國立交通大學

電子工程學系電子研究所碩士班

碩 士 論 文

可調適視訊壓縮串流在 WIMAX 之低視訊延遲回報機制設計

A Low Video Latency Feedback Scheme for SVC Streaming in WIMAX MAC

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A Thesis
Submitted to Department of Electronics Engineering & Institute of Electronics
College of Electrical and Computer Engineering
National Chiao Tung University
in Partial Fulfillment of the Requirements
for the Degree of
Master
in
Electronics Engineering

August 2008

Hsinchu, Taiwan, Republic of China

中華民國 九十七年八月

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摘　　要

可調視訊編碼(SVC)是一種可以針對播放設備以及網路頻寬，彈性的調整傳送出去位元流的影像壓縮技術，微波存取全球互通(WIMAX)是無線通訊上的新技術，我們的研究目的在於結合可調視訊編碼與微波存取全球互通的特性，在即時視訊傳送端提出一個系統上跨層的設計方法，以減少頻寬變動與無線通道品質的干擾，讓可調視訊編碼的播放端可以流暢的播放視訊影像。

在本論文中，我們提出一個跨層的設計架構以及提出一個頻寬回報機制，這個頻寬回報機制將會根據最新的頻寬訊息，決定出一個最適合未來一段時間的估計頻寬，回報給可調視訊編碼應用層，進而調整所要傳送的位元流。利用我們所提出的跨層架構以及頻寬回報機制來傳送即時視訊影像,將會在播放延遲、頻寬使用率以及資料緩衝區所累積的資料大小有著不錯的表現.

A Low Video Latency Feedback Scheme for SVC Streaming in WIMAX MAC

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ABSTRACT

Scalable video coding (SVC) is a video compression technique which provides flexible bitstream extraction according to device types and network bandwidth, and WIMAX is a new wireless communication technique. Combining the two techniques to transmit multimedia information is an important application in the future. Our research goal is to reduce the impact of the variation of network bandwidth and channel condition on real time video communication based on above newly developed techniques.

In our thesis, we propose a cross-layer architecture between SVC system and WIMAX MAC, and we also propose a mechanism to decide the bitrate of the extracted SVC bitstream. According to the latest bandwidth information, this mechanism could decide a proper bitrate to fit the bandwidth of a period in the future, and then feedback this bitrate to SVC extractor. Thus SVC extractor can extract the proper size of SVC bitstream to transmit. Finally, with the proposed architecture and mechanism, we can achieve good performance in video latency, bandwidth utilization, and buffer condition during the real time video transportation.

誌　謝

時間過的很快，兩年的碩士生涯到了終點。在碩一升碩二之時，要感謝老師 黃經堯教授的收容，讓我進入無線通訊領域學習通訊知識，並且要感謝老師在研究方面的指導，點破我在思考上的盲點，也讓我的研究內容能更趨完整與豐富。這一年多來，從老師身上學到的做研究的方法，將會是我人生以後很重要的知識利器。除了鄭重的感謝老師們的指導外，更想要跟一起過日子的朋友們說聲感謝。

首先，就是要感謝實驗室裡面有問必答的學長們，鴻輝、域晨、傑堯、詠嵐、理銓、冠穎、玠原以及士恆，因為你們讓我在研究的時候多了很多的靠山並指引了我很多重要的觀念，再來是陪我一起走過兩年研究的同學、學弟們，明憲、家弘、仲麒、宗耿、純孝、彥甫、禹伸、易達、志強以及信駿，因為你們才能夠讓我在苦悶的實驗室裡得到精神上的歡樂。

再來，感謝我的兩個好朋友，哲嘉以及榮軒，每次當我無聊要找人歡樂時，你們總是在我身邊。有了你們，才讓我的研究所生活更多采多姿、更加順利。最後，感謝我的爸媽以及家人，謝謝你們的支持與關懷，沒有你們就沒有現在的我。

2008 年 陳志展撰

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CHAPTER 1

RESEARCH OVERVIEW

1.1 Problem Statement

The delivery of the real time video bitstream(video conference, video telephone) to the receiver over wireless channel is a challenging problem because the real time video transportation will suffer from bandwidth fluctuation, which is caused by the variation of the available bandwidth and the channel condition. Bandwidth fluctuation will result in video latency at the receiver and bandwidth wastage at the transmitter. And it will also make buffer condition unstable whether in the transmitter or receiver. In detail, when the transmission bitrate of the PHY layer is smaller than the video bitrate of the application layer for a period, a part of the video data will not be transmitted to reveiver in time, so it is obvious that the decoder buffer is whether in the underflow stage or in the null stage (there is no video data in the decoder buffer). The video latency will happen at the receiver when the decoder buffer is in the null stage. In the other hand, if the available bandwidth of the transmitter is larger than the size of video data comes from the application layer, then a part of available bandwidth will be wasted. Because total bandwidth resource of the wireless communication system is limited, we should utilize the available bandwidth of the transmitter efficiently.

There are some related researches which work on the same problem mentained before [1] [2]. Their contribution is to propose a SVC streaming architecture to solve this problem. But their issue is to make the video bitstream fit the network bandwidth efficiently. In this thesis, our issue is not only to fit network bandwidth, but also to let the probability that SVC decoder buffer has no data to decode as lower as possible. So our contribution is to propose a cross-layer architecture to solve the problem caused by bandwidth fluctuation, and to satisfy with our issue.

1.2 Research Approach

To solve the problem caused by bandwidth fluctuation, we choose the scalable video coding (SVC) as video compression technique to compress the video source. The SVC system can change the bitrate of the coded video bitstream to fit the available bandwidth. This property is helpful to the problem caused by bandwidth fluctuation. But the SVC system cannot change the bitrate all the time, in our thesis, we change at the beginning of each group of pictures (GOP, this term GOP will discuss in chapter 3). During the time of the GOP, the bitrate of the coded video bitstream will remain the same, but the available bandwidth will still vary. So we need to decide a proper bitrate of the coded video bitstream at every beginning of the GOP.

In our research, the video latency is the first and important issue. Therefore we will propose a mechanism to decide the proper bitrate of the coded video bitstream, and will also propose a cross-layer architecture to realize this mechanism. With this mechanism, we will reduce the video latency as much as possible, and let the bandwidth utilization efficiently.

1.3 Thesis Outline

In chapter 2, we will introduce the protocol of WIMAX MAC. In chapter 3, we will briefly introduce the scalable video coding (SVC). In Chapter 4, we will propose a mechanism and a cross-layer architecture to decide the proper bitrate of the coded video bitstream. In Chapter 5, we will simulate the performance of the proposed mechanism. In Chapter 6, we will present our conclusion and future work.

CHAPTER 2

OVERVIEW OF 802.16E SYSTEM

The mobile broadband wireless access system IEEE 802.16e [3], called IEEE 802.16e-2005 or mobile WIMAX, has been approved in December 2005. Mobile WIMAX is a enhancement to IEEE 802.16-2004 [4] to support subscriber stations moving at vehicular speeds. It specifies a system for combined fixed and mobile broadband wireless access, and tries to fill the gap between very high data rate WLAN and very high mobility cellular systems.

Because of the property of the high data rate, WIMAX system is expected to provide many kinds of services, like voice, internet, and multimedia services. For different services, they have different bandwidth request mechanism and different QoS definition according to the service characteristics. WIMAX system must do the scheduling well to meet the requirement of different services.

2.1 Introduction to PHY Layer of 802.16e PHY

There are three air interface defined in spec. One is WirelessMAN-SCa. Another is WirelessMAN-OFDM. The other is WirelessMAN-OFDMA. And there are two duplex methods defined in spec. One is frequency division duplex (FDD). The other is time division duplex (TDD). In our thesis, we adopt Wireless-OFDMA air interface and TDD mode. So the introduction to PHY Layer will be focused on the OFDMA PHY and TDD mode.

2.1.1 OFDMA PHY Frame Structure

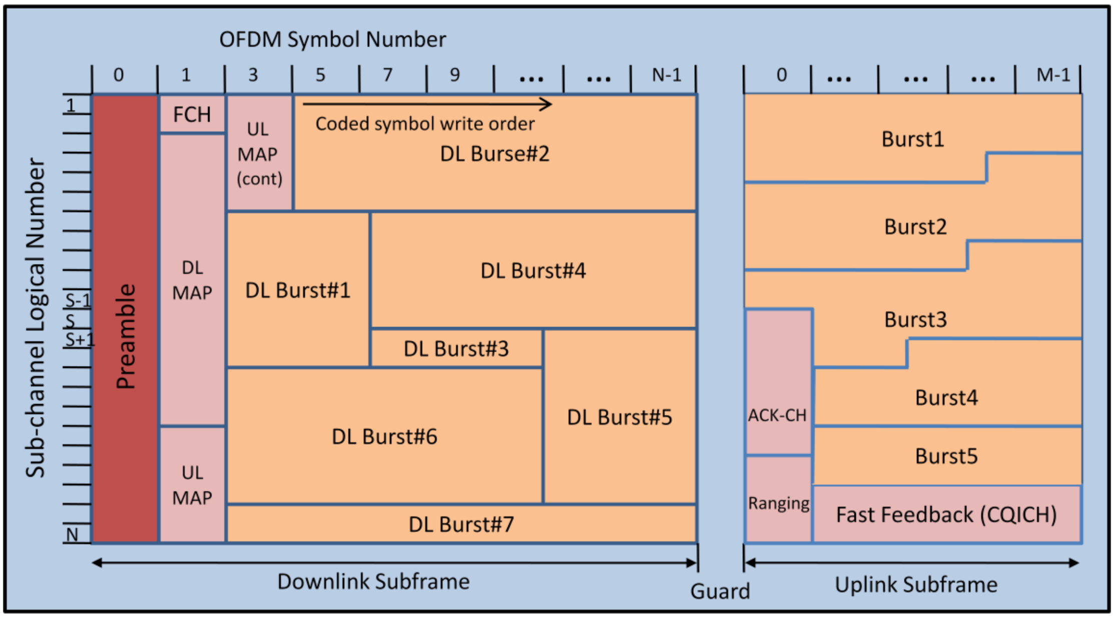

FIGURE 1 WIMAX OFDMA FRAME STRUCTURE FOR TDD MODE

The OFDMA physical layer [3][4] can use the resource flexible and efficient because of the two dimension allocation. It supports TDD, Full and Half-Duplex FDD operation. TDD mode is much common for three reasons. Firstly, TDD mode can adjust the downlink/uplink ratio to support asymmetric downlink/uplink traffic load. Secondly, TDD mode can provide global spectrum allocations. TDD mode just need a single channel for downlink and uplink. Downlink and uplink are separated from the guard time. The first part of transmission frame is downlink subframe, and the second part of transmission frame is uplink subframe. Thirdly, Transceiver designs for TDD mode implementations are less complex.

Now we will introduce the elements in the frame structure of Figure 1,

1) Preamble : The preamble occupies the first OFDMA symbol of the frame, and it is used for synchronization and channel estimation.

2) Frame Control Header (FCH) : The FCH contains DL_Frame Prefix. It specifies the length of the DL-MAP message and the repetition coding used for the DL-MAP message.
3) DL-MAP and UP-MAP : The DL-MAP and UL-MAP are used for resource allocation of DL and UL data bursts, and include burst-MS pairing information, the modulation and coding schemes of each data burst.
4) UL Ranging : The UL ranging channel is allocated for mobile stations to perform closed-loop time, frequency, power adjustment and bandwidth requests.
5) UL CQICH : The UL CQICH channel is allocated for mobile stations, mobile station can use it to feedback channel state information to base station.
6) UL ACK : The UL ACK channel is allocated for mobile stations, mobile stations use it to feedback DL HARQ acknowledge.

2.1.2 PHY Slot And Data Mapping

The OFDMA slot is a minimum unit for data transmission. One OFDMA slot occupies one subchannel and several OFDMA symbols. For downlink Full Usage of Subcarriers (FUSC) using the distributed subcarrier permutation, one slot is one subchannel by one OFDMA symbol. For downlink Partial Usage of Subcarriers (PUSC) using the distributed subcarrier permutation, one slot is one subchannel by two OFDMA symbols. For uplink Partial Usage of Subcarriers (PUSC) using the distributed subcarrier permutation, one slot is one subchannel by three OFDMA symbols. For downlink and uplink Band Adaptive modulation and coding (BandAMC) using the adjacent subcarrier permutation, one slot is one subchannel by one, two, three or six OFDMA symbols.

