

Multicast Power Control for VoIP in Mobile WiMAX

Jae-Heung Yeom and Yong-Hwan Lee
School of Electrical Engineering and INMC
Seoul National University
Kwanak P. O. Box 34, Seoul 151-600 Korea
E-mail: jhyeom@ttl.snu.ac.kr, ylee@snu.ac.kr

Abstract - In this paper, we consider the service of voice over internet protocol (VoIP) traffic with the service of non-real time traffic in the mobile WiMAX system. VoIP traffic is delay-sensitive and requires transmission at an almost constant rate. As a consequence, it may require a large amount of resource when users are in poor channel condition, leading to resource outage problem. To alleviate this problem, we consider the allocation of power according to the VoIP traffic load as well as the channel condition. When the VoIP traffic is fully loaded, the power is allocated to maximize the VoIP capacity. Otherwise, it is allocated to maximize the data throughput. Finally, the performance of the proposed scheme is verified by computer simulation.

Keywords-IEEE 802.16e; multicast; OFDMA; power control; VoIP; WiMAX

I. INTRODUCTION

IEEE 802.16e standard specifies a new wireless access system that can provide a state-of-the-art solution for the last-mile technology [1]. Mobile worldwide interoperability for microwave access (m-WiMAX) is a commercialized IEEE 802.16e system based on orthogonal frequency division multiple access (OFDMA) with time-division duplex. It can support bursty data traffic with high peak rates, while simultaneously supporting streaming video and latency-sensitive voice traffic over the same channel [2]. It employs adaptive modulation and coding (AMC) and hybrid automatic repeat request (HARQ) for the enhancement of coverage and capacity [2].

Voice service is possible in a packet switched wireless system by means of voice over internet protocol (VoIP). VoIP traffic is delay sensitive and requires transmission at an almost constant bit rate during the active period. As a consequence, it may require a large amount of resource for users in low signal-to-interference-plus-noise ratio (SINR) condition, deteriorating the overall system performance due to excessive use of resource [3]. On the other hand, users in high SINR condition require a small amount of resource for VoIP service. This problem can be alleviated by controlling the transmit power for each packet in OFDMA packet switched systems [4]–[6]. However, optimization of power and resource allocation may require high implementation and computation complexity [6].

The m-WiMAX can control the transmit power in a group-

wise manner to reduce the signaling overhead [1]. However, it has not supported the power control for VoIP packets, but it has supported only AMC [7], [8]. Moreover, it has not considered the link adaptation for concurrent transmission of VoIP and non-real time traffic (NRT) such as web browsing. In this paper, we consider a link adaptation scheme for the concurrent transmission of VoIP and NRT traffic with the use of AMC and multicast power control. To enhance the system capacity for the service of mixed traffic, we consider the power control according to the VoIP traffic load. When the VoIP traffic is fully loaded, the proposed scheme allocates the power to maximize the VoIP capacity. Otherwise, it allocates the power to maximize the data throughput. Furthermore, it differently allocates the power to VoIP users according to the channel condition.

This paper is organized as follows. Section II briefly describes the IEEE 802.16e system. Section III presents the proposed multicast power control. Section IV verifies the performance of the proposed scheme by computer simulation. Finally, Section V concludes the paper.

II. POWER CONTROL IN IEEE 802.16E

The m-WiMAX has a slot comprising 48 data subcarriers as the minimum unit for resource allocation. We consider partial usage of subcarriers (PUSC) permutation that provides frequency diversity gain [2]. In the PUSC mode, a single slot consists of one subchannel (corresponding to 24 data subcarriers) and two symbols. HARQ is supported by HARQ DL MAP IE that defines the allocation of resource in a two-level process. This allocation first defines a rectangular region comprising two-dimensional (2-D) frequency-time domain, where the power can be boosted at a step of 3dB in a range of -12dB to +9dB. Each region is partitioned into a number of bursts in the frequency domain [1].

