

**Application Note**

**RTSP Extensions to  
Dialogic<sup>®</sup> 3G-324M  
Multimedia Gateway  
Demo and Dialogic<sup>®</sup>  
Multimedia Demo**

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

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## Executive Summary

This application note describes Real Time Streaming Protocol (RTSP) extensions to two existing demos, the Dialogic® 3G-324M Multimedia Gateway demo (also referred to as the 3G-324M demo) and the Dialogic® Multimedia demo (also referred to as the Multimedia demo). An open-source RTSP stack from Live Networks, Inc., called LIVE555, was used to add RTSP client functionality to those demos. Demo code changes, LIVE555 usage and Dialogic® device considerations for RTSP are explained.

“Developing Media Solutions using RTSP and Dialogic® Products” application note is intended to be used as a reference for this application note.

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# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

## Introduction

As described in [RFC2326], “Real Time Streaming Protocol, or RTSP, is an application-level protocol for control over the delivery of data with real-time properties. RTSP provides an extensible framework to enable controlled, on-demand delivery of real-time data, such as audio and video. Sources of data can include both live data feeds and stored clips. This protocol is intended to control multiple data delivery sessions, provide a means for choosing delivery channels such as UDP, multicast UDP and TCP, and provide a means for choosing delivery mechanisms based upon RTP.” [RFC2326]

This application note describes RTSP extensions to two existing demos, Dialogic® 3G-324M Multimedia Gateway demo (also referred to as the 3G-324M demo) and the Dialogic® Multimedia demo (also referred to as the Multimedia demo). It also describes how an open source project from Live Networks, Inc., called “LIVE555 Streaming Media” (<http://www.LIVE555.com/liveMedia/>), can be used for RTSP control, and how Dialogic® IPM devices can be used for the delivery of media to SIP and 3G-324M endpoints.

The application note, “Developing Media Solutions using RTSP and Dialogic® Products,” is intended to be used as a reference for this application note (see the For More Information section).

## RTSP-Enabled System

Figure 1 shows various endpoints (clients), an RTSP server, content feeds, and a multimedia system based on Dialogic® components in the heart of this system. The RTSP server not only streams media to endpoints connected via the Dialogic® server (in this document, the “Dialogic server” can be Dialogic® Host Media Processing Software [Dialogic HMP Software], Dialogic® Multimedia Software for AdvancedTCA [MMP Software for ATCA], or Dialogic® Multimedia Kit Software Release 1.0 for PCIe [MMK Software 1.0 for PCIe]), but also to the “web” infrastructure, as shown by the RTSP Client in Figure 1.

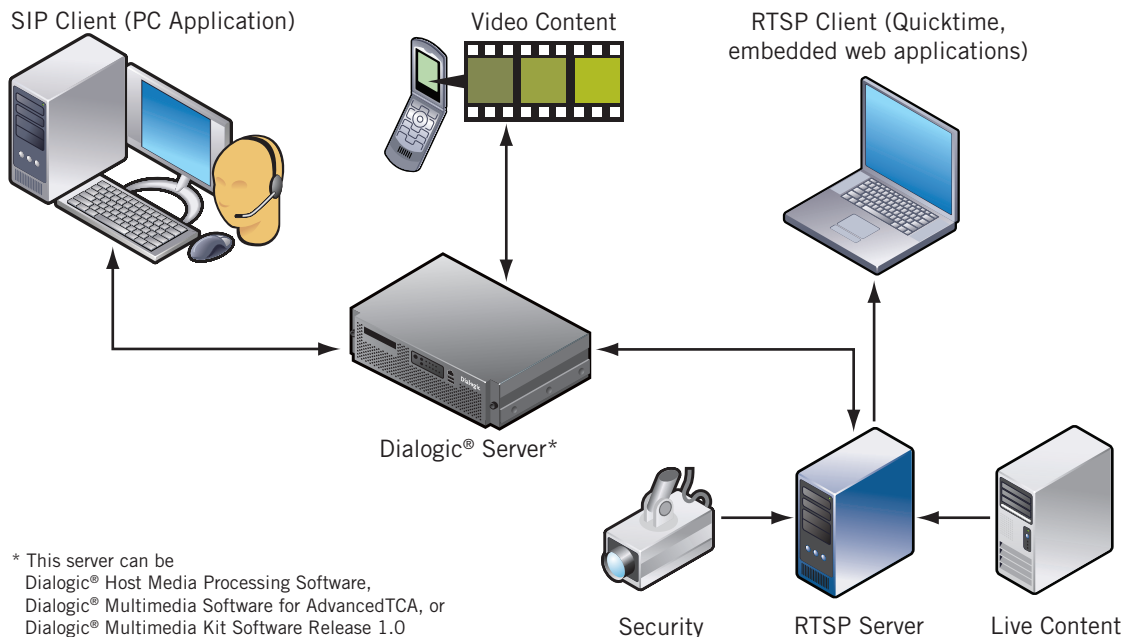


Figure 1. Multimedia System with RTSP

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## Protocols

The following protocols are used to enable the RTSP solution in the Dialogic server shown in Figure 1:

### 3G Endpoint

- ISDN, SS7, or BICC for establishment of call from 3G handset
- 3G-324M for video session media establishment between the server and the 3G User Equipment (3G UE)
- Media flow is bi-directional
- Digits are conveyed using a H.245 UII control message

Transport of call control and media may be over TDM or IP interfaces. The API for call establishment may be Dialogic® Global Call API. Media session establishment is accomplished using the M3G API.

### SIP Endpoint

- SIP for establishment of the call
- RTP for media transport
- Media flow is bi-directional
- Digits are conveyed using RFC2833 signals. In-band audio may be used for certain CODECs, but RFC2833 signals are the preferred method.

Transport of call control and media is via host IP interface or with IP interface on the Dialogic® Multimedia Accelerator Board for PCIe (MMA Boards) or the Dialogic® Multimedia Platform for AdvancedTCA (MMP for ATCA). The API for call establishment may be Global Call API. Media session establishment is accomplished using the IPM API.

## RTSP Server

The Dialogic server needs to act as an RTSP Client. RFC2326 describes this protocol. The demo applications use Live Networks' RTSP stack for establishment of the media session.

Media is uni-directional. In the case of an endpoint wanting to view (Play) some content, RTSP streams content through the Dialogic server to the endpoint. In case of an endpoint wanting to store (Record) some content from endpoint, the RTSP server receives the media from the endpoint through the Dialogic server.

Transport of the RTSP session establishment between the Dialogic server and the RTSP server is TCP-based. The Dialogic server is considered an RTSP Client in this context. The media flow between the Dialogic server and the RTSP server occurs using RTP/UDP/IP.

## 3G Phone and RTSP Server

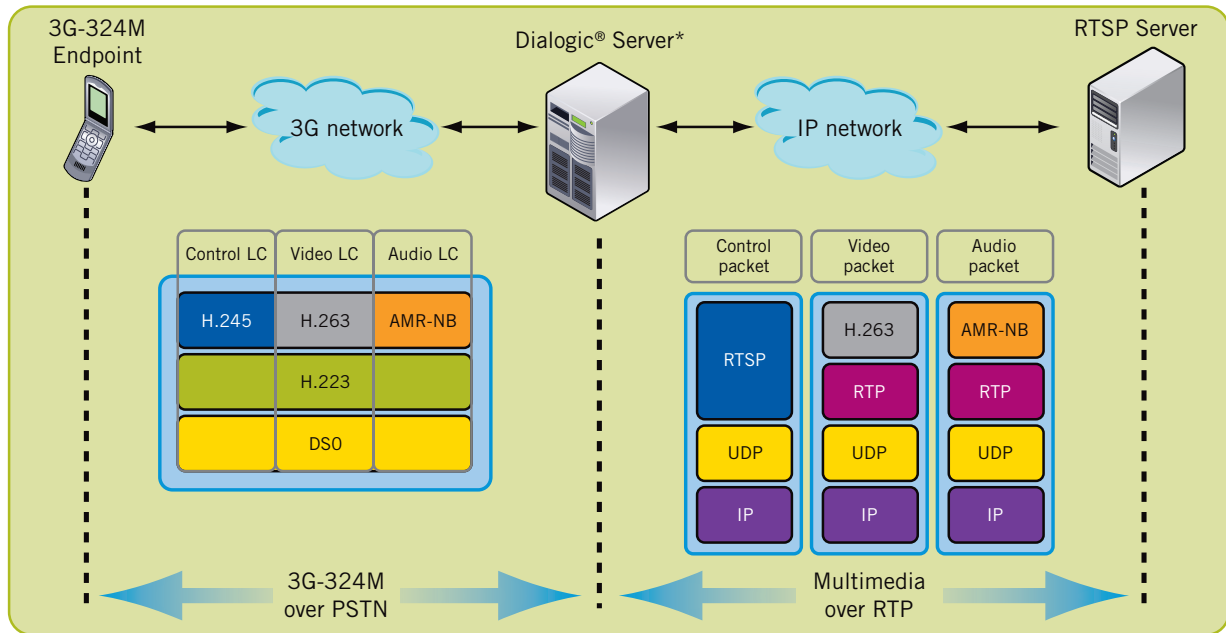
Figure 2 shows the control and media flow between a 3G-324M endpoint and an RTSP server. From the Dialogic® device's perspective, the application connects an m3g device's audio and video ports to an IPM device. The IPM device in turn connects to the RTSP server media using RTP.