A Data Region is a two dimensional allocation which contents a group of contiguous subchannels and OFDMA symbols. All the allocation refers to logical subchannels. The minimum unit of data mapping is an OFDMA slot.

In Mobile WIMAX system, the BS decides that how many and which resource units are assigned to a connection. And the assignment of resource units is different in downlink and uplink. In downlink, system will consider the data size and try to fulfill the resource units in frequency domain first. After the frequency domain is fulfilled, and then system will try to fulfill the resource units in time domain. In uplink, the system will try to fulfill the resource unit in time domain first. After the time domain resource units are fulfilled, and then go to another frequency domain and repeat the procedure. Figure 2 and Figure 3 show downlink resource allocation and uplink resource allocation mechanism.

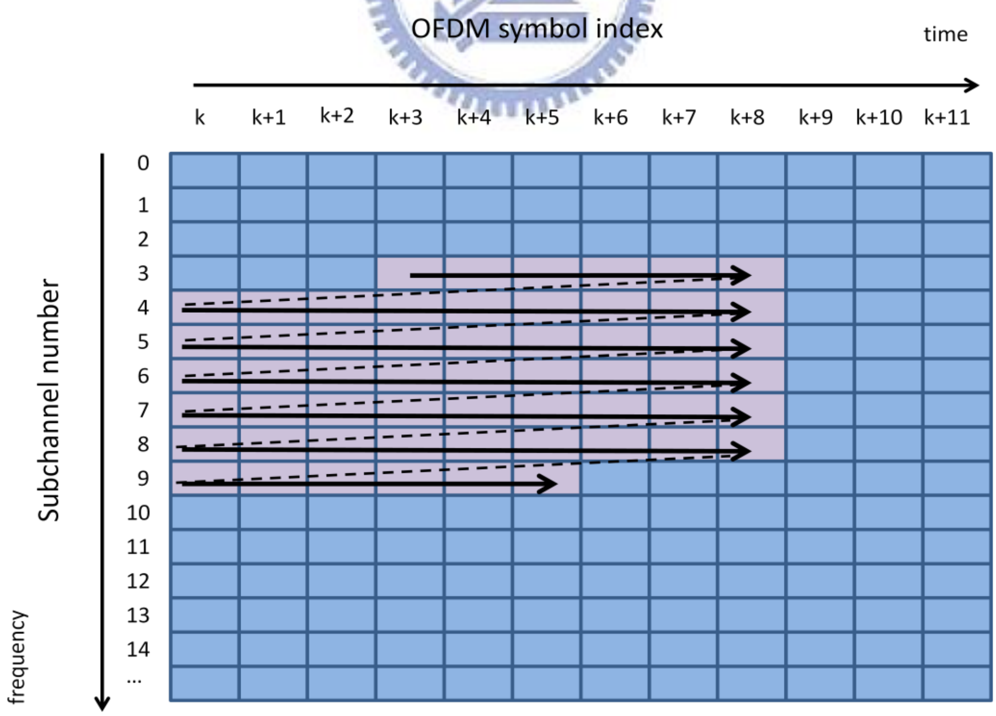

FIGURE 2 DL RESOURCE ALLOCATION

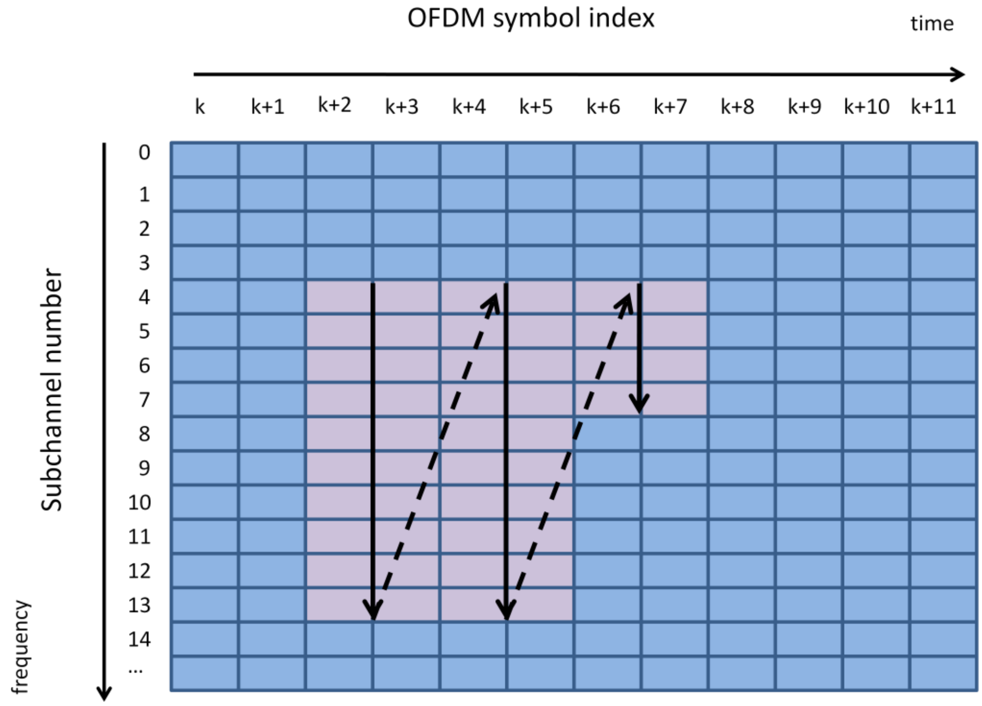

FIGURE 3 UL RESOURCE ALLOCATION

2.1.3 Subcarrier Permutation

Subcarrier permutation is a method to assign frequency subcarrier into subchannels. There are two kinds of permutation types. One is distributed subcarrier permutation. The other is adjacent subcarrier permutation.

Distributed permutation is that subcarriers belonging to a subchannel are selected pseudo randomly from all subcarriers. It can average intercell interference and avoid fading effect. The adjacent subcarrier permutation chooses adjacent subcarriers to form the subchannel. With adjacent subcarrier permutation, system can take advantage of frequency select fading and get multiuser diversity on the frequency domain.

IEEE 802.16e provides three ways to group subcarriers into the subchannel,

1) Full Usage of Subchannels (FUSC) :

This method is distributed permutation mode and is used in downlink only. It can use all subcarriers to do permutation for one subchannel and achieve the best frequency diversity by spreading subcarriers over entire band.

2) Partial Usage of Subchannels (PUSC) :

This method is distributed permutation mode and is used in both downlink and uplink. Firstly, subcarriers are grouped into several clusters. Then subcarriers are chosen one by one from each cluster to form a subchannel.

3) Band Adaptive Modulation and Coding (BandAMC) :

This method is adjacent permutation mode and is used in both downlink and uplink. The total bandwidth is divided into sub-bands and tries to utilize the frequency select fading to enhance the system performance.

2.1.4 Adaptive Modulation And Coding (AMC)

In Mobile WIMAX, the system will adaptively change the modulation coding scheme according to the channel condition of the radio link. The system supports severval modulation types : Quadrature Phase Shift Keying (QPSK), 16-state Quadrature Amplitude modulation (16-QAM), and 64-state Quadrature Amplitude modulation (64-QAM). And the system also supports several coding schemes : Convolution Code (CC), Low Density Parity Check Code (LDPC), Block Turbo Code (BTC), and Convolution Turbo Code (CTC). TABLE 1 summarizes the modulation coding scheme supported in the Mobile WIMAX profile.

TABLE 1   SUPPORTED CODE AND MODULATION

		DL	UL
Modulation		QPSK,16QAM,64QAM	QPSK,16QAM,64QAM
Code Rate	CC	1/2, 2/3, 3/4, 5/6	1/2, 2/3, 5/6
	CTC	1/2, 2/3, 3/4, 5/6	1/2, 2/3, 5/6
	Repetition	x2, x4, x6	x2, x4, x6

2.2 Introduction to MAC Layer of 802.16e MAC

2.2.1 Layer Structure

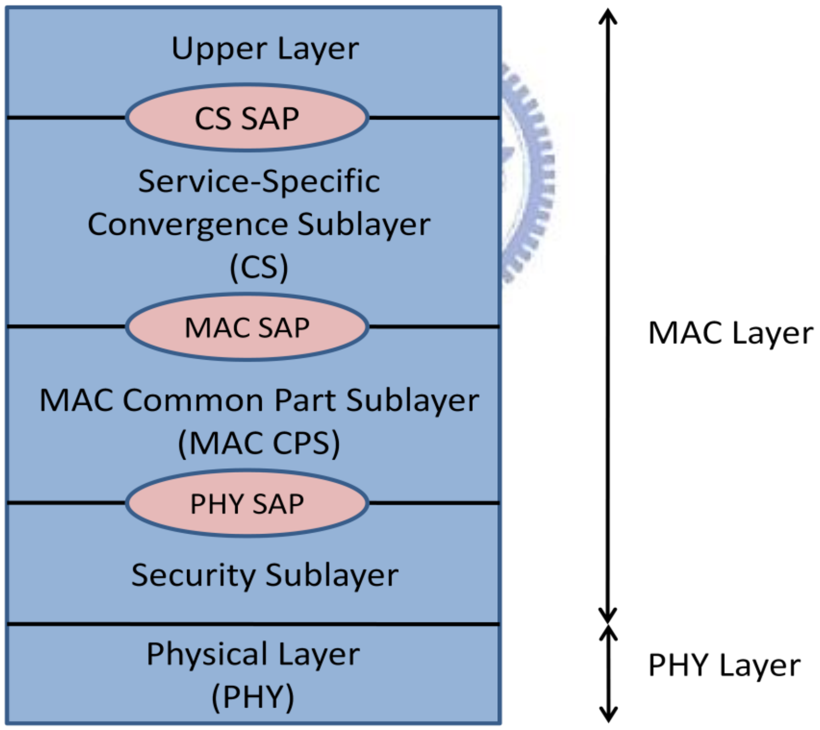

FIGURE 4   PHY-MAC STRUCTURE IN IEEE 802.16E

As shown in Figure 4, there are three sublayers in the MAC layer. We will introduce the functionality of each sublayer.

1) Service-Specific Convergence Sublayer (CS) : This sublayer is an interface between upper layer and MAC layer. The most important functionality of CS is to identify different traffic from upper layer and to assign connection ID (CID) to each connection.
2) Common Part Sublayer (CPS) : This sublayer manages the main function of controlling the whole radio resource, such as QoS control, fragmentation, packing, scheduling, request-and-grant, admission control, handover and QRQ.
3) Security Sublayer : This sublayer performs the authentication of network access, registration, key exchange and encryption of PDUs.

2.2.2 MAC PDU Formats

The MAC PDU is a data unit between the BS MAC layer and MS MAC layer. A MAC PDU consists of a 48 bits MAC header, a variable length data payload, and an optional 32 bits Cyclic Redundancy Check (CRC). Some MAC PDU will not include payload and CRC bits. These kinds of PDUs are used only in the uplink to transmit control message. These MAC signaling headers include bandwidth request, uplink transmit power report, CINR report, CQICH allocation request, PHY channel report, uplink sleep control, SN report, and feedback functionalities. MAC PDUs also include some subheaders. Those subheaders will be inserted in MAC PDUs following the generic MAC header. Those subheaders help system perform grant management, packing, ARQ feedback, and so on.

2.2.3 Fragmentation And Packing

In WIMAX system, the MAC SDU coming from CS will be formatted according to the MAC PDU format in the CPS, possibly with fragmentation and packing due to efficient utilization of the radio resource and packet error rate.

Fragmentation process is to divide a SDU into several PDUs payload areas. There are two reasons for fragmentation. One is that the SDU size is larger than the maximum size of PDU payload. The other is for preventing high packet error rate. The larger PDU size is, the higher packet error rate is. So the WIMAX system needs to divide the SDU size properly according to the channel condition. Figure 5 shows the process of fragmentation.

Packing process is to pack several SDUs into a single PDU payload. In this way, system may avoid resource waste due to the overhead caused by MAC header and CRC. Figure 6 shows the process of packing.

