QoS Packet-Scheduling Scheme for VoIP Services in IEEE 802.16e Systems

Jaeshin Jang, Jong-Hyup Lee, Seung-Kook Cheong, and Young-Sun Kim

Abstract: **The IEEE 802.16 wireless metropolitan area network (WMAN) standard is designed to correct expensive communication costs in CDMA-based mobile communication systems and limited coverage problems in wireless LAN systems. Thus, the IEEE 802.16e standard can provide mobile high-speed packet access between mobile stations and the Internet service provider through the base station with cheap communication fees. To efficiently accommodate voice over IP (VoIP) services in IEEE 802.16 systems, an uplink quality of service packet-scheduling scheme is proposed, and its performance is evaluated with an NS-2 network simulator in this paper. Numerical results show that this proposed scheme can increase the system capacity by 100% more than in the unsolicited rand service (UGS) scheme and 30% more than the extended real-time polling service (ertPS) scheme, respectively.**

Index Terms: **extended real-time polling service (ertPS), IEEE 802.16e, packet-scheduling, quality of service (QoS), unsolicited grant service (UGS).**

I. INTRODUCTION

Since CDMA-based mobile communication systems are targeted to provide voice and medium-rate data services for mobile stations moving at high-speed, providing data services in those systems results in high communications costs. In addition, shortages of resources make it impossible to provide high-speed data communication through those systems. Although wireless LAN systems can offer high-speed data services up to 54 Mbps, they have limited coverage when attempting to provide a wide-area service outdoors. In order to make up for weak points in mobile communication and wireless LAN systems, the IEEE 802.16 wireless metropolitan area network (WMAN) standard was proposed. The IEEE 802.16d [1] standard finalized in 2004, however, cannot support a terminal mobility, although it provides high-speed wireless communications between wireless terminals and the base station. Therefore, the IEEE 802.16e [2] standard was created in 2005 to provide a high-speed terminal mobility; this IEEE 802.16e system has been under commercial service in Korea from the latter half of 2006. The physical layer of this standard has orthogonal frequency division multiplexing (OFDM), multiple input multiple output (MIMO) and smart antenna technologies for enabling high-speed transmission and improving system performance and capacity. The MAC layer of this standard classifies all traffic services into 5 classes: Unsolicited grant service (UGS), extended real-time polling service (ertPS), real-time polling service (rtPS), non-real-time polling service (nrtPS), and best effort (BE) services; the MAC layer also describes the related QoS parameters. The ertPS service has been added in the IEEE 802.16e standard in order to efficiently accept variable rate voice over IP (VoIP) services. However, there is no essential quality of service (QoS) packet-scheduling scheme in the IEEE 802.16e standard, except QoS signaling between the base station and mobile stations. This is because implementing a QoS packet-scheduling scheme is considered to be a system developer's role. The typical type of initial data service is an asymmetric type, such as downloading text, images, or moving pictures from web servers. Thus, the CDMA2000 1xEV-DO or W-CDMA systems allocate more capacity on the downlink than on the uplink link and, as a result, researchers have interest only in the downlink packet-scheduling schemes. However, as symmetric data services such as video telephone and peer-to-peer (P2P) services become popular, finding a solution for efficient use in the uplink channel requires increasing attention. As a result, a large amount of research on uplink packet-scheduling schemes in IEEE 802.16e systems has been done.

In this paper, to keep pace with these research trends, an uplink QoS packet-scheduling scheme in IEEE 802.16e systems that can efficiently accommodate VoIP traffic services is proposed. Related research results on QoS packet-scheduling schemes are surveyed in Section II, while the uplink QoS packet-scheduling scheme proposed in this paper is described in Section III. After the numerical results of the proposed scheme are described and compared with those in the UGS and ertPS schemes in Section IV, this paper is concluded in Section V.

