Multimedia Transmission based on PRMA Wireless Networks

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ABTRACT

This paper proposes a new scheme for the integration of voice and video transmission over the Packet Reservation Multiple Access (PRMA) system that is a modification of Reservation-ALOHA protocol. We focus on low bit-rate video applications like video conferencing and visual telephony for wireless communications. The ITU-T H.263 standard provides a solution to the need for low bit-rate video compression under 64Kbps. The proposed protocol assumes that each voice terminal follows a traffic pattern of talk spurts and silent gaps with fixed permission probability (p = 0.3), and each video terminal has the higher permission probability (p = 1) to access the available slot based on ITU-T H.263 standard. Again, we present a "pseudo reservation" scheme to release slots reserved by video terminals according to the contents of each video transmission buffer, and the active voice terminals can temporarily access the additional slots to improve the performance without sacrificing the video capacity of the system. The packet dropping probability of the active voice terminals and bandwidth utilization of the system are superior to the original PRMA, as indicated in simulation results.

Keyword- Permission Probability, Pseudo Reservation.

I. INTRODUCTION

Packet reservation multiple access (PRMA) is a modification of Reservation-ALOHA protocol for the statistical multiplexer of speech packets transmission among wireless terminals and the base station. Goodman et al. proposed the protocol for packet voice terminals in cellular systems [1]. The PRMA channel is slotted, and the slots are grouped together into frames. An active wireless terminal is required to monitor each slot as either "reserved" or "available" according to the feedback message received from the base station at the end of the slot. But unlike Time Division Multiple Access (TDMA), where the wireless terminals are granted access to a time slot for the entire duration of a call, PRMA voice terminals are granted access to the time slot only during talk spurts and release the time slot during silence periods. Again, the terminals must contend for access an available slot with the permission probability, p, when they move from silence periods to talk spurts. If more than one terminal contend the available slot simultaneously, a collision occurs. We ignore the capture effect of colliding packets with different signal levels [2]. The base station fails to detect each packet, and all colliding packets require retransmission as waiting for the random time delay.

Several articles studied the performance of voice-only or joint voice-data PRMA system [3]-[10]. In this paper, we present a new protocol for the integration of voice and video transmission over PRMA. We assume each voice terminal follows a traffic pattern of talk spurts and silent gaps with fixed permission probability (p = 0.3), which has been shown to perform well [3]. Furthermore, it is assumed that each video terminal has the highest permission probability (p = 1) to access the available slot based on ITU-T H.263 standard [11][12] which provides a solution to the need for low bit-rate video compression. And we propose a "pseudo reservation" scheme to release reserved slots of the video terminals of the contents of each transmission buffer been under the predefined threshold. The voice packet dropping probability and system bandwidth utilization performance are superior to the original PRMA without sacrificing the video capacity of the system, as indicated in the results simulated by BONeS[13].

The rest of the paper is divided as follows. Section II describes the H.263 video compression standard. Section III shows the integration of voice and video transmission protocol over PRMA. Section IV displays the performance results and discussions in comparison with original PRMA. Conclusions are presented in Section V.

II. DESCRIPTION OF H.263

The proposed protocol of video transmission is based on the H.263 video compression standard that provides for the coding rate below 64Kbps. At the beginning of the H.263 draft presented in International Telecommunications Union — Telecommunications Standards Sector (ITU-T), there were no clear consensus among the experts about the scope and objectives. But H.263, derived from the earlier version, H.261, is in principle network-independent and can be used for large range of applications. Its target applications are video conferencing and visual telephony, and target networks are low bit-rate networks like the General Switched Telephone Network (GSTN), Integrated Services Digital Network (ISDN), and wireless networks.

The ITU-T H.263 Recommendation specifies a coded representation that can be used for compressing the moving picture component of video conferencing and visual telephony services at low bit-rate. The basic configuration of video coding algorithm is based on ITU-T H.261 that is a hybrid of inter-picture prediction to utilize temporal redundancy and transform coding of the remaining signal to reduce spatial redundancy. Like MPEG and H.261, H.263 also uses motion compensation and the Discrete Cosine Transform (DCT) for video compression.

