TECHNICAL RESEARCH REPORT

Medium Access Protocols for Interconnecting ATM and Wireless Networks

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MEDIUM ACCESS PROTOCOLS FOR INTERCONNECTING ATM AND WIRELESS NETWORKS *

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ABSTRACT

The current trend in modern telecommunication is that most of the traffic (CBR, VBR and ABR, real-time and delay-insensitive) in the fixed broadband networks will be carried by fixed-length packets. It is desirable to extend the services to wireless networks as well. However, one major difficulty in wireless networks is the limited spectrum available. Another difficulty is that the bit error rate (BER) of the wireless channel is typically much higher than that of the channels of the wireline broadband networks. The latter problem is amplified for military wireless networks that experience substantial link interference (e.g., multipath fading, other-user interference, jamming). Therefore, we can only expect that the same services are being carried in a similar fashion but with a lower source rate. In this paper, a medium access protocol for interconnecting ATM and wireless networks is discussed, followed by some alternatives, and other issues pertaining to guaranteeing low BER. This research was conducted in support of Task 1.1 of the ATIRP project during FY 96 and continues under Task 4.3 for FY 97.

NCPRMA/DQRUMA: A MEDIUM ACCESS PROTOCOL FOR WIRELESS ATM NETWORKS

A medium access protocol for wireless ATM networks using time division multiple access has been developed. Voice and data users are supported. By voice users, we also include services which require real-time delivery. For data users, we imply that the service is delay-insensitive. In order to utilize the resource efficiently, information slots are allocated to users that are in talkspurt only. When a user changes its state from silent to talkspurt, it has to request for an information slot. This demand assignment approach increases the number of users that can be supported by allowing a small packet dropping probability.

Using this protocol, information can be distributed between mobile users and a control center (base station). This is useful in the Army digital battlefield. The information sources include voice where small packet dropping is acceptable, and data (such as computer files containing maps, documents, etc). In the latter case, packet dropping is undesirable. However, it can be overcome by requesting for re-transmissions through the network layer of the protocol if necessary. In addition, for services where real-time and zero packet dropping are required, dedicated resources can be assigned (i.e. without going through the demand assignment process). The objective of the proposed medium access protocol is to provide a throughput that is better than the traditional circuit switching approach by allowing a small but acceptable packet dropping probability. Detailed descriptions of the protocol are given below.

The medium access scheme for voice users is based on non-collision packet reservation multiple access (NCPRMA), which itself is a modification of PRMA ([3], [1]). Each voice user has its own control slot which is used to send requests to the base station. As a result, the base station will receive the requests

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promptly and the delay requirements of voice users are met with high probability. On the other hand, the scheme for data users is based on distributed-queueing request update multiple access (DQRUMA) [4]. Data users send their requests for information slots through the data random access slots which are shared with other data users. The base station will receive the requests only if there are no collisions of packets during the random accessing process.

The uplink TDMA frame (Figure 1) is made up of voice control slots, data random access slots, and information slots. The downlink TDMA frame is made up of a status slot, polling slots, attention slots, and information slots. The functions of the voice control slots and data random access slots have already been described above. The functions of the polling and attention slots are to notify the users the outcomes of their requests and to inform them to tune to downlink information slots for packets, respectively. The function of the status slot is to inform the users about the number of control slots in each frame and some other system information. The number of control slots determines the maximum number of voice users the system can support. The information slots are used to carry the user packets. A piggybacking bit is present in the information slot which is used to reserve information slots for the next frame. This reduces the amount of contention in the data random access slots. In scheduling the information slots, priority is given to voice users over data users. In addition, we can limit the number of voice and data users that are assigned information slots. As a result, we can control the dropping probabilities of the users. Refer to Tables 2 to 5 for procedures of sending packets between the voice/data users and the base station.

Another issue we need to resolve is how handoff can be implemented. The proposed protocol is compatible with the user-executed handoff scheme discussed in [5] by allowing a user that has just moved into a new cell to randomly transmit in one of the unused control or random access slots, depending on whether it is a voice or data user. The base station will notify the user if its transmission is received successfully. As a result, the workload involved in handoff is distributed over the network.