Dialogic® device and network connections are shown in Figure 3 and Figure 4. When the 3G handheld connects to the Dialogic-based service, it will negotiate either H.263 or MPEG4 for video and AMR-NB for audio.

**Note:** Using the AMR-NB resource in connection with one or more Dialogic products mentioned herein does not grant the right to practice the AMR-NB standard. To seek a patent license agreement to practice the standard, contact the VoiceAge Corporation at <http://www.voiceage.com/licensing.php>.

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\* This server can be  
Dialogic® Host Media Processing Software,  
Dialogic® Multimedia Software for AdvancedTCA, or  
Dialogic® Multimedia Kit Software Release 1.0

Figure 2. Control and Media Flow between the 3G (Release 99) Endpoint and the RTSP Server

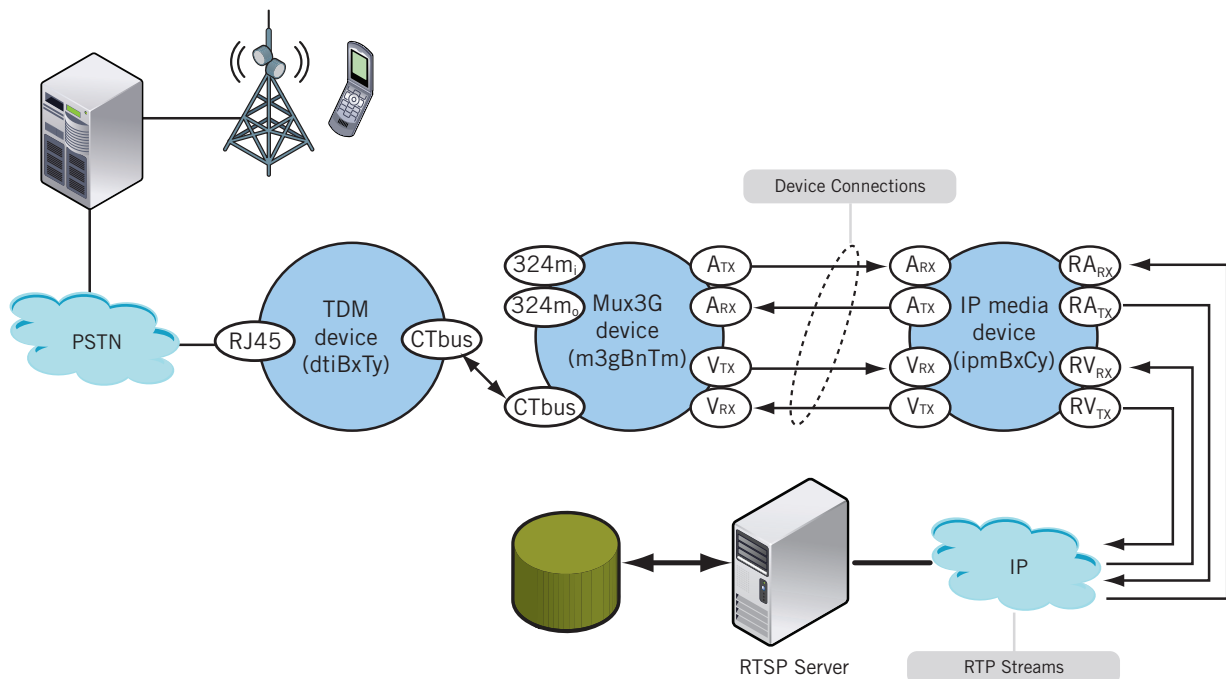


Figure 3. 3G (Release 99) to RTSP Server

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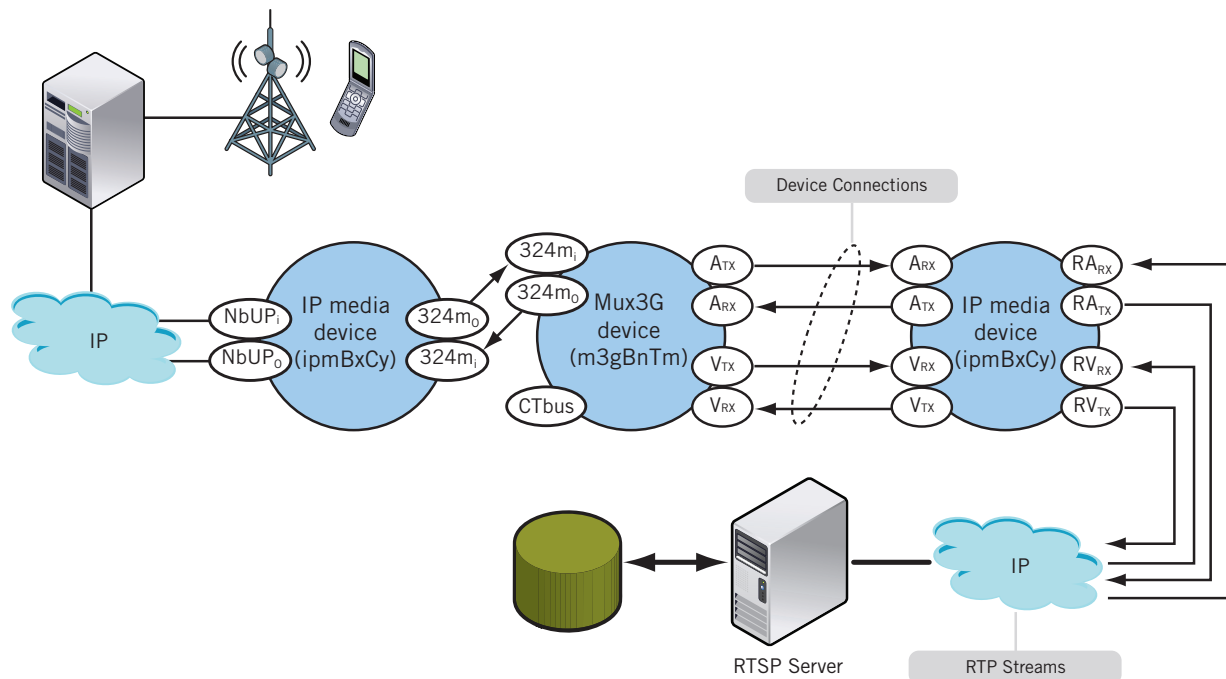


Figure 4. 3G (Rel 4/5) to RTSP Server

## RTSP Stack

### Installing the RTSP Stack

Installation of the RTSP Stack can be done as follows:

1. Download Live Networks, Inc.'s stack source-code from <http://www.LIVE555.com/liveMedia/public/>
2. Uncompress the downloaded archive into a directory. For example,
  - a. `mkdir /export/RTSP`
  - b. `cd /export/RTSP`
  - c. `tar -xvzf live.2008.07.25.tar.gz`
3. Files should be uncompressed into `/export/RTSP/live` after step 2.c.
4. Make live555 libraries:
  - a. `cd live`
  - b. `./genMakefiles linux`
  - c. `make`

### Using RTSP Stack with Dialogic Demos

The following two demo applications make use of RTSP stack installed in the "Installing the RTSP Stack" section:

- 3G-324M demo
- Multimedia demo

### Dialogic Demos' Makefile Changes

Update the location of LIVE555 stack source code as follows:

1. Makefiles included with the 3G-324M and Multimedia demos contain an environment variable for the location of LIVE555 stack. The variable is LIVE555. Assuming the stack was installed per instructions in the "Installing the RTSP Stack" section, in `/export/RTSP/live` directory, modify the Makefile so that:

```
LIVE555 = /export/RTSP/live
```

2. Allow use of RTSP as follows:

The default behavior of the demos is to function without RTSP

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support. The demos' Makefile includes a flag for allowing the demo to operate in RTSP mode. This flag is set in Makefile with the USE\_RTSP variable:

To use RTSP: uncomment the line, so that line is:  
"USE\_RTSP=1"

To disable RTSP: comment out the line: "#USE\_RTSP=1"

**Note:** Only the "SIP to RTSP" and "3G to RTSP" functions of these demos are supported. Other demo variations (such as SIP to MM, 3G to SIP, and so on) are not supported by the RTSP version of the demos.

