영상회의 시스템을 위한 RTP/RTCP 구현 및 오디오 데이터 전송을 이용한 QoS 분석

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요 약

본 논문은 ITU-T에서 제안한 영상회의 시스템에서 오디오/비디오 스트림 데이터를 회의 참여자에 송수신하거나 또는 수신된 멀티미디어 데이터들에 대한 QoS 정보를 송신측에 피드백 하기 위해 제공되는 RTP/RTCP 프로토콜(RFC1889, 1890)에 관한 설계 및 구현에 관해 기술 한다. RTP는 인코더로부터 전달된 오디오/비디오 데이터를 고정 포맷으로 패킷화하여 모든 회의 참여자에 멀티캐스팅하고, RTCP모듈은 RTP와 함께 연동되면서 수신RTP 패킷을 모니터하여 지연, 지연 변이 및 패킷 손실 등의 QoS 값들을 검출하고, 이를 비-정기적으로 송신측에 피드백하도록 구현하였다. 이들 프로토콜은 Windows NT에서 멀티쓰레드 방식으로 구현됐으며, 하위 프로토콜로 socket I/F을 통해서 UDP/IP-Multicast를 이용하였다. 또한, 인터넷 환경에서 영상회의 시스템을 수행했을 때 나타나는 여러 QoS 값들을 검출하여 분석하였다. 시험은 오디오 데이터 전송을 이용하였으며 통신 부하가 심한 시간 구간에서 지연과 지연 변이는 음성 인식에 대체로 허용 범위에 충족되나 다량의 패킷 손실에 따른 품질 저하를 분석할 수 있었으며, 대부분의 손실된 패킷들은 비-연속적인 특성을 갖는 것으로 나타났다.

Implementation of RTP/RTCP for Teleconferencing System and Analysis of Quality-of-Service using Audio Data Transmission

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ABSTRACT

This paper describes the design and the implementation of the Realtime Transport Protocol(RTP)/Realtime Control Protocol(RTCP) (RFC 1889, 1890) that is used to transmit the audio/video data to any destination and to feedback the Quality of Service (QoS) information of the received media data to the sender, in the teleconferencing systems proposed by ITU-T. These protocols are implemented with multi-thread technique and run on top of UDP/IP-Multicast through the socket interface as the underlying protocol. The upper layer is implemented such that it can be accessed by the H.245 conference control protocol. The RTP packetizes the digitized audio/video data from the encoder into a fixed format, and multicast to the participants. The RTCP monitors RTP packets and extracts the QoS values from it such as round-trip delay, jitter and packet loss to form RTCP packets and non-periodically sends them to the sender site.

In this paper, we also describe the study of measurement and analysis for QoS factors that observed on performing teleconferencing system over Internet. The results from this experiment is indicate that RTT and Jitter value are acceptable even network load is high. However, it appears that packet loss rate is high in daytime and most losses periods have length one or two.

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1. Introduction

In the past few years, teleconferencing system has a great deal of attention from both the general public and the research community. With the growth in bandwidth and improvement in compression techniques, multimedia conferencing systems are emerging as an inexpensive solution. With the introduction of IP multicast technology, it is possible to conduct ad-hoc conferences between large numbers of participants [5]. The ITU has recently completed the development of the H.323 standard for desk-top conferencing system over non-guaranteed QoS LAN, and it also accepts the RTP/RTCP as a part of specification, in order to transfer the audio-visual data between end systems. The RTP and RTCP provide the real-time properties such as synchronized playback, packet loss, security and payload type identification. They are announced by IETF as RFC1889, 1890 in 1996 and ITU-T accepted them as audio/video transmission protocols [1].

This paper describes the study of the design and development of RTP/RTCP to guarantee the maximum QoS of audio/video data transmission to the conference participants, which was a part of H.323 terminal development for the teleconferencing. Present network protocols including the Internet were developed with objectives of transmitting data such as messages or files that has non-realtime bursty traffic property rather than the support of the multimedia applications of audio-visual information that has realtime property. Generally, non-real time transmission requires reliable transfer without packet loss even delay may occur. But in real-time transmission QoS parameters such as delay and delay variation have great influence on the qualitative aspect of services while it is very generous to small amount of packet loss. Therefore to solve the performance issue there are generally two approaches. The first approach is to extend or modify the existing transport layer protocol to guarantee the quality that multimedia services require. The reservation and the admission control mechanisms are in this category but there are many problems to be solved. The second method is to adjust application services to be adaptive to current network services. In this method many QoS capabilities are dynamically observed and the result is notified to application programs and based on this control the multimedia devices are controlled or through certain mechanism the stream data from the audio/video devices is formatted adaptively to the network services. The emergence of RTP/RTCP belongs to the second category. The video conferencing, a typical application service that requires real-time data transmission, in distributed environments such as the Internet, uses the UDP as the transport protocol but the performance in bandwidth, packet loss or transmission delay is not guaranteed. If there is no recovery mechanism of any packet loss, the RTP/UDP/IP may experiences the distortion from the audio data loss. Therefore, RTP/RTCP doesn't improve the quality of the network itself but adjust the stream data rate by controlling the audio/video codec by observing QoS parameters [6], [8].

The remainder of this paper is structured as follows. In the following chapter we describes the structure of the H.323 teleconferencing system and the functions of the RTP/RTCP. In the third chapter describes the design of the RTP/RTCP of the teleconferencing system in functional point of view. The implementation with the object oriented method is given in third chapter. The fifth chapter describes the practical results and its analysis which being appear during teleconference over the Internet. In the last chapter, we concluded this paper with the direction of the future study.

