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# High Throughput Performance and Quality Issues of a Wimax System

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*Abstract:* - Today there is a strong demand for broadband multimedia services and therefore it is important that a technology provides high bit rates to the end users. These high bit rates need to be provided to the end user in a wireless manner any time, in any location, even in harsh environmental conditions and at an acceptable quality. This paper presents experimental results of a Wimax system that follows the specifications of 802.16d 2004 standard, it depicts high bit rate measurements, it presents techniques for achieving low Bit Error Rates (BER) and discusses QoS issues for wireless 802.16d 2004 links.

*Key-Words:* - Multimedia Services, Wireless Systems

## 1. Introduction

Recently the Wimax (Worldwide Interoperability for Microwave Access) is poised to become an important player of fixed, portable and mobile data networks. Wimax is an implementation of the IEEE 802.16d 2004 standard now that uses orthogonal frequency division multiplexing (OFDM) for optimization of wireless data services. The OFDM technology transmits the information by assigning small subcarriers (KHz) to end users depending on the radio frequency conditions [1]. This achieves high spectral efficiency and makes Wimax networks very well suited to high speed data connections for fixed and mobile users. This paper shows experimental results with throughput links of up to 47 Mbps approximately for UDP data transmission, for short distances between the base station (BS) and the end user's terminal (SS) something that decreases as the distance increases. At the physical level, the throughput can be as high as 72 Mbps. Such a high throughput can be used to support a number of applications, including 'last mile' broadband connections, hot spot and cellular backhaul, and high speed enterprise connectivity for businesses.

A parameter of a wireless link that needs to be taken care, especially for high bit rate throughputs as that offered by Wimax technology, is the BE R parameter. This paper shows methods of setting a wireless link between a BS and an SS in order to achieve low BER. These methods can be applied by adjusting the transmitted and the received power of either the BS or the SS for different modulations (QPSK, 16QAM and 64

QAM). Curves are also shown that depict the relation of BER to Carrier to Interference Noise Ratio (CINR).

Wimax is an Ethernet layer technology in which voice is packetized and transmitted in Ethernet frames. Depending on the packetization size on the Ethernet frame, the voice quality experienced in a communication between two users can vary in different levels. This paper discusses QoS issues for the voice service transmitted in a wireless Wimax link.

## 2. BER-CINR measurements

An important relation regarding the quality of wireless systems links is BER versus Signal to Noise Ratio (SNR) function for different types of modulation schemes [2]. This relation is a strong quality indicator for a link as it provides a complete description of how a system actually operates under certain conditions of signal power over noise interference. So, it is important to derive the BER vs SNR curves of a Wimax system in a real operational environment. In our case, we shall provide the BER vs Carrier to Interference plus Noise Ratio (CINR) curves as the Wimax system that we tested provided the CINR parameter at a wireless link.

The system we tested was operating in the 3.5 GHz range and was able to operate in adaptive modulation mode which is generally used, but we were able to force our system to operate with specific modulation schemes (QPSK ¾ , 16QAM ¾ , 64QAM 2/3, where ¾ and 2/3 are the FEC coding rate) by properly adjusting certain thresholds. The topology under which these measurements were taken place is illustrated in Figure [1] and includes two BER meters at the end of the BS and the SS, directly connected to the E1 interfaces provided. The use of two BER meters enabled the simultaneous measurement of both uplink and downlink. Also, there was an LOS condition between the BS and the SS, and the channel bandwidth was set to 14MHz and the cyclic prefix or guard interval was set to ¼, which gives the maximum separation between the subcarriers and thus provides robustness against multipath.

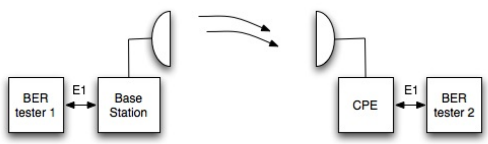


**Figure 1:** Block diagram of the system for BER-CINR measurements

In order to accomplish increasingly high BER measurement points at the SS side, the transmission power of the Base Station has been gradually decreased, something that led to the decrement of the received CINR at the SS side and as a result the BER increments. Thus, for adjusting the values of power transmittion at the BS, the BER over CINR curves were obtained Figure 2.

