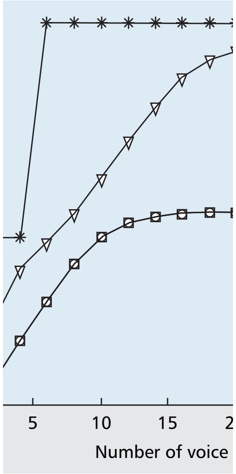
PERFORMANCE OF PACKET VOICE TRANSMISSION USING IEEE 802.16 PROTOCOL

DONGMEI ZHAO, MCMASTER UNIVERSITY
XUEMIN (SHERMAN) SHEN, UNIVERSITY OF WATERLOO



The authors study the performance of voice packet transmissions and BS resource utilization using the three types of scheduling services in IEEE 802.16-based backhaul networks, where each SS forwards packets for a number of voice connections.

ABSTRACT

The IEEE 802.16 standard defines three types of scheduling services for supporting real-time traffic, unsolicited grant service (UGS), real-time polling service (rtPS), and extended real-time polling service (ertPS). In the UGS service, the base station (BS) offers a fixed amount of bandwidth to a subscriber station (SS) periodically, and the SS does not have to make any explicit bandwidth requests. The bandwidth allocation in the rtPS service is updated periodically in the way that the BS periodically polls the SS, which makes a bandwidth request at the specified uplink time slots and receives a bandwidth grant in the following downlink subframe. In the ertPS service, the BS keeps offering the same amount of bandwidth to the SS unless explicitly requested by the SS. The SS makes a bandwidth request only if its required transmission rate changes. In this article we study the performance of voice packet transmissions and BS resource utilization using the three types of scheduling services in IEEE 802.16-based backhaul networks, where each SS forwards packets for a number of voice connections. Our results demonstrate that while the UGS service achieves the best latency performance, the rtPS service can more efficiently utilize the BS resource and flexibly trade-off between packet transmission performance and BS resource allocation efficiency; and appropriately choosing the MAC frame size is important in both the rtPS and ertPS services to reduce packet transmission delay and loss rate.

INTRODUCTION

Wireless metropolitan area networks (MANs) based on IEEE 802.16 are expected to have wide deployment in the near future. Beyond just providing a single last-hop access to wireline backbone networks, such as the Internet, the IEEE 802.16-based technology can be used for creating wide-area wireless backhaul networks, such as backhaul for connecting the radio network controllers with base stations in cellular networks and for connecting Wi-Fi-based routers, for coverage extension with rapid and low-cost deployment. The IEEE 802.16 standard is designed for point-to-multipoint configurations, where several subscriber stations (SSs) are associated with a central base station (BS). Optional mesh deployment is also available, where SSs can communicate with each other. When used for backhaul transmissions, each SS is usually responsible for forwarding traffic for a number of end stations or connections.

The IEEE 802.16 standard specifies five different scheduling services in the uplink: unsolicited grant service (UGS), real-time polling service (rtPS), extended real-time polling service (ertPS), nonreal-time polling service (nrtPS), and best effort (BE) service. The first three types of services are used to support real-time traffic. In the UGS service, the BS offers a fixed size burst in time slots to an SS periodically, and the SS does not have to make any explicit bandwidth requests. Therefore, the UGS service is simple and suitable for real-time constant bit rate traffic. When it is used to support packet voice traffic with alternate active and silent periods, the reserved bandwidth is wasted when a corresponding UGS connection is inactive. Furthermore, when used in backhaul networks, the fixed transmission rate cannot catch the changes of aggregate packet arrival rate from multiple voice connections. For the rtPS service, the BS periodically polls the SS, which makes a bandwidth request at the specified uplink time slot, and waits for the bandwidth grant from the BS through downlink transmission. The rtPS service is designed for real-time services with variable packet generation rates. In the rtPS service the SS can request for a different rate every time when it is polled and update the bandwidth requests based on the traffic load changes. In this way, the bandwidth allocation in the rtPS service is updated periodically. The rtPS service is more flexible, but introduces extra signaling overhead and delay. The ertPS service is a new addition in IEEE 802.16e. In this service, the BS keeps offering the same amount of bandwidth to the SS unless explicitly requested by the SS. The

SS makes a bandwidth request only if its required transmission rate changes. Therefore, if the SS generates packets at a constant rate, it does not need to update its bandwidth requests; then the ertPS service works in the same way as the UGS service. On the other hand, when the packet generation rate from the source is changed, the SS can request for bandwidth changes, similar to that in the rtPS service. The ertPS service was originally designed to combine the simplicity of the UGS service and flexibility of the rtPS service for supporting real-time packet voice services with alternate active and silent periods. When used in backhaul networks, the packet arrival process is an aggregate process from a number of connections, and the SS may have to update bandwidth requests frequently whenever any of the associated connections changes its status. The performance of the ertPS service in such a scenario should be investigated. The nrtPS and BE services are for non-real-time traffic. In the nrtPS service, the BS polls the connections on a regular basis, depending on requested minimum rates and other parameters. The nrtPS connections receive fewer polling opportunities during network congestion and are allowed to use contention-based bandwidth requests. In the BE service, the SS can either use unicast bandwidth requests or contention request opportunities to request bandwidth.

