

From an IP camera via RTSP

Overview

A video stream is captured from an RTSP source that provides audio and video in the supported codecs. Then, the stream can be played from server in browsers and mobile devices.

RTSP sources

- IP cameras
- Media servers
- Surveillance systems
- Conference servers

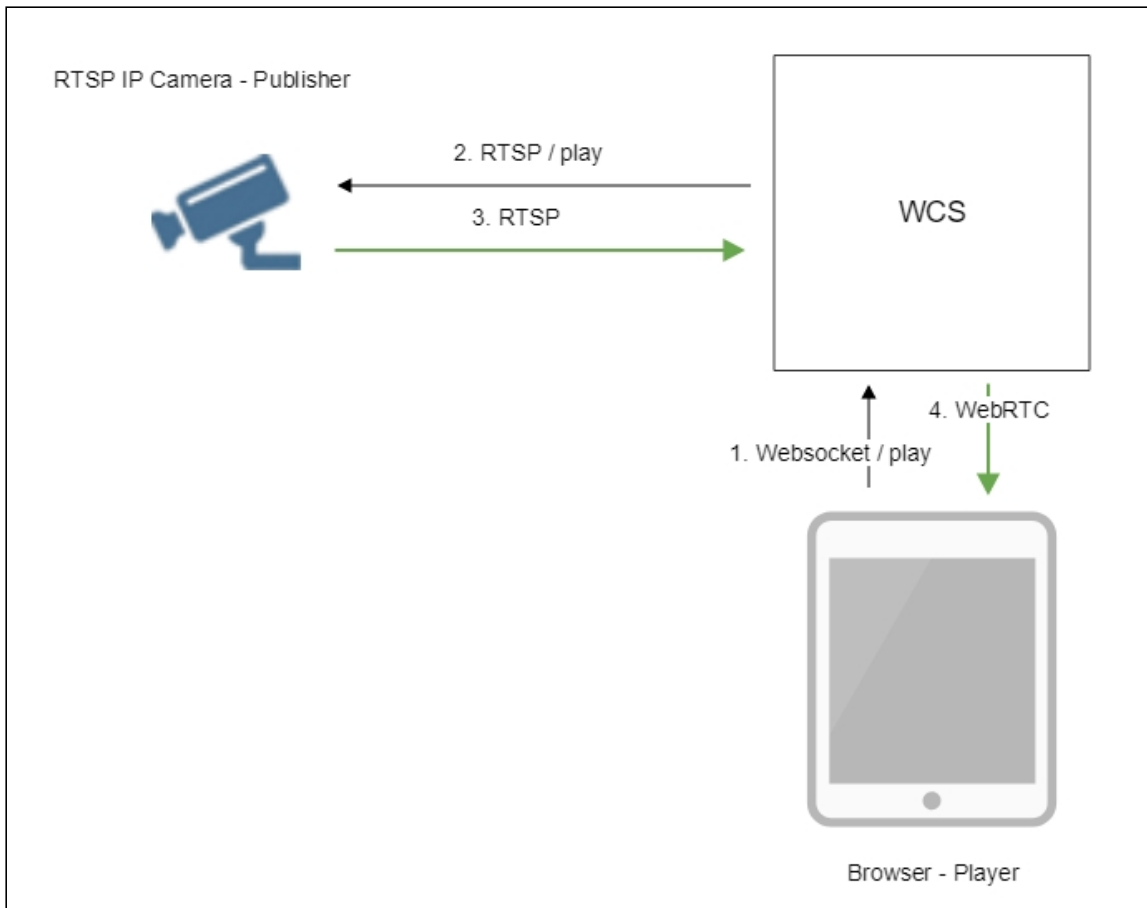
Supported codecs

- Video: H.264, VP8, H265 (since [5.2.1579](#))
- Audio: AAC, G711 (PCMA, PCMU), Speex

Supported platforms and browsers

	Chrome	Firefox	Safari	Edge
Windows	✓	✓	✗	✓
Mac OS	✓	✓	✓	✓
Android	✓	✓	✗	✓
iOS	✓	✓	✓	✓

Operation flowchart



1. The browser establishes a connection to the server via the Websocket protocol and sends the `playStream` command.
2. The server connects to the RTSP source and send the `PLAY` command.
3. The RTSP source sends the RTSP stream to the server.
4. The server send the stream as WebRTC to the browser.