## RTSP Stack Code Integration

### RTSP Client Class

A simple wrapper class, CRTSPClient, was created around the Live555 RTSP stack. The class resides in CRTSPClient.cpp/h files. The 3G and Multimedia demos both use this class.

The following functions are of interest:

- **CRTSPClient(unsigned short ipmRxAudioPort, unsigned short ipmRxVideoPort, RTSPEndpoint\* pRTSPEndpoint)**

Constructor — If this is the first RTSP Client object allocated by the application, the underlying Live RTSP stack environment, which is based on a Singleton, is initialized.

The application specifies a particular IPM device's audio and video ports to which the RTSP server will stream. The Dialogic application will control a set of IPM devices. Each IPM device will get its own CRTSPClient object.

The object is created every time an RTSP session is requested, and the object is destroyed once another RTSP session is requested, and a new CRTSPClient object is created. For example:

- A. New inbound SIP or 3G call, create CRTSPClient object for URL A.
- B. User presses a DTMF button to view another stream.
- C. Object associated with URL A is destroyed and another CRTSPClient object is created for URL B.

- **CreateRTSPSession(const char\* url, bool startPlayAlso)**

Before the application calls ipm\_StartMedia(), this function is called to establish an RTSP session between the specified URL and IPM ports provided during object construction. The following steps are taken within this function:

1. Validate URL, including parsing of the IP address in the URL.
2. Send OPTIONS command to the RTSP server and get the response. OPTIONS response is currently not used by the sample applications.
3. Get media description (Session Description Protocol [SDP]) using the DESCRIBE command. Information retrieved during the SDP contains the URL's media information, such as CODEC types, payload types. See the SDP's RFCs for more information.
4. Send two SETUP messages to the RTSP server, one for the audio stream and another for the video stream. The RTSP server is provided with the IPM receive port numbers in these SETUP messages. The RTSP server's response contains the RTSP source UDP addresses that the IPM device needs in order to receive the RTP media. The RTSP server, at this point, allocates the resources to stream, but does not actually start streaming.
5. Optionally, start the RTSP media stream (see Play()).

- **Play()**

Send PLAY messages to the RTSP server. Two PLAY messages are sent, one for audio stream and another for video stream.

The RTSP stream is now active.

- **Teardown()**

The 3G and Multimedia demo applications use this method to stop an RTSP session already in progress. The demos call this method upon receiving UII or DTMF from the endpoint.

**Note 1:** The SDP contains media duration; for example, 151 seconds is the duration in: "a=range:npt=0-150.900". Notably, when the application calls Play() above, it would kick off a timer to know when to call Teardown(). However, this functionality is not implemented in the demos.

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**Note 2:** When a user enters a DTMF during an existing RTSP stream, the current RTSP stream is torn down using Teardown(), the RTSP client object is destroyed, and a new object is created for the next RTSP session.

**Note 3:** When an RTSP media stream stops, the user viewing the stream will normally press another DTMF or hang up. When either of these actions takes place, the current RTSP client object is destroyed.

The following functions return information (use with ipm\_StartMedia() ):

**Note:** See RFC2326 for further information on the various messages discussed.

- **decoder\_config\_info()**

Use with MPEG4. If “mpeg4” is the CODEC type, this config information is available, but not used by the demo.

- **GetAudioPayloadType()**
- **GetVideoPayloadType()**
- **GetAudioCodecName()**

If “AMR-NB”, IPM CODEC type CODER\_TYPE\_AMRNB\_12\_2k is used. Otherwise, CODEC is assumed to be CODER\_TYPE\_G711ULAW64K.

RTSP streamed 3gp files should contain AMR-NB audio.

**Note:** Using the AMR-NB resource in connection with one or more Dialogic products mentioned herein does not grant the right to practice the AMR-NB standard. To seek a patent license agreement to practice the standard, contact the VoiceAge Corporation at <http://www.voiceage.com/licensing.php>.

- **GetVideoCodecName()**

“H263-1998” and “H263-2000” are treated as IPM CODER\_TYPE\_H263\_1998. These are supported in Native device connection mode only.

“H263” and “MP4V” are other CODECs supported by the demos

- **GetRTSPAudioPort()**
- **GetRTSPVideoPort()**
- **GetRTSPAudioIP()**
- **GetRTSPVideoIP()**
- **GetRTSPVideoProfile()**

Used to determine QCIF or CIF resolution. If the profile is 3, CIF resolution is used. Otherwise, QCIF resolution is assumed.

## Multimedia Demo

The modified Multimedia demo is included with Dialogic HMP Software, MMP Software for ATCA, and MMK Software 1.0 for PCIe releases. This demo, as shipped with the associated product, allows inbound SIP calls. Once a call connects, the demo plays locally stored audio/video files using the Dialogic® Multimedia API.

The modified Multimedia demo does not use the MM device for audio/video playback. Instead, a second IPM device is used to stream media from an RTSP server to the incoming SIP call. See Figure 5 for the configuration realized by the modified Multimedia demo.

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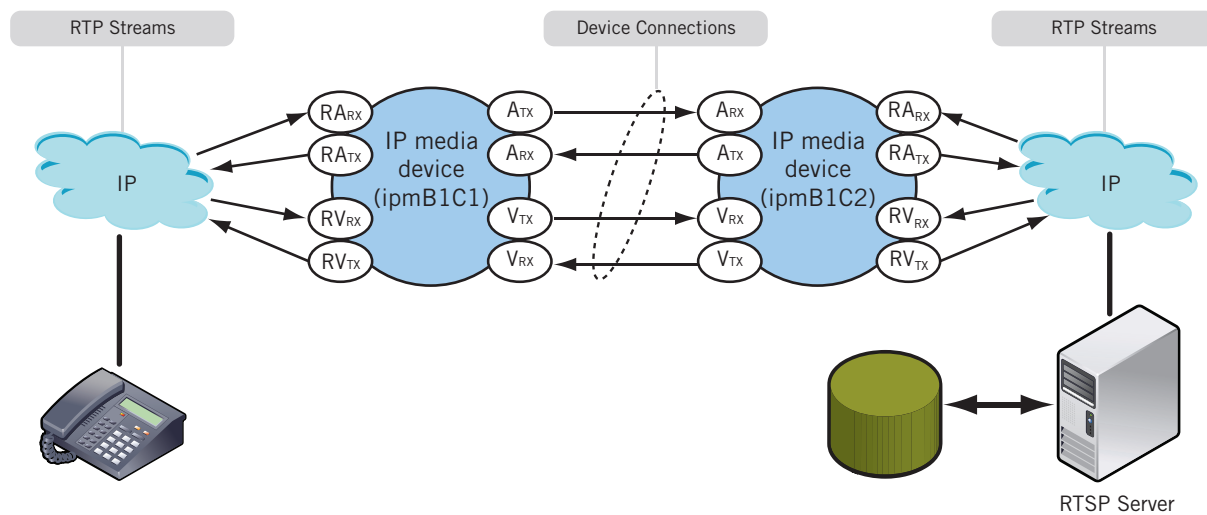


Figure 5. RTSP Server to SIP Call

During application startup, two IPM devices are routed to each other with Transcoding or Native mode, based on the xcode parameter in the config file.

