A HISTORICAL PERSPECTIVE

The Internet has gone from near invisibility to near ubiquity and penetrated into every aspect of society in the past few years (Department of Commerce, 1998). The application scenarios have also changed dramatically and now demand a more sophisticated service model from the network. In the early 1990s, there was a large-scale experiment in sending digitized voice and video across the Internet through a packet-switched infrastructure (Braden, Clark, & Shenker, 1994). These highly visible experiments have depended upon three enabling technologies: (a) Many modern workstations now come equipped with built-in multimedia hardware, (b) IP multicasting, which was not yet generally available in commercial routers, is available, and (c) highly sophisticated digital audio and video applications have been developed. It became clear from these experiments that an important technical element of the Internet is still missing: Real-time applications often do not work well across the Internet. The Internet, as originally conceived, offers only a very simple quality-of-service (QoS), point-to-point, best-effort data delivery. However, for a real-time application, there are two aspects of the problem with using this service model. If the sender and/or receiver are humans, they simply cannot tolerate arbitrary delays; on the other hand, if the rate at which video and audio arrive is too low, the signal becomes incomprehensible. To support real-time Internet applications, the service model must address those services that relate most directly to the time of delivery of data. Real-time applications like video and audio conferencing typically require stricter guarantees on throughput and delay. The essence of real-time service is the requirement for some service guarantees in terms of timing. In response to these demands of real-time multimedia applications, the Internet Engineering Task Force (IETF) has significantly augmented the Internet protocol stack based on the Internet integrated-services model, which is the focus of this article.

THE INTERNET INTEGRATED-SERVICES MODEL

An Internet service model consists of a set of service commitments; that is, in response to a service request, the network commits to deliver some service. The Internet is conventionally designed to offer a very simple service model, best effort, providing no guarantee on the correct and timely delivery of data packets. Each request to send is honored by the network as best as it can. This is the worst possible service: Packets are forwarded by routers solely on the basis that there is a known route, irrespective of traffic conditions along that route. This simplicity has probably been one of the main reasons for the success of IP technology. The best-effort service model, combined with an efficient transport-layer protocol (TCP [transmission-control protocol]), is perfectly suited for a large class of applications, which tolerate variable delivery rates and delays. This class of applications is called elastic applications.

However, demanding real-time applications require more sophisticated service models beyond the best effort. There has been a great deal of effort since 1990 by IETF to add a broad range of services to the Internet service model, resulting in the Internet integrated service model (Braden et al., 1994; Crowcroft, Handley, & Wakeman, 1999). The Internet integrated services model defines five classes of services that should satisfy the requirements of the vast majority of future applications.
integrated-layer processing (Clark & Tennenhouse, 1990). In this approach to protocol architecture, the different functions are next to each other, not on top of one another. The Internet multimedia protocol architecture is shown in Figure 1.

As shown in Figure 1, the overall multimedia data and control architecture currently incorporates a set of real-time protocols, which include the real-time transport protocol (RTP) for transporting real-time data and providing QoS feedback, the real-time streaming protocol (RTSP) for controlling delivery of streaming media, the session-announcement protocol (SAP) for advertising multimedia sessions via multicast, and the session-description protocol (SDP) for describing multimedia sessions. In addition, it includes the session-initiation protocol (SIP), which is used to invite the interested parties to join the session. But the functionality and operation of SIP does not depend on any of these protocols. Furthermore, the resource-reservation protocol (RSVP) is designed for reserving network resources. These protocols, together with reliable multicast (Handley, Floyd, Whetten, Kermode, Vicisano, & Luby, 2000), are the underlying support for Internet multimedia applications. While all the protocols above work on top of the IP protocol, the Internet stream protocol, version 2 (ST-II), is an IP-layer protocol that provides end-to-end guaranteed service across the Internet.

### The Real Time Transport Protocols: RTP and RTCP

The real-time transport protocol, named as a transport protocol to emphasize that RTP is an end-to-end protocol, is designed to provide end-to-end delivery services for data with real-time characteristics, such as interactive audio and video (Schulzrinne, Casner, Frederick, & Jacobson, 2003). Those services include payload-type identification, sequence numbering, time-stamping, and
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