

From an IP camera via RTSP

- [Overview](#)
 - [RTSP sources](#)
 - [Supported codecs](#)
 - [Supported platforms and browsers](#)
- [Operation flowchart](#)
- [Configuration](#)
- [Quick manual on testing](#)
 - [Capturing of a video stream from the IP camera and playing it in a browser](#)
- [Stream capture from the IP camera management by REST API](#)
 - [Testing](#)
 - [REST-queries](#)
 - [REST-methods and response statuses](#)
 - [Parameters](#)
- [Call flow](#)
- [RTSP connection reuse](#)
- [Stream capture authentication](#)
- [Known issues](#)

Overview

A video stream is captured from an RTSP source that provides audio and video in the supported codecs. Then, the server transforms this video stream for playing in browsers and mobile devices.

RTSP sources

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

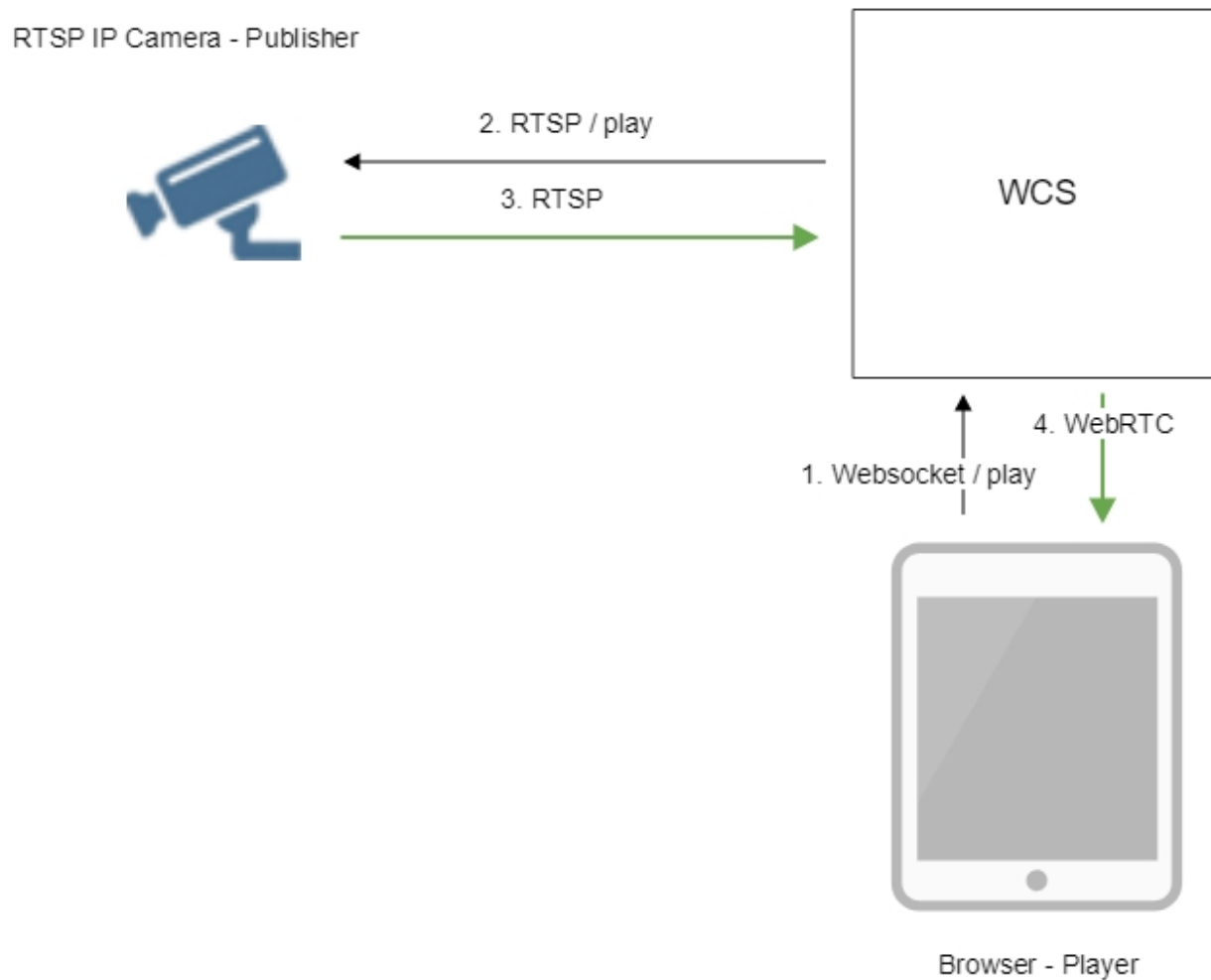
Supported codecs

- H.264
- VP8
- AAC
- G.711
- Speex

Supported platforms and browsers

	Chrome	Firefox	Safari 11	Internet Explorer	Edge
Windows	+	+		+	+
Mac OS	+	+	+		
Android	+	+			
iOS	-	-	+		

Operation flowchart



1. The browser establishes a connection to the server via the Websocket protocol and sends the play 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 transforms the stream to WebRTC and gives the stream to the browser.

Configuration

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

```
rtsp_client_address=172.16.0.3
```

Quick manual on testing

Capturing of a video stream from the IP camera and playing it in a browser

1. For this test we use:
 - the demo server `atdemo.flashphoner.com`;
 - the `Player` web application to play the captured stream in the browser.