Although the 802.16 standard specifies different scheduling services and QoS mechanisms, it does not provide details of scheduling and reservation management. Research has been done for efficient resource management and some scheduling schemes have been proposed in the literature. In [1], a scheduling scheme is proposed for VoIP connections with alternate ON and OFF activities, and one reserved bit is used in the MAC header for the SS to inform the BS of the status transitions of its voice connection. The MAC layer performance of 802.16 is studied through simulation in [2], which compares queuing performance of different scheduling schemes, effect of the ratio between the downlink and uplink subframe durations, channel conditions, and other factors on packet level performance. In the cross-layer scheduling scheme proposed in [3], each connection can use adaptive modulation and coding schemes at the physical layer to optimize the MAC layer throughput. A queue-aware bandwidth allocation and rate control mechanism is proposed in [4] for polling services. Bandwidth allocation given in [5] is adaptive to the channel conditions and jointly considered with connection admission control. A subchannel and rate allocation framework is proposed in [6] for IEEE 802.16-based mesh networks. In the QoS performance study of [7], deficit round robin and weighted round robin are used for uplink and downlink scheduling, respectively.

In this article we study the packet voice transmission performance and BS resource utilization in IEEE 802.16-based backhaul networks, where each SS is responsible for forwarding packets for a number of voice connections. Instead of studying how to share the BS resources among connections associated with multiple SSs for a given scheduling service, we emphasize the effect of different scheduling services on the voice packet transmission performance and BS resource allocation efficiency. The remainder of the article is organized as follows. We describe the system that this work is based on, and describe details of the real-time scheduling services when used for supporting packet voice traffic. We demonstrate the simulation results of the voice packet transmission performance and BS resource allocation efficiency using the different scheduling services, followed by the conclusion of the article.

SYSTEM DESCRIPTION

We consider that N_s SSs are connected to the BS using the IEEE 802.16 protocol. A number of voice connections are associated with each of the SSs. The packet generation process for a voice connection has alternate active and silent periods and can be modeled as an ON-OFF source. The ON and OFF time intervals are both approximated as exponentially distributed. During the OFF states, there is no packet generated. During the ON states, there is a constant packet generation rate, R_{voice} b/s. Let T_{on} and T_{off} respectively represent the average length of the ON and OFF intervals. Denote P_{on} as the probability that the source is being ON. Then we have

$$P_{\text{on}} = \frac{T_{\text{on}}}{T_{\text{on}} + T_{\text{off}}}.$$

All voice packets have the same length, and the bit sequence from an active voice source every T_{voice} seconds is packed into a voice packet. Therefore, the transmission time required for each voice packet is given by

$$T_p = \frac{R_{\text{voice}} T_{\text{voice}} + L_{\text{head}}}{R_b},$$

where R_b is the physical-layer transmission throughput in b/s, and L_{head} is the total header size at the physical, MAC and higher layers. Let N_i be the number of connections associated to SS i, $i = 1, 2, \ldots, N_s$. Then the number of active connections, or connections in the ON states, associated to SS i at a particular moment is a random variable following a binomial distribution with parameters N_i and P_{on}. All generated voice packets from the active connections associated with the same SS are stored in a shared buffer at the SS and are served on first-come-first-serve basis. Each voice packet has a maximum tolerable delay, $D_{\max}$, and a packet is discarded if it cannot be transmitted within $D_{\max}$ after it is generated. We study the average delay and packet loss performance for voice packet transmissions, as well as the BS uplink resource utilization. In the remaining part of the section, we briefly describe the three real-time scheduling services, UGS, rtPS, and ertPS, for packet voice traffic.

UGS

In the UGS service, the BS allocates the peak rate to each SS. With N_i connections, the total required transmission rate from SS i is $N_i R_{\text{voice}}$

Although the 802.16 standard specifies different scheduling services and QoS mechanisms, it does not provide details of scheduling and reservation management. Research has been done for efficient resource management and some scheduling schemes have been proposed in the literature.

The idea of the rtPS_A service is to reduce the packet transmission delay by allowing the SS to request bandwidth not only for packets that are in its buffer, but also for packets that may arrive before the next bandwidth request.

b/s, or $\lfloor N_i T_{\text{MAC}}/T_{\text{voice}} \rfloor$ packets per MAC frame, where T_{MAC} is the duration of one MAC frame. The maximum number of voice connections that can be accommodated using the peak rate allocation is given by

$$C_{\max,0} = \left\lfloor \frac{T_{\text{UGS,max}} T_{\text{voice}}}{T_p T_{\text{MAC}}} \right\rfloor,$$

where $T_{\text{UGS,max}}$ is the maximum amount of time that an SS is allowed to use in each uplink subframe. The advantage of using the peak rate allocation is that, once a connection is admitted, no further resource request/grant signaling exchange is needed, and packets can be transmitted with the minimum delay. However, since each voice connection has a fraction of $(1 - P_{\text{on}})$ time in the OFF states, the same fraction of the reserved BS resources is wasted using the peak rate allocation. The low latency of packet transmissions is achieved at a price of low efficiency in utilizing the BS resources. On the other hand, when a certain queuing delay and packet loss rate can be tolerated, more than $C_{\max,0}$ voice connections can be admitted in the system by statistically sharing resources among connections associated with the same SS.