NUMERICAL RESULTS

Computer simulations have been performed to compare the performance of the proposed protocol with two other protocols, the PRMA [1] and Reservation TDMA [2] protocols. The parameters used are given in Table 1. The results are plotted in Figures 2 to 4.

Frame period	25 ms
Channel rate	394.24 kbps
Size of an infomation slot	896 bits
Num. of information slots per frame	10
Size of an overhead slot	64 bits
Num. of overhead slots per frame	14
Average talkspurt duration	1 s
Average silence duration	1.35 s
Buffer size of voice users	1 packet
Buffer size of data users	10 packets

Table 1: Parameters used in the simulation.

The observation is that for the same number of voice and data users in the system, the average voice and data dropping probabilities are smaller in the proposed protocol. An advantage of our protocol is that an idle user is required to listen to the channel in the "overhead" period only, during which time polling and attention slots are being sent from the base station. This saves the battery life of the mobile. However, the access delay of this protocol is slightly higher than PRMA because the users have to wait until the beginning of the next frame before sending their requests. Another advantage of our protocol over the other two is about optimizing the system throughput. In our case, we first optimize the performance of the voice portion of the system, followed by the data portion. However for the other two schemes, both voice and data users have to be considered together. This makes the optimization more difficult.

WIRELESS ATM NETWORKS & CDMA

There are alternatives to the medium access protocol described above. One example is the use of CDMA. A natural way is to assign different spreading codes to different users and allow the users to transmit their packets as soon as they arrive. However, the system throughput will be poor because of the lack of control. (Incidentally, this phenomena is similar to what happens in PRMA.) Demand assignment type of schemes is also possible for CDMA system. Specifically, dedicated PN codes can be used as voice control channels and data random access channels, and thus the same protocol as in TDMA can

Direction	Slot used	Functions performed
M→B	Voice Control Slot	Request for a certain number of uplink information slots.
B→M	Voice Polling Slot	Reply with the mobile's ID, the number and location of the information slots allocated. Or, if the request is not accepted, do not send any reply.
M→B	Voice Control Slot, Information Slot	If polling is received, acknowledge through the control slot, and send the packets in the allocated infomation slot and piggyback with the number of information slots requested for the next frame. Otherwise, retransmit the request in the next frame.
M→B	Information Slot	Keep sending the packets in the allocated infomation slots and piggybacking. When the talkspurt ends, inform the other end through piggybacking.

Table 2: Sending Voice Packets from Mobile.

Direction	Channel used	Functions performed
B→M	Voice Attention Slot	Inform the mobile to listen to the specified downlink information slots. The mobile's ID, the number and location of the information slots to be used are sent.
M→B	Voice Control Slot	Send an acknowlegement and start receiving the packets.
B→M	Information Slot	Send the packets and piggyback with the number of information slots to be used for the next frame. If an acknowledgement is not received, retransmit the attention.
B→M	Information Slot	Keep sending the packets in the designated infomation slot and piggybacking. When the talkspurt ends, inform the other end through piggybacking.

Table 3: Sending Voice Packets from Base.

be used. Multi-code CDMA can be used to transmit multi-rate user packets. The spreading codes for different packet streams of the same user are orthogonal and thus the interference within the transmissions of the same user is very small. An advantage of a CDMA protocol over a TDMA one is that when the number of active users is smaller than the system capacity, the CDMA is actually performing better because the interference is smaller. In addition, a better throughput is expected using CDMA because of the capture phenomenon. However, there are also issues about supporting high data rate users and performing power control in CDMA.

ADAPTIVE HYBRID FEC/ARQ

In the simulations above, we assume that the channel is noiseless. It is important that some form of error correction is performed in order to provide the suggested throughput of the system. Various FEC/ARQ schemes have been proposed in the literature. One such scheme is the adaptive hybrid FEC/ARQ scheme. For real-time users, high-rate FEC can be used when the channel is in the good state while delay-bounded ARQ combined with low-rate FEC can be used when the channel is in the bad state. For delay-insensitive users, the users may be allowed to send more re-transmissions when the channel is bad. Powerful codes such as concatenated Reed-Solomon and turbo codes can be used to provide the required bit error rate. We are currently investigating the performance of one such scheme to provide different qualities of service (in terms of bit error rate and delay) to different user types.

