Packet Loss Control for Continuous Media over Heterogeneous Network

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ABSTRACT: In this paper, QOS control of continouous multimedia communication system under heterogeneous environment by the wired and the wireless networks is proposed. In our suggested system, as channel coding, FEC (Forward Error Correction) method with Reed-Solomon coding is introduced to reduce the packet error rate on the wireless network. On the other hand, as source coding, transcoding methods including transformation of various video codings such as M-JPEG, MPEG and Quicktime, controls of Q-factor within a frame, frame rate and color depth is introduced to maintain the required QOS, particularly the end-to-end throughput. The increases of the required bandwidth by redundant packet addition with FEC can be suppressed by the transcoding functions while the packet error rate is reduced to the accepted value. In order to verify the functionality and the efficiency in our suggested system, numerical simulation was carried out. As the result, our suggested system by combination of transcoding and FEC could correct the packet error rate with accepted order while maintaining the frame rate and the amount of data transform at a constant.

複合ネットワークにおける連続メディア転送のためのパケットロス制御

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本稿では、有線と無線が相互接続される環境において、双方向にリアルタイムに連続メディア転送を可能とするため、利用者要求や資源環境の変化に対応できる QOS 制御法について提案する。提案システムでは、特に無線ネットワークにおけるパケットロスを低減させるために、Channel Coding として Reed-Solomon 符号を導入した前方誤り訂正(Forward Error Correcting)を導入する．一方では、End-to-End のスループットを一定に維持するために、Source Coding として、Motion-JPEG, MPEG や QuickTime といったビデオ符号の相互変換方式、フレーム内の Q-factor 制御、フレームレートや解像度の制御といったトランスコーディング機能を導入する．これにより FEC によってパケット誤り率は、許容値内に低減できる一方、FEC の冗長データによる必要帯域の増加は、トランスコーディング機能により、単位時間あたりのデータ量を一定に保つことができる．本方式の有効性を示すために、プロトタイプに対するシミュレーションにより解析を行ない、性能評価を行なったので報告する。

1.INTRODUCTION

In recent years, various multimedia services like multimedia conference system, streaming video, and VOD services have been realized by the development of the high speed and broadband of networks. Not only the present wired networks, such as Cupper-based LANs, optical fiber networks, or CATV-based networks but also wireless and mobile networks, have been used to heterogeneous network environments where the bidirectional multimedia communication is realized beyond the limits of time and space. Then, the usage of the heterogeneous network by the wired and wireless networks makes us to expect the realization of new applications like the advanced traffic system, the disaster prevention system, and the adhoc network system. However, the wireless network has essential problems as follows:

1) network bandwidth is not sufficient.
2) packet delay is large

3) the bit error, namely packet error is high, compared with wired networks. Those problems cause difficulties for seamless communication through the wired and wireless networks. As example, current popular wireless network such as IEEE 802.11b with 2.4GHz and 11Mbps provides the packet loss by the bit error over wireless and causes service quality degradation when the communication distance is larger than a couple of Km or obstacles are existed between communicating stations. On the realtime bidirectional communication by the audio/video, the delay and jitter on packet arrival at the receiver make the realtime communication very difficult, eventually conducts the service quality degradation. To avoid these problems, it is necessary to introduce end-to-end QoS (Quality of Services) guarantee mechanism into the heterogeneous network environment. Moreover, if the reliable protocol like TCP is applied, the delay time due to retransmission for the lost packets would be increased, eventually the realtime communication would become difficult.

In order to solve those problems, we introduce, a new dynamic QOS control method based on the combination of channel coding and transcoding. As channel coding, FEC (Forward Error Correction) with Reed-Solomon coding is used while various transcoding methods are used.

