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Real-Time Protocols (RTP/RTCP)

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INTRODUCTION

Real-time protocols cover specific needs by applications with real-time characteristics. Real-time applications, such as voice over IP (VoIP), videoconferencing applications, video on demand, continuous data applications, and control and measurement applications have specific requirements from the lower layers, mainly in terms of packet loss, delay, and jitter. Traditional transport protocols such as **TCP** and **UDP** have been designed for general use and are not specialized for such specific purposes. In particular, real-time protocols have to be able to deliver high throughput, handle multicast, manage the transmission quality, and be friendly to the rest of the traffic, and, more importantly, to the congestion-sensitive TCP traffic.

BACKGROUND

An early attempt at a protocol designed for transferring real-time data was **NVP** (Cohen, 1981). It was first implemented in 1973 by Danny Cohen of the Information Sciences Institute (ISI), University of Southern California. The project's stated goals (Cohen, 1976) were "to develop and demonstrate the feasibility of secure, high-quality, low-bandwidth, real-time, full-duplex (two-way) digital voice communications over packet-switched computer communications networks…[and to] supply digitized speech which can be secured by existing encryption devices. The major goal of this research is to demonstrate a digital high-quality, low-bandwidth, secure voice handling capability as part of the general military requirement for worldwide secure voice communication." NVP was used to send speech between distributed sites on the ARPANET, using several different voice-encoding techniques.

RTP/RTCP protocol was first defined in RFC 1889 (Schulzrinne et al., 1996), which was later updated with RFC 3550 (Schulzrinne et al., 2003). The discussions on the rationale and design choices behind RTP were summarized in Schulzrinne (1993), which provides a good reference to the desired characteristics for an efficient and flexible real-time protocol.

REAL TIME PROTOCOLS DESCRIPTION

Desired Characteristics of Real-Time Protocols

High Throughput

Multimedia data and especially video require continuous high-rate transmission. The real-time protocol that takes over the transport of data has to be fast enough to support the application requirements, and in particular the protocol throughput has to be faster than the network access speed, otherwise the bandwidth will not be used efficiently and the transport protocol will be a bottleneck.

Another approach to the throughput requirements for a transport protocol is the total communications system view. The throughput of a transport protocol has to be higher than the access speed of the network, otherwise it would not be possible to fully utilize the bandwidth offered by the network access points, and the transport protocol would become a bottleneck of the whole communications system (Rosenberg et al. 1998).

Multicast Capability

Multicast support is also essential, because many applications specify multiple recipients and transmitting the same large amount of data over multiple unicast connections wastes available network resources.

Transmission Quality Management

Multimedia data flows require quality of service guarantees regarding bandwidth, delay, and jitter. In order to satisfy these requirements, a transport system has to provide the applications with a mechanism for determining and negotiating their **quality of service** (QoS) requirements. These QoS requirements are transferred by the transport layer to the network layer, which is responsible for propagating them and for making the necessary reservations of resources over a network connection. This network connection often supports multicast functionality which is useful for many multimedia applications. In order to support QoS guarantees, the cooperation of all the subsystems of a transport system is necessary, which includes resource management, network access control, and queue management at network devices. The operating system should also be able to support multimedia applications.

In the case that the network is not able to provide quality of service guarantees, the real-time protocol has to be able to adapt the transmitted multimedia data to the current network conditions. Although this technique does not offer specific QoS guarantees, as in Jacobson et al. (1999) and Heinanen et al. (1999), it can improve the network performance as a whole because of the reduction in congestion and packet losses.

TCP Friendliness

The TCP protocol implements a congestion avoidance mechanism that is best suited to the transmission of non real-time data such as HTTP or FTP. Real-time applications have to be based on UDP, which is faster and offers no reliability or congestion control. The lack of congestion control mechanism in UDP can lead to congestion problems if a UDP sender exceeds the transmission rate that can be handled by the network. TCP traffic is very sensitive to congestion because of TCP's congestion avoidance mechanism, and therefore the UDP traffic rate has to be somehow controlled. These mechanisms should not only aim at avoiding network overload, but also transmiting TCP-friendly traffic. TCP-friendly traffic is a traffic stream that does not consume more bandwidth than a TCP stream would consume on the same network path (Bouras et al., 2005).

Extensibility

Real-time multimedia services are still a field of research where new ideas and implementations occur often, and therefore, a real-time protocol should be able to incorporate additional services as practical experience with the protocol is gathered and as applications that were not originally anticipated use its services. Furthermore, experimental applications should be able to exchange application-specific information without jeopardizing interoperability with other applications (Schulzrinne et al., 2003).

