SONY

Technical Guide | G6

# RTSP Streaming

Network Camera
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IPELA

# CONTENTS

# 1. Overview

## 1.1. About This Document

This document describes technical details about RTSP streaming function supported by the 6th generation (the G6) of Sony Network Cameras (hereafter referred to as “the cameras”).

## 1.2. About CGI Version

If you would like to identify whether your camera belongs to G6 or not, please kindly use the CGI command below.

```
http://camera_address/command/inquiry.cgi?inq=system
```

In case of G6 models, the following description is in the response.

```
CGIVersion=6.*. *.*.*                    (Where each * is a number.)
```

Otherwise, this parameter does not exist.

# 2. Supported Codecs

The following codecs are supported with RTSP streaming function of the cameras.

| Video Codec |
| --- |
| H.264 |
| Motion JPEG |

| Audio Codec |
| --- |
| G.711 |
| G.726 |
| AAC |

* Some cameras may not support some of the codecs with RTSP streaming function depending on capabilities of each model/series.

For details about the codecs listed above, please refer to relevant documents such as “User’s Guide” or “CGI Commands Technical Guide”.

# 3. RTSP Request URL

RTSP request URLs of the cameras for getting live stream are as follows.

| Request URL | Description |
|---|---|
| `rtsp://<camera_address>/video1` | Requests video* bit stream corresponding to the "`ImageCodec1`"** and its related CGI parameters.<br>* Audio bit stream can be transmitted together with this video stream depending on situations. See chapter 5 "Getting Stream".<br>** `ImageCodec1` corresponds to the "Image1" settings in administrator menu of the cameras. |
| `rtsp://<camera_address>/video2` | Requests video* bit stream corresponding to the "`ImageCodec2`"** and its related CGI parameters.<br>* Audio bit stream can be transmitted together with this video stream depending on situations. See chapter 5 "Getting Stream".<br>** `ImageCodec2` corresponds to the "Image2" settings in administrator menu of the cameras. |
| `rtsp://<camera_address>/video3` | Requests video* bit stream corresponding to the "`ImageCodec3`"** and its related CGI parameters.<br>* Audio bit stream can be transmitted together with this video stream depending on situations. See chapter 5 "Getting Stream".<br>** `ImageCodec3` corresponds to the "Image3" settings in administrator menu of the cameras. |
| `rtsp://<camera_address>/audio` | Requests audio bit stream corresponding to the "`AudInCodec`"* and its related CGI parameters.<br>* `AudInCodec` corresponds to the "Audio sending" settings in administrator menu of the cameras. |

* Total number of codec instances in the camera may vary depending on capabilities of each model/series.

RTSP port of the camera (RTSP server) is 554 by factory default. If needed, the port can be changed by using "`camera.cgi`" CGI command with "`RTSPPort`" CGI parameter.

For details about the each live stream listed above (such as "what the CGI parameter `ImageCodec1` is", "how to configure the Image1 settings" and so on), please refer to the relevant documents — "CGI Commands Technical Guide" and "User's Guide".

Regarding request URLs for SSM (source-specific multicast), please refer to chapter 7 "Source-Specific Multicast".

# 4. RTSP Methods

## 4.1. Supported Methods

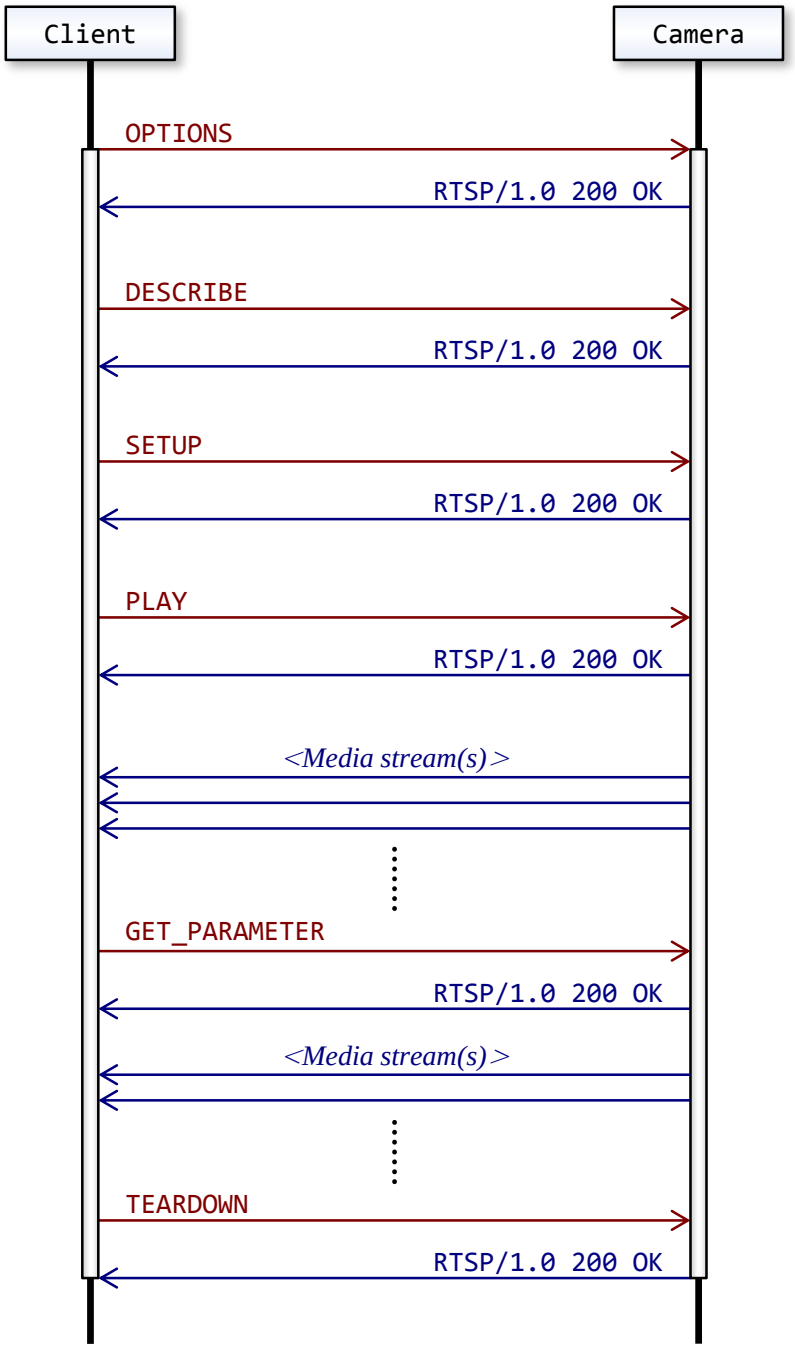
The cameras support the following RTSP methods.