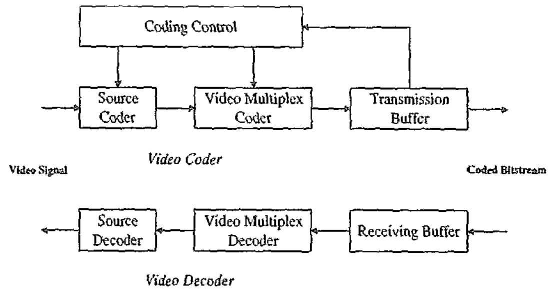

Figure 1. The block diagram of H.263 video codec

The block diagram of the H.263 codec is shown in Figure 1, where the video input and output signals may be composite or component, analog or digital and the methods of performing any necessary conversion to and from the source coding format are not subject to recommendation. Again, the video encoder provides a self-contained digital bit stream, and the encoding algorithm can be decomposed into four major steps: motion estimation, transform coding, linear quantization, and run length coding. However, the video decoder performs the reverse process.

In the proposed joint voice-video PRMA scheme, we need to determine the threshold value (according to the packet size of one slot) of the transmission buffer of each video terminal shown in Figure 1, and to decide when the video terminals release and recover the reservation slots. The next section will present the scheme in detail.

III. A NEW SCHEME FOR THE INTEGRATION OF VOICE AND VIDEO TRANSMISSION

In this section, we focus on the integration of voice and video service in a PRMA system. Figure 2 shows the frame structure of the proposed PRMA protocol like TDMA/TDD (Time Division Duplexing) for up-link and down-link data transmission. In general, each voice terminal follows a traffic pattern of talk spurts and silent gaps for each slot with the data rate of 32Kbps (512 bit / 16 ms). While each video terminal accesses two available slots and the maximum data rate is 64Kbps (1024 bits / 16 ms) based on H.263 video compression for the entire duration of the transmission. But for the VBR (Variable Bit Rate) traffic like the video conferencing and visual telephony services, the data-rate is always under 32Kbps. Thus, how to allocate the empty slots of each video terminal for the other voice terminals using to improve the bandwidth usage of the system is an important issue in our study.

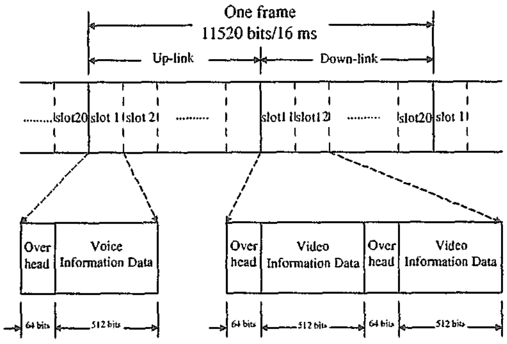

Figure 2. The frame structure of the joint voice-video PRMA system

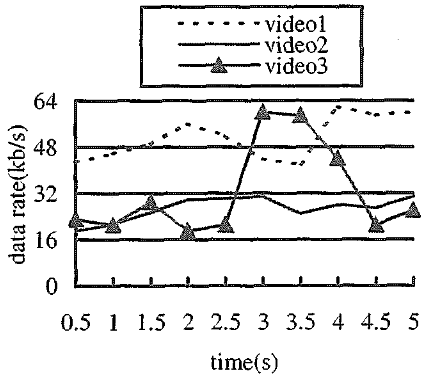

Figure 3. Traffic models of the different video streams

For instance, Figure 3 illustrates the traffic models of different video streams encoded by the H.263 compression scheme [11], which is a hybrid algorithm of inter-picture prediction to utilize temporal redundancy and transform coding of the remaining signal to reduce spatial redundancy. As we can see, the data rate of video 2 is almost under 32Kbps like the video telephony service, which just needs to occupy one slot during the transmission time. But the video terminal must assign two time slots to satisfy the maximum data rate (64Kbps) based on the PRMA system. In order to improve the usage of system bandwidth, we propose a new scheme for the integration of voice and video transmission using the PRMA protocol.

The new scheme is described below in detail.