During application runtime, upon receiving a SIP call, the SIP call is answered. Next, the application creates an RTSPClient object, and `CreateRTSPSession(url, startPlay=false)` is called. At this point, the application either gets an error from creating the RTSP session or the application has the RTSP media information needed to start RTP streaming using `ipm_StartMedia()`.

Once the SIP to RTSP session is established, pressing a DTMF button will cause the demo to look for the RTSP URL associated with the input DTMF. If a URL is configured for the DTMF, the current RTSP session:

1. Will be torn down.
2. `ipm_StopMedia()` will be initiated
3. A new RTSP session setup will be performed.
4. `ipm_StartMedia()` will be initiated based on new session setup.

When the SIP call is disconnected, `ipm_StopMedia()` is called on both the SIP and RTSP IPM devices.

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Figure 6 shows the Multimedia demo SIP RTSP interaction.

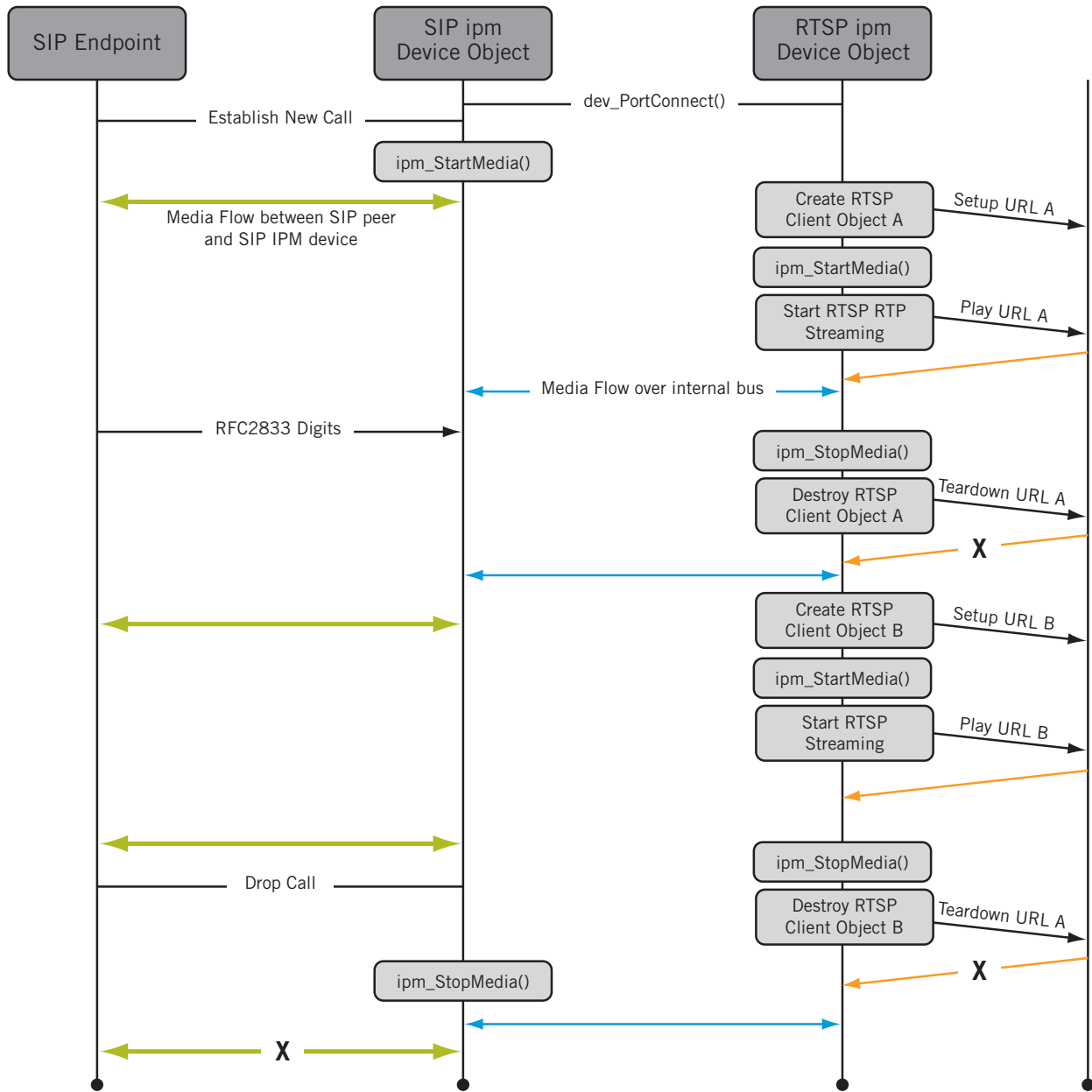


Figure 6. Multimedia Demo SIP RTSP Server Interaction

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Multimedia demo config file changes:

Configuration items of importance to the RTSP-capable Multimedia demo are lines that begin with “xcode” and “RTSPINFO” parameters.

1. “xcode” determines use of Transcoding or Native connection between two IPM devices.

**Note:** H.263+ support is with Native only.

2. RTSPINFO line maps a DTMF to an RTSP URL. When the user presses the configured DTMF, the corresponding RTSP URL session is established and streamed to the user. This line has the following template:

- a. RTSPINFO: <dtmf> <RTSP URL>

- b. Example: “RTSPINFO: 0 rtsp://146.152.82.228/shuttle\_landing.3gp”

— When the user presses DTMF “0”, the RTSP stream in the example plays.

- c. The URL must have numeric IP address.

## 3G-324M Demo

The modified 3G-324M demo is included with Dialogic HMP, MMP Software for ATCA, and MMK Software 1.0 for PCIe releases. This demo, as shipped with the applicable product, allows inbound 3G or SIP calls. When the call connects, either an outbound call is made to a 3G or SIP endpoint or an audio/video file is played back. For more details on various configurations with the 3G-324M demo, consult the *Dialogic® 3G-324M Multimedia Gateway Demo Guide* (see the For More Information section).

See Figure 4 for the configuration realized by the RTSP configuration of the 3G-324M demo.

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Figure 7 shows the interaction between a 3G call and an RTSP server. The 3G and RTSP IPM objects are created based on the demo configuration. During demo initialization, the objects are instantiated. The m3g and IPM devices' internal ports are also connected to each other based on the last two columns of endpoint configuration lines. See the example configuration that follows.

