

RTSP Interfaces and Development Guide

1. RTSP Protocol Description

1.1 Live View URL `rtsp://<address>:<port>/Streaming/Channels/<id>`

(?parm1=value1&parm2=value2...).

ipaddress is the IP of the camera, speed dome or other devices, port is the RTSP port.

Example:

`rtsp://192.0.1.100:554/Streaming/Channels/101?transportmode=unicast` /*get the main stream of the 1st channel with unicast*/

`rtsp://192.0.1.100:554/Streaming/Channels/102`/* get the sub stream of the 1st channel with unicast */

`rtsp://192.0.1.100:554/Streaming/Channels/102?transportmode=multicast` /* get the sub stream of the 1st channel with multicast */

Playback URL `rtsp://<address>:<port>/Streaming/tracks/<id>`

(?parm1=value1&parm2=value2...).

ipaddress is the IP of the camera, speed dome or other devices, port is the RTSP port.

Example:

`rtsp://192.0.1.100:554/Streaming/tracks/101?starttime=20120802t063812z&endtime=20120802t064816z` /* Playback the recording files of the 1st channel from start time to end time. The format of should meet the requirement of ISO 8601, which is YYYYMMDD"TT"HHmmSS.fraction"Z", Y stands for year, M stands for month, D stands for day, T is a separator between date and time, H stands for hour, M stands for minute, S stands for second, Z is optional, which means Zulu (GMT).

*/

Old URL Format `rtsp://<ipaddress>/<videotype>/ch<number>/<streamtype>`

/av_stream,

ipaddress is the IP address of device, videotype is mpeg4 or h.264, number is numeric, streamtype is main or sub stream. It is not sensitive to the capital letter or lowercase letter.

For example: `rtsp://192.0.1.100/mpeg4/ch1/main/av_stream` /*the video type is MPEG4 and it comes from main stream of device's channel 1 which IP address is 192.0.1.100*/

1.2 Authentication

(1) Support URL carry with user name and password.

Example:

`rtsp://admin:12345/Streaming/Channels/101`

(2) Support authentication in Describe step.

Example:

```

Real Time Streaming Protocol
Request: DESCRIBE rtsp://172.8.6.48:554/Streaming/channels/1 RTSP/1.0\r\n
CSeq: 4\r\n
Authorization: Basic YWRtaW46MTIzNDU=\r\n
User-Agent: LibVLC/1.1.4 (LIVE555 Streaming Media v2010.08.22)\r\n
Accept: application/sdp\r\n
\r\n

```

1.3 Grammar

Request grammar::

```

COMMAND URI RTSP/1.0<CRLF>
Headerfield1: val1<CRLF>
Headerfield2: val2<CRLF>
...
<CRLF>

```

Feedback grammar:

```

RTSP/1.0 ResultCode ResultString<CRLF>
Headerfield3: val3<CRLF>
Headerfield4: val4<CRLF>
...
<CRLF>

```

1.4 Supported Command Formats

Support the following header-field and omit other header-field

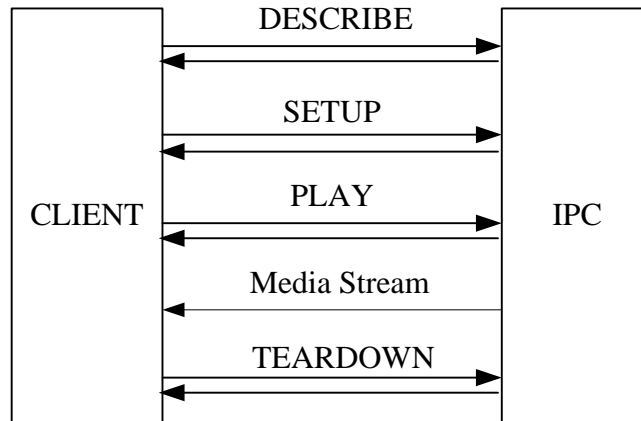
Field	Description
Authorization	the certification information come from client terminal(only support certification information that dragged-in in Describe period
CSeq	Request the serial NO.
Session	Session remark NO. (return by server in SETUP period
Content-Length	Length of content

Head information generated when RTSP server feedback

Field	Description
CSeq	Serial NO feedback(matching with the request serial NO)
Session	Session NO.

1.5 Take all of the RTSP commands that interacted between player and device for example:

C means rtsp client and S means rtsp server



RTSP OPTIONS, C inquire S for the method that can be used and all the methods that can be used are in the reply information of S.

