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# A Novel Mobile WiMAX Solution for Higher Throughput

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*Abstract*— **The IEEE 802.16 standard, also known as WiMAX, has emerged as an exciting technology for broadband wireless communications with potentials to offer high throughput and support high bandwidth demanding applications. WiMAX, however, has yet to prove its effectiveness when the end terminals are not fixed and have the capacity to move from one place to another at different speeds. Recent studies suggest that while WiMAX (802.16e) has the potential to deliver a data rate up to 75 Mb/s for fixed wireless communications, it fails drastically for mobile wireless communications, often providing a data rate less than 1 Mb/s when the mobile nodes travel at high speeds, which offers a huge challenge for QoS management. Multipath fading that causes high bit error rate at the receiver end is a key reason for low throughput at high speed. Bit error rate and maximum packet size determine the packet error rate, and error recovery for higher number of corrupted packets is not always an attractive option for many real-time applications with delay and jitter constraints. In this paper, we propose a mathematical model to estimate the bit error probability when the mobile station travels at different speeds. The estimated value of bit error probability is then taken into account to proactively compute the appropriate maximum packet size that offers the best chance to achieve improved throughput at different operating conditions. We simulated the proposed scheme for a centralized video surveillance system in a public train where the train is the mobile node and sends real-time video data to the base stations. The results show that the proposed scheme achieves significantly higher throughput and lower jitter compared to other standard schemes.**

***Keywords*- Wireless communication, WiMAX, throughput, bit error rate.**

## I. INTRODUCTION

The IEEE 802.11 standard [1] has been a huge success both in terms of wide spread penetration and commercial interests for wireless local area networks (WLAN). Following the success of the IEEE 802.11, the need for further improvement in transmission rate and Quality of Service (QoS) was realized, which ultimately leads to the introduction of another set of standards, termed as the IEEE 802.16 [2] for wireless broadband networks. The IEEE 802.16, also commonly known as WiMAX, is designed to support high scalability, rapid deployment and high speed data rate of up to 75 Mbps for fixed wireless metropolitan access networks (MAN). The IEEE 802.16-2004 standard, previously known as 802.16d or 802.16-REVd, was published for fixed access and later on in October 2005, the standard was updated and extended to the IEEE 802.16e to support mobile access [3, 4]. The 802.16e uses scalable orthogonal frequency division multiplexing (OFDM) and multiple antenna support through multiple input multiple output (MIMO) communications. The802.16e offers benefits in terms of coverage, power savings, frequency usage, higher bandwidth support and QoS. Considering the capacity and QoS aspects of the IEEE 802.16e, the standards have wide spread applicability for wireless communications with mobility supports for the end users. The ATM-like guaranteed QoS offered by the IEEE 802.16e is highly suitable for voice, video and data applications in mobile end nodes. The future of the wireless internet, video surveillance, video conferencing and video on demand in public transports as well as high quality video and voice conversion highly depends on the successful deployment of the IEEE 802.16e standards.

Although the theoretical capacity of the IEEE 802.16e is 75 Mbps over a 112.6 km range, this capacity is only valid for an ideal situation and in practice, the IEEE 802.16e can support up to 10 Mbps at around 10 km for the line-of-sight range. This capacity further drops significantly when the users are on the move. The other feature of the IEEE 802.16e is that available bandwidth is shared among many users in a given radio sector, which essentially provides much lower bandwidth to the individual user. Lower and unreliable throughput at high speed offers a huge challenge for QoS management. The low data rate at high speed is mainly caused by fading of radio signal when the end terminals move at high speed and the carrier frequency is in the lower range. Multipath fading at high speed causes signal noises at the receiver end, which results in higher bit and packet error rate, limiting the effective data transmission capacity. For error prone channels as such, packet size has strong impacts on overall throughput because higher levels of received signal error cause higher number of corrupted bits and larger packets have higher corruption probability than smaller packets. Smaller packets however, have higher overheads and are inefficient when the level of received signal error is low. An appropriate packet size under different error scenarios has significant impact on the overall throughput, which motivates the research work presented in this paper.

