Real-Time Transport Protocol

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Introduction

The distribution of audio and video across modern Ethernet networks is creating a new dimension in how businesses communicate. The ability to make a phone call across the Internet, to watch a lecture that is taking place across the country, or collaborate with co-workers breaks down distance barriers allowing us to share ideas, information, and resources to be more productive. Several technologies have evolved to provide the means to send audio and video across a network such as:

QuickTime™

Shockwave™

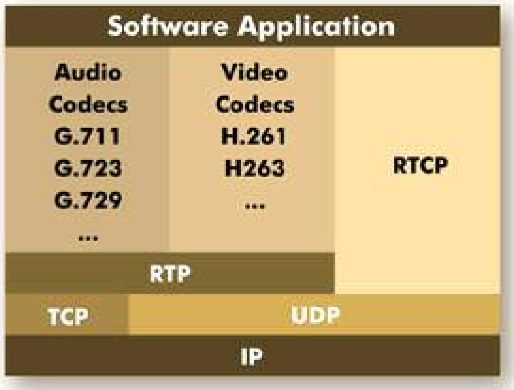
RealAudio™

Microsoft's Windows Media™

However, many of these popular technologies are not robust enough to scale to service thousands of individual users during a single session. To meet the need to deliver video and audio content over a network to a large audience, several powerful and scalable protocols were designed for specific implementations such as RTSP for streaming applications, H.323 and SIP for IP telephony applications, and SAP/SDP for multicast sessions. One thing all of these robust protocols have in common is the underlying use of the Real-Time Transport Protocol (RTP) to deliver real-time content.

What is RTP?

Real-Time Transport Protocol (RTP) (RFC3550) is an application layer protocol which essentially means it uses lower level protocols such as UDP and TCP for transport across the network, but the information contained in the RTP packet is meant for an application like Windows Media Player, SIP based phones, or H.323 media gateways.



UDP or TCP?

RTP can be used over UDP or TCP, but which is better? It really depends on how RTP will be used. In most situations, UDP is favored over TCP because UDP is much quicker at creating packets. This is because UDP has less overhead since it does not provide several TCP functions such as sequencing the datagrams it sends, packet receipt verification, missing packet retransmissions, and other flow control services. In most cases where RTP is used there is no need to have the functions that TCP provides, making UDP an even better choice.

Regardless of the underlying network protocol used, RTP provides data transport for real-time data. It provides several functions to ensure data is synchronized for all users and will be recombined correctly at the receiving end by using the information contained in the RTP packet.

RTP Packet

The RTP packet is a combination of an RTP header, the data payload, and the underlying UDP framework that allows the packet to be properly transmitted across a network. Below is a graphical depiction of an RTP packet header and the payload:

Version: The version field identifies the version of RTP, which is currently version 2

Padding (P): Some protocols or algorithms require a packet to be a specific size. The padding field is a variable field that fills space to make the header the proper size.

CSRC count: The CSRC field contains the number of CSRC identifiers that follow the fixed header.

Marker: The marker field is used in specific applications to identify groups of RTP packets.

Payload type: This field identifies what type of information is in the payload portion of the RTP packet and is used by applications to determine how to handle the packet.

Sequence number: The sequence number keeps a count of RTP packets by incrementing the field by one for each packet that is sent, and is used to detect packet loss and to maintain the sequence of the packets that are received.

Timestamp: The timestamp field records the time when the RTP packet was created and is derived from a clock that increments monotonically and linearly in time to allow synchronization and jitter calculations.

Synchronization source (SSRC): The SSRC field contains a randomly generated 32-bit number to identify individual RTP sessions for the purpose of separating individual RTP sessions and synchronizing the data transmission across multiple end-points.

Contributing source (CSRC): The CSRC field is another layer of identification for sessions that have the same SSRC number, but the data in the stream needs to be differentiated further. For example, in a conference call, all the RTP packets in the session would contain the same SSRC identifier but the CSRC could be used to identify each caller who is speaking.

However, an RTP packet is still just a packet and there are several problems packets have that are magnified when dealing with real-time information. First, packets may or may not be received and when they are, they may not be in the same order as when they were sent. This makes reconstructing the packet stream in the proper order and requesting missing packets very important.

Second, a real-time transmission is sensitive to the delay created from the time a packet is sent to when it is received. Third is the proper identification of the senders and receivers to make sure the transmissions are valid.

Finally, there must be a way to monitor the quality of service for the recipients of the real-time transmission so adjustments can be made to maintain an acceptable level of transmission quality. To summarize, the potential problems are:

Packet order

Network delay

Monitoring functions

Payload identification

RTP provides several functions in the RTP header that help overcome the problems real-time transmissions have when sent across a network, and works closely with the real-time control protocol (RTCP) to provide monitoring and feedback capabilities.

Packet order /network delay

When a transmission is sent to one or many end-points packet order arrival is always a problem. RTP provides two fields in the packet header to help reconstitute a transmission, a time-stamp field, and a packet sequence number.

The time stamp field uses the network time protocol (NTP) to mark the packet with the time it was sent. The receiver can use this information to decide if the packet is fresh enough to use and can be used to calculate the jitter of the packets (the variation in delay between when packets are sent and when they are received). Jitter provides a useful statistic for configuring quality of service measures that are external to RTP.

The packet sequence field gives the packets in an individual transmission a specific order by which they can be recombined. Missing packets are retransmitted until the entire message is received.

Besides delay and packet order other network conditions can hamper the transmission of real-time data. One common situation in an enterprise network is mixed networks where parts of the network have large amounts of bandwidth and other parts have limited bandwidth such as remote offices connected via a WAN link. RTP has the ability to take varying network bandwidth into account during a single transmission through the use of mixers and translators.