RTPS

The rtPS connections are polled by the BS regardless of network load. The polling rate should meet the QoS requirements of the connections, such as delay and packet loss rate. We define a polling interval, k, for an SS to be the number of MAC frames between two successive polls to the SS. The BS periodically polls the SS every k MAC frames through the downlink. Upon receiving the polling information, the SS makes a bandwidth request through the next uplink subframe. The bandwidth request/grant process takes about one MAC frame time: if the bandwidth request is sent from the SS in the current uplink subframe, the bandwidth grant is sent back from the BS in the following downlink subframe, and the SS is allowed to use the allocated bandwidth in the next uplink subframe. Before the new bandwidth can take effect packets in the buffer are served using the old bandwidth. We consider two different variations when the SS makes bandwidth requests: rtPS_B and rtPS_A.

In the rtPS_B, upon receiving a poll, the SS checks its own buffer size and makes a bandwidth request to the BS for transmitting B packets per polling interval, where B is the total number of packets in the SS buffer at the time of making the bandwidth request. Upon receiving this request, the BS checks its resource availability and sends a bandwidth grant message with a transmission rate of $\min(B, M_{\text{rtPS,max}})$ packets per polling interval to the SS in the following downlink subframe, where $M_{\text{rtPS,max}}$ is the maximum number of packets that the SS can transmit in the uplink during one polling interval (k MAC frames). Using the rtPS_B service, the SS only requests bandwidth for packets in its buffer. Because of the bandwidth request/grant process, all packets have to experience at least one MAC frame delay. Besides this, all packets should wait for the next poll from the BS after they arrive at the SS buffer, and this approximately introduces a delay of $k/2$ MAC frames on average. There is also queuing delay at a high traffic load when B is larger than $M_{\text{rtPS,max}}$.

The idea of the rtPS_A service is to reduce the packet transmission delay by allowing the SS to request bandwidth not only for packets that are in its buffer, but also for packets that may arrive before the next bandwidth request. In the rtPS_A, the requested bandwidth from SS i to the BS is $B + \lceil n_i k T_{\text{MAC}}/T_{\text{voice}} \rceil$ packets per polling interval, where n_i is the number of active connections currently associated with SS i. This request is sent to the BS every k MAC frames. The bandwidth grant process in the rtPS_A is the same as that in the rtPS_B.

ERTPS

In the ertPS service, the SS keeps using its current transmission rate until the aggregate packet generation rate from its associated connections changes (i.e., the total number of active connections changes). When this occurs, the SS updates its bandwidth request. According to the standard, the SS can use allocated bandwidth, piggyback its new bandwidth request, or use contention-based transmission opportunities if the current available transmission rate is zero. Since contention-based transmission does not guarantee a strict delay requirement, and piggyback request is optional for the SS and only for incremental request, we consider to use preallocated bandwidth from the BS for the SS to update its bandwidth requests. In this way, bandwidth requests can be updated with a minimum delay. The requested bandwidth should be $T_{\text{ertPS,req}} = \max(T_{\text{BW\_REQ}}, n_i \lceil T_{\text{MAC}}/T_{\text{voice}} \rceil T_p)$ seconds per MAC frame, where n_i is the total number of active connections at the time of making the bandwidth request, and $T_{\text{BW\_REQ}}$ is the time for transmitting one bandwidth request message. That is, the amount of requested bandwidth is equal to the aggregate packet generation rate of all active connections if at least one connection is active, or the amount of bandwidth for transmitting one bandwidth request message if no connection is active. The amount of granted bandwidth is equal to $\min(T_{\text{ertPS,req}}, T_{\text{ertPS,max}})$ seconds per MAC frame, where $T_{\text{ertPS,max}}$ is the maximum amount of time available for the SS in an uplink subframe.

In the ertPS service, packets are delayed for transmission when the new rate is higher than the current rate. On the other hand, some bandwidth may be wasted during the bandwidth updating process if the new rate is lower than the current one.

PERFORMANCE EVALUATION

We consider an IEEE 802.16-based backhaul network where five SSs (i.e., $N_s = 5$) are connected to the BS, and an equal number of voice connections are associated with each of the SSs. The system may also have other types of traffic, such as real-time video traffic and best effort traffic. Although having real-time video traffic in the same network may affect the amount of available resources for voice traffic, it should only affect the voice packet transmission performance quantitatively but not qualitatively in low and medium traffic loads, since packet transmission delay and loss are mainly caused by the mechanism of bandwidth requests and grants. Having best-effort traffic in the network should not affect the voice traffic performance, since voice traffic is usually given a higher priority than best-effort traffic. The physical layer is OFDM, and both the uplink and the downlink share the same frequency. All MAC frames have the same length. In each MAC frame, half of the time is for the uplink subframe, and the other half is for the downlink subframe. Except for a fixed amount of time, T_{overhead}, for contention-based transmissions, initial ranging, and other functions, the remaining time in the uplink subframe can be used for voice traffic, including both bandwidth requests and packet transmissions. In order to isolate the effect of different scheduling services on the voice transmission performance, we do not consider statistical resource sharing among different SSs. All the SSs share an equal amount of the BS uplink subframe time. For the UGS service, the maximum amount of time that each SS is allowed to use per MAC frame is given by

$$T_{\text{UGS,max}} = \frac{T_{\text{MAC\_UP}} - T_{\text{overhead}}}{N_S}.$$

For the rtPS service, the maximum number of packets that each SS can transmit in one MAC frame is given by

$$M_{\text{rPS,max}} = \left\lfloor \frac{T_{\text{MAC\_UP}} - T_{\text{overhead}} - N_s T_{\text{BW\_REQ}}}{N_S T_p} \right\rfloor.$$

