RADIO RESOURCE MANAGEMENT OF HETEROGENEOUS SERVICES IN MOBILE WIMAX SYSTEMS

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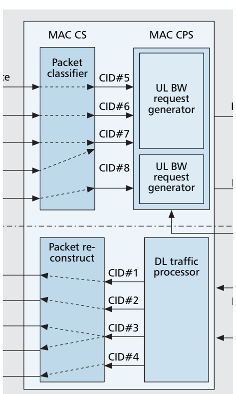


The authors evaluate the downlink performance of the Mobile WiMAX cellular system with different radio resource management, especially the scheduler for QoS control and the implementation of multi-connection for streaming applications.

ABSTRACT

IEEE 802.16e, known as Mobile WiMAX, has gained much attention recently for its capability to support high transmission rates in cellular environments and QoS for different applications. Beyond what the standard can define, in order to effectively support video streaming, VoIP, and data services, proprietary radio resource management, including multiconnection assignment, scheduling controls, and call admission controls, are essential. In this study we evaluate the downlink performance of a Mobile WiMAX cellular system with different radio resource management, especially the scheduler for QoS control and the implementation of multiconnection for streaming applications.

INTRODUCTION

The standards of the IEEE 802.16 family [1, 2] provide fixed and mobile broadband wireless access (BWA) and promise to deliver multiple high-data-rate services over large areas. The Worldwide Interoperability for Microwave Access (WiMAX) Forum streamlines the implementation of IEEE 802.16 standards. Based on complexity and flexibility management of the medium access control (MAC) and physical (PHY) layers, the IEEE 802.16 family is expected to support better quality of service (QoS).

The first IEEE 802.16 standard, approved in 2001, is in the 10–66 GHz range for line-of-sight wireless broadband services. In order to overcome the disadvantage of line-of-sight links, IEEE 802.16a, completed in 2003, is in the 2–11 GHz band for non-line-of-sight wireless broadband services. IEEE 802.16d, approved in 2004, named IEEE 802.16-2004, is designed for fixed wireless communications. The new IEEE 802.16e standard extends the 802.16d standard and provides mobility support in cellular deployments [3].

Even flexible bandwidth allocation and QoS mechanisms are provided in the IEEE 802.16 standard; the details of scheduling, admission control, and reservation management are left undefined. This article will focus on evaluating IEEE 802.16e system-level performance with different radio resource management for mixed VoIP and non-real-time services. Besides, the performance of video streaming services will also be investigated.

The rest of this article is organized as follows. We discuss the PHY and MAC layers of mobile WiMAX. The service classes and service data flow are discussed. We introduce the radio resource management implemented in mobile WiMAX. The performance of mobile WiMAX is investigated. Finally, conclusions are drawn.

OVERVIEW OF THE PHY AND MAC LAYER OF WIMAX

IEEE 802.16e defines only the PHY and MAC layers. Based on the specifications, a simulation platform is developed to investigate the performance of mobile WiMAX. In the next few subsections we discuss the PHY and MAC layers in more detail.

PHY LAYER

Basically, the PHY layer of WiMAX comprises different configurations. Among them, the time-division duplex (TDD) mode of the wireless MAN (WMAN) orthogonal frequency-division mutliple access (OFDMA) PHY layer is selected by most vendors. Here, our discussion focuses only on the TDD mode.