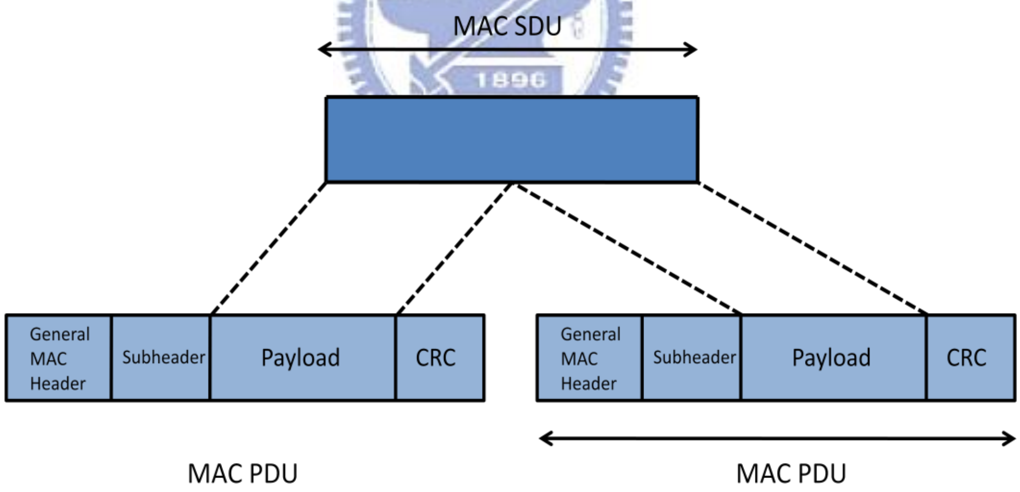

FIGURE 5 FRAGMENTATION

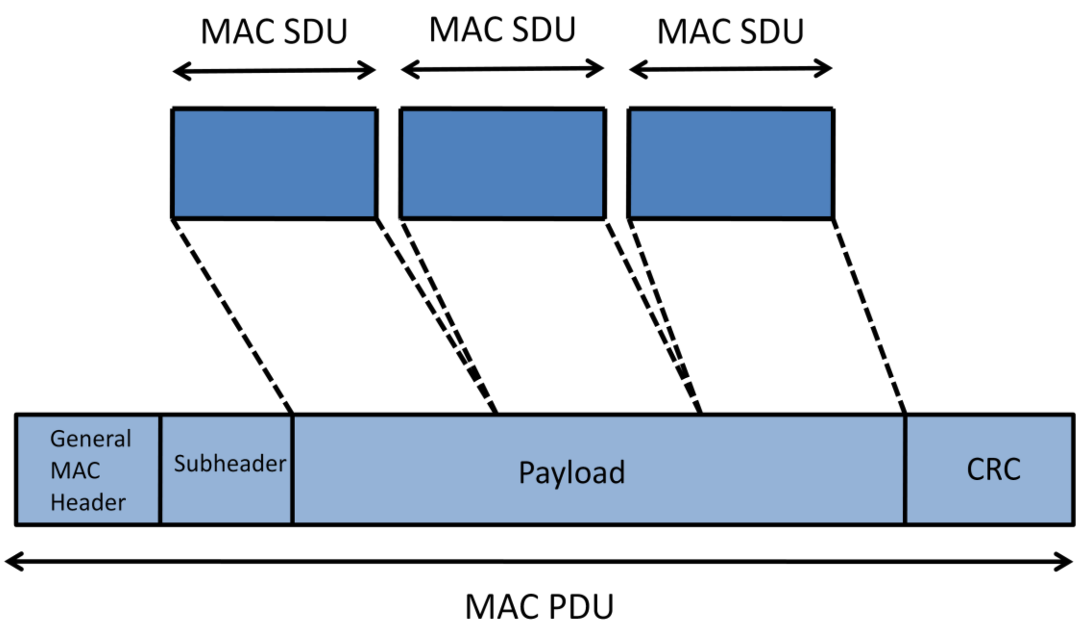

FIGURE 6 PACKING

2.2.4 QoS Based Service Classes

The IEEE 802.16e standard provides several QoS classes for different kinds of services. For different QoS classes, system sets different parameters and transmission/request methods to meet the requirement of different kinds of service. Here will introduce these classes:

1) Unsolicited Grant Service (UGS) : Designed to support real-time service flows that generate fixed-size data packets periodically, such as T1/E1 and VoIP without silence suppression.

2) Real-time Polling Service (rtPS) : Designed to support real-time service flows that generate variable-size data packets, such as Moving Picture Experts Group (MPEG) video.

3) Extended Real-time Polling Service (ertPS) : A scheduling mechanism that builds on the efficiency of both UGS and rtPS. The BS provides unsolicited unicast

grants as in UGS, thus saving the latency of a bandwidth request. However, UGS allocations are fixed in size, whereas ertPS allocations are dynamic.

4) Non-real-time Polling Service (nrtPS) : The nrtPS is designed for non-real-time service that can tolerate more delay, such as FTP, web-browsing and so on.

5) Best Effort Service (BE) : BE service is with the lowest QoS level. These kinds of service are designed to support data streams for which no minimum service level is required and therefore may be handled on a space-available basis.

CHAPTER 3

OVERVIEW OF SCALABLE VIDEO CODING

3.1 Introduction to Scalable Video Coding

Data networks for video communication are growing fast nowadays. The environment varies from broadband cable/ADSL networks to wireless/mobile networks. Besides, the display monitors of the devices are also diversified. It may be a small size screen on a mobile device or a high definition projection system. For different applications on various devices or under different network conditions, the available bandwidth and resource may be highly divergent. To overcome different application scenarios, the idea of scalable video coding is proposed.

The scalable video coding (SVC) standard [5] is an extension of the H.264/AVC standard [6] developed by the Joint Video Team (JVT) that uses a single bit-stream to provide multiple frame rates, frame sizes and quality levels while achieving a reasonable coding efficiency.

3.2 Encoder Overview

The SVC encodes the video into multiple spatial, temporal, and SNR layers for combined scalability. Figure 7 shows the generic structure of an SVC encoder with three spatial layers.

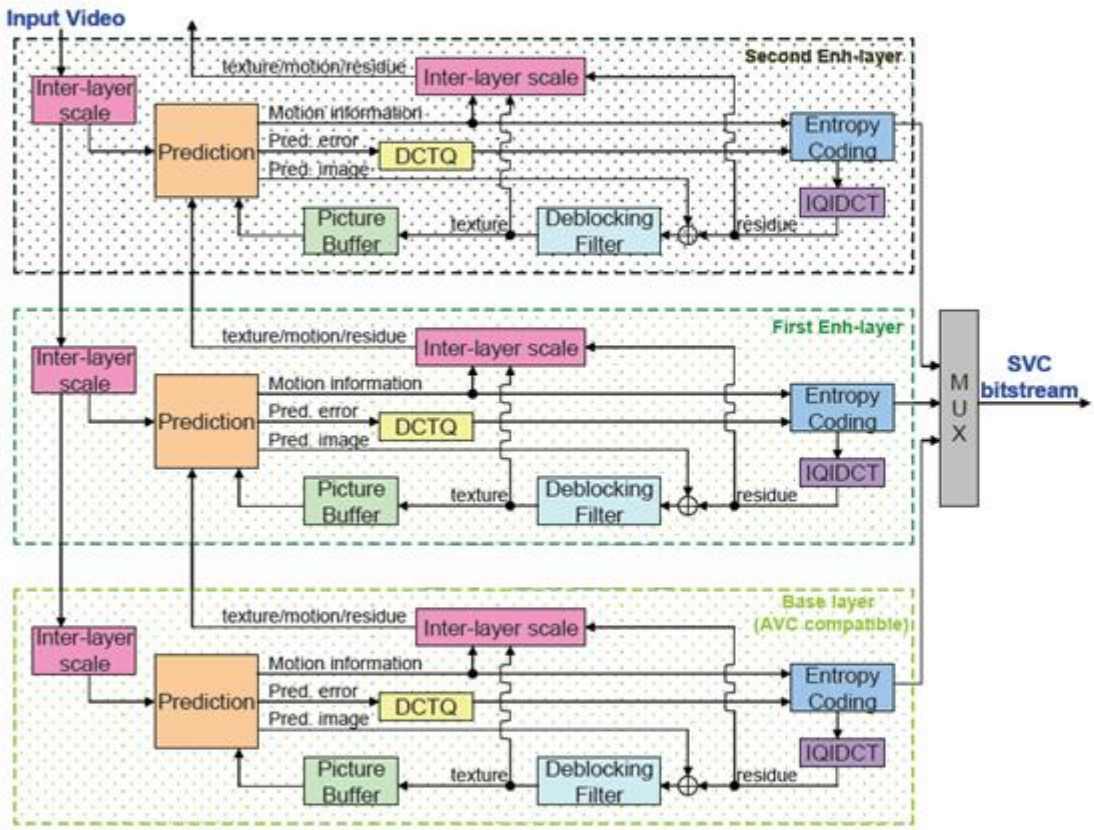

Figure 7 SVC encoder structure with three spatial layers [7]

SVC encoder provides three different scalable features in spatial, temporal and SNR layer respectively. Spatial scalability and CGS are achieved by multiple layers with a pyramid structure. Temporal scalability is achieved by a temporal decomposition using hierarchical B pictures. FGS is achieved by encoding successive refinements of the transform coefficients.

3.3 Hierarchical-B Prediction Structure

SVC encoder uses hierarchical-B prediction structure to achieve multilevel temporal scalability. Figure 8 depicts a hierarchical-B prediction structure with 4 temporal levels and a GOP size of 8. Each key picture is either an intra-coded frame(I frame) or a P frame that uses the previous key picture as the reference picture. The picture number from 1 to 7 are B frames. Each B-frame is bi-directionally predicted using both previously and future displayed reference pictures from the lower temporal level. The hierarchical-B structure has better coding efficiency using more efficient frame level bit allocation, especially for sequences with fine texture and regular motion.

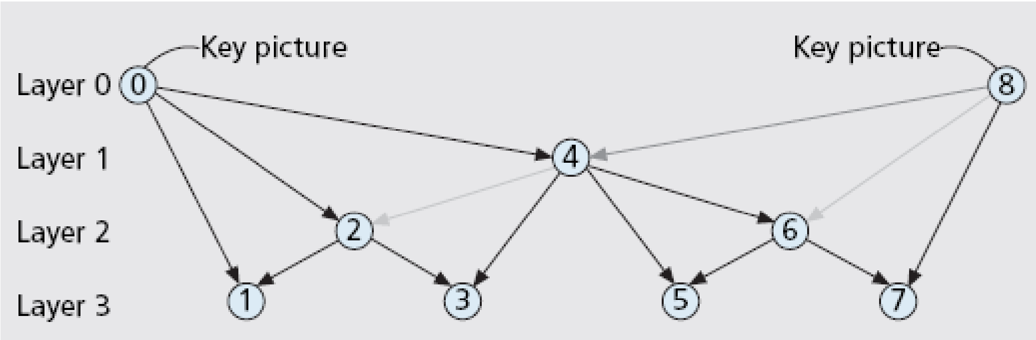

Figure 8 Hierarchical-B prediction structure [7]