Configuration

RTSP client binding to a certain address

Sometimes, when IP camera should be connected through VPN, RTSP client should be bound to a certain IP address. The option `rtsp_client_address` in `flashphoner.properties` file defines this address, for example:

```
rtsp_client_address=172.16.0.3
```

RTSP stream capturing via UDP

By default, RTSP streams are captured via TCP. Stream capturing over UDP can be turned on if necessary with the following parameter

```
rtsp_interleaved_mode=false
```

Audio and video track selection in RTSP stream

By default, audio and video tracks in RTSP stream are selected dynamically according to camera SDP. Tracks order can be set explicitly if necessary with the following parameter, for example

```
rtsp_interleaved_channels=2-3;0-1
```

Where

- `2-3` - audio track channels
- `0-1` - video track channels

Playing RTSP stream in AnnexB form

Some IP cameras are publishing H264 stream in AnnexB format, Honeywell MAXPRO Video Streamer for example. To play video from such cameras, the following parameter is added since build [5.2.636](#) to enable AnnexB format parsing

```
h264_check_and_skip_annexb=true
```

Attention

Since build [5.2.946](#), this parameter is removed from settings, and AnnexB frames are detected and played automatically.

Audio codecs exclusion

Sometimes it is necessary to capture a stream from camera without audio, or disable some audio codecs to escape sound transcoding. This can be done with the following parameter listing audio codec names, as provided in camera SDP, to be excluded while capturing stream, for example

```
rtsp_client_strip_audio_codecs=PCMA,PCMU
```

This setting will exclude PCMA (alaw) and PCMU (ulaw) codecs. A stream with audio track in those codecs will be captured from a camera as video only.

The listed codecs are excluded at SDP level, by names.

H264 packetization mode configuration

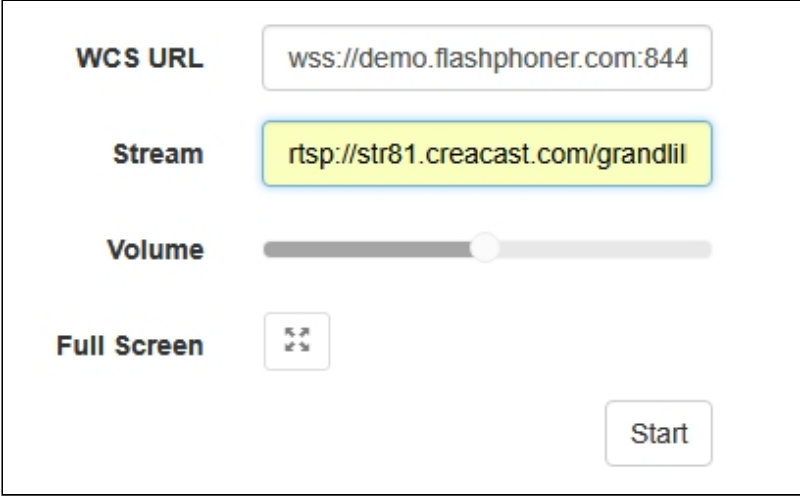
According to H264 specifications, if H264 packetization mode is not set explicitly in stream SDP, it should be set to 0. However, some RTSP cameras may send a stream encoded with packetization mode 1, and may omit this mode setting in SDP. This behaviour may lead to stream transcoding and picture quality dropping while playing the stream in Safari browser.

Since build [5.2.820](#) it is possible to set default packetization mode to use such cameras with the following parameter

```
default_packetization_mode=1
```

Quick manual on testing

1. For this test we use:
2. the demo server at `demo.flashphoner.com`;
3. the [Player](#) web application to play the captured stream in the browser.
4. Open the Player web app and specify the URL of the camera in the `Stream` field:




The screenshot shows a web application interface with the following elements:

- WCS URL:** A text input field containing `wss://demo.flashphoner.com:844`.
- Stream:** A text input field containing `rtsp://str81.creacast.com/grandlil`, which is highlighted with a yellow background.
- Volume:** A horizontal slider control.
- Full Screen:** A button with a full-screen icon.
- Start:** A button located at the bottom right of the interface.

