Using RTMP encoder

To run online broadcasts you can use special hardware or software video capturing devices (Live Encoder). Such devices or programs capture a video stream and send it to the server via the RTMP protocol.
Web Call Server 5.1 can receive an RTMP video stream from such a device or software (Wirecast, ffmpeg, OBS Studio, FMLE etc.) encoded to H.264 + AAC or Sorenson Spark + Speexand broadcast this video stream to browsers and mobile devices.

Overview

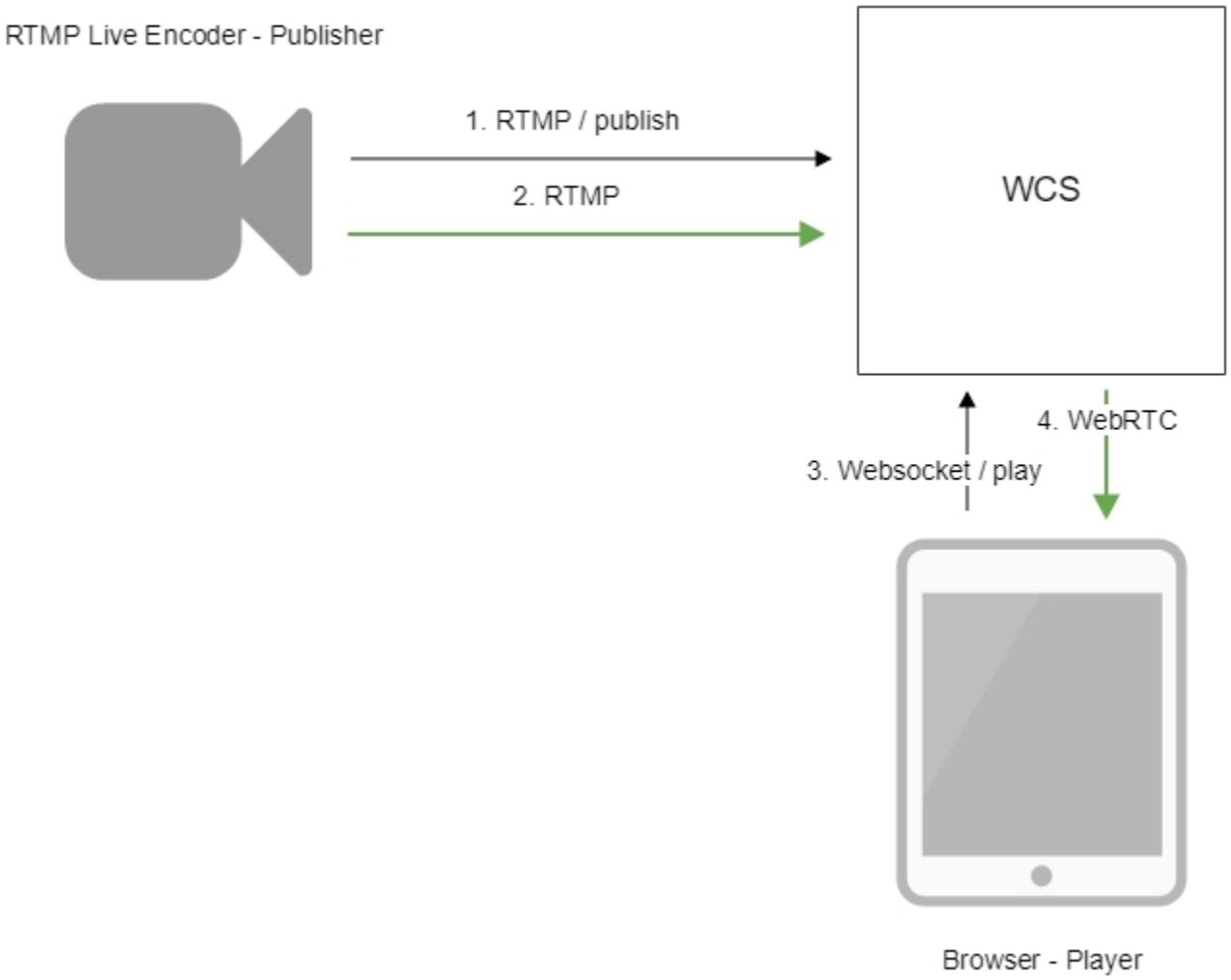
Technical specifications

- Receiving incoming audio- and video streams via the RTMP protocol
- Broadcasting of the received video stream to browsers and platforms: any among ones supported by WCS
- Uses video stream playback technologies: any among ones supported by WCS