II. SURVEY OF RELATED RESEARCH

One packet-scheduling scheme that has been implemented and is commonly used in commercial mobile communication systems is the proportional fairness (PF) algorithm [3] in the CDMA2000 1xEV-DO system. Although this PF algorithm attained throughput maximization with assigning more transmission chances to mobile stations with a higher signal to noise ratio (SNR) value, it cannot provide QoS differentiation. To solve this limitation, an M-LWDF scheme [4] is proposed that could meet delay requirement and guarantee minimum transmission rates, but its performance is not confirmed. On the other hand, a lot of research on QoS packet-scheduling schemes in the IEEE 802.16 system has been conducted [5]–[8]. In reference [5], various scheduling schemes are surveyed and compared, including the proportional fairness scheme used in CDMA2000 1xEV-DO system. Reference [6] proposes uplink packet-scheduling and connection admission control (CAC) schemes for the IEEE 802.16d standard. Although the research output outlined in ref-

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erence [6] is good, the related database needed when implementing the proposed uplink packet-scheduling scheme is too complicated; it is assumed for the proposed CAC scheme that the total uplink capacity and each allocated capacity for each traffic service are constant. This, however, has a critical limitation, as link capacities in wireless communication are time-varying, especially due to the fading effect. In reference [7], an ertPS packet-scheduling scheme for VoIP service is proposed, and its performance is evaluated when VoIP traffic is assumed to use an enhanced variable rate coder (EVRC) codec. Reference [7], however, does not consider the additional packet arrivals during each scheduling interval when the base station determines uplink bandwidth assignments for all the mobile stations located within its coverage. Thus, this scheme can cause each packet's queuing delay to increase when the traffic load increases. In addition, this scheme has no solution for the problem of which mobile station the base station chooses first when there are many mobile stations in its coverage. A queuing-theoretic and optimization-based model for radio resource management in IEEE 802.16-based broadband wireless access networks was presented in reference [8]. The packet-scheduling scheme proposed in this paper is similar to the opportunistic scheduling described in references [9]–[11]. The modulation-assisted two-user opportunistic scheduling scheme was proposed in order to reduce average packet delay by offering more frequent channel access to users [9], [10]. And these literatures provided throughput performance which was only constrained by an SNR value of each modulation and coding scheme (MCS). However, packets from real-time traffic can be dropped additionaly due to excessive delay. The scheduling scheme proposed in this paper achieves both maximizing throughput and minimizing packet loss probability due to excessive delay more than maximum latency by choosing adaptive scheduling intervals. The VoIP traffic model considered in this paper is the EVRC traffic model, which is a voice codec widely used in CDMA-based mobile communication systems. In the EVRC traffic model, variable-sized voice packets are generated at 20msec intervals in four different modes: Full, half, quarter, and eighth modes [7]. All the previous research to date on the UGS and ertPS schemes considered only a constant scheduling interval for VoIP traffic service. However, it may be difficult to consistently assign a constant scheduling interval for VoIP traffic service especially, when the traffic load becomes heavy. Although using a constant scheduling interval for VoIP traffic service can decrease the delay jitter value, the four service classes other than the UGS service do not use the delay jitter parameter as a mandatory QoS parameter, according to the IEEE 802.16e standard. In addition, if the receiver has a solution for the delay jitter problem (i.e., a play-out buffer), it would be better to find a new QoS packet-scheduling scheme that could meet the QoS requirements and accommodate a much greater number of subscribers.

III. DESCRIPTION OF PROPOSED SCHEME

A TDM frame structure used in the IEEE 802.16e physical layer is shown in Fig. 1. One TDM frame consists of a downlink subframe and an uplink subframe in the TDD mode. Each subframe has numerous time slots. Information concerning in which time slots and in how many time slots each subscriber should send data packets is described in each UL-MAP control message and transmitted at the beginning of every TDM frame.