Preliminary results based on analysis are shown in Figure 5. The scheme used is based on a type-II hybrid FEC/ARQ scheme. The encoder can be in one of two states depending on the channel condition. User packets are transmitted uncoded in state 1. If the

Direction	Slot used	Functions performed
M→B	Data Random Access Slot	Request for a certain number of uplink information slots.
B→M	Data Polling Slot	If the mobile's request is received successfully, send the mobile's ID, the number and the location of the information slots allocated.
M→B	—	If polling is received, send the packets . Otherwise, retransmit the request after a random number of frames.

Table 4: Sending Data Packets from Mobile.

Direction	Channel used	Functions performed
B→M	Data Attention Slot	Inform the mobile to listen to the specified downlink information slots. The mobile's ID, and the number and location of the information slots to be used are sent.
B→M	Information Slot	Send the packets.

Table 5: Sending Data Packets from Base.

transmission is unsuccessful, the parity bits of the previous packet will be sent, using a rate 1/2 turbo code. The parity bits are combined with the previously transmitted packet (information bits) and decoded. If this also results in a decoding failure, the encoder will switch to state 2 and all packets are sent using a rate 1/2 turbo code. The initial state at which the system starts at is determined by the channel condition. The graph shows the throughput of this adaptive hybrid scheme as a function of SNR. It can be observed that by adapting to the channel condition, the hybrid scheme achieves good throughput in both low and high SNR.

CONCLUSIONS

In this paper, a new medium access protocol based on TDMA is proposed. Computer simulations show that the protocol gives a better performance than the PRMA and RTDMA protocols. A CDMA version of the protocol is also briefly described. Preliminary results of an adaptive hybrid FEC/ARQ scheme using turbo code are given.

The views and conclusions contained in this document are those of the authors and should not be interpreted as representing the official policies, either expressed or implied, of the Army Research Laboratory or the U.S. Government.

REFERENCES

[1] D. J. Goodman, R. A. Valenzuela, K. T. Gayliard, and B. Ramamurthi, "Packet reservation multiple access for local wireless communications," *IEEE Trans. Commun.*, vol. COM-37, no. 8, pp. 885-890, Aug. 1989.

[2] D. Raychaudhuri, and N. D. Wilson, "ATM-Based transport architecture for multiservices wireless personal communication networks," *IEEE J. Select. Areas Commun.*, vol. 12, no. 8, pp. 1401-1414, Oct. 1994.

[3] J.-H. Wen, and J.-W. Wang, "A new protocol for wireless voice communications – non-collision packet reservation multiple access," in *Proc. PIMRC'95*, pp. 638-642.

[4] M. J. Karol, Z. Liu, and K. Y. Eng, "Distributed-queueing request update multiple access (DQRUMA) for wireless packet (ATM) networks," in *Proc. ICC'95*, pp. 1224-1231.

[5] A. S. Acampora, and M. Naghshineh, "An architecture and methodology for mobile-executed handoff in cellular ATM networks," *IEEE J. Select. Areas Commun.*, vol. 12, no. 8, pp. 1365-1374, Oct. 1994.

[6] C. Berrou, A. Glavieux, and P. Thitimajshima, "Near Shannon limit error-correcting coding and decoding: turbo-codes ," in *Proc. ICC'93*, pp. 1064-1070.

[7] S. B. Wicker, *Error Control Systems for Digital Communication and Storage*, Prentice Hall, 1995.

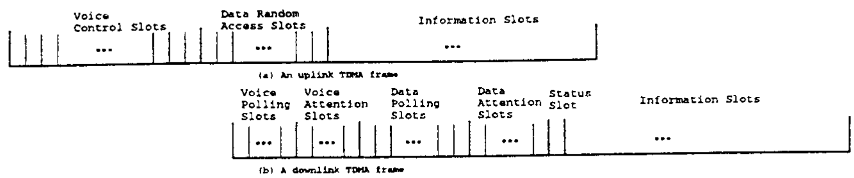

Figure 1: Structure of the TDMA frames.