| Supported Method |
| --- |
| OPTIONS |
| DESCRIBE |
| SETUP |
| PLAY |
| PAUSE *(To be supported)* |
| TEARDOWN |
| GET_PARAMETER |
| SET_PARAMETER |

For details about the RTSP methods listed above, please refer to RFC 2326.

## 4.2. Typical Sequence of RTSP Communication

Overview of a typical RTSP communication sequence between the camera (RTSP server) and a client is as shown below.




RTSP "GET_PARAMETER" method in the sequence above is used to keep the RTSP streaming alive. Please refer to chapter 5 "Getting Stream" for further descriptions.

# 5. Getting Stream

### < Transfer Protocols >

RTSP function of the cameras supports the following transfer protocols to stream video and/or audio from the camera to client(s).

**a)** TCP bit stream
**b)** UDP unicast bit stream
**c)** UDP multicast bit stream

Details of each case above are described in the following sections.

### < Number of Media Streams >

The cameras support multiple codec instances simultaneously as mentioned in the previous chapter 3 "RTSP Request URL". Number of media streams in an RTSP session of the camera varies depending on situations as described below.

**Case 1)** In a situation where audio codec is disabled and a video stream is requested:
Each RTSP session delivers just one video stream instances at a time.
(Examples of this case are deeply described in section 5.1 "Getting Video Stream" below.)

**Case 2)** In a situation where audio codec is enabled and an audio stream is requested:
Each RTSP session delivers just one audio stream at a time.
(This case is described in section 5.2 "Getting Audio Stream" below.)

**Case 3)** In a situation where audio codec is enabled and just a video stream is requested:
Each RTSP session delivers two media streams at a time — not only a video stream but also an audio stream.
(Example of this case is deeply described in section 5.3 "Getting Both Video and Audio Bit Streams" below.)

### < RTSP Session Timeout >

RTSP session timeout value of the cameras – "`RTSPTimeout`" CGI parameter in seconds – is set to zero by factory default. The cameras treat the value zero as infinite duration and RTSP session never timeout. To set `RTSPTimeout` to an arbitrary duration, "`camera.cgi`" CGI command with `RTSPTimeout` CGI parameter (in seconds) can be used. Please refer to relevant document "CGI

Commands Technical Guide" for details about `camera.cgi` and `RTSPTimeout`.

RTSP session timeout value of the camera is indicated to a client as "`timeout`" parameter (in seconds) in RTSP response to "`SETUP`" request. However, when the timeout value is set to zero, `timeout` parameter does not exist in the response.

To keep an RTSP streaming alive, examples in this document use RTSP "`GET_PARAMETER`" method before the camera automatically closes RTSP session in accordance with `timeout` pamameter.