Figure 1 shows the concept of a WMAN-OFDMA frame structure in TDD mode. The frame structure contains the downlink subframe and the uplink subframe. Transmit/receive transition gap (TTG) and receive/transmit transition gap (RTG) are the transmission gaps between the downlink and uplink subframes. These two gaps allow the antenna to switch from transmit mode to receive mode and from receive mode to transmit mode. In 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 for multiple users. The downlink (DL) subframe contains the following parts. The preamble occupies the first symbol that can be used for synchronization and channel estimation. The frame control header (FCH) with a fixed slot size specifies the resource allocation of the DL_MAP. The DL_MAP and UL_MAP messages are used for the resource allocation of DL and UL data bursts, including burst-mobile station (MS) pairing information, modulation, and coding schemes of each data burst. The remaining parts of the downlink subframe are DL data bursts. The uplink (UL) subframe contains the following: The ranging subchannel specified in the UL_MAP message is used for initial ranging, periodic ranging, and contention-based bandwidth request. The initial ranging transmission shall be used by the MS to synchronize with the system during the first setup. Periodic ranging will be executed periodically to update the system time, frequency, and transmission power. The bandwidth request is used by the MS to request the uplink allocations. The remaining parts of the uplink subframe are UL data bursts.

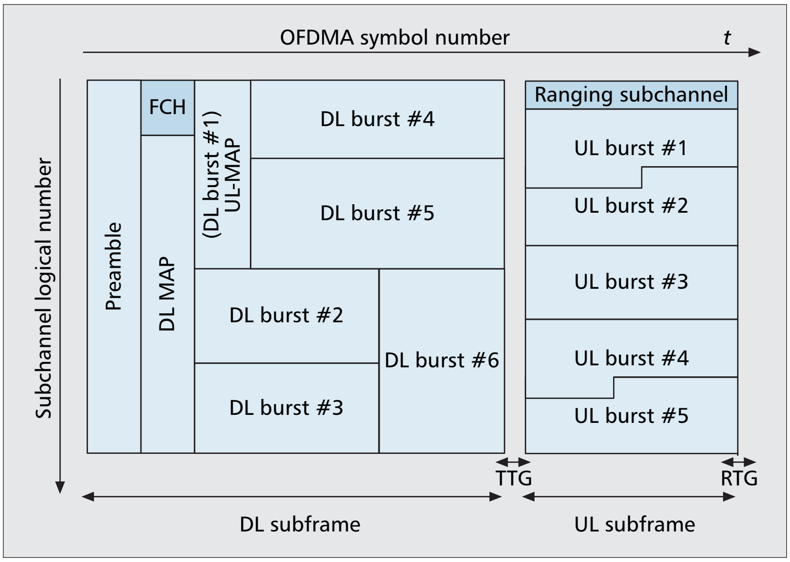


Figure 1. *Example of an OFDMA frame in TDD mode.*

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

MAC Layer

The MAC layer of WiMAX mainly supports a point-to-multipoint (PMP) architecture and a mesh architecture (optional). The MAC layer is designed for handling applications with different quality of service (QoS) requirements. In mobile WiMAX, all services are connection-oriented; as shown in Fig. 2, each service is mapped to one connection or multiple connections, and is handled by the convergence sublayer (CS) and then the common part sublayer[1] (CPS). Because the MAC layer of WiMAX must support various backhaul networks such as asynchronous transfer mode (ATM) and IP-based networks, the CS needs to be able to handle a mapping from different types of transport-layer traffic to a MAC formatted connection (or multiple connections). As mentioned before, the MAC is connection-oriented; each service, including the connectionless service, is mapped to at least one connection. Each connection is identified by a 16-bit connection identifier (CID). This sublayer classifies the service data units (SDUs) to a proper connection with specific QoS parameters. After the CS, the CPS controls most MAC functionalities (fragmentation, packing, scheduling, retransmission, etc.). Besides, the IEEE 802.16 MAC implements a request-and-grant mechanism for allocating resources. This provides more centralized QoS control of all applications.

Service Class and Service Data Flow

Service Classes

The service classes for mobile WiMAX consist of five different types: unsolicited grant service (UGS), real-time polling service (rtPS), extended rtPS (ertPS), non-real-time polling service (nrtPS), and best effort service (BE).

Unsolicited Grant Service — The UGS is designed to support real-time service flows that will generate fixed-size data packets periodically. The UGS will be granted periodically without a polling-request procedure; hence, it can reduce the latency. With UGS, the grant management subheader will include the poll-me bit, and no random access opportunity is allowed for bandwidth requests.