We assume that the total resource is divided into K 2-D regions in a rectangular shape, as shown in Fig. 1, and that the power of the k -th region is boosted by a factor of β_k , where

$$\sum_{k=1}^K \beta_k P_0 f_k \leq P_t. \quad (1)$$

Here f_k denotes the number of subchannels in the k -th

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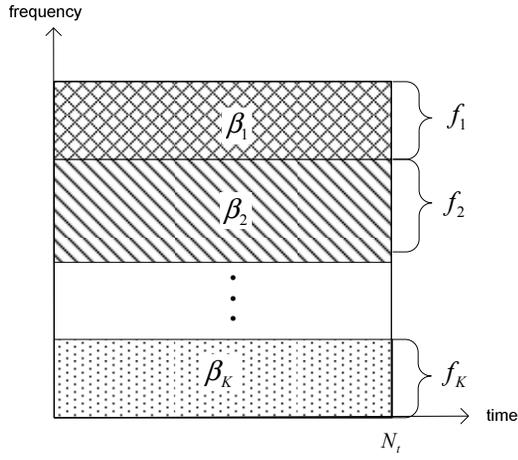


Figure 1. Power boosting pattern for the downlink subframe.

region, P_0 denotes the nominal power per subchannel, and P_t denotes the maximum allowable transmit power. For ease of description, assume that $\beta_1 \geq \beta_2 \geq \dots \geq \beta_K$. Note that $\beta_k = 1$ for all k , when only AMC is employed. The amount of frequency-time resource for the k -th region in the PUSC mode can be represented as

$$R_k = f_k \cdot \frac{N_t}{2} \quad (\text{slots}) \quad (2)$$

where N_t denotes the number of OFDM symbols and the denominator '2' indicates the symbol duration of a slot.

III. PROPOSED MULTICAST POWER CONTROL

We consider the allocation of transmit power and packets according to the channel condition and VoIP traffic load. Assume that each of K resource regions of which each is boosted by a specific power gain. Then, there can be N_B possible boosting patterns determined as

$$N_B = \# \left\{ b \mid \sum_{k=1}^K \beta_{b,k} P_0 f_{b,k} \leq P_t \right\} \quad (3)$$

where $\#A$ denotes the number of elements in set A , and $\beta_{b,k}$ and $f_{b,k}$ denote the power gain and the number of subchannels for the k -th region of boosting pattern b , respectively. Let $\Upsilon[b, k]$ denote the region boosted by the k -th gain of boosting pattern b .

Letting γ_n be the SINR of VoIP packet n , the effective SINR in region $\Upsilon[b, k]$ can be represented as

$$\bar{\gamma}_{b,k,n} = \beta_{b,k} \gamma_n \quad (4)$$

For a given effective SINR $\bar{\gamma}_{b,k,n}$, assume that a VoIP packet of L_v bits can be transmitted using the number slots given by

$$r(\bar{\gamma}_{b,k,n}) = \left\lceil \frac{L_v}{J \cdot C(\bar{\gamma}_{b,k,n})} \right\rceil \quad (5)$$

where J denotes the number of subcarriers for traffic transmission in each slot (e.g., $J=48$ in m-WiMAX), $C(\gamma)$ denotes the spectral efficiency at γ , and $\lceil x \rceil$ denotes the smallest integer larger than or equal to x .

When N VoIP packets with effective SINR $\{\gamma_n\}$ are in the queue, they can be sorted in an ascending order of the SINR as

$$(j_1, j_2, \dots, j_N) = \Xi(\gamma_1, \gamma_2, \dots, \gamma_N) \quad (6)$$

where $\Xi(\cdot)$ outputs the packet index according to the SINR in an ascending order (i.e., j_m denotes packet index with the m -th lowest SINR). We consider the transmission of VoIP packets with different power gain according to the channel condition (e.g., VoIP packets in low SINR condition can be sent with high power gain and vice versa). The packets can be allocated in two ways. One way is to allocate the packets in a descending order of the SINR (i.e., transmit packet j_N , then j_{N-1} , and so on), and the other way is to allocate the packets in an ascending order of the SINR (e.g., transmit packet j_1 , then j_2 , and so on). We will call the former allocation 'up-allocation' (denoted by packet allocation mode $d=0$) since packets are filled in order of from the region with the lowest power gain to the region with the highest power gain. The latter allocation will be called 'down-allocation' (denoted by $d=1$) since packets are filled in order of from the region with the highest power gain to the region with the lowest power gain.