For the ertPS service, the maximum amount of time that each SS is allowed to transmit is given by

$$T_{\text{ertPS,max}} = \frac{T_{\text{MAC\_UP}} - T_{\text{overhead}}}{N_S}.$$

The parameters used for the OFDM physical layer and the MAC layer are listed in the first and second parts of Table 1, respectively. With BPSK modulation and a channel coding rate of 1/2 at the physical layer, packet transmissions at the MAC layer are assumed to be error free. Parameters for the voice traffic are listed in the third part of Table 1. We examine voice packet transmission average delay and loss rate and the BS resource allocation efficiency using the UGS, rtPS (both rtPS_B and rtPS_A), and ertPS scheduling services. It is possible that different connections may require different scheduling services, but this work emphasizes the effect of the scheduling services on the packet transmission performance. In the simulated system, all connections require the same scheduling service.

PHY layer parameter	Value
Channel bandwidth BW	20 MHz
Sampling factor n	144/125
Number of carriers N_{FFT}	256
Number of data carriers N_{data}	192
Sampling rate F_s	$\lfloor n \times BW/8000 \rfloor \times 8000 = 23.04$ MHz
Useful symbol time T_B	$256/F_s = 11.11$ µs
CP time T_G	$1/4T_B = 2.78$ µs
Symbol time T_{sym}	$T_G + T_B = 13.89$ µs
Channel coding rate r	1/2
Physical data transmission throughput R_b	$rN_{data}/T_{\text{sym}} = 6.91$ Mb/s
MAC layer parameter	**Value**
MAC frame duration T_{MAC}	5 ms, 10 ms, 20 ms
TTG (RTG) duration	26 µs
Initial ranging period duration T_{overhead}	312 µs
Time for each BW request message T_{BWreq}	27.78 µs
Uplink burst preamble T_{pre}	11.11 µs
Time for each voice packet T_{packet}	$(L_{\text{packet}} + L_{\text{head}})/R_b + T_{pre} = 203.3$ µs
Polling interval for rtPS service k	1 MAC frame
Voice traffic parameter	**Value**
Voice mean ON time T_{on}	240 ms
Voice mean OFF time T_{off}	400 ms
Voice packet generation rate R_{voice}	64 kb/s
Voice packetization time T_{voice}	20 ms
Voice packet maximum delay D_{max}	60 ms
Voice packet payload L_{packet}	$R_{\text{voice}} \times T_{\text{voice}} = 1280$ bits

Table 1. *Default simulation parameters.*

Average Transmission Delay

Figure 1 compares the average packet delay performance of the three scheduling services with different MAC frame sizes. It can be seen that using the UGS service always achieves the best mean delay performance among all the scheduling services, since bandwidth is allocated to the SS before packets arrive at the buffer. For the UGS service, all packets generated during the BS downlink subframe periods need to be delayed until the next uplink subframe. This is the main reason causing packet transmission delay at a low traffic load. When the number of connections per SS is larger than $C_{\max,0}$, the connections cannot be served at the peak rate. As N_i increases, the queuing delay due to statistically sharing the BS resources increases and gradually dominates the total packet transmission delay.