Over the past decade, researchers have proposed many innovative techniques to address the problem of low throughput in mobile wireless communications. In [5], Li *et al.* showed the impact of channel noises on transmission control protocol (TCP) performance for the IEEE 802.11 wireless communications. Although TCP is a well studied technique for the Internet, it has insignificant and sometimes detrimental impacts on many real-time applications where the communication channel is not entirely reliable. This is because TCP involves a lot of packet retransmissions, reducing the overall throughput and introducing unacceptably high jitter for real-time applications that often operate with low delay and jitter constraints. Makela *et al.* [6] provided analytical results on average throughput for the IEEE 802.11 multi-rate wireless communication when a mobile terminal goes far away from an access point. Their work however, is restricted to multi-rate mobile data communication frequently used for tactical communication. Aramvith *et al.* [7] proposed a conditional retransmission strategy for corrupted packets of real-time applications with low delay constraints in wireless communications. They proposed a low delay interleaving scheme that combines the video encoder buffer with the interleaving memory so that interleaving does not increase the delay and memory in the video encoder. This scheme is particularly suitable for low bit rate low delay video with the H.263 standard. Chen *et al.* [8] proposed an adaptive error control scheme to adaptively insert error– resilience features into a compressed video for error prone channels. The key idea was to determine the retransmission schedule based on the impact of packet loss. Wang *et al.* [9] developed a new model to follow the statistical nature of bursty packet error sequences encountered in digital wireless communication. For WiMAX, there is an adaptive modulation and coding scheme [10] that changes the modulation scheme and hence the transmission rate in response to a drop in signal quality. This however, does not address the problem of higher packet corruption probability when the packet size is large and the mobile node travels at high speeds.

In this paper, we propose a mathematical model for the IEEE 802.16e to compute the appropriate packet size under different error scenarios when a mobile node travels at various speeds. The novelty of the paper is two folds – first we show how to estimate the bit and packet error probability in WiMAX in proactive fashion when the mobile node travels at different speeds and secondly, how to use this information to compute the appropriate packet size so that the packet error rate at the receiver end remains within an acceptable bound and the overall throughput improves. The motivation for this research is derived from the observation that usable data rate is very low in mobile WiMAX communications and reactive error control schemes (e.g., TCP) are very costly in these scenarios because the system can not afford the extra overheads caused by the reactive protocols when the usable data rate is already very low. In mobile WiMAX communications, any reactive scheme often requires multiple attempts to send information to the mobile nodes because of high error rate. Also when the mobile nodes move at various changing speeds (e.g., a train is accelerating fast from a stop), information provided by the reactive schemes are often obsolete in context of the current changed condition of the mobile nodes.

## II. PROPOSED SCHEME

The key idea of the proposed scheme is to first compute an estimated bit error probability as the speed of a mobile terminal changes. Based on that estimated bit error probability and desired packet corruption rate set by the network provider, the maximum packet size that is likely to yield lower packet corruption rate and thereby offer higher throughput, is then calculated. In Section 2.1, we show how to calculate the estimated bit error probability at various mobile terminal's speeds and then in Section 2.2, we show how to calculate the maximum packet size as the speed changes.

### 2.1 *Bit Error Rate and Mobile Terminal's Speed*

For multipath fading, Rayleigh fading [11][12] is proven to be an excellent model that can emulate the error in radio signal when there are many objects in the environment scattering the radio signal before the receiver receives the signal. Now let us consider a wireless communication channel that is characterized by the parameters: $N$ be the number of OFDM sub carriers, $f_m$ be the Doppler frequency where $f_m = f_c(v/c)$, $v$ be the speed of mobile nodes, $c$ be the speed of light, $f_c$ be the carrier frequency, $T_s$ be the duration of each M-ary QAM symbol, $E_s$ be the average symbol energy, $E_b$ be the average energy per bit, $N_o$ be the noise energy, $\gamma_b$ be the received bit-energy-to-noise ratio, $\overline{\gamma_b}$ be the average received bit-energy-to-noise ratio, $\gamma_s$ be the received symbol-energy-to-noise ratio, $\overline{\gamma_s}$ be the average received symbol-energy-to-noise ratio, $P_b(\gamma_b)$ be the probability of received bit error, $P_P$ be the packet error probability. Following the Rayleigh fading model [12], we can express the density function of received symbol energy to noise ratio as

$$p_{\gamma_s}(x) = \frac{1}{\overline{\gamma_s}} e^{-x/\overline{\gamma_s}}, x \geq 0 \tag{1}$$

and the average symbol error probability for a such channel can be expressed as

$$P_M = \int_0^{\infty} P_M(x)\, p_{y_s}(x)\ dx \tag{2}$$

where the average received symbol-energy-to-noise-ratio is given by

$$\overline{\gamma_s} = \frac{1}{1 - \frac{1}{N^2}[N + 2\sum_{i=1}^{N-1}(N-i)J_0(2\pi f_m T_s i)] + \frac{NT_s}{\frac{E_s}{N_0}}} \tag{3}$$