Codec support

- Video H.264 + audio AAC
- Video Sorenson Spark + audio Speex 16 kHz

Operation flowchart



1. Live Encoder establishes a connection to the server via the RTMP protocol and sends the publish command.

2. Live Encoder sends the RTMP stream to the server.

3. The browser establishes a connection via Websocket and sends the play command.

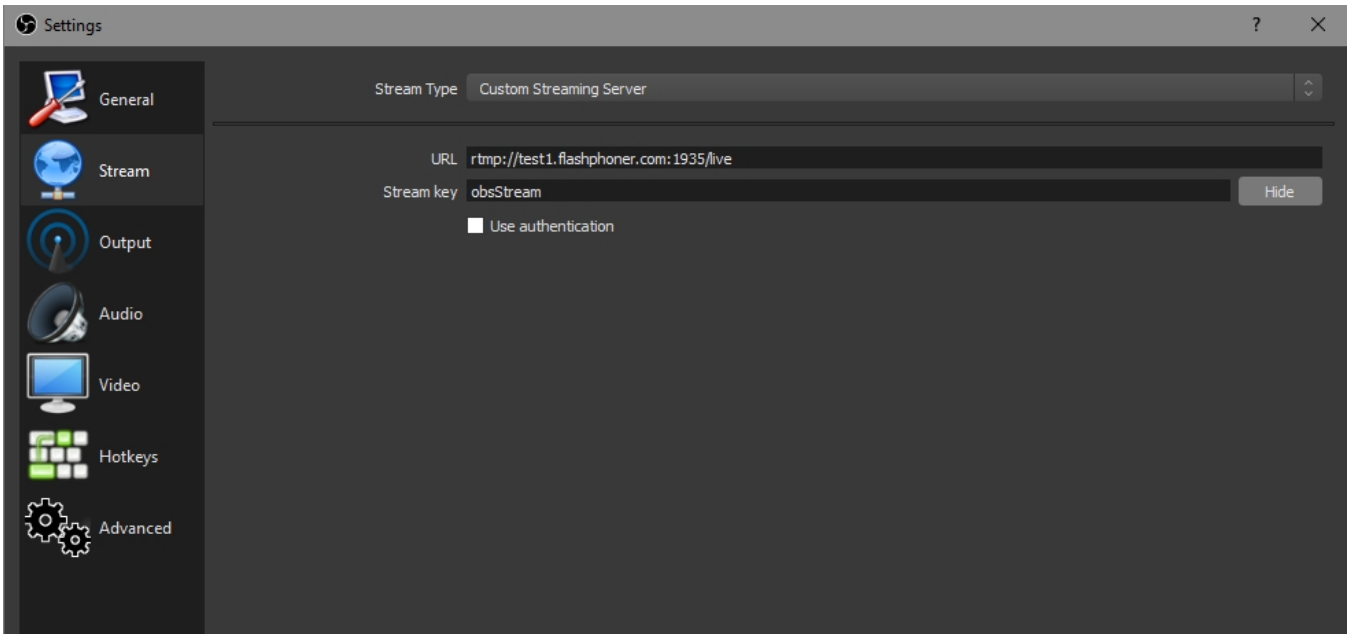
4. The browser receives the WebRTC stream and plays that stream on the page.