C-S:OPTIONS rtsp://192.0.1.100/Streaming/Channels/101 RTSP/1.0

CSeq: 1// there is label in every message, normally, the first packet is request message from option

User-Agent: Hikplayer

S-C:RTSP/1.0 200 OK

CSeq: 1

Public: DESCRIBE,PLAY,SETUP,TEARDOWN

RTSP DESCRIBE, return back the described information of SDP(RFC2327) appointed by URL. And DESCRIBE command accepts the following header-field:

Accept	The content list type supported by client software(only support application/sdp)
--------	--

DESCRIBE command generate the extra following information of header::

Content-Type	Content type(application/sdp)
Content-Length	The length of SDP that described

C-S:DESCRIBE rtsp://admin:12345@192.0.1.100/Streaming/Channels/101 RTSP/1.0

CSeq: 2

Accept: application/sdp

User-Agent: Hikplayer

S-C:RTSP/1.0 200 OK

CSeq: 2

Content-type: application/sdp

Content-length: 692

v=0//SDP version information

o=- 1109162014219182 1109162014219192 IN IP4 x.y.z.w// owner/creator and session label symbol

s=Media Presentation///session name

e=NONE//Email address

c=IN IP4 0.0.0.0// connection information

t=0 0//active session time

User-Agent: Hikplayer
 S-C:RTSP/1.0 200 OK
 CSeq: 3
 Session:1389957320;timeout=60// the session label symbol that responded from server
 Transport: RTP/AVP;unicast;client_port=1094-1095;server_port=12028-12029

RTSP PLAY, this command is for client to start the data transmit and it requires to generate the following header-field:

Range	Prescribe the play range, for it only support real time stream, so it only has beginning time but not ending time
RTP-Info	information of RTP stream, including next RTP serial NO.

C-S:PLAY rtsp://192.0.1.100/Streaming/Channels/101 RTSP/1.0
 CSeq: 4
 Session: 1389957320
 Range: npt=0.000- // Set the range of playing time. During playback, except the URL, this field should contain the start time and end time parameters as well.
 User-Agent: Hikplayer
 S-C:RTSP/1.0 200 OK
 CSeq: 4
 Session: 1389957320
 Range: npt=now-
 RTP-Info: url=trackID=1;seq=29626 // seq is information that in rtp packet

RTSP TEARDOWN, this command stop the data transmission of the server

C-S:TEARDOWN rtsp://192.0.1.100/Streaming/Channels/101 RTSP/1.0
 CSeq: 5
 Session: 1389957320
 User-Agent: Hikplayer
 S-C:RTSP/1.0 200 OK
 CSeq: 5
 Session: 1389957320

The above methods is usually used in the process of interaction

2. Transmission Protocols

2.1 RTP OVER UDP

- (1) User send a SETUP command with trackID=1 during SETUP phase to get the video stream.
- (2) User send a SETUP command with trackID=2 during SETUP phase to get the audio stream.
- (3) User send two SETUP commands with trackID=1 and rackID=2 during SETUP phase to get the video and audio stream.
- (4) Allocated ports. The default UDP port (for video data) at the server side is calculated with login port. The formula is: the valid UDP ports start from [login port value +200+ channel ID], and ends at [login port value +800+ channel ID]. The default login port is 8000, so the UDP port

for channel 1 is from 8201 to 8801.

Once the port value is changed, please reboot the camera to take effective. In the future, it will be editable.

Example of SETUP procedure:

```
C-S:SETUP rtsp://192.0.1.100/Streaming/Channels/101/trackID=1 RTSP/1.0
CSeq: 3
Transport: RTP/AVP;unicast;client_port=13000-13001
User-Agent: Hikplayer
S-C:RTSP/1.0 200 OK
    CSeq: 3
    Session:1389957320
    Transport: RTP/AVP;unicast;client_port=13000-13001;server_port=8200-8201;
ssrc=3FBF041A
```

```
C-S:SETUP rtsp://192.0.1.100/Streaming/Channels/101/trackID=2 RTSP/1.0
    CSeq: 3
    Transport: RTP/AVP;unicast;client_port=13002-13003
    User-Agent: Hikplayer

S-C:RTSP/1.0 200 OK
    CSeq: 3
    Session:1389957320
    Transport: RTP/AVP;unicast;client_port=13002-13003;server_port=8200-8201;
ssrc=3FBF041A
```

2.2 Multicast RTP

(1) The default multicast address and port number are used when there are not multicast address and port in the address.

(2) If there are multicast address and port in the address, they will be used to send data

(3) Allocated ports. The default ports are from 8600-9001. Now, the value is fixed. In the future, it will be editable by the user.

(4) SSRC, it is used for distinguishing the device which sent the multicast data according to the last 4 bits MAC address.