[1] *The security sublayer is the third sublayer in the MAC layer. However, since it is has no impact on transmission performance, discussion of its functionalities is excluded.*

The nrtPS is designed to support delay-tolerant data streams which consist of variable-sized data packets. In general, these services can tolerate longer delays and are relatively insensitive to the delay jitter. The nrtPS is suitable for Internet access with a minimum guaranteed rate.

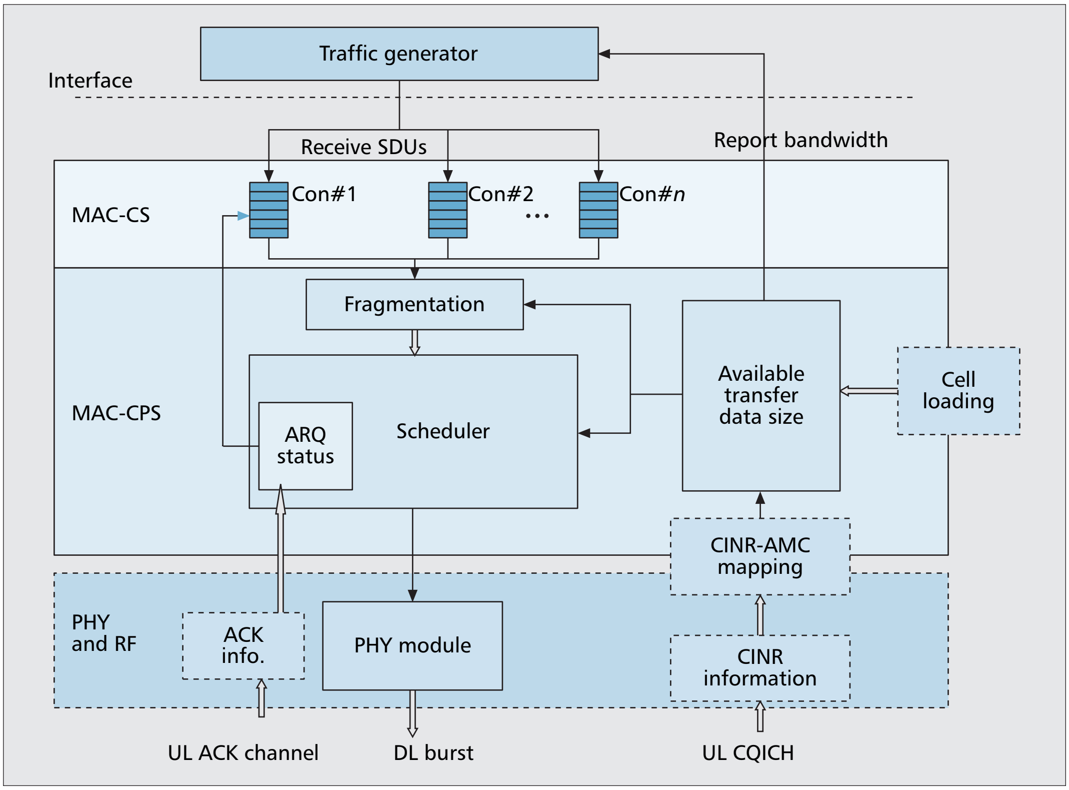


Figure 2. *IEEE 802.16e protocol layer.*

Real-Time Polling Service — The rtPS is designed to support real-time service flows, which will generate variable-size data packets on a periodic basis. This service requires more request overhead and latency than UGS, but can support variable grant sizes. The rtPS is well suited for connections carrying services such as voice over IP (VoIP) or video streaming services.

Extended Real-Time Polling Service — The ertPS is designed to support real-time service flows that generate variable-size data packets on a periodic basis, such as VoIP services. Extended rtPS is to utilize the efficiency of both UGS and rtPS. In ertPS, the BS provides unicast grants in an unsolicited manner as in UGS to save the latency caused by bandwidth request.