5. Click the **Start** button. Broadcasting of the captured stream begins:

Player

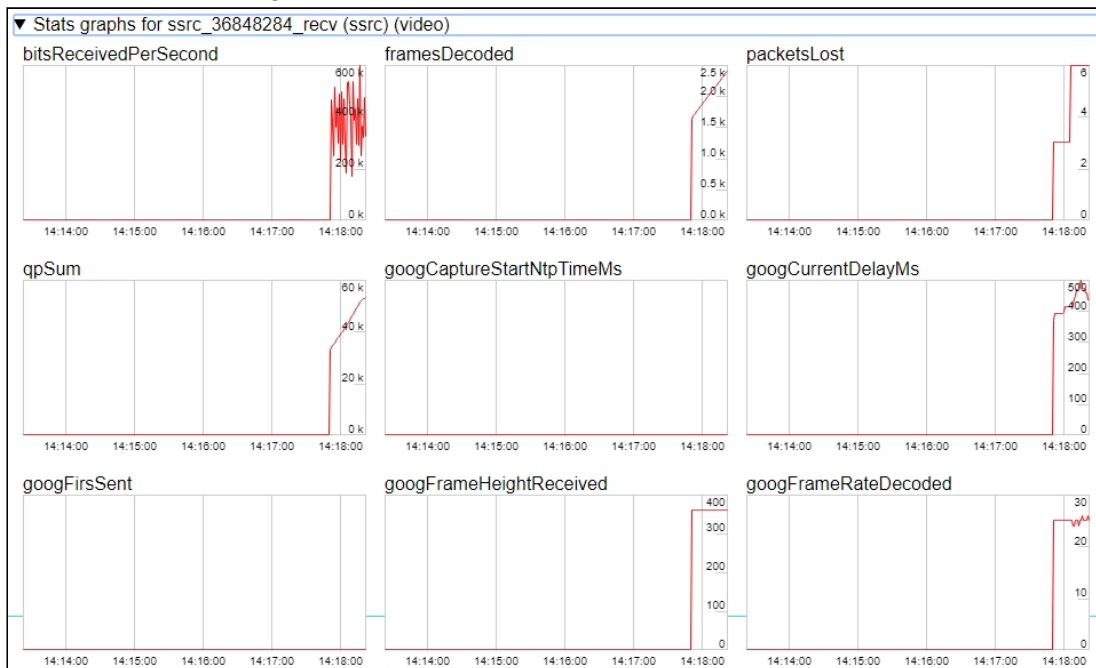


WCS URL

Stream

Volume

6. WebRTC internals diagrams:



RTSP stream capture management by REST API

Usually, it is enough to set the camera URL as stream name to capture stream from IP camera. However, it is possible to manage RTSP stream capture by REST API if necessary.

A REST-query should be HTTP/HTTPS POST request as follows:

- HTTP: `http://test.flashphoner.com:8081/rest-api/rtsp/startup`
- HTTPS: `https://test.flashphoner.com:8444/rest-api/rtsp/startup`

Where:

- `test.flashphoner.com` - is the address of the WCS server
- `8081` - is the standard REST / HTTP port of the WCS server
- `8444` - is the standard HTTPS port
- `rest-api` - is the required part of the URL
- `/rtsp/startup` - REST-method to use

REST methods and responses

REST method	Request body	Response body	Response status	Description
<code>`/rtsp/startup`</code>	<pre>{ "uri": "rtsp://myserver.com/live/myStream", "localStreamName": "myRTSPStream" }</pre>		200 OK 409 Conflict 500 Internal error	Pull the RTSP stream by the specified URL

REST method	Request body	Response body	Response status	Description
<code>`/rtsp/find_all`</code>		<pre>{ "uri": "rtsp:// myserver .com/liv e/myStre am", "status" : "PLAYING ", "localSt reamName ": "myRTSPs tream" }</pre>	200 OK 404 Not found	Find all pulled RTSP-streams
<code>`/rtsp/terminate`</code>	<pre>{ "uri": "r tsp://my server.c om/live/ myStream " }</pre>		200 OK 404 Not found	Terminate the pulled RTSP stream

Parameters

Current status of the stream

Parameter	Description	Example
uri	URL of the RTSP stream	<code>`rtsp://myserver.com/live/myStream`</code>
localStreamName	Name to set to the stream captured	<code>`myRTSPstream`</code>