3.4 Inter-layer Prediction Structure

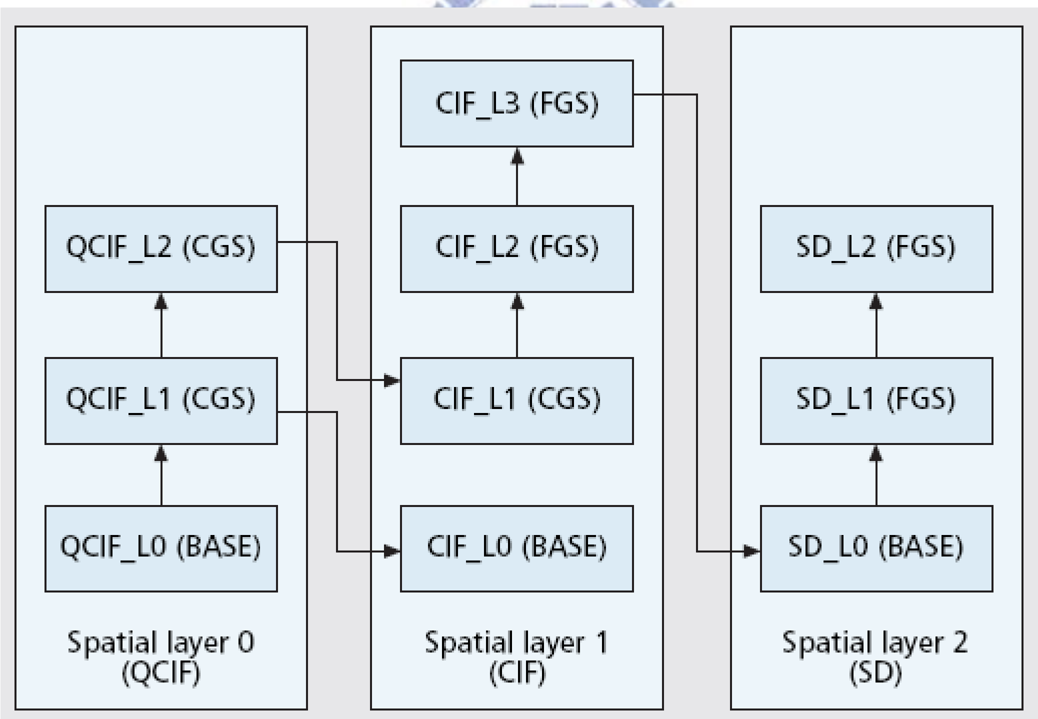

Figure 9 Inter-layer prediction structure with three spatial layers [7]

Interlayer prediction is dependent on the types of layers used. The spatial and CGS layers can flexibly select the reference layer from any lower layers while the FGS layer must be predicted from the previous SNR layer at the same resolution. As demonstrated by an example in Figure 9, the three columns represent three spatial resolutions: QCIF, CIF, and 4CIF. Each spatial resolution contains several SNR layers, and the arrow specifies the reference layer.

CHAPTER 4

THE PROPOSED CROSS-LAYER DESIGN

In this chapter, we will propose a mechanism and a cross-layer architecture between the SVC system and the WIMAX MAC to solve the problem discussed in chapter 1.1 and chapter 1.2.

4.1 The Design Issues

Bandwidth fluctuation in Mobile WIMAX system is caused by the variation of bandwidth allocation and the channel condition. And it brings many problems during the transportation of real time video bitstream (video conference or video telephone), such as video latency, available bandwidth wastage and buffer condition unstable. So the first design issue is latency issue, and the second design issue is the utilization of the available bandwidth.

What we have to do is to estimate a proper bitrate of the coded video bitstream for each GOP and send it to the SVC extractor. With this bitrate, the SVC extractor will extract the proper size bitstream for each GOP to transmit. So during the transportation of real time video bitstream, we could have the better performance in video latency in the receiver, and also have the efficient utilization of available bandwidth in the transmitter.

4.2 The Proposed Cross-Layer Architecture

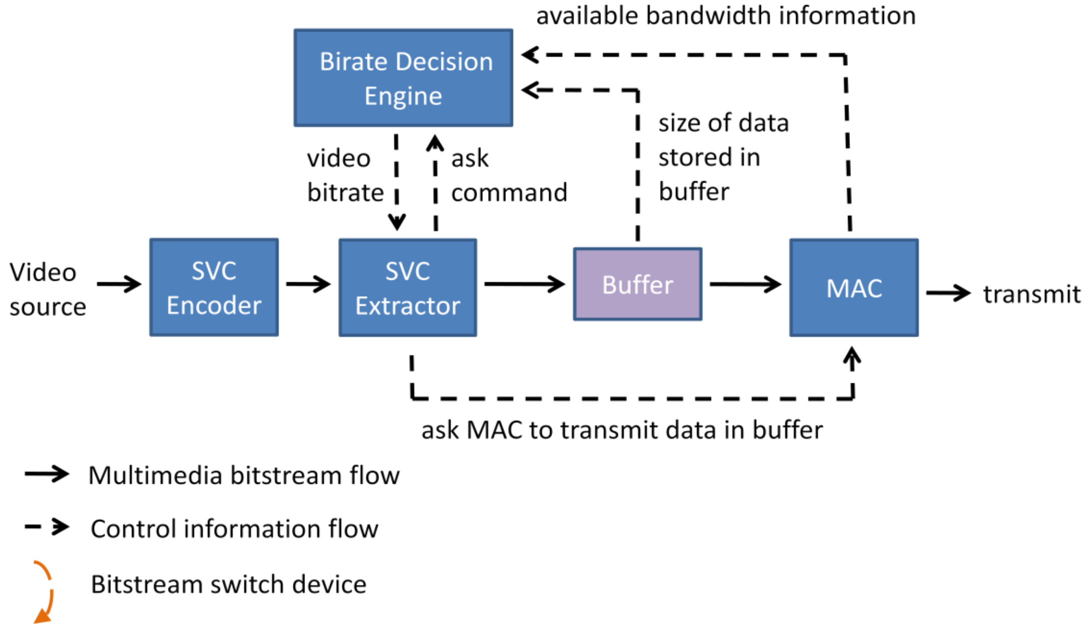

FIGURE 10 THE PROPOSED CROSS-LAYER ARCHITECTURE

The most important component of the proposed architecture is the Bitrate Decision Engine. How to design this component will be discussed in chapter 4.3. Now we should focus on the overall coded video bitstream flow and control information flow of the proposed cross-layer architecture, as shown in Figure 10.

The video source is sent to SVC encoder, and then the coded bitstream is sent to the SVC extractor. Before the beginning of each GOP, SVC extractor will ask Bitrate Decision Engine to feedback the extracted video bitrate, and uses this bitrate to extract the proper size of bitstream for this GOP.

There is one buffer between MAC and SVC extractor. It is FIFO buffer. When the SVC extractor starts to extract the SVC bitstream, it will ask MAC to transmit the data in buffer at the same time.

There are two inputs of the Bitrate Decision Engine. One is the size of data stored in buffer. The other is the available bandwidth information of the period in the past. With the input information, the bitrate decision engine will decide the bitrate of the coded video bitstream for the present GOP.

4.3 The Bitrate Decision Engine

In chapter 4.2, we have mentioned that SVC extractor will ask Bitrate Decision Engine to feedback the extracted video bitrate before the beginning of each GOP. To achieve a better performance, an enhanced feedback control mechanism will be proposed.

Before the discussion of the proposed enhanced feedback control mechanism, we will first discuss four different conventional feedback control mechanisms.

Four different conventional feedback control mechanisms:

1) Mean Mechanism: There are 53 transmission frames (one frame is 5 ms in WIMAX system) in each GOP period (one GOP period is 8/30 seconds in our thesis). The Mean Mechanism uses the average transmission bitrate of the last GOP period to be the extracted video bitrate for the present GOP period. The formula is shown in eq(1).

$$E_N = \left(\frac{D_N + \cdots + D_{N-52}}{T \times 53}\right) \tag{1}$$

2) Median Mechanism: There are 53 transmission bitrates for each transmission frame in the last GOP period. And the Median Mechanism uses the median of these transmission bitrate to be the extracted video bitrate for the present GOP period. The formula is shown in eq(2).

$$E_N = \text{median of } \left\{\frac{D_{N-52}}{T} 、 \ldots 、 \frac{D_N}{T}\right\} \tag{2}$$

3) IIR Mechanism : IIR Mechanism is adopted by [1]. The formula is shown in eq(3).

$$E_N = p \times E_{N-1} + (1-p) \times \frac{D_N}{T} \tag{3}$$

4) Instant Mechanism: Instant Mechanism uses the first transmission bitrate in the present GOP period to be the extracted video bitrate for the present GOP period. The formula is shown in eq(4).

$$E_N = \frac{D_N}{T} \tag{4}$$

where

E_N : the extracted video bitrate for the present GOP period (bits/sec)

E_{N-1}: the extracted video bitrate for last GOP period (bits/sec)

D_N : the data size which can be transmitted at the first transmission frame of this GOP period (bits)

D_{N-i}: the data size which can be transmitted at the $(53-i)$th transmission frame of the last GOP period (bits)

p: 0.2

T: the time of one transmission frame (sec)

Enhanced feedback control mechanism(proposed):

Enhanced feedback control mechanism is proposed to satisfy the latency issue. It modifies the extracted video bitrate decided by conventional feedback control mechanism based on the data size stored in the buffer. The modification size is shown in eq(5) :

$$\Delta = \alpha \times \frac{D_B}{N \times T} \tag{5}$$

where

Δ: the modification size (bits/sec)

D_B: the data stored in the buffer which the MAC is transmitting (bits)

T: the time of one transmission frame (sec)

N: the number of transmission frames in one GOP period

α: the cofficient of the modification size, $0 \le \alpha \le 1$. In our thesis, α is set to 0.3 by experiment

Finally, the Bitrate Decision Engine will send the modified extracted video bitrate to the SVC extracter. The formula is shown in eq(6) :

$$\text{Modified extracted video bitrate} = E_N - \Delta \tag{6}$$

where

E_N: the extracted video bitrate decided by conventional feedback control mechanism for the present GOP period (bits/sec)

4.4 Discussion: The reason why enhanced feedback control mechanism can improve the performance of latency issue

Sometimes, the extracted SVC data which can't be transmitted to the receiver in its GOP period will be stored in MAC buffer, and it will be transmitted during the next GOP period. The data which can't be transmitted to the receiver in its GOP period is named Delay Data.

There are two types of data which need to be transmitted during the present GOP period. One is the Delay Data. The other is the extracted SVC bitstream of the present GOP. Therefore, we need to divide the transmission bitrate of the present GOP period into two parts. One part is used to transmit the extracted SVC bitstream of the present GOP. The other part is used to transmit the Delay Data.

The transmission bitrate for the present GOP period, E_N , is decided by conventional feedback control mechanism, and we divide it into two parts, Δ and $E_N - \Delta$. The Δ decided by enhanced feedback control mechanism is used to transmit the Delay Data. $E_N - \Delta$ is used to transmit the extracted SVC bitstream of the present GOP.

So the reason why enhanced feedback control mechanism can improve the performance of latency issue is because we reserve one part of the transmission bitrate, Δ , and use it to transmit the Delay Data.

CHAPTER 5

SIMULATION SETUP AND RESULTS

5.1 Simulation Setup

5.1.1 Cell Plane

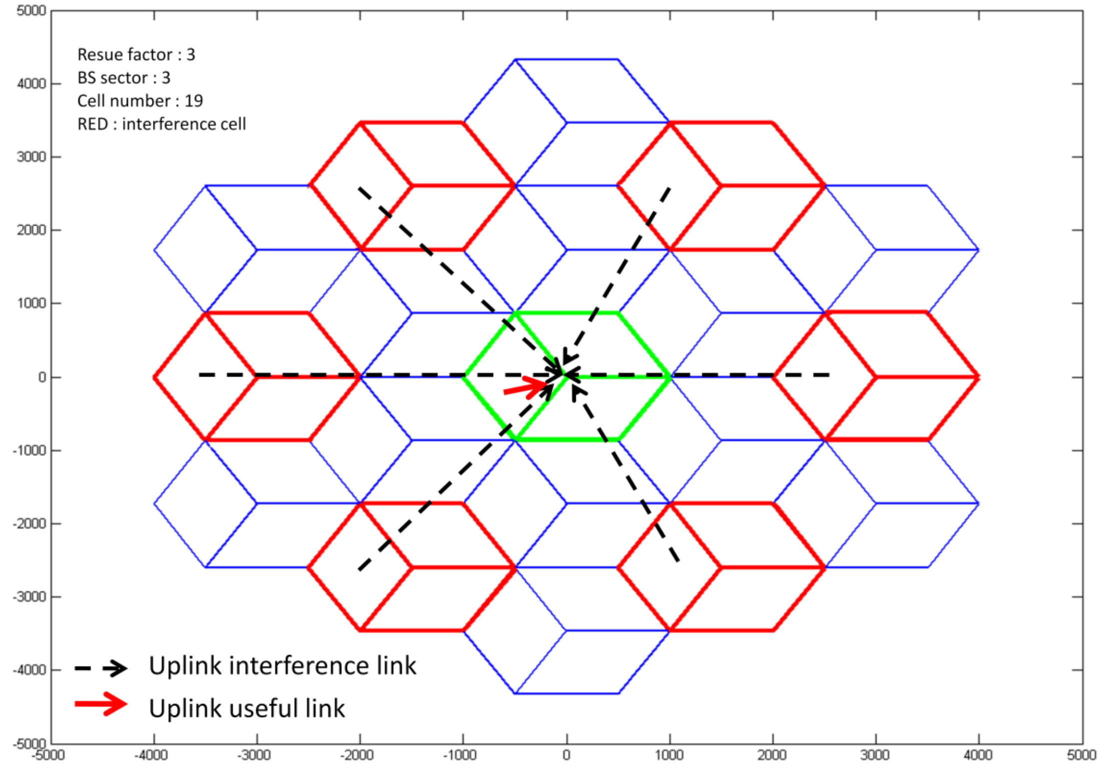

FIGURE 11 CELL PLANE

As shown in Figure 11, there are 19 cells in the simulation platform. The cell radius is 1km. And the resue factor and BS sector are both three. The interference cells are marked red color. So there are six interference links and one useful link during uplink transmission.