Multiple Content

Real-time protocols are mainly motivated by audio and video for conferences (Basso et al., 1997). However, other applications, such as distribution of voice/video, distributed simulations, and loss-tolerant remote data acquisition may also use the services provided by such a protocol. Also, new formats of established media, for example, high-quality multi-channel audio or combined audio and video sources, should be anticipated (Rosenberg et al., 1996).

The RTP/RTCP Protocol

The RTP/RTCP protocols have been specifically created for transferring multimedia data such as voice and video. Initially designed for multicast communication, they have also been widely used for unicast communications. They can be used for one-direction communication, such as for video on demand services, but also for full-duplex communication, such as for videoconferencing and VoIP applications. They provide a common platform for data transfer and synchronization of information.

RTCP (real-time control protocol) is the control protocol for RTP (real-time protocol). RTP operates in cooperation with RTCP, which provides the information regarding the connection quality and the participants in the RTP **session**.

RTP offers end-to-end transport services for data with real-time characteristics. In particular, RTP enables the specification and identification of the payload type, sequential numbering of packets, timestamps, and control of the transport procedures. RTP offers end-to-end services, but does not offer the complete functionality of a transport layer protocol. An application can use RTP over TCP or UDP in order to take advantage of multiplexing and checksum functions of the TCP or UDP protocol, but any other suitable transport protocol can also be used. RTP is not aware of the connection and can therefore operate over both connection-oriented and connectionless lower level protocols.

RTP offers no mechanism for guaranteeing the delivery of data in specific time intervals, and no quality of service guarantees for the transmission, because that is an issue for the lower level protocols. For an application that requires such guarantees, RTP can be accompanied by mechanisms such as **RSVP**, which can provide resource reservation and reliable services.

Multimedia applications usually pose strict time constraints regarding transmission of data, which does not fit very well with the architecture of the Internet. The RTP protocol provides several mechanisms that take into account these issues. Such mechanisms are timestamps and sequential numbering of packets.

Timestamps provide useful information to real-time applications. The sender inserts a timestamp into each packet, which is used by the receiver in order to determine how the data should be presented to the end user. In other words, timestamps provide the synchronization signals so that the receivers can properly reconstruct the initial data. Timestamps are also used for the synchronization of separate data flows, such as audio and video (RTP/RTCP transmits audio and video using separate data flows). RTP is not accountable for this synchronization, which is the responsibility of the applications.

UDP, which is typically used for the transmission of RTP/RTCP packets, does not deliver packets in the order they were transmitted. Therefore, RTP packets are sequentially numbered upon transmission, so that the receiver can properly arrange them. These sequence numbers are also used in order to detect packet losses.

Since RTP is often used for multicast communication, an RTP data packet contains the identity of the information sender, so that the session group can identify which member of the session transmits data. The sender's identity is provided in the source identification field.

RTP is typically used over the UDP transport protocol. TCP and UDP are the most widely used transport protocols on the Internet. While TCP offers connection-oriented services and reliable data transmission, UDP offers connectionless services and no reliability. UDP is preferable as the transport protocol for RTP because:

- RTP is mainly designed for multicast transmissions which do not fit well with the connection-orientated TCP.
- Especially for multimedia data, reliability is not as important as timely transmission. Reliable transmission is usually achieved through retransmission of lost packets, which might not be desirable, since it can lead to network overloading and can hinder the steady transmission of data.

The idea behind the control protocol RTCP is that applications that have recently transmitted multimedia data generate a sender report which is sent to all the participants in the RTP session. This report includes counters for the packet data and the bytes sent, and the receivers can use them to estimate the actual data transmission rate.

In order to establish an RTP session, an application determines a pair of destination addresses (which is comprised of an IP network address and two ports, one for RTP and one for RTCP). The address can be either a unicast or a multicast network address. During a multimedia session, each medium is transmitted in a separate RTP session, and RTCP packets report the transmission quality for each separate session. This means that audio and video are transmitted at separate RTP during a videoconference.

Although RTP/RTCP packets are transferred inside UDP packets, data packets and control packets use two sequential ports, with the RTP port always being the lower one and with an odd number. In the case of other protocols below RTP at the protocol stack (such as the case of having RTP directly over AAL5: ATM Adaptation Layer type 5), it is possible to transfer both data (RTP) and control information (RTCP) within a single data unit of the lower layer protocol, with data following the control information.