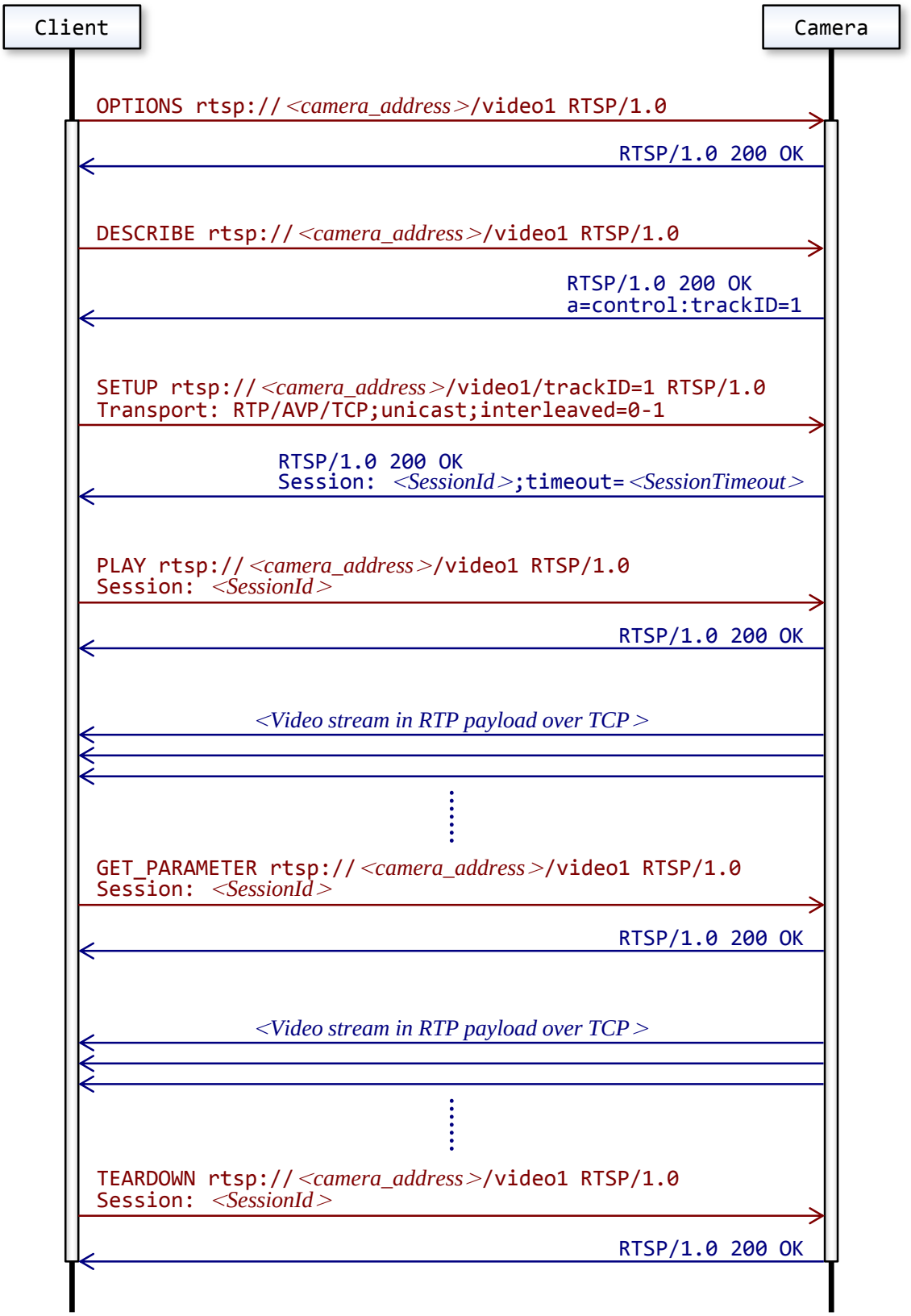
### < Closing RTSP Session >

To safely close an RTSP session and its related UDP ports invoked by it (RTP, RTCP), the cameras require clients to use RTSP "`TEARDOWN`" method.
Technically sending RTCP "`BYE`" packet from a client to the camera might trigger closing all these session/ports in the camera eventually, however, the cameras do not recommend using RTCP "`BYE`" packet in this case.

## 5.1. Getting Video Stream

### 5.1.1. TCP Bit Stream (Video)

The following diagram and captured packets in this section show an example of getting a video bit stream from the camera over TCP in a situation where audio codec is disabled and a client requests a video stream.


Client
Camera
OPTIONS rtsp://<camera_address>/video1 RTSP/1.0
RTSP/1.0 200 OK
DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0
RTSP/1.0 200 OK
a=control:trackID=1
SETUP rtsp://<camera_address>/video1/trackID=1 RTSP/1.0
Transport: RTP/AVP/TCP;unicast;interleaved=0-1
RTSP/1.0 200 OK
Session: <SessionId>;timeout=<SessionTimeout>
PLAY rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
<Video stream in RTP payload over TCP>
GET_PARAMETER rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
<Video stream in RTP payload over TCP>
TEARDOWN rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK

```
OPTIONS rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 1\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 1\r\n
Public: DESCRIBE, SETUP, TEARDOWN, PLAY, OPTIONS, SET_PARAMETER, GET_PARAMETER\r\n
\r\n

DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 2\r\n
User-Agent <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 2\r\n
Content-Length: <Length>\r\n
Content-Type: application/sdp\r\n
Content-Base: rtsp://<camera_address>/video1/\r\n
\r\n

v=<ProtocolVersion>\r\n
o=- <SessionIdForOrigin> 1 IN IP4 <camera_address>\r\n
s=<SessionName>\r\n
t=0 0\r\n
a=range:npt=now-\r\n
c=IN IP4 <ConnectionAddress>\r\n
m=<MediaNameAndTransportAddress>\r\n
a=rtpmap:<PayloadType> <EncodingName>/<ClockRate>\r\n
a=control:trackID=1\r\n
a=framerate:<FrameRate>\r\n
a=fmtp:<Format> <FormatSpecificParameters>\r\n
\r\n

SETUP rtsp://<camera_address>/video1/trackID=1 RTSP/1.0\r\n
CSeq: 3\r\n
Transport: RTP/AVP/TCP;unicast;interleaved=0-1\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 3\r\n
Session: <SessionId>[;timeout=<SessionTimeout>]\r\n
Cache-Control: must-revalidate\r\n
Transport: RTP/AVP/TCP;interleaved=0-1;ssrc=<SSRC>\r\n
```

```
\r\n

PLAY rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 4\r\n
Session: <SessionId>\r\n
Range: npt=0.000-\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 4\r\n
Session: <SessionId>\r\n
RTP-Info: url=trackID=1;seq=<SequenceNumber>;rtptime=0\r\n
\r\n

<Video stream in RTP payload over TCP>

GET_PARAMETER rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 5\r\n
Session: <SessionId>\r\n
User-Agent: <UA>\r\n
\r\n

<Video stream in RTP payload over TCP>

TEARDOWN rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 6\r\n
Session: <SessionId>\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 6\r\n
Session: <SessionId>\r\n
```

