A New AAL2 Scheduling Algorithm for Mobile Voice and Data Services over ATM

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Abstract

AAL2 has been adopted for bandwidth-efficient transmission of low bit rate traffic over ATM networks in ITU-T and ATM Forum. Since ATM/AAL2 is expected to be used as a switching technology in third-generation mobile access networks and mobile data traffic is expected to increase rapidly in near future, there must be a need for efficient scheduling scheme satisfying the QoS requirement of low bit rate voice as well as the one of high bit rate data. In this paper, we propose a new class-scheduling scheme to improve data packet loss probability, while QoS of voice traffic is guaranteed, when data traffic is multiplexed together with mobile voice traffic into a single ATM VCC. The proposed scheme can efficiently support data traffic by assigning a time threshold value to voice traffic. Through simulation study, we show that the proposed scheme does not only achieve better efficiency for providing both mobile voice and data services than HOL class-scheduling scheme and normal FIFO scheme, but also guarantees mean voice packet delay under a certain criteria.

I. INTRODUCTION

Asynchronous Transfer Mode (ATM) based on fast packet switching principle with a fixed size 53-octet is a key technology in the construction of a high speed multi-media backbone network. ATM supports various kinds of applications with diversified quality service requirements including voice and video conferencing. Since ATM has been adopted as the standard for the future high-speed network, the interconnection between wireless systems and wired ATM networks has become an important issue.

For providing multimedia services through interconnection of wireless/wired networks, we have to consider a requirement for a fast and efficient transport of cellular voice and data between the base station (BS) and the mobile switching center (MSC). One possible solution that has been proposed is to use ATM links between BS and MSC. For the typical wireless applications, the data rate is relatively low and the packet size is much smaller than those of the normal applications of the wired networks. If this small sized packet is carried by one ATM cell which can transport 48 bytes of payload, significant amount of link bandwidth would be wasted and the multiplexing delay to make one ATM cell would be increased.

Since the existing AALs such as AAL-1, AAL-3/4 and AAL-5 provide neither bandwidth efficiency nor low multiplexing delay characteristic, AAL type-2 for low bit rate traffic transmission over ATM has been developed to be standardized at the ITU-T. AAL2 allows variable sized packets from multiple channels to be multiplexed into an ATM VCC thus improving the network bandwidth utilization and reducing the multiplexing delay [1].

Mobile telephony is the dominant application in the wireless networks today and is expected to remain so for many years. At the same time, wireless data is expected to increase rapidly due to the success of Internet and recent advances in wireless technology. In order to meet the expected demand for the number of mobile users, as well as Internet applications in wireless networks, the third generation wireless networks standards are being developed by various telecommunicaton providers [2].

ATM/AAL2 is expected to be applied as a switching technology in the third-generation wireless access networks [3]. When both mobile voice and data are multiplexed into a single ATM VCC, an efficient scheduling method is required to provide QoS related to each traffic characteristic at AAL2 level. Generally, for QoS of voice and data service, voice is sensitive to delay time and data to loss. Therefore, voice

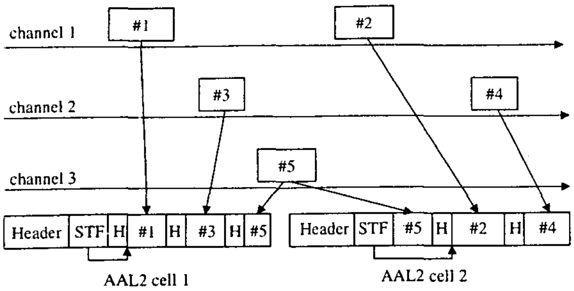

Fig. 1. AAL2 Cell Packing.

has higher priority than data traffic in viewpoint of their waiting times in scheduler.

In this paper, we propose a new class-scheduling scheme to improve mobile data packet loss probability and also to guarantee mobile voice packet delay under a certain criteria using a time threshold value when both mobile voice traffic and data traffic are multiplexed into a single ATM VCC.

The rest of this paper is organized as follows. The concept of AAL2 is described in section II. The section III describes the proposed class-scheduling scheme. In section IV, we evaluate simulation results of the proposed scheme compared with those of normal FIFO scheme and the HOL class-scheduling scheme [4]. Finally we conclude this work in section V.

