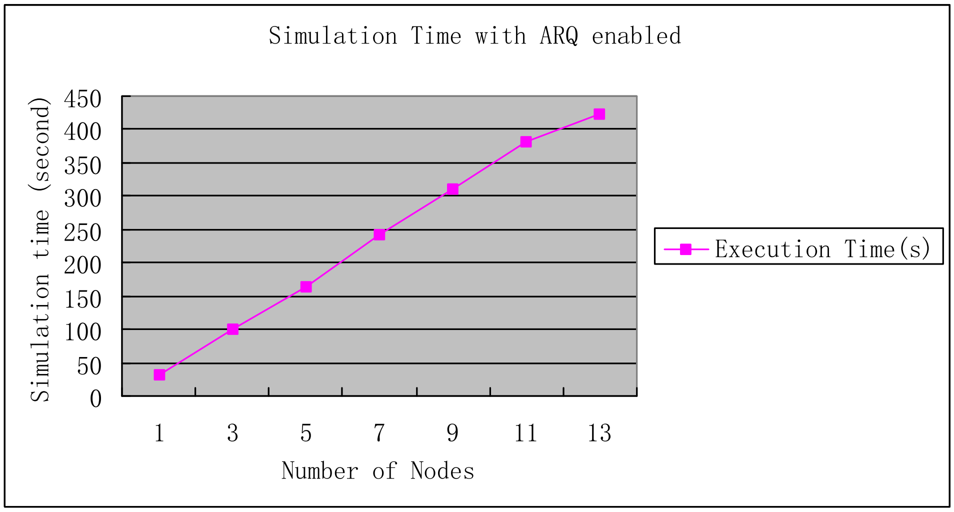


Figure 34: Number of Node and Simulation Time diagram with ARQ

With ARQ, each MAC SDU will be divided into ARQ blocks. The smaller the size of the ARQ block, the longer the simulation time will be.

7. Conclusion and Discussion

In this thesis we mainly presented the technologies of WiMAX which we implemented in the network simulator ns. These technologies together not only could address the link level simulation accurately but also systematically simulate the WiMAX performance. Statistics module can provide valuable information related to the data traffic transmission like delay, jitter, loss rate, throughput etc. Flexible configuration design makes the WiMAX simulation more controllable and the results more accurate. Configurable DL-UL radio, frequency bandwidth, rtg, ttg can make the physical layer properties more realistic. Series of timers and corresponding management messages organically assemble the WiMAX simulation processes and can maximally simulate the behavior of each simulation block especially during the load test. Reasonable buffer and queue management schemes can could accurately simulate conditions both under low and heavy load upon the WiMAX system.

We are currently working on some more advanced features like MIMO (Multiple Input Multiple Output), HARQ (Hybid Automatic Repeat request). MIMO with STC (Space-Time Coding) implementation could help us quantitatively know how much MIMO technology can improve the WiMAX system performance. HARQ and AMC with CQICH (Channel Quality Indication Channel) mechanism can allow an additional degree of control mechanism to enhance the system performance. Sectorization can measure the mobility properties of mobile WiMAX system.

Besides the above features which are defined in the IEEE802.16 standard, some other useful technologies were not included in standard but could be valuable supplements to WiMAX system:

1) *Power Balance mechanism*. BS could provide Tx power indication to each SS as to that minimize the near-far effect and cell breath phenomena.

2) *Enhanced Cell Reselection*. This mechanism could improve the interoperability between WiMAX and other wireless communication systems like 2G, 3G networks etc.
3) *Radio Admission Control*. It could help the BS know the current cell load and could make reasonable decision when new SS(s) want to register into it.
4) *Traffic Management*. It helps BS quantitatively know its CPU, buffer and queue consumption. Based on these information, BS can send increase or decrease traffic indication to some SS(s) so that BS will not be overwhelmed or waste bandwidth.
5) *Radio Overload protection.* It can help BS control its transmission power within a reasonable range so that its transmission electronic components would not be damaged.

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