### 5.1.2. UDP Unicast Bit Stream (Video)

The following diagram and captured packets in this section show an example of getting video bit stream from the camera over UDP unicast in a situation where audio codec is disabled and a client requests a video stream.

In the example below in this section, client uses "`client_port`" parameter in RTSP "`SETUP`" request to specify UDP unicast ports of the client's end for RTP session and RTCP session. One of the following ways can be used to specify the ports.

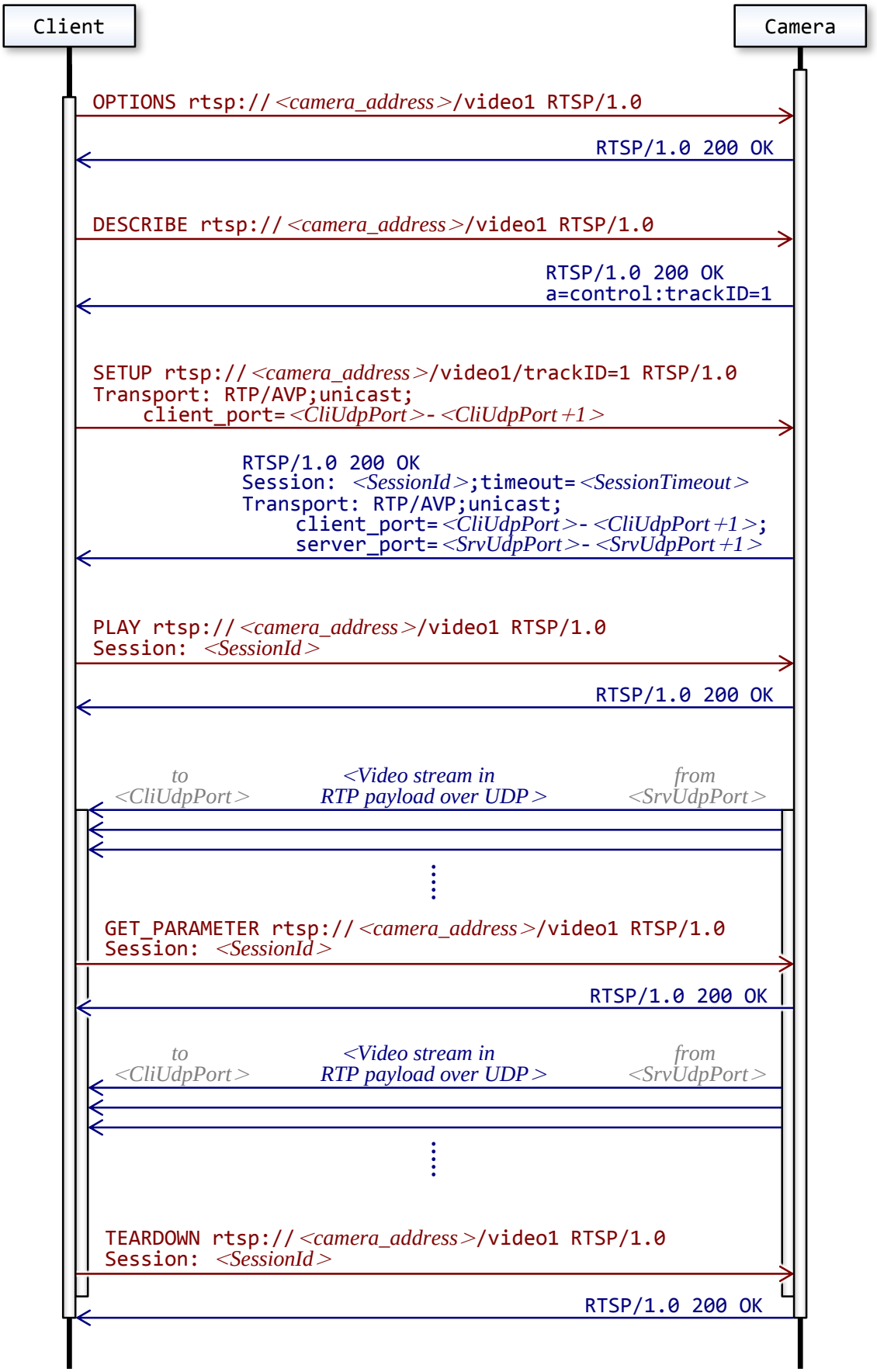
**a)** Specifying ports by using "`client_port`" parameter in RTSP "`SETUP`" request — This is used in the example mentioned above.

