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Rate-Adaptive Real-Time Multicast TV Conference System with Locally Adaptive Packet Flow Control

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SUMMARY As TCP/IP networks develop, various type of applications or services are appearing. Especially, many people want to use real time multicast applications over TCP/IP networks like a TV conference system. Most of the current TCP/IP networks, however, still do not support QoS guarantees using RSVP, so that they provide only a best-effort service. Therefore, such real time applications must control data transmitting rate by the network or receiver's condition. However, it is difficult to control data rate over a multicast session, since every receiver on a multicast network does not necessarily have the same environment. To solve this problem, the authors proposed a locally adaptive control intermediate system. This paper describes a rate-adaptive real-time multicast system with locally adaptive packet flow control.
key words: *multicast, real-time, RTP*

1. Introduction

Currently, as the Internet develops, several novel application services are appearing. WWW services provide not only text but also multimedia information, so it is natural for users to want to transmit and receive multimedia data, voice or movies, in real-time.

Applications such as ftp, telnet or WWW using usual file transfer services demand only one "Quality of Service (QoS)"; that is, a reliable data transfer service. Unlike these, applications that transmit a large amount of real-time data (voice or movie) need other types of QoS. In some cases, it will be guaranteed bandwidth, in another case it will be maximum delay time. There are two methods to provide QoS for application demands, i. e. first, the network may provide enough QoS on demand by an application, and second, an application may adaptively control the data transmitting rate as is flow control to obtain the minimum QoS or not affect other traffic. The former is called an "application-initiated system" and the latter is called a "network-initiated system" [1].

In general, the current Internet can provide only one QoS, a best-effort data transfer service, but recently, IETF is standardizing RSVP (Resource reSerVation setup Protocol) [2] as a new protocol to guarantee QoS. RSVP is a multicast-oriented protocol, so it is suitable for a real-time multicast application over TCP/IP networks such as a TV conference system. However, RSVP is no more than a signaling protocol to carry resource request messages, so every node over the path between end nodes must have some mechanism to actually reserve resources. For example, each node needs traffic control modules to reserve bandwidth, but now, because of the difficulties concerned with policy control and accounting, RSVP is expected to function as a protocol to support next generation services and to be used within only limited networks (for example, private networks).

Therefore, over the current TCP/IP networks like the Internet, which provide only a best effort service, an application should be a network-initiated system. Many applications, such as nv [3] and vic [4], that provide video transmission over the IP multicast backbone were developed. In addition, experiences with rate control mechanism using RTP/RTCP [5] were reported [6]-[9]. However, under multicast networks, it is very difficult for a source to perform control rate-adaptively by feedback from receivers, since receivers are not always in the same network environment. To transmit multicast data efficiently in such a heterogeneous network, Heterogeneous MultiCast (HMC)[10], Receiver-driven Layered Multicast (RLM) [11] or a video gateway [12] were proposed.

In this paper, we report a rate-adaptive real-time multicast system with a locally adaptive control RTP mixer/translator. Section 2 describes real-time data transmission over TCP/IP, and the problem of real-time multicast transmission over TCP/IP and related matters. In Sect. 3, we propose a new architecture to perform control rate-adaptively in a real-time multicast environment. Section 4 and Sect. 5 give a simulation-based evaluation of this architecture and related work. Section 6 concludes the paper.

2. Real-Time Data Transmission Over TCP/IP Networks

2.1 Real-Time Data Communication Using RTP

IETF (Internet Engineering Task Force) is standardizing RTP (Realtime Transport Protocol) for real-time data communication. Unlike RSVP or other reservation protocols, RTP does not guarantee any QoS. RTP carries data that has real-time properties, and a timestamp, sequence number, etc. Also, RTP control protocol (RTCP) monitors the QoS and conveys information about the participants in a session between senders and receivers. With RTP/RTCP, therefore, applications can perform a network-initiated QoS