Based on the cell plane, the SINR formula of the uplink transmission is shown in eq(7).

$$\mathrm{SINR_{dB}} = 10\log\left(\frac{P_t}{\sum_{i=1}^{6} p_i + \mathrm{Noise}}\right) \tag{7}$$

where

p_t : received power at BS

p_i : interference power at the BS

Noise : the noise power

p_t and p_i is impacted by pass loss, shadowing and BS antenna pattern. And we will introduce BS antenna pattern, path loss model and shadowing model in chapter 5.1.2 , chapter 5.1.3 and chapter 5.1.4.

5.1.2 BS Antenna Pattern For 3-sector Cells

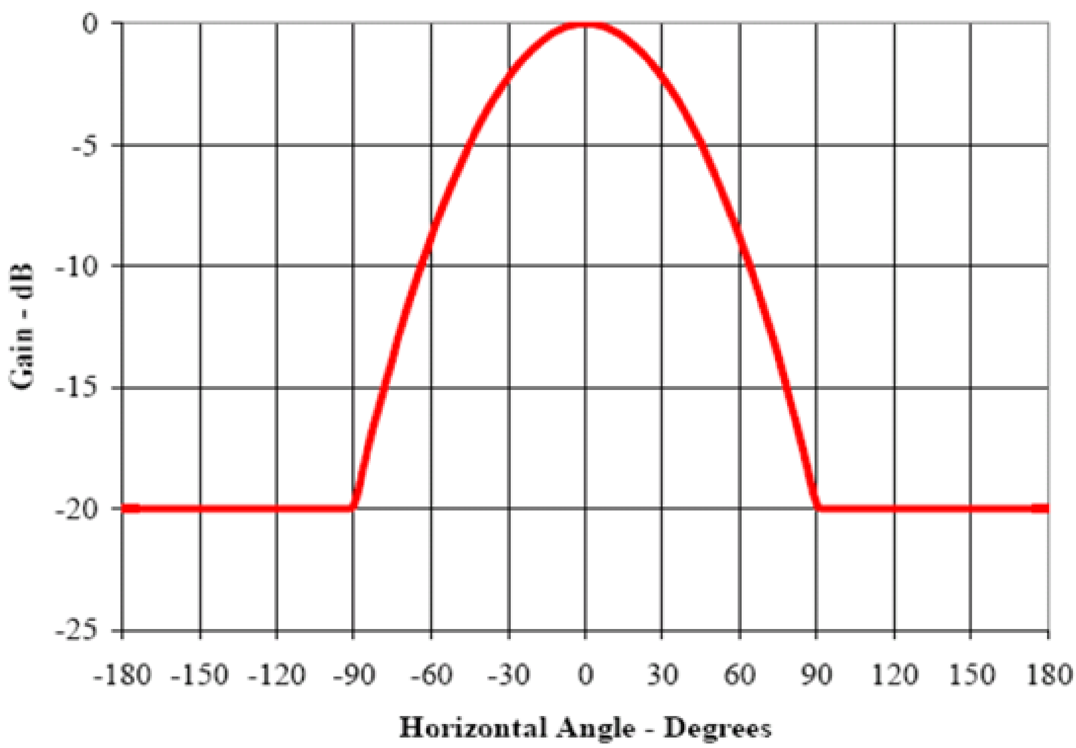

FIGURE 12 ANTENNA PATTERN FOR 3-SECTOR CELLS

As shown in Figure 12, this antenna pattern [8] is used in our simulation platform. And the formula of antenna gain is shown in eq(8).

$$A(\theta) = -\min\left[12\left(\frac{\theta}{\theta_{3dB}}\right)^2 , A_m\right] \tag{8}$$

where

$A(\theta)$ is the antenna gain in dBi in the direction θ

θ_{3dB} is the 3 dB beamwidth (corresponding to $\theta_{3dB} = 70^{\circ}$)

$A_m = 20dB$ is the maximum attenuation

5.1.3 Path Loss Model

The pathloss model [8] is used to simulate the degradation of signal strength with increasing distance between transmitter and receiver. The formula is shown in eq(9).

$$PL(dB) = \begin{cases} 20\log\left(\frac{4\pi d}{\lambda}\right) & \text{for } d \le d_0' \\ A + 10\gamma\log\left(\frac{d}{d_0'}\right) + \Delta PL_f + \Delta PL_{ht} & \text{for } d > d_0' \end{cases} \tag{9}$$

where

$A = 20\log\left(\frac{4\pi d_0'}{\lambda}\right)$

$d_0 = 100m$

$d_0' = d_0 10^{-\left(\frac{\Delta PL_f + \Delta PL_{ht}}{10\gamma}\right)}$

$\gamma = a - bh_b + \frac{c}{h_b}$

$\Delta PL_f = 6\log\left(\frac{f(MHz)}{2000}\right)$

$$\Delta PL_{ht} = \begin{cases} -10\log\left(\frac{h_t}{3}\right) & for\ h_t \le 3m \\ -20\log\left(\frac{h_t}{3}\right) & for\ h_t > 3m \end{cases}$$

d = distance between BS and MS

h_b = height of BS

h_t = height of RS

$a = 3.6$，$b = 0.005$，$c = 20$

5.1.4 Shadowing Model

Shadowing fading effect is the increase or decrease of signal strength due to the shelters, like buildings or mountains, on the signal transmitted path. According to the test result of the real wireless environment, it is known that the variant of shadow fading is a log-normal distribution [8]. Hence, log-normal distribution is applied to simulate shadowing fading effect. In our simulation platform, the standard deviation is set to 8 dB.

If the location of the MS does not change too much, the variance of shadow fading will have a correlated relationship associated with the distance the MS moves during the time duration between two neighbor simulation drops. The correlation model [8] is shown in eq(10):

$$\rho(\Delta x) = e^{-\frac{|\Delta x|}{d_{cor}}\ln 2} \tag{10}$$

where

ρ is the auto − correlation constant between two simulation drops

Δx is the distance of two simulation time drops

d_{cor} is decorrelation distance, in our simulation platform, we set it to 50m

5.1.5 Available Slots of UL-MAP Simulation

In our thesis, we simulate the available slots in UL-MAP for each uplink subframe by the formula (11).

$$\text{Available\_slots}_i = G \times N \tag{11}$$

where

$Available\_slots_i$: the available slots for the MS in the ith uplink subframe

G : G is a Gaussion random variable with the predetermined mean and variance, and the predetermined mean and variance are fixed number during simulation. G represents the percentage of the available slots in the ith uplink subframe.

N : the total number of slots in the ith uplink subframe

5.1.6 System Parameters Setting

TABLE 2 SYSTEM PARAMETERS SETTING

Cell plane	19 cells
Sectors per cell	3
Frequency reuse factor	(3,3,3)
Available Bandwidth	30MHz with reuse factor (3,3,3)
Antenna pattern	70° with 20dB front to back ratio
Cell radius	1 km
Duplex	TDD
DL/UL subframe ratio	28:19
Frame length	5ms,according to []
OFDMA symbols per slot	3 symbols (UL-PUSC slot)
Thermal noise density	-173.93 dB/Hz, according to []
MS Tx power	27dBm
MS antenna gain	3 dBi, according to []

5.2 Simulation Results

5.2.1 Performance of conventional feedback control mechanism

5.2.1.1 Video Latency

TABLE 3 VIDEO LATENCY PERFORMANCE

	Variance = 0.1	Variance = 0.5	Variance = 1	Variance = 5
P_Zero_Mean	1.2%	1%	1.1%	1.4%
P_Zero_Median	2.9%	2.4%	1.4%	0.7%
P_Zero_IIR	3.2%	2.6%	2.6%	4.9%
P_Zero_Instant	2.1%	2%	1.7%	4%

Where

P_Zero : the probability that decoder buffer has no data to decode

Variance : the variance of the available slots

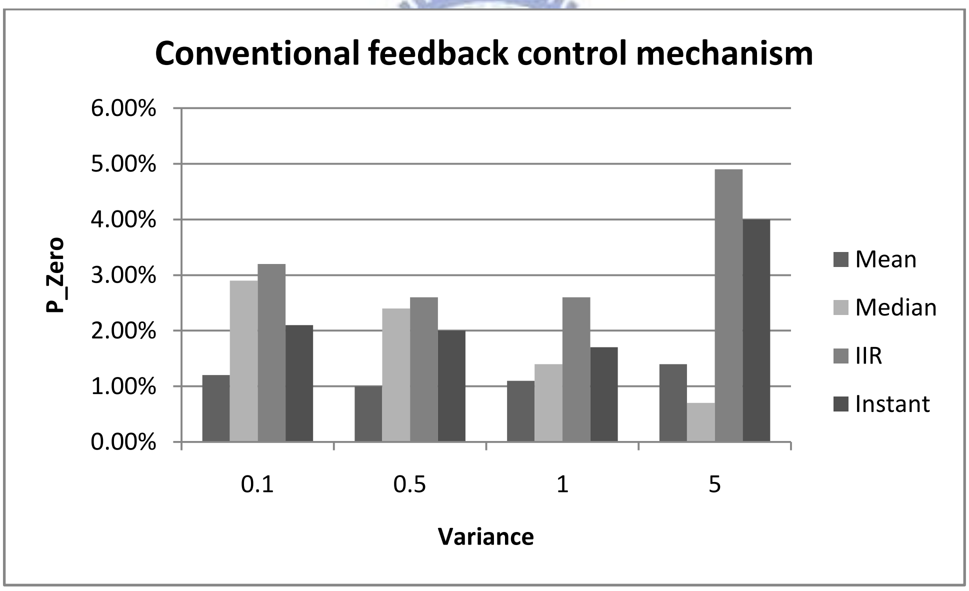

FIGURE 13 GRAPH OF TABLE 3

5.2.1.2 MAC Buffer Condition

TABLE 4 MAC BUFFER CONDITION PERFORMANCE

	Variance = 0.1	Variance = 0.5	Variance = 1	Variance = 5
Av_MB_Mean	2006.4	2035.4	2038.5	2405.7
Av_MB_Median	4783.7	4151.5	2504.2	1362.9
Av_MB_IIR	6693.8	5939.8	5253.8	12693.3
Av_MB_ Instant	6573.4	6206	5309.1	12587.6