Normally a network-based transmission can only perform up to the level of the lowest common denominator. For networks with mixed network links, those users with fast network connections would have to receive real-time transmissions at the same speed as those users with slower connections.

Mixers and translators allow a single real-time transmission to use different compression and decompression algorithms based on bandwidth limitations so that users with fast connections to the network are able to receive real-time transmissions at faster speeds and higher quality while users with limited bandwidth are able to receive the same transmissions, by using a more aggressive compression/decompression algorithm to save bandwidth.

Monitoring Functions

When dealing with a real-time transmission feedback is very important. Since most implementations of RTP are built using UDP packets for transport there is no real way to send information about performance, therefore another protocol like the real-time transmission control protocol (RTCP) is needed to share performance information. RTCP (RFC 3550, 3551) is the control protocol designed to work along side the real-time transport protocol (RTP) to provide feedback for flow control manage several aspects of the delivery of real-time content. RTCP and RTP are very scalable in that they can support sessions with a few users up to thousands of users; as such they are used by several protocols such as H.323 and SIP to deliver real-time streaming content.

As mentioned above, RTP and RTCP work together to deliver real-time content. RTP is a protocol used to format blocks of real-time data. During the process of packetizing the real-time data stream RTP gathers information such as packet jitter, packet delay, lost packets, and time stamp. RCTP allows the source and the end-points receiving the real-time content to periodically exchange the information gathered by RTP in the form of RTCP packets to provide feedback on the quality of the data distribution, track session participants, and provide session control information.

Payload Identification

As mentioned before, RTP is an application layer protocol and therefore it is important to be able to classify what type of information is in an RTP packet. This allows proper direction to its destination and enables external QoS mechanisms to classify the traffic so priority can be given to the real-time traffic. Identification is provided by RTP through the payload identification field in the RTP packet header.

RTP is not implemented as a separate layer in the scheme of network protocols, but is meant to piggyback upon other transport protocols such as TCP and UDP. Because RTP operates in conjunction with UDP and TCP, it is compatible with multicast or unicast network services. RTP does not provide quality of service (QoS) functions and therefore must rely on the network and its QoS functions, such as DiffServ, ToS, 802.1p and Q, or RSVP, for any level of reliability.

RTP provides many functions through the RTP header, but it does not have any way to ensure packet network quality, or stop packet loss and network delay. RTP packets are still at the mercy of the overall network when it comes to timely delivery. The real-time transport control protocol (RCTP), which was built to work along side RTP, provides valuable feedback about the quality of service for an RTP session and it provides information about who is participating in the session to allow network-based measures to be implemented or configured to improve network transmission quality.

RTCP

Since RTP is mainly a way to format and transmit real-time data, there must be a mechanism to monitor the RTP session and provide feedback so that problems can be corrected to ensure optimal RTP session quality. The Real-time Transport Control Protocol (RTCP) is designed to be independent of the underlying transport and network layers and provides several functions to enhance RTP.

The first function is to provide feedback on the network's data transmission quality. The participants in the real-time session use RTCP to exchange the information gathered by RTP to help with multicast configuration and performance. Senders and receivers can use the RTCP information to optimize performance for multicast and real-time traffic by changing aspects of the transmissions such as timing and the encoding of the audio/video payload.

Second, RTCP packets are used to synchronize the real-time traffic between sending and receiving endpoints. For instance, if a user who is active in the real-time session suddenly reboots their machine and reconnects to the session, RTCP is used to resync the user with the RTP session so they see or hear the same thing that everyone else sees or hears.

Third, RTCP is used to gather information about the participants of the real-time session such as the total number of participants or user information (i.e., email address). Finally, if a problem occurs where packets are dropped or there is a loss of network connectivity, RTCP information can be used by network administrators to determine if the problem is local, regional, or global.

RTP Security

Most video and voice functions do not require a level of security beyond what is already implemented within the normal network (i.e., VPN, firewalls, RADIUS, etc.) However, there are times when a real-time transmission must be protected such as when sending confidential real-time data that could be intercepted. Therefore steps have been taken to create secure RTP (SRTP) where the header of the RTP packet is left untouched so regular packet handling can occur without problems, but the payload will be encrypted using an implementation of the Advanced Encryption Standard (AES). SRTP is still in development and the implementation may change over time until the final specification is validated. For now, SRTP when used with other security measures can be a powerful way to transmit real-time data across a network.

Conclusion

RTP itself does not have a specific use. It is more of a building block for applications based on the transmission of real-time data. As such, RTP has been integrated into several mainstream protocols such as SIP and H.323 and therefore, is important in many different enterprise technologies such as VoIP, streaming media presentations, distance learning, and collaboration software.

However, RTP is not just good for video and voice transmission – it is also used for real-time data transmission for applications. Uses include customer relationship management and inventory management where real-time replication of databases is required and also anywhere data needs to be sent in real-time such as updating network components (firewalls, PBXs or switches) across the entire enterprise, or updating a companies employee directory that has several nodes located around the world. RTP is not something you will see as a marquee feature, but it is one of the fundamental engines that make the transmission of real-time data across a network practical and possible.

Glossary

NTP	Network Time Protocol
RTCP	Real-time Transport Control Protocol
RTP	Real-time Transfer Protocol
RTSP	Real-time Streaming Protocol
SAP	A client/server database architecture developed by IBM
SDP	Streaming download project
SIP	Session Initiation Protocol

Sources of additional information

RTP RFC 3550

Introduction to streaming media

Multimedia Over IP: RSVP, RTP, RTCP, RTSP

RTP Data Transfer Protocol