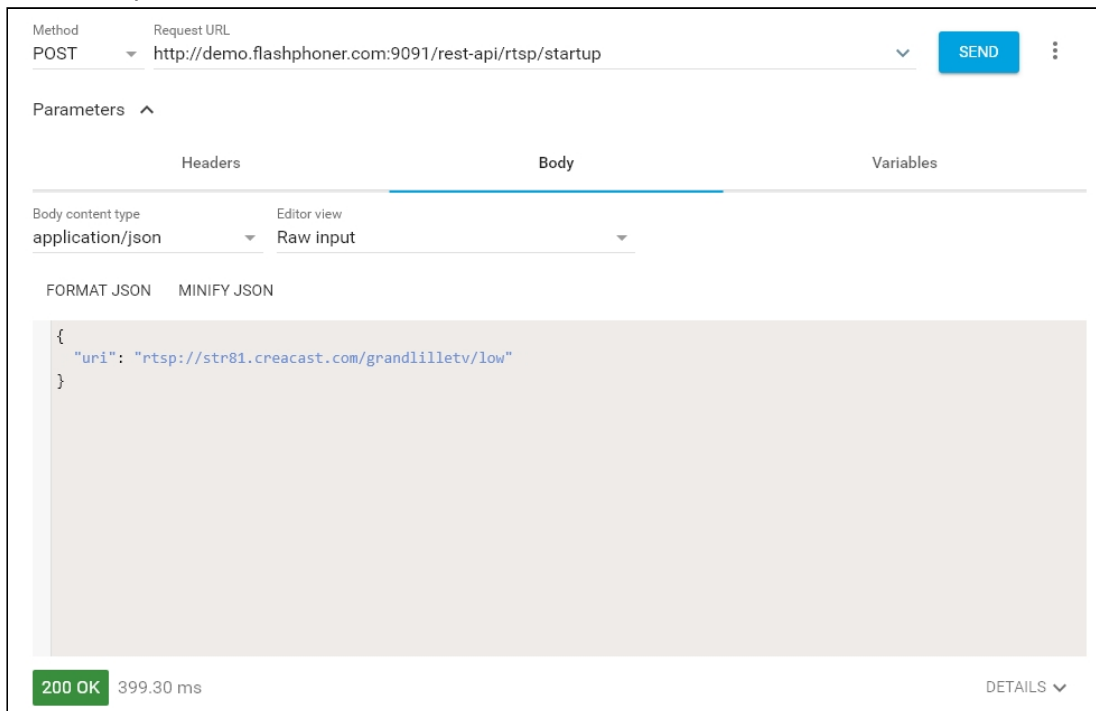
Parameter	Description	Example
status	`PLAYING`	