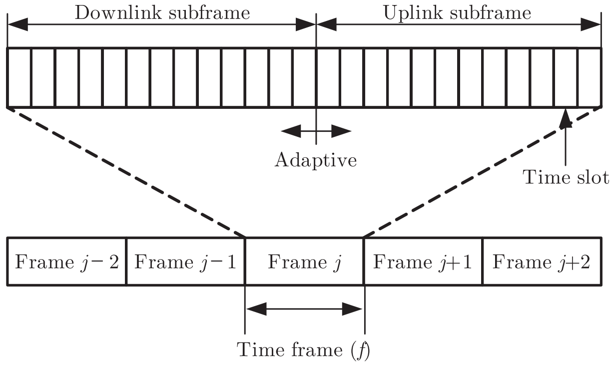

Fig. 1. IEEE 802.16e TDM frame structure.

In the IEEE 802.16 system with a link adaptation scheme, a mobile station uses a high-level modulation scheme such as 64QAM, when its channel status with the base station is good, while a low-level modulation scheme, such as QPSK, is used when its channel status is bad. In general, 64QAM can send three times more data during a symbol period than QPSK. The QoS packet-scheduling scheme proposed in this paper is able to allocate greater scheduling intervals to mobile stations working under good channel conditions and, thus, reduce the channel occupancy needed for sending the same size of data. Therefore, system capacity, i.e., the number of VoIP mobile stations that can be accommodated in a base station without violating the VoIP service's maximum latency, increases.

The key concepts of the QoS packet-scheduler proposed in this paper are to determine scheduling intervals and the amount of resource assignment, and to decide which mobile station the base station should serve first among multiple mobile stations. In order to make the QoS packet-scheduling scheme work properly in the uplink, as shown in the Fig. 2, each base station must extract information on queue sizes from packets sent in a piggy-back manner by mobile stations, and select the adequate MCS scheme based on the current channel status. It is assumed that channel status can be decided through measuring RSSI or SNR values at any mobile station's side. Subsequently, the base station derives scheduling intervals based on the selected MCS scheme (step 1), calculates how many resources to assign to each mobile station (step 2), and makes a transmission list (step 3) in which eligible mobile stations are lined up in the decreasing order of their transmission priorities, based on their transmission deadlines. This transmission list indicates which mobile station the base station should serve first within one TDM frame. The base station should load that scheduling information on a UL-MAP control message and send it to the mobile stations within its coverage according to QoS signaling procedure.

Mobile stations should store each packet arriving from the upper layer into the appropriate packet buffer. When mobile stations receive a UL-MAP message from the base station, they should fetch the packets from the appropriate packet buffer, of which the amount is described in the received UL-MAP mes-

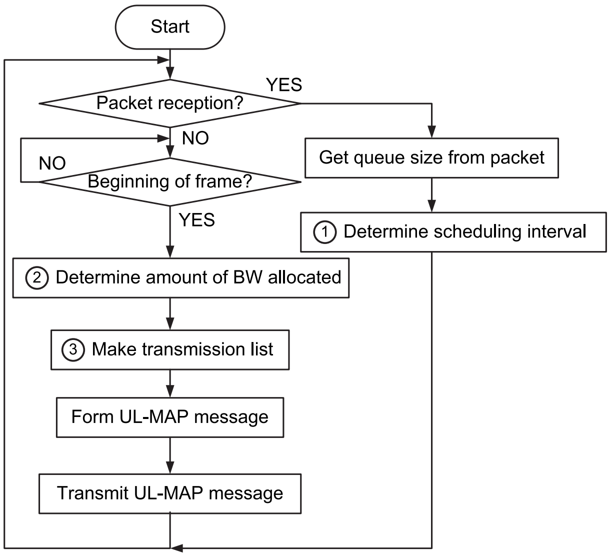

Fig. 2. Operation procedure at the base station side.

