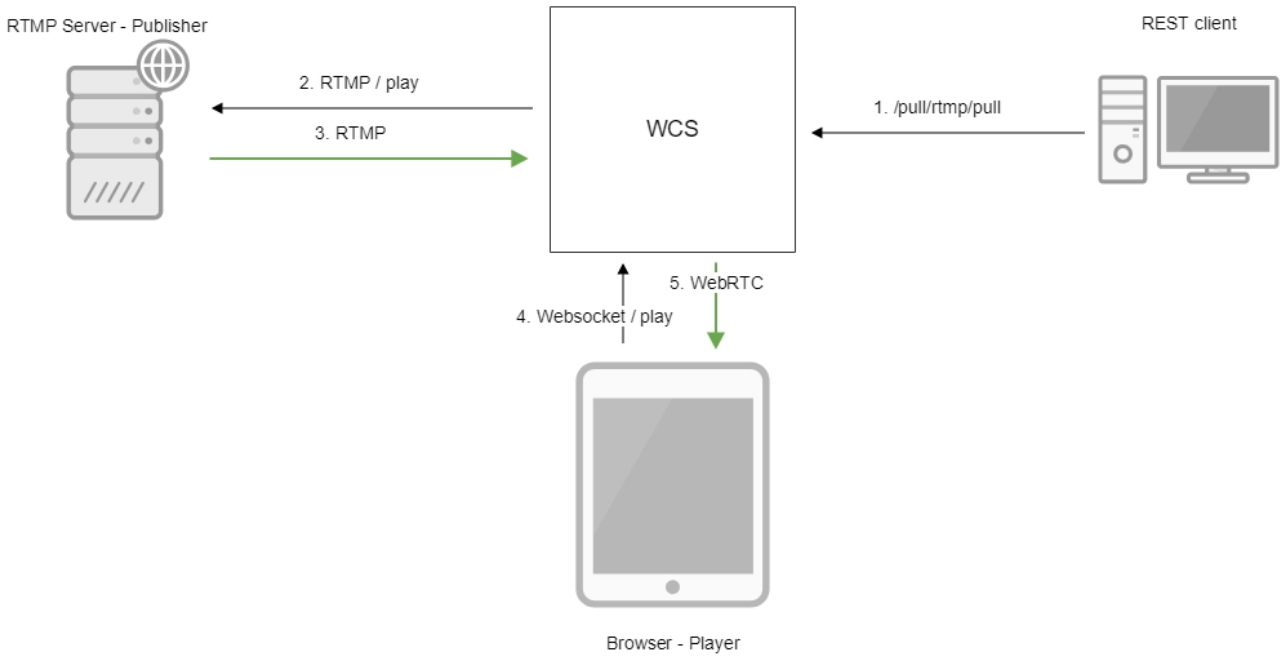
From another server via RTMP

Overview

WCS can capture an RTMP video stream published by another server, by request. The captured stream can be broadcast toany of supported platformsus ingany of supported technologies. Managing of RTMP stream capturing is performed using REST API.

Operation flowchart



1. The /pull/rtmp/pull REST query is sent to the WCS server
2. The WCS server requests the RTMP stream from the specified server
3. The RTMP stream is broadcast to the WCS server
4. The browser requests playing the captured stream via WebSocket
5. The browser receives the stream via WebRTC

REST queries

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

- - HTTP:http://test.flashphoner.com:8081/rest-api/pull/rtmp/pull
- - HTTPS:https://test.flashphoner.com:8444/rest-api/pull/rtmp/pull

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
- - /pull/rtmp/pull- is the REST method used

REST-methods and response statuses

REST-method	Example of REST-query	Example of REST response	Response status	Description

/pull/pull	<pre>{
	"uri":"rtmp://myserver.com
	/live/myStream",
	"localStreamName":"stream1",
	"record": "false",
	"hasAudio": true,
	"hasVideo": true
	}</pre>
			409 - Conflict
			
			500 - Internal error
				Pull the RTMP stream at the specified URL

/pull
/find_all
		<pre>{
		"localMediaSessionId": "5a072377-
		73c1-4caf-abd3"
		"localStreamName": "stream1"
		"uri": "rtmp://myserver.com/live
		/myStream",
		"status": "PROCESSED_REMOTE",
		"hasAudio": true,
		"hasVideo": true,
		"record": false
		}</pre>
			200 – streams are found
			