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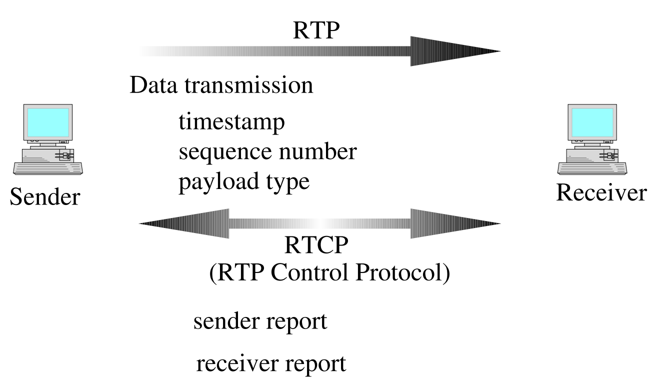
control. For example, an application can estimate its network status from receivers' reports and when it detects congestion, it controls its data transmitting rate (Fig. 1).

The authors developed a dynamically rate-adaptive TV conference system with RTP/RTCP and showed that feedback rate control with RTP/RTCP like this can work well in point-to-point communications [9].

2.2 The Problem of Rate-Adaptive Real-Time Multicast Communication

It is difficult to extend the feedback control method with RTP/RTCP proposed in [4] to real-time multicast communication. The reason is that a multicast environment is generally heterogeneous; that is, the network environment of each receiver is not always the same. Some users may access the network via a ten kbps telephone line with modems, and others may use hundreds of kbps bandwidth via fast LANs. On the other hand, in a network that has many users, users may hardly use any bandwidth because of the network congestion.

Considering network-initiated control with RTP/RTCP in a multicast environment, if a sender receives a RTCP receiver report from one receiver and estimates the network congestion of that receiver, the sender, however, may estimate no congestion from receiver reports from other receivers. In this case, the sender will be unable to decide whether he should decrease the data-transmitting rate (Fig. 2). If the sender decides to decrease the rate on the demand



RTP itself cannot guarantee any QoS.

Fig. 1 Control with RTP/RTCP.

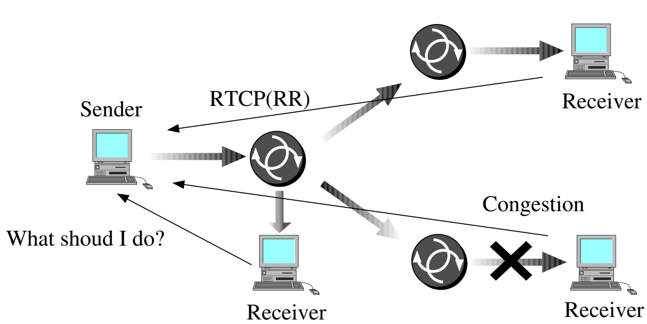


Fig. 2 The problem under a simple rate-adaptive multicast.

of the receiver who lives in a congested network, other receivers who live in a "silent" network have to receive low-quality data although they have enough resources to receive higher quality data. Conversely, if he decides to continue to send data at the same rate at the sacrifice of one receiver, the congestion in his network becomes worse.

3. Rate-Adaptive Real-Time Multicast System with Locally Adaptive RTP Mixer/Translator

3.1 Locally Adaptive Control RTP Mixer/Translator

RTP defines a mixer and a translator as intermediate nodes. Some kinds of mixer or translator, for example, can decode received data and re-encode changing the coding rate, so that they can send data at an appropriate transmitting rate per receiver. By this definition, however, an intermediate node itself does not change the transmitting rate depending on its network status, so it must have knowledge about the environment of all receivers. Further TCP/IP networks cannot guarantee a constant network status, so a mixer/translator can not control local fluctuations of TCP/IP network traffic.

When an application is transmitting real-time data with a UDP that has no flow control mechanism, it has to control congestion in order to prevent a network from becoming worse or increasing packet loss. In the case of using an ordinary mixer/translator, only the data source dynamically controls congestion, so the data-transmitting rate to all multicast receivers decreases because of congestion in a partial network. Here, by extension of this ordinary mixer/translator, we propose a Local Adaptive Control RTP Mixer/translator (LACRM) which controls data transmitting rate according to the local congestion of networks when it relays data from a data source. Using a LACRM, an application can transmit data at an appropriate transmitting rate in response to each receiver's status. A LACRM is placed in each local network, and RTCP receiver report (RR) packets from receivers are acquired by the LACRM. A LACRM sends its own receiver report to a sender or a parent LACRM (Fig. 3).