Therefore, an RTP session is characterized by the following parameters:

- **IP address of participants:** This can be either a multicast IP address, which corresponds to the multicast session of the participants group, or a set of unicast addresses
- **RTP port:** The port number used by all participants in the session for sending data
- **RTCP port:** The port number used by all participants in the session for sending RTCP control messages

The RTP header provides the synchronization information necessary for synchronizing and presenting audio and video data, and also for determining whether any packets have been lost or arrived out of order. Furthermore, the header determines the data type and therefore allows multiple types of data and compression. RTP can be tuned to meet the specific requirements of each application by using auxiliary data structure and shape specifications.

In order to allow a higher level of synchronization or to synchronize non-periodical data flows, RTP uses a clock that increments monotonically. This clock is usually increased in time units smaller than the smallest block size of the data flow, and its initial value is random. An application does not use the RTP timestamps directly but it rather uses the NTP (network time protocol) timestamps and the RTP timestamps from the transmitted RTCP packets for each flow that needs synchronization.

Session participants produce receiver reports for all the senders—sources of audio and video data from which they have recently received data. The reports contain information regarding the highest sequence number received, the number of lost packets, the jitter, and the timestamps needed for calculating an estimate of the transmission **round trip time** (RTT).

As RTP and RTCP create separate sessions for separate data streams, an RTCP sender report contains an indication of the actual time, and an RTP timestamp which can be used for synchronizing multiple data flows at the receiver.

RTP data packets identify their source only through a 32-bit randomly generated number, while RTCP messages include a source description (SDES) which contains relevant information. Such a body of information is the canonical name, a globally unique identification code of the session participant. Other possible SDES objects are the user's name, e-mail address, telephone number, and application information.

RTCP offers feedback capabilities related to the current network conditions and the reception quality, allowing the applications to automatically adapt to changing network conditions. For example, a slowdown by many receivers could possibly be due to a network problem (for example a faster link has failed and has been substituted by a slower backup link) and are not due to a specific participant. In such a case, the sender could choose to immediately switch to another codec with lower bandwidth requirements, or temporarily stop transmitting video, or use some other technique to reduce the multimedia transmission rate.

In other cases, the network administrators can use information from RTCP packets in order to evaluate the performance of their networks. Since RTCP sends feedback information not only to the sender, but also to all other receivers of a multicast stream, it enables a user to realize whether a problem is due to the local node or is more general to the network.

The basis for traffic and congestion control is offered by the RTCP sender and receiver reports. By analyzing the jitter field, which is included in the RTCP sender report, the fluctuation during a certain time period can be measured and the possibility of congestion can be identified and dealt with before it appears and causes packet losses.

FUTURE TRENDS

Since RTP contains no specific assumptions about the capabilities of the lower layers, except the fact that they provide framing, it is capable of running over **IPv6** (Deering et al., 1998), the new Internet protocol that is expected to gradually replace the currently used IPv4 in the near future. RTP contains no network-layer addresses, so it is not affected by address changes. IPv6 includes enhanced support for lower-layer capabilities such as security or quality-of-service guarantees, and these features can be used by applications employing RTP. This combination makes RTP an attractive option for new multimedia applications that can benefit from both RTP support for real-time data and enhancements from IPv6 and other emerging Internet technologies

CONCLUSION

This article presented the main design characteristics for real-time protocols and a detailed presentation of RTP/RTCP, the most widely implemented and used protocol for transportation of real-time data. Real-time protocols should be extensible, provide for high throughput, be able to operate over multicast, and transfer multiple types of content. Naturally, no end-to-end protocol can ensure in-time delivery, since this always requires the support of lower layers that actually have control over resources in switches and routers. RTP/RTCP satisfactorily covers the above requirements and provides functionality suited for carrying real-time content, e.g., a timestamp and control mechanisms for synchronizing different streams with timing properties.

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KEY TERMS

Delay Jitter: Delay jitter is defined to be the mean deviation (smoothed absolute value) of the difference in packet spacing at the receiver compared to the sender for a pair of packets.

Internet Engineering Task Force (IETF): The organization comprised of a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of Internet architecture and the smooth operation of the Internet.

Multimedia Data: Multimedia data refers to data that consist of various media types like text, audio, video, and animation.

NVP (Network Voice Protocol): A pioneering network protocol for transporting human speech over packetized communications networks.

Packet Loss Rate: Packet loss rate is defined as the fraction of the total transmitted packets that did not arrive at the receiver.

Quality of Service (QoS): The ability to provide specific guarantees to traffic flows regarding the network characteristics, such as packet loss, delay, and jitter experienced by the flows.

TCP (Transmission Control Protocol): A connection-oriented, reliable protocol of the TCP/IP protocol suite used for managing full-duplex transmission streams.

UDP (User Datagram Protocol): A connectionless, unreliable protocol of the TCP/IP protocol suite used for sending and receiving datagrams over an IP network.