

Network Working Group  
Request for Comments: 1889  
Category: Standards Track

Audio-Video Transport Working Group  
H. Schulzrinne  
GMD Fokus  
S. Casner  
Precept Software, Inc.  
R. Frederick  
Xerox Palo Alto Research Center  
V. Jacobson  
Lawrence Berkeley National Laboratory  
January 1996

RTP: A Transport Protocol for Real-Time Applications

Status of this Memo

This document specifies an Internet standards track protocol for the Internet community, and requests discussion and suggestions for improvements. Please refer to the current edition of the "Internet Official Protocol Standards" (STD 1) for the standardization state and status of this protocol. Distribution of this memo is unlimited.

Abstract

This memorandum describes RTP, the real-time transport protocol. 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) to allow monitoring of the data delivery in a manner scalable to large multicast networks, and to provide minimal control and identification functionality. 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 mixers.

Table of Contents

1.	Introduction .....	3
2.	RTP Use Scenarios .....	5
2.1	Simple Multicast Audio Conference .....	5
2.2	Audio and Video Conference .....	6
2.3	Mixers and Translators .....	6
3.	Definitions .....	7
4.	Byte Order, Alignment, and Time Format .....	9
5.	RTP Data Transfer Protocol .....	10
5.1	RTP Fixed Header Fields .....	10
5.2	Multiplexing RTP Sessions .....	13

Schulzrinne, et al                      Standards Track                      [Page 1]  
RFC 1889                                      RTP                                      January 1996

5.3	Profile-Specific Modifications to the RTP Header.....	14
5.3.1	RTP Header Extension .....	14
6.	RTP Control Protocol -- RTCP .....	15

6.1	RTCP Packet Format .....	17
6.2	RTCP Transmission Interval .....	19
6.2.1	Maintaining the number of session members .....	21
6.2.2	Allocation of source description bandwidth .....	21
6.3	Sender and Receiver Reports .....	22
6.3.1	SR: Sender report RTCP packet .....	23
6.3.2	RR: Receiver report RTCP packet .....	28
6.3.3	Extending the sender and receiver reports .....	29
6.3.4	Analyzing sender and receiver reports .....	29
6.4	SDES: Source description RTCP packet .....	31
6.4.1	CNAME: Canonical end-point identifier SDES item .....	32
6.4.2	NAME: User name SDES item .....	34
6.4.3	EMAIL: Electronic mail address SDES item .....	34
6.4.4	PHONE: Phone number SDES item .....	34
6.4.5	LOC: Geographic user location SDES item .....	35
6.4.6	TOOL: Application or tool name SDES item .....	35
6.4.7	NOTE: Notice/status SDES item .....	35
6.4.8	PRIV: Private extensions SDES item .....	36
6.5	BYE: Goodbye RTCP packet .....	37
6.6	APP: Application-defined RTCP packet .....	38
7.	RTP Translators and Mixers .....	39
7.1	General Description .....	39
7.2	RTCP Processing in Translators .....	41
7.3	RTCP Processing in Mixers .....	43
7.4	Cascaded Mixers .....	44
8.	SSRC Identifier Allocation and Use .....	44
8.1	Probability of Collision .....	44
8.2	Collision Resolution and Loop Detection .....	45
9.	Security .....	49
9.1	Confidentiality .....	49
9.2	Authentication and Message Integrity .....	50
10.	RTP over Network and Transport Protocols .....	51
11.	Summary of Protocol Constants .....	51
11.1	RTCP packet types .....	52
11.2	SDES types .....	52
12.	RTP Profiles and Payload Format Specifications .....	53
A.	Algorithms .....	56
A.1	RTP Data Header Validity Checks .....	59
A.2	RTCP Header Validity Checks .....	63
A.3	Determining the Number of RTP Packets Expected and Lost .....	63
A.4	Generating SDES RTCP Packets .....	64
A.5	Parsing RTCP SDES Packets .....	65
A.6	Generating a Random 32-bit Identifier .....	66
A.7	Computing the RTCP Transmission Interval .....	68

A.8	Estimating the Interarrival Jitter .....	71
B.	Security Considerations .....	72
C.	Addresses of Authors .....	72
D.	Bibliography .....	73

1. Introduction

This memorandum specifies the real-time transport protocol (RTP), which provides end-to-end delivery services for data with real-time characteristics, such as interactive audio and video. Those services include payload type identification, sequence numbering, timestamping and delivery monitoring. Applications typically run RTP on top of UDP to make use of its multiplexing and checksum services; both protocols

contribute parts of the transport protocol functionality. However, RTP may be used with other suitable underlying network or transport protocols (see Section 10). RTP supports data transfer to multiple destinations using multicast distribution if provided by the underlying network.