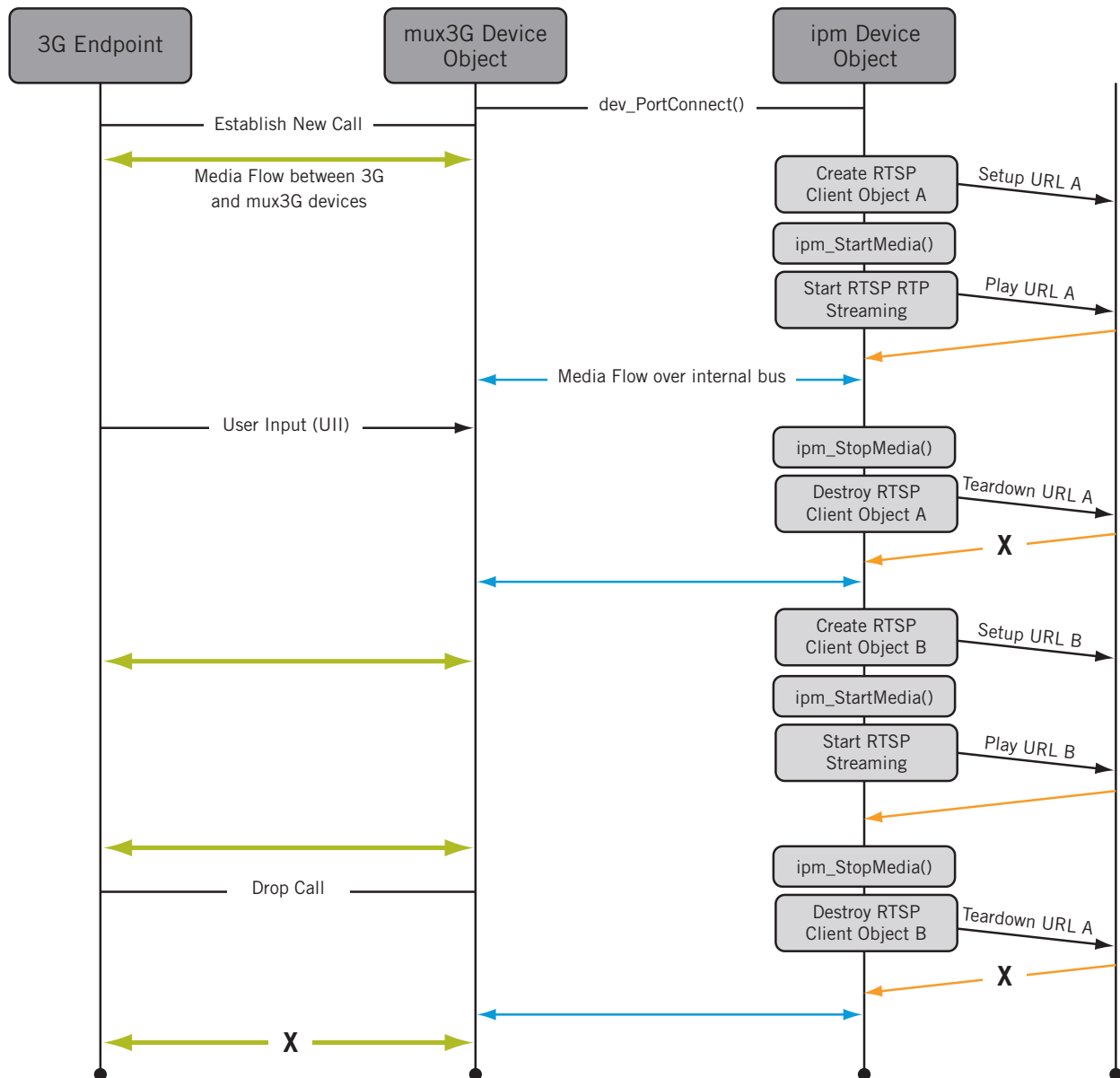


Figure 7. 3G-324M Demo 3G to RTSP Server Interaction

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3G-324M demo config file changes:

Configuration items of importance to the RTSP-capable 3G-324M demo are lines for RTSP endpoints and “RTSPINFO” parameters.

1. RTSPINFO line maps a DTMF to an RTSP URL. When the user presses the configured DTMF, the corresponding RTSP URL session is established and streamed to the user. This line has the following template:

a. RTSPINFO: <dtmf> <RTSP URL>

b. Example: RTSPINFO: 0 rtsp://146.152.82.228/shuttle\_landing.3gp

— When the user presses DTMF “0”, the RTSP stream in this example plays.

— On an inbound call, the first clip to be streamed is the one associated with DTMF “0”.

c. The URL must have a numeric IP address.

2. RTSP Endpoint:

a. An RTSP endpoint type needs to be paired with an m3g endpoint type in order to bridge the 3G call to an RTSP server.

b. Endpoint type configuration details are discussed in the Dialogic® 3G-324M Multimedia Gateway Demo Guide. A basic explanation, pertaining to RTSP, is described here.

c. Example config lines:

```
EP: 1 m3g m3gB1T1 IdtiB5T1 2 0 none N N
EP: 2 rtsp none ipmB1C3 1 0 none T T
```

— Endpoint 1 specifies the 3G configuration. dtiB5T1, running the ISDN protocol, is connected to m3gB1T1 to handle the 3G session.

—This endpoint is paired with endpoint 2; see next line for endpoint 2 details.

— Endpoint 3 is an RTSP endpoint due to “rtsp” in the “endpoint type column”.

— ipmB1C3 is the IPM device that will stream RTSP-related RTP audio/video data.

— This endpoint is paired with endpoint 1. Endpoint 1 specified the 3G configuration in this example.

— Last two columns of endpoints 1 and 2 signify the Audio and Video port connection method. An “N” causes the transmit port of a specific endpoint to connect to its peer endpoint using the DMFL\_TRANSCODE\_NATIVE flag. A “T” causes the transmit port of a specific endpoint to connect to its peer endpoint using DMFL\_TRANSCODE\_ON flag. In this example, m3gB1T1 will transmit to ipmB1C3 natively. In the opposite direction, ipmB1C3 will transmit to m3gB1T1 with transcoding for Audio and Video ports.

## CODEC Support

RTSP servers typically stream content as it was originally authored. General observations have shown that the servers do not transcode the media from one format to another. Typically, the media is stored in 3gp files.

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## 3GP Files

The 3gp files are containers of audio and video tracks. The audio and video tracks are encoded using particular CODECs. Audio CODECs can be AMR-NB\*, AAC, and so on. Dialogic supports AMR-NB\* but not AAC. Video CODECs can be H.263, MPEG4, and others. Dialogic supports H.263 and MPEG4. For H.263 and MPEG4 CODEC support details, refer to the specific Dialogic® product Release Guide. Only some special cases surrounding CODECs are discussed as follows:

**\*Note:** Using the AMR-NB resource in connection with one or more Dialogic products mentioned herein does not grant the right to practice the AMR-NB standard. To seek a patent license agreement to practice the standard, contact the VoiceAge Corporation at <http://www.voiceage.com/licensing.php>.

### **H.263+ (H.263-1998), H.263++ (H.263-2000)**

3gp files containing H.263 encoded video may be streamed using RFC2429 RTP packetization.

As of 3Q'08, H.263+ and H.263++ CODEC support in Dialogic® products implies support of RFC2429. Furthermore, an H.263 encoded RFC2429 RTP stream is supported in Native mode, meaning an IPM device is connected to another IPM or m3g device using the `dev_PortConnect()` API with `DMFL_TRANSCODE_NATIVE` flag. See “Device Connections” in Figure 3 and Figure 4.

### **AMR-NB**

AMR-NB encoded media stream in a 3gp file may be transmitted by the RTSP server with multiple AMR-NB frames per RTP packet. This is usually due to how the 3gp file was originally authored.

**Note:** Using the AMR-NB resource in connection with one or more Dialogic products mentioned herein does not grant the right to practice the AMR-NB standard. To seek a patent license agreement to practice the standard, contact the VoiceAge Corporation at <http://www.voiceage.com/licensing.php>.

The IPM device receiving the RTP stream from the RTSP server can support up to ten Frames Per Packet (FPP).

Various application scenarios, their device usage, and FPP behavior is provided in Table 1.

Application	Endpoint Facing Device	Device Connect Flag	RTSP Facing Device	FPP from RTSP	FPP to Endpoint
RTSP to SIP	ipmB1C1	DMFL_TRANSCODE_NATIVE	ipmB1C2	N (N = 1...10)	N (N=RTSP Tx)
RTSP to SIP	ipmB1C1	DMFL_TRANSCODE_ON	ipmB1C2	N (N = 1...10)	M (application configurable, 1...10, typically 1)
RTSP to 3G UE	m3g*	DMFL_TRANSCODE_NATIVE	ipmB1C1	N (N = 1...10)	NA – 3G does not use RTP
RTSP to 3G UE	m3g*	DMFL_TRANSCODE_ON	ipmB1C1	N (N = 1...10)	NA – 3G does not use RTP

\* m3g device is also connected to dti (Rel 99) or IPM (Rel 4/5) for 3G-324M data transport. See Figures 3 and 4.

Table 1. Application Scenarios, Device Usages, and FPP Behavior

## Hinted Tracks

3gp files also contain metadata describing their content. 3gp files may also contain what is known as hinted tracks. The audio and video tracks contain the actual encoded sound and video samples. The hinted tracks provide packetization information about the audio and video tracks. If hinted tracks are present in a 3gp file, the RTSP server may use the hinted track to seek, packetize, and transmit the audio and video streams using RTP. If hinted tracks are not present, an RTSP server, if it supports streaming non-hinted 3gp files, may need to index and packetize the media data when necessary in order to stream the media using RTP.

## Creating 3gp Files

It is critical that the 3gp files be created properly for target endpoint environment. Average bit rate for files being streamed to 3G handset endpoints should not exceed 42 kbps. Average bit rate for files being streamed to IP endpoints should not exceed 384 kbps. However, the IP network may have limits that in turn may require the bit rate to be less than 384 kbps.

