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Abstract— **WiMAX has emerged as a promising wireless communication technology with potential to deliver high throughput and guaranteed quality of service to the end applications. Recent studies suggest that while WiMAX (802.16e) is capable of delivering a data rate of up to 75 Mbps for fixed wireless communications, data rate decreases drastically for mobile wireless communications, often providing a data rate less than 1 Mb/s when the mobile nodes travel at vehicular speeds. High bit error rate caused at high vehicular speeds is the key reason for low throughput. In noisy mobile communication environments, standard error control mechanisms like the transmission control protocol (TCP) has limited and often detrimental impacts on the overall throughput because of the excessive retransmission overheads. To address this issue, WiMAX standard incorporates forward error correction (FEC) mechanism that eliminates the need for retransmissions. In FEC, extra parity bits are added to the original message to recover the corrupted information. Adaptive FEC that adjusts the size of extra parity bits in response to packet retransmission requests is an enhancement over standard FEC that uses fixed block of party bits. Existing adaptive FEC schemes, however, have limited efficiency when the end terminal moves at vehicular speeds. In this paper, we propose a new FEC scheme that estimates and adjusts the size of extra parity bits to suit the channel conditions. We apply the concept of interval based data sampling to address the dynamic nature of communication environments at high vehicular speeds. We simulate the proposed scheme for a centralized live video surveillance system in a public train where the train is the mobile node and sends real-time video data to the base stations on train platforms. The results show that the proposed scheme achieves significantly higher throughput and lower jitter compared to other standard schemes.**

Keywords-Mobility; vehicular speed; throughput.

I. Introduction

Following the deployment and commercial success of the IEEE 802.11 standard [1], the need for further improvement in the wireless transmission rate and quality of service (QoS) has become apparent, which has lead to the introduction of another set of standards, being the IEEE 802.16 [2]. The new standard, also commonly known as WiMAX, is designed to support rapid deployment, high scalability and high data rate of up to 75 Mbps for fixed wireless metropolitan access networks (MAN). The earlier version of the IEEE 802.16 standard (802.16-2004), commonly known as 802.16d, was aimed at fixed access networks. In October 2005, the standard was upgraded and extended to the IEEE 802.16e to support mobile access [3, 4, 10]. The 802.16e incorporates scalable orthogonal frequency division multiplexing (OFDM) and multiple antenna support through multiple input multiple output (MIMO) communications, which leads to higher data rate and improved QoS. The 802.16e standard also offers larger coverage, power savings, better frequency usage, higher bandwidth support and provision for QoS, which makes WiMAX a strong contender for future mobile wireless communication standards.

The IEEE 802.16e is capable of offering data rates of up to 75 Mbps and range of up to 112 km. This capacity, however, is only valid for an ideal communication environment and in practice, the IEEE 802.16e can support data rate of up to 10 Mbps at distance of around 10 km for the line-of-sight range. Transmission rates further drop when the users are on the move. The other feature of the IEEE 802.16e is that many users in a large radio sector share the available bandwidth, which ultimately provides much lower bandwidth to an individual user. Lower and inconsistent throughput at high vehicular speeds offers a huge challenge for QoS management for many multimedia applications, particularly with tight delay and jitter constraints. Multipath fading at high vehicular speeds causes signal noises at the receiver end, which results in higher bit and packet corruption rate, limiting the effective data transmission capacity. For wireless channels at high mobile speeds, error recovery schemes like the TCP that attempts to address the problem of error in received packets by requesting retransmissions, have limited, if not detrimental, impacts on the actual data transmission rate because the channel can not afford the extra overheads caused by the reactive protocols like the TCP when the usable data rate is already very low. The TCP scheme may often require multiple attempts to send uncorrupted data to the receiver nodes because of high error rate.

Realising the limitations of the error recovery schemes like the TCP at noisy environment, WiMAX proposes to incorporate forward error correction (FEC) schemes into their standard. Error control mechanisms like FEC are particularly suitable for noisy wireless communication as they do not require retransmissions of packets. In FEC, extra parity symbols are added to a packet to recover the corrupted symbols in a packet. The correct number of extra parity bits is highly critical as unnecessary parity bits limit the actual data transmission while insufficient parity bits result in unrecoverable corrupted packets. In this paper, we propose a new FEC scheme that adjusts the FEC size more efficiently in mobile communication at high vehicular speeds.