Quick manual on testing

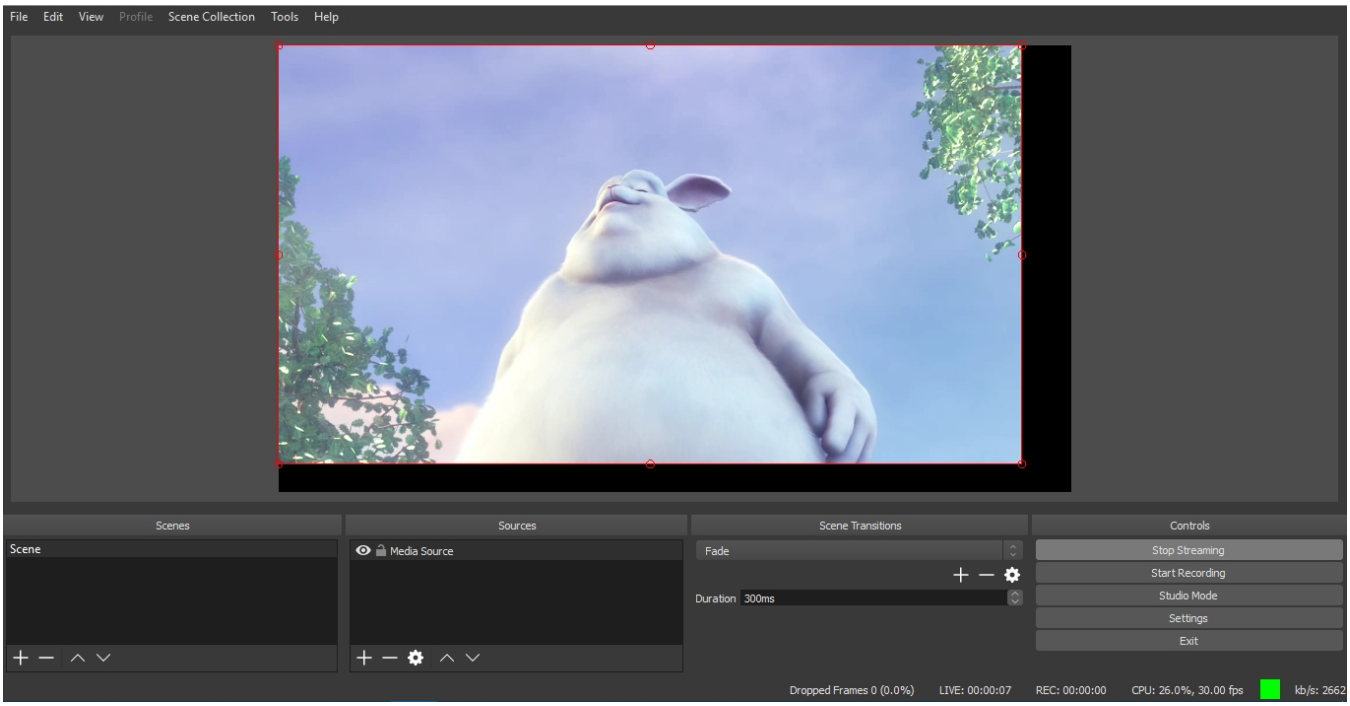
1. For test we use:

- WCS server
- OBS Studio
- Player web application in Chrome browser to stream playback