A. *The New Scheme — Pseudo Reservation Scheme*

If each video terminal is assigned a lot of available slots according to the maximum bit-rate initially, it dose not occupy all of the slots for the entire duration of the transmission. Thus, we propose a new scheme for the joint voice-video PRMA protocol to improve the bandwidth usage of the system. At first, the system needs to determine the threshold value of each video transmission buffer according to the packet size of one slot. For example, we assign 512 bits to be the threshold value because each slot contains 512 bits for the information field in our simulation system shown in Figure 2. And the base station decides when to release and recover the reservation slots of the video terminals according to the threshold value of each transmission buffer. Furthermore, these released slots will only allow the other active voice terminals to contend, and their permission probabilities are given by the highest value ($p = 1$) for increasing the chance to access empty slots. If any active voice terminal accesses the slots successfully, it can not make normal reservation but only "pseudo reservation" instead. The reason is that once the transmission buffer of the original video terminal goes over the threshold value (i.e. 512 bits in our system), the video terminal will recover the released slots without any exception, and the voice terminal will again contend the other available slots in the next frame. Thus, we call the above mentioned scheme as Pseudo Reservation scheme.

Four states exist for the active voice terminals in Pseudo Reservation scheme as follow.

1. When any voice terminal moves from silence periods to talk spurts, it contends the available slots to transmit packets at first, where the permission probability, p, is 0.3.
2. If there are no empty slots in the current frame, but only have the released slots by the video terminals, the voice terminal assigns the slots to make the pseudo reservation with $p = 1$.
3. Once the transmission buffer of the original video

terminal goes over the threshold value, the video terminal will recover the released slots without any exception, and the voice terminal will end the pseudo reservation to contend the other available slots in the next frame.

4. If neither empty slots nor released slots are available, some packets of the voice terminals will be dropped due to the delay limit (approximately 32 ms in general).

According to the above statuses or procedures to decide when to start and stop the pseudo reservation scheme. It is obvious that the better performance of voice traffic can be obtained without sacrificing the video capacity of the system.

B. Example for the Pseudo Reservation Scheme

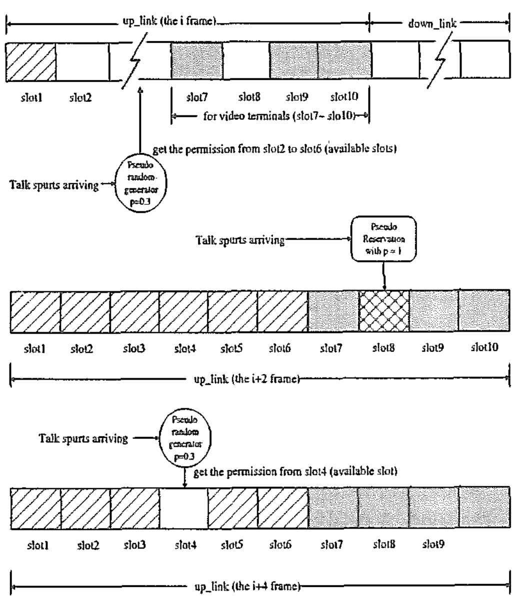

Figure 4. Illustration of the Pseudo Reservation scheme

In the following, we give an example for illustrating the joint voice-video PRMA system with 10 slots for the up-link bandwidth, and Figure 4 presents the operations of the Pseudo Reservation scheme. It assumes that two video terminals occupy four time slots (i.e. from slot 7 to slot 10 as shown in the i frame of Figure 4) to transmit video traffic. One of them releases a slot (e.g. slot 8) for the pseudo reservation with the highest permission probability (p = 1) by the other active voice terminal (e.g. in the i+2 frame), because the contents of the transmission buffer are under the threshold. When the video transmission buffer goes over the threshold, the slot with pseudo reservation will be returned, and the voice terminal must contend for the other available slots in the next frame (e.g. in the i+4 frame). Instinctively, the voice terminals will get more chances to transmit speech packets based on the pseudo reservation scheme, so that the performance of packet dropping probability of each voice user and the bandwidth utilization of the system get better.