First, our suggested system can dynamically control the FEC redundancy to enable bidirectional realtime video communication and to reduce the packet error rate under the heterogeneous environment where the wired and wireless networks are interconnected. The packet error rate is periodically observed at the receiver side and feed backed to the sender side when the error rate varied. The number of redundant packets for error correction in the unit time is determined by observing packet error rate and the desired packet error rate. The FEC with Reed-Solomon coding is applied to both data packets and redundant packets at the sender side and the calculated packets are sent to the receiver side. The receiver side recalculates whether the packet error happened or not. If packet error happened, then error correction process is executed. Thus, the length of FEC redundancy is dynamically controlled to maintain the actual packet error rate at a constant on the end-to-end communication.

Next, transcoding method as source coding is introduced to provide stable end-to-end multimedia service quality between the wired and wireless networks. Transcoding can control the required network throughput even the FEC method generates additional redundant packet transmission and increases the required communication bandwidth. The transcoding is executed by the system functions by changing Q-factor, frame rate, color depth and pixel resolution at the sender side, and transforming the video formats coding like from/to Motion JPEG or MPEG-1, 2, 4. Here, the controls of Q-factor, frame rate, and the transform of video format are dynamically fitted by the corresponding to change the network bandwidth and the user resources.

The reminder of this paper is organized as follows. The next section provides system configuration of interconnected wire and wireless networks. Section 3 introduces our suggested system architecture for QOS control of realtime multimedia communication system. Section 4 explains various transcoding methods which are considered in this paper. Section 5 explains the error recovery method by FEC and it dynamic control. Section 6 deals with simulation and its results for a simple prototyped network system to verify the functionality and performance of the suggested system. Finally section 7 presents the conclusions.

2. SYSTEM CONFIGURATION

The multimedia communication services network currently assumed in this paper is organized by integration of wireless network such as IEEE 802.11b (2.4GHz, 11Mbps) and the wired network based on an optical fiber, as shown in Fig. 1. This interconnected network is constructed by fixed hosts (FH) like desktop typed personal computers, and mobile hosts (MH) like notebook typed personal computers.

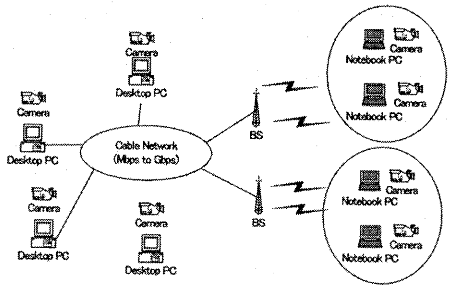


Fig. 1 The Heterogeneous Network by the Wired and Wireless Networks

FH and MH have video cameras and video capturing functions to facilitate realtime TV conference system or net meetings. The wired and wireless networks are interconnected by a base stations (BS) which performs as gateway functions. Therefore, FH and MH can communicate with each other in both directions by end-to-end manner.

However, since the interconnected network is consisted of the wired network which is based on sufficient resource environment and the wireless network which is not, especially because of the higher packet error rate due to the bit error rate on wireless environment, the end-to-end audio/video communication services cause service quality degradation. Therefore, when real time bidirectional communication by the audio/video is implemented using the interconnected network, the quality deterioration over audio/video streams due to the delay and jitter on packet arrivals at the receiver make the realtime communication very difficult. In order to solve these problems, it is necessary to introduce novel these problems, it is necessary to introduce novel functions to guarantee end-to-end QoS in audio/video communication system through the wired and the wireless networks, as introduced in the next section.

3. SYSTEM ARCHITECTURE

The system architecture of our suggested video communication system is based on the peer-to-peer model and is organized by the media coordination system (MCS) [5] which consists of three layers, including the synchronization layer, the data transform layer, and the media flow control layer between the application layer and the transport layer in the OSI reference model to realize end-to-end QoS guarantee as depicted in Fig 2. The synchronization layer performs inter/intra media synchronizations in between audio and video frames. The media transform layer performs transcoding functions including transformation of a video coding to another one, such as from/to Motion JPEG to/from MPEG-1, 2, 4, changing Q-factor, frame rate, color depth and pixel resolution at the sender side. The controls of Q-factor, frame rate, and the transform of video format are dynamically fitted according to change of the user resource environment, such as CPU load and memory buffer occupancy. The media flow control layer performs variable packet flow control and packet error detection and correction functions according to dynamic change of traffic load conditions of the computers and networks resource environment.

