13.1 Real-time Transport Protocol

In the following we provide a brief overview of the RTP. For the full specification of the RTP refer to the Internet draft on RTP [rtp].

Streaming media has certain characteristics as against the normal data. Therefore streaming media warrants a separate transport protocol. Streaming media can be of the following two types:

(a) Real-time delivery of media as in the case of the media-on-demand application. In this case a server may stream a stored media from a file while the client receives and playback the media in real-time.

(b) Delivery of a real-time data. A typical example is a video-conferencing application wherein the captured audio and video are being streamed in real-time.

A very important characteristic of multimedia contents is that they can tolerate small amount of errors. Reason being the human brain can reconstruct the original information from imperfect audio or video signals. However the timing details of the multimedia content should be preserved for the proper playback. Further interactive applications demand a low latency in the transmission of the streaming media.

The default transport layer protocol for the Internet is the Transmission Control Protocol (TCP) [tcp/ip]. TCP is not an appropriate choice for carrying real-time multimedia contents for the following reason. Multimedia contents do not demand the 100% reliability in the transmission, that is being offered by the TCP. On the other hand the overhead introduced by TCP (acknowledgement processing and retransmission) causes large amount of delay in receiving the data and makes the application less interactive. Therefore TCP is seldom used for real-time streaming.

The other popular transport protocol of the Internet is the User Datagram Protocol (UDP). As such UDP is also unsuitable for multimedia transmission as UDP does not guarantee ordered delivery of packets.

Therefore the Audio Video Transport (AVT) working group of the Internet Engineering Task Force (IETF) came up with the specification for a new protocol called Real-time Transport Protocol [ietf]. RTP is an end to end transport layer protocol intended for transmitting real-time data such as audio and video.

RTP has a companion protocol called RTP Control Protocol (RTCP) whose role is to monitor the quality of data delivery offered by the RTP. We can use any transport layer and network layer protocol to carry RTP and RTCP packets. However the RTP and RTCP packets are normally carried by UDP packets while the UDP packets are in turn carried by the IP datagrams. However Figure 13.1 shows how and where RTP and RTCP fits in the protocol stack of the network architecture.

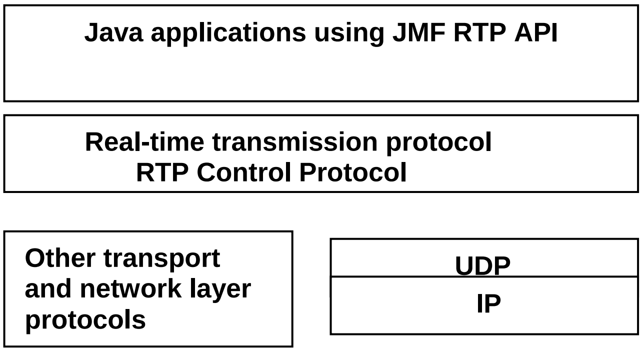

Figure 13.1 The networking protocol stack with JMF RTP API.