Note that RTP itself does not provide any mechanism to ensure timely delivery or provide other quality-of-service guarantees, but relies on lower-layer services to do so. It does not guarantee delivery or prevent out-of-order delivery, nor does it assume that the underlying network is reliable and delivers packets in sequence. The sequence numbers included in RTP allow the receiver to reconstruct the sender's packet sequence, but sequence numbers might also be used to determine the proper location of a packet, for example in video decoding, without necessarily decoding packets in sequence.

While RTP is primarily designed to satisfy the needs of multi-participant multimedia conferences, it is not limited to that particular application. Storage of continuous data, interactive distributed simulation, active badge, and control and measurement applications may also find RTP applicable.

This document defines RTP, consisting of two closely-linked parts:

- o the real-time transport protocol (RTP), to carry data that has real-time properties.
- o the RTP control protocol (RTCP), to monitor the quality of service and to convey information about the participants in an on-going session. The latter aspect of RTCP may be sufficient for "loosely controlled" sessions, i.e., where there is no explicit membership control and set-up, but it is not necessarily intended to support all of an application's control communication requirements. This functionality may be fully or partially subsumed by a separate session control protocol,

which is beyond the scope of this document.

RTP represents a new style of protocol following the principles of application level framing and integrated layer processing proposed by Clark and Tennenhouse [1]. That is, RTP is intended to be malleable to provide the information required by a particular application and will often be integrated into the application processing rather than being implemented as a separate layer. RTP is a protocol framework that is deliberately not complete. This document specifies those functions expected to be common across all the applications for which RTP would be appropriate. Unlike conventional protocols in which additional functions might be accommodated by making the protocol more general or by adding an option mechanism that would require parsing, RTP is intended to be tailored through modifications and/or additions to the headers as needed. Examples are given in Sections 5.3 and 6.3.3.

Therefore, in addition to this document, a complete specification of RTP for a particular application will require one or more companion documents (see Section 12):

- o a profile specification document, which defines a set of payload type codes and their mapping to payload formats (e.g.,

media encodings). A profile may also define extensions or modifications to RTP that are specific to a particular class of applications. Typically an application will operate under only one profile. A profile for audio and video data may be found in the companion RFC TBD.

- o payload format specification documents, which define how a particular payload, such as an audio or video encoding, is to be carried in RTP.

A discussion of real-time services and algorithms for their implementation as well as background discussion on some of the RTP design decisions can be found in [2].

Several RTP applications, both experimental and commercial, have already been implemented from draft specifications. These applications include audio and video tools along with diagnostic tools such as traffic monitors. Users of these tools number in the thousands. However, the current Internet cannot yet support the full potential demand for real-time services. High-bandwidth services using RTP, such as video, can potentially seriously degrade the quality of service of other network services. Thus, implementors should take appropriate precautions to limit accidental bandwidth usage. Application documentation should clearly outline the limitations and possible operational impact of high-bandwidth real-

time services on the Internet and other network services.

## 2. RTP Use Scenarios

The following sections describe some aspects of the use of RTP. The examples were chosen to illustrate the basic operation of applications using RTP, not to limit what RTP may be used for. In these examples, RTP is carried on top of IP and UDP, and follows the conventions established by the profile for audio and video specified in the companion Internet-Draft draft-ietf-avt-profile

### 2.1 Simple Multicast Audio Conference

A working group of the IETF meets to discuss the latest protocol draft, using the IP multicast services of the Internet for voice communications. Through some allocation mechanism the working group chair obtains a multicast group address and pair of ports. One port is used for audio data, and the other is used for control (RTCP) packets. This address and port information is distributed to the intended participants. If privacy is desired, the data and control packets may be encrypted as specified in Section 9.1, in which case an encryption key must also be generated and distributed. The exact details of these allocation and distribution mechanisms are beyond the scope of RTP.