When a LACRM detects congestion in a local network based on the packet loss rate in RTCP RR messages, it decreases the data-transmitting rate to the congested network. In Fig. 3, for example, congestion occurs in the network between LACRM A and C, so A decreases the data transmitting rate to C (Fig. 4). However, LACRM A does not report the local congestion information as a receiver report, so networks beyond B can communicate at the same rate as before.

3.2 Multicast Routing

We can select two types of behavior of LACRM when a LACRM relays data as follows. 1) router mode : A LACRM sends data to the session with the same multicast IP address and port number as received data. 2) gateway mode : A LACRM sends data to the session with a different multicast

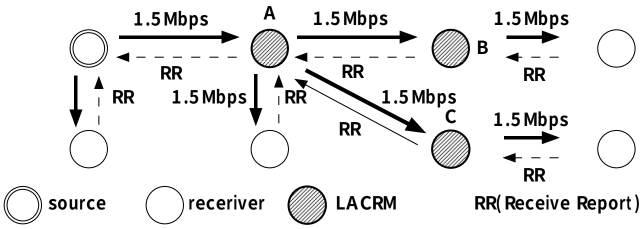


Fig. 3 Abstract of LACRM (before congestion).

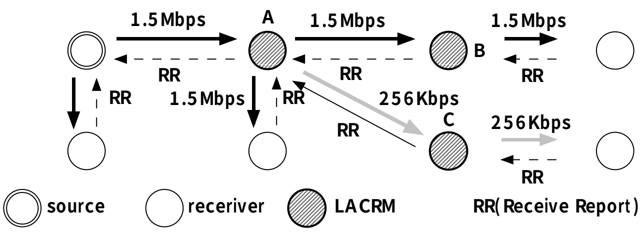


Fig. 4 Abstract of LACRM (after congestion).

IP address or port number from the received data. In router mode, an application uses only two multicast sessions as RTP and RTCP for one data, but needs to interact with its routing process in order to send data only to its subordinate networks within the same multicast network selectively. On the other hand, in gateway mode, the number of sessions that an application needs to communicate with RTP/RTCP is (the number of its subordinate networks) times two. This means that the processing load at a LACRM increases depending on the number of sessions but a LACRM can work independently of the multicast routing by using unique multicast sessions for each subordinates network. Therefore, a LACRM can coexist with current routers so that it is easy to shift.

3.3 Packet Flow Control

In order to control the data-transmitting rate at intermediate nodes, it is necessary to decode once and re-encode at an appropriate rate for each destination network. However in general, this process causes an intermediate node to take a very heavy load, so that the delay time between a sender and a receiver increases and this delay time causes a problem in real-time systems. One solution to this problem is as follows. First, the information to transmit is classified according to its priority with layered coding. Next, this classified information is encapsulated into packets and the priority is included within the packet header. Finally, when the network is congested, the priority within a header is checked and the packets whose priority is low are discarded. Hence, only the processing header can control the data-transmitting rate efficiently.

In our implementation as one example, we used MPEG-1 based layered encoding [13]. Each type of MPEG-1 frame is divided into two subtypes, low resolution information and enhancement information, so there exists six types of frames I_L, I_H, P_L, P_H, B_L and B_H. The subscript L means low-resolution information and H means enhancement information. Table 1 shows the relationship of each frame type.

Table 1 Relationship between frames.

	Previous	Next
I_L	N/A	N/A
I_H	I_L	N/A
P_L	P_L or I_L	N/A
P_H	P_L	N/A
B_L	P_L or I_L	P_L or I_L
B_H	B_L and P_H or I_H	P_H or I_H

I_L does not need information from other frames, but to decode the B_H frame, the previous B_L frame, previous I_H or P_H frame, next B_L frame, and next I_H or P_H frame are needed. Therefore, we can control data transmitting rate at 6-levels. If we consider that spatial information is more important than temporal information, the priority becomes $I_L > I_H > P_L > P_H > B_L > B_H$, otherwise, $I_L > P_L > B_L > I_H > P_H > B_H$. Also if the identifier of these frame types is included within a packet header, an application and a LACRM can control the rate by packet header processing alone.