			500 - Internal error
				Find all pulled RTMP streams

/pull
/terminate
	<pre>{
	"uri":"rtmp://myserver.com
	/live/myStream"
	}</pre>
			200 - stream terminated
			
			500 - Internal error
				Terminate the pulled RTMP stream

Parameters

Parameter name	Description	Example
uri	URL of the RTMP stream	rtmp://myserver.com/live /myStream
record	Is the pulled stream recording	true
localMediaSessionId	Session identifier	5a072377-73c1-4caf-abd3
localStreamName	Local name assigned to the captured stream. By this name the stream can be requested from the WCS server	stream1
status	Current stream status	PROCESSED_REMOTE
hasAudio	Stream has audio track	true
hasVideo	Stream has video track	true

Pulled stream publishing with a given name

Since build5.2.724it is possible to set the name to publish stream on server using `localStreamName` query parameter. If the parameter is not set, the stream name will be set to `uri`, as done in previous builds.

RTMP stream repeatedly capturing with the same URI

`/pull/rtmp/pull` query returns `409 Conflict` while trying to repeatedly capture RTMP stream with the same URI. If the stream is already published on the server, it is necessary to subscribe to it.

Audio only or video only stream publishing

Since build 5.2.1439 it is possible to set in REST query how to publish an RTMP stream captured on WCS. For example, to publish a video only stream, set

```
{
 "uri":"rtmp://myserver.com/live/myStream",
 "localStreamName":"stream1",
 "hasAudio": false,
 "hasVideo": true
}
```

Configuration

Codecs parameters definition

In the `/usr/local/FlashphonerWebCallServer/conf` directory you can find the SDP description file for the RTMP agent `rtmp_agent.sdp`:

```
v=0
o=- 1988962254 1988962254 IN IP4 0.0.0.0
c=IN IP4 0.0.0.0
t=0 0
a=sdplang:en
m=video 0 RTP/AVP 95
a=rtpmap:95 H264/90000
a=fmtp:95 profile-level-id=42e01f;packetization-mode=1
a=sendonly
m=audio 0 RTP/AVP 103 96 97 98 99 100 102 108 104
a=rtpmap:108 mpeg4-generic/48000/2
a=rtpmap:96 mpeg4-generic/8000/2
a=rtpmap:97 mpeg4-generic/11025/2
a=rtpmap:98 mpeg4-generic/12000/2
a=rtpmap:99 mpeg4-generic/16000/2
a=rtpmap:100 mpeg4-generic/22050/2
a=rtpmap:104 mpeg4-generic/24000/2
a=rtpmap:102 mpeg4-generic/32000/2
a=rtpmap:103 mpeg4-generic/44100/2
a=recvonly
```

To enable recording of both audio and video (instead of audio only) during captured stream recording specify the following attribute in this file

```
a=sendonly
```

for video.

Publishing a stream without audio or video

If audio or video description is removed from `rtmp_agent.sdp` , an RTMP stream captured will be published on WCS without audio or video respectively. For example, use the following SDP to publish video only:

```
v=0
o=- 1988962254 1988962254 IN IP4 0.0.0.0
c=IN IP4 0.0.0.0
t=0 0
a=sdplang:en
m=video 0 RTP/AVP 95
a=rtpmap:95 H264/90000
a=fmtp:95 profile-level-id=42e01f;packetization-mode=1
a=sendonly
```

Frame type detection

Sometimes, RTMP stream source may set a media frame type incorrectly at RTMP protocol level. In this case frame type should be detected according to frame content. To enable this, use the following parameter added in build 5.2.1446

```
rtmp_detect_h264_frame_type=true
```