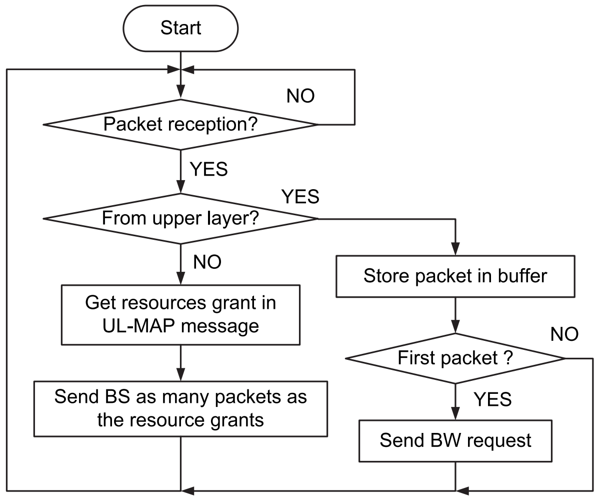

Fig. 3. Operation procedure at the mobile station side.

sage, and send those packets to the base station. Whenever mobile stations send packets, they should always add the information on current queue size of the packet buffer into the transmitted packet in a piggy-back manner. Mobile stations' operation is described in Fig. 3.

Before providing the full-scale description of the three key concepts for QoS packet-scheduling scheme, we define several notations. The frame size is f [sec], the maximum sustained traffic rate (MSTR) for ith subscriber is $r_i^{\max}$ [bps], and the maximum latency (ML) for ith subscriber is d_i [sec] or $m_i(= d_i/f)$ [frame]. MSTR and ML which is a kind of deadline are an example of traffic source descriptor. When a new connection arrives, the base station is informed of this traffic source descriptor. Each VoIP voice packet is assumed to be generated every k [frame]. As a result, the possible scheduling interval values become $1 \leq n_i \leq m_i - 1$ [frame]. Since, however, it is of no value that the scheduling interval is less than k [frame], the appropriate scheduling interval values are $k \leq n_i \leq m_i - 1$ [frame].

The first step for the QoS packet-scheduling scheme is to determine each mobile station's scheduling interval based on the selected MCS scheme. The key concept for the first step is to assign long scheduling intervals for mobile stations in good channel state and short scheduling intervals for mobile stations in bad channel state. Let us assume that there are M MCS schemes. Let's distribute all the possible scheduling interval values evenly among the M MCS schemes. Then, the problem of finding mobile stations' scheduling intervals can be solved as following two procedures:

- Assign $s_i(= \lfloor (m_i - k)/M \rfloor)$ scheduling intervals to every MCS scheme.
- Assign all the remainder $(m_i - k)\%M$ scheduling intervals to higher MCS schemes, one value per each MCS scheme from the highest-level MCS scheme in the descending order.

For instance, let us assume that the frame size is $f = 5$ msec, the ML is 100 msec, and the VoIP inter-packet generation time is $k = 4$ [frame]. We also assume that there are 6 MCS schemes in this example. Thus, the possible scheduling interval values are $n_i = 4, 5, \cdots, 19$ [frame] and the scheduling interval values for each MCS scheme become as follows according to the two procedures described above:

$$\{\mathrm{MCS}_1, \mathrm{MCS}_2, \mathrm{MCS}_3, \mathrm{MCS}_4, \mathrm{MCS}_5, \mathrm{MCS}_6\} = \{(4,5), (6,7), (8,9,10), (11,12,13), (14,15,16), (17,18,19)\}.$$

The second step for the QoS packet-scheduling scheme is to determine the amount of resources allocated for each eligible mobile station. With the values of the scheduling interval n_i determined from the first step and the queue size BR_i notified by ith mobile station, the amount of resources assigned for each mobile station can be calculated by the following (1) or (2). In (1), $(r_i^{\max} f n_i)$ means the maximum amount of resources that can be assigned during one scheduling interval and the parameter r_{time} described in (3) is used for compensating for the difference between the expected scheduling time and the current time for eligible mobile stations.

$$R_i = \begin{cases} r_i^{\max}(f n_i + r_{\text{time}}) \text{ [bit]}, & \text{for UGS service} \\ \min(r_i^{\max}(f n_i + r_{\text{time}}), K_i) \text{ [bit]}, & \text{for ertPS service} \end{cases} \tag{1}$$

$$\text{where} \quad K_i = BR_i + \nu_i(f n_i + r_{\text{time}})$$
$$S_i = \lceil R_i / b_i \rceil \text{ [slot]}. \tag{2}$$

The variable K_i in (1) is the sum of the queue size BR_i notified at about the previous scheduling time and the estimated amount of packets arriving during scheduling interval. The variable ν_i is the current VoIP coder status as shown in reference [7] or the average VoIP packet generation rate as a unit of bits per second. The parameter b_i in (2) is the size of packets in bits which can be transmitted during a slot for each MCS scheme; the values used in this paper are shown in the Table 1.