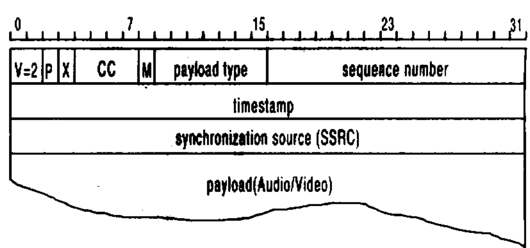
2. Related works

2.1 The H.323 Terminal Architecture

The H.323 is a standard approved by the ITU Study Group 15. It defines how the audiovisual stream data for conferencing is transmitted across networks. The ITU document, "Recommendation H.323, issued in 1996, describes the terminals, equipment, and services for multimedia communication over Local Area Networks not guaranteed by the Quality of Service. The H.323 standard provides a foundation for audio, video and data communications across IP based networks, including the Internet and the keystone for the LAN based product for real-time multimedia applications. This defines the components, procedures, and protocol necessary to provide audiovisual communication in LAN [1], [4]. Other recommendations in H.323-Series includes the H.225.0 packet and synchronization, H.245 control, H.261 and H.263 video codec. The H.225.0 describes how the audiovisual data and control information in non-guaranteed QoS LAN can be managed to provide inter-operation service in the H.323 terminals. The ITU accepts the RTP/RTCP into the H.225.0 standard, as Annex A, unless modified by text. In order to setup calls between each H.323 terminals, the call signaling function is required and this H.255.0 defines the procedure and messages for call setup. The H.245 is a control protocol that provides end-to-end signaling for proper operation of multimedia terminals, and signals other end-to-end system functions. It provides capability exchanges, signaling of commands and indications, and open messages to describe the contents of the logical channels. The H.323 makes use of the logical channel signaling procedures of the Recommendation of the H.245. Procedures are provided for the expression of the receiver and sender capabilities, so sender's encoder is limited to what the receivers has. In sender site, RTP/RTCP sample a audio/video signal, packetizes the data and transmitters it over network. The receiver then unpacketizes the data and plays it back to the screen or speaker [2], [4], [13].

2.2 Real-Time Transport Protocol

The Real-Time Transport Protocol has been developed for a few years by IETF. It has recently been standardized as rfc1889, 1890. RTP supports audio/video data transfer to multiple destinations, to realize distribution properties of the multimedia applications. This includes teleconferencing using the multicast protocol, if available, however, not limited to any particular application. There are recently considered to be use RTP as a part of web transaction for transmission of data with real-time constraints such as audio and video, over the Internet. RTP provides end-to-end delivery services for real-time data transmission over unicast and multicast network services. It does not normally provide transmission functionality handled by the transport protocol, but offers functional properties suited for carrying real-time data. RTP has no protocol state alone and makes no assumptions about the capabilities(reliability or security) of the underlying protocol. The services that RTP provides includes timestamping, sequence numbering, payload identification, and source identification. The important information in the RTP header are timing and sequence information. The receiver uses these timestampe to reconstruct the original timing before playing back the data. Timestampe contain relative timing information between packets. Therefore, the sender and receiver do not need to be synchronized. Sequence number information is necessary to detect packet loss problems and handle out of order packets. The format of the RTP header is simply illustrated in Figure 1 [2], [3], [6].



(Fig. 1) RTP Packet Format

2.3 Real-Time Control Protocol

The RTCP specification defines several packet types to carry a variety of control information. Multiple RTCP packets may be concatenated without any intervening separators to form a compound RTCP packet that is sent in a single packet. RTCP consists of the following packets: The SR and RR packets in the compound packet holds several reception report blocks(RRB). The number of RRB relies on the number of sources of the RTP packet after the last RTCP packet transmission. Each 24-bytes RRB contains the QoS information about the RTP transmission to carry to the source, as illustrated in Figure 2.

0 … 7	15 … 31
synchronization source (SSRC)	
fraction lost	cumulative number of packet lost
extended highest sequence number received	
interarrival jitter	
last SR(LSR)	
delay since last SR(DLSR)	