RTSP stream repeatedly capturing with the same URI

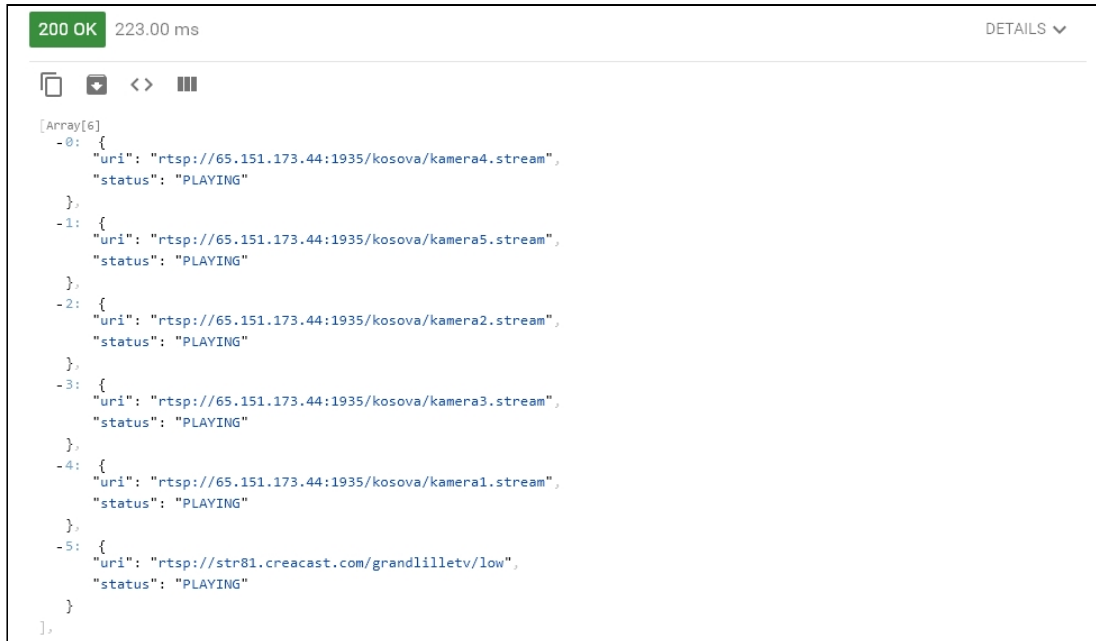
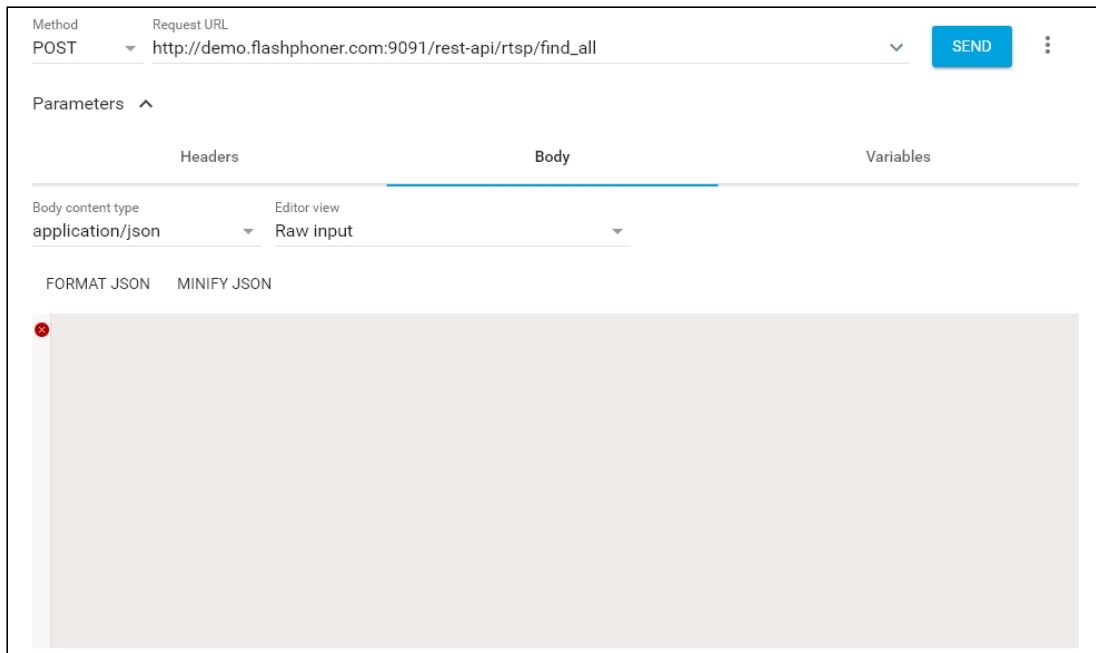
`/rtsp/startup` query returns 409 Conflict while trying to repeatedly capture RTSP stream with the same URI. If the stream from the IP camera is already published on the server, it is necessary to subscribe to it.

Testing

1. For this test we use:
2. the demo server at `demo.flashphoner.com`;
3. the Chrome browser and the [REST-client](#) to send queries to the server;
4. the [Player](#) web application to play the captured stream in the browser.
5. Open the REST client. Send the `/rtsp/startup` query specifying the URL of the RTSP camera in parameters:



6. Make sure the stream is captured by the server. To do this, send the `/rtsp/find_all` query:



7. Open the Player web app, specify the URL of the RTSP camera in the **Stream** field and click **Start**. Browser starts to play the stream:

Player



WCS URL `wss://demo.flashphoner.com:844`

Stream `rtsp://str81.creacast.com/grandlil`

8. Send the `/rtsp/terminate` query specifying the URL of the web camera in parameters:

Method Request URL

POST http://demo.flashphoner.com:9091/rest-api/rtsp/terminate SEND ⋮

Parameters ^

Headers Body Variables

Body content type Editor view

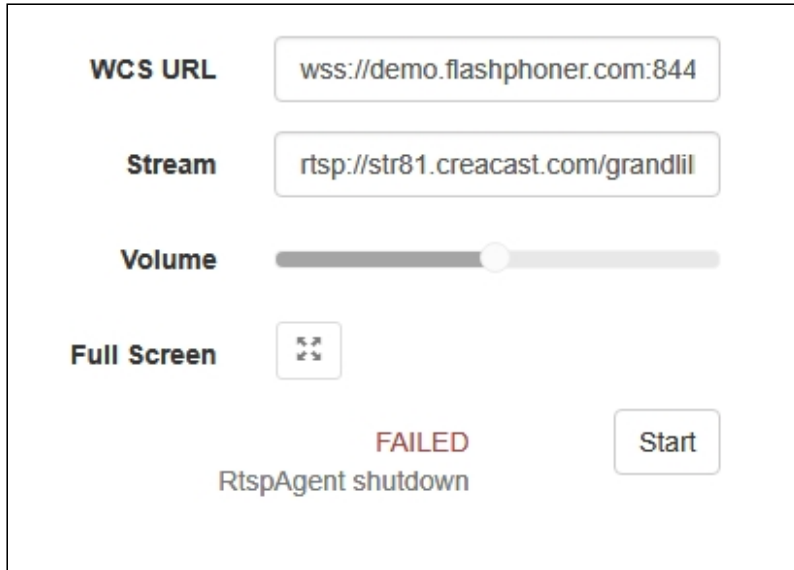
application/json Raw input

FORMAT JSON MINIFY JSON

```
{
  "uri": "rtsp://str81.creacast.com/grandliletv/low"
}
```

200 OK 224.20 ms DETAILS ⌵

9. Stream playback will terminate displaying an error:



Call flow

Below is the call flow when using the Player example

[player.html](#)

[player.js](#)

1. Establishing a connection to the server

`Flashphoner.createSession()` [code](#)

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED,
function(session){
    setStatus(session.status());
    //session connected, start playback
    playStream(session);
}).on(SESSION_STATUS.DISCONNECTED, function(){
    setStatus(SESSION_STATUS.DISCONNECTED);
    onStoped();
}).on(SESSION_STATUS.FAILED, function(){
    setStatus(SESSION_STATUS.FAILED);
    onStoped();
});
```

2. Receiving an event confirming successful connection

`SESSION_STATUS.ESTABLISHED` [code](#)

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED,
function(session){
    setStatus(session.status());
    //session connected, start playback
    playStream(session);
```

```

    }).on(SESSION_STATUS.DISCONNECTED, function(){
        ...
    }).on(SESSION_STATUS.FAILED, function(){
        ...
    });

```

3. Request to play the stream

`Session.createStream()`, `Stream.play()` [code](#)

IP camera URL is passed to `createStream()` method as stream name

```

function playStream(session) {
    var streamName = $('#streamName').val();
    var options = {
        name: streamName,
        display: remoteVideo,
        flashShowFullScreenButton: true
    };
    ...
    stream = session.createStream(options).on(STREAM_STATUS.PENDING,
function(stream) {
        ...
    });
    stream.play();
}

```

4. Request from WCS to the RTSP source to broadcast the stream

5. Broadcasting the RTSP stream

6. Receiving an event confirming successful capturing and playing of the stream

`STREAM_STATUS.PLAYING` [code](#)

```

stream = session.createStream(options).on(STREAM_STATUS.PENDING,
function(stream) {
    ...
}).on(STREAM_STATUS.PLAYING, function(stream) {
    $("#preloader").show();
    setStatus(stream.status());
    onStarted(stream);
    ...
});
stream.play();

```

7. Sending audio and video stream via WebRTC

8. Stopping playing the stream

`Stream.stop()` [code](#)

```

function onStarted(stream) {
    $("#playBtn").text("Stop").off('click').click(function(){
        $(this).prop('disabled', true);
        stream.stop();
    }).prop('disabled', false);
    $("#fullScreenBtn").off('click').click(function(){

```

```
    stream.fullScreen();
  }).prop('disabled', false);
  $("#volumeControl").slider("enable");
  stream.setVolume(currentVolumeValue);
}
```

9. Receiving from an event confirming successful stop of the stream playback

`STREAM_STATUS.STOPPED` [code](#)