The last step of the QoS packet-scheduling scheme proposed in this paper is to make a transmission list. Since scheduling intervals for mobile stations are variable based on the chosen MCS scheme, each subscriber's remaining time until its deadline is different. Especially, when there is a shortage of uplink

Table 1. System parameters for performance evaluation.

System parameters	Values (*)	System parameters	Values
64QAM-3/4	27 / 7.4	Simulation time	200 sec
64QAM-2/3	24 / 10.7	Maximum latency	100 msec
16QAM-3/4	18 / 14.8	# of slots per an uplink subframe	40
16QAM-1/2	12 / 20.8	Frame size	5 msec
QPSK-3/4	9 / 10.9	Uplink : Downlink (subframe)	3:2
QPSK-1/2	6 / 35.4	Cell topology	circle

(*) : b_i [byte] / occupance [%]

resources for accommodating all the eligible mobile stations within a TDM frame, it is needed to assign a transmission priority to each mobile station based on its remaining deadline, and allocate resources to mobile stations according to their transmission priorities. A new parameter r_{time} is defined here, which is somewhat related to the remaining deadline; this parameter signifies the extent to which the resource assignment is delayed in rel ation to the expected scheduling time. Thus, its value is proportionate to the mobile station's transmission priority.

$$r_{\text{time}} = \text{current time} - (\text{previous scheduled time} \\ + \text{scheduling interval}). \quad (3)$$

IV. NUMERICAL RESULTS

The cell diameter is assumed to be 1 km here and all the mobile stations are assumed to move only in four directions, i.e., east, west, south, and north, using the system parameters described in Table 1. The mobile station's speed and movement duration are assumed to be uniformly distributed among $1{\sim}167$ m/sec and $1{\sim}5$ sec, respectively. After movement direction and speed are determined, a mobile station is assumed to move with the decided speed until the end of the movement duration. The ratio of uplink subframe to downlink subframe is fixed at 3:2. VoIP packets of EVRC traffic model are assumed to be generated at full rate with 29%, half rate with 4%, quarter rate with 7%, and eighth rate with 60%. The sizes of VoIP packets generated in each rate are 171, 80, 40, and 16 bits, respectively [7]. The other system parameters used in this paper for performance evaluation are described in Table 1. When mobile stations are stationary, it is assumed that they use only the QPSK-1/2 modulation scheme.

Two packet-scheduling schemes are chosen for performance comparison with the QoS packet-scheduling scheme proposed in this paper.

- UGS scheme: Described in the IEEE 802.16e standard; 22 bytes are allocated every 20 msec for MSTR.
- ertPS scheme: Described in the IEEE 802.16e standard.
- New scheme: Proposed scheme in this paper.

For computer simulation, the NS-2 network simulator [12] is used and the performance measures are packet dropping probability, average packet delay in the buffer and average number of occupied slots. First, we describe the performance results when

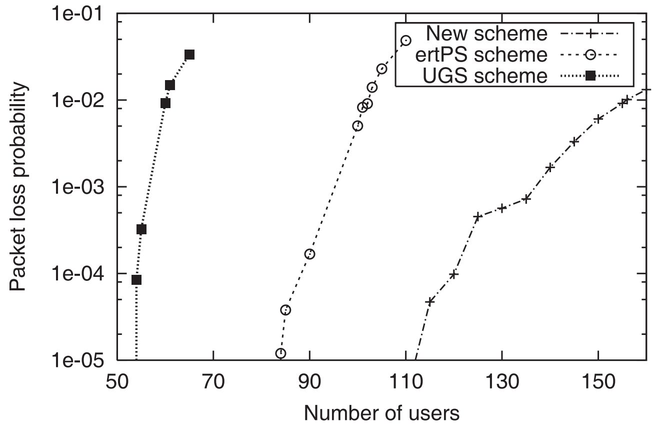

Fig. 4. Comparison of packet dropping probability.