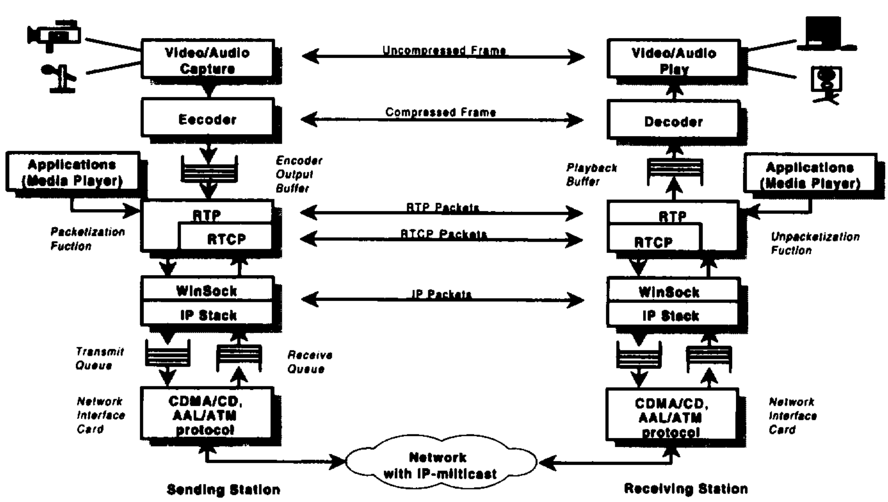
(Fig. 2) Reception Report Block

3. Design of the RTP/RTCP

The data flow in teleconferencing system means a series of procedure which transmits audio/video data to the codec board of the other system. In this stream controlling layer, the RTP/RTCP is located between the codec device driver and network layer. The RTCP monitors and processes lots of QoS parameters to control the quality during transmission. Thus, data transmission starts from the control of H.323 application program, and RTP synchronizes with the codec device and take the encoded data from the encoder, put it into the output buffer for transmission the data to the network. In receiving site, the RTP waits for the packet arrival from the network. It is processed as blocking mode and the header of received packet is transmitted to the RTCP and the data part is transmitted to the playback buffer. The detailed process describes in the next chapter. The relationship which use the RTP/RTCP between audio/video data flow and protocol shows as in the Figure 3 [1], [2].

3.1 Functional model of RTP

The major function of RTP is to transmit the audio/video stream data, which is created consecutively by encoder, to network with UDP/IP or to receive the packet of RTP format from network such as an Internet. In this case, all packets are fixed format and a media is assigned as a sampling



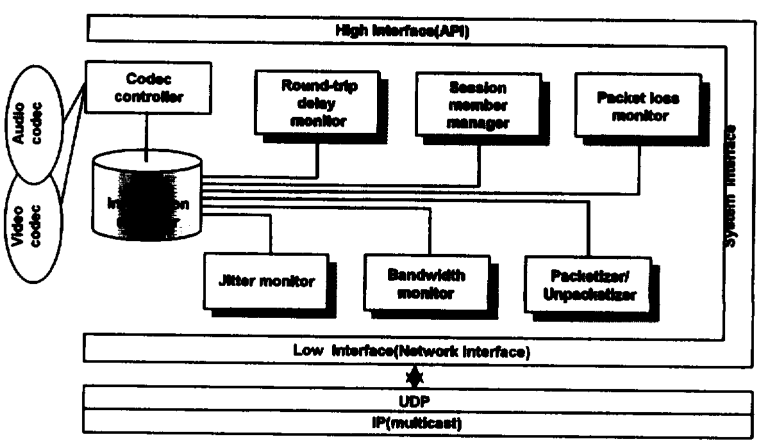
(Fig. 3) protocol and data flow for H.323 Terminal pointed timestamp and packet sequence. A receiver can provide several functions such as packet recovery mechanism or intermedia synchronization(lip-sync) by using it. In RTP media data transmission, since the data from device driver is transmitted to the network with using socket I/F, RTP packet transmission is processed after a device driver captures the stream data through encoder to transmit the stream data from a device driver to a buffer. The playout of a media data in a receiving site relies on the irregular arrival time of a packet. We design that the implementation architecture of the RTP protocol is consisted of three functional module.

- **sequence control module**
  Since the UDP/IP protocol stack does not guarantee the turn of a packet transmission, unless there is no sequence control in a receiver site, a quality deterioration can be caused by the playout of which turn is changed in video/audio data. Therefore, it is necessary to control a playout turn with a sequence number of RTP header. To verify the turn of a received RTP packet, a prefetching is necessary in the time point of initially receiving. Since a buffer for sequence control causes a playout delay, actually the size of buffer relies on a human perception.

- **playback module**
  The header of the RTP packet shows the time point of sampling the data packet in an encoder and it controls the time point of playback between each packet. The control can be processed by the interval in a timestamp when an arrived data part of RTP packet is transmitted to a playback buffer under the device driver's control.
- **packet processing module**
  The protocol engine of the RTP performs following functions:
  ✓ packetizing the media data or unpacketizing
  ✓ verifying the head of arriving packet
  ✓ transmitting the RTP packet to the network with socket I/F
  ✓ receiving packets from network
  ✓ interface with applications

3.2 Functional model of RTCP

The RTCP periodically transmits the control packet to all participants in conferencing with the transmission mechanism as in the RTP. The control packet includes the network's information extracted from the receiver or sender. The functional view shows as in the Figure 4. Each module is composed of one or several objects. The collected information, statistics information, saved on the information base under the control of the QoS information manager [10].



(Fig. 4) RTCP functional modules

- **Packet Loss Monitor**

Participants in the teleconferencing systems suffers from packet loss by more often because the RTP/UDP/IP-multicast protocol stack is unreliable in nature. The dropping of the packets is mainly due to the congestion or transmission error over networks. The RTP packets may be mainly dropped from node congestion. The quality of the audio and video streams the delivered from source to destination depends essentially on the number of lost packets. Since all the RTP packets has a sequence of number that increments monotonically, the fraction of all the packets that does not arrive can be measured, In general, Packet Loss Rate(PLR) defines the proportion between total number of lost RTPs, N_{loss}, and the number of totally transferred RTPs, N_{total}, which may occur in any given Interval time. Furthermore, the packet loss probability P_{loss} can be simply defined by $0 \leq P_{loss} = \frac{N_{loss}}{N_{transfer}} \leq 1$, where $N_{transfer} \geq N_{loss}$. Figure12 illustrates the examination of packet loss in teleconferencing system [7].

