Standards

Standardizing Real-Time Streaming Protocols

Interview with Rob Lanphier, RealNetworks

At the RealNetworks Conference in April 1998, RealNetworks announced its new G2 streaming media product. During the product announcement, they made a big deal about the fact that it was based on the newly approved Real-Time Streaming Protocol (RTSP) standard.

RealNetworks currently claims about a 93 percent market share in the streaming media area with its proprietary Progressive Networks Media (PNM) protocol. Because of this market dominance, I was curious to know why a relatively small company would choose to standardize what seemed to be one of its key technologies.

So I tracked down Rob Lanphier—known as the RealNetworks "standards nerd"—and asked him a few questions.

—Charles Severance

At its fundamental level the Internet is based on a protocol—the Internet Protocol (IP)—that does not guarantee the delivery of any particular packet. To

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compensate for this potential loss of packets, another protocol—called the Transmission Control Protocol (TCP)—adds careful error correction by marking data with sequence numbers and carefully retransmitting any data that appears to have been lost. TCP/IP guarantees that all the data will be delivered in proper order, no matter how long it takes.

TCP/IP is not the ideal protocol for sending real-time data like compressed audio. In an oversimplified example, we could broadcast audio by breaking it into 10 packets and sending those packets across the network. On the receiving end, if a packet gets lost and we receive only nine packets, it is usually better to play the best possible sound using only nine packets, rather than pausing the audio for several seconds and requesting the single lost packet. By using clever error-correction methods when spreading the audio information across packets, the loss in quality may be barely noticeable.

A real-time streaming protocol is more concerned about *when* a packet is supposed to arrive than *if* a packet has arrived. And it is more important adjust for packet loss or delay than to correct for either. A standard for a streaming protocol, then, specifies how streaming protocols operate and allows streaming software from multiple vendors to interoperate.

SOME TERMINOLOGY

It takes more than just a real-time streaming protocol to deliver audio, video, or other media content across the Internet. There are several other components that are necessary in order to stream, say, video across the Internet. As shown in Figure 1, first you need an encoder that digitizes and compresses the original raw video into a form capable of being delivered on the Internet.

This data can either be sent directly across a network or stored in a file using a particular file format. Example file formats include RealMedia, Quicktime, and AVI. Once these files are placed on a server, they can be streamed across the Internet to the destination computer that plays the media. There are three activities that the destination (client) computer must perform through the media player.

The player must:

- understand the real-time streaming protocol,
- decode the media and perform error correction (as necessary), and then
- play the media.

This is all a very complex process that self-adjusts to network loads and delays several times per second, all the while attempting to make it appear as if the audio and/or video is playing without interruption.

STANDARDIZING RTSP

Now that we have a basic understanding of the technology, we can begin to look at why RealNetworks chose to standardize their protocol.

Charles Severance: Why did you choose to standardize RTSP when you had a dominant market share with PNM?

Rob Lanphier: We felt it was an important step in moving streaming media into the mainstream. A critical part of doing that is getting industry consensus so that an infrastructure can be built (including firewalls, caching, and so forth). The net effect that comes from having a uniform standard is that we can channel all of the energy and buzz surrounding streaming media into productive activity designed to make the Internet a mass medium rather than staking out turf in a market share war.

CS: Why did you choose the Internet Engineering Task Force (IETF)?

RL: We got together with Netscape and submitted RTSP to the IETF in November 1996. It was never a question who we would go to, because the IETF is the recognized authority in all things Internet. And RTSP was really designed for Internet use.

CS: You came into the IETF with a complete, successful standard in your hands. Did they rubber stamp it or did they change it?

RL: They did not rubber stamp it. The standard was not drastically altered from the time we introduced it until the time it was approved. However, there were changes that occurred during the process. Much of the change came either from people with experience in similar situations or from people involved in other parts of Internet standardization who could see how this standard might affect their particular area. The standard that was approved as RTSP is a better standard than the first draft that we brought into the process.