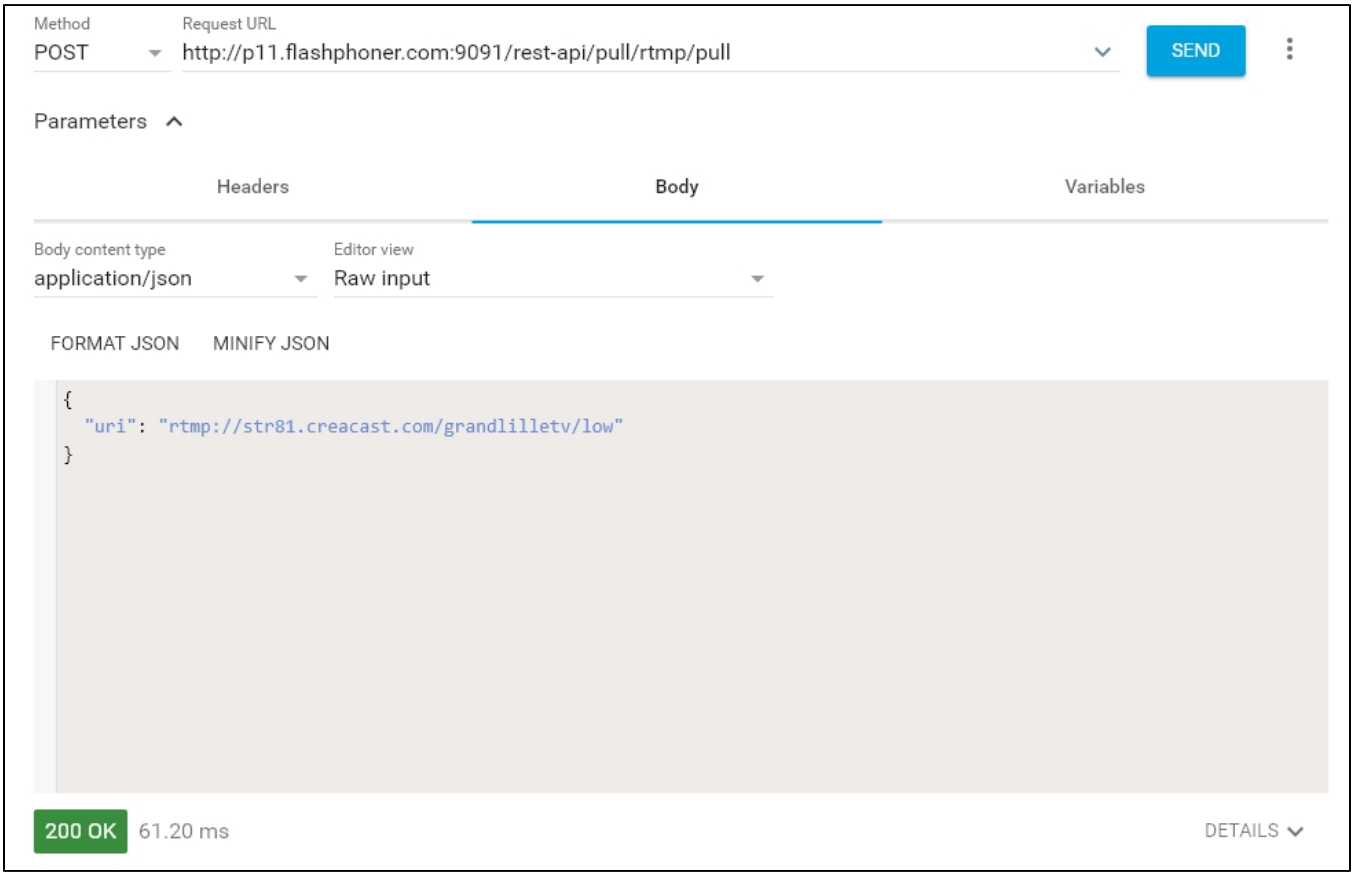
Quick manual on testing

Capturing of an RTMP stream broadcast by another server using the REST-query /pull/rtmp/pull

