Wireless Netw (2010) 16:113–123
DOI 10.1007/s11276-008-0118-1

Cross-layer mobile WiMAX MAC designs for the H.264/AVC scalable video coding

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Published online: 17 May 2008

Abstract To effectively provide the QoS of streaming services in mobile WiMAX systems, this paper proposes a cross-layer design between the streaming server and mobile WiMAX base stations. We will show that for each user the implementation of multiple connections with feedback information of the available transmission bandwidth is critical for supporting H.264/AVC-based scalable video coding in which the transmission packets can be further separated into multiple levels of importance. To quantify the performance, a mobile WiMAX simulation platform including both the video encoder/decoder and IEEE 802.16e MAC controls is developed.

Keywords Mobile WiMAX · IEEE 802.16e · SVC · Streaming service

1 Introduction

With increasing demands in high-data-rate services and mulitimedia applications in wireless communications, the IEEE 802.16 standard family [1, 2] and the associated Worldwide Interoperability for Microwave Access (WiMAX) forum are developed and formed to support the broadband wireless access (BWA) in a wireless metropolitan area network (WMAN). The IEEE 802.16-2004 standard [1] supports wireless data and multimedia services for fixed subscriber stations. More advanced, the IEEE 802.16e-2005 [2] has been standardized for supporting mobile broadband wireless access (MBWA) and the access technology will potentially be the fundamental technology for 4G standards.

For video streaming services, to achieve flexible bitstream adaptation for multimedia transmissions, the Joint Video Team (JVT) formed by the ISO/IEC MPEG and ITU-T is developing the scalable extension of the H.264/AVC standard [3–5]. The scalable video coding (SVC) uses layered structure to provide spatial, temporal, and quality (SNR, signal to noise ratio) scalability simultaneously. According to the network conditions and receiver capabilities, the pre-encoded SVC bitstream can be easily adapted by the streaming server to provide various spatial, temporal, and quality (SNR) resolutions. Further, the SVC layered structure put the data with different importance into different layers. The unequal erasure protection (UEP) can be easily incorporated with SVC to provide more protection to important data. With such features, the SVC bitstream is more suitable than the non-scalable bitstream when the video packets are transmitted over an error-prone channel with fluctuated bandwidth.

Some existing researches [6, 7] have discussed the scenario of scalable video streaming in wireless transmission. In this paper, we will propose cross-layer system architecture to effectively support quality of service (QoS) of streaming services which are encoded by SVC over the mobile WiMAX system. We also investigate the performance of multiple-connection mechanism based on simulation results and a subjective test. The rest of this paper is organized as follows: The scalable extension of H.264/AVC is introduced in Sect. 2. The basic medium access control (MAC) and physical (PHY) layers of IEEE 802.16e and the design flexibilities will be discussed in Sect. 3. Section 4 describes the proposed system architecture. Section 5 will provide simulation models and simulation results. Finally, conclusions are drawn in Sect. 6.

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2 Overview of scalable video coding

Scalable video coding (SVC) is a video coding technology that encodes the video at the highest resolution, and allows the bitstream to be adapted to provide various lower resolutions. There are three dimensions of scalability, including spatial, temporal, and quality (SNR) scalability. Spatial scalability means the bitstream can provide different spatial resolutions. Temporal scalability means various frame rates are available and SNR scalability means the visual quality is scalable.

To achieve the scalability, the video data is encoded into several layers. The lower layers contain lower resolution data. This data is more important because it provides basic video quality with low bit-rate. The higher layers contain the refinement data which refines the lower resolution data to provide higher resolution video. The refinement data is less important and can be removed when the bandwidth or decoding capability is not sufficient.

For wireless mobile communication, SVC has several advantages over the non-scalable video coding. For a single user, the transmission bandwidth is time-varying due to the mobility and the available resources are fluctuated. Besides, different users are located at different positions which lead to different transmission bandwidth. As a result, it is difficult to support all users with a single non-scalable bitstream. Moreover, there is no priority in the non-scalable bitstream. This leads to inefficient error protection because both more important data and less important data have the same performance in an error-prone channel. SVC provides simple solutions for these problems. The SVC bitstream can be easily adapted to the time-varying bandwidth at various receivers. The SVC layered data structure allows the important data to get more protection easily. As a result, these features provide more efficient video transmission over the error-prone channel with fluctuated bandwidth.

The scalable extension of the H.264/AVC is the latest SVC standard. It is developed by the Joint Video Team (JVT) formed by the ISO/IEC MPEG and ITU-T and is aimed to simultaneously provide three dimensioned scalabilities with good compression efficiency. To support the spatial scalability, the video is decomposed into several layers. Each layer can be encoded separately or get prediction from lower spatial layers to remove the redundancy. In each spatial layer, the data can be separated into several SNR layers with two means: The Coarse Grain Scalability (CGS) means that the bitstream can only be truncated at several pre-defined point, while the Fine Grain Scalability (FGS) means that the bitstream can be truncated at any position. In the standard, the first SNR layer in a spatial layer is restricted to CGS. To support the temporal scalability, a hierarchical prediction structure is used. The pictures in a group of picture (GOP) are dyadic decomposed into several layers. For more details of SVC, please refer to [8, 9].