Non-Real-Time Polling Service (nrtPS) — The nrtPS is designed to support delay-tolerant data streams that consist of variable-sized data packets. In general, these services can tolerate longer delays and are relatively insensitive to delay jitter. The nrtPS is suitable for Internet access with a minimum guaranteed rate, such as FTP and HTTP.

Best Effort Service — The BE service is designed to support data streams that have no minimum service requirement and therefore may be handled on a resource-available basis, such as email. In BE neither throughput nor delay guarantees are provided.

The service classes are distinguished by the service-specific convergence sublayer (CS). When the packets are classified in the CS, the connection is chosen based on the type of QoS requirements.

Service Data Flow

In Fig. 3 we show how different services are handled within an IEEE 802.16e system. The MAC layer defines QoS signaling mechanisms and functions that can control BS and MS data transmissions. The CS sublayer classifies the different QoS services into connections and assigns the connection a unique connection indicator (CID) on both downlink and uplink. The data of the connections are forwarded to appropriate queues. On the downlink, transmission is relatively simple because the BS is the only one that transmits during the downlink subframe in which the BS schedules all downlink connections. The assignments are broadcast to all MSs in the DL_MAP. The associated MS will then know exactly when to receive its own packets. The BS also determines the number of slots that each MS will be allowed to transmit in an uplink subframe. This information is broadcast by the BS through the UL_MAP at the beginning of the first DL burst. The UL_MAP contains an information element (IE) that includes the transmission opportunities (i.e., the slots in which the MS can transmit during the uplink subframe). After receiving the UL_MAP, each MS will transmit data in the predefined slots as indicated in the IE.

Radio Resource Management

The purpose of radio resource management is to improve the efficiency and reliability of wireless transmission. In this study the following radio resource management is considered:

Rate Control: The adaptive modulation and coding scheme (AMC) is a major method to