- **Jitter Monitor**

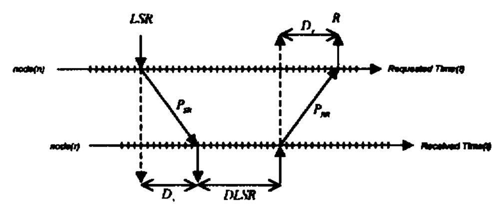
The timestamp in a RTP packet notes sampling instant that driven from a clock. The sampling clock must be linearly incremented, thus denoting when the packets will be played in relation to the other packets. If packets i and j are assigned with timestamps S_i and S_j when they are sent, and they are received at times R_i and R_j, separately, then $D_{i,j} = (R_j - S_j) - (R_i - S_i) = (R_j - S_i) - (R_i - S_i)$ where $j = i+1$, is a transit time in time stamp unit. Therefore, $D_{i,j}$ indicate the transit time between consecutively arriving packets, and if packets arrive at the same rate as they are sent, $D_{i,j}$ is 0. According to the description of the RTCP, the jitter is defined as a smoothed, averaged the delay variation between consecutive packets over time. Jitter function be smoothed:

$$J_i = J_{i-1} + (|D(i-1, i)| - J_{i-1})/16 = 15/16\, J_{i-1} + 1/16 * |D_{i-1,i}|$$

This gain factor 1/16 gives a good noise reduction ration, as optimal first-order estimator.

- **Round-Trip Delay Monitor**

Transferring audio and video stream data between the source and sink consumes time for processing and queuing in the nodes and for the transmission in the network. The RTP packets containing audio and video experience a certain delay between participants who joined in any teleconference. $D(D_s+D_r)$denotes the round-trip delay will need to be transferred, as be illustrated in Figure 5. Round-trip delay can be defined as $D = R - DLSR - LSR$, where $DLSR$ denotes the time between receiving the last SR packet from the source ($node(n)$) and sending the reception report block (RRB)[9].



(Fig. 5) time flow for RTT calculation

- **Bandwidth Monitor**

The teleconferencing system generally allows ranging from a few participants to thousands to attend. The RTCP traffic also increases with the number of participants. This overhead of the control traffic may become an obstacle to transfer media data, due to the sharing of the network bandwidth with data traffic. Therefore, the rate of the RTCP traffic should be scaled down. The RFC1889 suggest the fraction of the session bandwidth be allocated to the RTCP be fixed at 5%. In other words, the session bandwidth is the product of individual sender's bandwidth multiplied by the number of participants and the RTCP bandwidth is 5% of that product. The bandwidth of the stream data generally defines the amount of the user data transferred during a certain time interval between the sender and the receiver. The RTP bandwidth for stream data transmission depends upon the estimate of the number of sites participating in the conference session. It must maintain the number of the session members to estimate the data traffic of each participant, so the Session Manager module must keep track of them using the SSRC or CSRC identifier. The RTCP packet interval is given by the RTCP packet size S_{rtcp} and RTCP bandwidth B_{rtcp} : $I=S_{rtcp}/(B_{rtp}*0.05)$. The inter-arrival time, I, denotes the time distance between two subsequent RTCP transfers. Consequently, the Bandwidth Monitor module provides various functions monitoring data and control traffic, in order to determine the transmission interval of the RTCP packets.

- **The Session Member Manager**

The session member manager modules play two important roles. First, the number of session members should be maintained to roughly determine the value of the RTCP packet transmission interval. The second important role is to retain the user information of the participants. The SDES packet is composed of items describing the source. Once the participant is valid, this module retains the information of the participants using the SDES items received. The session member manager maintains the member table, and according users are joined or leave, creates or destroys the entry or exit at the member table according to the users entrance/exit.

- **The Packetizer / Unpacketizer**

The RTCP defines several types of packets in order to perform the functions of the control. Multiple RTCP packets may be concatenated into a single packet as a compound packet before being sent. Each individual RTCP packet embraced in a compound packet can be handled independently at the receiver sites. The Packetizer/ Unpacketizer module handles the compound packets in order to analyze the packets received from or will be sent to networks

- **The QoS Information Manager**

To guarantee the quality of the multimedia transmission, the QoS information must be collected. This information is used to control the codec parameter, such as the bit rate, quantization, or to apply any mechanism that may be able to enhance the transmission quality. To do this, the QoS Information Manager manages the table to do deal with the QoS information, and provide interface functions for other functional modules to access the QoS table. The QoS table handled by this module is described in further detail in next chapter.

4. Implementation Techniques

In this chapter we describe the object oriented methods of the implementation and the result of RTP/RTCP. The software architecture of the teleconferencing data processing consists of the part that handles RTP/RTCP and the user mode library to access the audio/video device driver. The user mode library is the collection of the functions that stream processing part can access to utilize the audio hardware and the video hardware. The audio library is a low level service and is 8bit sampling, 11.025KHz mono type. Users use the video I/O functions request the data in synchronization method and acquire the compressed video data via video I/O functions. Each library uses the functions that Windows NT provides and detailed description of these functions is omitted.

4.1 Stream Processing Elements

Below we give detailed description of the RTP/RTCP processing part implemented in DLL with Visual C++. The Stream Processing Elements are abstract objects for unified processing and in reality the protocol implementation is included in the stream objects [12].

- **Media Unit**

  The media unit is a structure that contains the digitized audio/video data unit sampled from the codec board during a period of time and has variable size. In this paper, the media unit doesn't refer to a single frame visually sensible but refers as the media data that will be placed to the encoder output buffer. The media units are included in the payload field of the RTP packets.

- **Source Object**

  The source object is an object that contains input media units and is the beginning point of threads. The source object reads media units from input devices through the connected medium object.