**b)** Specifying ports by using "`camera.cgi`" CGI command with CGI parameters listed below.

| CGI Parameter | Corresponding to |
|---|---|
| `RTSPUcVideoPort1` | A pair of UDP unicast ports for the "`ImageCodec1`" live stream |
| `RTSPUcVideoPort2` | A pair of UDP unicast ports for the "`ImageCodec2`" live stream |
| `RTSPUcVideoPort3` | A pair of UDP unicast ports for the "`ImageCodec3`" live stream |
| `RTSPUcAudioPort` | A pair of UDP unicast ports for the "`AudInCodec`" live stream |

For details about them, please refer to relevant document — "CGI Commands Technical Guide".

**c)** Not explicitly specifying UDP unicast ports — In this case, the ports are automatically specified by the camera (RTSP server) and are indicated in RTSP response to "`SETUP`" request.


Client
Camera
OPTIONS rtsp://<camera_address>/video1 RTSP/1.0
RTSP/1.0 200 OK
DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0
RTSP/1.0 200 OK
a=control:trackID=1
SETUP rtsp://<camera_address>/video1/trackID=1 RTSP/1.0
Transport: RTP/AVP;unicast;
client_port=<CliUdpPort>-<CliUdpPort+1>
RTSP/1.0 200 OK
Session: <SessionId>;timeout=<SessionTimeout>
Transport: RTP/AVP;unicast;
client_port=<CliUdpPort>-<CliUdpPort+1>;
server_port=<SrvUdpPort>-<SrvUdpPort+1>
PLAY rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
to <CliUdpPort>
<Video stream in RTP payload over UDP>
from <SrvUdpPort>
GET_PARAMETER rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
to <CliUdpPort>
<Video stream in RTP payload over UDP>
from <SrvUdpPort>
TEARDOWN rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK

```
OPTIONS rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 1\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 1\r\n
Public: DESCRIBE, SETUP, TEARDOWN, PLAY, OPTIONS, SET_PARAMETER, GET_PARAMETER\r\n
\r\n

DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 2\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 2\r\n
Content-Length: <Length>\r\n
Content-Type: application/sdp\r\n
Content-Base: rtsp://<Address>/video1/\r\n
\r\n

v=0\r\n
o=- <SessionIdForOrigin> 1 IN IP4 <camera_address>\r\n
s=<SessionName>\r\n
t=0 0\r\n
a=range:npt=now-\r\n
c=IN IP4 <ConnectionAddress>\r\n
m=<MediaNameAndTransportAddress>\r\n
a=rtpmap:<PayloadType> <EncodingName>/<ClockRate>\r\n
a=control:trackID=1\r\n
a=framerate:<FrameRate>\r\n
a=fmtp:<Format> <FormatSpecificParameters>\r\n
\r\n

SETUP rtsp://<camera_address>/video1/trackID=1 RTSP/1.0\r\n
CSeq: 3\r\n
Transport: RTP/AVP;unicast;client_port=<CliUdpPort>-<CliUdpPort+1>\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 3\r\n
Session: <SessionId>[;timeout=<SessionTimeout>]\r\n
Cache-Control: must-revalidate\r\n
Transport: RTP/AVP;unicast;client_port=<CliUdpPort>-<CliUdpPort+1>;source=<Sou
```

*rceAddress*>;server_port=<*SrvUdpPort*>-<*SrvUdpPort+1*>;ssrc=<*SSRC*>\r\n
\r\n

PLAY rtsp://<*camera_address*>/video1 RTSP/1.0\r\n
CSeq: 4\r\n
Session: <*SessionId*>\r\n
Range: npt=0.000-\r\n
User-Agent: <*UA*>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <*ServerName*>\r\n
CSeq: 4\r\n
Session: <*SessionId*>\r\n
RTP-Info: url=trackID=1;seq=<*SequenceNumber*>;rtptime=0\r\n
\r\n

<*Video stream in RTP payload over UDP unicast*>

GET_PARAMETER rtsp://<*camera_address*>/video1 RTSP/1.0\r\n
CSeq: 5\r\n
Session: <*SessionId*>\r\n
User-Agent: <*UA*>\r\n
\r\n

<*Video stream in RTP payload over UDP unicast*>

TEARDOWN rtsp://<*camera_address*>/video1 RTSP/1.0\r\n
CSeq: 6\r\n
Session: <*SessionId*>\r\n
User-Agent: <*UA*>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <*ServerName*>\r\n
CSeq: 6\r\n
Session: <*SessionId*>\r\n

### 5.1.3. UDP Multicast Bit Stream (Video)

The following diagram and captured packets in this section show an example of getting video bit stream from the camera over UDP multicast in a situation where audio codec is disabled and a client requests a video stream.

In the example below in this section, the client uses "`destination`" parameter and "`client_port`" parameter in RTSP "`SETUP`" request to specify multicast address and ports for RTP session and RTCP session. One of the following ways can be used to specify the multicast address and/or the ports.

**a)** Specifying address and/or ports by using "`destination`" parameter and/or "`client_port`" parameter in RTSP "`SETUP`" request — This way is used in the example mentioned below.