At the BS side, we were not allowed to adjust the transmission power of the SS, as it was dynamically adjusted and controlled by the BS depending on the Received Signal Strength Indicator (RSSI) level. So the measurements were based to the fact that by decreasing gradually this minimum accepted RSSI threshold at the BS side, it was possible to indirectly control the transmission power of the SS. As a result, as the transmission power at the SS was indirectly decreasing and so did the received CINR at the BS, leading to high BER values Figure 3.

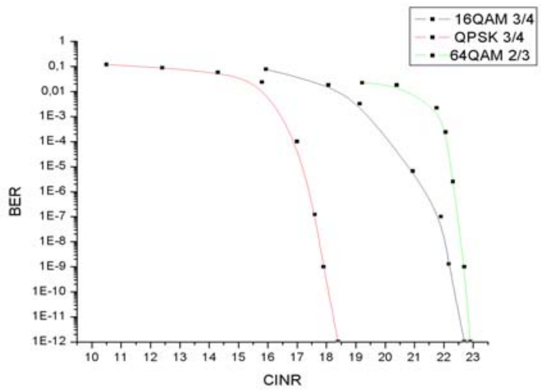


**Figure 2:** BER-CINR for QPSK, 16QAM, 64QAM at the Subscriber Station

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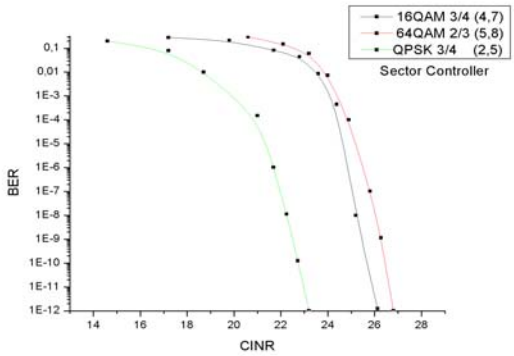


**Figure 3:** BER-CINR for QPSK, 16QAM, 64QAM for the Sector Controller (BS)

It is worth noting, that the measurements were taken for a time period of 20 minutes. That would provide the lowest BER ($10^{-12}$) as after that period the errors were occurring more frequently. Also, in the real measurements we cannot have the exact value that CINR is zero.

The experimental results are in accordance with the theory since from the curves it is seen that for a good link, meaning high values of CINR, the optimal modulation with the least bit error rate is the 64 QAM. On the contrary, for low CINR values we can achieve better BER with a lower modulation, as QPSK. This behavior relates to the fact that QPSK is a low spectral efficiency modulation and thus requires a lower energy level per bit and it is, thus, more energy efficient, as it is derived from the equation:

$$SNR = 10\log_{10}(N_b) + EBNR_{dB}, \text{ [3]}$$

where $N_b$ is the number of bits per symbol for the modulation scheme and $EBNR_{dB}$ is the Energy per Bit to Noise Ratio.

The importance of these curves is that we can specify an acceptable BER value and this value will define the required CINR. So, for a fixed BER value, the 64QAM modulation scheme requires better CINR.

## 3. Throughput measurements

In this part of the paper we investigate the performance, in terms of throughput, of such a system, deployed in the 3.4 - 3.6GHz licensed band with 3.5, 7 and 14MHz channel bandwidths, considering the Orthogonal Frequency Division Multiplexing transmission scheme with 256 points FFT and Time Division Duplex.

