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Study Group 15
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TITLE: Real-Time Transport Protocol in H.32Z Systems in ATM and Other Packet-Switched Computer Networks
PURPOSE: Discussion

1 Introduction

In this documents the following points are considered:

- Real-Time Transport Protocol (RTP)
- Timing Model without bandwidth reservation
- Packet routing in packet-switched computer networks
- Multicast addresses for data transportation

2 Real-Time Transport Protocol

In the Internet community, IETF's Audio/Visual Transport Working Group is defining a Real-Time Transport Protocol (RTP) that provides sufficient means of clock recovery when network clock is available [1]. This document relates to RTP version 2 where some of the concepts are changed from version 1. Several security, clock recovery, media synchronization and bandwidth considerations have been taken into account when designing the RTP [2].

2.1 RTP Overview

RTP provides end-to-end network transport functions suitable for applications transmitting real-time data, such as audio, video or simulation data over multicast or unicast network services.

RTP does not address resource reservation and does not guarantee quality-of-service for real-time services. The data transport is augmented by a control protocol (RTCP) designed to provide minimal control and identification functionality, particularly in multicast networks. Within multicast associations, sites can also direct control messages to individual sites. RTP and RTCP are designed to be independent of the underlying transport and network layers. The protocol supports the use of RTP-level translators and bridges.

2.2 RTP Status

RTP is currently at working draft stage. However, recent modifications have made RTP more workable. In case ITU shows interest in RTP, it should also take measures to affect the work done by IETF.

3 RTP and H.320 Systems

Clock recovery is one of the biggest problems for real-time multimedia services. The problem can be solved using a simple and efficient transport protocol.

RTP version 1 has been widely used in the Multicast Backbone (MBONE), an international virtual multimedia network which is based on regular Internet connections and multicast addresses. Therefore, if H.32Z supports RTP it is expected that H.32Z would gain better acceptance and wider user base. This precisely due to the fact that it is independent of lower layer protocols, and can thus be used as a common working protocol between various packet-switched computer networks (e.g. ATM, ATM-LAN, Ethernet, ISO-Ethernet) and protocols (e.g. UDP, TCP, IP, IPX).

4 RTP Timing Model

The timing model in RTP version 2 is based on a media specific time stamp. The nominal clock frequency of the media time stamp is transferred using the control protocol (RTCP). The sender also sends periodically timing relations between the sample clock and network clock using RTCP. The receiver may decide to follow only sender nominal clock frequency or to synchronize the playback to the network clock. Where inter-media synchronization between different media types is required, the receiver synchronizes the nominal clock frequency and the network clock periodically based on the senders time messages.

Each sender and receiver sends periodically reports on the corresponding control channel containing information on how well the transmission is received. From these reports senders and receivers can calculate the quality of the channel (e.g. round-trip delay, packet loss rate).

In the case of H.320 systems, the different medias can be transferred separately and each of the streams would be synchronized to the network clock. The H.320 data channel could be used as the base communication channel for parameter negotiation and transportation reports.

This method requires that network clock is available to both ends of the transmission. The RTP timing model does not require bandwidth reservation nor fixed delay network.

5 Backchannel and Signaling

Each RTP data connection has a bidirectional backchannel associated with it for control purposes. In fact, the aforementioned timing information is carried in this control channel.

RTP allows the connection to specify service-dependent control packets. These packets can be defined to carry call setup, and other signaling needed in a call.

6 Routing Principles in ATM and UDP/IP

RTP does not handle routing and multiplexing. Therefore, when crossing between ISDN and computer networks, there should be a mapping between the ISDN-address (phone-number) and the computer network address (e.g. IP address together with UDP port number or VCI/VPI on ATM).

Routing in computer networks is based on the address of the receiver. If unicast addresses are used, all data will be routed to the receiver whether it wants to only use audio data or both audio or video, because the routers cannot distinguish between different streams for one receiver. If multicast addresses are used, media streams can be split on different addresses and selective routing is possible. Also multicast addresses greatly reduce the network load in a case where several receivers share a common transport link.

However, multicast addresses affect seriously the design of transport protocol because there may be many receivers for each stream. The quality of service parameters can be very different to each receiver. Also the security issues become more important because the data is easily available to everybody between the sender and receiver. The transport protocol should be defined so that it provides the best possible quality to each receiver, which is one of the most important design goals for RTP.

RTP has been designed to work with both unicast and multicast addresses and the selective routing and multicast has been effectively used within MBONE.

7 Bandwidth Reservation

There is no guaranteed bandwidth available in MBONE. The reservation mechanism for global computer network, if available, would had to be very complex due to, e.g., large number of users, tariffing, fairness and multicast traffic.

The only necessary requirement for the network in MBONE system is that the network must be able to handle the average data rate. In MBONE system, the receiver can factor out how much time has passed while the data traveled from the sender by examining the time stamp provided by RTP. The receiver has to accommodate enough buffering to compensate for the variations in network delay.

If H.32Z system is only defined for bandwidth guaranteed networks, it cannot be used on current computer networks which do not provide bandwidth reservation. Also the bandwidth guaranteed networks are very similar to ISDN situation and the regular H.320 timing model can probably be extended for the slightly bigger jitter present in these networks.

Wouldn't it be better to have the audio-visual services with possibly slightly worse quality available to installed base of computers rather than no service at all?

References

[1] Schulzrinne H., Casner S. "RTP: A Protocol for Real-Time Applications", IETF working draft, draft-ietf-avt-rtp-05.txt

[2] Schulzrinne H. "Issues in Designing a Transport Protocol for Audio and Video Conferences and Multiparticipant Real-Time Applications", IETF working draft, October 1993