3 Overview of IEEE 802.16e/mobile WiMAX

The IEEE 802.16 Working Group works on the technology of broadband wireless access and develops the standard of Wireless Metropolitan Area Networks (WMAN). The fixed broadband wireless access system IEEE 802.16d has been standardized in 2004, the standard is also called IEEE 802.16-2004. The mobile broadband wireless access system IEEE 802.16e, called IEEE 802.16e-2005 or mobile WiMAX, has been approved in December 2005. IEEE 802.16e provides mobility enhancements to IEEE 802.16-2004 for the purpose of supporting subscribers at a vehicular speed. The IEEE 802.16 Working Group mainly standardizes the specification of the medium access control (MAC) protocols and the physical layer (PHY). As compared with DSL and WiFi, the WiMAX family of standards expects to support a large number of subscribers over large coverage at high data rates. Besides, the WiMAX can provide broadband wireless access to locations in the rural areas or the locations that DSL cable is hard to build [10]. Due to the flexibility and efficiency, the WiMAX is expected to provide many kinds of services, including voice, internet, and multimedia services.

The IEEE 802.16-2004 PMP (point-to-multipoint) mode architecture consists of two kinds of station: subscriber station (SS) and base station (BS). In the IEEE 802.16e standard, the SS is replaced by the MS (mobile station) due to the support of the mobility. In the PMP mode, the radio resource management is centralized, where the BS controls the allocation of all resources within the cell.

3.1 Overview of the PHY layer of WiMAX

The PHY layer of WiMAX comprises different configurations. In this study, the TDD (time division duplex) mode of WirelessMAN-OFDMA PHY layer will be considered.

Figure 1 shows the concept of WirelessMAN-OFDMA frame structure in TDD mode. The frame structure contains the downlink subframe and uplink subframe. TTG (transmit/receive transition gap) and RTG (receive/transmit transition gap) are the transmission gaps between the downlink subframe and uplink subframe. The two gaps allow the antenna to switch from the transmit mode to receive mode and from the receive mode to transmit mode. In the TDD mode, the uplink subframe follows the downlink subframe on the same frequency band. Moreover, each subframe is further divided into physical slots for the purpose of bandwidth allocation and identification of PHY

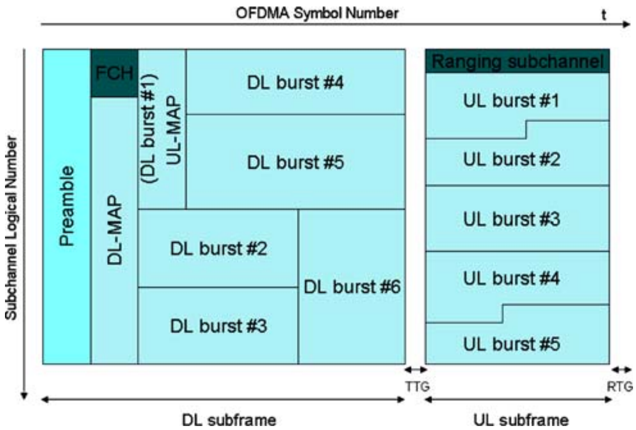

Fig. 1 The frame structure of WirelessMAN-OFDMA in TDD mode

transitions. The downlink subframe contains the following parts. The preamble occupies the first symbol that can be used for synchronization and channel estimation. FCH (frame control header) with a fixed slot size specifies the resource allocation of the DL-MAP. DL-MAP message specifies the resource allocation of DL data bursts and UL-MAP specifies the resource allocation of UL data bursts. The remaining parts in the downlink subframe are DL data bursts. The DL-MAP and UL-MAP messages are used for the resource allocation of DL and UL data bursts, including burst-MS pairing information, the modulation and coding schemes of each data burst. The uplink subframe contains the following parts. The ranging subchannel is used for initial ranging, periodic ranging and contention-based bandwidth request. Ranging subchannel is dynamically allocated by the MAC layer and is specified in the UL-MAP message. The initial ranging transmission shall be used by the MS to synchronize to the system for the first setup. Periodic-ranging will be periodically executed for updating the system time and transmission power. Bandwidth request is used by the MS to request the uplink allocation. The remaining parts of the uplink subframe are UL data bursts.

3.2 Advance modulation and coding (AMC)

The IEEE 802.16 defines several burst profiles that are the combination of modulation and coding scheme in each PHY configuration. With link adaptation, the system can decide the proper modulation and coding scheme according to the current CINR (carrier to interference and noise ratio) value. In IEEE 802.16e, the CINR of each mobile station may change with time. At the beginning of a frame, the base station should decide the burst profile of each DL and UL data burst. For the DL data burst, the base station can decide the burst profile of the DL data burst according to the feedback DL channel condition in the UL fast feedback channel (or called channel quality indication channel, CQICH). For the decision of the burst profile of the UL data burst, the base station can measure the signal strength of the transmitted UL data burst and decide the burst profile for that mobile station. Besides, there is an optional mechanism called UL sounding that supports smart antenna or MIMO (multiple input multiple output). The UL sounding signal is similar to uplink pilot. The base station can measure the signal strength of the UL sounding signal transmitted by the mobile station. Then the base station can translate the measured UL channel condition to a proper burst profile for the uplink transmission of the mobile station.

3.3 Overview of the MAC layer of WiMAX

The MAC layer of WiMAX mainly supports a point-to-multpoint (PMP) architecture, and can support mesh architecture optionally. The MAC layer is designed for supporting the different QoS (quality of service) requirements of different types of applications. Besides, it is connection-oriented; each service is mapped to the minimum of one connection. The MAC layer of WiMAX consists of three sublayers including convergence sublayer, common part sublayer, and privacy sublayer. Because the MAC layer of WiMAX must support the various backhaul network like asynchronous transfer mode (ATM) network and IP-based networks, *the convergence sublayer* is to converge different types of transport-layer traffic to a MAC connection. As mentioned before, the MAC is connection-oriented; each service, including the connectionless service is mapped to a connection. Each connection will be identified by a 16-bit connection identifier (CID). This sublayer classifies the service data units (SDUs) to a proper connection with specific QoS parameters. *The common part sublayer* controls most parts of MAC functionalities like fragmentation, packing, scheduling, retransmission and so on. First, for the tradeoff between the transmission efficiency and robustness, fragmentation, packing, or combined fragmentation and packing mechanisms are adopted. Besides, the 802.16 MAC uses a request-grant mechanism for resource allocation. The request-grant mechanism can provide some features like scalability, efficiency, and self-correcting. *The privacy sublayer* is for data encryption and it can provide the security on network transmission.