Now the average received bit-energy-to-noise ratio $\overline{\gamma_b}$ can be derived from the average received symbol-energy-to-noise ratio $\overline{\gamma_s}$ according to the following equation:

$$\overline{\gamma_b} = \frac{\overline{\gamma_s}}{\log_2 M} \tag{4}$$

where $M$ is the number of symbols for $M$-ary QAM modulation scheme. For 16-QAM, M equals 16 and for QPSK M is 4. Combining Eq. (3) and (4), we can express $\overline{\gamma_b}$ as

$$\overline{\gamma_b} = \frac{\frac{1}{\log_2 M}}{1 - \frac{1}{N^2}[N + 2\sum_{i=1}^{N-1}(N-i)J_0(2\pi f_m T_s i)] + \frac{NT_s}{\log_2 M}(\frac{1}{\frac{E_b}{N_0}})} \quad (5)$$

For a fixed channel with unchanged values of $N$, $NT_s$, relatively constant $E_b / N_0$ over time and a known modulation scheme, the main source of error for a mobile terminal is the velocity $v$ that has direct impact on $f_m$, which in turn influences received symbol/bit energy to noise ratio. According to the equation, higher will be the velocity, lower will be the received symbol/bit energy to noise ratio. For OFDMA air interface, the main source of bit error is the inter-carrier interference instead of interference between the OFDMA users. Additive white Gaussian noise (AWGN) [12] is often used to successfully approximate the OFDMA inter-career interference. Taking AWGN into consideration, we can derive the bit error probability as

$$P_b = Q(\sqrt{2\gamma_b}) \quad (6)$$

where bit error probability $P_b$ and symbol error probability $P_M$ of OFDM are related to each other in the form of

$$P_b \approx \frac{P_M}{\log_2 M} \quad (7)$$

For M=2 or M=4, the average bit error probability is given as

$$\begin{aligned} P_b &= \int_0^{\alpha} P_b(x)\, p_{\gamma_b}(x)\, dx \\ &= \frac{1}{2}[1 - \sqrt{\frac{\overline{\gamma_b}}{1+\overline{\gamma_b}}}] \end{aligned} \quad (8)$$

For a scenario where carrier frequency $f_c$ is 2.6 Ghz, bandwidth is 12 MHz, number of sub carriers $N$ equals 2048, symbol period equals 149.33s, modulation scheme is QPSK and $E_b/N_0$ equals 5 dB, the above expressions lead to a relationship between the average bit error probability and mobile terminal's speed that can be depicted as shown in Figure 1. The Figure demonstrates that higher the mobile terminals' speed, higher is the average bit error probability. Higher bit error probability significantly increases the probability of error in a received packet when the packet size is large. Conversely, when the bit error probability is low, larger packet size reduces the overheads of extra header bits, achieving higher throughput. From packet error perspective, it is therefore highly important to calculate an optimal packet size in response to various mobile end's speeds so that the overall packet error rate at the receiving end does not the cross an acceptable range.

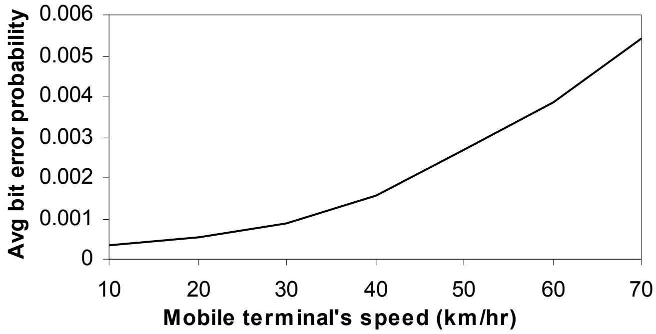


Figure 1: Average bit error probability at various mobile terminal's speed.