Where

Av_MB : the average size of data stored in MAC buffer

Variance : the variance of the available slots

Units : byte

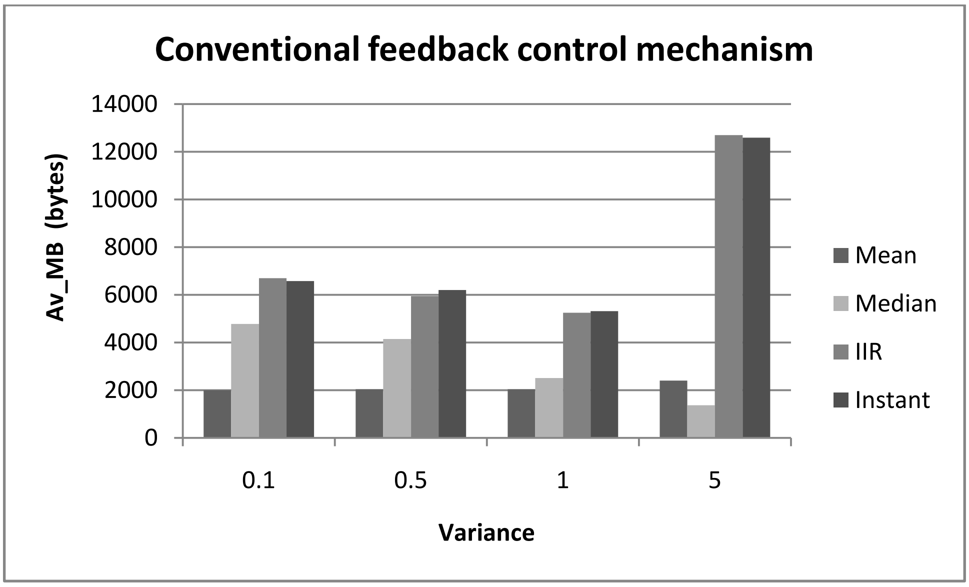

FIGURE 14 GRAPH OF TABLE 4

5.2.1.3 Available Bandwidth Utilization

TABLE 5 AVAILABLE BANDWIDTH UTILIZATION PERFORMANCE

	Variance = 0.1	Variance = 0.5	Variance = 1	Variance = 5
Ut_Mean	99.7%	99.7%	99.7%	99.8%
Ut_Median	99.3%	99.4%	99.4%	98.2%
Ut_IIR	99.5%	99.5%	99.2%	99.6%
Ut_ Instant	99.5%	99.5%	99.2%	99.6%

Where

Ut : represents the utilization of available bandwidth, the formula is shown as eq(12):

$$\mathrm{Ut} = \frac{\sum_i \mathrm{TData}_i}{\sum_i \mathrm{AData}_i} \tag{12}$$

TData_i is the data size transmitted in the ith uplink subframe

AData_i is the maximun data size which the MS can transmit in the ith uplink subframe

Variance : the variance of the available slots

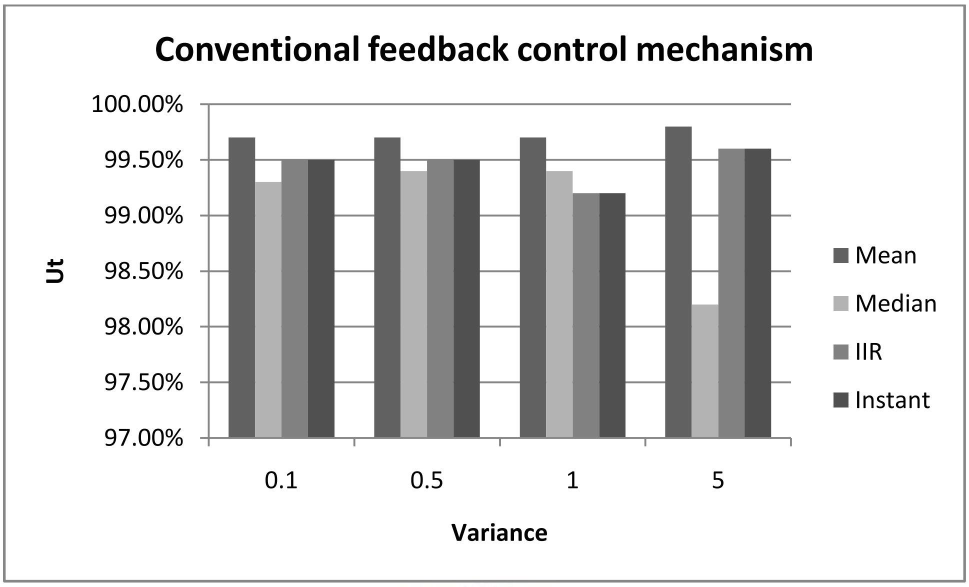

FIGURE 15 GRAPH OF TABLE 5

5.2.1.4 Discussion

The ideal case is that the probability which decoder buffer has no data to decode is as lower as possible, the average size of data stored in MAC buffer is as smaller as possible, and the utilization of the available bandwidth is as higher as possible.

With each conventional feedback control mechanism, although the performance of available bandwidth utilization is good, but we can see that the probability which decoder buffer has no data to decode is not lower enough, and the average size of data stored in MAC buffer is not smaller enough also. So we need to improve the performance of conventional feedback control mechanism in these two issues.

5.2.2 Performance of enhanced feedback control mechanism

5.2.2.1 Video Latency

TABLE 6 VIDEO LATENCY PERFORMANCE

	Variance = 0.1	Variance = 0.5	Variance = 1	Variance = 5
P_Zero_Mean	0.44%	0.44%	0.44%	0.5%
P_Zero_Median	0.48%	0.47%	0.46%	0.51%
P_Zero_IIR	0.01%	0.09%	0.09%	0.34%
P_Zero_Instant	0.09%	0.16%	0.18%	0.44%

