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.

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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,

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