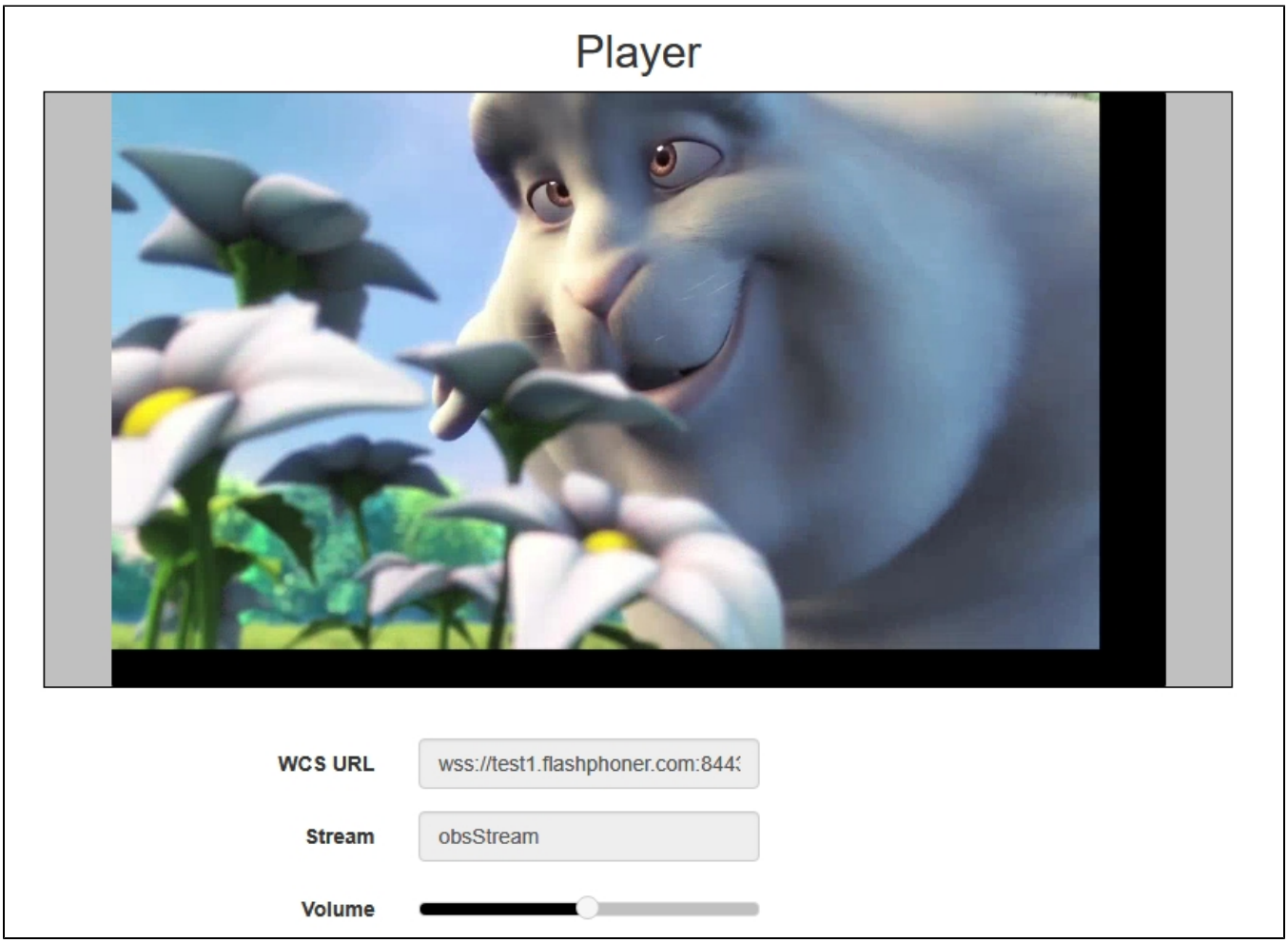
2. Set up RTMP strteaming to the server address, for example rtmp://test1.flashphoner.com:1935/live/, set the stream key obsStream:



3. Start streaming in OBS Studio:

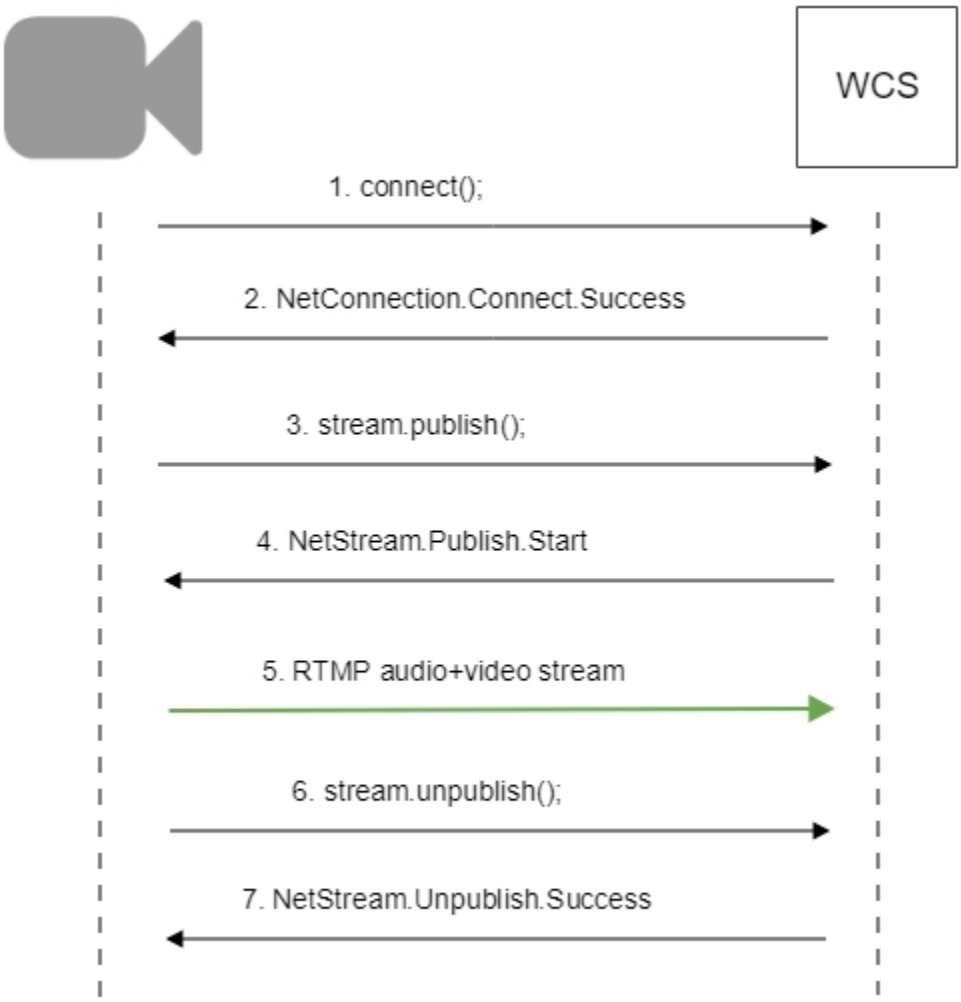


4. Open Player application. Set the stream key in 'Stream' field and press 'Start' button. The stream captured playback begins.



Call flow

Below is the call flow when an RTMP stream is broadcast from an external source (Live Encoder)to the WCS server



Parsing stream URL parameters

When RTMP stream is published or played on WCS, RTMP connection and stream parameters may be set in stream URL like this:

```
rtmp://host:1935/live?connectParam1=val1&connectParam2=val2/streamName?streamParam1=val1&streamParam2=val2
```

Where

- host is WCS server hostname;
- connectParam1, connectParam2 are RTMP connection parameters;
- streamName is stream name on server;
- streamParam1, streamParam2 are stteam parameters.

WCS server passes the parameters to backend server in REST hook in `customfield`, for example:

Connection parameters

```
URL:http://localhost:8081/apps/EchoApp/connect
OBJECT:
{
"nodeId" : "Qb3rAjf3lzoy6PEl1WZkUhRG1DsTykgj@192.168.1.1",
"appKey" : "flashStreamingApp",
"sessionId" : "/127.0.0.1:5643/192.168.1.1:1935",
"useWsTunnel" : false,
"useWsTunnelPacketization2" : false,
"useBase64BinaryEncoding" : false,
"keepAlive" : false,
"custom" : {
"connectParam1" : "val1",
"connectParam2" : "val2"
},
"login" : "rQq83sodiCPY0pJXCxGO"
}
```

Publishing parameters

```
URL:http://localhost:8081/apps/EchoApp/publishStream
OBJECT:
{
"nodeId" : "Qb3rAjf3lzoy6PEl1WZkUhRG1DsTykgj@192.168.1.1",
"appKey" : "flashStreamingApp",
"sessionId" : "/127.0.0.1:5643/192.168.1.1:1935",
"mediaSessionId" : "627990f9-8fe5-4e92-bb2a-863cc4eb43de",
"name" : "stream1",
"published" : true,
"hasVideo" : false,
"hasAudio" : true,
"status" : "NEW",
"record" : true,
"width" : 0,
"height" : 0,
"bitrate" : 0,
"minBitrate" : 0,
"maxBitrate" : 0,
"quality" : 0,
"mediaProvider" : "Flash",
"custom" : {
"streamParam1" : "val1",
"streamParam2" : "val2"
}
}
```