- **Destination Object**

  The destination object holds the output media and is the end point of the media unit generated by a source object. Therefore the destination objects of the transmitter define the functions to access the network and on the receiver side, the destination objects offer the functionality of the transmission or the monitoring of the medium unit in the playback buffer of the medium object.

- **Stream Object**

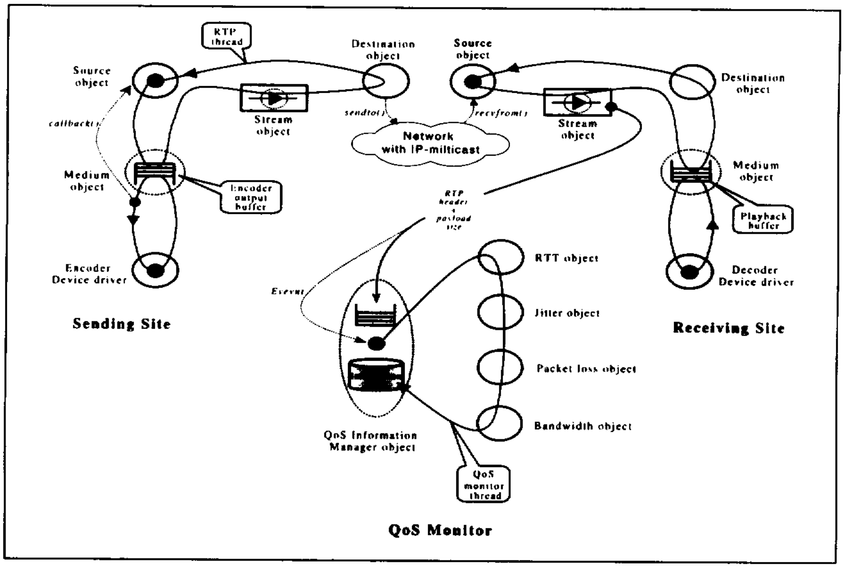
  The stream object is the object that connects the source object and the destination object, in real RTP threads a stream object contains RTP objects and in RTCP threads stream objects contain RTCP objects. The stream objects transmit media data input from the medium objects connected to the source objects to the destination objects.

- **Medium Object**

  The medium objects represent the audio, the video and the network devices. The medium objects are functions that input from and output to the corresponding device with connection to source or destination objects and implemented in the form of a DLL.

4.2 RTP Thread Operation

The RTP thread formats the media units transferred from the codec board into RTP packets and multicasts to the network. The receiving RTP receives RTP packets in blocking mode, detects related QoS information and send the data to the decoder. As can be seen in the Figure 6, the encoder device driver of the sender transfers the media unit sampled by the board to the encoder output buffer and controls the beginning of the RTP thread by using the callback. Since the medium object itself controls the data processing beginning, the time required for the media unit to reside in the encoder output buffer can be minimized. On the other hand, the RTP thread of the receiver waits in the blocking mode for the RTP packets to be received from the network and the received packets are sent to the stream object. The stream object that process the RTP controls the sequence using the sequence number field of the RTP header (if needed re-ordering is performed) or executes the synchronizing playout functions with the timestamp field. For this functions, small amount of internal buffer is required. Also the RTP thread of the receiver transfers the RTP header and the payload size to the RTCP thread for the purpose of QoS information collection. This payload size information is used to determine the transmission timing of the subsequent RTCP packets. At this time, the stream object transfer the RTP header and payload size information to the QoS Information Manager object and triggers off start of the QoS Monitor thread using event. The QoS Monitor thread calculate the QoS value via the RTT object, Jitter object, Packet

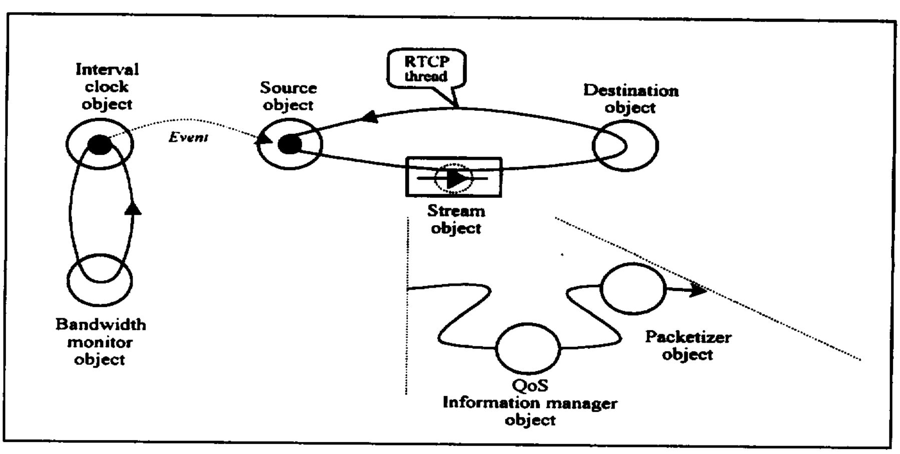


.(Fig. 6) A graphical representation of the RTP thread operation

Loss object, and Bandwidth object with RTP header and size information, and put it into the information base which managed by itself. These Information stored in table are send to the sender who already sent RTP packet, using RTCP packets.

