

Real Time Streaming Protocol

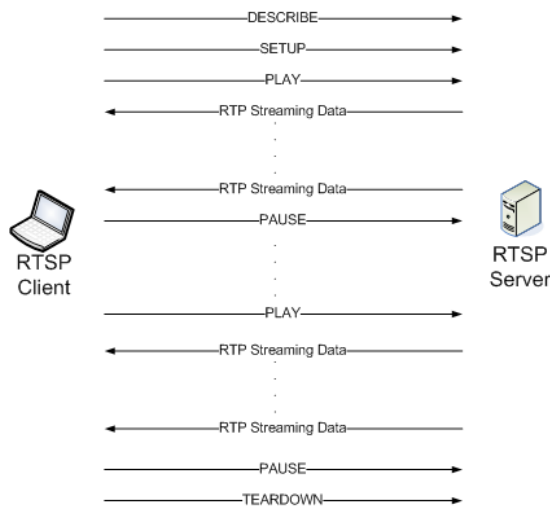
Emulate Hundreds of Thousands of RTP Streams controlled via RTSP

The Real Time Streaming Protocol (RTSP) was originally submitted to the Internet Engineering Task Force (IETF) by engineers from Netscape and Progressive Networks in November 1996. A product of the Multiparty Multimedia Session Control Working Group (MMUSIC WG), RTSP evolved through ten different drafts before being officially published as RFC 2326.

The purpose of RTSP is to provide a framework for on-demand multimedia content such as audio and video. RTSP can negotiate multimedia transmission over the Real-time Transport Protocol (RTP) and can handle multiple sessions simultaneously. The types of media that may be controlled via RTSP include both real-time data such as live feeds, as well as cached feeds. While RTSP generally negotiates a 3rd party protocol for streaming the actual media content, the capability does exist for RTSP to stream the content intertwined with the control mechanisms.

RTSP is a stateful protocol, although that state is not maintained by the layer 4 transport protocol, rather it uses an identifier to maintain state for each stream. Additionally, RTSP may run over both UDP and TCP on port 554, but since RTSP provides application-level reliability, rendering on TCP is less useful.

Figure 1



A typical RTSP setup scenario to stream multimedia content over RTP.

BreakingPoint Testing Tools Emulate the RTSP Protocol:

- One BreakingPoint Elite 3-slot chassis is able to generate hundreds of thousands of stateful RTSP streams
- Utilizing blended applications and security, users can fully test their network infrastructure to ensure RTSP/RTP quality of service conditions are met under full network load
- Utilizing Application Manager allows users to create RTSP/RTP flows to fully qualify RTSP stream establishment under both normal and abnormal protocol adherence

There are twelve unique commands which are used by RTSP to control the state machines on the clients and servers, resembling HTTP, including the reuse of status codes. The primary commands that are used for acting as a “remote control” for the media streams are “DESCRIBE”, “SETUP”, “PLAY”, “PAUSE” and “TEARDOWN”. The “DESCRIBE” command contains the resource indicator as well as a description in Session Description Protocol (SDP) format of the type of media that may be played. The “SETUP” command is responsible for selecting the method for streaming the media.

In order to setup a media stream, the client will send the “SETUP” command, RTSP sequence number and the method of transport with the appropriate port numbers to the server with the URI. The media server will respond with a “200 OK” message that includes the source port so the client will understand the incoming media stream. This transaction allows intermediate network devices such as firewalls to understand and allow the appropriate streams as they start to transmit data. The “PLAY” command commences the RTP streaming of the media, while “PAUSE” stops the RTP streaming. The “TEARDOWN” command terminates the streaming session and frees all resources on the remote server.

Figure 1 illustrates a typical RTSP sequence for streaming multimedia audio. After obtaining a URI, usually via HTTP, the RTSP client sends a “DESCRIBE” command followed by the “SETUP” command to finalize RTP parameters. Once the streams have been setup, a “PLAY” command is issued to start the multimedia stream. If desired, the “PAUSE” command may be issued to halt playback with “PLAY” reinitiating the stream. Once the client has completed its streaming, it issues the “TEARDOWN” command to free the streaming resources on the server.