2. Open the Player web app and specify the URL of the camera in the "Stream" field:

WCS URL

wss://demo.flashphoner.com:844

Stream

rtsp://str81.creacast.com/grandlil

Volume

Full Screen

Start

3. Click the "Start" button. Broadcasting of the captured stream begins.

Player



WCS URL

wss://demo.flashphoner.com:844

Stream

rtsp://str81.creacast.com/grandlil

Volume

4. WebRTC internals diagrams:



Stream capture from the IP camera 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.

Testing

1. For this test we use:

- the demo server at demo.flashphoner.com;
- the Chrome browser and the [REST-client](#) to send queries to the server;
- the [Playerweb](#) application to play the captured stream in the browser.

2. Open the REST client. Send the /rtsp/startup query specifying the URL of the web camera in parameters:

Method

Request URL

POST

http://demo.flashphoner.com:9091/rest-api/rtsp/startup

SEND

Parameters ^

Headers

Body

Variables

Body content type

Editor view

application/json

Raw input

FORMAT JSONMINIFY JSON

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

200 OK399.30 ms

DETAILS v

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

Method

Request URL

POST

http://demo.flashphoner.com:9091/rest-api/rtsp/find_all

SEND

Parameters ^

Headers

Body

Variables

Body content type

Editor view

application/json

Raw input

FORMAT JSONMINIFY JSON

200 OK223.00 ms

DETAILS

[Array[6]]

-0: {

"uri": "rtsp://65.151.173.44:1935/kosova/kamera4.stream",

"status": "PLAYING"

}

,

-1: {

"uri": "rtsp://65.151.173.44:1935/kosova/kamera5.stream",

"status": "PLAYING"

}

,

-2: {

"uri": "rtsp://65.151.173.44:1935/kosova/kamera2.stream",

"status": "PLAYING"

}

,

-3: {

"uri": "rtsp://65.151.173.44:1935/kosova/kamera3.stream",

"status": "PLAYING"

}

,

-4: {

"uri": "rtsp://65.151.173.44:1935/kosova/kamera1.stream",

"status": "PLAYING"

}

,

-5: {

"uri": "rtsp://str81.creacast.com/grandlilletv/low",

"status": "PLAYING"

}

}

,

4. Open the Player web app and in the "Stream" field specify the URL of the web camera and click Start. Browser starts to play the stream:

Player

11:11

Lille 10.5° C

GRAND LILLE TV

Robersart Color One, ce Samedi, à Wambrechies. Un hommage sera rendu à la petite Angélique...

HOMMAGE

WCS URL

wss://demo.flashphoner.com:844

Stream

rtsp://str81.creacast.com/grandlill

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

The screenshot shows a REST client interface with the following details:

- Method:** POST
- Request URL:** http://demo.flashphoner.com:9091/rest-api/rtsp/terminate
- Parameters:** Expanded section showing Headers, Body, and Variables tabs.
- Body content type:** application/json
- Editor view:** Raw input
- Body content:**

```
{  "uri": "rtsp://str81.creacast.com/grandlilletv/low"}
```
- Buttons:** FORMAT JSON, MINIFY JSON
- Status:** 200 OK, 224.20 ms
- Details:** DETAILS button with a dropdown arrow.

6. Stream playback will terminate displaying an error:

The screenshot shows a stream playback control interface with the following details:

- WCS URL:** wss://demo.flashphoner.com:844
- Stream:** rtsp://str81.creacast.com/grandlil
- Volume:** A slider control.
- Full Screen:** A button with a full screen icon.
- Error Message:** FAILED RtspAgent shutdown
- Start Button:** A button labeled Start.

REST-queries

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](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 response statuses

REST-method	Example of REST-query	Example of response	Response statuses	Description
/rtsp /startup	<pre>{ "uri": "rtsp://myserver.com /live/myStream" }</pre>		409 - Conflict 500 - Internal error	Pull the RTMP stream by the specified URL
/rtsp /find_all		<pre>{ "uri": "rtsp://myserver.com /live/myStream", "status": "PLAYING" }</pre>	200 – streams found 404 – streams not found	Find all pulled RTMP-streams
/rtsp /terminate	<pre>{ "uri": "rtsp://myserver.com /live/myStream" }</pre>		200 - stream terminated 404 - stream not found	Terminate the pulled RTMP stream

Parameters

Parameter name	Description	Example
uri	URL of the RTSP stream	rtsp://myserver.com/live/myStream
status	Current status of the stream	PLAYING

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);
  onStopped();
}).on(SESSION_STATUS.FAILED, function(){
  setStatus(SESSION_STATUS.FAILED);
  onStopped();
});
```


2. Receiving from the server an event confirming successful connection.

ConnectionStatusEvent 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 from the server an event confirming successful capturing and playing of the stream.

StreamStatusEvent, cratyc PLAYING`code`

```
stream = session.createStream(options).on(STREAM_STATUS.PENDING, function(stream) {
    ...
}).on(STREAM_STATUS.PLAYING, function(stream) {
    $("#preloader").show();
    setStatus(stream.status());
    onStart(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 the server an event confirming successful stop of the stream playback.

StreamStatusEvent, cratyc [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 differ, 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 escaped such as

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

Where

- user is user name
- p@ssword is password with character '@', it is escaped in URL.

Known issues

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

Symptoms:

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

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

Solution: change the encoder settings so, that B-frames were not used (lower encoding profile, specify in the command line etc).

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

In this case, warnings will be 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 when 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 escaped

Symptoms: RTSP stream is not played with 'Bad URI' error

Solution: any character that is not allowed in URI, should be escaped in stream URL, for example

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

should be set 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.