3.4 Congestion Detection and Recovery

In this system, a data source detects network congestion by monitoring packet losses at each receiver as with TCP. However, a data source cannot distinguish losses due to congestion from ones due to temporal lowering of line quality. On the other hand, as a method to recover the data-transmitting rate, we can envisage a slow-start rate recovery, but these are not always appropriate solutions. As a future topic, we will consider not only packet losses but also fluctuation of jitter as an indication of network congestion, since accumulating packets in queues of intermediate systems during congestion increases delay time.

3.5 Real-Time Socket Interface

The authors considered RTP as a transport protocol for real-time communication and proposed an extended socket interface with SOCK-REALTIME [14]. This interface lowers the cost of developing a real-time application. The LACRM needs to be implemented for each application, and we developed our system with this interface for wide use. This section simply describes this real-time socket interface.

In general, RTP uses a service of an under layer protocol (UDP, etc.) to identify RTP/RTCP or to identify multi-sessions. RTP is implemented on UDP in this interface. Figure 5 shows the protocol stack of this interface.

An application transmits data, parameters and receiver reports etc. via this real-time socket interface. Adding to TCP/UDP parameters, an application must set RTP parameters, for example CNAME, when using this real-time socket interface. For example, with Windows Sockets 2 API [15] or RAPI [2], an application can set QoS parameters

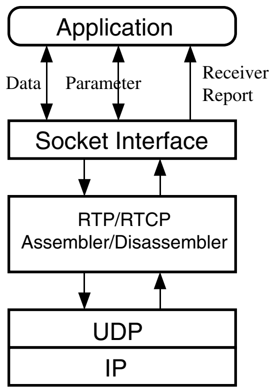


Fig. 5 Protocol stack of real-time socket interface.

based on RFC1363 but cannot set RTP specific parameters like CNAME or RTCP interval. In this interface, to provide a uniform interface, we extended the SETSOCKOPT call, which is the standard function of the UNIX/socket interface. With this new SETSOCKOPT call, an application can set RTP parameters. The real-time socket interface, then, includes this information in RTP/RTCP packets, encapsulates UDP packets, and sends them to a network.

4. Evaluation of LACRM Using Simulation Models

In this section, we evaluate LACRM using simulation models. We used Alta Group BONeS DESIGNER [16] as a simulation platform. Figure 6 shows a real-time multicast environment without LACRM and Fig. 7 shows one with LACRM.

In Figs. 6 and 7, the video frame generator generates layer-encoded pseudo-frames, whose type is I_L, I_H, P_L, P_H, B_L or B_H. The video frames are generated periodically (30 frames per second) in a fixed order and the length of each type of frame has a normal distribution with each mean and variance based on real movie data [13]. Table 2 shows means and variances of these types of frame length [13].

The video frame receiver receives and assembles these pseudo-frames. Each receiver monitors the sequence number of packets, calculates packet losses, and reports packet losses to a sender with RTCP RR packets. A receiver sends RTCP RR packets when it receives RTP data packets. Except for I_L frames, if a frame that is used for other participating frames is lost, the frames generated using that frame are discarded. In each environment, there are two video frame receivers via a 2Mbps serial line. To make the analysis simpler, in this model we considered that the transmission delay times over every serial line are all zero. The network that has receiver B has a Jamming generator that generates other traffic. As general traffic, the jamming traffic consists of packets based on a normal distribution packet length and Poisson distribution packet interval. Considering the default MTU length over unknown networks, the mean length is 500 [bytes], and the variance of length is 100000. To make the mean bandwidth used by the jamming traffic 2 [Mbps], the mean packet interval is 0.002 [sec]. In this environment, we measured used bandwidth for each receiver and the number of frames that each receiver could assemble. Here, we considered three cases. First, a video frame generator sends a multicast packet to receiver A and B, and does not control data transmitting rate. Second, a video frame generator controls data transmitting rate depending on the status of either receiver without LACRM. Third, a video frame generator controls data transmitting rate depending on the status of a LACRM, and a LACRM controls rates according to each receiver. The first case uses a Fig. 6 configuration, and the second case also uses a Fig. 6 configuration. The third case uses a Fig. 7 configuration. In each case, a video frame generator sends data for a hundred seconds. A jamming generator generates jamming traffic from 30 [sec] to 50 [sec]. These results are shown in Figs. 8, 9 and 10.

