REAL TIME TRANSPORT PROTOCOL (RTP)

In real-time applications, a stream of data is sent at a constant rate. This data must be delivered to the appropriate application on the destination system, using real-time protocols. The most widely applied protocol for real-time transmission is the Real-Time Transport Protocol (RTP), including its companion version: Real-Time Control Protocol (RTCP).

UDP cannot provide any timing information. RTP is built on top of the existing UDP stack. Problems with using TCP for real-time applications can be identified easily. Real-time applications may use multicasting for data delivery. As an end-toend protocol, TCP is not suited for multicast distribution. TCP uses a retransmission strategy for lost packets, which then arrive out of order. Real-time applications cannot afford these delays. TCP does not maintain timing information for packets. In real-time applications, this would be a requirement.

The real-time transport protocol (RTP) provides some basic functionalities to real-time applications and includes some specific functions to each application. RTP runs on top of the transport protocol as UDP. As noted in Chapter 7, UDP is used for port addressing in the transport layer and for providing such transport-layer functionalities as reordering. RTP provides application-level framing by adding application-layer headers to datagrams. The application breaks the data into smaller units, called application data units (ADUs). Lower layers in the protocol stack, such as the transport layer, preserve the structure of the ADU.

Real-time applications, such as voice and video, can tolerate a certain amount of packet loss and do not always require data retransmission. The mechanism RTP uses typically informs a source about the quality of delivery. The source then adapts its sending rate accordingly. If the rate of packet loss is very high, the source might switch to a lower-quality transmission, thereby placing less load on the network. A real-time application can also provide the data required for retransmission. Thus, recent data can be sent instead of retransmitted old data. This approach is more practical in voice and video applications. If a portion of an ADU is lost, the application is unable to process the data, and the entire ADU would have to be retransmitted.

Real-Time Session and Data Transfer

The TCP/IP and OSI models divide the network functionalities, based on a layered architecture. Each layer performs distinct functions, and the data flows sequentially between layers. The layered architecture may restrict the implementation on certain functions out of the layered order. Integrated layer processing dictates a tighter coupling among layers. RTP is used to transfer data among sessions in real time. A session is a logical connection between an active client and an active server and is defined by the following entities:

- RTP port number, which represents the destination port address of the RTP session. Since RTP runs over UDP, the destination port address is available on the UDP header.
- IP address of the RTP entity, which involves an RTP session. This address can be either a unicast or a multicast address.

RTP uses two relays for data transmission. A relay is an intermediate system that acts as both a sender and a receiver. Suppose that two systems are separated by a firewall that prevents them from direct communication. A relay in this context is used to handle data flow between the two systems. A relay can also be used to convert the data format from a system into a form that the other system can process easily. Relays are of two types: mixer and translator.

A mixer relay is an RTP relay that combines the data from two or more RTP entities into a single stream of data. A mixer can either retain or change the data format. The mixer provides timing information for the combined stream of data and acts as the source of timing synchronization. Mixers can be used to combine audio streams in real-time applications and can be used to service systems that may not be able to handle multiple RTP streams.

The translator is a device that generates one or more RTP packets for each incoming RTP packet. The format of the outgoing packet may be different from that of the incoming packet. A translator relay can be used in video applications in which a high-quality video signal is converted to a lower-quality in order to service receivers that support a lower data rate. Such a relay can also be used to transfer packets between RTP entities separated by an application-level firewall. Translators can sometimes be used to transfer an incoming multicast packet to multiple destinations.

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