II. THE CONCEPT OF AAL2

In AAL2 voice or data traffic are generated into a short packet, and these short packets(CPS packets) are accumulated into a standard ATM cell. The CPS-Packet header consists of a Channel Identifier (CID), Length Indicator (LI), User-to-User indication (UUI), and Header Error Control (HEC). The length indicator (LI) represents the voice or data length, which can be set to arbitrary values smaller than 45 or 64.

AAL2 can distinguish different channels using an 8-bit field assigned to CID, which is available to 255 users. In order to maximize the packing density, AAL2 has a multiplexing function that divides the last packet of cells into two appropriate parts. To avoid the problem of incorrect selection of cells during reception, AAL2 has an offset field (OSF) with the 1-byte start field on the head of each cell payload to indicate the next packet's starting point. Finally, packets divided into two are combined correctly by comparing the OSF with the difference between the LI and the length of the first part. In addition to that, the STF has a sequence number (SN) to detect cell loss and a parity field for detecting bit errors in the STF [1].

Consequently, the basic idea of AAL2 is to multiplex voice or data packets from different channels into one ATM VCC by using the above functions so that the time to fill a cell can be reduced significantly. Figure 1 illustrates a scenario of voice or data packets from three channels being packed into cells [6].

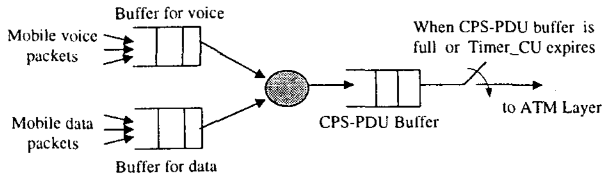

Fig. 2. The Structure of the Proposed Scheduling Scheme.

III. THE PROPOSED SCHEME DESCRIPTION

When the network operator wishes to minimize the bandwidth of the leased ATM VCCs, all AAL2 traffic such as mobile voice and data must be multiplexed into an ATM VCC. Once the AAL2 traffic is multiplexed into an ATM VCC, the ATM layer is unable to differentiate a certain AAL2 service class from the others. Therefore, it is very important to differentiate service classes at AAL2 level so that each mobile traffic characteristic and required QoS can easily be mapped to the AAL2 level.

The proposed scheme has two service classes at the AAL2 level as shown in Figure 2, in which one is for mobile voice traffic and the other is for mobile data traffic. Using a time threshold value for the class of mobile voice traffic, the proposed scheme can provide service efficiently guaranteeing the delay performance of voice traffic under a proper criteria and the loss performance of data traffic. For describing the proposed class-scheduling scheme, we select the AAL type 2 system parameters as shown below.

- Timer_CU value is 20msec.
- Maximum length of a CPS-SDU is 45 octets.

In the proposed class-scheduling scheme, voice packets and data packets are stored in voice buffer and data buffer, respectively. The voice buffer memorizes the time of inserting into buffer every voice packet from different channels, and manages the timer representing how long each voice packet is waited at the buffer. If that time value of a voice packet is larger than the time threshold value TH at a certain time slot, then that voice packet is transmitted into CPS-PDU buffer. And if that time value of a voice packet is not over, then data packet is instead transmitted into CPS-PDU buffer. When no voice packet in the voice buffer, the scheduler transmits the data packet of the buffer into CPS-PDU

Transmission rate	Packet size	Probability
$8Kbps$	22octets	0.291095
$4Kbps$	10octets	0.03882
$2Kbps$	5octets	0.072328
$800bps$	2octets	0.597757

Table I. Voice Packet Size Distribution with 8Kbps Vocoder for CDMA System

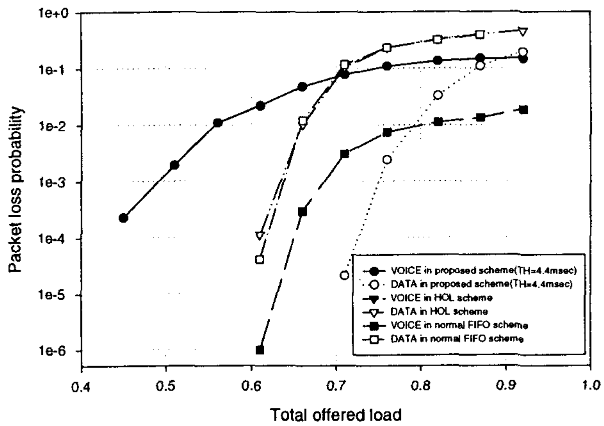

Fig. 3. Packet loss probability.