IV. PERFORMANCE RESULTS AND DISCUSSIONS

Definition	Value
Channel rate	720 kb/s
Speech rate	32 kb/s
Slot size	576 bytes
Overhead per packet	64 bits
Frame duration	16 ms
Slots/frame	20
Maximum delay limit	32 ms
Mean talk-spurt duration	1 s
Mean silence duration	1.35s
Video permission probability	1
Speech permission probability	0.3

Table 1. Parameters used in the system simulation

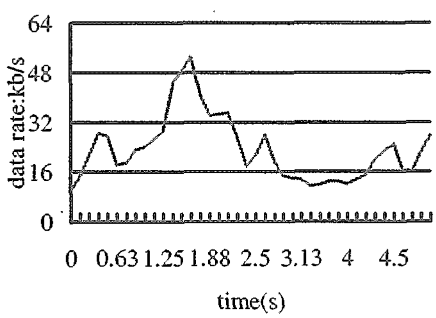

Figure 5. Video traffic model of "Suzie"

We use the integration protocol and pseudo reservation scheme in previous sections to simulate the performance of the joint voice-video PRMA system by the workstation based on system parameters as listed in Table 1. We assume that each voice terminal contains a voice activity detector which outputs talk-spurt and silence alternatively, and the probability distributions of talk-spurt and silence duration are the exponential distribution for each voice terminal, where the mean values are shown in Table 1 respectively [3]. An active voice terminal drops all packets in the FIFO (First In First Out) buffer that incurs a contention delay longer than some maximum delay limit. In order to obtain the voice packet dropping probability, we also define the maximum delay limit to be 32 ms, and the buffer size is 40 packets for each active voice terminal.

Again, the simulation system assumes that two video terminals request total four time slots to transmit the video traffic with the video traffic model of "Suzie" (a video telephony service) shown in Figure 5, which have been coded by ITU-T H.263 compression scheme [11]. The system must determine the threshold value of each video transmission buffer (i.e. 512 bits in our simulation system) at intervals of one frame-duration, and to decide when the video terminal releases and recovers the reserved slots according to the pseudo reservation scheme. Figures 6~8 will show the performance of the pseudo reservation scheme compared to the original PRMA system with conventional slot reservation.