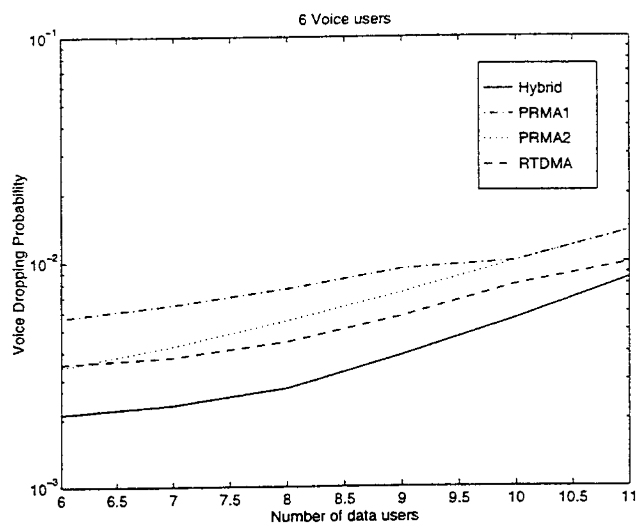

Figure 2: Voice Dropping Probability (6 voice users).

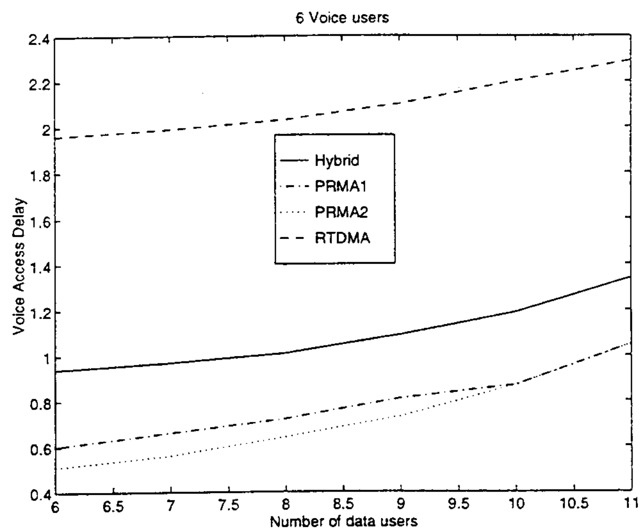

Figure 4: Voice Access Delay (6 voice users).

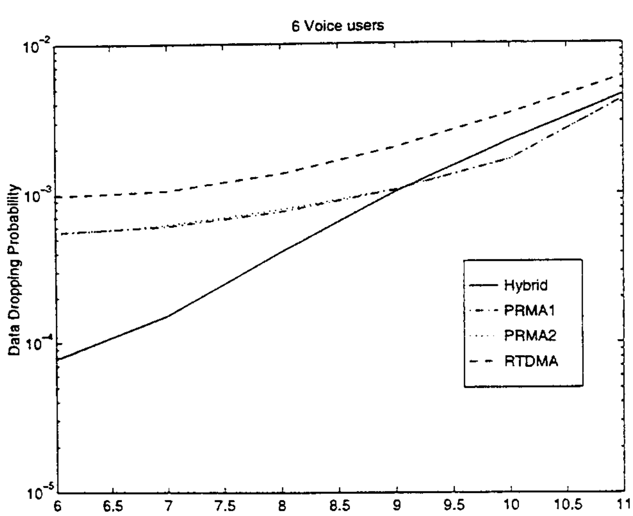

Figure 3: Data Dropping Probability (6 voice users).

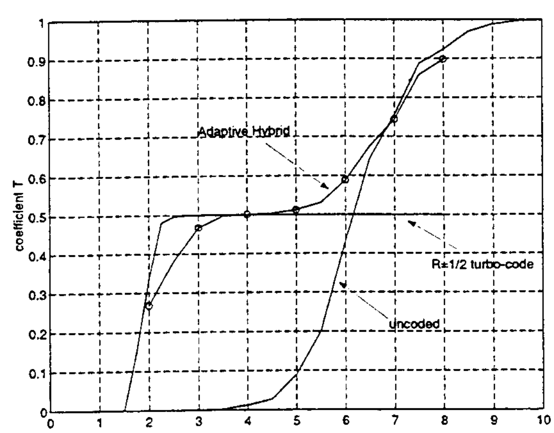

Figure 5: Throughput of the Adaptive Hybrid Scheme vs. SNR.