We present the throughput measurements at the test sites of our choice for LOS and NLOS operation conditions and a fixed modulation scheme, 64QAM with ¾ FEC code rate. The quality of the link between the BS and the SS, as this is indicated by the RSSI and CINR for the LOS scenarios, was maintained to a satisfactory level that was varied between -55 dBm to -85 dBm and 20 dB to 30 dB, respectively for 23 dBm BS Tx power and 16 dBm SS Tx power. On the other hand for the NLOS scenarios, an effort to achieve the optimum link quality, based on the geographical area density, was made and achieved by properly adjusting the orientation of the BS antenna, where the values for RSSI and CINR maintained within the same limits as for the LOS case. It is worth mentioning at this point, that is some situations we experienced high interference from unlicensed transmittions in the frequency band allocated for our test purposes, indicated by the increased number of CRC errors measured. To avoid this problem we had to change the carrier frequency so that we get an interference free channel. Nevertheless, in all NLOS scenarios, the antennas of the BS and SS were facing the same building and the angle between their beamwidths was varied from a few degrees up to $45^0$.

Our test measurements were performed using an open-source software package, MGEN, where traffic of 40 Mbps between the BS and SS was generated. The resultant UDP uplink and downlink throughputs are shown in Figures [4-7].

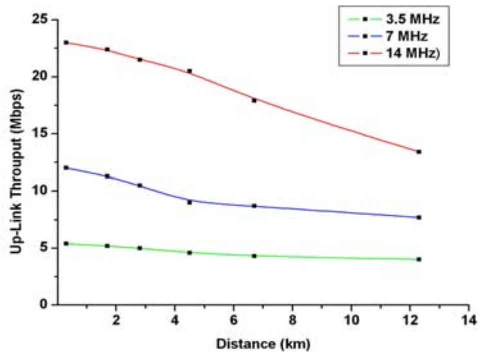


**Figure 4:** UDP Uplink Throughput Vs Distance for LOS scenario.

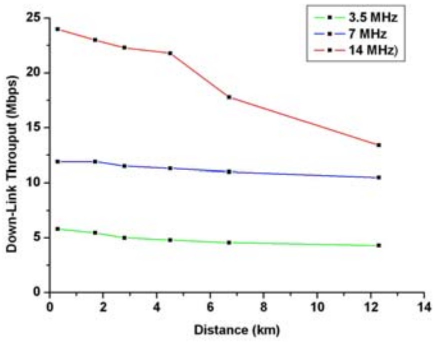


**Figure 5:** UDP Downlink throughput vs Distance for LOS scenario.

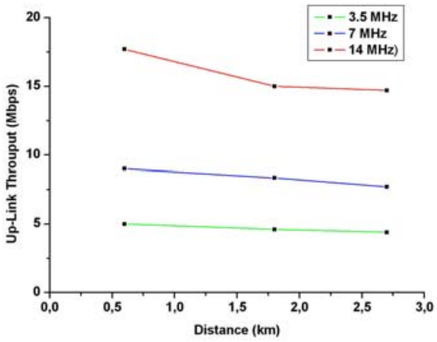


**Figure 6:** UDP Uplink Throughput vs Distance for NLOS scenario.

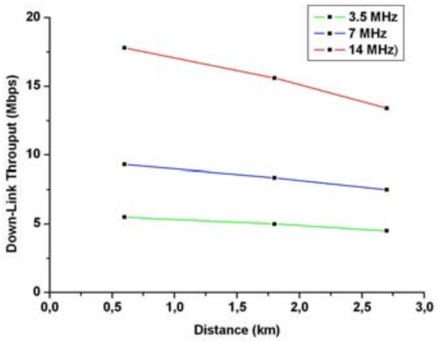


**Figure 7:** UDP Downlink Throughput vs Distance for NLOS scenario.

## 4. VoIP and Wimax

The Wimax broadband wireless access model is aimed at the delivery of high data rates to the end user. That is why we choose to evaluate the Quality of common Services, over 802.16d 2004. In order to be able to cope with the demand of Voice over IP (VoIP) or video streaming, the wireless link must meet strict limitation in delay and jitter [4]. In order to do that we need to configure the QoS of the corresponding Service Flow at the SS at the best available setting provided by the vendor in order to reserve the best priorities according to the 802.16 MAC layer for the video or VoIP streams.