Figure 8 shows the results using the second environment, and Fig. 9 shows the results using the third environment. In each figure, the horizontal axis means the

Table 2 Mean and variance of each type of frame.

	Mean [bits]	Variance
I_L	56095	34683223
I_H	76247	254603849
P_L	28807	28573353
P_H	41618	111231894
B_L	20728	56043053
B_H	12109	23945851

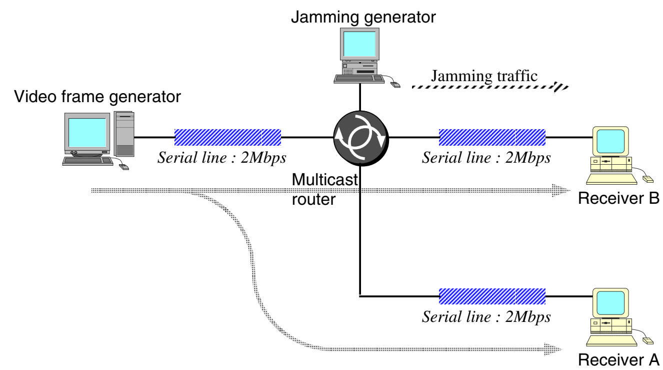


Fig. 6 Experimental environment without a LACRM.

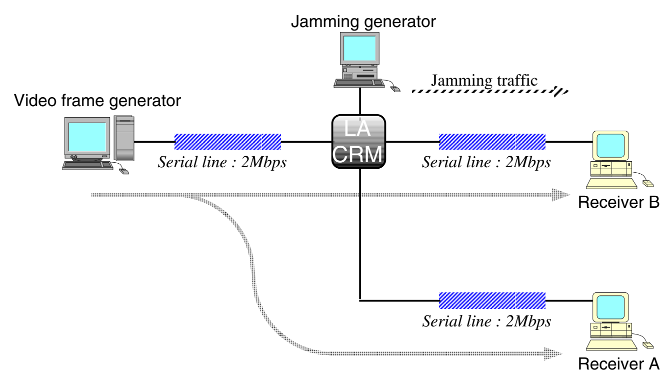


Fig. 7 Experimental environment with a LACRM.

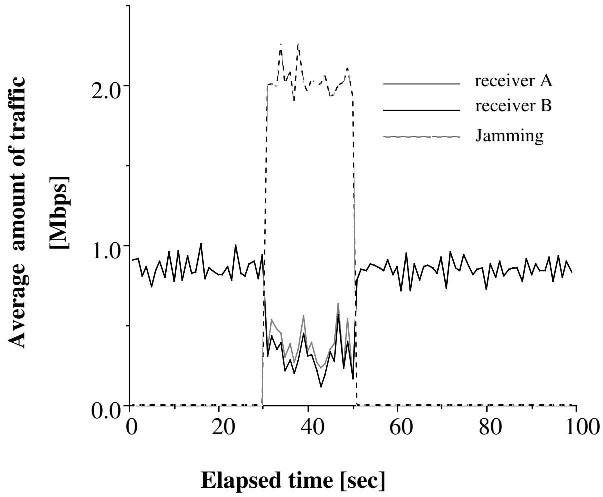


Fig. 8 Average amount of traffic (usual multicast).