Playback parameters

```
URL:http://localhost:8081/apps/EchoApp/playStream
OBJECT:
{
"nodeId" : "Qb3rAjf3lzoy6PEl1WZkUhRG1DsTykgj@192.168.1.1",
"appKey" : "flashStreamingApp",
"sessionId" : "/127.0.0.1:5643/192.168.1.1:1935",
"mediaSessionId" : "stream1/127.0.0.1:5643/192.168.1.1:1935",
"name" : "stream1",
"published" : false,
"hasVideo" : true,
"hasAudio" : true,
"status" : "NEW",
"record" : false,
"width" : 0,
"height" : 0,
"bitrate" : 0,
"minBitrate" : 0,
"maxBitrate" : 0,
"quality" : 0,
"mediaProvider" : "Flash",
"custom" : {
"streamParam1" : "val1",
"streamParam2" : "val2"
}
}
```

This feature can be used for example to authenticate client on backend server while publishing or playing RTMP-stream on WCS server.

Setting a server application while RTMP stream publishing

While publishing RTMP stream to WCS server, a server application can be set that will be used to backend server interaction. It can be done with parameter in stream URL:

```
rtmp://host:1935/live?appKey=key1/streamName
```

Where

- host is WCS server;
- key1 is application key on WCS server;
- streamName is stream name to publish

By default, if application key parameter is not set, the standard application `flashStreamingApp` will be used.

Besides, an application can be explicitly specified as stream URL part. To do this, the following parameter in flashphoner.properties file should be set

```
rtmp_appkey_source=app
```

Then application key must be set in stream URL as

```
rtmp://host:1935/key1/streamName
```

In this case, `live` is also an application name, therefore when stream is published with URL

```
rtmp://host:1935/live/streamName
```

`live` application must be defined on WCS server.

Sorenson Spark + Speex 16 kHz stream publishing

WCS server can capture RTMP stream encoded with Sorenson Spark + Speex 16kHz to FLV container. This stream can be published, for example, using ffmpeg as follows:

```
ffmpeg -re -i BigBuckBunny.flv -preset ultrafast -ar 16000 -ac 1 -acodec speex -vcodec flv -strict -2 -f flv
rtmp://test1.flashphoner.com:1935/live/test
```

Known limits

1. To handle such stream including stream recording, the stream will be transcoded to H.264 + AAC.

2. Payload types 127 for video and 97 for audio should be set in SDP when publishing such stream, for example

```
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 127
a=rtpmap:127 FLV/90000
a=sendonly
m=audio 0 RTP/AVP 97 8 0
a=rtpmap:97 SPEEX/16000
a=rtpmap:8 PCMA/8000
a=rtpmap:0 PCMU/8000
a=sendonly
```

Using RTMP connection timeouts

In some cases, if RTMP encoder does not support Keep Alive packets sending, or Keep Alives are disabled due to another reason with the following parameter

```
keep_alive.algorithm=NONE
```

it is necessary to control RTMP connection and close it when no data was transmitted for a long time. To do this, use the following parameters.

Read timeout

Read timeout is set with the following settings inflashphoner.propertiesfile:

```
rtmp.server_read_socket_timeout=120
```

In this case RTMP connection will be closed if no data was received in last 120 seconds.

Write timeout

Write timeout is set with the following setting

```
rtmp.server_write_socket_timeout=120
```

In this case RTMP connection will be closed if no data was sent in last 120 seconds.

Read and write timeout

Read and write timeout is set with the following setting

```
rtmp.server_socket_timeout=120
```

In this case RTMP connection will be closed if no data was received and sent in last 120 seconds.

RTMP stream picture rotation

When publishing RTMP stream to WCS, stream picture can be rotated using RTMP metadata. It may be useful for picture handling from mobile publishers.

To turn a picture to a certain angle RTMP metadata containing ‘orientation’ field should be sent. The following orientations are supported:

Orientation value	Rotation angle, degrees
0	0
1	90
2	180
3	270

A picture is rotated clockwise.