Let $U_{b,k,d}$ be a set of VoIP packets allocated in region $\Upsilon[b, k]$ with packet allocation mode d , where packets are transmitted with power gain $\beta_{b,k}$. Initial amount of resource for region $\Upsilon[b, k]$ is given by

$$R_{b,k,d} = f_{b,k} \cdot \frac{N_t}{2} \quad (\text{slots}). \quad (7)$$

After packet j_m is allocated in region $\Upsilon[b, k]$ with packet allocation mode d , the amount of resource left in region $\Upsilon[b, k]$ is

$$R_{b,k,d} := R_{b,k,d} - r(\beta_{b,k} \gamma_{j_m}) \quad (8)$$

where $:=$ denotes update. Provided that $R_{b,k,d} \geq 0$, $U_{b,k,d}$ is updated as

$$U_{b,k,d} := U_{b,k,d} \cup \{j_m\}. \quad (9)$$

When chase-combining HARQ (CC-HARQ) is employed, the effective SINR of retransmit packet n can be represented as

$$\tilde{\gamma}_{b,k,n} := \gamma_n(0)\beta_{b,k} + \sum_{t=-T_n+1}^{-1} \gamma_n(t)\tilde{\beta}(t) \quad (10)$$

where T_n denotes the number of transmissions, t denotes the transmission index, and $\tilde{\beta}(t)$ denotes the power gain at the t -th transmission. Here, negative transmission index t means previous transmission. The power gain of current transmission can be determined from (10) as

$$k = \max \left\{ i \mid \beta_{b,i} \geq \frac{th_{M(0)} - \sum_{t=-T_n+1}^{-1} \gamma_n(t)\tilde{\beta}(t)}{\gamma_n}, 1 \leq i \leq K \right\} \quad (11)$$

where $\max A$ denotes a maximum value in set A , $th_{M(0)}$ denotes the SINR threshold for modulation and coding scheme (MCS) level $M(0)$ for the current transmission. By increasing $M(0)$ by means of power boosting, it is possible to reduce the repetition number. The retransmit packet n is allocated to region $\Upsilon[b,k]$ as

$$U_{b,k,d} := U_{b,k,d} \cup \{n\} \quad (12)$$

There can be $N_{b,d}$ VoIP packets with the use of boosting pattern b and packet allocation mode d , given by

$$N_{b,d} = \# \left\{ n \mid R_t - \left\{ \sum_{k=1}^K \sum_{n \in U_{b,k,d}} r(\tilde{\gamma}_{b,k,n}) \right\} \geq 0 \right\}, \quad 1 \leq b \leq N_B, d = 0, 1 \quad (13)$$

where R_t denotes the number of total slots for payload transmission. Assuming that N_{VoIP} VoIP packets are in the queue, they can be supported by a set of boosting pattern b and packet allocation mode d , given by

$$I(b,d) = \{(b,d) \mid N_{b,d} \geq N_{VoIP}, 1 \leq b \leq N_B, d = 0, 1\}. \quad (14)$$

When $I(b,d)$ is an empty set, this indicates that these VoIP packets cannot properly be transmitted by any boosting pattern, i.e., fully loaded. In this case, it may be desirable to maximize the VoIP capacity that is defined as the number of users in a sector experiencing a voice outage less than a desired outage percent. A voice outage occurs when the user experiences a packet loss rate (PLR) larger than a desired PLR, say p_e [9]. Note that the up-allocation mode can support more VoIP packets than the down-allocation mode since it first transmits packets in high SINR condition. However, it may delay the transmission of packets in low SINR condition, which may cause voice outage due to excessive delay. When the VoIP

traffic is fully loaded, it may be desirable to provide transmission opportunity for packets in low SINR condition as well as to maximally accommodate VoIP packets. To this end, we first allocate packets with PLR larger than p_e with a power gain larger than or equal to '1', reducing the amount of resource. Therefore, the remained resource can be represented as