The audio conferencing application used by each conference participant sends audio data in small chunks of, say, 20 ms duration. Each chunk of audio data is preceded by an RTP header; RTP header and data are in turn contained in a UDP packet. The RTP header indicates what type of audio encoding (such as PCM, ADPCM or LPC) is contained in each packet so that senders can change the encoding during a conference, for example, to accommodate a new participant that is connected through a low-bandwidth link or react to indications of

network congestion.

The Internet, like other packet networks, occasionally loses and reorders packets and delays them by variable amounts of time. To cope with these impairments, the RTP header contains timing information and a sequence number that allow the receivers to reconstruct the timing produced by the source, so that in this example, chunks of audio are contiguously played out the speaker every 20 ms. This timing reconstruction is performed separately for each source of RTP packets in the conference. The sequence number can also be used by the receiver to estimate how many packets are being lost.

Since members of the working group join and leave during the conference, it is useful to know who is participating at any moment and how well they are receiving the audio data. For that purpose,

each instance of the audio application in the conference periodically multicasts a reception report plus the name of its user on the RTCP (control) port. The reception report indicates how well the current speaker is being received and may be used to control adaptive encodings. In addition to the user name, other identifying information may also be included subject to control bandwidth limits. A site sends the RTCP BYE packet (Section 6.5) when it leaves the conference.

## 2.2 Audio and Video Conference

If both audio and video media are used in a conference, they are transmitted as separate RTP sessions RTCP packets are transmitted for each medium using two different UDP port pairs and/or multicast addresses. There is no direct coupling at the RTP level between the audio and video sessions, except that a user participating in both sessions should use the same distinguished (canonical) name in the RTCP packets for both so that the sessions can be associated.

One motivation for this separation is to allow some participants in the conference to receive only one medium if they choose. Further explanation is given in Section 5.2. Despite the separation, synchronized playback of a source's audio and video can be achieved using timing information carried in the RTCP packets for both sessions.

## 2.3 Mixers and Translators

So far, we have assumed that all sites want to receive media data in the same format. However, this may not always be appropriate. Consider the case where participants in one area are connected through a low-speed link to the majority of the conference participants who enjoy high-speed network access. Instead of forcing everyone to use a lower-bandwidth, reduced-quality audio encoding, an RTP-level relay called a mixer may be placed near the low-bandwidth area. This mixer resynchronizes incoming audio packets to reconstruct the constant 20 ms spacing generated by the sender, mixes these reconstructed audio streams into a single stream, translates the audio encoding to a lower-bandwidth one and forwards the lower-bandwidth packet stream across the low-speed link. These packets might be unicast to a single recipient or multicast on a different address to multiple recipients. The RTP header includes a means for mixers to identify the sources that contributed to a mixed packet so

# Explore Litigation Insights

Docket Alarm provides insights to develop a more informed litigation strategy and the peace of mind of knowing you're on top of things.

## Real-Time Litigation Alerts



Keep your litigation team up-to-date with **real-time alerts** and advanced team management tools built for the enterprise, all while greatly reducing PACER spend.

Our comprehensive service means we can handle Federal, State, and Administrative courts across the country.

## Advanced Docket Research



With over 230 million records, Docket Alarm's cloud-native docket research platform finds what other services can't. Coverage includes Federal, State, plus PTAB, TTAB, ITC and NLRB decisions, all in one place.

Identify arguments that have been successful in the past with full text, pinpoint searching. Link to case law cited within any court document via Fastcase.

## Analytics At Your Fingertips



Learn what happened the last time a particular judge, opposing counsel or company faced cases similar to yours.

Advanced out-of-the-box PTAB and TTAB analytics are always at your fingertips.

## API

Docket Alarm offers a powerful API (application programming interface) to developers that want to integrate case filings into their apps.

## LAW FIRMS

Build custom dashboards for your attorneys and clients with live data direct from the court.

Automate many repetitive legal tasks like conflict checks, document management, and marketing.

## FINANCIAL INSTITUTIONS

Litigation and bankruptcy checks for companies and debtors.

## E-DISCOVERY AND LEGAL VENDORS

Sync your system to PACER to automate legal marketing.