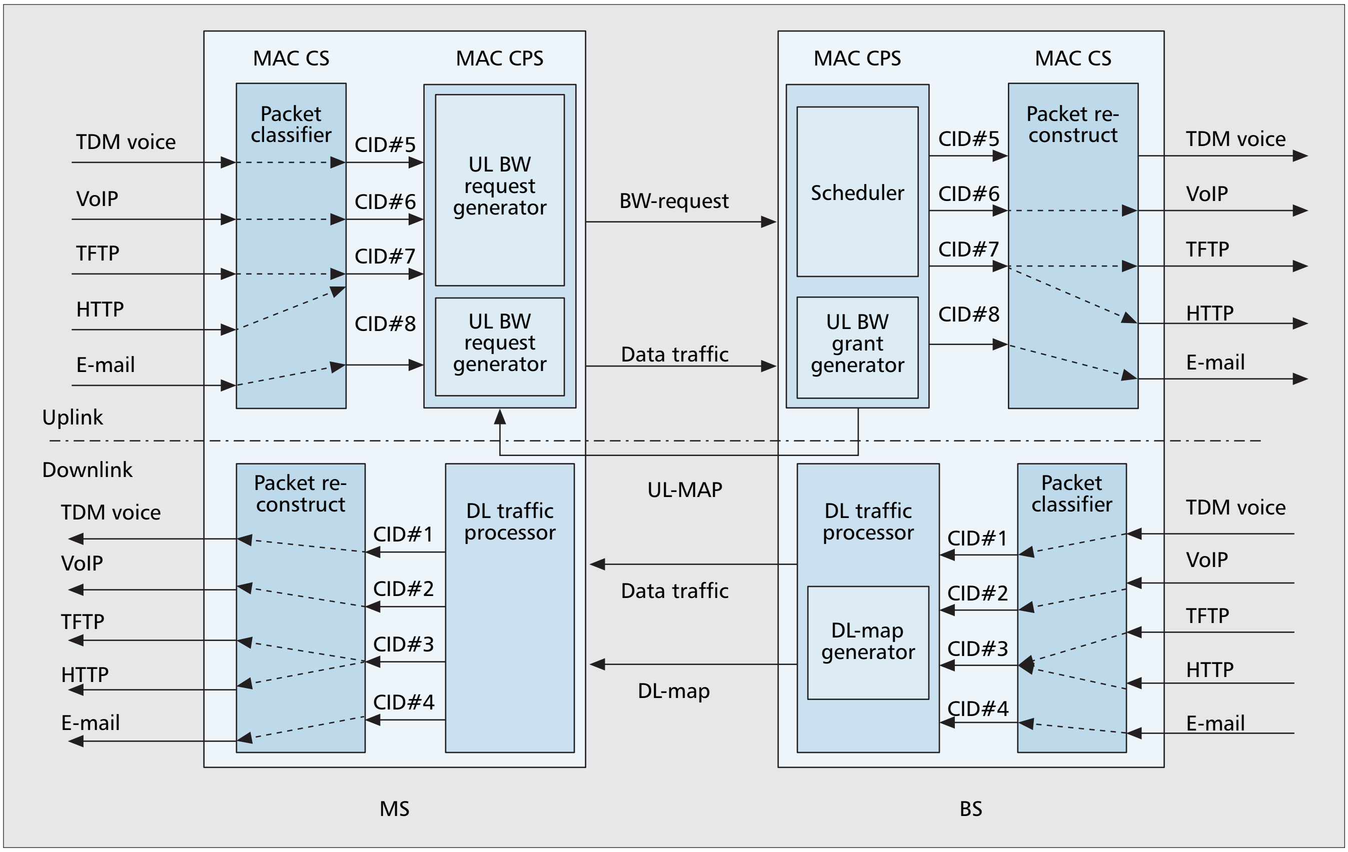


Figure 3. *Example of 802.16 service flow.*

maintain the quality of wireless transmission. IEEE 802.16e supports a variety of modulation and coding schemes. In our study, with 1/2 convolution code rate, we have considered quaternary phase shift keying (QPSK), 16-quadrature amplitude modulation (QAM), and 64-QAM schemes. As a result, based on different radio frequency (RF) conditions, each transmission slot could carry 48, 96, or 144 bits, respectively.

Power Control: Even IEEE 802.16e allows dynamic power control; on the downlink, with rate control, power control is not very critical to maintain transmission quality. Here, fixed power is assumed for all users in different RF conditions. In that case, rate control is applied to benefit from changing RF conditions.

Channel Assignment: The OFDMA frame structure has two dimensions where each transmission slot is consisted of subchannel numbers and OFDMA symbols. Basically, packets are segmented and fitted into one OFDMA slot. To construct the transmission slot, subchannel numbers (the frequency) will be decided prior to the number of OFDMA symbols.

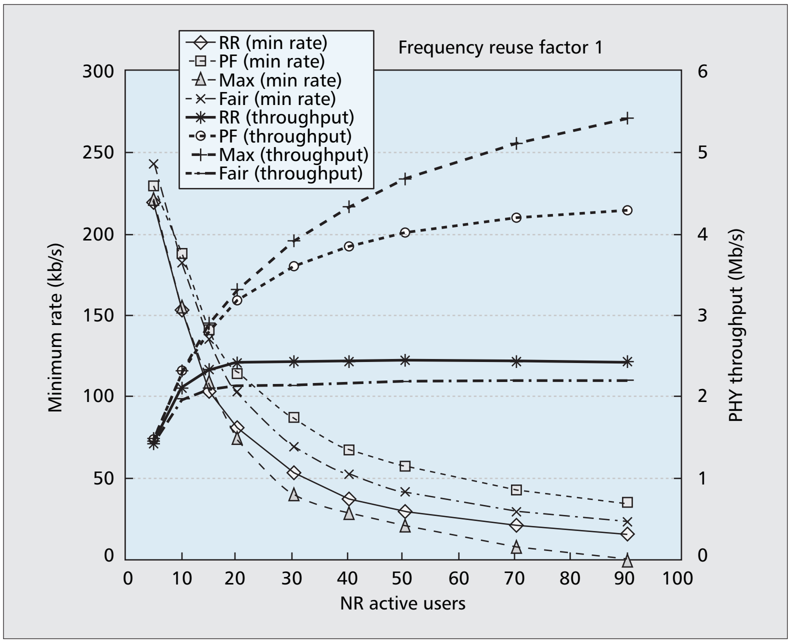
Subcarrier Permutation: In mobile WiMAX, there are two types of permutation:

- Distributed subcarrier permutation: This method is designed for averaging intercell interference by assigning subcarriers pseudo-randomly across the entire transmission spectrum. In this case users all have cross-spectrum subcarrier assignment. In-band interference and frequency selective fading affect all users evenly (approximation). This permutation method is adopted by Partial Usage of Subchannels (PUSC) and Full Usage of Subchannels (FUSC).
- Adjacent subcarrier permutation: The adjacent subcarrier permutation method is designed to make use of the frequency selection gain where the quality of different subcarriers is not the same. To define adjacent subcarrier permutation, a bin, which consists of a set of nine contiguous subcarriers within an OFDMA symbol, is a basic transmission unit in both the downlink and uplink. A typical implementation example is the AMC band designed for an adaptive antenna system (AAS) with the beamforming technique.

In this article, with sectorized antennas the PUSC distributed subcarrier permutation method is chosen.

Scheduling method: Scheduling control is an important radio resource management. For comparison, round-robin (RR), proportional fair (PF), max CINR (MC), fair throughput (FT), and early deadline first (EDF) [4–7] are considered in this study. The detail of each scheduler algorithm is discussed as follows:

- **Round robin**: With respect to RR scheduling, users are cyclically scheduled irrespective of the channel condition.
- **Proportional fair**: A PF scheduler allocates user m^* with the maximum ratio of achievable instantaneous data rate over the average received data rate. The user in the OFDMA system is scheduled at frame n using the following function:



■ **Figure 4.** *Minimum rate and system throughput.*

$$m^* = \arg_m \max\left\{\frac{\sum_{s=1}^{S} IR_m^s(n)}{R_m(n)}\right\}, \quad (1)$$

where $IR_m^s(n)$ denotes the achievable instantaneous data rate for user m at time n on subcarrier s. S is the total subscribers assigned to the user. $R_m(n)$ denotes the moving average of datarate at user m who has received up to time n according to the following equation:

$$R_m(n) = \left(1 - \frac{1}{N_T}\right) R_m(n-1) + \frac{1}{N_T} IR_m^{assigned}(n-1), \quad (2)$$

where N_T denotes the length of moving average, which has been set to 750.

- **Max CINR**: The MC scheduler allocates the user m^* with the maximum received CINR. The user in the OFDMA system is scheduled at frame n using the following function:

$$m^* = \arg_m \max\left\{\sum_{s=1}^{S} CINR_m^s(n)\right\}, \quad (3)$$

where $CINR_m^s(n)$ denotes the CINR for user m at time n on subcarrier s.

Fair throughput: The FT scheduler allocates the user m^* with the minimum average received data rate. The user is scheduled at frame n using the following function:

$$m^* = \arg_m \min\{R_m(n)\}, \quad (4)$$

where $R_m(n)$ denotes the moving average of the data rate at user m that has received up to time n and is equal to Eq. 2.

- **Early deadline first**: The EDF scheduler allocates the user m^* with the minimum remaining time that needs to be transmitted. EDF provides priority treatment for real-time services. The user is scheduled at frame n using the following function:

$$m^* = \arg_m \min(DB - Age - T_t\}, \quad (5)$$

where DB is the delay bound, Age is the time that the user's packet has stayed in the MAC layer, and T_t is the required time to finish transmission of the packet.