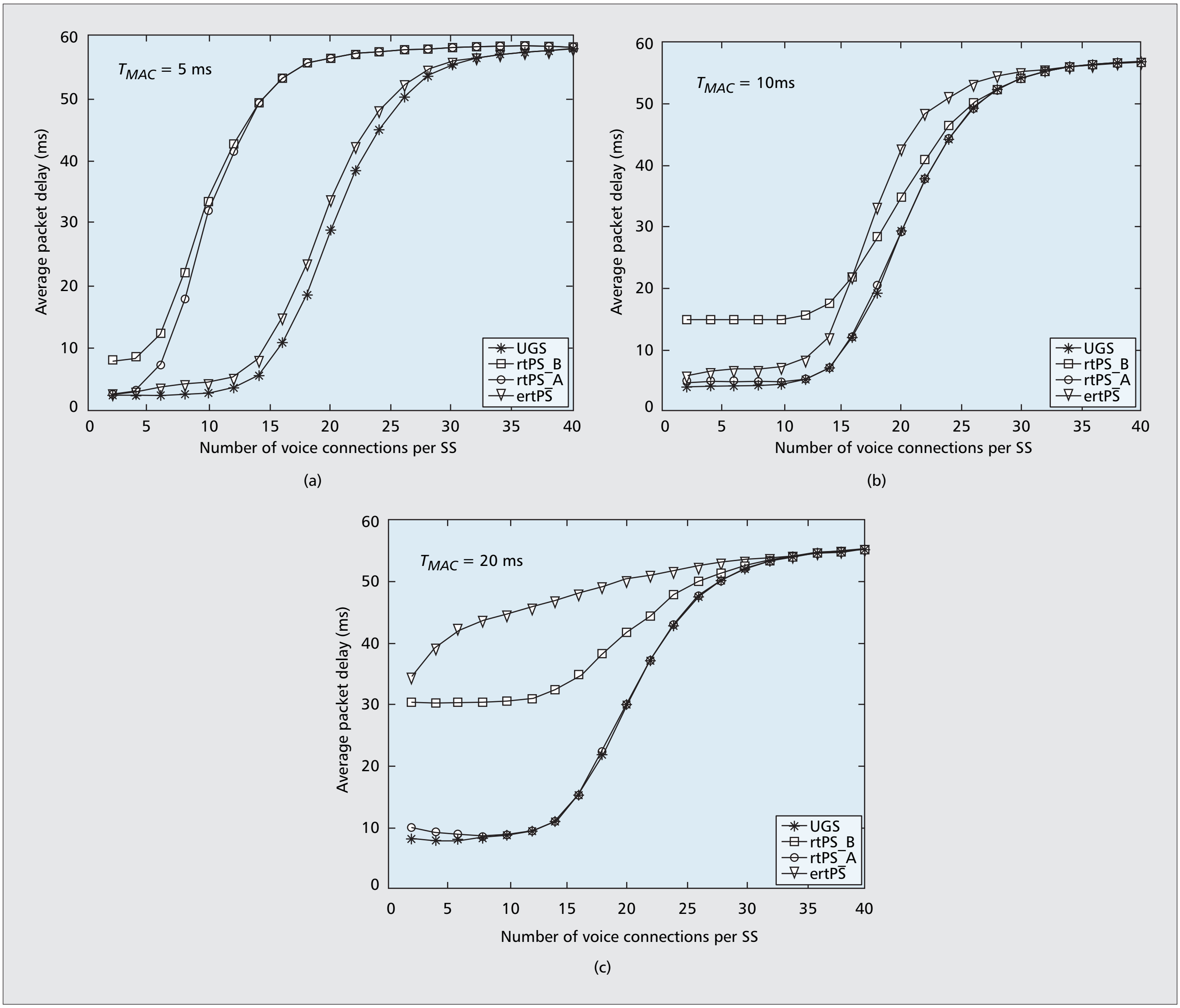
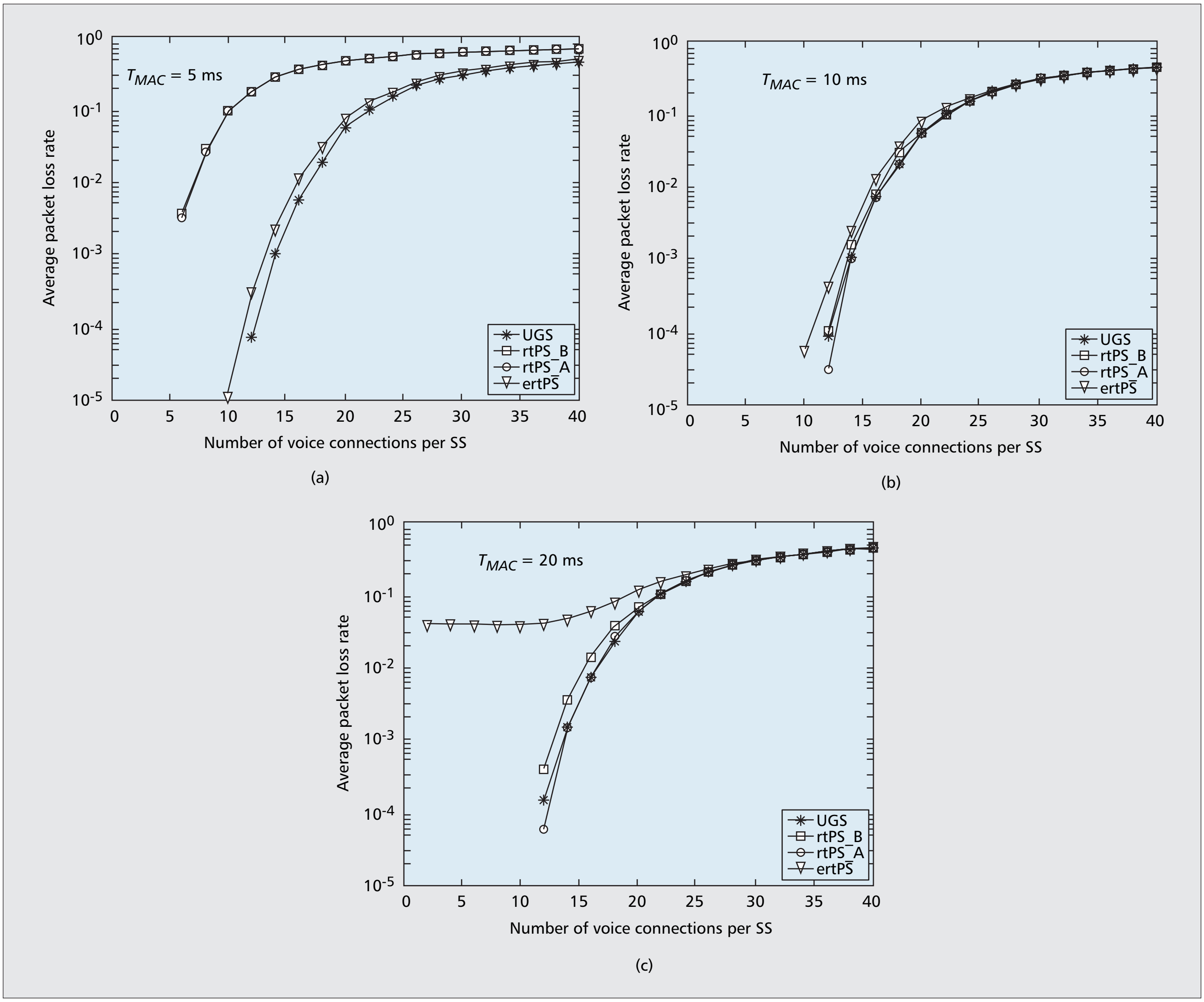