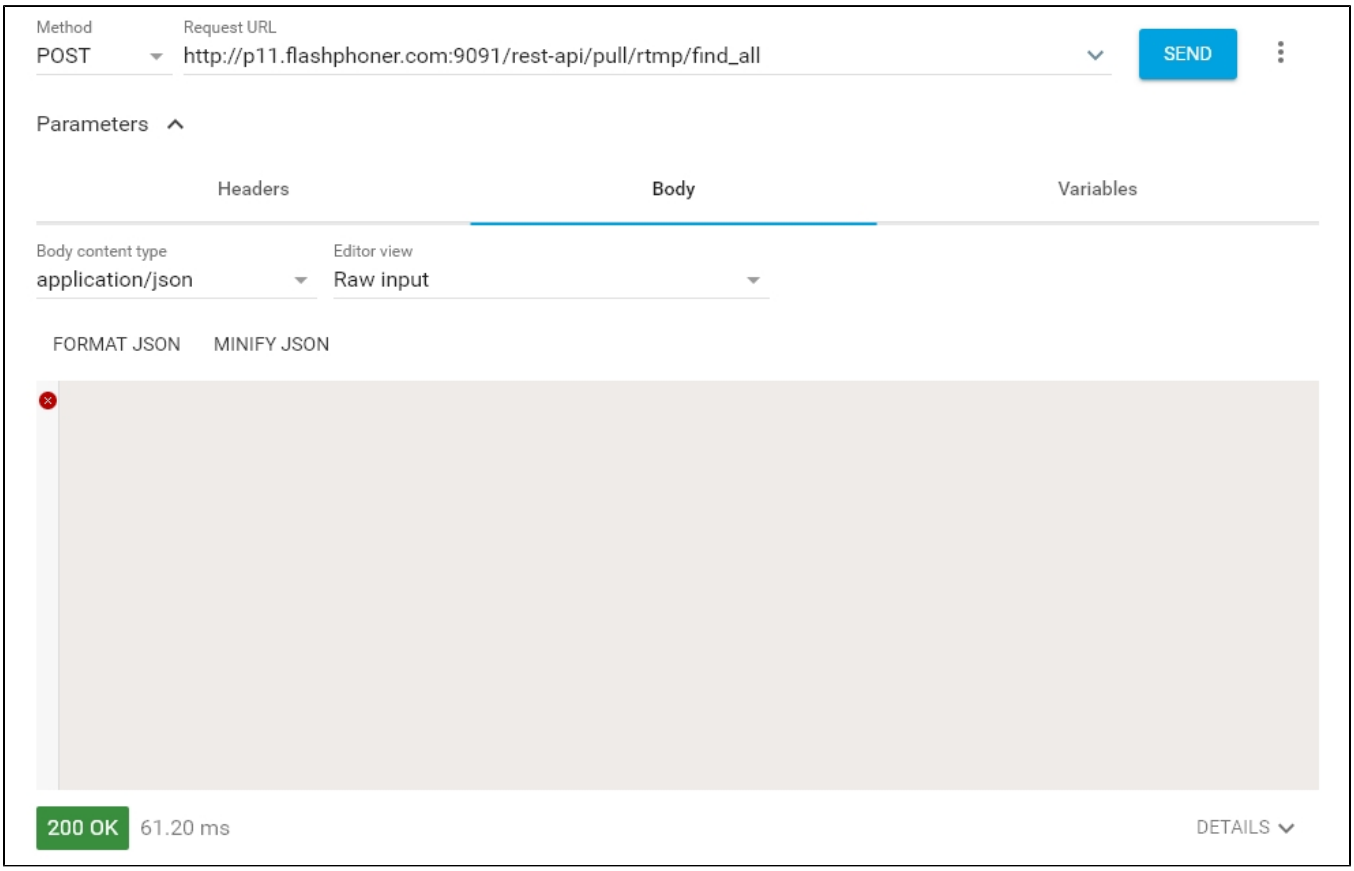
1. For the test we use:

- the demo server at demo.flashphoner.com;
- the Chrome browser and theREST-clientto send queries to the server;
- theTwo Way Streamingweb application to play the captured stream in a browser.