II. Background

Researchers have been working for long to improve the FEC based error control mechanism. The major research interest remains how to make the FEC code size dynamic/adaptive instead of using a fixed FEC code under all communication environments. Bolot *et al.* proposed an adaptive FEC scheme [5] for the Internet Telephony that adjusts the code size according to an optimisation model based on the assumption that the audio packet loss in a network follows a Bernoulli process. Padhye *et al.* raised concern about the assumption and further improved the FEC scheme by employing methods to better control the amount of redundancy [6]. Yao *et al*. proposed a dynamic FEC scheme [7] for digital video broadcasting that dynamically adjusts the code size based on the assumption that error sequence generated by data transmission channels follows a Gilbert-Elliot model. The method of employing fixed models to determine the characteristics of wireless channel, works reasonably well for an environment, where the end nodes are fixed or have low mobility. For an environment that changes dynamically with time and speeds, finding an appropriate model is still a major research issue. Many researchers have proposed to use feedback loop to determine the changing channel conditions. Ahn *et al.* [8] proposed an adaptive FEC code control algorithm for sensor networks that dynamically tunes the code size based on the arrival of acknowledgment packets. In [9], Smadi *et al*. proposed an error recovery service protocol that adjusts the strength of FEC code depending on the notification of corrupted packets at the receiver end. All of the above mentioned research works are suitable for wireless communication where the end nodes are either static or move at very low speed. In a true mobile environment, where the end node moves at vehicular speeds, the communication environment changes very quickly (e.g., an electric train is accelerating from a platform to reach its top speed) and the existing FEC schemes fail to deliver higher network performance.

III. PROPOSED SCHEME

The key idea of this paper is to use interval based monitoring of channel conditions at the receiver end and notify the sender as the channel condition changes at various vehicular speeds, so that the sender can promptly adjust FEC size for the next interval such that it offers the best chance to correct the error without the need for retransmissions. The feedback loop FEC method as proposed in [9] that uses the request for packet retransmission as the feedback information to incrementally increase the strength of FEC size until the packet is delivered without an error, is not efficient in a true mobile environment as it will require too many retransmissions at noisy environments at high vehicular speeds, thereby limiting the overall data transfer rate. Interval based monitoring/sampling of data is used in many dynamic system [11] where it is hard to estimate the future data deterministically. In this paper, we apply this method to address the dynamic conditions of communication channels. Other innovations of this work include how to adapt the sampling interval as the speed changes and what feedback information to use and how to use that information to determine the near appropriate FEC code size.

Let us consider a wireless communication scenario (Fig. 1) where a live video surveillance system is installed in a train and real-time video data need to be uploaded at various vehicular speeds. Video data will then be sent to a central control room where the security experts will analyse the contents. As the train moves, radio signals are reflected from various

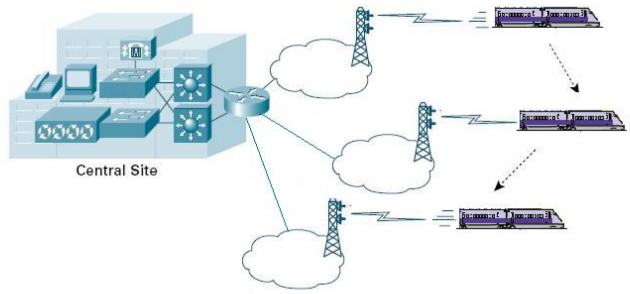

Figure 1: Wireless communication for a video surveillance application on trains.

environmental elements and arrive at the receiver end at different times, causing multipath fading. Due to fading, received bit error rate increases, which ultimately causes many packets to be received as corrupt. Bit error and packet corruption rate increases as the multipath fading becomes more severe when the speed of the moving train increases (no-FEC in Fig. 2). Reed-Solomon (RS) [9] is an efficient FEC code and used in many wireless systems. With RS code, error in a received packet can be recovered as long as the number of the received symbol errors does not exceed half the size of FEC code size. Packet corruption at the receiver end, therefore, can be avoided, if the sender has the information about the amount of received error at the receiver end at current speed and uses the correct FEC size that suits the current environment. Feedback information in the form of bit error rate can assist the sender to adjust it FEC size. However, excessive feedback information (e.g., piggybacking) can cause extra overhead and reduce the overall transmission rate.