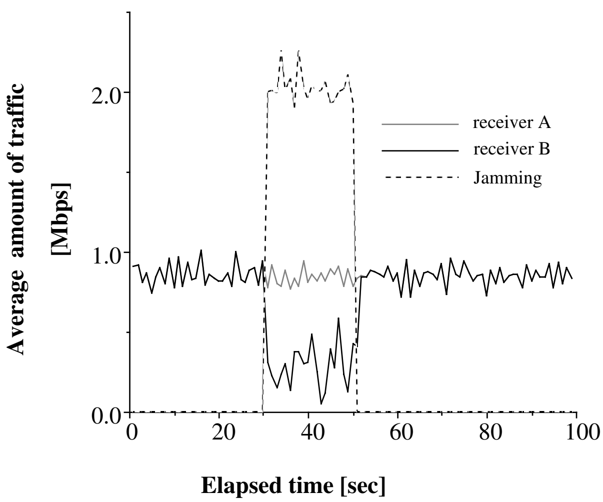


Fig. 9 Average amount of traffic (with LACRM).

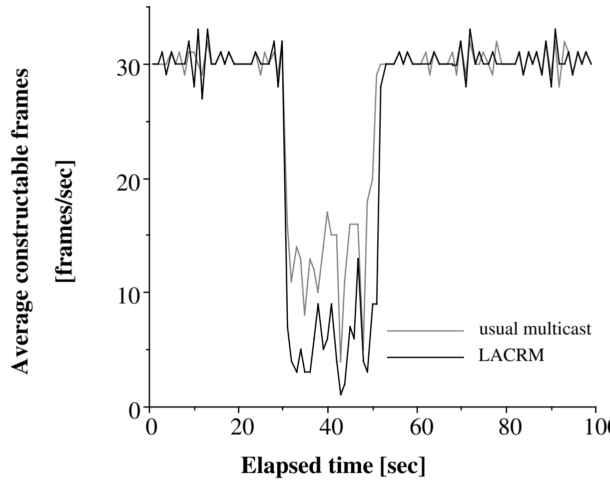


Fig. 10 Average constructable frames at receiver B.

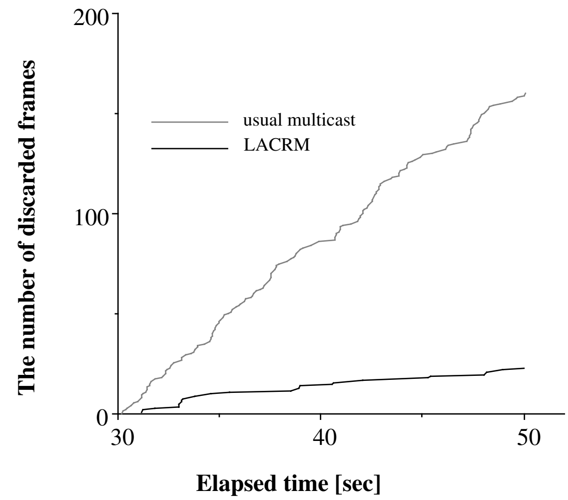


Fig. 11 Number of discarded frames at receiver B.

elapsed time [sec]. In Figs. 8 and 9, the vertical axis means average amount of traffic per second [Mbps]. From these results, it is seen that, with a LACRM, each receiver could acquire data at appropriate rates.

Figure 10 shows the average number of frames that were received by a receiver and also able to be decoded in the receiver per second. Figure 11 shows the number of total unconstructable frames that were discarded in the receiver during congested periods. Before and after congested periods, no frames were discarded. This figure includes results of the first, second and third cases.

From Fig. 10, the average frame rate under a usual multicast is larger than the one with a LACRM during congestion. At a glance, a LACRM seems to be unnecessary. This was caused by inappropriate congestion control. However, regarding the number of frames discarded, the number for a usual multicast for exceeds the one with a LACRM. This means the following. Without an LACRM, a large amount of data was transmitted even when the network was congested. This makes the network worse and may disrupt the network.

In this model, we used RTP data packets receives as RTCP RR transmit triggers. In general, RTCP RR packets are transmitted periodically and the available bandwidth and the number of receivers define the interval. Therefore as the number of receivers increases, the interval increases. This large interval dulls the control, however the system using LACRMs is controlled locally, so the number of receivers which are assigned one LACRM will not become very large.

From these results, it was found that a LACRM could control transmitting rate locally, and this makes total performance under a real-time multicast session more efficient.

5. Related Work

In this section, we give a comparison with other related work.