4.3 RTCP Thread Operation

The RTCP packets that transfer the feedback information are transferred regularly depending upon the RTP bandwidth. The bandwidth monitor object calculates the bandwidth for each session with the payload size of the received data, and the RTCP thread determines the packet transmission interval with the information from the bandwidth monitor object. This object hands an event to the beginning point of the RTCP thread and the RTCP thread begins. At this point on the transmitter's side in the stream object of the RTCP thread calculate corresponding QoS values are transferred to the information manager object. The QoS information manager object maintains 3 tables as shown in the table 1 and is a collection of the functions that store the transferred QoS values to and search the table. The



(Fig. 7) A graphical representation of the RTCP thread operation

〈Table 1〉 Information Base for storing of QoS values

Identifier Table

SSRC	4237
Media_Type	Audio
Payload_Type	G.711
Status	on

User Information Table

Type	Length	Value
CNAME	26	makang@computer.etri.re.kr
NAME	12	MinGyu Kang
EMAIL	26	makang@computer.etri.re.kr
PHONE	15	82-42-860-5966
LOC	34	161 Kajong-Dong, Yusong Gu, Teajon
TOOL	12	GIaNT 1.0
NOTE	25	"on the phone, can't talk"
PRIV	0	

QoS Parameter Table

QoS Parameter	Recent	Max	Min	Mean
Packet loss rate	3.04%	34%	0.41%	5.80%
Interarrival Jitter	18 ms	124 ms	7 ms	24 ms
Round Trip Delay	174 ms	251 ms	107 ms	165 ms
Bandwidth	132.44 kbps	145.46 kbps	[illegible] kbps	129.42kbps
Packet count	23450			
Octet count	84567327 octets			
Fraction packet lost	67.00%			
Cumulative packet lost	143 packets			
LSR	5.250 sec			
DLSR	46553.125 sec			
Highest sequence number	23456			

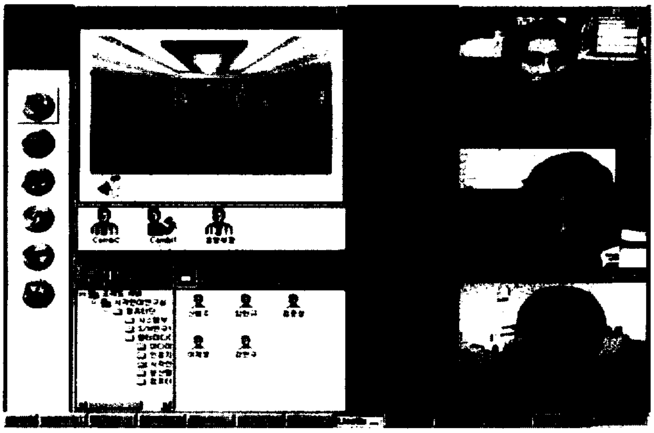
identifier table and the QoS parameter table are filled with QoS values using the RTP packet header and the user information table maintains the personal information of the conference participants using the SDES packet of the compound RTCP packet that arrives periodically. The destination object includes the functions with the socket I/F to transmit and receive the compounded RTCP packets to and from the network.

5. Results and Analysis

5.1 Result of Implementation

As the result of the H.323 conferencing system, as in Figure 8, the chairman and the speaker will appear during the conferencing on the screen. It is due to the codec board can be processed to the maximum 2 streams and your screen is overlaid. It allow users to enter and leave conference as specified by their multicast IP address and port number. The speaker can be frequently changed under the chairman's allowance, as each participant's screen is changed. It's easy operated with convenient visual metaphor for group works, and provide integrated service with T.120 data conferencing as a multipoint group collaboration system. This audiovisual conferencing tool supports a number of codeces, including PCM, GSM, and H.261 for video and supports RTP/RTCP as transport protocol.

In this chapter, we describes the practical results and its analysis which being appear on performing teleconference over the Internet. We have measured the round-trip delay, delay variance and loss of audio packets using RTP/RTCP protocol and audio codecs. The coding schemes of audio packets available at this time use 8-KHz sampled speed and, all the measurements presented in this paper have been obtained with the PCM coder. The PCM coder send periodically audio packets during talkspurts. Recently, Applications over the number of hosts share with the Internet. It is impossible to study the delay and loss characteristics for all possible connection. In this paper, we focus on some specific connection between ETRI in the Teajon and Ajou University and Pusan University. Destination hosts are geometrically distinct, and the physical characteristics of