## Installing the Darwin RTSP Server

Darwin is an open-source RTSP server. It can be installed on Mac OS X Server, Windows®, Linux, and Solaris. It can be downloaded from: <http://dss.macosforge.org/>.

The current version of the RTSP server is provided in source code for OSs other than Mac OS X. Previous versions are available in binary form for other OSs. Previous versions can be downloaded using the “Previous releases” link on <http://dss.macosforge.org/>.

## Steps for Installing the RTSP Server on Linux

1. uncompress tgz file.
2. Run Install script: “./Install”.
3. Copy .3gp files to /usr/local/movies.

## Running Darwin Streaming Server

1. Run “DarwinStreamingServer” if it is not already running. It should be in the path.
2. Check process list for DarwinStreamingServer.

## Testing the RTSP Server Setup

Tools that aid in testing the setup are Wireshark, Quicktime, and/or VLC. Quicktime and VLC are some of the applications that can play content from the RTSP server. Wireshark aids in tracing the messages between the RTSP Client and Server.

Test the RTSP server with standalone RTSP client first. Once the RTSP server is deemed working, test with the 3G-324M demo.

During the course of navigating various URLs through the demos, if a “NULL options response” message is seen, it most likely means that the URL is invalid. Check the configuration of the demo, as well as the RTSP server.

Various RTSP servers were tested with the modified demos. Some information about downloading video files from the internet for testing are provided in Appendix A. Sample RTSP traces from various RTSP servers are provided in Appendix B.

## References

[RFC2326] Real Time Streaming Protocol (rfc2326) —  
<http://www.ietf.org/rfc/rfc2326.txt>

## Appendix A: RTSP Protocol Sample

A video of the Space Shuttle landing was downloaded from YouTube using <http://www.savevid.com> and converted to a 3gp file using Helix mobile producer. Helix mobile producer was used to convert the downloaded file to a 3gp file (3GPPv5) with following attributes:

- MPEG 4 (37.8 kbps)
- AMR-NB (12.2 kbps)
- 176 x 144 resolution

Original video can be viewed at: <http://www.youtube.com/watch?v=YOxZsbyjSb8>.

The converted 3gp file contains several tracks, as shown by MP4Box. MP4Box can be downloaded from [http://gpac.sourceforge.net/doc\\_mp4box.php](http://gpac.sourceforge.net/doc_mp4box.php).

## Shuttle\_Landing.3gp File Information

The output that follows shows information about the shuttle\_landing.3gp file that is used in Appendix B, which contains an RTSP network trace. Some items of interest in this output are:

- **Tracks** — Tracks are audio and video tracks, along with hint information for those tracks. During RTSP SETUP phase, track ID is provided by the client to serve to the request the desired audio and/or video track. Some RTSP servers will play the hinted track and others may play the raw track. See the upcoming table, “Tracks Played by Various Servers.”
- **Video resolution** — Track ID 201 shows the video resolution is 176x144 Using the RTSP Stack with Dialogic Demos. This is QCIF.
- **Video CODEC** — Track ID 201 also indicates the CODEC. It is mpeg4, simple profile @ Level 0.
- **Payload types** — Payload types are shown as 96 for the video track and 97 for audio track.
- **Audio CODEC** — Track ID 101 indicates AMR-NB as the CODEC.

```
# MP4Box -info shuttle_landing.3gp
* Movie Info *
  Timescale 90000 - Duration 00:02:31.000
  Fragmented File no - 6 track(s)
  File Brand 3gp5 - version 0
  Created: GMT Mon Jul 21 19:02:47 2008

File has root IOD
Scene PL 0xff - Graphics PL 0xff - OD PL 0xff
Visual PL: Simple Profile @ Level 0 (0x08)
Audio PL: Not part of MPEG-4 audio profiles (0xfe)
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
Track # 1 Info - TrackID 201 - TimeScale 90000 - Duration 00:02:30.900
Media Info: Language "Undetermined" - Type "vide:mp4v" - 1509 samples
MPEG-4 Config: Visual Stream - ObjectTypeIndication 0x20
MPEG-4 Visual Size 176 x 144 - Simple Profile @ Level 0
Pixel Aspect Ratio 1:1 - Indicated track size 176 x 144
Synchronized on stream 1
```

```
Track # 2 Info - TrackID 65335 - TimeScale 90000 - Duration 00:02:31.000
Media Info: Language "Undetermined" - Type "hint:rtp " - 1510 samples
Streaming Hint Track for track ID 201
    Payload ID 96: type MP4V-ES/90000
```

```
Track # 3 Info - TrackID 101 - TimeScale 8000 - Duration 00:02:30.900
Media Info: Language "Undetermined" - Type "soun:samr" - 7543 samples
    3GPP AMR stream - Sample Rate 8000 - 2 channel(s) 16 bits per samples
```

```
Track # 4 Info - TrackID 65435 - TimeScale 8000 - Duration 00:02:31.000
Media Info: Language "Undetermined" - Type "hint:rtp " - 755 samples
Streaming Hint Track for track ID 101
    Payload ID 97: type AMR/8000
```

```
Track # 5 Info - TrackID 1 - TimeScale 90000 - Duration 00:02:30.900
Track is present in Root OD
Media Info: Language "Undetermined" - Type "odsm:mp4s" - 1 samples
MPEG-4 Config: ObjectDescriptor Stream - ObjectTypeIndication 0x01
Self-synchronized
```

```
Track # 6 Info - TrackID 2 - TimeScale 90000 - Duration 00:02:30.900
Track is present in Root OD
Media Info: Language "Undetermined" - Type "sdsm:mp4s" - 1 samples
MPEG-4 Config: SceneDescription Stream - ObjectTypeIndication 0x02
BIFS Scene description - Command stream
Synchronized on stream 1
```

## Tracks Played by Various Servers

Server	Audio Track/Stream ID Provided in SDP	Video Track/Stream ID Provided in SDP
Lucent	control:trackID=101	control:trackID=201
Helix	control:streamid=65435	control:streamid=65335
Darwin	control:trackID=65435	control:trackID=65335

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

## SDP Video Bandwidth Attributes Provided

Lucent	Helix	Darwin
b=TIAS:32800	b=TIAS:29193	
b=AS:40	b=AS:35	b=AS:36
b=RR:1000	b=RR:1094	
b=RS:1000	b=RS:364	

## SDP Audio Bandwidth Attributes Provided

Lucent	Helix	Darwin
b=TIAS:12200	b=TIAS:12828	
b=AS:18	b=AS:15	b=AS:14
b=RR:450	b=RR:481	
b=RS:450	b=RS:160	

## SDP Video a= Attributes Provided

Lucent	Helix	Darwin
a=maxprate:14.0	a=maxprate:10.108380	
a=control:trackID=201	a=control:streamid=65335	a=control:trackID=65335
a=rtpmap:96 MP4V-ES/90000	a=rtpmap:96 MP4V-ES/90000	a=rtpmap:96 MP4V-ES/90000
a=range:npt=0-150.900	a=range:npt=0-151	a=range:npt=0-150.89999389
	a=length:npt=151	
a=fmtp:96 profile-level-id=8; config=000001b008000001b50e a020202f000001000000012000c7 88ba9850584121463f;decode_ buf=2101	a=fmtp:96 profile-level-id=8; config=000001B008000001B50EA02 0202F000001000000012000C788BA 9850584121463F	a=fmtp:96 profile-level-id=8; config=000001B008000001B50EA02 0202F000001000000012000C788BA 9850584121463F
	a=cliprect:0,0,144,176	a=cliprect:0,0,144,176
	a=mimetype:string;"video/ MP4V-ES"	a=mpeg4-esid:201
	a=3GPP-Adaptation-Support:1	a=x-envivio-verid:00035A32
	a=Helix-Adaptation-Support:1	
	a=AvgBitRate:integer;29193	
	a=AvgPacketSize:integer;361	
	a=Width:integer;176	
	a=Height:integer;144	
	a=HasOutOfOrderTS:integer;1	
	a=Preroll:integer;1000	
	a=mpeg4-esid:201	
	a=x-envivio-verid:00035A32	