## 2.2 *Calculation of the Maximum Packet Size*

The relationship between bit error rate and packet error rate for a packet with $K$ bits can be expressed as:

$$P_P = 1 - (1 - P_b)^K \quad (9)$$

From Eq. (9), for a desired packet error rate of $P_P = \lambda$, set by the network provider, we can derive the optimal maximum packet size $K$ as

$$K = \frac{ln(1-\lambda)}{ln(1-P_b)} \quad (10)$$

A number of considerations can influence the network providers' choices of desired packet error rates. A key consideration is the error recovery mechanisms the network provider may employ. Preferred packet error rate may also depend on the type of applications based on their level of quality of service (QoS) requirements [5]. The overall algorithm for the proposed method to compute the maximum packet size at various mobile terminals' speed is given as

Procedure *compute_packet_size* ($v$, $\lambda$)

begin

avg_ber_prob = Find estimated average bit error probability at mobile terminals' speed $v$ from the relationship as depicted in Figure 1 obtained by using expression (8).

$K$ = compute maximum packet size based on the estimated avg_ber_prob using expression (10)

return $K$

end *compute_packet_size*($v$, $\lambda$)

When the mobile terminal is the transmitter, the velocity information is readily available to compute the packet size using the procedure *compute_packet_size*. When the mobile nodes are the receiver, the proposed scheme can be readily applied for many systems (e.g., fully automated public trains) where the speed information is known in advance and the transmitter can use this knowledge while using the procedure *compute_packet_size*. For mobile receivers that are highly dynamic in nature (e.g., bus, car), the proposed system relies on message exchanging system used in WiMAX. In WiMAX, the transmitter and the receiver exchange messages at intervals to maintain the connection and the required QoS [13, 14] and the proposed scheme needs to collect the speed information from these messages.

## III. SIMULATION RESULTS

The simulation is conducted in NS2 for a centralized real-time video surveillance system in a train as shown in Fig. 2. The train is equipped with 4 video cameras; each of them sending video data at a rate of 512 Kbps to the base stations using WiMAX technology. The video data are then sent through a wired optical communication network to a central control room where the security experts interpret/monitor the video contents and take actions accordingly. The maximum data rate capacity of the wireless channel is 10 Mbps and carrier frequency was 2.6 GHz with a bandwidth of 12 MHz, number of sub-carries is 2048 and the modulation style is QPSK. For the simulation scenario, the train moves from a stop and gradually increases its speed, reaching a speed of 40 km/hr at 20 sec and a top speed of 70km/hr at around 60 sec time. The train continues to cruise at that speed before the train starts to slow down at 110 sec. The train continues to slow down and finally stops at the next railway station at 180sec. We monitored the overall throughput and actual video data received by the base station while the train was moving.

Figure 3a shows the comparison of received bit rate at the receiver end in the proposed scheme, and the standard 64 bytes and 128 bytes schemes. The figure shows that when the mobile node moves at a higher speed, the standard scheme achieves nearly the same data rate compared to the proposed scheme. This is because at high speed the bit error rate is significantly higher and smaller packet size offers higher chance of uncorrupted packet arrivals at the receiver end. The standard schemes however, fail very badly when the speed is low because the capacity of actual transmission is reduced due to the extra overheads caused by higher number of packets and extra header bits. While the proposed scheme maintains the same level of throughput at high speed, at low speed it comfortably outperforms the standard 64 bytes and 128 bytes schemes. Standard scheme with 256 bytes packet size performs comparably against the proposed scheme at low speed, but suffers badly at high speed as shown in Figure 3b. Standard scheme with 512 bytes packet size has disastrous throughput at high speed compared to the proposed scheme as evident in Figure 3b. This confirms that the proposed scheme offers significant advantage over the standard schemes by adjusting the packet size in response to changing speeds. We also monitored the received video data by stripping out the header bits and discarding protocol related packets. The results are presented in Fig. 4 that confirms that the proposed scheme outperforms the standard 64 and 128 bytes scheme by a margin up to 400 Kbps and 200 Kbps, respectively at low speed while the proposed scheme achieves the same throughput at very high speed. The proposed scheme also outperforms the standard 256 and 512 bytes schemes. The proposed scheme considers the bit error probability and then carefully selects the appropriate packet size in order to reduce the number of

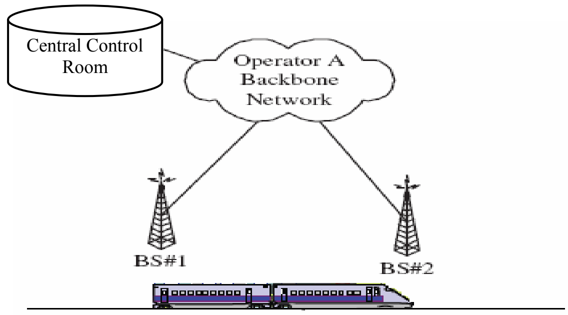


Figure 2: Simulation scenario of a real-time video surveillance system on a public train.