$$R_{b,k,d} := R_{b,k,d} - r(\beta_{b,k}\gamma_n) \quad (15)$$

where $k \in \{i \mid \beta_{b,i} \geq 1, 1 \leq i \leq K\}$ and $n \in \{j_m \mid p_{e,j_m} > p_e, 1 \leq m \leq N\}$. Here, p_{e,j_m} denotes the PLR of user j_m . When $R_{b,k,d} \geq 0$,

$$U_{b,k,d} := U_{b,k,d} \cup \{j_m\}. \quad (16)$$

When the VoIP traffic is fully loaded, its capacity can be maximized by determining as

$$(\hat{b}, \hat{d}) = \arg \max_{1 \leq b \leq N_B, d} \# \left\{ n \mid R_t - \sum_{1 \leq k \leq K} \sum_{n \in U_{b,k,d}} r(\tilde{\gamma}_{b,k,n}) \geq 0 \right\}. \quad (17)$$

When $I(b,d)$ is not empty, this indicates that the system can properly transmit VoIP packets. In this case, it is desirable to maximize the data throughput using the rest of resource. The amount of resource left in region $\Upsilon[b,k]$ can be represented as

$$R_{b,k,d} := f_{b,k} \frac{N_t}{2} - \sum_{n \in U_{b,k,d}} r(\tilde{\gamma}_{b,k,n}), \quad (b,d) \in I, 1 \leq k \leq K \quad (18)$$

and the index set $B_{b,d}$ of power gain for a resource left in $\Upsilon[b,k]$ is obtained by

$$B_{b,d} = \{k \mid R_{b,k,d} \geq 0, (b,d) \in I, 1 \leq k \leq K\}. \quad (19)$$

The optimum parameters that maximizes the data throughput can be obtained by

$$(\hat{b}, \hat{d}) = \arg \max_{(b,d) \in I} \sum_{k \in B_{b,d}} R_{b,k,d} C(\beta_{b,k}\gamma_d) \quad (20)$$

where γ_d denotes the SINR of the NRT user. It can be seen that the optimum parameters depend on the SINR of the NRT user. For example, it may be desirable for the NRT user in low SINR to obtain high power gain rather than large amount of resource. In this case, the proposed scheme may choose the up-allocation mode to allow high power gain to the NRT. On the contrary, it may be desirable for NRT user in high SINR choose the down-allocation mode to allocate the large amount of resource.

IV. PERFORMANCE EVALUATION

The performance of the proposed scheme is verified by computer simulation in a regularly placed 19-cell environment (with three sectors per cell). It is assumed that users are uniformly distributed in the cell while being at least 35 meters away from the sector, and the users receive the strongest power from the serving sector. It is also assumed that the minimum geometry for a call admission is set to -4 dB and that a packet loss occurs when a packet is not successfully delivered within an access delay of 80ms. Channel decoding error is calculated by an effective SINR mapping method in [10]. It is also assumed that G.729 voice codec generates a voice frame of 20 bytes at every 20msec at a bit rate of 8kbps and a conventional RTP (12 bytes)/UDP (8 bytes)/IP (20 bytes) header size (40 bytes) is compressed to a 2-byte data [11]. The frequency length of each boosting region has a minimum unit of the major group (i.e., 6 or 4 subchannels) and K is set to 3, which corresponds to the maximum number of SUB-MAP messages [1]. We consider the ITU multi-path power delay profile environments comprising 40 % Pedestrian-A, 30 % Pedestrian-B at a user speed of 3 km/h and 30% Vehicular-A at a user speed of 10 km/h. Table I summarizes the simulation parameters, and the others follow a common m-WiMAX system profile [12]. In figures, ‘AMC(-3 dB)’ or ‘AMC($+15$ dB)’ indicates that VoIP users are serviced by means of only the AMC, and the left resource and power are allocated to a single NRT user with a geometry of -3 dB or 15 dB. ‘Proposed(-3 dB)’ or ‘Proposed($+15$ dB)’ indicates that VoIP is serviced by combined use of the proposed multicast power control and AMC, and the left resource and power are allocated to a single NRT user with a geometry of -3 dB or 15 dB.