Now, let us assume that at a time t_i, the train is cruising at a speed of v_i and the monitoring/sampling interval is set as l_i. At the end of each sampling interval, the FEC size is updated based on the data recorded in the last interval. The sender is then informed about the updated FEC code size and the sender uses that code size for the next interval. A fixed sampling interval is not suitable in all environments because when the train moves at a higher speed (e.g., the train is accelerating to reach its top speed), the channel condition changes more rapidly. Under this scenario, a shorter interval is more suitable as a longer sampling interval records data for environments that may significantly differ from current environment (i.e., within 10 sec a train can reach 100 km/hr and hence a large change of environment). Conversely, when the train moves at a slower speed or stops at a station, channel condition is relatively deterministic and a shorter sampling interval provides unnecessary/redundant feedback information, causing extra overheads on the system. Therefore a longer interval will be more suitable when the speed is in the lower range. We, therefore, propose the sampling interval l_i to adjust as the speed of the train changes. The sampling interval l_i can be modelled as:

$$l_i = a + (b - a)(1 - \overline{v_i}) \tag{1}$$

where, a and b indicate the minimum and maximum length of sampling interval and represents the normalized speed at time t_i relative to the maximum allowed speed. Eq. (1) ensures that the sampling interval is updated in response to the different environments, caused by changed speeds. At the end of a sampling interval l_i, the receiver records N_i as the total number of received packets and C_i as the number of received packets with error(s). The packet corruption rate is, therefore, available as:

$$P_p(i) = C_i / N_i \tag{2}$$

In C_i packets, the ratio of bit error rate is recorded as $P_b(i)$ where

$$Pb(i) = E_i / D_i \tag{3}$$

Here, E_i and D_i represent the number of erroneous bits and total bits in C_i packets, respectively, received by the receiver end in the last interval l_i. Now for an m bit symbol, the relationship between P_b and average symbol error probability S_p can be expressed as:

$$S_P(i) = 1 - (1 - P_b(i))^m \tag{4}$$

For a packet comprising K symbols, number of estimated corrupted symbol T_i can be computed as:

$$T_i = K \text{ x } S_p(i) \tag{5}$$

For RS code to be used in the next interval l_{i+1}, the number of parity symbols can be given as:

$$(n-K)_{i+1} = 2\text{x}T_i = 2K \text{ x } S_p(i) \tag{6}$$

For priority differentiation and network provider's flexibility, Eq. (6) can be modified as:

$$(n-K)^r_{i+1} = \alpha \text{ x } 2K \text{ x } S_p \tag{7}$$

where, α ($\geqslant 1$) is the strength parameter of the RS code. The network provider can tune the strength parameter α for different levels of error recovery capabilities depending on the types of applications. For real-time application, the strength should be higher and for offline applications the strength can be set low.

The overall algorithm can be given as:

Procedure update_FEC_size ($\overline{v_{i+1}}$)

begin

FEC_size = compute the RS code size for the next sampling interval using expression (7) and send it to the sender.

l_{i+1} = update the sampling interval using expression (1).

end update_FEC_size($\overline{v_{i+1}}$).

The update_FEC_size will be revoked at the end of each sampling interval (i.e., only when current time $t \geqslant (t_i + l_i)$, where l_i was determined at time t_i).

IV. SIMULATION RESULTS

The simulation was conducted in ns-2 [12] for a centralised real-time video surveillance system in a train as shown in Fig. 1. The train is equipped with 4 video cameras; each of them uploading video data at a rate of 512 Kbps to the base stations using WiMAX technology. The video data are then transmitted through a wired optical communication network to a central control room where the security experts observe/interpret the video. The maximum data rate capacity of the wireless channel was set as 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. The maximum and minimum window of sampling interval was considered as 20 and 2 sec, respectively. In 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 station at 180sec. We monitored the received data and jitter at the base station while the train was moving.