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

SDP Audio a= Attributes Provided

Lucent	Helix	Darwin
a=maxptime:200		
a=control:trackID=101	a=control:streamid=65435	a=control:trackID=65435
a=maxprate:10.0	a=maxprate:5.010937	
a=rtpmap:97 AMR/8000	a=rtpmap:97 AMR/8000	a=rtpmap:97 AMR/8000
a=range:npt=0-150.900	a=range:npt=0-151	a=range:npt=0-150.89999389
	a=length:npt=151	
a=fmtp:97 mode-set=7;octet-align=1;decode_buf=32	a=fmtp:97 octet-align=1	a=fmtp:97 octet-align=1
	a=mpeg4-esid:101	a=mpeg4-esid:101
	a=mimetype:string;"audio/AMR"	a=x-envivio-verid:00035A32
	a=3GPP-Adaptation-Support:1	
	a=Helix-Adaptation-Support:1	
	a=AvgBitRate:integer;12828	
	a=AvgPacketSize:integer;320	
	a=HasOutOfOrderTS:integer;1	
	a=Preroll:integer;1000	
	a=x-envivio-verid:00035A32	

## Appendix B: RTSP Protocol Trace

### Using the Lucent PVNS Server

```
OPTIONS rtsp://146.152.82.223/public/shuttle_landing.3gp RTSP/1.0
CSeq: 55
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 55
Date: Mon, 11 Aug 2008 17:00:14 GMT
Supported: method.eos, com.pv.server_playlist, com.pvns.proxy-exchange
Server: PVSS/5.0.1_080124 (03:32:53 Jan 24 2008)
Public: DESCRIBE, SETUP, TEARDOWN, PAUSE, PLAY, OPTIONS, SET_PARAMETER

DESCRIBE rtsp://146.152.82.223/public/shuttle_landing.3gp RTSP/1.0
CSeq: 56
Accept: application/sdp
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 56
Date: Mon, 11 Aug 2008 17:00:14 GMT
Supported: method.eos, com.pv.server_playlist, com.pvns.proxy-exchange
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
Content-Type: application/sdp
Content-Base: rtsp://146.152.82.223/public/shuttle_landing.3gp/
Content-Length: 736
Server: PVSS/5.0.1_080124 (03:32:53 Jan 24 2008)

v=0
o=- 3427462814 2259191454 IN IP4 146.152.82.223
s=
e=support@localhost
c=IN IP4 146.152.82.223
t=0 0
a=creation_date:20080719T190247.000Z
a=range:npt=0-151
a=X-wmfversion:1.2
a=control:*
a=etag:08FB0C8EDE96BCB254EE680F7B1689B6
m=video 0 RTP/AVP 96
b=TIAS:32800
b=AS:40
b=RR:1000
b=RS:1000
a=control:trackID=201
a=maxprate:14.0
a=rtptime:96 MP4V-ES/90000
a=range:npt=0-150.900
a=fmtp:96 profile-level-id=8;config=000001b008000001b50ea020202f000001000000012000c788ba9850
584121463f;decode_buf=2101
m=audio 0 RTP/AVP 97
b=TIAS:12200
b=AS:18
b=RR:450
b=RS:450
a=control:trackID=101
a=maxprate:10.0
a=rtptime:97 AMR/8000
a=range:npt=0-150.900
a=fmtp:97 mode-set=7;octet-align=1;decode_buf=32
a=maxptime:200

SETUP rtsp://146.152.82.223/public/shuttle_landing.3gp/trackID=201
RTSP/1.0
CSeq: 57
Transport: RTP/AVP;unicast;client_port=57350-57351
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
RTSP/1.0 200 OK
CSeq: 57
Date: Mon, 11 Aug 2008 17:00:14 GMT
BufferSize: 7000
Session: 10396e24910aa8b8;timeout=60
Supported: method.eos, com.pv.server_playlist, com.pvns.proxy-exchange
Transport: RTP/AVP;unicast;client_port=57350-57351;server_port=3018-3019;ssrc=000e08b4
Server: PVSS/5.0.1_080124 (03:32:53 Jan 24 2008)
```

```
SETUP rtsp://146.152.82.223/public/shuttle_landing.3gp/trackID=101
RTSP/1.0
CSeq: 58
Transport: RTP/AVP;unicast;client_port=49158-49159
Session: 10396e24910aa8b8
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)
```

```
RTSP/1.0 200 OK
CSeq: 58
Date: Mon, 11 Aug 2008 17:00:14 GMT
BufferSize: 7000
Session: 10396e24910aa8b8;timeout=60
Supported: method.eos, com.pv.server_playlist, com.pvns.proxy-exchange
Transport: RTP/AVP;unicast;client_port=49158-49159;server_port=3018-3019;ssrc=010e08b4
Server: PVSS/5.0.1_080124 (03:32:53 Jan 24 2008)
```

```
PLAY rtsp://146.152.82.223/public/shuttle_landing.3gp/ RTSP/1.0
CSeq: 59
Session: 10396e24910aa8b8
Range: npt=0.000-150.900
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)
```

```
RTSP/1.0 200 OK
CSeq: 59
Date: Mon, 11 Aug 2008 17:00:14 GMT
Session: 10396e24910aa8b8
Supported: method.eos, com.pv.server_playlist, com.pvns.proxy-exchange
Range: npt=0-150.900
RTP-Info:
url=rtsp://146.152.82.223/public/shuttle_landing.3gp/trackID=201;seq=0;rtptime=0,
url=rtsp://146.152.82.223/public/shuttle_landing.3gp/trackID=101;seq=0;rtptime=0
RTCP-Interval: 1000
```

```
TEARDOWN rtsp://146.152.82.223/public/shuttle_landing.3gp/ RTSP/1.0
CSeq: 60
Session: 10396e24910aa8b8
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 60
Date: Mon, 11 Aug 2008 17:00:35 GMT
Session: 10396e24910aa8b8
Supported: method.eos, com.pv.server_playlist, com.pvns.proxy-exchange
```