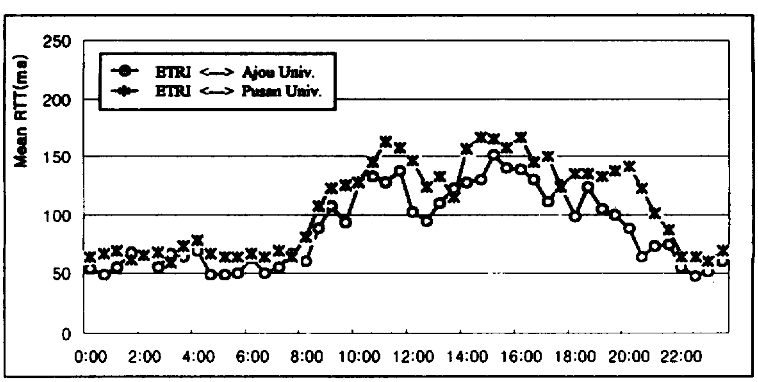


(Fig. 8) User Interface of H.323 Teleconferencing System

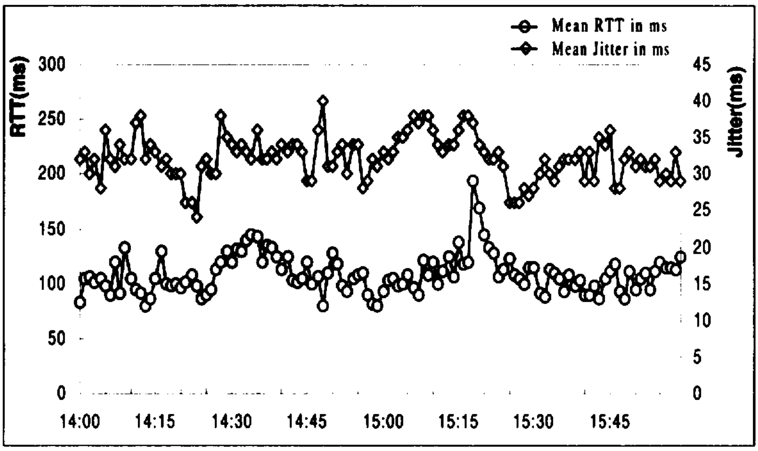
these link are very different. Our tool send audio data, as probe packet, at regular intervals from source host to destination. Throughout the rest of the paper, they provide merely very coarse-grain information. We have found the characteristics of various QoS parameters that being occurred from audio data transmission.

5.2 Analysis of Round-trip delay and Jitter

We will present the results of a study of Internet round-trip delay and variance. Since this link frequently accessed academic hosts as well as research institute hosts, the characteristics of QoS, such as RTT and jitter very difficulties to predict even during short term. We performed the measurement about Internet for a day using the feedback information of RTCP. Round-trip delay extracted from RTCP which defined in chapter 3 is illustrated in Figure 9. This Figure has show that RTT value is specially large between 10:00 am and 12:00 am, and between 2:00 p.m. and 4:00 p.m. By inspection of the graph, we also found that there is small temporal variation in round-trip time, and that the ratio of smallest RTT to largest RTT is similar, RTT distributions change slowly, and sharp changes are undone very shortly. We have observed more detail between 2:00 p.m. to 4:00 p.m. in the Figure 10, when relatively shown high delay. End-to-end one-way transmission time for telecommunication system that described by Recommendation G.114[10], pro-



(Fig. 9) RTT Distribution (for a day)



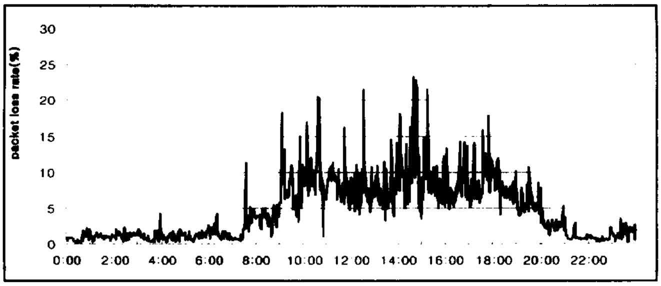
(Fig. 10) RTT & Jitter Distribution (between ETRI and Ajou Univ.)

posed values below 150 ms, as one-way transmission time. Thus, we are considered to be acceptable for most users even during peak period, and to be not needed to consider the delay impact.

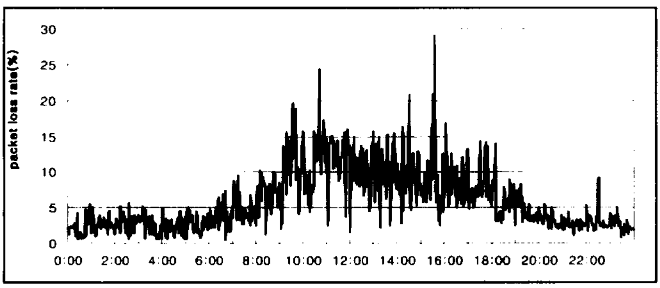
The other measurement was the delay variance. It is defined to be the mean deviation of the difference in packet spacing at the receiver compared to the sender for a pair of packets. If the distortion of interpacket times is large, it need to buffer for smoothness of listen in receiver site. Fortunately, the result measured indicates that there is enough to do not need to minimize the jitter value by following two reasons, and someone who take part in teleconference may be not feel the phenomenon of distortion of voice. The first is that the average of jitter turns out small. The second, since audio packets are experimented of buffering over the playback buffer at front of decoder, as we'd above seen in the Figure 3, the most of all packets with an exception of spikes which is large jitter, did smooth in buffer.

5.3 Analysis of Packet Loss Rate

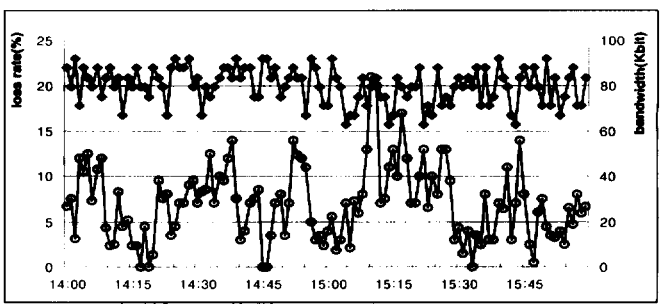
We consider the loss problem of audio data over Internet in this chapter. Internet do not support the real-time property that the distributed multimedia systems (such as teleconferencing system, tele-ducation system, etc.) have inherent. Especially, the UDP/IP protocol stack as under of RTP, do not guarantee the reliable transmission. Therefore, the teleconferencing systems including RTP/RTCP are frequently experience in packet loss, and this lost leads to serious degradation of quality in interactive multimedia communication. Figure 11 and Figure 12 show packet loss rate of links between ETRI and Ajou University, and between ETRI and Pusan university for a day. In these figures, we can find the fact that the phenomenon of packet loss primarily occur during daylight as expected, and tracing of loss by night need not to consider for problems which lead to degradation of QoS. The total number of losses is high between 8:00 am and 8:00 p.m., and the peak of loss process is between 2:00 p.m. and 4:00 p.m., i.e. when the network load is higher in a day. From this reason, we did determine the time interval for investigation. The statistics information that made an observation in Figure 11, 12 during this period, describe in the Table 2. It notes that typical values for loss rate range between 5 % to 15 % during heavy load of these links. In case of the receiver Ajou Univ., it appears that high loss rate period is around 10:00 a.m., 15:00 p.m. and 18:00 p.m., and those hugely contribute to total number of loss very large. Many different quantities can be used to characterize the loss of audio packets, but the