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING,
function(stream) {
  ...
}).on(STREAM_STATUS.PLAYING, function(stream) {
  ...
}).on(STREAM_STATUS.STOPPED, function() {
  setStatus(STREAM_STATUS.STOPPED);
  onStopped();
}).on(STREAM_STATUS.FAILED, function(stream) {
  ...
}).on(STREAM_STATUS.NOT_ENOUGH_BANDWIDTH, function(stream){
  ...
});
stream.play();
```

RTSP connection reuse

If other subscribers request the stream captured from RTSP IP camera, the previous RTSP connection will be used if all subscribers set the same camera URL. For example, two requests to the same IP camera

```
rtsp://host:554/live.sdp
```

and

```
rtsp://host:554/live.sdp?p=1
```

are different, then two RTSP connections will be created if streams from both URLs are requested.

Stream capture authentication

WCS supports RTSP stream capture authentication by user name and password, user data should be set in stream URL, for example

```
rtsp://user:password@hostname/stream
```

If name or password contains any special characters, they should be encoded such as

```
rtsp://user:p%40ssword@hostname/stream
```

Where

- `user` is user name
- `p@ssword` is password with character `@`, it is encoded in URL.

Another IP address redirection handling

Some IP cameras return `302 Moved Temporarily` in response to `DESCRIBE` or `OPTIONS` query to redirect a client to another IP address for RTSP connection establishing. WCS supports this feature since build 5.2.179.

In this case, if the IP camera redirects requests to another address, and if client establishes connections separately to this camera and directly to the camera where requests are redirected, it is two different streams for WCS. The pulling agents are created for every of those streams, and subscribers connect to one of those agents depending on address set on connection establishing.

RTSP pulled stream publishing with a given name

The ability was added to publish RTSP pulled stream with a given name since build 5.2.479. The stream name should be set with `toStream` parameter of `/rtsp/startup` REST query, for example

```
POST /rest-api/rtsp/startup HTTP/1.1
Host: localhost:8081
Content-Type: application/json

{
  "toStream": "stream1",
  "uri": "rtsp://myserver.com/live/myStream"
}
```

By default, if `toStream` parameter is not set, the stream name will be formed from RTSP URI. If RTSP stream with such URI is already pulled, or the stream with given name is already exists, server returns `409 Conflict`.

In latest WCS builds, the parameter is renamed to `localStreamName`:

```
POST /rest-api/rtsp/startup HTTP/1.1
Host: localhost:8081
Content-Type: application/json
```

```
{
  "uri": "rtsp://myserver.com/live/myStream",
  "localStreamName": "stream1"
}
```

If the name is set to the RTSP stream pulled, this stream can be played in [CDN](#) by its name (by default, CDN playback is not available for RTSP streams because they always are pulled locally).

Capturing H265 RTSP stream

Since build [5.2.1579](#) it is possible to capture RTSP stream published by camera as H265. To enable this, H265 should be added to supported codecs list

```
codecs=opus,alaw,ulaw,g729,speex16,g722,mpeg4-generic,telephone-
event,h264,vp8,flv,mpv,h265
```

and to exclusion lists

```
codecs_exclude_sip=mpeg4-generic,flv,mpv,h265
codecs_exclude_sip_rtmp=opus,g729,g722,mpeg4-generic,vp8,mpv,h265
codecs_exclude_sfu=alaw,ulaw,g729,speex16,g722,mpeg4-generic,telephone-
event,flv,mpv,h265
```

Stream captured may be played as [WebRTC](#), [RTMP](#), [MSE](#), [HLS](#) with transcoding and as [RTSP without transcoding](#)

Warning

A stream should not contain B-frames! If B-frames occur in the stream, it may be played as RTSP without transcoding only

A first subscriber issue

Before WCS build [5.2.1760](#) RTSP streams may start to play with a huge delay for the first subscriber. This was due to key frames missing caused by subscriber thread starting after publisher thread. Since build [5.2.1760](#) the behavior was changed: publisher thread starts after subscriber thread. This may be reverted if necessary by the following parameter

```
agent_use_subscriber_listener=false
```

Stream timestamp fix

In some RTSP streams a frame timestamps may be in wrong order, for example two subsequent frames may have the same timestamp. Such stream may not be displayed or periodically display a gray square while playing via WebRTC. Since build [5.2.1794](#) the following parameter is available to fix a broken timestamps