2. Open the REST client. Send the /pull/rtmp/pull query and specify the URL of the RTMP stream in parameters:

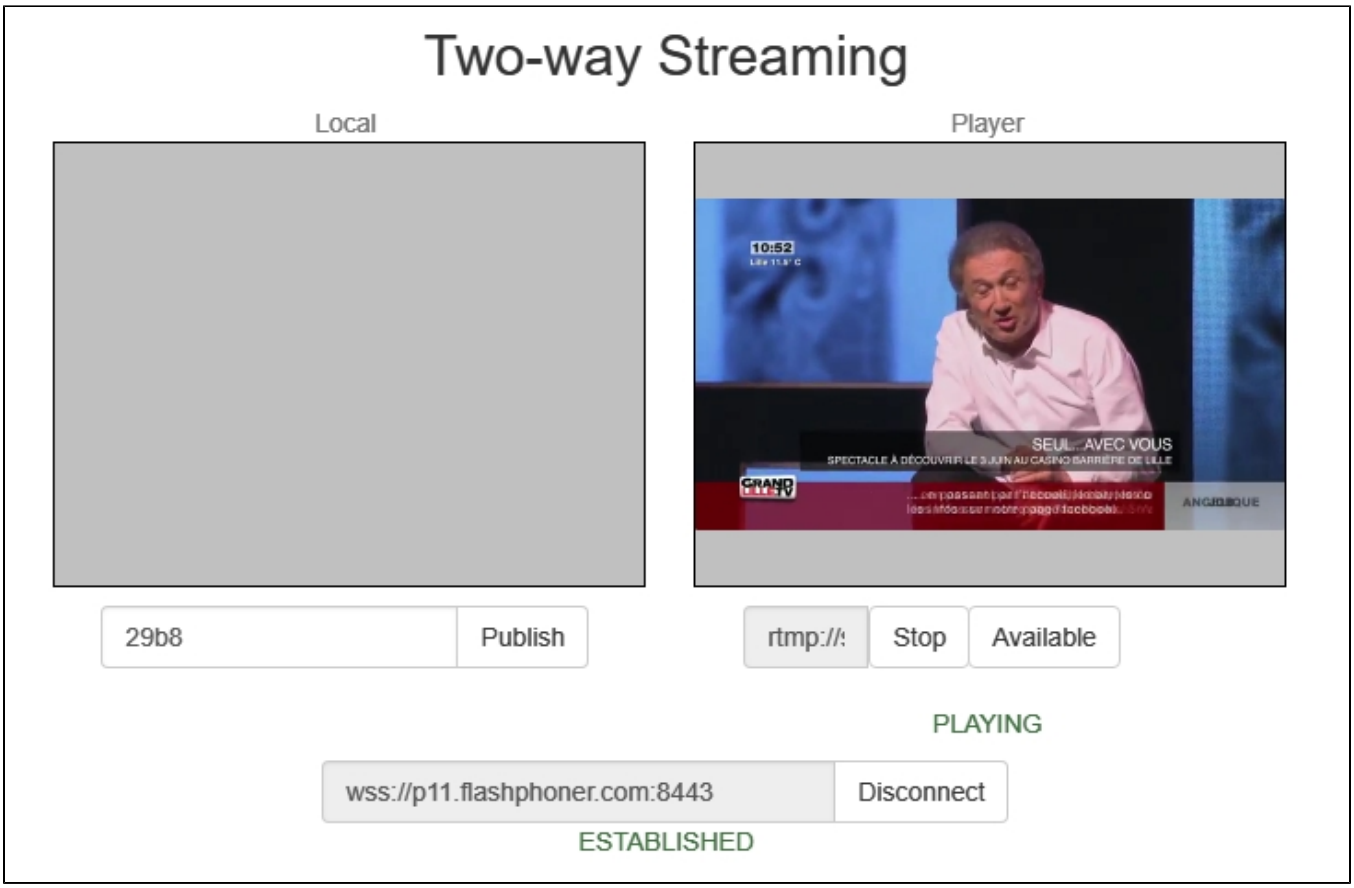


3. Make sure the stream is captured by the server. To do this, send the /rtmp/pull/find_all request:



and copy the local name of the stream from the localStreamName response parameter:

4. Open the page of the Two Way Streaming web application. Click "Connect" and specify the local stream name, then click "Play":