(Fig. 11) Packet Loss Rate Distribution (between ETRI and Ajou Univ.)



(Fig. 12) Packet Loss Rate Distribution (between ETRI and Pusan Univ.)



(Fig. 13) Packet Loss Rate and Bandwidth(between ETRI and Ajou Univ.)

〈Table 2〉 Statistics Value (14:00~16:00)

Destination	Statistics parameter	Packet Loss Rate(%)	Delay (ms)	Jitter (ms)
Ajou Univ. / Pusan Univ.	Mean	6.938 / 9.387	108 / 258.28	32.225 / 38.34
	Standard Dev.	4.141 / 4.405	17.805 / 8.635	3.049 / 2.93
	Max	21 / 28.12	194 / 167	40 / 59
	Min	0.5 / 2.5	78 / 145	24 / 32
	Variance	17.149 / 19.402	317.01 / 74.57	9.302 / 8.123

obvious measure is the average loss. In accordance with high loss rate in daylight, it make degrade the quality of voice, and many participants in teleconference feel inconvenience. In another experiment, we observe the relation between packet loss rate and its bandwidth. It indicate in inspection of Figure 13 that loss rate is independent upon bandwidth of audio data. This means that the major part of occurrence of packet loss is more influenced by background load that is derived from diverse applications being share the Internet than by load with respect to transmit audio packets for examination.

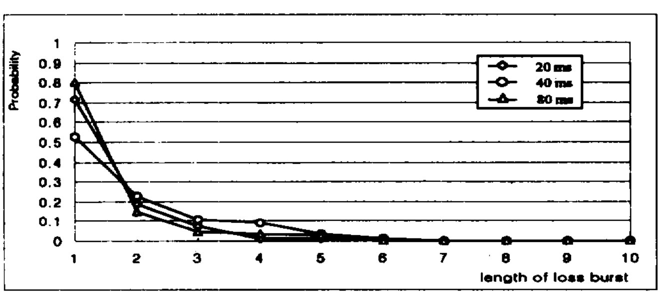
5.4 Analysis of Loss Burstiness

Figure 14 shows the evolution of the number of consecutively lost packet as a function of the sequence number measured 14:00 p.m. to 16:00 p.m. We have examined the loss of audio packets over unicast connection of ETRI - Ajou Univ. It appears that most loss periods involve one or two packets. This observation is confirmed by looking at the probability of loss burst in Figure 14. Hence, we have found that probability distribution of the number of consecutively lost packets is similar to that described above[8]. The observations that we make, are confirmed by looking at the probability distribution of the number of consecutive losses, when the network load is higher. Inspection of the graph shows that the slope of the distribution decreases linearly, the probability distribution decreases geometrically fast away. We also observed that most packet losses are isolated except for "almost periodic" loss periods when the number of consecutive losses occur. The measurements have all be done the PCM coder with 20ms, 40ms, and 80ms packets in order to recognize whether characteristics of burstiness is influence by size of audio packet. However, the probability of bursty remains a few changed according to alteration of packet size. Therefore, the result from this experiment may define as follow.

1) almost number of consecutively lost packets is one or two packets
2) the distribution of packet loss burstiness is approximately geometric.
3) the loss burstiness nearly independent on packet size.

6. Conclusions and the Future Work

In this paper, we described the design and the implementation of the RTP/RTCP for real-time transmission of the stream data and the characteristics of transmission quality-of-service in ITU-T H.323 audio/video conferencing. The RTP operates with audio/video devices and transmits the real-time stream to the network to make conferencing with remote users possible. Generally, interactive multimedia application services such as audio/video conferencing systems suffers from the requirement of QoS provision within the limit of the real-timeness of data transmission. Therefore, ITU-T adopted the RFC 1889 and 1890 so that the QoS can be provided in audio/video stream data transmission in audio/



(Fig. 14) Percentage of Packet Loss

ideo conferencing system. In this paper, the design and the implementation of the RTP/RTCP in Window NT environment are described in detail and we presented the audio/video conferencing system developed as the result and the QoS values of the system. Using audio data transmission over Internet, we also describe the study of measurement and analysis for QoS factors that observed on performing teleconferencing system. In this experiment, we indicate the fact that RTT and Jitter value are acceptable even network load is high, but the packet loss rate is high in daytime and most losses periods have length one or two.

The future work plan includes the development and the application of an improved algorithm of measuring the audio/video transmission quality in the case of H.323 conferencing system on the Internet and will support the QoS guaranteed Teleconferencing System Service even on the network shared by various application services.

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