Example of SETUP procedure:

```
C-S:SETUP rtsp://192.0.1.100/Streaming/Channels/101/trackID=1 RTSP/1.0
    CSeq: 3
    Transport: RTP/AVP;mcast
    User-Agent: Hikplayer
S-C:RTSP/1.0 200 OK
    CSeq: 3
    Session:1389957320
```

```
Transport: RTP/AVP;mcast;destination=224.0.1.2;port=8602-8603;ssrc=3FBF041A
C-S:SETUP rtsp://192.0.1.100/Streaming/Channels/101/trackID=2 RTSP/1.0
CSeq: 3
Transport: RTP/AVP;mcast
User-Agent: Hikplayer
S-C:RTSP/1.0 200 OK
CSeq: 3
Session:1389957320
Transport: RTP/AVP;mcast;destination=224.0.1.2;port=8602-8603;ssrc=3FBF041A
```

2.3 RTP OVER RTSP

- (1) RTSP protocol, all RTP data stream is transmit in 554 port
- (2) Binary data embedded into the bit stream
 - The first byte is symbol \$, and it is expressed as 0x24 in hex
 - The second byte is the number of video/audio channel
 - The third and fourth bytes are about the length of RTP packet.

- (3) Example of SETUP procedure::

```
C-S:SETUP rtsp://192.0.1.100/Streaming/Channels/101/trackID=1 RTSP/1.0
CSeq: 3
Transport: RTP/AVP/TCP;unicast
User-Agent: Hikplayer
S-C:RTSP/1.0 200 OK
CSeq: 3
Session:1389957320
Transport: RTP/AVP/TCP;unicast;interleaved=0-1
C-S:SETUP rtsp://192.0.1.100/Streaming/Channels/101/trackID=2 RTSP/1.0
CSeq: 4
Transport: RTP/AVP/TCP;unicast
User-Agent: Hikplayer
S-C:RTSP/1.0 200 OK
CSeq: 4
Session:1389957320
Transport: RTP/AVP/TCP;unicast;interleaved=2-3
```

2.4 RTP OVER RTSP OVER HTTP

In order to pass firewall, RTSP needs to use HTTP channel. In order to do this, two sessions need to be built. One is for GET command, which responds to commands and sends video data; the other is for POST command, which is used for sending commands. RTSP commands are encrypted with base64 before being sent through POST. But GET commands are not encrypted. To bind two sessions together, the device needs a unified ID, which is in the x-sessioncookieField of the http head.

Example:

```
C-S:GET Streaming/Channels/101 HTTP/1.0
```

```
x-sessioncookie: 123456789
S-C: HTTP/1.0 200 OK
Content-Type: application/x-rtsp-tunnelled
C-S: POST Streaming/Channels/101HTTP/1.0
x-sessioncookie: 123456789
Content-Length: 32767
Content-Type: application/x-rtsp-tunnelled
```

This command does not need to be responded. After sending this command, the client could send RTSP commands after base64 encryption.

```
C-S: DESCRIBE rtsp://myserver/ Streaming/Channels/101 RTSP/1.0
CSeq: 14
User-Agent: Axis AMC
Accept: application/sdp
```

2.5 RTP OVER TCP

(1) Specialness when get bit stream:

- During setup phase, user send a setup command with trackID=1 and take over the ports
- User uses the port to connect to the device
- User sends the playing command
- Device sends the video/audio stream to user

(2) Some examples for different periods in the process of SETUP:

```
C-S:SETUP rtsp://192.0.1.100/mpeg4/ch1/main/av_stream/trackID=1 RTSP/1.0
CSeq: 3
Transport: RTP/AVP/TCP;unicast;client_port=13000-13001
User-Agent: Hikplayer
S-C:RTSP/1.0 200 OK
CSeq: 3
Session:1389957320
Transport: RTP/AVP/TCP;unicast;client_port=13000-13001;server_port=8200-8201
```

(3) We use the TPKT protocol to guarantee the transmission and get the whole RTP packet, and distinguish the video from audio via RTP load, the value of video is 96 and audio is 0.

1. The first 4 bytes is TPKT protocol.
2. The first byte is version of TPKT
3. The second byte is pre-reserve field of TPKT protocol
4. The third byte is TPKT header and 4th is length of RTP packet
5. From the 5th byte to 16th bytes are the header of RTP
6. The first two bits of 5th byte are the version No of RTP packet
7. The third bit of 5th byte is the padding bit of RTP packet
8. The 4th bit of 5th byte is the extend bit of RTP packet.
9. The 5th, 6th, 7th, 8th bytes are CSRC of RTP packet
10. The 9th, 10th, 11th, 12th bytes are timestamp of RTP packet

11. The 13th 14th 15th 16th bytes are SSRC of RTP packet(fixed value:0x55667788)



(3) Allocated ports. The TCP port(for data transmission) at the server side depends on the login port. The formula is: login port value +200+(chan-1). The default login port is 8000, so the TCP port for channel 1 is: 8000+200+(1-1)=8200.

Once the port value is changed, please reboot the camera to take the settings effective. Now, the ports are fixed. In the future, it will be editable by the user.

Reference Protocols:

1. RFC2327 SDP(Session Description Protocol) protocol
2. RFC2326 RTSP protocol
3. RFC3551 TPKT protocol
4. RFC3984 RTP Payload Format for H.264 Video
5. RFC3016 RTP Payload Format for MPEG-4 Audio/Visual Streams