## Using the Helix Server

```
OPTIONS rtsp://146.152.82.228/shuttle_landing.3gp RTSP/1.0
CSeq: 1
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 1
Date: Tue, 05 Aug 2008 18:09:30 GMT
Server: Helix Server Version 12.0.0.1095 (linux-rhel4-i686) (RealServer compatible)
Public: OPTIONS, DESCRIBE, PLAY, PAUSE, SETUP, GET_PARAMETER, SET_PARAMETER,
TEARDOWN
TurboPlay: 1
RealChallenge1: 125226d135c1c1512e1d0528a53a172c
StatsMask: 8

DESCRIBE rtsp://146.152.82.228/shuttle_landing.3gp RTSP/1.0
CSeq: 2
Accept: application/sdp
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 2
Date: Tue, 05 Aug 2008 18:09:30 GMT
Set-Cookie: cbid=ikcgchlmjkhkildmeokoqppqerjrktlufkejkielfjdkilplqnmrkultrresntmpdfdgi
hhl;path=/;expires=Thu,31-Dec-2037 23:59:59 GMT
vsrc: http://146.152.82.228:8080/viewsource/template.
html?nuyhtg2twtz60p8hzgyoomxdj4C6me8Ckcs6y5yq20nnn2n7s7A158uel657
B9m5tgf8zcse6e8s5gnix5b71Bi1E68ra5E6
Last-Modified: Mon, 04 Aug 2008 13:11:39 GMT
Content-base: rtsp://146.152.82.228/shuttle_landing.3gp/
Vary: User-Agent, ClientID
Content-type: application/sdp
x-real-usestrackid: 1
Content-length: 1387
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
v=0
o=- 1217855499 1217855499 IN IP4 146.152.82.228
s=shuttle_landing.3gp
i=<No author> <No copyright>
c=IN IP4 0.0.0.0
t=0 0
a=SdpplinVersion:1610641560
a=StreamCount:integer;2
a=control:*
a=DefaultLicenseValue:integer;0
a=FileType:string;"MPEG4"
a=LicenseKey:string;"license.Summary.Datatypes.RealMPEG4.Enabled"
a=range:npt=0-151
m=video 0 RTP/AVP 96
b=AS:35
b=TIAS:29193
b=RR:1094
b=RS:364
a=maxprate:10.108380
a=control:streamid=65335
a=range:npt=0-151
a=length:npt=151
a=rtpmap:96 MP4V-ES/90000
a=fmtp:96 profile-level-id=8; config=000001B008000001B50EA020202F000001000000012000C788BA985
0584121463F
a=mimetype:string;"video/MP4V-ES"
a=3GPP-Adaptation-Support:1
a=Helix-Adaptation-Support:1
a=AvgBitRate:integer;29193
a=AvgPacketSize:integer;361
a=Width:integer;176
a=Height:integer;144
a=HasOutOfOrderTS:integer;1
a=Preroll:integer;1000
a=cliprect:0,0,144,176
a=mpeg4-esid:201
a=x-envivio-verid:00035A32
m=audio 0 RTP/AVP 97
b=AS:15
b=TIAS:12828
b=RR:481
b=RS:160
a=maxprate:5.010937
a=control:streamid=65435
a=range:npt=0-151
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
a=length:npt=151
a=rtpmap:97 AMR/8000
a=fmtp:97 octet-align=1
a=mimetype:string;"audio/AMR"
a=3GPP-Adaptation-Support:1
a=Helix-Adaptation-Support:1
a=AvgBitRate:integer;12828
a=AvgPacketSize:integer;320
a=HasOutOfOrderTS:integer;1
a=Preroll:integer;1000
a=mpeg4-esid:101
a=x-envivio-verid:00035A32
SETUP rtsp://146.152.82.228/shuttle_landing.3gp/streamid=65335 RTSP/1.0
CSeq: 3
Transport: RTP/AVP;unicast;client_port=57350-57351
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 3
Date: Tue, 05 Aug 2008 18:09:30 GMT
Session: 853127572-3;timeout=79
Transport: RTP/AVP;unicast;client_port=57350-57351;server_port=14076-14077
Reconnect: true
RDTFeatureLevel: 0
StatsMask: 8

SETUP rtsp://146.152.82.228/shuttle_landing.3gp/streamid=65435 RTSP/1.0
CSeq: 4
Transport: RTP/AVP;unicast;client_port=49158-49159
Session: 853127572-3
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
CSeq: 4
Date: Tue, 05 Aug 2008 18:09:30 GMT
Session: 853127572-3;timeout=79
Transport: RTP/AVP;unicast;client_port=49158-49159;server_port=31878-31879
RDTFeatureLevel: 0
StatsMask: 8

PLAY rtsp://146.152.82.228/shuttle_landing.3gp RTSP/1.0
CSeq: 5
Session: 853127572-3
Range: npt=0.000-151.000
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
RTSP/1.0 200 OK
CSeq: 5
Date: Tue, 05 Aug 2008 18:09:30 GMT
Session: 853127572-3;timeout=79
RTP-Info: url=rtsp://146.152.82.228/shuttle_landing.3gp/streamid=65335;seq=0;rtptime=0,
url=rtsp://146.152.82.228/shuttle_landing.3gp/streamid=65435;seq=0;rtptime=0
Range: npt=0-151
```

## Using the Darwin Server

```
OPTIONS rtsp://146.152.82.225/shuttle_landing.3gp RTSP/1.0
CSeq: 7
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
Server: DSS/5.5.5 (Build/489.16; Platform/Linux; Release/Darwin; state/beta; )
Cseq: 7
Public: DESCRIBE, SETUP, TEARDOWN, PLAY, PAUSE, OPTIONS, ANNOUNCE, RECORD

DESCRIBE rtsp://146.152.82.225/shuttle_landing.3gp RTSP/1.0
CSeq: 8
Accept: application/sdp
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
Server: DSS/5.5.5 (Build/489.16; Platform/Linux; Release/Darwin; state/beta; )
Cseq: 8
Last-Modified: Mon, 11 Aug 2008 15:55:25 GMT
Cache-Control: must-revalidate
Content-length: 1133
Date: Wed, 20 Aug 2008 14:59:48 GMT
Expires: Wed, 20 Aug 2008 14:59:48 GMT
Content-Type: application/sdp
x-Accept-Retransmit: our-retransmit
x-Accept-Dynamic-Rate: 1
Content-Base: rtsp://146.152.82.225/shuttle_landing.3gp/

v=0
o=StreamingServer 3428233187 1218470125000 IN IP4 146.152.82.225
s=/shuttle_landing.3gp
u=http:///
e=admin@
c=IN IP4 0.0.0.0
t=0 0
t=0 0
```



# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

```
SETUP rtsp://146.152.82.225/shuttle_landing.3gp/trackID=65435 RTSP/1.0
CSeq: 10
Transport: RTP/AVP;unicast;client_port=49154-49155
Session: 786924391097519769
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
Server: DSS/5.5.5 (Build/489.16; Platform/Linux; Release/Darwin; state/beta; )
Cseq: 10
Session: 786924391097519769
Last-Modified: Mon, 11 Aug 2008 15:55:25 GMT
Cache-Control: must-revalidate
Date: Wed, 20 Aug 2008 14:59:48 GMT
Expires: Wed, 20 Aug 2008 14:59:48 GMT
Transport: RTP/AVP;unicast;source=146.152.82.225;client_port=49154-49155;server_port=6970-6971;ssrc=136E7A2C
PLAY rtsp://146.152.82.225/shuttle_landing.3gp/ RTSP/1.0
CSeq: 11
Session: 786924391097519769
Range: npt=0.000-150.900
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
Server: DSS/5.5.5 (Build/489.16; Platform/Linux; Release/Darwin; state/beta; )
Cseq: 11
Session: 786924391097519769
Range: npt=0.00000-150.90000
RTP-Info: url=rtsp://146.152.82.225/shuttle_landing.3gp/trackID=65335;seq=11566;rtptime=1772451650,url=rtsp://146.152.82.225/shuttle_landing.3gp/trackID=65435;seq=61746;rtptime=1813260943

TEARDOWN rtsp://146.152.82.225/shuttle_landing.3gp/ RTSP/1.0
CSeq: 12
Session: 786924391097519769
User-Agent: MultiMediaDemo (LIVE555 Streaming Media v2008.06.25)

RTSP/1.0 200 OK
Server: DSS/5.5.5 (Build/489.16; Platform/Linux; Release/Darwin; state/beta; )
Cseq: 12
Session: 786924391097519769
Connection: Close
```

# RTSP Extensions to Dialogic® 3G-324M Multimedia Gateway Demo and Dialogic® Multimedia Demo

Application Note

## For More Information

### Reference Application Note

Developing Media Solutions using RTSP —  
<http://www.dialogic.com/goto/?10580>

### Dialogic Documentation

Dialogic® 3G-324M Multimedia Gateway Demo Guide —  
[http://www.dialogic.com/manuals/docs/3g324m\\_demo\\_v1.pdf](http://www.dialogic.com/manuals/docs/3g324m_demo_v1.pdf)

Dialogic® Multimedia Demo Guide —  
[http://www.dialogic.com/manuals/docs/multimedia\\_demo\\_hmp\\_v4.pdf](http://www.dialogic.com/manuals/docs/multimedia_demo_hmp_v4.pdf)

Dialogic® Multimedia API Programming Guide —  
[http://www.dialogic.com/manuals/docs/multimedia\\_programming\\_v5.pdf](http://www.dialogic.com/manuals/docs/multimedia_programming_v5.pdf)

### Utilities

MP4Box: [http://gpac.sourceforge.net/doc\\_mp4box.php](http://gpac.sourceforge.net/doc_mp4box.php)

RTSP Servers:

Darwin: <http://dss.macosforge.org/>

PVNS Streaming Server: <http://www.pvnetsolutions.com/downloads/register-pvserver.html>

Helix: [http://www.realnetworks.com/products/media\\_delivery.html](http://www.realnetworks.com/products/media_delivery.html)

### Specifications and Tutorials

Real Time Streaming Protocol (from Wikipedia) —  
[http://en.wikipedia.org/wiki/Real\\_Time\\_Streaming\\_Protocol](http://en.wikipedia.org/wiki/Real_Time_Streaming_Protocol)

Henning Schulzrinne, “Internet Media-on-Demand: The Real-Time Streaming Protocol” —  
<http://www.cs.columbia.edu/~hgs/teaching/ais/slides/2003/RTSP.pdf>

[www.dialogic.com](http://www.dialogic.com)

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