Fig. 2 depicts the average amount of resource and the average power gain left after accommodating VoIP users according to the number of active VoIP. Since the AMC provides the same power for VoIP users, the amount of resource and the power left after accommodating VoIP are the same regardless of the SINR of the NRT (-3 dB or $+15$ dB). However, the proposed scheme determines the boosting pattern and packet allocation mode by considering the SINR of the NRT when the VoIP traffic is not fully loaded. Note that the data throughput depends on the power gain in low SINR environments, whereas it depends on the degree of freedom (i.e., the amount of resource) in high SINR environments [13]. In this manner, it can be seen that the proposed scheme transmits the NRT at geometry of -3 dB using the small amount of resource with high power gain while the NRT user at geometry of 15 dB using a large amount of resource with low power gain. That is, it determines the boosting pattern and packet allocation mode adaptively to maximize the data throughput when the VoIP traffic is not fully loaded.

Fig. 3 depicts the average data throughput according to the number of active VoIP users. It can be seen that the proposed scheme improves the data throughput over the AMC as the

TABLE I
SIMULATION PARAMETERS

| PARAMETERS | Values |
|-----------------------------|--|
| Carrier frequency | 2.3 GHz |
| Channel bandwidth | 8.75 MHz |
| Downlink symbols | 27 |
| Overhead resource | 150 slots |
| MCS | QPSK- $\{1/12, 1/8, 1/4, 1/2, 3/4\}$, 16QAM- $1/2$, 64QAM- $\{1/2, 2/3\}$ |
| HARQ | CC with max retransmission of 3 |
| Path loss model (d: meters) | $28.6+35*\log_{10}(d)$ dB |
| Antenna configuration | (1×2) |
| Sector antenna pattern | $70^\circ(-3\text{dB})$ with 20dB front-to-back ratio |
| Receiver algorithm | Maximal ratio combining |

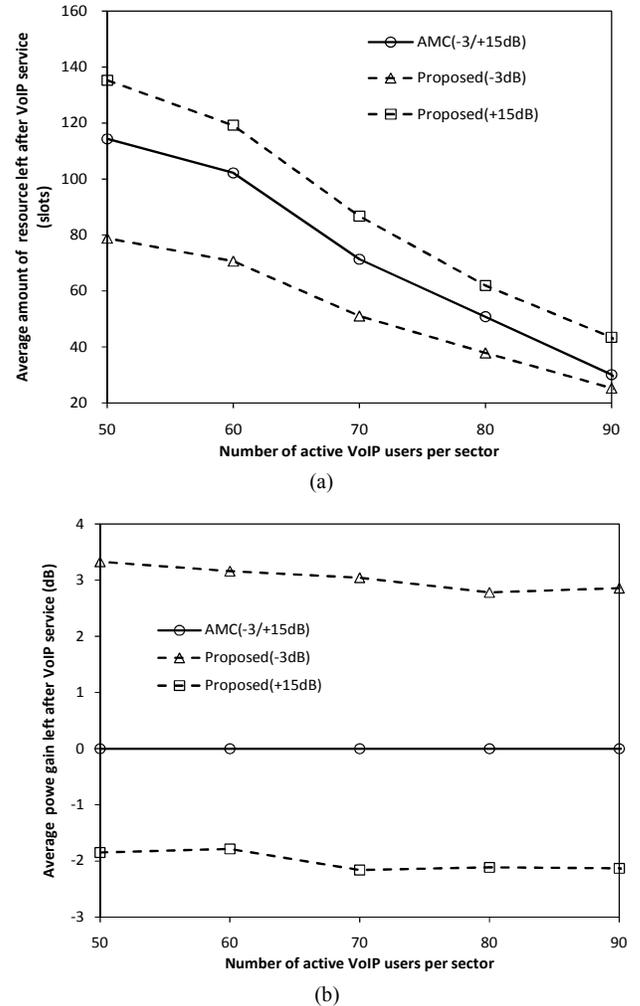


Figure 2. Resource and power allocated to a NRT user according to the number of active VoIP users; (a) Average amount of resource (slots), (b) Average power gain (dB).