MCS is furthermore vertically divided into four planes such as the user plane, the QoS maintenance plane, the control plane, and the stream management plane by referring ATM architecture and adopted QoS-A model of Lancaster university [8]. In the user plane, synchronization function between different media, such as audio and video streams, data transform function between different media attributes, media flow control for both constant and variable bit rate transmission for video/audio streams are performed. In the control plane, connection establishment/release of the media streams and QoS renegotiation are maintained. In the QoS maintenance plane, each entity for video/audio services is responsible for the fine-grained monitoring and maintenance of their associated protocol entities. In the stream management plane, the most suitable QoS parameter values on each protocol values on each protocol layer are determined according to the user's QoS requirements, characteristics of the source media data, output device attributes, and available computing and network resources.

In this research, it is especially important how MCS deal with packet error by the rate of bit errors occurred on the wireless networks, and perform new recovery control functions in a media flow control layer. In order to realize the above functions, the packet error rate is periodically measured at receiver side in the media flow layer as follows.

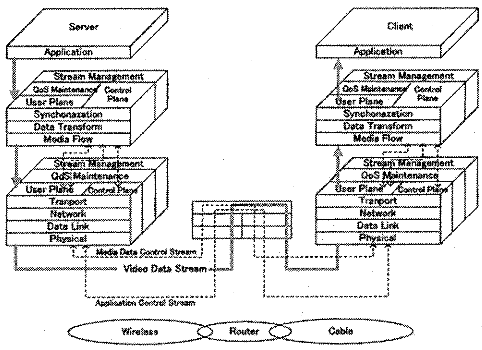


Fig. 2 System Architecture

If the packet error rate exceeds the permissible value, the measured value would be feed back to the media flow control layer of the sender side. Then the control function to keep the packet error rate under the permissible value would be carried out by

1) changing the value Q factor (Compression rate)
2) changing the frame rate by sub-sampling the intermediate frames
3) changing the video coding scheme
4) changing the color depth
5) Changing the pixel resolution

Also, as mentioned in the following section, the increase of the number of the transmitted packets per unit time between the data transform layers in both hosts by FEC redundancy when the burst error in bit transmission occurred can be kept constant by the same way during video communication service.

4.TRANSCODING

As mentioned in the previous section, the assumed end-to-end communication was realized by the introduction of transcoding. As typical network environment, wireless side doesn't have sufficient resources like the available bandwidth although the wired side does. Here, in order to realize the seamless end-to-end communication, the control system of Q factor of intra frame, frame rate control, and the transform of video coding were introduced for our transcoding between wired and wireless side.

Q-factor (Quality factor) is a number to generate quantization tables for intra frame and determines the degree of compression rate in M-JPEG and MPEG video streams. That is, the video quality was transcoded by the controlling Q-factor between wired and wireless networks, depending on the host and network conditions.

Secondly, video frame rate is also controlled according to the user's requirement and resource conditions. In case of Motion JPEG, any frames are simply sub-sampled to adjust to the desired rate. In the case of MPEG video which is consisted of a number of group of pictures (GOP). Furthermore, one GOP is consisted of I, B, P-pictures for inter-frame prediction and has mutual relation to each video frame. It is obvious the priority of I-picture is the highest and the priority of P-picture is higher than the B-picture, because I-picture is required to predict the P- and B-pictures and P-picture is required to predict B-pictures. Therefore, when sub-sampling of the MPEG frames is required, some of B-pictures are sub-sampling first, then P-picture and finally I-picture depending on the host and network load conditions. Besides, the transform of video coding was adopted among one coding to another such as Motion JPEG, MPEG-1, 2 4, H-261, -323 or Quicktime, etc.