4 System architecture

The end-to-end transmission of a streaming video in the mobile WiMAX system is depicted in Fig. 2. As shown, the last mile transmission system is IEEE 802.16e/Mobile

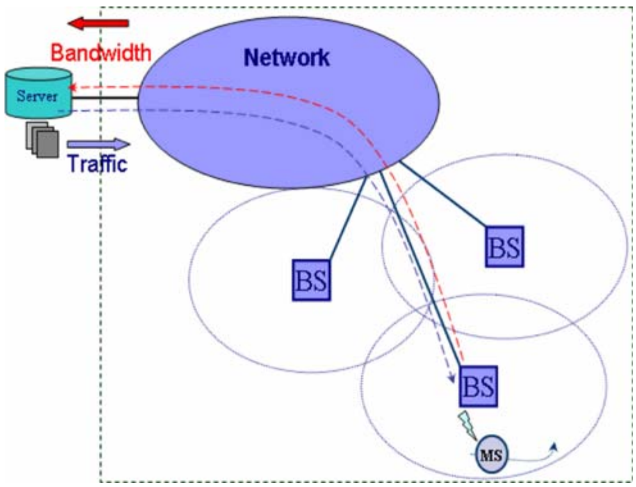

Fig. 2 The end-to-end mobile WiMAX system

WiMAX which consists of base stations (BSs) and mobile stations (MSs). A multi-media server (based on H.264/AVC) and WiMAX subsystem are inter-connected by an IP-based backhaul network. The multimedia service in this study is the streaming service encoded by H.264/AVC based scalable video coding (SVC).

When the MS intends to initiate a streaming service, it starts the call setup process (Dynamic Service Addition ~ DSA dialogue) with the connected BS. The handshaking process exchanges system parameters and QoS parameters including minimum latency, maximum sustain rate, minimum reserved rate, request policy and so on. In the proposed cross-layer design, the BS will periodically report the average bandwidth to the streaming server according to the residual BS capacity and RF condition on the period of *Report Period*. The streaming server can adjust the number of the video packets for the downlink transmission. Because the video files are encoded by the scalable video coding, it comprises some base layers and enhancement layers. The streaming server can prioritize the packets and decide which packets can be delivered depending on the available bandwidth. As shown in Fig. 3,

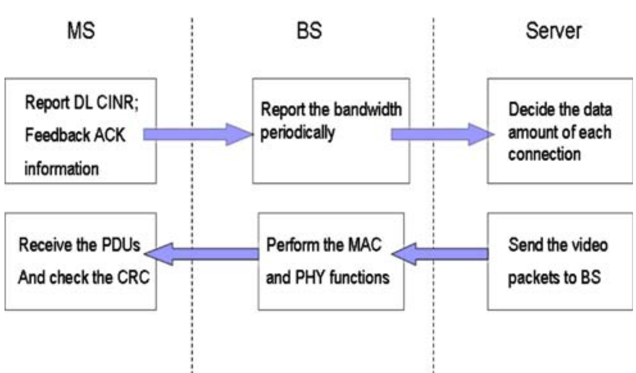

Fig. 3 The control flow between each entity

the following subsections will describe the jobs of the streaming server, base station MAC, the mobile station, and the interaction between each entity.

4.1 The jobs of the streaming server

In each of the *Report Period*, the BS will request a target bit-rate from the SVC streaming server. The SVC streaming server analyzes the bitstream at the group of pictures (GOPs) that covers the current *Report Period*, as shown in Fig. 4. In the SVC bitstream, the data at lower spatial-SNR resolution is more important. In each spatial-SNR layer, the lower temporal layer is more important. Therefore, according to the requested bit-rate from the BS, the lower spatial-SNR layers are firstly extracted. At the spatial-SNR layer where the bit-rate cannot cover all the data, only the data at the lower temporal layers will be extracted. With FGS coding, the data at the same temporal layers can be further truncated at any position to provide the exactly request bit-rate. The extracted data are then sent to the BS.

It should be mentioned that some GOPs may belong to two *Report Periods* (such as the GOP(x + 2) in Fig. 4). For such GOPs, the data that already sent in the first *Report Period* will not be sent again. However, if the second *Report Period* allows higher bandwidth, the remaining data in such GOPs will be transmitted. This makes the video quality smoother when the bandwidth changes frequently. Depend on the pre-load time of the related streaming services, the number of the overlapped GOPs between the *Report Periods* can be further extended to provide more smooth video quality. Further, this structure is also possible to enable the retransmission at the streaming server with suitable pre-load time.