Figure 2 shows the comparison of received bit rate at the receiver end in the original scheme (without FEC termed as no-FEC in this section) and existing scheme [9] for 256 bytes packet size. It is evident that the received data rate in the no-FEC scheme falls drastically as the speed of the train increases. This is because packet corruption rate increases sharply due to bit error rate caused by multipath problem at higher speed. Existing scheme that adjusts the size of FEC code size in response to subsequent packet retransmission requests, provides higher data rate compared to the no-FEC scheme. The improvement, however, is not consistent and follows a zigzag pattern, mainly because, in the existing scheme. FEC code size is incrementally increased, following subsequent notifications of corruption of the same packet. By the time, the sender increases its FEC size, many packets are already received as corrupted at the receiver end, resulting a drop in throughput. Once the packet is delivered good, FEC code size returns to its initial value, causing the same cycle to repeat at a noisy environment. Figure 3 shows that the proposed scheme addresses this problem quite well and provides consistently higher throughput even at high vehicular speeds. At some stages, the improvement in the proposed scheme is in the range of 800 Kbps. This is highly suitable for real-time multimedia applications (e.g., video surveillance) that operate with delay and jitter constraints and allow no retransmission.

Figure 4 shows the received data rate for 512 bytes packet size and the proposed scheme maintains its superior performance compared to others. It is also evident that larger packet size offers higher throughput at lower speeds because of smaller overheads, but as the speed increases and error rate increases smaller packet size (256 bytes in this case) offers better performance. This is because at noisy environments, larger packet size has higher corruption probability. The proposed scheme, however, efficiently considers the changing environments and generates valuable feedback information for the sender so that the FEC size reflects the latest

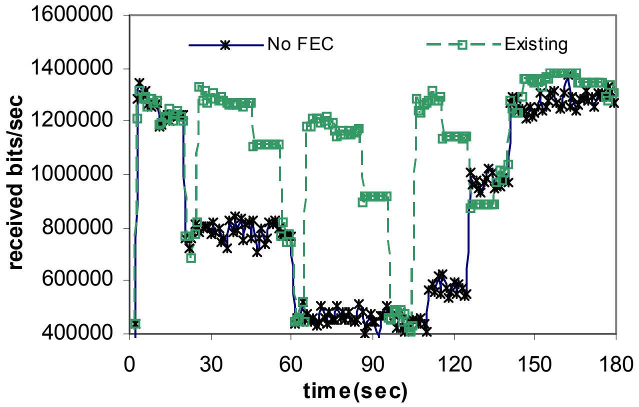

Figure 2: Comparison of received bits in no-FEC and existing FEC schemes for 256 bytes packet size.

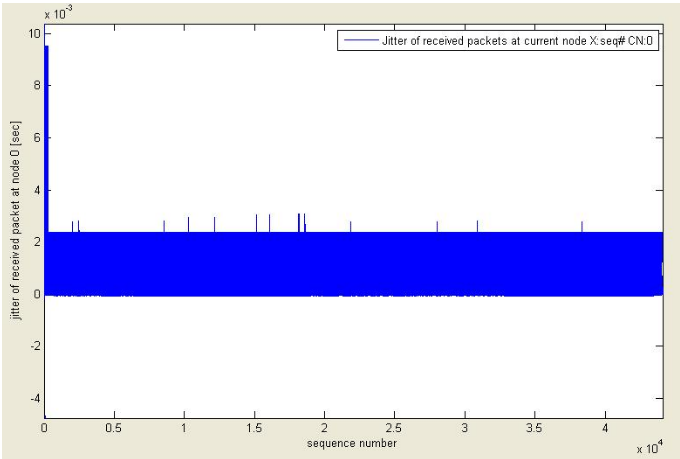

Figure 5(a): Jitter of the received packets (sec) in the proposed scheme.