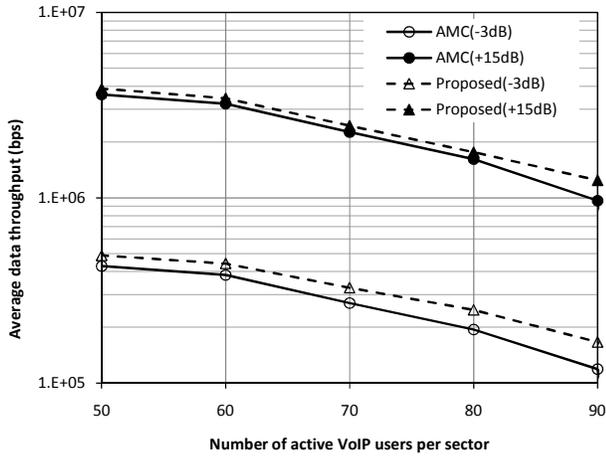


Figure 3. Average data throughput according to number of active VoIP users

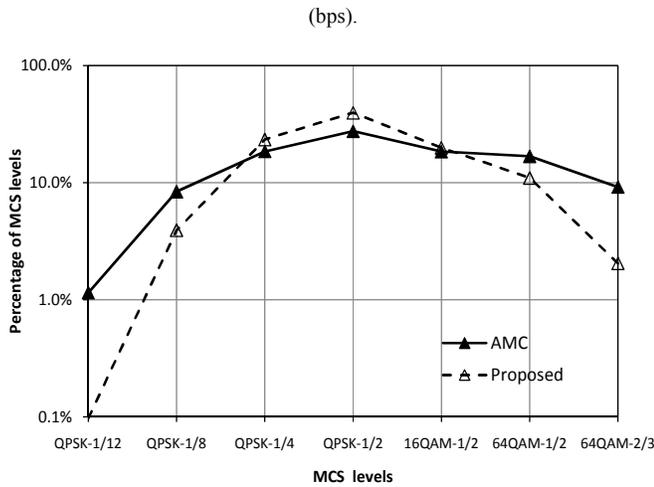


Figure 4. Percentage of MCS levels allocated to VoIP users.

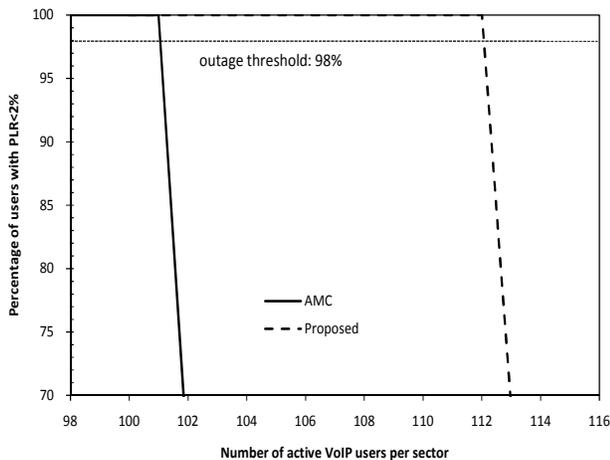


Figure 5. Percentage of users with PLR < 2% according to the number of active VoIP users per sector.

number of VoIP users is large by utilizing the optimized amount of resource and power gain. The proposed scheme controls the power for VoIP users considering the NRT, making it effective when the VoIP traffic is heavy.

Fig. 4 depicts the percentage of the MCS levels for VoIP packets when the VoIP traffic is fully loaded. It can be seen that the proposed scheme reduces the use of low and high MCS levels compared to the AMC, while increasing the use of moderate MCS levels.

Fig. 5 depicts the VoIP capacity, which is the number of users in a sector experiencing a voice outage less than 2%. It can be seen that the proposed scheme yields a VoIP outage capacity gain of about 13% over the AMC. This is mainly because when the VoIP traffic is fully loaded, the proposed scheme determines the boosting pattern and packet allocation mode to maximize the VoIP capacity.

V. CONCLUSIONS

In this paper, we have considered combined use of multicast power control and AMC according to the VoIP traffic load and the channel condition. The proposed scheme has enhanced the system capacity by seamlessly maximizing the VoIP capacity or the data throughput according to the VoIP traffic load. The simulation results have shown that the proposed scheme can effectively be applied to the m-WiMAX system for the service of VoIP as well as data traffic.

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