**b)** Specifying address and/or ports by using "`camera.cgi`" CGI command with CGI parameters listed below.

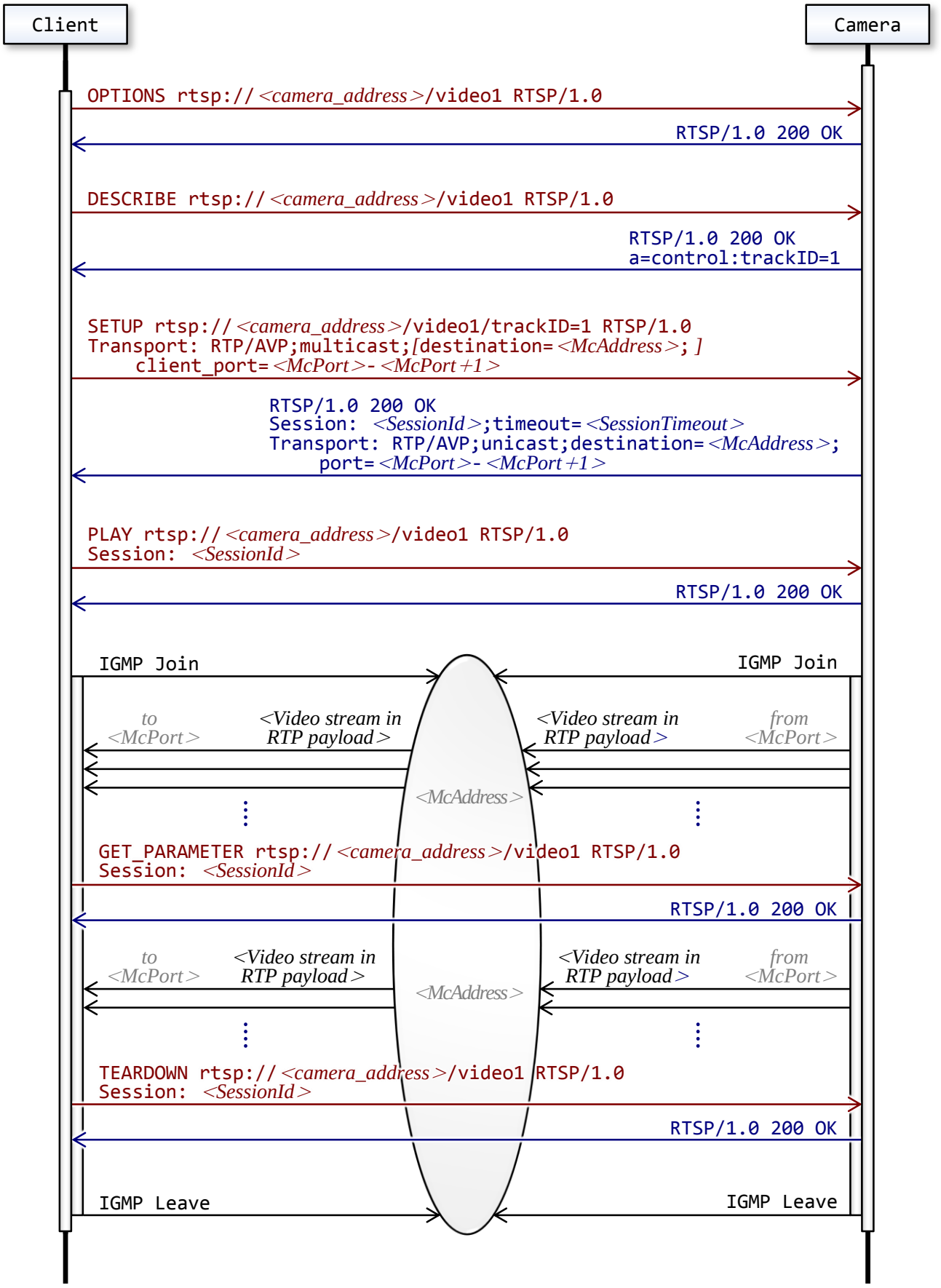
| CGI Parameter | Corresponding to |
|---|---|
| `RTSPMcAddress` | IPv4 multicast address |
| `RTSPMcVideoPort1` | A pair of multicast ports for the "`ImageCodec1`" live stream |
| `RTSPMcVideoPort2` | A pair of multicast ports for the "`ImageCodec2`" live stream |
| `RTSPMcVideoPort3` | A pair of multicast ports for the "`ImageCodec3`" live stream |
| `RTSPMcAudioPort` | A pair of multicast ports for the "`AudInCodec`" live stream |

For details about them, please refer to relevant document — "CGI Commands Technical Guide".

**c)** Not explicitly specifying multicast address and/or ports — In this case, multicast address and/or ports are automatically specified by the camera (RTSP server) and are indicated in RTSP response to "`SETUP`" request.

Regarding multicast "time to live" (TTL) value, the cameras indicate it to clients as "`ttl`" parameter in RTSP response to "`SETUP`" request.
Clients can set an arbitrary TTL value in RTSP "`SETUP`" request if needed. Or, when clients do not explicitly specify TTL value in "`SETUP`" request, the cameras automatically use "`McTtl`" CGI parameter as TTL value instead. `McTtl` can be changed by using "`camera.cgi`" CGI command if needed. Please refer to relevant document "CGI Commands Technical Guide" for details about `camera.cgi` and `McTtl`.

Regarding SSM (source-specific multicast), please refer to chapter 7 "Source-Specific Multicast".