To support multiple connections between the BS and MS, the data sent to the BS from the streaming server is allocated into several connections according to importance levels, as shown in Fig. 5. The more important data is allocated to the connection that has more protection with higher transmission priority and MAC retransmission. To address the effect of bandwidth fluctuation, in the proposed two-connection implementation, the server allocates 80% data which includes mostly the important data at the first connection, and put the remaining data at the second connection. This allows the BS to re-transmit only the

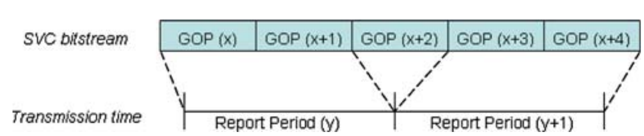

Fig. 4 The GOP(x), (x + 1), (x + 2) and the GOP(x + 2), (x + 3), (x + 4) will be analyzed and transmitted during Report Period (y) and Report Period (y + 1), respectively

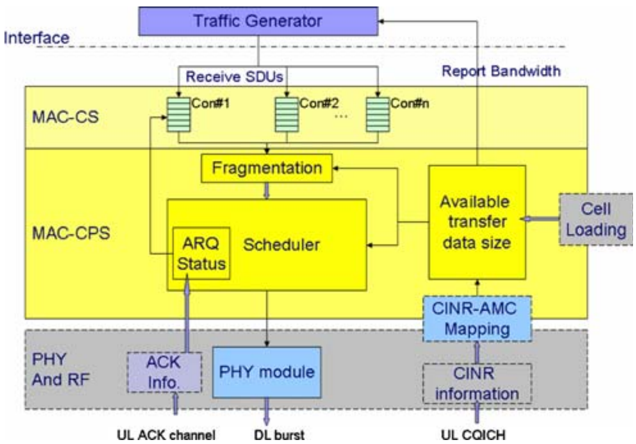

Fig. 5 Base station MAC control modules

important data when the real bandwidth is smaller than the expected bandwidth. In the future work, the percentage of the data that put in the first connection should be adaptively according to the estimate channel model, which may further improve the overall video quality.

4.2 The jobs of the base station

Upon receiving those packets in the WiMAX subsystem, as shown in Fig. 6, those packets will be first stored as MAC service data units (SDUs) in the queues. The stored SDUs will be treated differently depending on the information types and MAC controls. The base station should collect the downlink (DL) channel condition and translate the information to relative available bandwidth. Besides, the base station will support multiple connections and have the capability of handling the connections by different means. In other words, if it establishes multiple MAC connections, then it should provide the flexibility of providing different QoS and priorities to different connections. Because the corresponding service data unit of each connection is from the server, the server should also have the capability of categorizing the video bitstream to multiple level of importance as mentioned.

4.2.1 RF bandwidth estimation and bandwidth report

The communication period between the BS and the server is called *Report Period*. The *Report Period* parameter must be a proper value and it depends on the buffer size in the mobile station and base station. The base station should report the average bandwidth to the server for a period of *Report Period*. In this study, the value of *Report Period* is set to 0.5 s which is larger than the network delay of 100–200 ms. In a time-varying channel, the reporting period might not be sufficient to reflect the real channel condition and any further delay will make this situation worse. Later in this study, we will show that with the increase of cell loading and fluctuated bandwidth, different treatments of transmitted frames are important to maintain the performance of streaming services.

For each WiMAX frame, the BS collects the reported DL CINR information from CQICH in the uplink subframe and then the BS decides the proper modulation and coding scheme for DL transmission according to the CINR-AMC mapping table, as shown in Fig. 6.

Besides, we define the ratio of occupied DL OFDMA slots to total DL OFDMA slots as *Cell Loading* (the ratio does not consider the overheads like preamble, FCH, and MAP message). The OFDMA slot is defined by the standard [2] and Fig. 7 shows an example. By combining the information of AMC scheme and residual capacity of the cell, the BS can calculate the available bandwidth for the streaming service. Finally the BS can calculate the average available bandwidth for the streaming service over the duration of *Report Period* and then report it to the server.

4.2.2 Multiple connections

In WiMAX system, the service flow is mapped to a connection in the MAC layer. In fact, the 802.16 MAC is connection-oriented and each service will be mapped to a connection even though it is a connectionless service. Each MAC connection will be identified by a 16-bit CID field and each MAC connection will be related to a set of QoS parameters according to its service type. The 802.16 MAC is

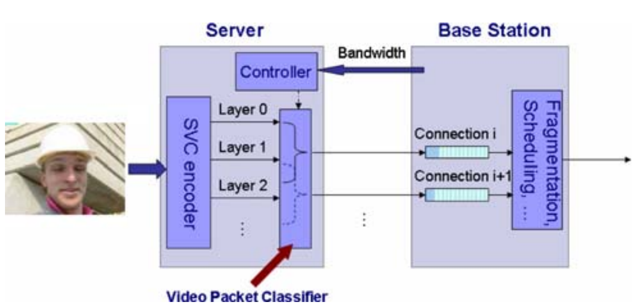

Fig. 6 The interaction between the server and the base station

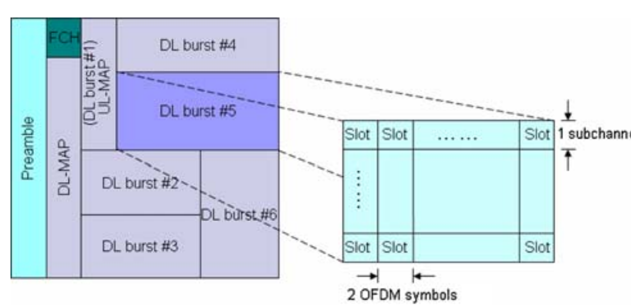

Fig. 7 The definition of an OFDMA slot in PUSC mode

flexible and it allows multiple connections for a mobile subscriber station at the same time. The multiple connections can belong to same or different service types and can be related to same or different QoS parameters. In this study, the streaming service flows can be classified to rtPS or ertPS. Because the streaming server adopts scalable video coding and it can decide which video packet is more important and which is not, the BS can establish two or more connections with the server. In this study, a two-connection scenario is adopted and compared with the performance of one-connection scenario. In the two-connection scenario, the first connection is for important video packets and will be given for higher priority in transmission and protection. The second connection is for less important video packets (like some enhancement packets) and we give it lower priority as compared to the first connection.