Where

P_Zero : the probability that decoder buffer has no data to decode

Variance : the variance of the available slots

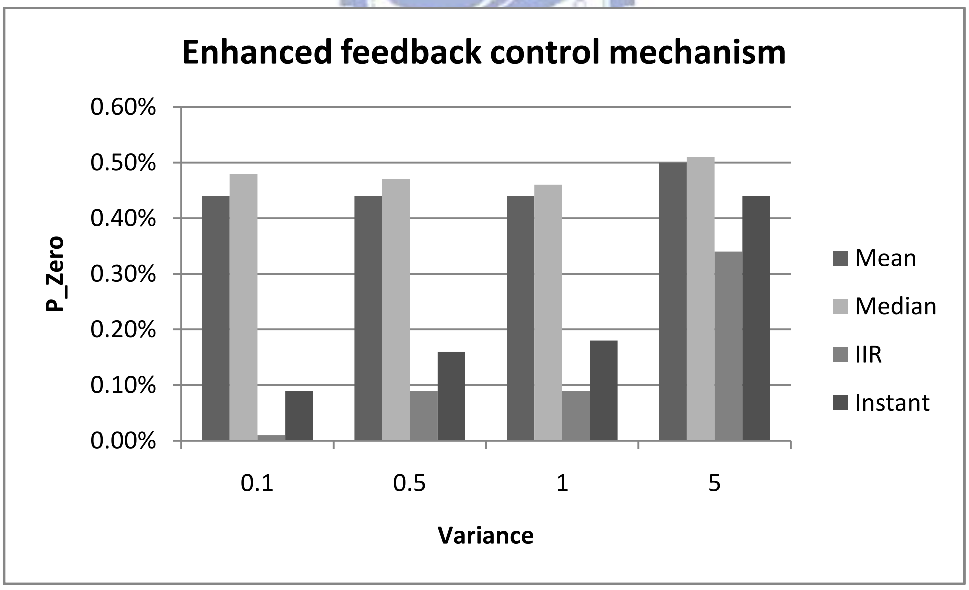

FIGURE 16 GRAPH OF TABLE 6

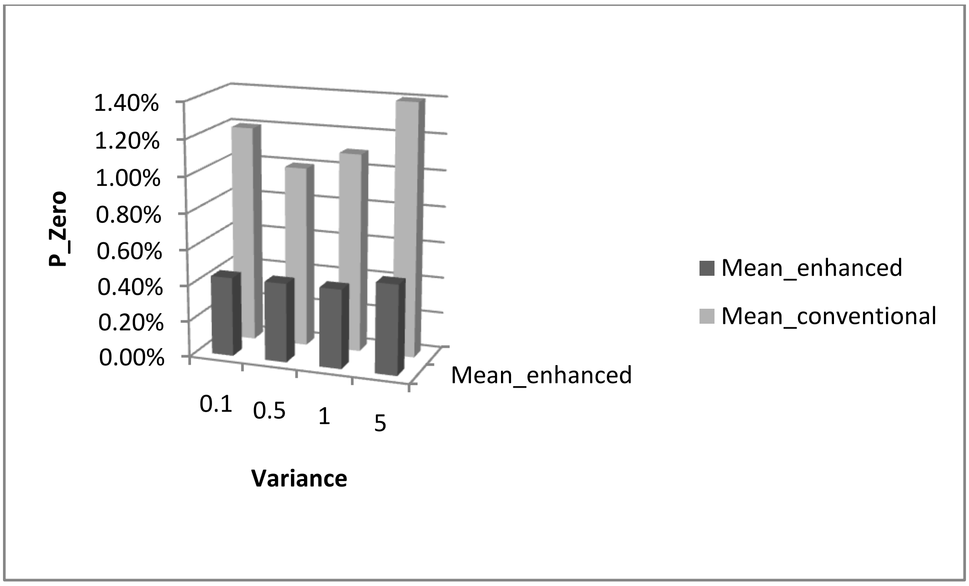

FIGURE 17 COMPARED WITH CONVENTIONAL IN MEAN

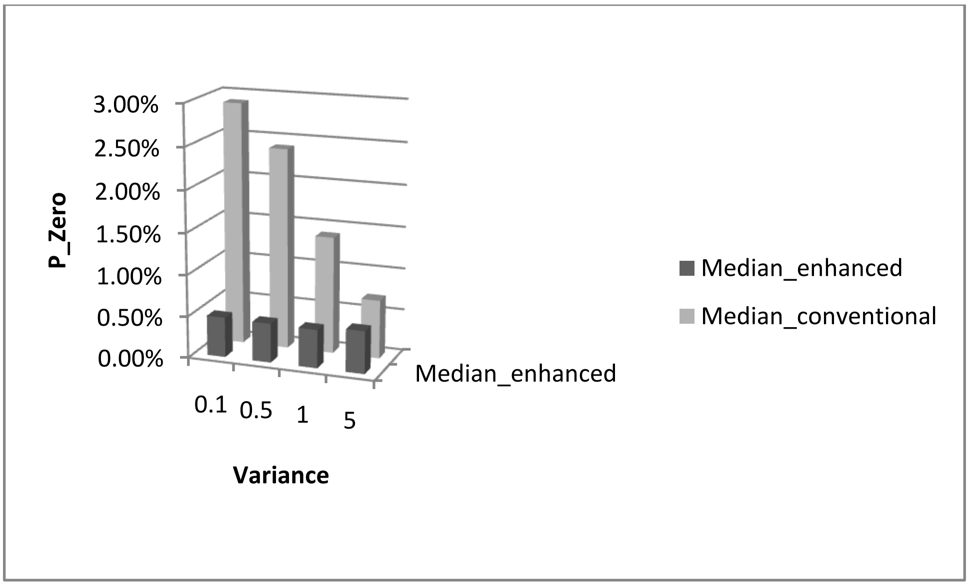

FIGURE 18 COMPARED WITH CONVENTIONAL IN MEDIAN

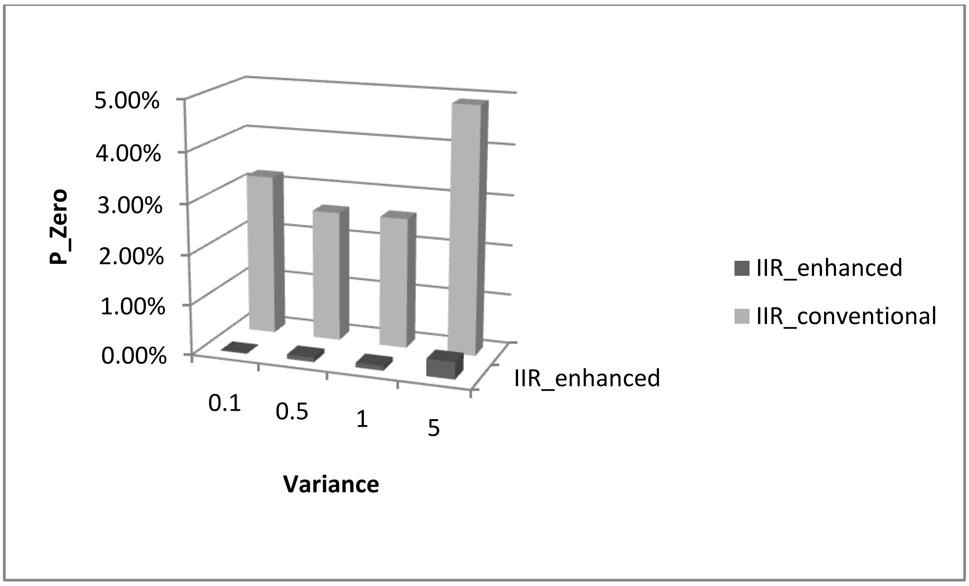

FIGURE 19 COMPARED WITH CONVENTIONAL IN IIR

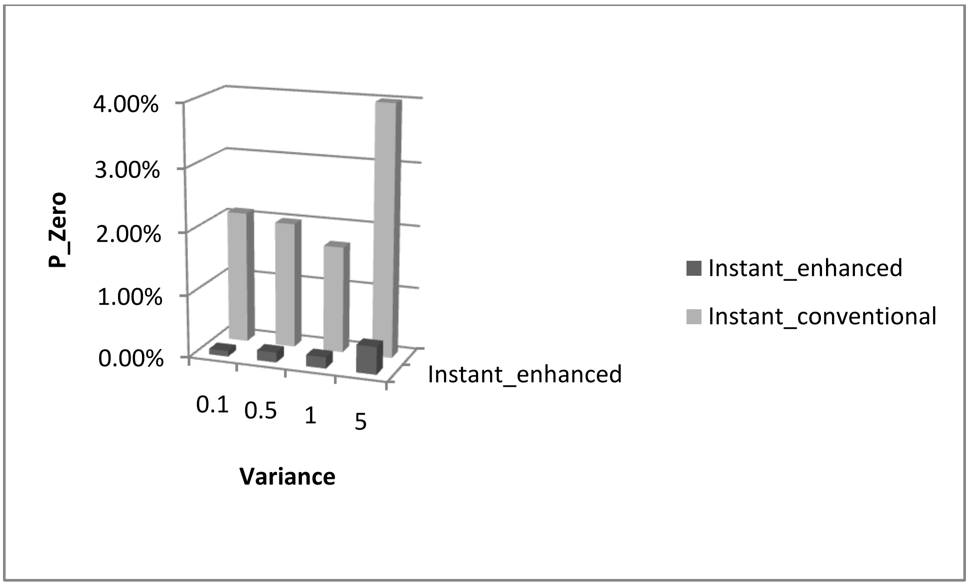

FIGURE 20 COMPARED WITH CONVENTIONAL IN INSTANT

5.2.2.2 MAC Buffer Condition

TABLE 7 MAC BUFFER CONDITION PERFORMANCE

	Variance = 0.1	Variance = 0.5	Variance = 1	Variance = 5
Av_MB_Mean	409.3	416.7	418.3	450.7
Av_MB_Median	440.1	432.7	415.8	416.9
Av_MB_IIR	364.8	384.5	407.6	546.3
Av_MB_Instant	437.8	463.7	487.8	683.2

Where

Av_MB : the average size of data stored in MAC buffer

Variance : the variance of the available slots

Units : byte

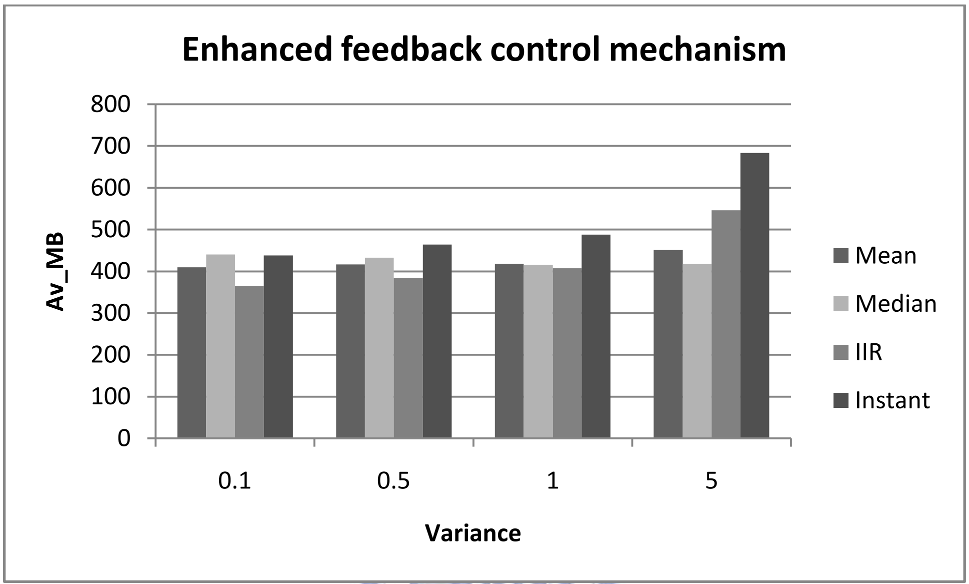

FIGURE 21 GRAPH OF TABLE 7

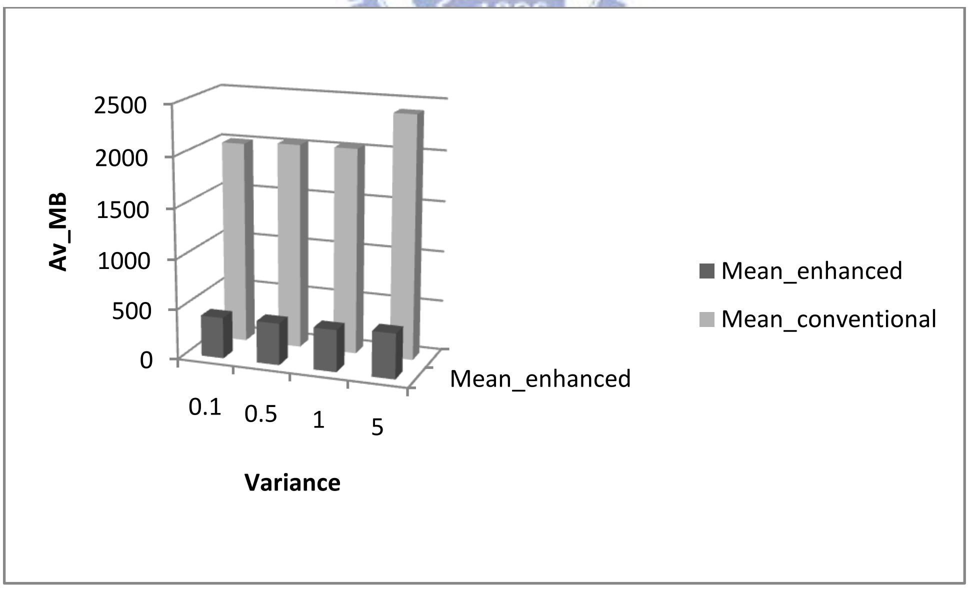

FIGURE 22 COMPARED WITH CONVENTIONAL IN MEAN

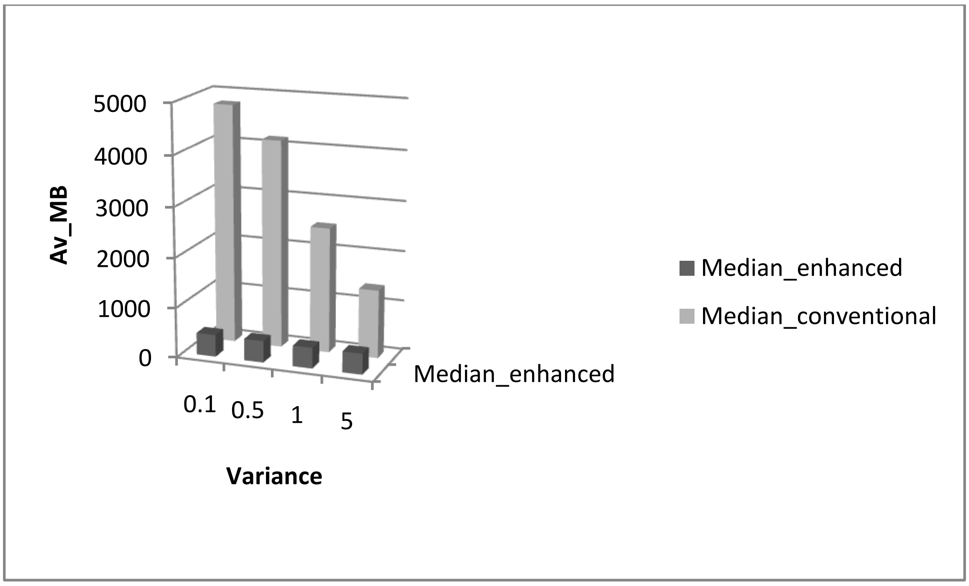

FIGURE 23 COMPARED WITH CONVENTIONAL IN MEDIAN

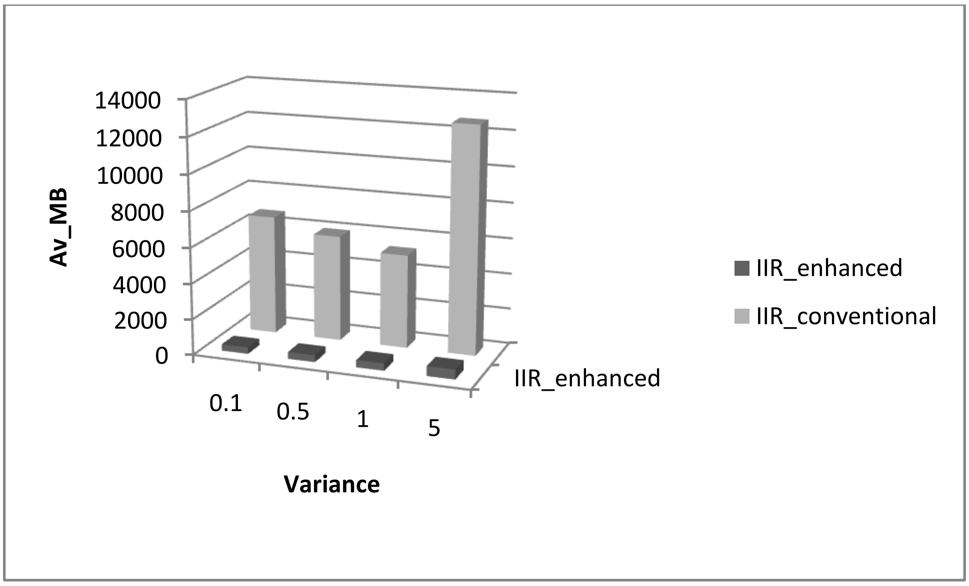

FIGURE 24 COMPARED WITH CONVENTIONAL IN IIR

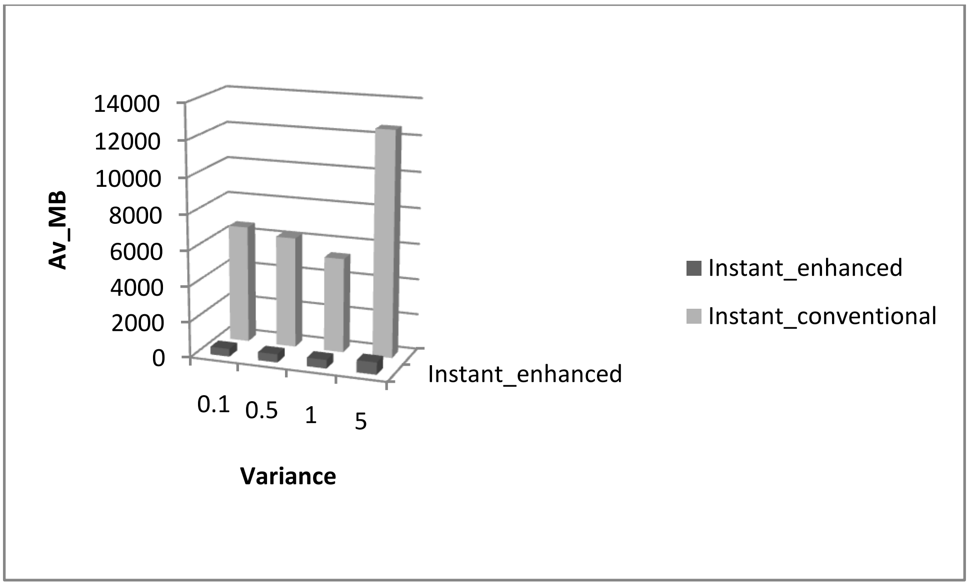

FIGURE 25 COMPARED WITH CONVENTIONAL IN INSTANT

5.2.2.3 Available Bandwidth Utilization

TABLE 8 AVAILABLE BANDWIDTH UTILIZATION PERFORMANCE

	Variance = 0.1	Variance = 0.5	Variance = 1	Variance = 5
Ut_MB_Mean	98.9%	98.7%	98.6%	98.2%
Ut_MB_Median	98.6%	98.6%	98.2%	97.8%
Ut_MB_IIR	98.8%	98.2%	98.2%	98%
Ut_MB_Fast	98.6%	98.4%	98.1%	98%