Mobile WiMAX Performance

In this section we examine mobile WiMAX performance by considering non-real-time services, mixed VoIP, and video streaming services. With different applications, we examine the effects of implementing different radio resource management. All performances are simulated based on a three-sector 19-cell mobile system considering slow fading channels and PUSC distributed subcarrier permutation. The total transmission bandwidth is 6 MHz with a frequency reuse factor of one. The detail simulation parameters are listed in [8].

Non-Real-Time Services: HTTP

Considering only non-real-time services, Fig. 4 shows the throughput and minimum transmission rate each scheduler algorithm can achieve.

As shown in Fig. 4, MC, which can achieve the highest system aggregate throughput, has the lowest number of users that can meet the minimum transmission rate. This is because even MC always chooses the best CINR for transmission but has the worse fairness control among users. Thus, any poor CINR users will suffer from low transmission rates. On the other hand, in the mobility environment, the proportional fair (PF) algorithm, considering both the RF opportunity and fairness in average transmission rate simultaneously, can effectively improve the system aggregate throughput and at the same time also maintain the highest number of users with the minimum transmission rate. Even though the RR and FT scheduling control algorithms try to provide fairness in the transmission time and transmission rate, respectively, neither algorithm takes the opportunity to transmit in good RF conditions, which degrades overall performance including the number of users with the minimum transmission rate. The above results can be used as a reference design for call admission control if the minimum rate is the QoS for non-real-time services. For example, if the target minimum rate is set at 50 kb/s, the call admission control might need to be applied when the number of active users exceeds 26, 32, 40, and 60 for MC, RR, FT, and PF respectively.

Mixed VoIP and Non-Real-Time Services

Instead of considering only non-real-time services, mixed VoIP and non-real-time services are investigated. With 1 percent packet loss rate for VoIP and a minimum transmission rate of 50 kb/s for non-real-time services, Fig. 5 shows the trade-off in capacity between the VoIP and non-real-time services. As shown, for RR, without priority treatment (no EDF) for VoIP users, VoIP capacity will be degraded significantly when non-real-time traffic is introduced. But with EDF control, VoIP capacity will be held initially until the non-real-time users reach a certain number. On the other hand, with PF alone the VoIP capacity can hold and will be degraded only linearly with the increase of non-real-time users. With EDF, all VoIP capacity in different scheduler controls is improved. In other words, the priority treatment on VoIP can smooth the degradation caused by the increase of non-real-time traffic.

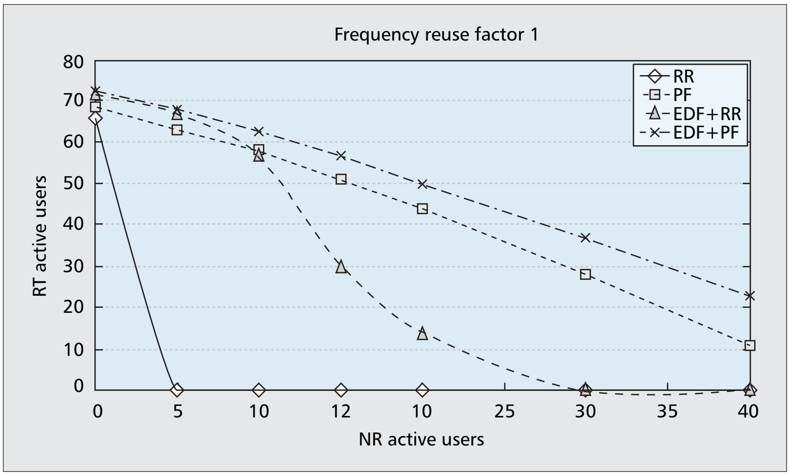
Video Streaming Services

Finally, we examine the transmission quality of video streaming services by exploring the option of the multiconnection feature. In mobile WiMAX, there is an option of establishing multiple connections for a single application. In other words, it is possible to transmit/receive different priority packets on different connection IDs. To examine the transmission performance of the video streaming service, we consider H.264/AVC-based scalable video coding (SVC) developed to provide high-quality streaming under various transmission bandwidths.

