The Multimedia Transport Protocol RTP

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The Multimedia Transport Protocol RTP
Introduction

- How should one transmit audio and video streams over an internetwork?
- Do we need a new protocol?
  » What’s wrong with TCP?
  » With UDP?
Multimedia Data Transport

Requirements

- Media synchronization & codec control
  - Inter-stream synchronization
  - Inter-receiver synchronization

- Quality-of-service management
  - Receiver selectable services
  - Feedback to media generation sources on network performance

- Application control
  - Media adaptation

- Reliable transmission
  - Depends...

Multimedia Data Transport

Multicast requirements

- The peculiarities of multicast and IP-multicast require special protocol/application support
## Multicast Primer

### Multicast addressing: The 4 major classes of IP addresses

- **Class A addresses**
  - 128 networks
  - more than 65,536 hosts

- **Class B addresses**
  - 16,384 networks
  - 256 to 65,536 hosts

- **Class C addresses**
  - \(2^{21}\) networks
  - less than 256 hosts

- **Class D addresses**
  - 28-bit multicast addresses
  - no origin or network information is encoded

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## Multicast Primer

### Multicast addressing & multicast groups

- Multicast addresses can only be used as a destination address
- Multicast addresses correspond to a *multicast group*
  - Groups may be of *any* size
  - Group members may be located *anywhere* in the Internet
  - Hosts can join and leave groups at will
  - There is no “list” of group members
  - A sender cannot tell who, or if anyone, received any message
  - *Senders need not be members of the group*
The peculiarities of multicast and IP-multicast require special protocol/application support

- Mixing, bridging, transcoding
- Tunneling through firewalls
- Encryption

A media gateway is an entity on the path from a source to a destination that receives, manipulates, and retransmits media

Gateways can be either:
- Active — emulating the behavior of a media source
- Passive — forwarding streams in a transparent manner
Media Gateways

Active gateways

- Active gateways modify streams through some form of:
  - Transcoding, recoding
- Applications
  - Mixing/bridging for composite media transmission
  - Transcoding/mixing for low bandwidth links

Media Gateways

Passive gateways

- Passive gateways provide filtering/transcoding functions that are logically transparent to the receivers
- Applications
  - Application-level filters
  - Multicast to unicast replicators
  - Encryption
  - Simple transcoding
The Multimedia Transport Protocol RTP
Overview

◆ RTP data delivery services:
  » Multicast & unicast stream delivery
  » Bridging, translation, transcoding, & encryption

◆ RTP conference control services:
  » Stream timing and synchronization
  » Performance monitoring & media adaptation
The Multimedia Transport Protocol RTP

Protocol components

- RTP
  - Multicast & unicast stream delivery
- RTCP
  - Conference control and feedback
- Mixers, monitors, & translators
  - RTP media gateways
- RTP profiles
  - RTP/RTCP message formats and semantics
- RTP payload specifications
  - RTP message formats

RTP Protocol Architecture

Protocol layering

- RTP is an application-level, datagram protocol
- Traditional transport services such as:
  - addressing,
  - segmentation/reassembly,
  - quality-of-service, and
  - delivery semantics
  - are all provided by a lower level protocol
RTP Concepts and Terms

Sessions

- An RTP *session* is the sending and receiving of RTP data by a group of participants
  - For each participant a session is a (pair of) transport addresses used by a participant to communicate with the group

- If multiple media types are communicated by the group, the transmission of each medium constitutes a session
RTP Concepts and Terms

Synchronization sources

- Each source of RTP packets is called a *synchronization source*
  - Identified by a unique, randomly chosen 32-bit ID (the SSRC)
- A host generating multiple streams within a single RTP session must use a different SSRC per stream

RTP Concepts and Terms

Basics of data transmission

- A basic RTP message consists of
  - Synchronization source identifier of sender
  - Sequence number
  - Timestamp
  - Media data unit(s)
An RTP mixer is an intermediate system that receives & combines packets of one or more RTP sessions into a new packet

» Streams may be transcoded, special effects may be performed
» A mixer will typically have to define synchronization relationships between streams
» Resulting stream is multicast to a new group address

A mixer defines synchronization relationships between streams, thus...

Mixers are *synchronization sources*

» Sources that are mixed together become *contributing sources* (*CSRC*)
An intermediate system that...

» Connects two or more transport-level networks
  - Multicasting through a firewall
  - RTP dissemination across/into non-IP networks

» or...