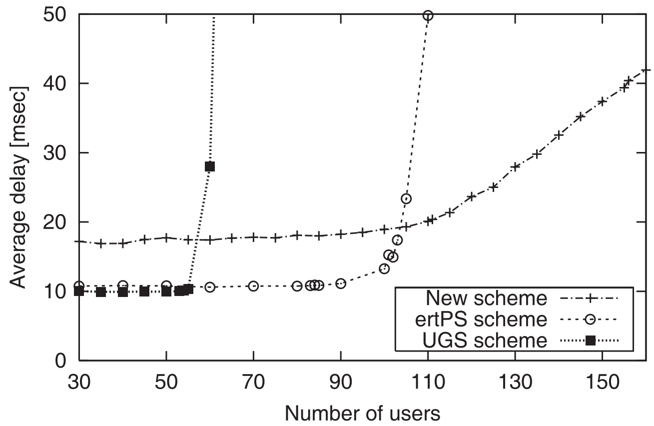

Fig. 5. Comparison of average packet delay.

subscribers are mobile. Performance results on packet dropping probability are shown in Fig. 4. Every VoIP packet is dropped only when its waiting delay in the buffer is larger than the maximum latency. This figure shows that the UGS scheme can accommodate up to 53 subscribers without any packet drop and up to 60 subscribers with packet dropping probability less than 1%. The ertPS scheme can accommodate up to 83 subscribers without any packet drop and up to 102 subscribers with packet dropping probability less than 1%. However, the scheme proposed in this paper can accommodate up to 110 subscribers without any packet drop and up to 155 subscribers with packet dropping probability less than 1%.

The performance results regarding average packet delay are shown in Fig. 5. According to the results, the average packet delay of the proposed scheme is larger than those of the other two schemes. That is because the scheduling interval at the proposed scheme is variable and larger than 20 msec, which causes the average waiting time in the buffer to increase. However, the average delay value increases more slowly than in UGS or ertPS schemes as the number of users increases. That is because managing the transmission list contributes highly to transmitting lots of packets within their deadlines under heavier traffic load and thus causing the average delay to increase slowly.

The performance results in terms of the average number of slots used in a TDM frame are shown in Fig. 6. The number of

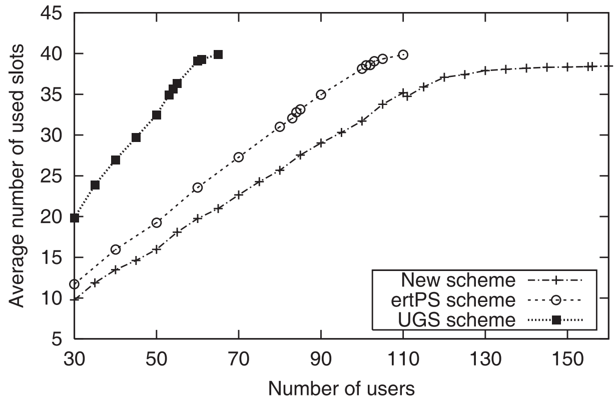

Fig. 6. Comparison of average number of occupied slots.