Figure 1. *Average packet transmission delay: a)* T_{MAC} *= 5 ms; b)* T_{MAC} *= 10 ms; c)* T_{MAC} *= 20 ms.*

Using the rtPS_B service always results in a higher delay than that using the UGS. In the rtPS_B service, packets are first stored in the SS buffer before the SS requests resources from the BS. This process greatly delays the packet transmissions. The resource request and grant process takes about one MAC frame duration. Therefore, a longer MAC frame results in higher packet transmission delay. In addition, as the traffic load increases, the number of packets arriving in one polling interval may exceed $M_{\text{rtPS,max}}$, and transmissions of the packets generated in the polling interval need to be delayed to a later polling interval. This causes queuing delay, which increases with the traffic load.

Using the rtPS_A service always achieves better delay performance than that using rtPS_B, and the difference is more obvious when T_{MAC} is larger. This is because that the rtPS_A service allows the BS to request resources before packets arrive, which reduces the average packet transmission delay. More packets can benefit from this mechanism when T_{MAC} is larger.

The packet delay performance using the ertPS service strongly depends on the MAC frame duration. As shown in Fig. 1, when T_{MAC} is 5 ms, the average packet transmission delay using the ertPS service is very close to that using the UGS service, and much less than that using either the rtPS_A or rtPS_B service; when T_{MAC} is 20 ms, using the ertPS service achieves the worst average delay performance among all three types of services; and when T_{MAC} is 10 ms, the average delay using the ertPS service is larger than that using both the UGS and rtPS_A, but smaller than that using the rtPS_B at a low traffic load. When the traffic load is high, using the ertPS service results in the highest delay among all the scheduling services. The above observations can be explained from the way that the ertPS service works. With a short MAC frame, the bandwidth update for the SS can be done quickly when the value of n_i changes. The SS can receive at most one bandwidth update in every MAC frame, while the value of n_i may change more than once during one MAC frame, especially for a long MAC frame size or high traffic load. Therefore, when



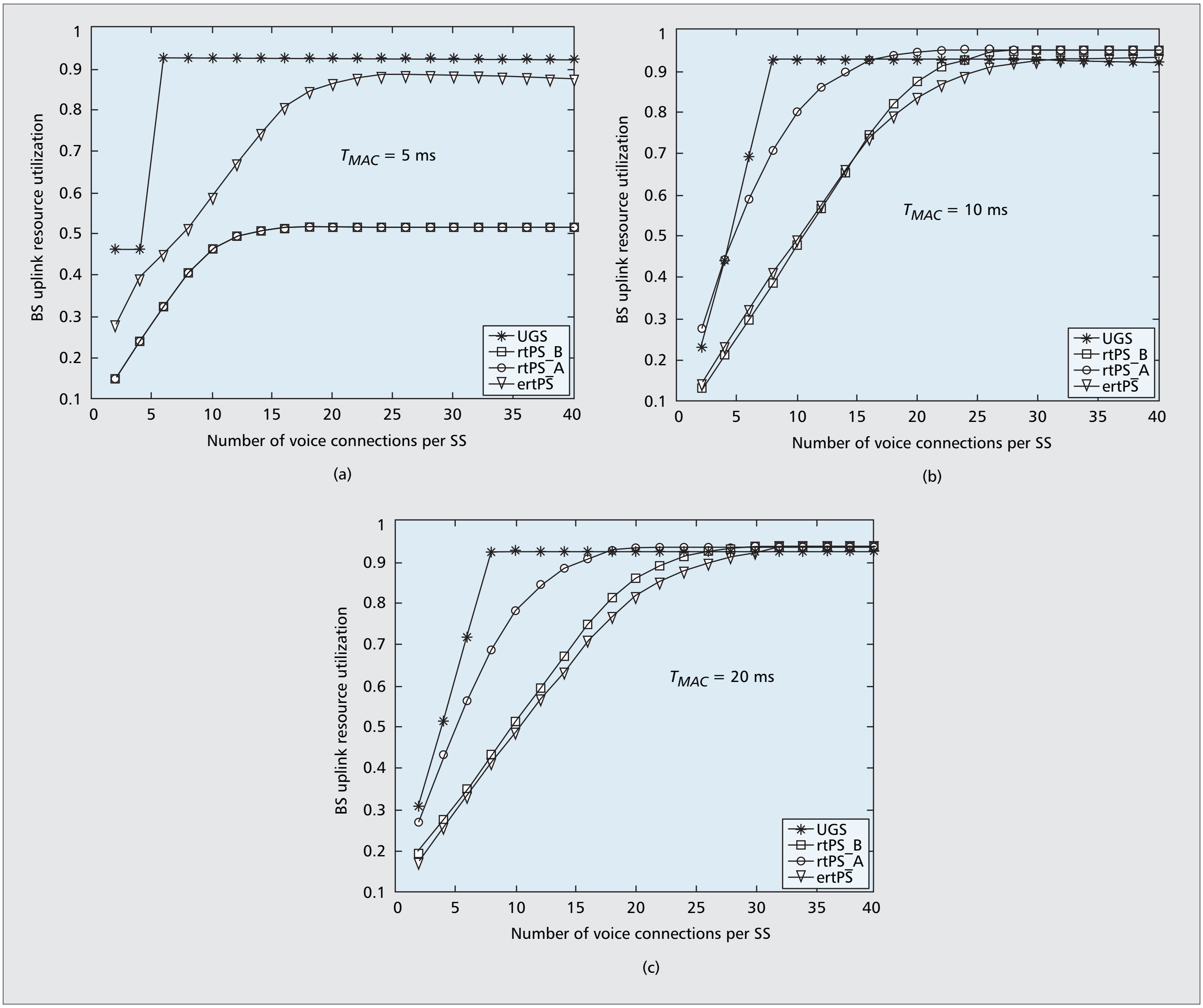
■ **Figure 2.** *Average packet loss rate: a)* T_{MAC} *= 5 ms; b)* T_{MAC} *= 10 ms; c)* T_{MAC} *= 20 ms.*