Client
Camera
OPTIONS rtsp://<camera_address>/video1 RTSP/1.0
RTSP/1.0 200 OK
DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0
RTSP/1.0 200 OK
a=control:trackID=1
SETUP rtsp://<camera_address>/video1/trackID=1 RTSP/1.0
Transport: RTP/AVP;multicast;[destination=<McAddress>; ]
client_port=<McPort>-<McPort+1>
RTSP/1.0 200 OK
Session: <SessionId>;timeout=<SessionTimeout>
Transport: RTP/AVP;unicast;destination=<McAddress>;
port=<McPort>-<McPort+1>
PLAY rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
IGMP Join
IGMP Join
to <McPort>
<Video stream in RTP payload>
<Video stream in RTP payload>
from <McPort>
<McAddress>
GET_PARAMETER rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
to <McPort>
<Video stream in RTP payload>
<Video stream in RTP payload>
from <McPort>
<McAddress>
TEARDOWN rtsp://<camera_address>/video1 RTSP/1.0
Session: <SessionId>
RTSP/1.0 200 OK
IGMP Leave
IGMP Leave

```
OPTIONS rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 1\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 1\r\n
Public: DESCRIBE, SETUP, TEARDOWN, PLAY, OPTIONS, SET_PARAMETER, GET_PARAMETER\r\n
\r\n

DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 2\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 2\r\n
Content-Length: <Length>\r\n
Content-Type: application/sdp\r\n
Content-Base: rtsp://<camera_address>/video1/\r\n
\r\n

v=0\r\n
o=- <SessionIdForOrigin> 1 IN IP4 <camera_address>\r\n
s=<SessionName>\r\n
t=0 0\r\n
a=range:npt=now-\r\n
c=IN IP4 <ConnectionAddress>\r\n
m=<MediaNameAndTransportAddress>\r\n
a=rtpmap:<PayloadType> <EncodingName>/<ClockRate>\r\n
a=control:trackID=1\r\n
a=framerate:<FrameRate>\r\n
a=fmtp:<Format> <FormatSpecificParameters>\r\n
\r\n

SETUP rtsp://<camera_address>/video1/trackID=1 RTSP/1.0\r\n
CSeq: 3\r\n
Transport:
RTP/AVP;multicast;destination=<DestinationAddress>;client_port=<McPort>-<McPort
+1>\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 3\r\n
Session: <SessionId>[;timeout=<SessionTimeout>]\r\n
Cache-Control: must-revalidate\r\n
```

```
Transport:
RTP/AVP;multicast;destination=<DestinationAddress>;port=<McPort>-<McPort+1>;ttl
=<MulticastTimeToLive>\r\n
\r\n

PLAY rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 4\r\n
Session: <SessionId>\r\n
Range: npt=0.000-\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 4\r\n
Session: <SessionId>\r\n
RTP-Info: url=trackID=1;seq=<SequenceNumber>;rtptime=0\r\n
\r\n

<Video stream in RTP payload over UDP multicast>

GET_PARAMETER rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 5\r\n
Session: <SessionId>\r\n
User-Agent: <UA>\r\n
\r\n

<Video stream in RTP payload over UDP multicast>

TEARDOWN rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 6\r\n
Session: <SessionId>\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 6\r\n
Session: <SessionId>\r\n
```

**<Note>**

To get video and/or audio bit stream over UDP multicast, the camera requires you to enable multicast streaming function in advance. Otherwise, a request for multicast streaming fails during RTSP communication. For details about ways to enable multicast streaming function of the camera, please refer to relevant documents — “User’s Guide” and “CGI Commands Technical Guide”.

## 5.2. Getting Audio Stream

The following captured packets in this section show an example of getting an audio bit stream from the camera over TCP in a situation where audio codec is enabled and a client requests an audio stream.

Detailed explanations about RTSP audio streaming over UDP unicast and UDP multicast are left out — these can be known by analogy of the cases of video streaming described in section 5.1.2 “UDP Unicast Bit Stream (Video)” and 5.1.3 “UDP Multicast Bit Stream (Video)”.

```
OPTIONS rtsp://<camera_address>/audio RTSP/1.0\r\n
CSeq: 1\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 1\r\n
Public: DESCRIBE, SETUP, TEARDOWN, PLAY, OPTIONS, SET_PARAMETER, GET_PARAMETER\r\n
\r\n

DESCRIBE rtsp://<camera_address>/audio RTSP/1.0\r\n
CSeq: 2\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 2\r\n
Content-Length: <Length>\r\n
Content-Type: application/sdp\r\n
Content-Base: rtsp://<camera_address>/audio/\r\n
\r\n

v=<ProtocolVersion>\r\n
o=- <SessionIdForOrigin> 1 IN IP4 <camera_address>\r\n
s=<SessionName>\r\n
t=0 0\r\n
a=range:npt=now-\r\n
c=IN IP4 <ConnectionAddress>\r\n
m=<MediaNameAndTransportAddress>\r\n
a=rtpmap:<PayloadType> <EncodingName>/<ClockRate>[/<EncodingParameters>]\r\n
a=control:trackID=2\r\n
\r\n

SETUP rtsp://<camera_address>/audio/trackID=2 RTSP/1.0\r\n
CSeq: 3\r\n
```

```
Transport: RTP/AVP/TCP;unicast;interleaved=0-1\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 3\r\n
Session: <SessionId>[;timeout=<SessionTimeout>]\r\n
Cache-Control: must-revalidate\r\n
Transport: RTP/AVP/TCP;interleaved=0-1;ssrc=<SSRC>\r\n
\r\n

PLAY rtsp://<camera_address>/audio RTSP/1.0\r\n
CSeq: 4\r\n
Session: <SessionId>\r\n
Range: npt=0.000-\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 4\r\n
Session: <SessionId>\r\n
RTP-Info: url=trackID=2;seq=<SequenceNumber>;rtptime=0\r\n
\r\n

<Audio stream in RTP payload over TCP>

GET_PARAMETER rtsp://<camera_address>/audio RTSP/1.0\r\n
CSeq: 5\r\n
Session: <SessionId>\r\n
User-Agent: <UA>\r\n
\r\n

<Audio stream in RTP payload over TCP>

TEARDOWN rtsp://<camera_address>/audio RTSP/1.0\r\n
CSeq: 6\r\n
Session: <SessionId>\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 6\r\n
Session: <SessionId>\r\n
```