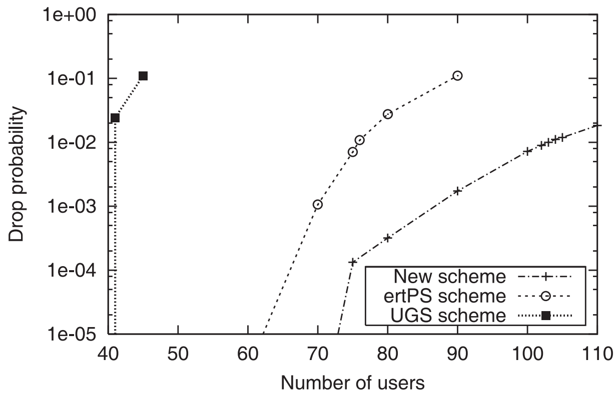

Fig. 7. Comparison of packet dropping probability (stationary).

slots for the TDM frame used in this performance evaluation is 40. For the UGS and ertPS schemes, the average number of occupied slots increases to 40 when the number of accommodated subscribers reaches 60 and 110, respectively. The new scheme, on the other hand, can occupy fewer than 40 slots on average even when the number of accommodated users is about 160. This result shows that the proposed scheme uses its resources more efficiently.

We now turn to describing the performance results for stationary subscribers. The performance results in terms of packet dropping probability are shown in Fig. 7. In the UGS scheme, the base station should assign 22 bytes of resources every 20 msec in order to accept a VoIP service with a variable rate. When using the QPSK-1/2 MCS scheme, a mobile station needs at least four slots for 22 bytes of resources as shown in Table 1. Since there are 40 slots in a TDM frame and there are 4 TDM frames during the EVRC packet inter-arrival time, it is not difficult to determine that the maximum number of subscribers for UGS scheme is 40 as shown in the following equation.

$$\frac{1}{4}\ [\text{mobile/slot}] \times 40\ [\text{slot/TDM frame}] \times 4\ [\text{TDM frame} \\ /\text{frame interval}] = 40\ [\text{mobile/frame interval}].$$

This same result can be easily ascertained from Fig. 7. When one base station accommodates more than 40 subscribers in the UGS scheme, packet dropping probability jumps abruptly to above 1%. The maximum number of subscribers without any

Table 2. System capacities of the three scheduling schemes.

Schemes	Mobility	System capacity	
		$P_{\text{drop}} = 0\%$	$P_{\text{drop}} < 1\%$
UGS scheme	Mobile	53	60
	Static	40	40
ertPS scheme	Mobile	83	102
	Static	60	75
New scheme	Mobile	110	155
	Static	70	103

P_{drop}: Packet dropping probability

packet drop is 40, 60, and 70 subscribers for the UGS, ertPS, and the new scheme, respectively.

From the packet dropping probability results, the system capacity which is defined as the number of VoIP mobile stations that can be accommodated in a base station without violating the VoIP service's maximum latency is derived in Table 2. From the results, it appears that the proposed scheme increases the system capacity by about 100% more than in the UGS scheme and by about 30% more than in the ertPS scheme.

V. CONCLUSION

The IEEE 802.16e WMAN system is called a 3.5G system because this system can increase the system capacity much more than the 3G system. The MAC layer describes QoS parameters and signaling procedures required for accommodating five traffic services. However, prescribing a QoS packet-scheduling scheme is out of standard and, thus, base station developers should implement the scheduling scheme in order to provide a differentiated QoS service and increased system capacity. In this paper, an uplink QoS packet-scheduling scheme is proposed and its performance is evaluated with the help of an NS-2 network simulator. The key concept of this proposed scheme is to allocate long scheduling intervals for mobile stations with good channel state and short scheduling intervals for mobile stations with bad channel state. In addition to determining scheduling intervals, this scheme determines the amount of resources to be assigned at each scheduling time, and makes a transmission list indicating which mobile stations the base station chooses first for packet transmission. Average packet delay, packet dropping probability and average amount of occupied slots within a TDM frame are used as performance measures. From the performance results using the NS-2 network simulator, this proposed scheme can increase the system capacity by about 100% more than in the UGS scheme about 30% more than in the ertPS scheme. Therefore, if this QoS packet-scheduling scheme is applied in base stations of a commercial IEEE 802.16e system, it will help to increase the system capacity.

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