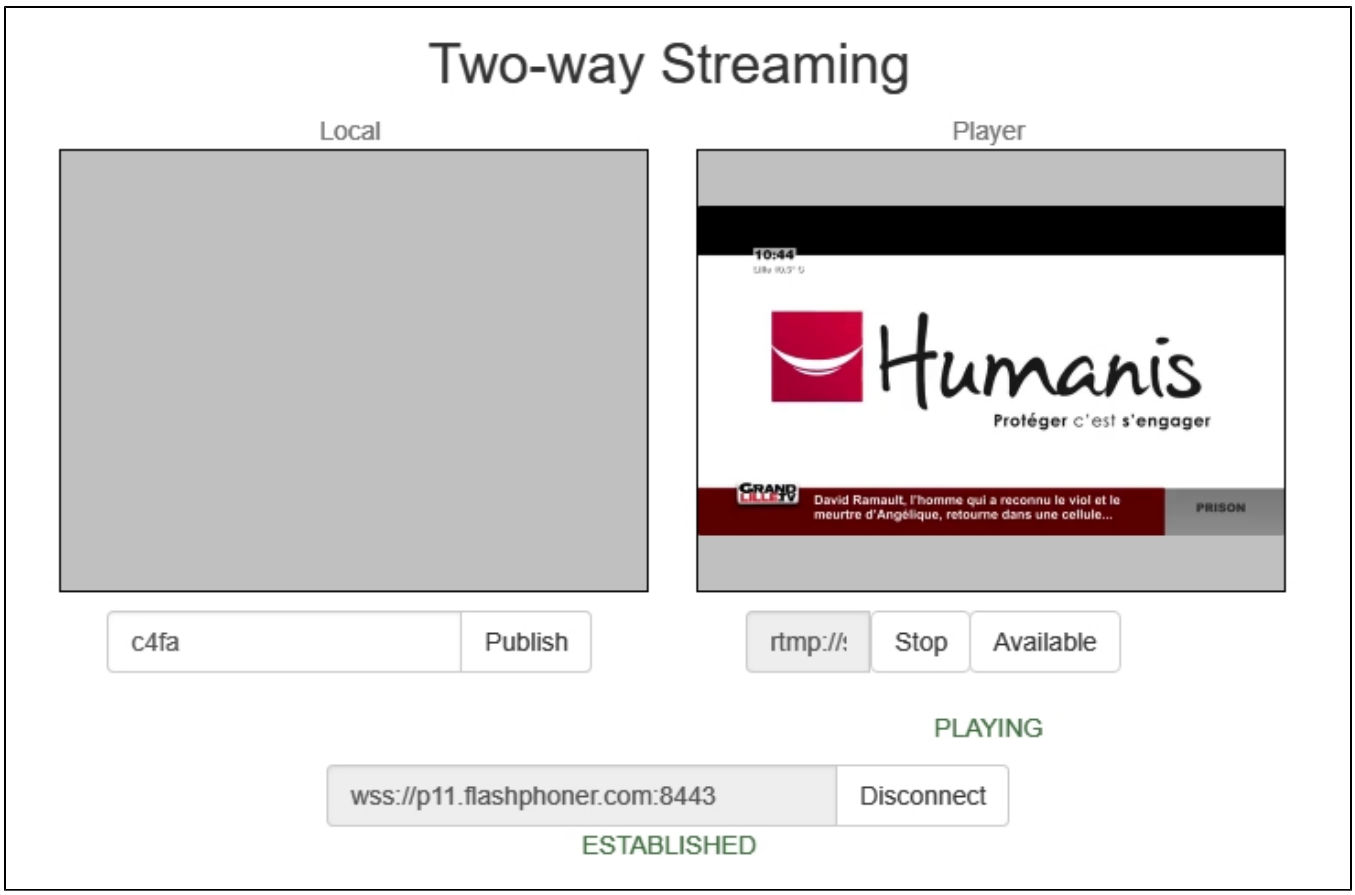
5. WebRTC internals diagrams in a browser:

Capturing of an RTMP stream broadcast by another server without using REST queries