By these transcoding, it is possible to maintain the traffic constant even if the FEC redundancy is increased or the condition of network characteristics is changed.

5. ERROR RECOVERY BY FEC

In the environment where a certain amount of packet error is allowed like this research, FEC is considered to be the very effective method when focusing on the importance of a time-critical characteristic. Compared with ARQ (the Automatic Repeat reQuest) which is a method to repeat transmitting the error or lost packets, FEC which carries the additional redundant data by the error correction code has the smaller calculation time during the recovery than the packet delay time by ARQ. Reed-Solomon (RS) coding as the FEC code was introduced to the media flow control layer in our system architecture as indicated in Fig 3.

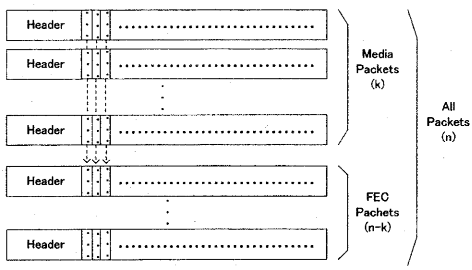


Fig. 3 Reed Solomon Coding

When the number of the RS coding packets is set as n and the data packets set as k in a unit time, RS coding has the capability to correct n-k error packets when the position of the bit error in a packet is known. When many packet errors occur more than n-k pieces among n packets, the error probability after RS coding and recovering processes can expressed as,

$$\mathrm{E} = \sum_{i=n-k+1}^{n} {}_{n}C_{i} e^{i} (1-e)^{n-i} \tag{1}$$

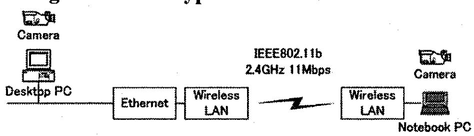
Here, e presents the rate of the packet error between the transmitted and receipt hosts. Since k is known value, E is determined by the value of n. If E is calculated for various cases of n in advance, target error rate can be calculated. Thus the error recovery power of RS coding is determined by n. Therefore, the packet error rate can be kept within the admissible value below by measuring e periodically, then calculating the E using formula (1) for the measured e and various n, and by feeding back the value of n which is equivalent to the admissible error rate E.

Even though the unexpected packet error may randomly cause beyond the predefined packet error rate, those packet error rate is periodically measured at the receiver side and suitable length n of RS coding packets can be calculated and dynamically feed back to the sender side. Therefore, the dynamic redundancy control for to maintain the packet error rate at constant can be attained.

6. SIMULATION

The numerical simulation was evaluated in order to verify the functionality and effectiveness of our suggested video communication system. The prototyped system of this simulation is shown as Fig 4.

Fig. 4 The Prototype of the Simulation



100Mbps Ethernet was used as the wired network and IEEE802.11b wireless LAN (2.4GHz, DSDD, 11Mbps) was used as the wireless network. Then, the real time video data with speech by man by Motion JPEG (average compression rate = 1/15) of 320x240x30fps was transmitted to each other. The numerical values of video source are shown in Table 1.

Assuming the network environment of wireless LAN dynamically changes, the packet error rate was abruptly changed from 10^{-6} to 0.5. In this condition, we would like to recover the packet error rate up to 10^{-6} by applying FEC method. By adding three FEC packets to ten video packets of 1500 bytes, the packet error could be recovered to 10^{-6}.