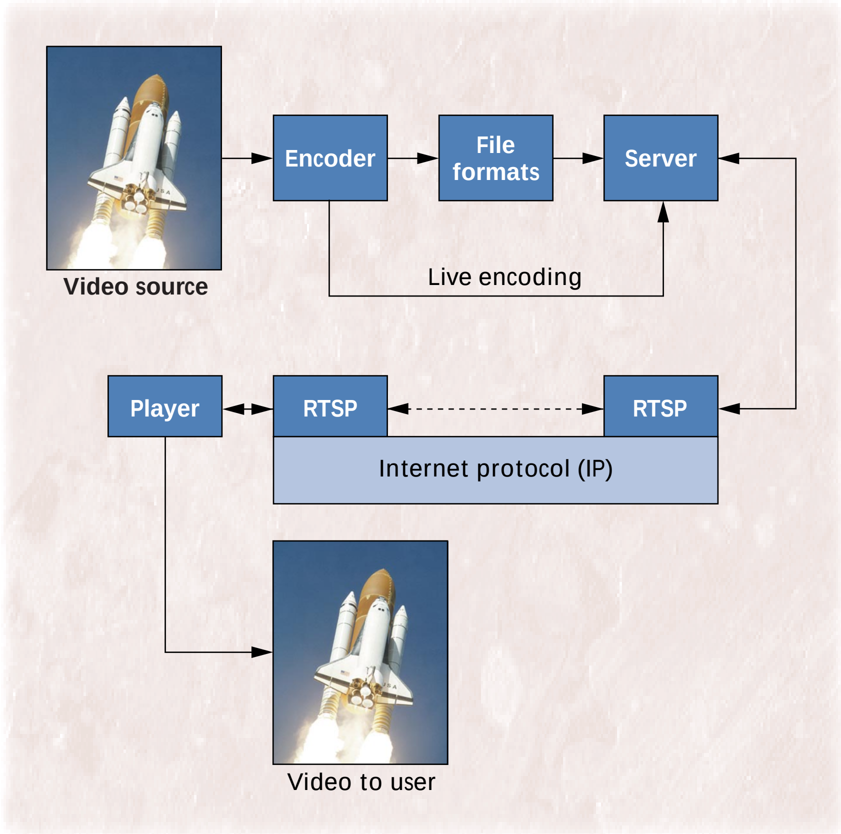

Figure 1. Streaming video across the Internet requires a number of components, including the player, which must understand the real-time streaming protocol, decode the media (while performing error correction), and then play the media.

MAKING THE MARKET SUCCESSFUL

CS: It is pretty convenient that the RTSP standard was approved in April and you announced the G2 product in April. How hard was that to pull off?

RL: Though not required, it's a practical reality that to get to the Proposed Standard stage where we are now, everyone needed to have enough implementation experience to be confident that we could reach subsequent stages of standardization (Draft Standard and Internet Standard). Since our implementation experience of RTSP is primarily based on G2, it was only natural that RTSP and G2 had to be done at roughly the same time, since shipping one is almost a necessary condition for shipping the other.

CS: Now that RTSP is a standard and any new company can create a compliant media player or media server, aren't you afraid that you will begin to lose market share?

RL: Not really. The technology that makes our media successful is not the underlying streaming protocol. When the company started, if an appropriate Internet streaming media standard existed, we would have adopted it. Our strength is providing a complete, cohesive end-to-end solution for the delivery of streamed media, of which RTSP is only a part. The complete solution includes encoders, codecs, APIs of all flavors, Netscape and Internet Explorer integration, and much more.

For example, our new G2 encoder and player can maintain very high audio quality over a modem with 10 percent packet loss. This is the best in the industry and far better than our RealPlayer 5.0 product. In fact, our new G2 architecture allows many forms of media to be streamed and played. Our new player can play Vivo streams, among other media formats.

Again, for us to be successful and keep growing, the entire market must be successful, which makes participating in standardization very important.

For more information on RTSP, see http://www.real.com/rtsp/. To examine the text of the RTSP standard itself, visit ftp://ftp.isi.edu/in-notes/rfc2326.txt. ❖

Rob Lanphier is the standards nerd for RealNetworks and can be reached at robla@real.com.

Correction

Last month we mislabeled Bob Schaumann's title. At Digital, he is Publisher of Strategic Standards Information and publishes the *Open Systems Standards Tracking Report*, a free monthly newsletter. Contact him at bob.schaumann@digital.com.