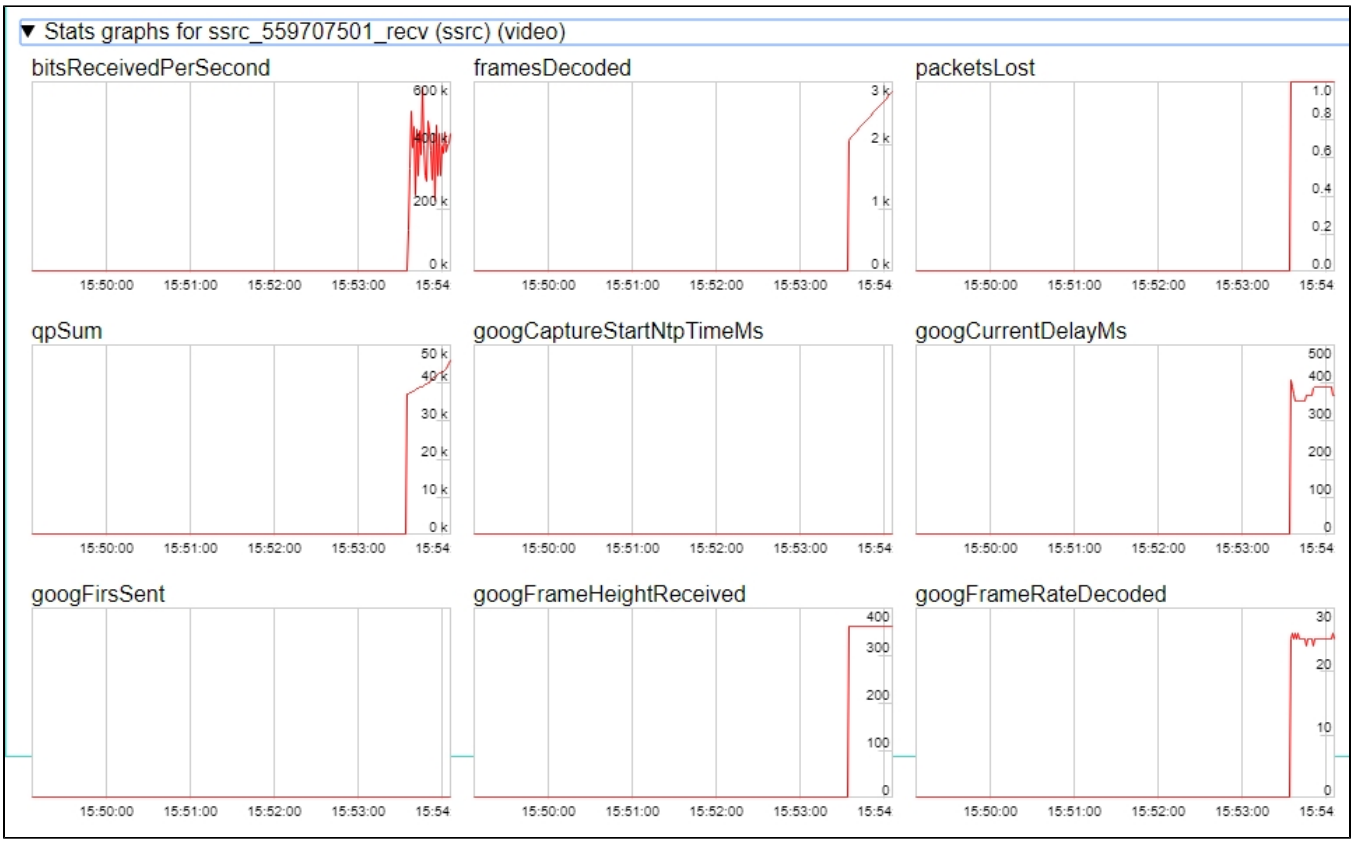
1. For the test we use:

- the demo server at demo.flashphoner.com;
- the web application,Two Way Streaming, to capture and play the captured stream in a browser.

2. Open the page of the Two Way Streaming web application. Click "Connect" and specify the name of theRTMP stream you want to capture, then click "Play":