## 5.3. Getting Both Video and Audio Bit Streams

In a situation where audio codec is enabled and a client requests just a video stream, the camera transmits the video stream and also an audio stream at a time in an RTSP session. The following captured packets in this section show an example of getting both a video bit stream and an audio bit stream at a time from the camera.

As you can see in the example, session descriptions in RTSP response to "DESCRIBE" request contains two media descriptions — the first one is for video, the second one is for audio.

```
DESCRIBE rtsp://<camera_address>/video1 RTSP/1.0\r\n
CSeq: 2\r\n
User-Agent: <UA>\r\n
\r\n

RTSP/1.0 200 OK\r\n
Server: <ServerName>\r\n
CSeq: 2\r\n
Content-Length: <Length>\r\n
Content-Type: application/sdp\r\n
Content-Base: rtsp://<camera_address>/video1/\r\n
\r\n

v=0\r\n
o=- <SessionIdForOrigin> 1 IN IP4 <camera_address>\r\n
s=<SessionName>\r\n
t=0 0\r\n
a=range:npt=now-\r\n
c=IN IP4 <ConnectionAddress>\r\n

m=video 0 RTP/AVP 105\r\n
a=rtpmap:105 H264/90000\r\n
a=control:trackID=1\r\n
a=framerate:60.0\r\n
a=fmtp:105 packetization-mode=1; profile-level-id=2742e0; sprop-parameter-sets
=Z0LgH41oBQBbsBbIAAAfSAADqYNaAD0IAQBXvdQ8UIqA,KM4ESSAAAA==\r\n

m=audio 0 RTP/AVP 101\r\n
a=rtpmap:101 mpeg4-generic/16000/1\r\n
a=control:trackID=2\r\n
a=fmtp:101 profile-level-id=15; streamtype=5; mode=AAC-hbr; config=1408; SizeL
ength=13; IndexLength=3;IndexDeltaLength=3; constantDuration=1024; Profile=1;
bitrate=64000; \r\n
\r\n
```

## 5.4. rtpmap Attribute

Values of "`rtpmap`" attribute(s) in RTSP response to "`DESCRIBE`" request vary with codec of media stream(s). Here are examples with the cameras.

| Codec | rtpmap Attribute Value |
|---|---|
| H.264 | `a=rtpmap:105 H264/90000\r\n` |
| JPEG | `a=rtpmap:26 JPEG/90000\r\n` |
| G.711 | `a=rtpmap:0 PCMU/8000\r\n` |
| G.726 (40 kbps) | `a=rtpmap:97 G726-40/8000\r\n` |
| G.726 (32 kbps) | `a=rtpmap:98 G726-32/8000\r\n` |
| G.726 (24 kbps) | `a=rtpmap:99 G726-24/8000\r\n` |
| G.726 (16 kbps) | `a=rtpmap:100 G726-16/8000\r\n` |
| AAC (64 kbps) | `a=rtpmap:101 mpeg4-generic/16000/1\r\n` |
| AAC (128 kbps) | `a=rtpmap:102 mpeg4-generic/48000/1\r\n` |

# 6. Bit-Packing Formats of G.726 Audio

The cameras support the following two bit-packing formats for G.726 audio streaming.

| Format | Description |
|---|---|
| RFC 3551 compliant | G.726 audio bit stream in this bit-packing format from the camera is compatible with ONVIF conformant clients.<br>Non-ONVIF clients (such as generic media players) may not be able to correctly recognize G.726 audio bit stream in this format. |
| ITU-T compliant | G.726 audio bit stream in this bit-packing format from the camera is compatible with ITU-T recommendation.<br>It is expected that many of generic media players can recognize G.726 audio bit stream in this format. |

The cameras automatically choose one of the two formats listed above according to string in "`User-Agent:`" field in RTSP request.

When the user agent string in RTSP request matches with "`RTSPReUserAgentG726Itu`" CGI parameter in regular expression, the cameras transmit G.726 audio bit stream in ITU-T compliant bit-packing format. Otherwise, the cameras transmit RFC 3551 compliant bit-packing format. "`RTSPReUserAgentG726Itu`" CGI parameter is "`LibVLC.*`" by factory default. This CGI parameter can be changed by using "`camera.cgi`" CGI command. Please refer to relevant document "CGI Commands Technical Guide" for details about `camera.cgi` and `RTSPReUserAgentG726Itu`.

# 7. Source-Specific Multicast

To be supported.

# 8. RTSP/RTP Tunneling over HTTP

To be supported.

# Revision History

| Date | Revision | Description |
|---|---|---|
| 2012/10/26 | 1.0.0 | First edition. |

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