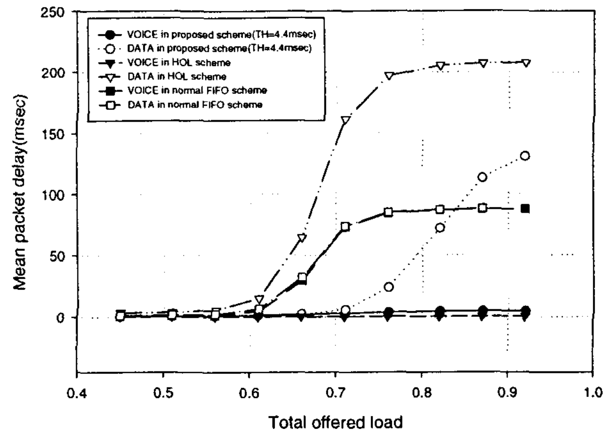

Fig. 4. Mean packet delay.

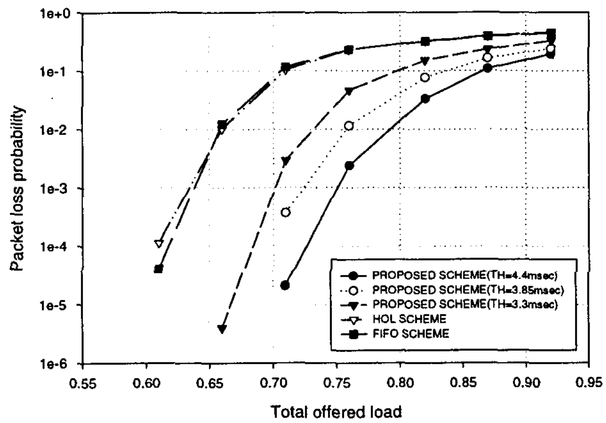

Fig. 5. The comparison of data packet loss probability of proposed scheme with different threshold values and the ones of the other two schemes.

buffer, and vice versa. The scheduler repeats the procedure every time slot until CPS-PDU buffer is filled or Timer_CU value is expired.

For the proposed scheme, the selection of the time threshold value TH is important. The time threshold value TH is determined considering the upper bound of queuing delay requirement for mobile voice service. For example, when the queuing delay time of mobile voice packet in AAL2 system is necessary to be bounded to 10msec, the time threshold value TH should also become 10msec to improve at best QoS of mobile data service, while guaranteeing QoS of mobile voice service. In AAL2 system transmission delay consists of queuing delay and multiplexing delay. Since the time threshold value TH means guaranteeing queuing delay of voice traffic at a certain criteria less than it, the proper selection of the time threshold value TH can provide efficient services for both voice and data traffic satisfying each QoS requirement. The effect of the time threshold value TH was simulated and its results will be shown in Section IV.

IV. PERFORMANCE EVALUATION

In this section, we evaluate the performance of the proposed class-scheduling scheme in terms of packet loss probability and mean packet delay through computer simulation, and compare it with HOL class-scheduling scheme and normal FIFO scheme.

In the case of HOL class-scheduling scheme, we consider that voice traffic always has higher priority than data traffic.

A. Simulation Model.

We consider a Base Station where the link between the BS and MSC is T1 (1.544Mbps). Figure 2 shows the process that voice packets and data packets from multiple mobiles are arrived in each buffer, scheduled for guaranteeing each QoS at the AAL2 level, packed into CPS-PDUs which are sent to ATM layer. The above procedure is simulated to study the performance of the proposed class-scheduling scheme.

In this simulation, the mobile voice traffic source is assumed as the CDMA system equipped with an 8Kbps vocoder that changes its transmission rate among 4 different levels based on input traffic's energy including silence period, as investigated in [5]. Accordingly, it periodically generates variable sized packets every 20msec and the packet size distribution used for this simulation is shown in Table 1. For the mobile data traffic source, we use ON-OFF source model where the duration of ON and OFF periods is exponentially distributed with mean 137msec and 411msec, respectively. Since the data rate of the mobile data user is assumed to be 64kbps, it generates 160byte packet every 20msec during the ON state and keeps silence during the OFF period. We simulate three schemes as increasing the number of mobile data users from 5 to 50, when the number of mobile voice users is 200 (the offered load from the mobile voice users is 200*3.1k/1544k=0.40). The sizes of voice buffer and data buffer are, as an example, 300 and 12000 bytes, respectively.