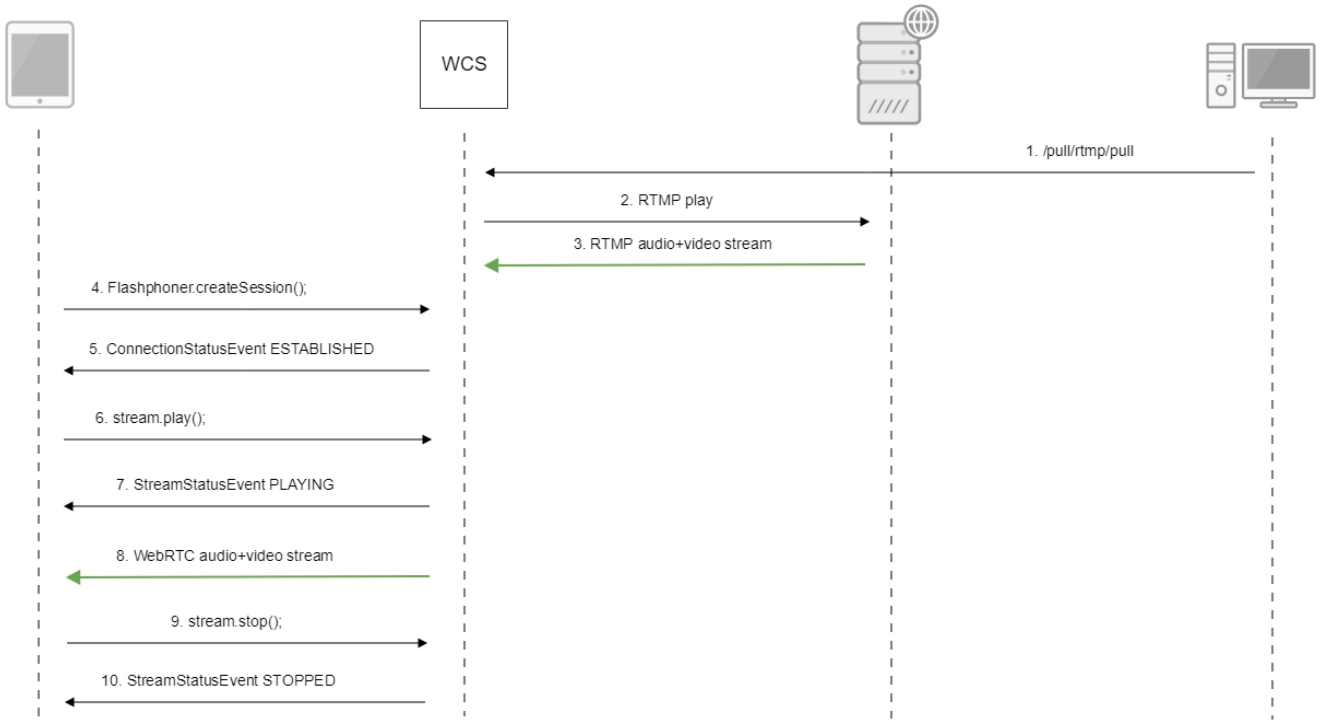


3. WebRTC internals diagrams in a browser:



Call flow

Below is the call flow when capturing an RTMP stream from another server



Authentication on a source server

WCS supports Adobe authentication on RTMP server while capturing a stream from it using RTMP URL parameters:

```
rtmp://username:password@server:1935/live/streamKey
```

Note that if an RTMP stream is requested from another WCS server, this kind of authentication is not supported.

Since build 5.2.1069 it is possible to pass authentication parameters after stream name

```
rtmp://server:1935/live/streamKey?user=username&password=password
```

In this case the parameters will be passed to RTMP server in `connect` message.

The parameters can be set after application name too

```
rtmp://server:1935/live?user=username&password=password/streamKey
```

In this case the parameters will also be passed to RTMP server in `connect` message. If the stream is requested from another WCS server via RTMP, authentication parameters will be available in REST hook /connect.

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 theclient 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)
- 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

In this case, warnings will be displayed in theclient 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 inflashphoner.propertiesfile

```
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 recordingsuch stream, sound is out of sync with video or absent

Solution:

a) set the following parameter inflashphoner.propertiesfile

```
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. Some RTMP functions does not supported and will be ignored:

- FCSubscribe
- FCPublish
- FCUnpublish
- onStatus
- onUpstreamBase
- releaseStream

6. When recording the captured stream to the disk, only audio is recorded

Symptoms: when the "record": "true" parameter is set in the /pull/rtmp/pull REST query, the received file lacks video track, only audio is present.
Solution: in theSDP settingsset the

```
a=sendonly
```

attribute for the video track.

7. Freezes are possible when RTMP stream is captured from Adobe Media Server withAggregate messagesenabled

Symptoms: freezes can occur, publishing can fail by timeout when capturing RTMP stream from Adobe Media Server

Solution: disable Aggregate messages in AMS configuration

- вall EnableAggMsgs in Server.xml
- Client/AggregateMessages and Queue/AggregateMessages in conf/_defaultRoot_/_defaultVHost/Application.xml
- AggregateMessages in Vhost.xml