4.2.3 Fragmentation

Fragmentation is a process that a MAC SDU is divided into multiple MAC protocol data units (PDUs). Because of the mobility, the channel condition will change with time and the proper PDU size will also change with time. The fragmentation of the SDU packets is critical to improve the error performance since the SDU sizes of the streaming video have a large range.

To achieve better error performance, we will suggest a cross-layer design by utilizing the RF information for having the better decision of how to fragment the MAC SDUs. In our simulations, we adopt the Gilbert-Elliott channel model [14] to acquire the bit error rate of each WiMAX frame. We will set a target PDU error rate and hence the proper PDU size can be calculated.

4.2.4 ARQ and MAC retransmission

Besides of the fragmentation, the MAC retransmission can also be adopted to increase the robustness and hence decrease the video packet loss rate. When the MS receives its downlink data burst, it should check the correctness by the CRC filed of each MAC PDU and should feedback the ACK information to the BS through the fast feedback channel in the uplink. In our proposed scheme, the first connection used to transmit important video packets will be managed as an ARQ-enable connection. The other connections can be managed as either an ARQ-enable connection or not. The decision should depend on the tradeoff between the robustness and the resource efficiency.

4.2.5 Handover operations

Because the mobile subscriber station can move from cell to cell, the handover control needs to be considered. There are some existing literatures [11, 12] discussing the enhanced MAC layer handover algorithms. These algorithms can be adopted for the purpose of reducing the handover gap which may result in packet loss. It is clear that an effective handover operation is critical to improve the seamless transition from cell to cell. In mobile WiMAX, fast base station switch (FBSS) and macro diversity handover (MDHO) are proposed to establish robustness of the handover transmission. We believe that, with proper handover controls and upper layer's coordination, the transmission gap can be managed under an acceptable range. Besides, the buffering mechanism in our proposed architecture can be used to further reduce the effect of the transmission gap caused by the handover process. In this study, a perfect seamless handover is assumed for simplicity and the impact of the handover gap will be taken into consideration in the future works.

4.3 The jobs of the mobile station

The mobile station should monitor the downlink subframe and report the DL CINR to the base station through the channel quality indication channel (CQICH) in the uplink subframe. Besides, it should feedback the ACK information for each received ARQ block.

5 Simulations

5.1 Simulation model

The simulation model of WiMAX environment is established by MATLAB. For PHY configuration, WirelessMAN-OFDMA TDD mode and PUSC (Partial Usage of Subchannel) are adopted. The FFT size is 2048 and the bandwidth is 6 MHz. From the standard, we can find the OFDMA channelization parameters [1]. The relative parameters are listed in Table 1.

The error model is referred to the Gilbert-Elliott channel model [13–16]. The CINR ranges of the burst profile (the adaptive modulation and coding scheme) are listed in Table 2 which is referred to the settings of ITRI's[1] simulation platform.

The controls of each MAC-level connection mainly comprise three parts: the first is the fragmentation, the second is the scheduling, and the third is the MAC retransmission. For the fragmentation, we will set a target PDU error rate of 5% and decide the proper fragment PDU size. For the scheduling controls, the early deadline first (EDF) concept is adopted since the traffic is the real-time traffic. For the MAC retransmission, the ARQ mechanism

[1] ITRI is Industrial Technology Research Institute of Taiwan.

Table 1 The WirelessMAN-OFDMA PHY parameters

Parameters	Values
Channel bandwidth	6 MHz
FFT size (N_{FFT})	2048
Number of sub-channels	60
Useful symbol time (T_b)	298.667 µs
Guard time ($T_g = 1/8T_b$)	37.333 µs
OFDMA symbol time ($T_s = T_g + T_b$)	336 µs
Frame duration	5 ms

Table 2 The CINR range of each burst profile

Burst profile	CINR range [dB]
64QAM-1/2	CINR ≧ 24.78
16QAM-1/2	24.78 > CINR ≧ 16.02
QPSK-1/2	16.02 > CINR ≧ 7.51

Table 3 The parameters used in the simulation

Parameters	Value
Maximum latency	550 ms
Max. MAC retransmission times	6
Average BER of the channel model	2×10^{-4}
Bad state BER of the channel model	10^{-3}
Good state BER of the channel model	10^{-4}
B_{av} (defined in [15])	2

of each MAC connection can be enabled. Table 3 shows the parameters used in the simulations.

The SVC bitstream used in the simulation is encoded to provide two spatial layers including QCIF (Quarter Common Intermediate Format, 176 × 144) and CIF (Common Intermediate Format, 352 × 288). As shown in Table 4, the QCIF resolution has one SNR layer and the CIF resolution includes four SNR layers. The purpose of this configuration is to provide a minimum video quality at QCIF resolution even during heavy cell loading and poor network conditions, while allowing good video quality at CIF resolution at normal conditions.

In each of the spatial-SNR layers, the GOP size is limited to 8 pictures to reduce the buffer requirement at the mobile station but still provide four layers temporal scalability through the hierarchical prediction structure. The intra-pictures are inserted every 64 pictures (8 GOPs) to provide the error recovery point. With the five spatial-SNR layers and the four temporal layers in each spatial-SNR layer, up to 20-layer bitstream adaptation is allowed. Further, the FGS is used from the CIF-SNR1 to CIF-SNR3 layers, such that the bitstream can be truncated at any point to achieve the requested bit-rate.

The test sequence is making up from 13 commonly used MPEG test sequences to form a 3,600 pictures video, which including bus, football, foreman, mobile, city, crew, harbour, soccer, coastguard, container, mother daughter, stefan, and table tennis. The average bit-rate of the SVC bitstream at various spatial-SNR and temporal resolutions are shown in Table 4.

5.2 Simulation results

In the simulations, we will compare the transmitting performance of one-connection scenario and multiple-connection scenario. In general, the number of the connection for transmitting the SVC-based video bitstream can be more than two. In this paper, for simplicity, we adopt two-connection scenario in order to investigate the necessity of having the multiple-connection for streaming services.