Video Format Coding	Motion JPEG
Frame Size	320x240
Frame Rate	30fps
Color Depth	3byte(Full Color)
Average Compression Rate	1/15
Video Scene	Speech by Man
Network Bandwidth	3.68Mbps(for1/15)
Observed Packet Error Rate	0.5～1.0E-6
Required Correct Packet Error	1.0E-6
Maximum End-to-End	3.68Mbps
Compression Rate(Q-factor)	1/5～1/40
Frame Rate	1～30fps

Table 1 The Numerical Values of Video Source

In the case of PentiumIII 800MHs as the host computers at sender side, the calculation time for a generating FEC packet was less than 0.004 seconds from actual measurement. Moreover the calculation times for the error correction at receiver side were 0.09 seconds, 0.11 seconds, and 0.14 seconds for the cases of one, two, and three packet errors respectively. The packet error rate with 10^{-3} is equivalent to one error for every 1000 packets. On the other hand, on the assumed video data transmission, about ($3.68x10^6/1500/8=307$packets/sec) 307 packets are transmitted in 1 second. Therefore, the error correction of Motion JPEG video will be performed about every (1000/307=3.26sec) 3.26 seconds. Although 0.09 seconds will be spent to correct packet error during 3.26 seconds. This means that the extra time for packet correction within 1 second is equivalent to 0.027 second. This value is smaller than 0.033(for 30 frame/sec) and the frame rate is not degraded by the packet correction time with FEC method.

On the other hand, although the amount of the transmitted video data per unit time is increased 30 percents by transmitting of ten video data packets and extra three FEC packets, the amount of video data can be kept the same, for instance, by changing compression ratio (namely, Q-factor) of Motion JPEG from 1/15 to 1/20. Therefore, by combining the transcoding function of video data at data transform layer and the FEC function for the packet error in media flow layer, the rate of a packet error can be improved while keeping the same amount of the frame rate and that of the transmitted data even though the change of wireless network resource environment gets worse.

As the second consideration, the packet error rate for different number of data packets (K=10, 20, 30) with different additional FEC packets was calculated to clear how the FEC packets can improve the original error rate. Fig.5 depicts the relation between the number of FEC packets in x-axis and the improved packet error rate in y-axis when the original packet error rate is 0.2. Fig.5 shows the corrected packet error rate after FEC as the number of the FEC packets increases for three different the number of data packet for video stream when the observed packet error rate is fixed at e=0.2. Obviously, the corrected packet error rate can be improved as the number of FEC packets increases. It is also said that when the number of data packet for video stream k in unit time is smaller, the number of the required redundancy packets could be smaller to maintain the same admissible error rate 10^{-6}.

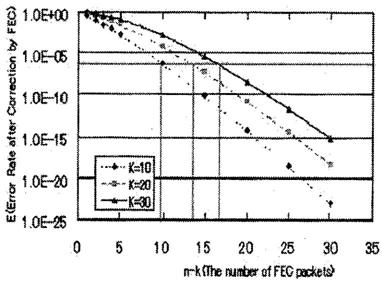


Fig.5 Simulation Result (1)

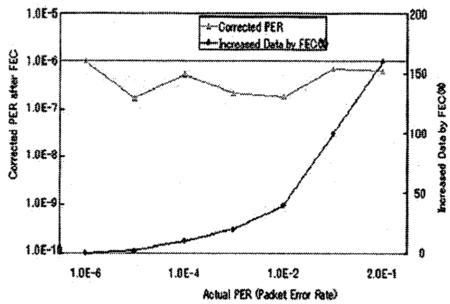


Fig.6 Simulation Result(2)
Corrected Packet Error Rare after FEC

Fig.6 shows the corrected packet error rate after FEC when the actual packet error rate increase.It is found that the corrected packet error rate after FEC can be maintained under 1.0E-6, although the total number o f video data increases as actual the packet error rate increases.This increase of the total data for video data can be reduced by video data control method.