Or... an intermediate system that modifies the stream without changing the stream’s timing

Sample translator functions:

» Encryption
» Frame-level transcoding or re-packing
» Protocol translation
Translators forward packets with SSRC identifiers intact
  » Translators are transparent to participants

As with mixers, output must be sent to a new multicast group address

RTP Gateways
Mixers and Translators example
Mixers and Translators
Message forwarding loops

- If gateways transmit to multicast group addresses to which they subscribe, then transmission loops form

![Diagram](image)

Mixer and Translator Anomalies
Message forwarding loops

- However, having translators transmit to a new group address is not sufficient to avoid loops
  » Participants must be prepared to dynamically detect loops as a session progresses

![Diagram](image)
Mixer and Translator Anomalies
Detecting forwarding loops

-loops through mixers can be detected by recognizing one’s SSRC in a CSRC list

Mixer and Translator Anomalies
Forwarding loops and SSRC conflicts

-but what’s the difference between a loop and an SSRC conflict?
  » How can S1 determine what is going on?
Mixer and Translator Anomalies
Dealing with forwarding loops and SSRC conflicts

Senders: if a SSRC conflict is found:
- leave the session and rejoin with a new SSRC
- if conflict persists, then a loop exists

 Receivers: ignore packets from one of the sources
- assume sources/gateways will fix the problem

Dealing With Loops & SSRC Conflicts
Receiver requirements

- All receivers maintain a table of all synchronization sources they have heard from recently
  - Table entries contain (SSRC, source transport address) pairs
  - Entries time out and are deleted if no packets from a source have been received for a sufficiently long time

- If a SSRC conflict is detected then packets from the new source are ignored
Dealing With Loops & SSRC Conflicts

**Sender requirements**

- If a source’s packets are looped then the source keeps track of other sources that conflict with it
  - Record the conflicting source’s transport address and the time the most recent conflicting packet from each source was received
  - Entries time out if no conflicting packets are received over a specified interval

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**The Multimedia Transport Protocol RTP**

**Outline**

- **RTP concepts**
  - Entities and abstractions

- **Protocol definition**
  - Header format and packet structure

- **Developing interoperable applications with RTP**
  - RTP profiles

- **Quality-of-service monitoring and reporting**
  - Real-time control protocol RTCP
RTP Protocol Definition
Message encapsulation

- RTP is an application-level protocol
  - UDP is the canonical transport protocol
- Underlying transport protocol handles
  - addressing
  - segmentation/reassembly
  - delivery semantics
  - quality-of-service

Protocol Definition
Fixed header format

![Fixed header format diagram]

- 12 octets (bytes)
  - A version number (= 2)
  - Payload type
  - Padding (yes/no)
  - Sequence number (16 bits)
  - Extension bit (yes/no)
  - Timestamp (32 bits)
  - Contributed source count (0-15)
  - Synchronization source identifier (32 bits)
RTP Header Format
Header extensions/optional fields

| 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 0 | 1 |
| v | p | x | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 0 | 1 | 2 | 3 | 4 | 5 | 6 | 7 | 8 | 9 | 0 | 1 |

- Contributing source identifier(s)
  - 0-15 entries, 32-bits each
  - Set by mixers
- Header extension
  - (Not expected to be used)

Creating an Instance of the RTP Protocol
RTP profiles

- RTP is a protocol framework that needs to be instantiated for a specific application or use
  - what are markers?
  - what is the clock frequency?
  - what are payload types?
  - how is the payload formatted?
The RTP Profile for Audio & Video Conferences with Minimal Control

◆ The default profile for 2-way interactive multimedia communication

◆ Defines markers & payload types for common audio and video encoding schemes and defines payload formats for audio encodings

  » PCM ("L16")
  » µ-Law ("PCMU")
  » IMA ADPCM ("DVI4")
  » MPEG Audio ("MPA")
  » CELP ("1016")
  » LPC
  » GSM
  » ...

A profile for simple audio video conferences
IMA ADPCM (DVI4) audio conferencing example

◆ Payload type = 5
◆ Clock frequency = 8 kHz
  » ticks are 125 µs apart
◆ The marker bit identifies the start of a talkspurt
◆ Packets contain ≈20 ms of audio data
  » 161 audio samples per packet
◆ Payload format:
  » uncompressed predicted sample (16-bits)
  » current index into the step-size table (8-bits)
  » reserved byte (8-bits)
  » 160 compressed samples (4-bits each)