5.2.1 MAC retransmission

In this subsection, we will show that the MAC retransmission is needed to ensure the enough robustness for the video packets, especially for the important video packets. Figure 8 shows the SDU error rate under the different maximum MAC retransmissions. This figure is based on the cell loading of 75%. As shown, the SDU error rate decreases if we allow more MAC retransmissions. If we want to guarantee the SDU error rate to a level about 10^{-3} to 10^{-4}, then the maximum MAC retransmissions are about 5–7 times. In our simulations, we will set the maximum MAC retransmission times to 6, as shown in Table 3.

5.2.2 One-connection scenario

5.2.2.1 Data rate Figure 9 shows the feature of SVC. When the cell loading increases, the available bandwidth

Table 4 The average bitrate of the SVC bitstream at various spatial-SNR and temporal resolutions

Spatial-SNR layers	Bitrate at 3.75fps	Bitrate at 7.5fps	Bitrate at 15fps	Bitrate at 30fps
QCIF-SNR0	10.9801	14.8971	19.3172	23.9405
CIF-SNR0	56.6238	67.7029	81.5013	103.4000
CIF-SNR1	152.9394	172.2780	205.6717	247.6827
CIF-SNR2	335.3745	387.9433	456.7107	539.1969
CIF-SNR3	616.8722	747.8123	911.1053	1095.1940

becomes lower and the base station will report the bandwidth to the streaming server. The server will respond to the available transmission bandwidth. As a result, if the cell loading increases, the number of video packet delivered by the server will be reduced. In consequence, the data rate will decrease when the cell loading increases. The behavior of the server is important to make sure the queuing delay of the video packets will not be increased significantly.

5.2.2.2 SDU failure rate and Type0 SDU failure rate The SDU failure rate is defined by the sum of SDU error rate and SDU lost rate. The SDU lost rate results from the violation of the maximum latency of the real-time service. Figure 10 shows the SDU failure rate under the different cell loading. As shown, the SDU failure rate will increase with the increase of the cell loading.

By defining the Type0 SDU as important SDU for the video streaming, the Type0 SDU failure rate will then be the failure rate of important video packets. In one-connection scenario, the SVC-encoded video packets of different layers are sent within one connection. As shown in Fig. 10, the SDU failure rate of important video packets (Type0 SDUs) and the SDU failure rate of total video packets in the one-connection scenario are increased together with the increase of the cell loading. Since the one-connection does not provide extra protection to the important video packets, the SDU failure rate of important video packets will also increase when the overall SDU failure rate increases. It also means that the one-connection scenario cannot guarantee the enough performance of the important video packets when cell loading is high.

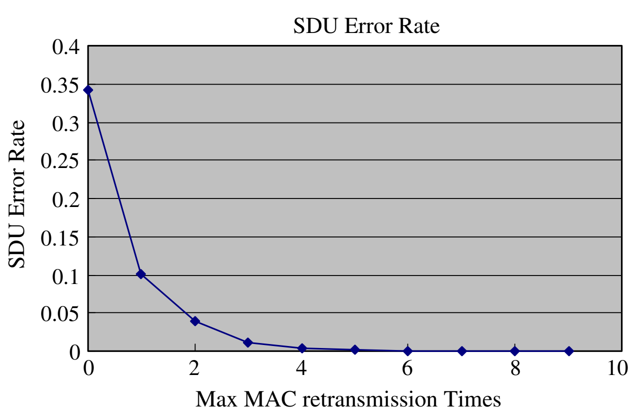

Fig. 8 The SDU error rate under the different maximum MAC retransmission times

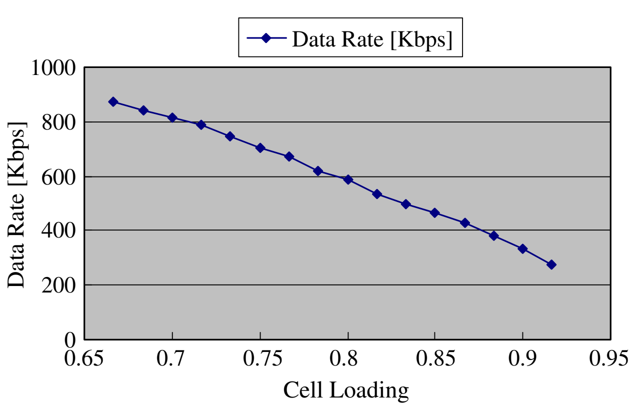

Fig. 9 The data rate of the one-connection scenario under the different cell loading

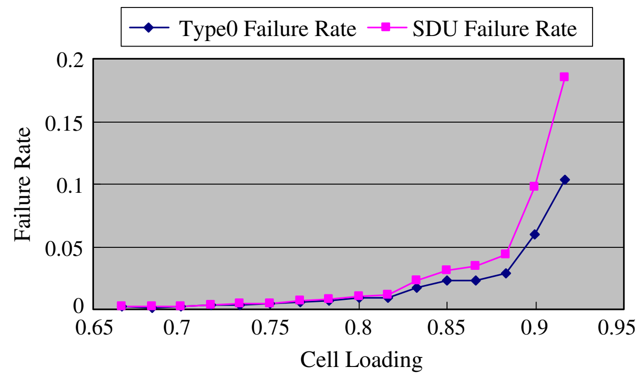

Fig. 10 The overall SDU failure rate and important SDU failure rate of the one-connection scenario under the different cell loading