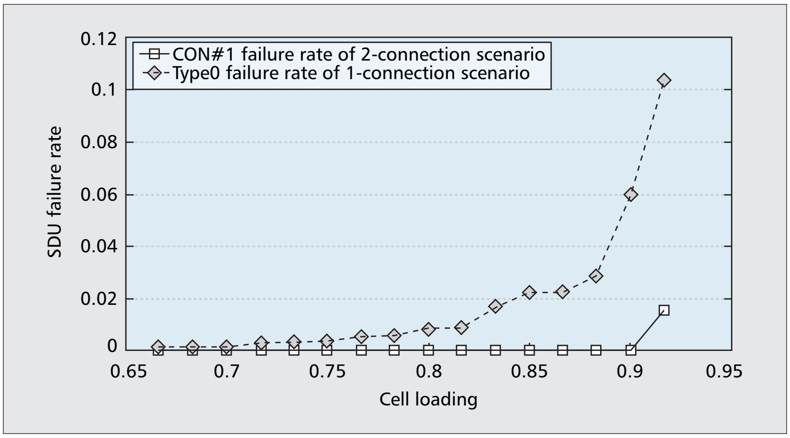
The scalable extension of H.264/AVC is the latest SVC standard. It is developed by the Joint Video Team (JVT) formed by the International Standards Organization/International Electrotechnical Commission (ISO/IEC) MPEG and ITU-T, and is aimed to simultaneously provide three-dimensional scalabilities with good compression efficiency. To support spatial scalability, the video is decomposed into several layers. Each layer can be encoded separately or get prediction from the lower spatial layers to remove redundancy. In each spatial layer, the data can be separated into several signal-to-noise ratio (SNR) layers by two means. Coarse-grained scalability (CGS) means the bitstream can only be truncated at several predefined points, while fine-grained scalability (FGS) means the bitstream can be truncated at any position. Note that in the standard, the first SNR layer in a spatial layer is restricted to CGS. To support temporal scalability, a hierarchical prediction structure is used. The pictures in a group of pictures (GOP) are dyadic decomposed into several layers, including base layers and enhanced layers. Basically, compared to enhanced-layer packets, base-layer packets are critical and need to be delivered successfully. For more details of SVC, please refer to [9–12].

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. 2. The more important data is allocated to the connection that has more protection (i.e., higher transmission priority and MAC retransmission). To address the bandwidth fluctuation effect, in the proposed two-connection implementation the server allocates important data to the first connection and the remaining data to the second connection. Thus, the BS needs to retransmit only the more important data when the real bandwidth is smaller than the expected bandwidth.

Figure 6 shows the SDU failure rate of important video packets in the two-connection and one-connection scenarios. By considering the failure rate of important video packets only, from Fig. 6 it is obvious that the two-connection scenario has better control of the failure rate of important video packets even when cell loading increases. The type0 failure rate of one connection is the loss of important packets from the one-connection-only delivery.



■ **Figure 5.** *The trade-off in capacity between VoIP and non-real-time services.*



■ **Figure 6.** *The SDU failure rate of important video packets in two-connection and one-connection scenarios.*

Conclusions and Recommendations

In this article, to support mixed VoIP and non-real-time services in mobile WiMAX, we have shown that a proper choice of good scheduling control is critical. Of all options, PF can be considered a good scheduling control algorithm for non-real-time services. Priority treatment for VoIP services is important and can minimize the degradation caused by the introduction of non-real-time services. For H.264/AVC-based scalable video streaming, it is critical to assign multiple connections to a single streaming application by separating the layered video packets into two different connections with different treatments in protection and retransmission. With the implementation of multiconnection, transmission quality can be maintained even with increasedd cell loading.

To support mixed VoIP and non-real time services in Mobile WiMAX, we have shown that a proper choice of good scheduling control is critical. Among all, PF can be considered as a good scheduling control algorithm for non-real time services.

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BIOGRAPHIES

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