there is a large number of voice connections, or the MAC frame is long, the bandwidth updating in the ertPS service may not well track the change of active connection numbers, and packet transmission delay is relatively high in these cases. Figure 1 shows that when the traffic load is low and the MAC frame is short, the packet delay performance using the ertPS service is relatively good.

Packet Loss Rate

Packet transmissions experience delay, and packets are dropped if the delay exceeds the maximum tolerable value. Figure 2 shows that when T_{MAC} is 5 ms, the ertPS achieves a slightly higher packet loss probability than the UGS service, and both the rtPS_A and rtPS_B services have a much higher packet loss probability. This observation is consistent with the delay performance shown in Fig. 1a, which indicates that the ertPS service achieves slightly higher packet transmission delay than the UGS, while both the rtPS_B and rtPS_A service result in much higher packet transmission delay than the ertPS service. When T_{MAC} is 20 ms, using the ertPS service results in the highest packet loss rate among all the scheduling services, and the loss rate is high even at a low traffic load, because of the long packet transmission delay as explained earlier. The difference in the packet loss performance among different scheduling services is not quite obvious when T_{MAC} is 10 ms. This is also consistent with the delay performance shown in Fig. 1b, where the packet transmission delay performance using different scheduling services has a slight difference.

Since the latency performance is always guaranteed by dropping packets with delay longer than D_{max}, the packet loss performance can be translated into connection-level performance in connection admission control (CAC). Consider that the CAC makes an acceptance decision for a new connection request if, after its admission, the average packet loss rate is less than a certain threshold, and the CAC blocks a new connection otherwise. Then a better loss rate performance shown in Fig. 2 for a given traffic load is equivalent to a better connection blocking probability for given maximum delay and packet loss rate requirements.



■ **Figure 3.** *BS uplink resource utilization: a)* T_{MAC} *= 5 ms; b)* T_{MAC} *= 10 ms; c)* T_{MAC} *= 20 ms.*

BS Uplink Resource Utilization

The resource utilization shown in Fig. 3 is the percentage of BS uplink resources allocated for voice connections. For the UGS service, it is the percentage of BS uplink resources reserved for the voice connections, while for the rtPS and ertPS services, it is for both bandwidth requests and voice packet transmissions. Although the resource utilization using the UGS service is almost always the highest due to the fact that the SS is allocated either the peak rate or the maximum available bandwidth from the BS, there are exceptions, as shown in Figs. 3b and 3c. That is, when the traffic load is high, the amount of resources reserved for voice connections using the UGS service is comparable to that using other scheduling services, since the amount of the allocated resource is limited by the maximum available amount in the BS. Using the rtPS or ertPS services may result in a slightly higher resource utilization than using the UGS due to the extra resources required by bandwidth requests.

When T_{MAC} is 5 ms, using the rtPS_B requires the least amount of BS resources. Because of the "arrival before bandwidth requests" mechanism used in the rtPS_B, there is little resource overallocated for packet transmissions. The rtPS_A has almost the same resource utilization as the rtPS_B when T_{MAC} = 5 ms, as shown in Fig. 3a. Since the MAC frame duration is short, the number of packets that may potentially arrive between two bandwidth requests is small, and the extra amount of requested resource using the rtPS_A (compared to using the rtPS_B) does not affect the total amount of allocated resources very much. As T_{MAC} is increased to 10 or 20 ms, using the rtPS_A requires much more resources than using the rtPS_B service, and the resource utilization using the rtPS_A is much higher than that using the rtPS_B, as shown in Figs. 3b and 3c.

The ertPS service requires more BS resources than both the rtPS_A and rtPS_B services when T_{MAC} is 5 ms. This is due to the granularity problem, as the BS allocates at least one packet per MAC frame for the SS as long as there is at least one active connection associated with the SS. The granularity effect becomes less obvious when T_{MAC} is 10 or 20 ms, and the amount of required BS resources using the ertPS becomes very close to that using the rtPS_B. It is also possible that the ertPS may require a less amount of the BS resources than the rtPS_B, since the former requires a fewer number of bandwidth requests.