5.2.3 Two-connection scenario

5.2.3.1 Total data rate of the two connections In the two-connection scenario, the first connection (CON#1) carries more important video packets and the second connection (CON#2) carries less important video packets. Figure 11 shows the data rate of the two-connection scenario under different cell loading. Since the scalable video coding is adopted and the server will send fewer video packets to the base station when the bandwidth is lower, the total data rate decreases when the cell loading increases. As mentioned

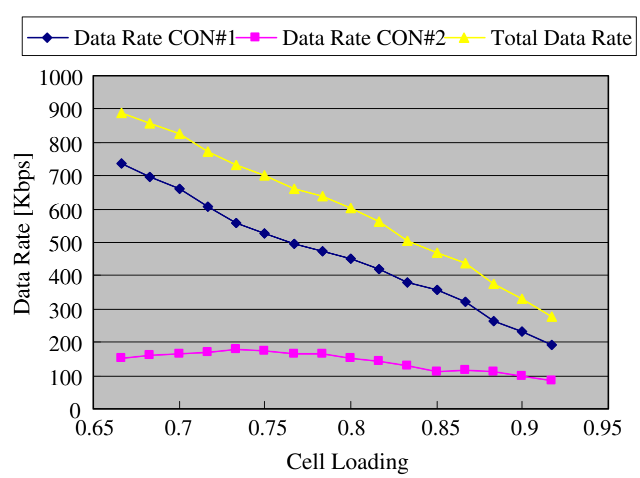

Fig. 11 The data rate of the two-connection scenario under the different cell loading

before, the server will allocate the 80% bandwidth to the first connection and the first connection contains the more important video packets including the base layer data and some enhancement layer data. When the reported bandwidth decreases, the server will still send the base layer data (the most important data) but reduce the enhancement layer data to the first connection. This behavior will lead to the fact that the data rate of the first connection becomes lower when the cell loading increases.

5.2.3.2 SDU failure rate From Fig. 12, as compared to one-connection case, the two-connection case provides better protection to important video packets where the first connection (CON#1) is basically insensitive to the cell loading. The second connection (CON#2), on the other hand, will increase its failure rate with the increase of the cell loading.

5.2.3.3 Two-connection scenario versus One-connection scenario Figure 13 shows the SDU failure rate of important video packet in two-connection scenario and one-connection scenario. By considering the failure rate of the important video packets only, from Fig. 13, it is obvious that the two-connection scenario has the better control of the failure rate of the important video packets even when the cell loading increases. This performance has a direct impact on the actual perceived visual quality which will be shown later in the subjective test.

5.3 Subjective test

In the subjective test, there are 18 viewers and we will compare the visual performance between two-connection scenario and one-connection scenario. There are four cases, each case has different average data rate. The average data rate of each case is: case 1 > case 2 > case 3 > case 4. For each video sequence, there are five scores, ranged from 1 to 5.

From Fig. 14, the subjective test indicates that the visual performance of two-connection scenario is better than

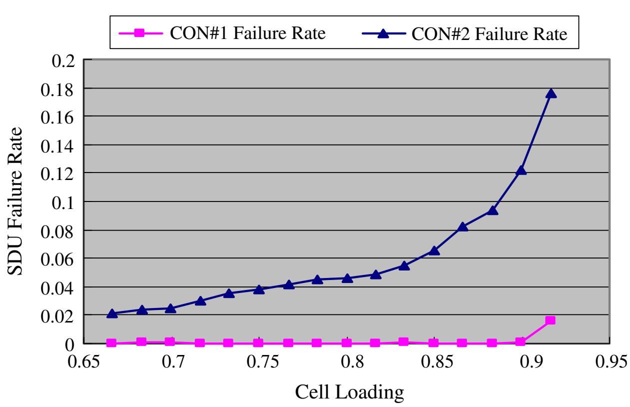

Fig. 12 The SDU failure rate of the two-connection scenario under the different cell loading

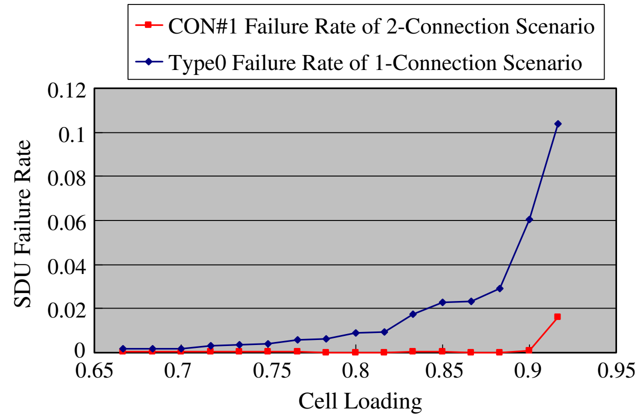

Fig. 13 The important SDU failure rate of the two-connection and one-connection scenario under the different cell loading

one-connection scenario in all cases especially when the data rate is lower. Figure 15 shows the distribution of the score comparisons of the two-connection scenario and the one-connection scenario. For each case, if the ranking of two-connection scenario is higher than the ranking of one-connection scenario, we classify it as a "Better" case. If the ranking of two-connection scenario and the ranking of one-connection scenario are the same, we classify it as the "Similar" case. If the ranking of two-connection scenario is lower than the ranking of one-connection scenario, we classify it as a "Worse" case. We can see that most viewers consider that the visual performance of two-connection scenario is better than that of one-connection scenario.