```
jitter_buffer_attempt_to_correct_broken_timestamp=true
```

In this case RTSP capturing client log may contain a messages as follows

```
Non-monotonous timestamp in input stream; previous: 453424, current: 453424;  
changing to 453425. This may result in incorrect timestamps in the output
```

and the problem stream should play normally.

Known issues

1. A stream containing B-frames does not play or plays with artifacts (twitches, lags)

Symptoms

- a stream sent by the RTMP encoder does not play or plays with twitches or lags
- warnings in the [client log](#):

```
09:32:31,238 WARN 4BitstreamNormalizer - RTMP-pool1-10-thread-5 It is B-  
frame!
```

Solution

- change the camera settings to prevent B-frames encoding (use lower encoding profile, set B frames count to 0 in the command line etc)
- [transcode](#) the stream, in this case there will be no B-frames in transcoded stream

2. AAC frames of type 0 are not supported by decoder and will be ignored while stream pulled playback

Symptoms

Warnings are displayed in the [client log](#):

```
10:13:06,815 WARN AAC - AudioProcessor-c6c22de8-a129-43b2-bf67-1f433a814ba9 Dropping AAC frame that starts with 0, 119056e500
```

Solution

Use Fraunhofer AAC codec with the following parameter in [flashphoner.properties](#) file

```
use_fdk_aac=true
```

3. When publishing and then playing and recording H264 + AAC stream video may be out of sync with sound, or no sound at all

Symptoms

When playing H264 + AAC stream published on server and recording such stream, sound is out of sync with video or absent

Solution

a) set the following parameter in [flashphoner.properties](#) file

```
disable_drop_aac_frame=true
```

This parameter also turns off AAC frames dropping.

b) use Fraunhofer AAC codec

```
use_fdk_aac=true
```

4. Sound may be distorted or absent when resampled to 11025 Hz

Symptoms

When H264 + AAC stream published on WCS server is played with AAC sample rate 11025 Hz, sound is distorted or absent

✓ Solution

Do not use 11025 Hz sample rate, or escape AAC sound resampling to this rate, for example, do not set this sample rate in [SDP settings](#).

5. Connection to the IP camera is lost on error in any track (audio or video)

🚫 Symptoms

Connection to the IP camera is lost if one of tracks returns error `4**`

✓ Solution

This behavior is enabled by default. However if one-time errors are not critical and should not terminate broadcasting, in the [flashphoner.properties](#) files set

```
rtsp_fail_on_error_track=false  
rtp_force_synchronization=true
```

6. All the characters in a stream name, that are not allowed in URI, should be encoded

🚫 Symptoms

RTSP stream is not played with `Bad URI` error

✓ Solution

Any character that is not allowed in URI, should be encoded in stream URL, for example

```
rtsp://hostname/c@@lstream/channel1
```

should be encoded as

```
rtsp://hostname/c%40%40lstream/channel1
```

7. Some IP cameras do not support `cnonce` field in RTSP connection message header

Symptoms

RTSP stream is played with VLC, but is not played with WCS

Solution

Set the following parameter in `flashphoner.properties` file

```
rtsp_auth_cnonce=
```

with empty value.

8. Streams from some cameras cannot be played due to buffer size lack to write Rbsp

Symptoms

RTSP stream is not playing with exceptions in server log

```
13:10:16,988 ERROR BitstreamNormalizer - pool-56-thread-1 Failed to  
normalize SPS 674d002a95a81e0089f950  
java.lang.RuntimeException: Failed to write sps rbsp
```

Solution

Increase Rbsp buffer size setting (1.5 by default)

```
h264_sps_rbsp_scale=2
```

9. Some camera streams can loose audio and video sync

Symptoms

RTSP stream freezes or cannot be played by HLS (some segments cannot be written), in stream statistics there are ubnormally big AV synchronization values

```
streams_synchronization=camera1/-21800;camera2/2079600704
```

✓ **Solution**

In build before [5.2.1775](#) increase synchronization buffers for audio and video tracks

```
audio_incoming_buffer_size=100  
video_incoming_buffer_size=100
```

since build [5.2.1775](#) increase force synchronization timeout for audio and video tracks

```
video_force_sync_timeout=1000  
audio_force_sync_timeout=1000
```