5.1 Reliable Multicast Protocols

The current TCP/IP protocol suites support two transport layer protocols, i. e., TCP and UDP. TCP can provide congestion avoidance and control [17], but cannot support multicast. Many TCP-like reliable multicast protocols have been proposed [18], [19]. However, in order to guarantee error-free data transmission, TCP and TCP-like protocols use acknowledge-based retransmission. This may cause unexpected delay time, so TCP and many other reliable multicast protocols are not adequate to transfer data in real time. Therefore, almost all applications that perform real time multicast transmission use UDP. UDP supports multicast data transmission but does not have any flow control mechanism.

5.2 HMC

To transmit multicast data efficiently in a heterogeneous network, which has many users in various types of environments, Heterogeneous MultiCast (HMC) was proposed [5]. HMC uses layered encoding and transmits each layered data with an individual multicast session. A receiver selects necessary sessions according to its environment so that every receiver can acquire data appropriately. The HMC adapts to a heterogeneous receiver environment statically, so HMC works very well under networks that guarantee QoS like a bandwidth guarantee for every receiver with ATM or RSVP, that is, networks where applications can perform application-initiate control. Therefore, HMC cannot adapt to networks that provide only best effort service like the Internet; that is, applications must perform network-initiated control.

5.3 RLM

To control rate adaptively under a real-time multicast environment, Receiver-driven Layered Multicast (RLM) was proposed [6]. Like HMC, RLM uses layered encoding and transmits each layered data with an individual multicast session. In RLM, receivers join and leave the multicast sessions according to their network environment or status, so that the amount of traffic can be controlled. When no receivers who join a session which sends a layer exist further than some router, the path beyond that router will be deleted by a multicast routing protocol, disused traffic will not flow and congestion will be avoided. In RLM, a sender or an intermediate system does not detect and control congestion but receivers do. RLM proposes shared learning when the number of receivers increases; however, as the number of receivers increases, the behavior is thought to be unstable. In addition, as a common point with HMC, both senders and receivers must manage a large number of (RTP/RTCP) sessions in proportion to the number of layers. Moreover, it is difficult for many multicast sessions to be synchronized.

5.4 Video Gateways

The video gateway proposed in [12] connects pairs of RTP multicast sessions transparently so that they are identified one logically same conference. A video gateway can perform bandwidth adaptation over a heterogeneous multicast network by transcoding multiple video formats and rate-control, and manipulating RTCP packets when it transmits data from one session to another session. In addition, it can connect between sessions using a same coding scheme. As one prototype, a video gateway transcoding Motion-JPEG to Intra-H.261 was developed.

A video gateway looks like our system, but it is an application-level gateway, so it needs some multicast sessions to perform bandwidth adaptation over a heterogeneous network. In general, transcoding multiple video formats may become a heavy load for video gateways. Therefore, high CPU performance is required to handle many sessions at a video gateway.

6. Conclusion

In this paper, we proposed a rate-adaptive real-time multicast system with LACRM and showed the effectiveness of a LACRM using simulation. We have already implemented a video frame generator and receiver on Windows95/NT, and now we are implementing a LACRM on FreeBSD and Windows95/NT.

In future work, we should consider some other things. First, we need more consideration of congestion detection, avoidance and recovery. Second, we should consider how to initially join sessions, that is, which LACRM a receiver should access when he wants to join a certain session. This needs a session control protocol. Third is implementation using other layered or hierarchical encoding systems. In this implementation, we used a real-time socket interface that we developed, and this interface make the cost of implementing a LACRM for other encoding low. Fourth is the placement of a LACRM. The number of LACRMs and their appropriate placement to communicate efficiently depends on the scale and the environment of the networks. We will evaluate these using our implementation.

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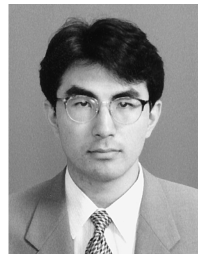

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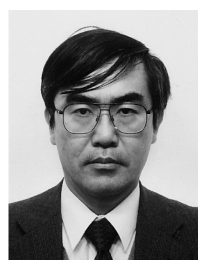

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