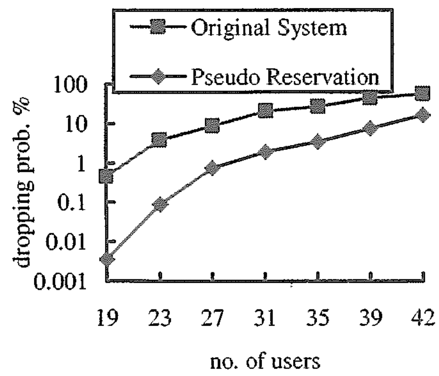

Figure 6. Mean dropping probability of the voice terminals

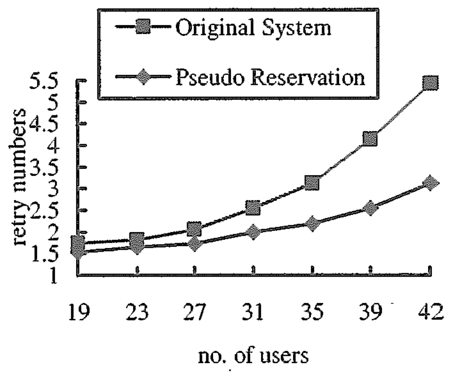

Figure 7. Mean number of retrials of voice terminals

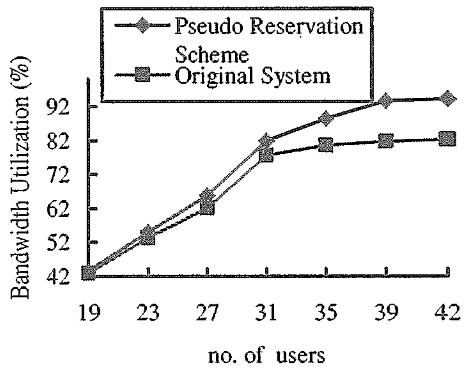

Figure 8. Bandwidth utilization of the PRMA system

Figure 6 displays the relationship of mean voice packet dropping probability, P_{drop}, to the number of simultaneous transmission for voice and video users (we assume that there are two video users in the system). It reveals that the pseudo reservation scheme can provide at least 28 voice users for simultaneous transmission under the condition of P_{drop} = 0.01, while the original system just only provides 20 voice users due to the waste of the released slots by video users. Figure 7 indicates the mean number of retrials for active contending voice users from silence to talk-spurt duration until accessing the available slots or released slots successfully. The smaller number of retrials reflects the lower power consumption of system. Figure 8 shows the bandwidth utilization of the joint voice-video PRMA system with and without the pseudo reservation scheme. Because the video users frequently release redundant slots (about 2 slots based on the video traffic model), the voice terminals get more chances to transmit speech packets based on the pseudo reservation scheme. Thus, it owns the merit of increasing system bandwidth for the active voice terminals. The pseudo reservation scheme leads to all above performance being better than the original system.

V. CONCLUSION

We have proposed a new scheme, Pseudo Reservation scheme, for the integration of voice and video transmission based on the PRMA protocol. The scheme adaptively releases reserved slots of the video terminals according to each transmission buffer under the threshold value, and each active voice terminal can access the additional slots with the highest permission probability to improve the system capacity. However, it provides efficient performance thereby leading to overall improvement compared with the original PRMA system without sacrificing the video capacity of the system. But how to satisfy the various QoS (Quality of Service) for integrated multimedia transmission with voice, data and video traffic using the PRMA protocol is currently under investigation.

REFERENCES

[1] D.J. Goodman, R.A.Valenzuela, K.T.Gayliard and B.Ramamurthi, "Packet Reservation Multiple Access for Local Wireless Communications", IEEE Transactions on Communications, Vol.37, No.8, pp.885-890, August 1989.

[2] Xiaoxin Qiu, Victor O. K. Li, "On the Capacity of Packet Reservation Multiple Access with Capture in Personal Communication System", IEEE Transactions on Vehicular Technology, Vol.45, No.4, pp.666-675, November 1996.

[3] D. J. Goodman and Sherry X. Wei, "Efficiency of Packet Reservation Multiple Access", IEEE on Vehicular Technology, Vol.40, No.1, pp.170-176, February 1991.

[4] P. Narasimhan and R. Yates, "A New Protocol for the Integration of Voice and Data over PRMA", IEEE Journal on Selected Areas in Comm., Vol. 14, No. 4, pp.623- 631, May 1996.

[5] Takahiro Suzuki and Shuji Tasaka, "A Performance Comparison of ALOHA-Reservation and PRMA in Integrated Voice and Data Wireless Local Area Networks", TENCOM'92, Vol.2, pp.754-758, November 1992.

[6] D. J. Goodman and Hong Y. Chung, "Transmission of Speech and Data using Packet Reservation Multiple Access", ICC'91, Vol.1, pp.99-104, June 1991.

[7] Wai-Choong Wong and D. J. Goodman, "Integrated data and speech transmission using packet reservation multiple access", ICC'93, Vol.1, pp.172-176 May 1993.

[8] Liang-Seng Koh and Ming T. Liu, "A Wireless Multiple Access Control Protocol for Voice-Data", 1996 International Conference on parallel and distributed system, pp.206-213, 1996.

[9] Shuji Tasaka, Keisaku Hayashi and Yutaka Ishibashi, "Congestion Control for Integrated Video and Data Transmission in a Wireless LAN", GLOBECOM'95, Vol.1, pp.332-338, May 1995.

[10] Jae-Shin Jang and Byung-Cheol Shin, "Performance Evaluation of a Voice/Data Integrated Fast PRMA Protocol", IEICE Trans. Commun. Vol. E80-B, No. 7, pp. 1074-1089, July 1997.

[11] DRAFT ITU-T Recommendation H.263, "Video Coding for Low Bit-rate Communication", International Telecommunication Union, May 1996.

[12] Karel Rijkse, "H.263: Video Coding for Low-Bit-Rate Communication", IEEE Communications Magazine, Vol. 34, No. 12, December 1996.

[13] Block Oriented Network Simulator (BONeS), Release 3.0, Alta Group of Cadence Design Systems, Inc., 1994.