In our measurements, we tested the performance of VoIP for several bandwidth profiles. More specifically we conducted experiments using videophones with H.263 video codec using bitrates of 64Kbps, 128Kbps, 384Kbps and 512Kbps and also with Cisco VoIP Phones and the use of G729.r8 and G711ulaw as voice codecs and varying RTP Payloads [5]. We used SER (SIP Express Router as SIP proxy server and a Cisco 2611 as a voice gateway to the PSTN network through an E1 TDM interface). The architecture that was used for the testing is shown in Figure [8].

| *Codec* | *BR* | *NEB* |
|---|---|---|
| G.711 | 64 Kbps | 87.2 Kbps |
| G.729 | 8 Kbps | 31.2 Kbps |

**Table 1**: Bit rate and Nominal Ethernet Bandwidth (one direction) for common voice codecs

By setting the bandwidth on the VoIP RTP flow near the required call bandwidth, we can reach the saturation limit could be reached in order for the service to be in acceptable levels. In the case of a LOS link with good ear link parameters (CINR, RSSI), VoIP is working with delays below 50 msec which is acceptable compared to the near 150 msec upper delay limit. In cases of NLOS link though, the delay and packet loss may vary and if the VoIP packetization used is larger than 80 Bytes, then voice loss is experienced between the calling parties.

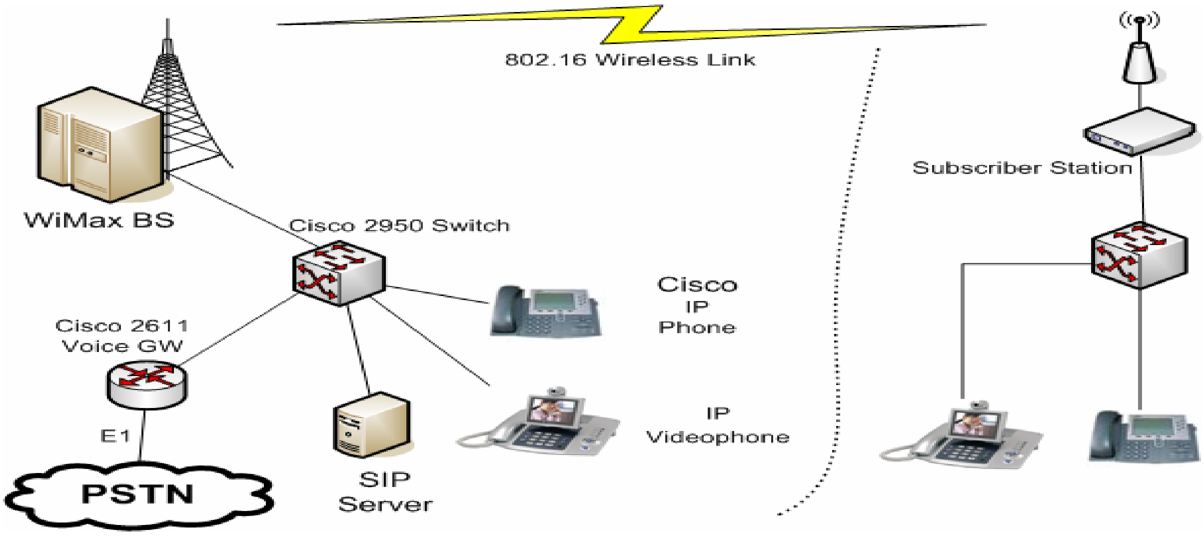


**Figure 8**: VoIP Testing Architecture

## 5. Conlusions

In this paper a series of throughput data have been conducted with a Wimax system operating at 3.5 GHz frequency and at channel bandwidths of 3.5 MHz, 7MHz and 14 MHz. These data show that a Wimax system is a very promising one for delivering broadband services to fixed and mobile users. It is shown how low BER can be achieved in a wireless link by adjusting the RF parameters of the transmitted and received power. The paper also shows methods and examples for achieving a good quality VoIP communication in a Wimax link.

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