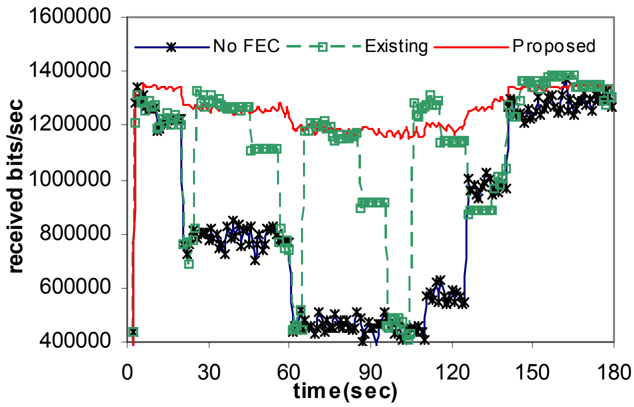

Figure 3: Comparison of received bits in no-FEC, existing FEC, and proposed scheme for 256 bytes packet size.

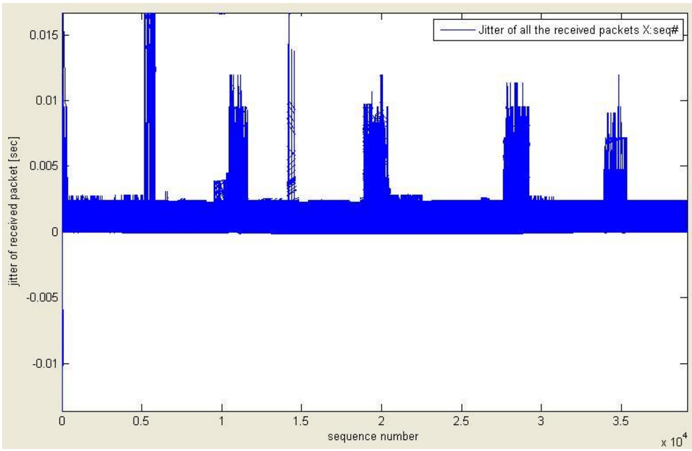

Figure 5(b): Jitter of the received packets (sec) in the existing scheme.

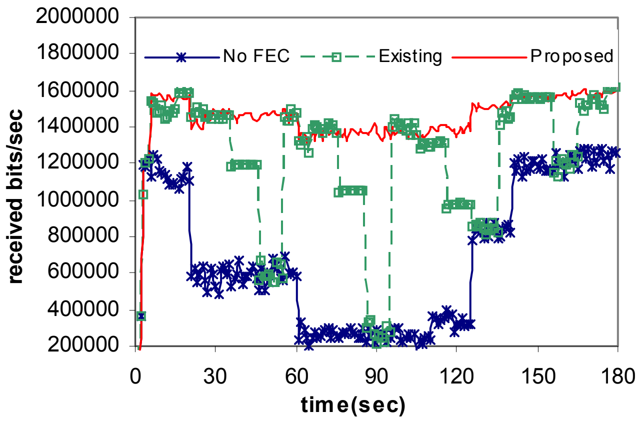

Figure 4: Comparison of received bits in no FEC, existing FEC, and proposed scheme for 512 bytes packet size.

communication environment. The benefit is quite evident as the data rate remains fairly high even at high speeds.

We also captured the delay jitter in different schemes and the results are shown in Fig. 5. Delay jitter for most of the packets in the proposed scheme remains within 2x10-3 sec range (Fig. 5a) whereas delay jitter in the existing varies significantly often reaching up to 0.015 sec (Fig. 5b). This is because in the existing scheme, higher numbers of packets are received as corrupted at the receiver end. In the proposed scheme, packet corruptions are in the lower range, which provides lower jitter.

V. CONCLUSION

Achieving high data rate in mobile wireless communication at high vehicular speed is a major challenge, and existing wireless technologies, including WiMAX, fail to deliver high data rate at 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 lower frequency carrier. Bit error rate and packet size determine the packet error rate, and error recovery mechanisms are of little assistance when a large number of packets are corrupted and the application is sensitive to delay and jitter. In this paper, we proposed a new FEC scheme to address the dynamic nature of mobile communication environments caused by high vehicular speeds. The new scheme employs interval based monitoring and sampling of received packets to compute the FEC size that best suits the current channel conditions. We simulated the proposed and other standard schemes for a centralized video surveillance system in a public train. The results confirm the suitability of the proposed scheme and show that the proposed scheme achieves significantly higher throughput and lower jitter compared to other schemes.

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