CONCLUSIONS

We have studied voice packet transmission performance in IEEE 802.16-based backhaul networks using different scheduling services. Our results demonstrate that:

- Although the UGS service achieves the best delay performance for voice packet transmissions among all the real-time scheduling services, it has very low efficiency in utilizing the BS resources, even at a light or medium traffic load. Therefore, using the rtPS or ertPS services is preferred as long as the transmission delay is within the maximum tolerable value. With a MAC frame size of 5 or 10 ms, the ertPS service can achieve almost as good packet loss rate performance as the UGS, and the rtPS service achieves almost the same loss rate performance as the UGS for a MAC frame size of 10 or 20 ms.
- The rtPS service provides more flexibility in scheduling voice packet transmissions. The BS can trade-off between the packet transmission performance and the BS resource utilization. The rtPS_B saves the BS resources with longer packet transmission delay and a slightly higher packet loss rate, while the rtPS_A achieves better delay and loss performance at a price of more allocated BS resources.
- Packet transmission performance using the ertPS service strongly depends on the MAC frame size. With a short MAC frame duration, using the ertPS service achieves approximately as good packet transmission performance as using the UGS service. As the frame size increases, the packet transmission performance using the ertPS service degrades.

REFERENCES

[1] H. Lee, T. Kwon, and D. H. Cho, "An Enhanced Uplink Scheduling Algorithm Based on Voice Activity for VoIP Services in IEEE 802.16d/e System," *IEEE Commun. Lett.*, vol.9, no.8, Aug. 2005, pp. 691–93.

[2] G. Gusak, N. Oliver, and K. Sohraby, "Performance Evaluation of the 802.16 Medium Access Control Layer," *ISCIS 2004*, 2004, pp. 228–37.

[3] Q. Liu, X. Wang, and G. B. Giannakis, "Cross-Layer Scheduler Design with QoS Support for Wireless Access Networks," *Proc. 2nd IEEE Int'l. Conf. QoS in Heterogeneous Wired/Wireless Networks)*, Florida, Aug. 22–24, 2005, pp. 21–27.

[4] D. Niyato and E. Hossain, "Queue-Aware Uplink Bandwidth Allocation and Rate Control for Polling Service in IEEE 802.16 Broadband Wireless Networks," to appear, *IEEE Trans. Mobile Comp.*

[5] D. Niyato and E. Hossain, "Joint Bandwidth Allocation and Connections Admission Control for Polling Services in IEEE 802.16 Broadband Wireless Networks," *Proc. IEEE ICC '06*, Istanbul, Turkey, June 11–15, 2006.

[6] D. Niyato and E. Hossain, "A Radio Resource Management Framework for the IEEE 802.16-Based OFDM/TDD Wireless Mesh Networks," *Proc. IEEE ICC '06*, Istanbul, Turkey, June 11–15, 2006.

[7] C. Cicconetti *et al.*, "Quality of Service Support in IEEE 802.16 Networks," *IEEE Network*, vol. 20, no. 2, Mar./Apr. 2006, pp. 50–55.

ADDITIONAL READING

[1] C. Hoymann, "Analysis and Performance Evaluation of the OFDM-Based Metropolitan Area Network IEEE 802.16," *Comp. Networks*, vol. 49, Oct. 2005, pp. 341–63.

[2] D. Anick, D. Mitra and M. M. Sondhi, "Stochastic Theory of A Data-Handling System with Multiple Sources," *Bell Sys. Tech. J.*, vol. 61, no. 8, 1982, pp. 1871–94.

BIOGRAPHIES

XUEMIN (SHERMAN) SHEN [M'97, SM'02] received a B.Sc. (1982) degree from Dalian Maritime University, China, and M.Sc. (1987) and Ph.D. degrees (1990) from Rutgers University, New Brunswick, New Jersey, all in electrical engineering. He is with the Department of Electrical and Computer Engineering, University of Waterloo, Canada, where he is a professor and associate chair for graduate studies. His research focuses on mobility and resource management in interconnected wireless/wireline networks, UWB wireless communications systems, wireless security, and ad hoc and sensor networks. He is a coauthor of two books, and has published more than 200 papers and book chapters in wireless communications and networks, control, and filtering. He serves or has served as Technical Program Committee Chair for IEEE Globecom '07, IEEE WCNC '07 Network Symposium, Qshine '05, IEEE Broadnet '05, WirelessCom '05, IFIP Networking '05, ISPAN '04, IEEE GLOBECOM '03 Symposium on Next Generation Networks and Internet. He also serves as Associate Editor for *IEEE Transactions on Wireless Communications*, *IEEE Transactions on Vehicular Technology*, *ACM Wireless Networks*, *Computer Networks*, and *Wireless Communications and Mobile Computing* (Wiley), and has served as Guest Editor for *IEEE JSAC*, *IEEE Wireless Communications*, and *IEEE Communications Magazine*. He received the Premier's Research Excellence Award (PREA) from the Province of Ontario, Canada, for demonstrated excellence of scientific and academic contributions in 2003, and the Outstanding Performance Award from the University of Waterloo for outstanding contribution in teaching, scholarship, and service in 2002. He is a registered Professional Engineer of Ontario, Canada.

DONGMEI ZHAO [M] (dzhao@mail.ece.mcmaster.ca) received a Ph.D. degree in electrical and computer engineering from the University of Waterloo in June 2002. Since July 2002 she has been with the Department of Electrical and Computer Engineering, McMaster University, Hamilton, Ontario, Canada, where she is an assistant professor. Her research interests include modeling and performance analysis, QoS provisioning, access control, and admission control in wireless networks. She is a Professional Engineer of Ontario, Canada.

The rtPS_B saves the BS resources with longer packet transmission delay and a slightly higher packet loss rate, while the rtPS_A achieves better delay and loss performance at a price of more allocated BS resources.