6 Conclusions

In this study, the transmission performance of the SVC-based streaming services over the mobile WiMAX system is

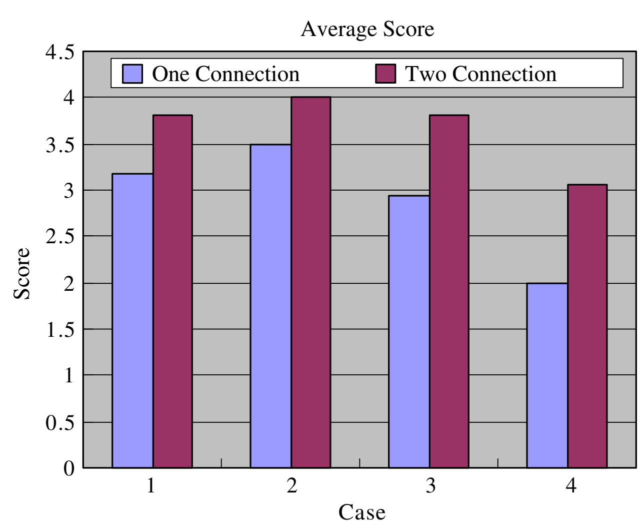

Fig. 14 The average scores of the subjective test

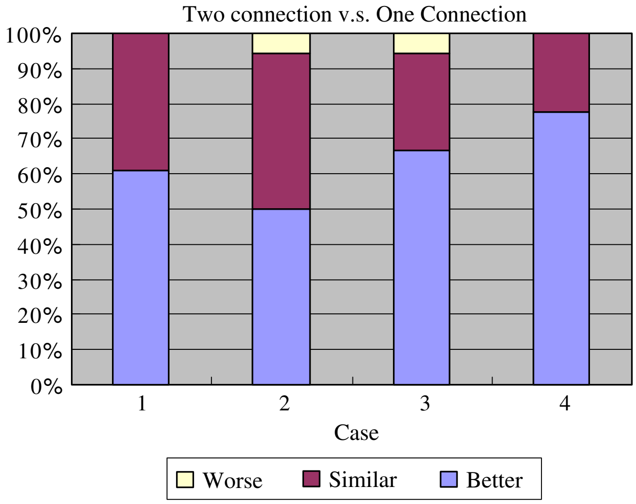

Fig. 15 The score comparison of the subjective test

investigated. By introducing a cross-layer design between the transmission server and MAC of the base station, we have shown that the multiple-connection scenario, which separates SVC packets into multiple levels of importance, and the feedback of the effective average bandwidth, considering both the residual BS capacity and RF conditions, can effectively control the SDU failure rate of the important video packets regardless of the cell loading. Also, as compared to one-connection scenario, the implementation of two-connection scenario is necessary if the error-prone channel with fluctuated bandwidth is considered. Finally, the subjective tests have also been done to verify above conclusions by performing user perceived video quality tests.

Acknowledgement This work is sponsored by MediaTek-NCTU research center and Taiwan MOE ATU Plan.

References

1. IEEE 802.16-2004. (2004). IEEE standard for local and metropolitan area networks. Part 16: Air interface for fixed broadband wireless access systems, June 2004.
2. IEEE 802.16e-2005. (2005). IEEE standard for local and metropolitan area networks. Part 16: Air interface for fixed and mobile broadband wireless access systems. Amendment 2: Physical and medium access control layers for combined fixed and mobile operation in licensed bands and Corrigendum 1, February 2005.
3. Advanced Video Coding for Generic Audiovisual Services. (2003). ITU-T and ISO/IEC JTC1, ITU-T Recommendation H.264—ISO/IEC 14496–10 AVC.
4. Joint Draft 5: Scalable Video Coding. (2006). ITU-T and ISO/IEC JTC1, JVT-R201, January 2006.
5. Joint Scalable Video Model JSVM-5. (2006). ITU-T and ISO/IEC JTC1, JVT-R202, January 2006.
6. Schierl, T., Hellge, C., Mirta, S., Grüneberg, K., & Wiegand, T. (2007). Using H.264/AVC-based scalable video coding (SVC) for real time streaming in wireless IP networks. *IEEE International Symposium on Circuits and Systems*.
7. Hillestad, O. I., Perkis, A., Genc, V., Murphy, S., & Murphy, J. (2007). Adaptive H.264/MPEG-4 SVC video over IEEE 802.16 broadband wireless networks, Packet Video 2007 (pp. 26–35), 12–13 Nov. 2007.
8. Schwarz, H., Marpe, D., & Wiegand, T. (2005). Comparison of MTCF and closed-loop hierarchical B pictures, ITU-T and ISO/IEC JTC1, JVT-P059, July 2005.
9. Ohm, J. R. (2005). Advances in scalable video coding. *Proceedings of the IEEE , 93*(1), 42–56.
10. Ghosh, A., Wolter, D. R., Andrews, J. G., & Chen, R. (2005). Broadband wireless access with WiMax/802.16: Current performance benchmarks and future potential. *IEEE Communications Magazine*, February, 129–136.
11. Choi, S., Hwang, G. H., Kwon, T., Lim, A. R., & Cho, D. H. (2005). Fast handover scheme for real-time downlink services in IEEE 802.16e BWA system, VTC 2005-Spring. 2005 IEEE 61st.
12. Lee, D. H., Kyamakya, K., & Umondi, J. P. (2006). Fast handover algorithm for IEEE 802.16e broadband wireless access system, *Wireless Pervasive Computing*.
13. Elliott, E. O. (1963). Estimates of error rates for codes on burst-noise channels. *Bell System Technical Journal, 42*(9), 1977–1997.
14. Gilbert, E. N. (1960). Capacity of a burst-noise channel. *Bell System Technical Journal, 39*, 1253–1265.
15. Sakakibara, K. (2000). Performance analysis of the error-forecasting decoding for interleaved block codes on Gilbert-Elliott channels. *IEEE Transactions on Communications*, March, 386–395.
16. Yee, J. R., & Weldon, E. J. (1995). Evaluation of the performance of error-correcting codes on a Gilbert channel. *IEEE Transactions on Communications*, August, 2316–2323.

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