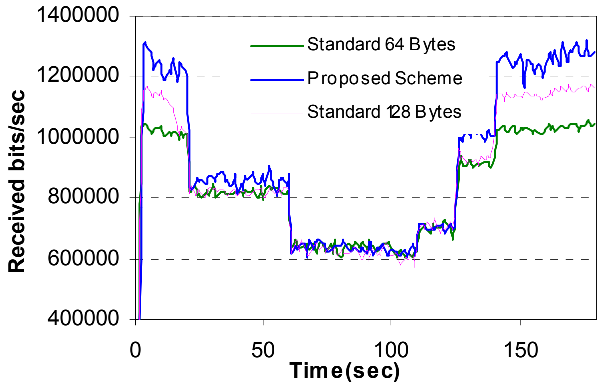


Figure 3a: Comparison of received bits in the proposed, standard 64 bytes and standard 128 bytes schemes .

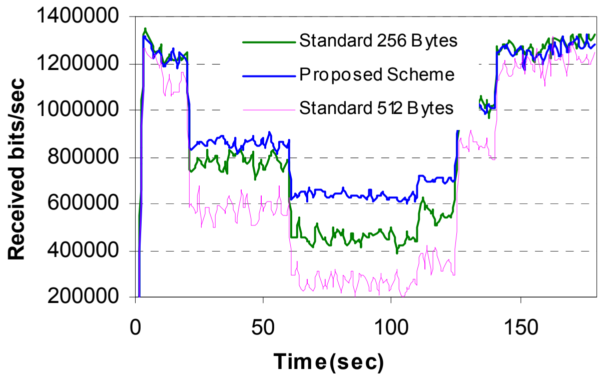


Figure 3b: Comparison of received bits in the proposed, standard 256 bytes and standard 512 bytes schemes .

corrupted packets at the receiver end. Increased numbers of uncorrupted packets offer improved jitter and simulation results suggest that the proposed scheme outperforms standard schemes in terms of delay jitter and end-to-end delay.

## IV. CONCLUSION

Although WiMAX is a promising technology for fixed wireless MAN, its effectiveness is limited when the wireless mobile nodes have the freedom to move at various high speeds. Multipath fading that causes high bit error rate at the receiver end is a major reason for low throughput in mobile WiMAX communications, especially for low frequency carrier. Bit error rate and packet size determine the packet error rate and error recovery mechanism fails when the number of corrupted

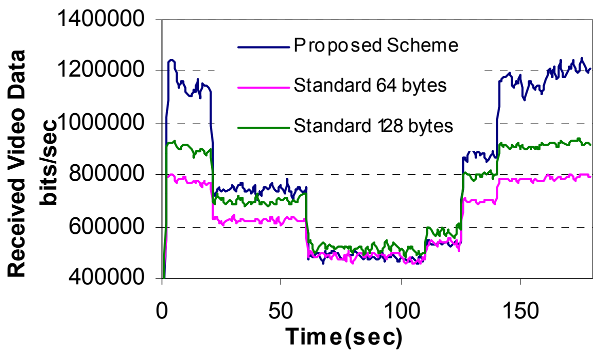


Figure 4a: Comparison of received video data in the proposed, standard 64 and standard 128 bytes schemes.

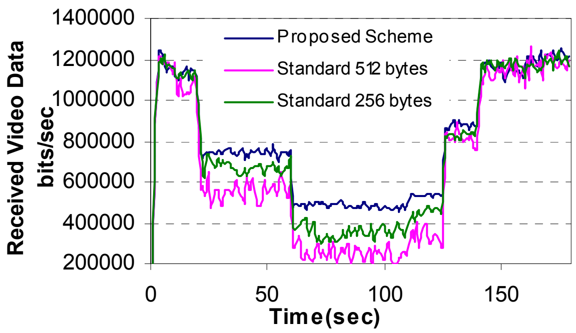


Figure 4b: Comparison of received video data in the proposed, standard 256 and standard 512 bytes schemes.

packets is very large and the application is sensitive to delay and jitter. In this paper, we proposed a new mathematical model to compute the maximum packet size based on the estimated value of bit error probability as the mobile terminals move at various speeds. We simulated the proposed and other standard schemes for a centralized video surveillance system in a public train. The results confirmed the suitability of the proposed scheme and suggested that the proposed scheme achieves significantly higher throughput and improved jitter compared to the other schemes.

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