B. Simulation Results.

Figure 3 shows simulation results of packet loss probability for three schemes. In this figure, we can find out that the proposed scheme gives better performance than the other two schemes for data packet loss probability, though the loss probability of voice packet rather high. Note that voice traffic is insensitive to QoS of the loss probability. Since voice traffic is not multiplexed into the CPS-PDU buffer before the time threshold TH except for the case that only voice buffer has packets, the proposed scheme can efficiently serve data traffic. We can observe the fact that HOL class-scheduling scheme can guarantee only voice service. This fact results in bad performance for the data. For normal FIFO scheme, the loss probability of data packet is too larger than that of voice packet. Therefore, normal FIFO scheme is not appropriate scheme for multiplexing both voice and data traffics into a single ATM VCC.

Figure 4 shows simulation results of mean packet delay for three schemes. Transmission delay of AAL2 system consists of queuing delay and multiplexing delay. In that figure, HOL class-scheduling scheme has the best performance. However, we can find out that mean voice delay of the proposed scheme is maintained under a certain criteria. For the proposed scheme, mobile voice packets are multiplexed into a cell under the time threshold value TH and transmitted into ATM layer. Thus queuing delay of mobile voice traffic in AAL2 system can be guaranteed at a certain criteria less than the time threshold value TH in our scheme.

We can understand the effect of the time threshold value on data packet loss probability from the result of Figure 5. As the threshold value increases, data packet loss probability decreases. Although increase of the time threshold value TH means decrease of data packet loss probability, the time threshold value TH should be determined in the range of guaranteeing the queuing delay of voice traffic in AAL2 system. To improve at best QoS of mobile data service guaranteeing QoS of mobile voice service, the time threshold value TH can be determined as the upper bound of delay requirement for mobile voice service as mentioned in section III. Therefore, the selection of the accurate upper bound of queuing delay requirement for mobile voice service is remained to determine the time threshold value TH as a further study.

V. CONCLUSIONS

In this paper, we proposed a scheduling scheme to provide AAL2 service efficiently when mobile voice traffic and mobile data traffic are multiplexed into a single ATM VCC. The proposed class-scheduling scheme has adopted the time threshold value for mobile voice traffic, which could contribute to serve mobile data traffic efficiently guaranteeing mobile voice traffic's QoS. From simulation results, we could find out that the proposed class-scheduling scheme can provide the better performance in terms of data packet loss probability than both HOL class-scheduling scheme and normal FIFO scheme. Also, we verified through simulation that the proposed class-scheduling scheme could maintain mean voice packet delay under a certain criteria. Therefore, we can conclude that the proposed class-scheduling scheme can simultaneously satisfy each QoS of mobile data traffic and mobile voice traffic at the AAL2 level.

REFERENCES

[1] ITU-T Recommendation, "B-ISDN ATM Adaptation Layer Specification," *I.363.2*, Sep 1997.

[2] B. Subbiah, S. Dixit, "ATM Adaptation Layer 2 (AAL2) for Low Bit Rate Speech and Data: Issues and Challenges," *Proc. IEEE ATM Workshop'98*, pp. 225-233, 1998.

[3] G. Eneroth, G. Fodor, G. Leijonhufvud, A. Racz, I. Szabo, "Applying ATM/AAL2 as a Swiching Technology in Third-Generation Mobile Access Networks," *IEEE Communications Magazine*, Vol. 376, pp. 112-123, June 1999.

[4] V. Casares-Giner, J. Paradells-Aspas, "An Integrated VOICE/DATA Traffic Model With Variable Voice Service Rate," *ICC'89*, Vol. 2, pp. 971-976, 1989.

[5] Z. A. Cui, "Analysis on variable cell size and short cell size ATM," *ATM Forum/96-0056*, Feb. 1996.

[6] C. Liu, S. Munir, R. Jain, "Packing Density of Voice Trunking using AAL2," *IEEE Globecom'99*, Volb. 1, pp. 611-615, Dec. 1999.