Where

Ut : represents the utilization of the available bandwidth, the formula is shown as (13):

$$\mathrm{Ut} = \frac{\sum_i \mathrm{TData_i}}{\sum_i \mathrm{AData_i}} \tag{13}$$

$\mathrm{TData_i}$ is the data size transmitted in the ith uplink subframe

$\mathrm{AData_i}$ is the maximun data size which the MS can transmit in the ith uplink subframe

Variance : the variance of the the available slots

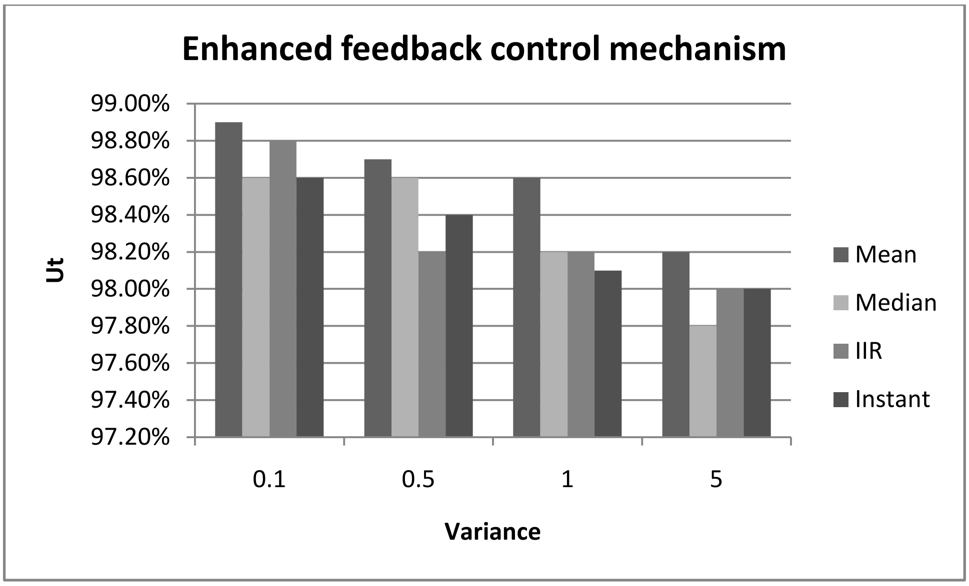

FIGURE 26 GRAPH OF TABLE 8

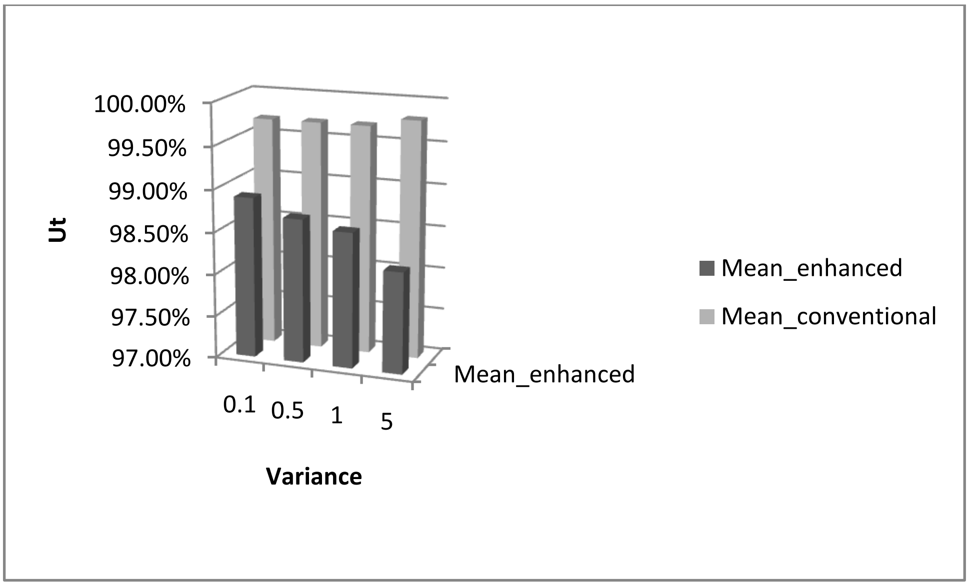

FIGURE 27 COMPARED WITH CONVENTIONAL IN MEAN

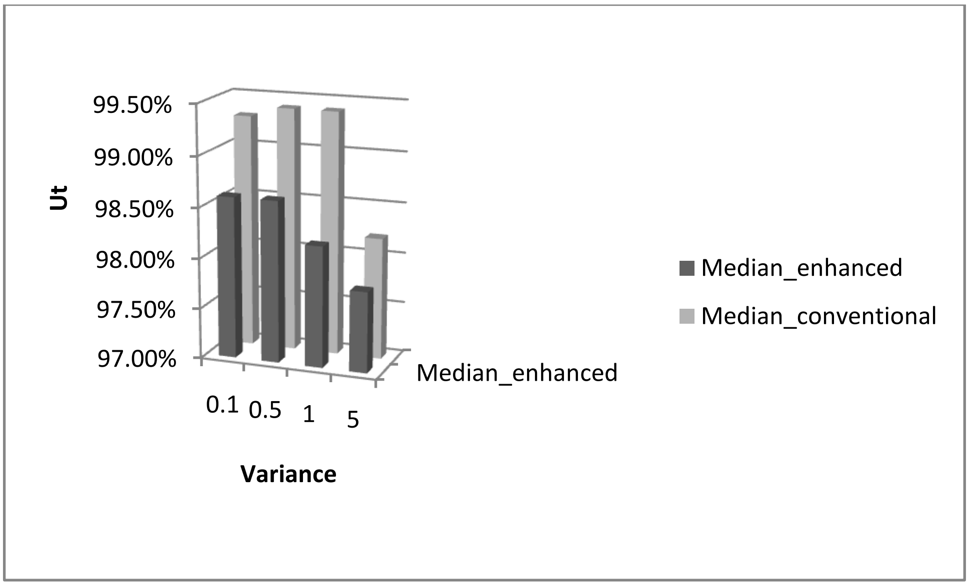

FIGURE 28 COMPARED WITH CONVENTIONAL IN MEDIAN

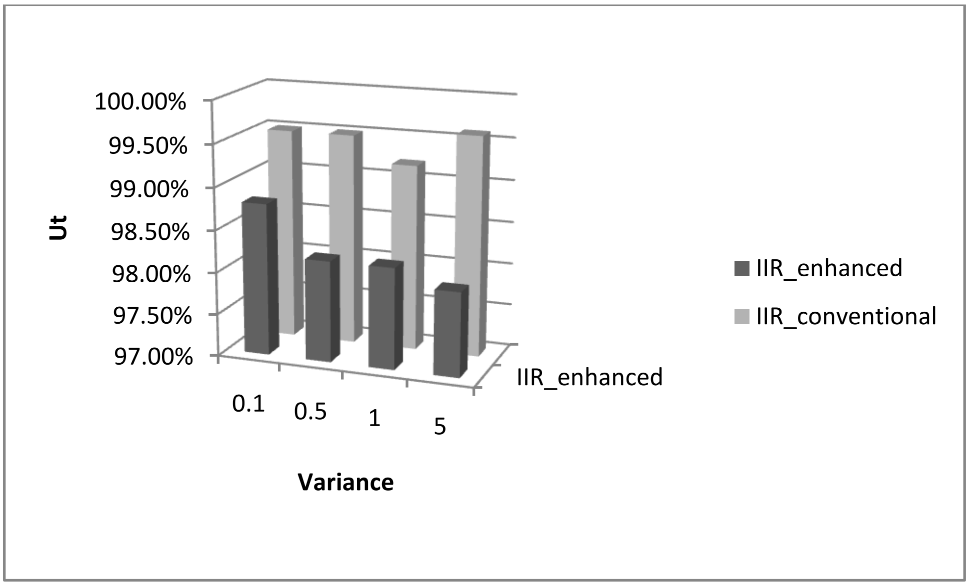

FIGURE 29 COMPARED WITH CONVENTIONAL IN IIR

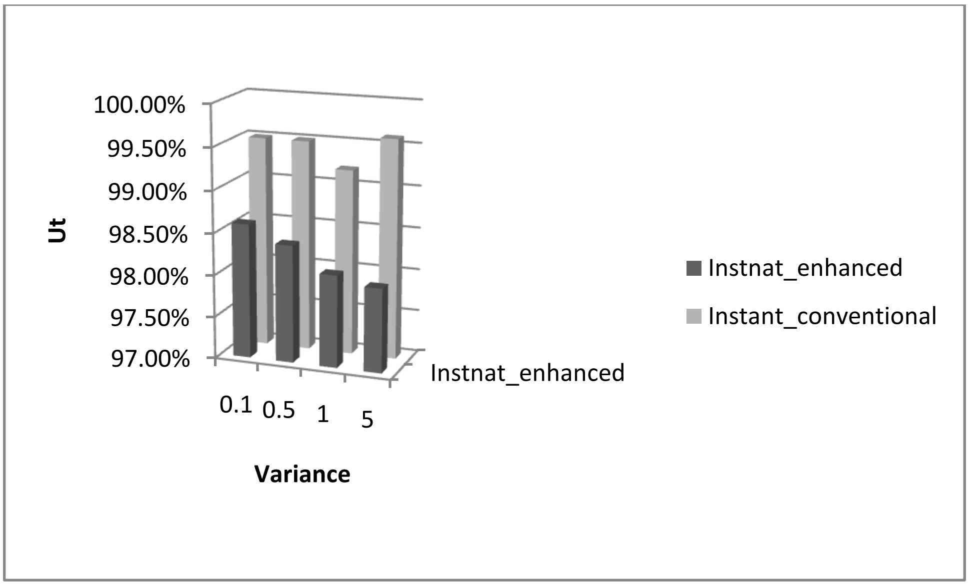

FIGURE 30 COMPARED WITH CONVENTIONAL IN INSTANT

5.2.2.4 Discussion

We should notice that the Delay Data stored in MAC buffer is bad to SVC decoder, because the SVC decoder will have less data to decode, and may cause video latency in the receiver. With enhanced feedback control mechanism, we modify the extracted video bitrate according to the size of data stored in MAC buffer. So we can greatly reduce the size of data stored in MAC buffer, and also improve the performance of the video latency. From the performance of enhanced feedback control mechanism, we can see that the probability of video latency is much lower than conventional feedback control mechanism and the size of data stored in MAC buffer is much smaller than conventional feedback control mechanism, too.

CHAPTER 6

CONCLUSION

6.1 Conclusion

In our thesis, the goal is to transmit real time video bitstream over wireless channel with less video latency at the receiver. Based on the goal, we have two contributions. One is that we propose a cross-layer architecture to transmit to real time video bitstream. The other is that we propose a enhanced feedback control machanism, which can decide a proper video bitrate for each GOP. With the contributions, we can have two benefits,

1) Smaller MAC Buffer
2) Lower probability of video latency

In the other hand, our research can also apply to the most popular compression system, H.264 system, because H.264 also has rate control mechanism which can control the video bitrate according to the bandwidth condition. And this mechanism just like the function of the SVC extractor. So in our proposed architecture, the SVC encoder can be replaced by the H.264 encoder, and the SVC extractor can be replaced by the H.264 rate control mechanism. And the enhanced feedback control mechanism can apply to this new cross-layer architecture undoubtedly.

6.2 Future Work

In our thesis, we focus on the optimization of unicast transmission. But the most powerful function of SVC system is multicast or broadcast transmission. We need to find solution to optimize the multicast or broadcast transmission. And thus we will have the total solution to transmit the real time video bitstream in WIMAX system, whether in unicast, multicast or broadcast.

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