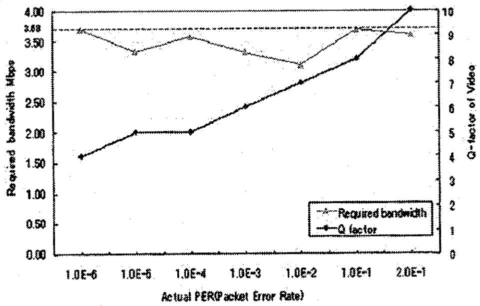


Fig.7 Simulation Result(3)
When Q-factor is controlled

Fig.7 shows the required bandwidth for video data

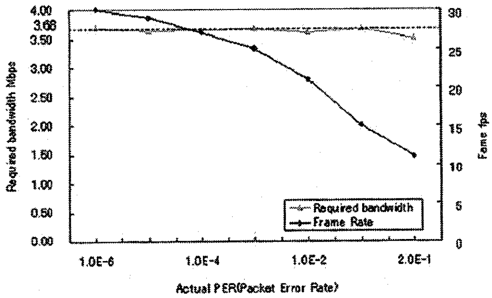


Fig8. Simulation Result (4)
When frame rate is controlled

Fig.8 shows the required bandwidth for video data when the frame rate video is controlled while Q-factor is constant.
Obviously the bandwidth can be maintained within the original rate, 3.68 Mbps although the frame rate decreased.

Through those simulation results, the required band with can be maintained at all most constant even though the packet error rate dynamically changed using the combination of packet error rate control method and video data control method.

7. CONCLUSIONS

In this paper, we introduced a bidirectional realtime video communication system such as a TV conference under the integrated network by the wired and wireless networks. We proposed the transcoding system by Qfactor, frame rate (GoP), and the transform of video coding, and the system which provide the best quality of video service by the combination of the packet loss control function by FEC, the video compression control, the frame rate control, and the resolution control even if under the environment where the calculation capability or network resources are changed. Then, the simulation was carried out to verify the functionality and effectiveness of our suggested system. The results showed the calculation time of the data correction was 0.09 seconds during 3.26 seconds, and it is almost negligible. Therefore realtime video communication can be attained as long as video coding Motion JPEG can be carried out in realtime on the host. This is the effective communication method especially in wireless LAN environment, and it enables the seamless communication by the interconnection with the wired and wireless network. At last, the experimentation of the real-time communication method in the condition of delay or jitter is planned as a future study.

REFERENCES

[1] J. Vass and X. Zhuang.: “A Novel Video Communication System Utilizing Adaptive and Integrated System Design for Mobile Wireless ATM”, Proc. on IEEE ICME, August, 2000.
[2] T. Komura, K. Fujikawa and K. Ikeda,: “Forward Error Correction for QoS-Reserved Internet Broadcasting”, Information Processing Society of Japan, DPS100-18, P81-85, November, 2000.
[3] N. Yamanouchi: “Internet Multimedia Transmission Using Multiple FEC Recovery Classes”, IPSJ journal, Vol.42, No.2, P206-212, Feb.2001.
[4] ISO/TC184/SC5/WG2: Draft Technical Report: Identifying user requirements for systems supporting time-critical communication
[5] K. Hashimoto, T. Chinen, J. Sato and Y. Shibata,: Packet and Frame Rate Control Methods for Compressed Video Transmission (Special Issue on Multimedia Distributed and Cooperative Computing”, IPSJ Journal, Vol.30, No.2, P337-347, 1998.
[6] K. Hashimoto, M. Katsumoto, M. Watanabe and Y. Shibata,: "End-to-End QoS Architecture for Continuous Media Service, Proc. ICOIN-10, P578-583, 1996.
[7] J. Sato, Y. Kousaka, K. Hashimoto, Y. Shibata, and N. Shiratori,: "Compressed Video Transmission Protocol Considering Dynamic QoS Control", Proceeding of the ICPP Workshops, P95-104, August, 1998.
[8] A. Campell, G. Coulson, and D. Hutchson,: “A Quality of Service Architecture”, ACM SIGCOM Computer Communication Review, Vol.24, No.2, pp.1-27, 1995.
[9] Kazuo Takahata, Norihiko Uchida and Yoshitaka Shibata. “QOS Control of Multimedia Communication over Wireless Network”.IEEE Proc. On ICDCS Work Shop,MNSA,P336-340,July,2002.