Set up

To enable stream rotation, set the following parameter inflashphoner.propertiesfile:

```
video_filter_enable_rotate=true
```

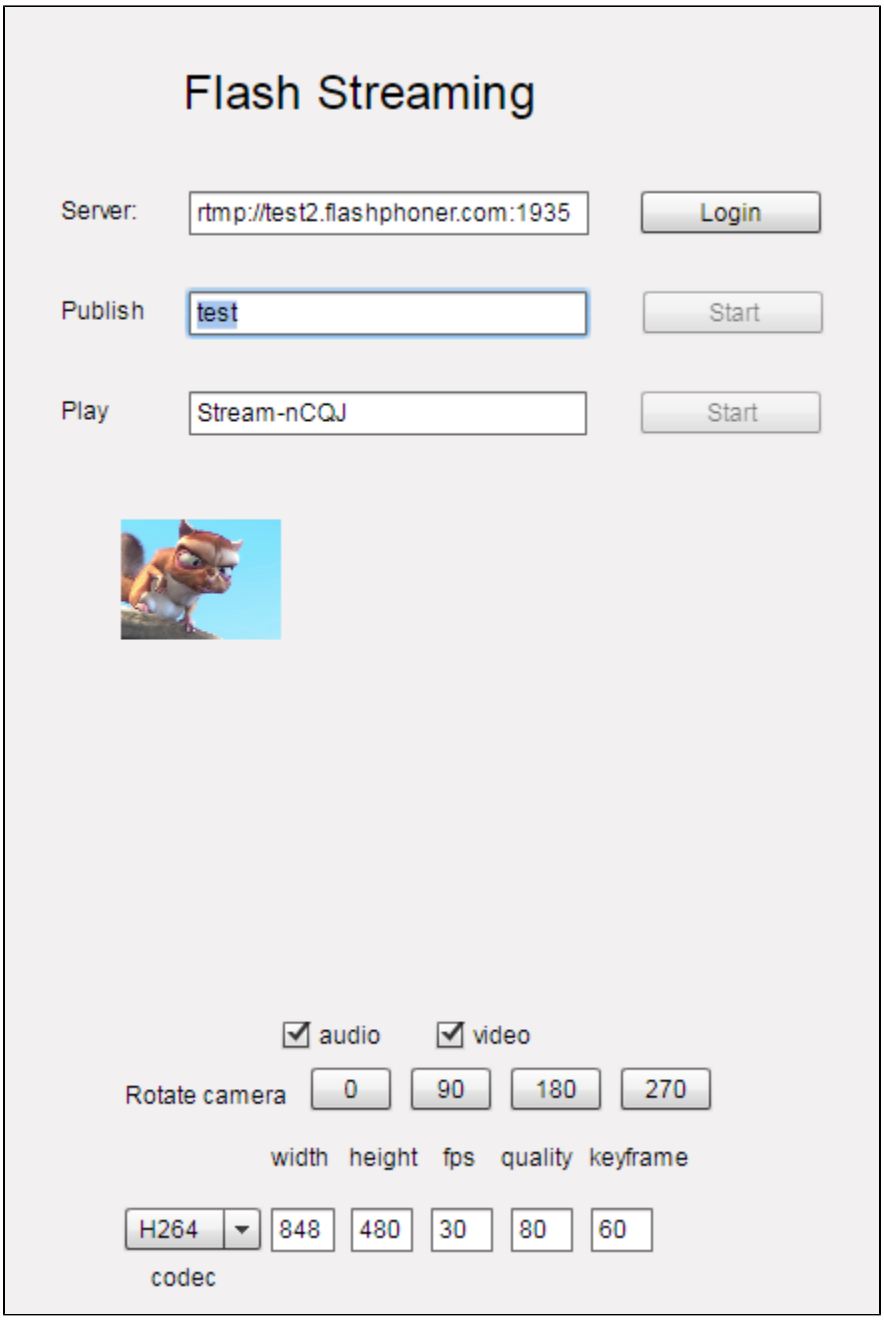
Note that stream rotation works for transcoded streams only.

Testing

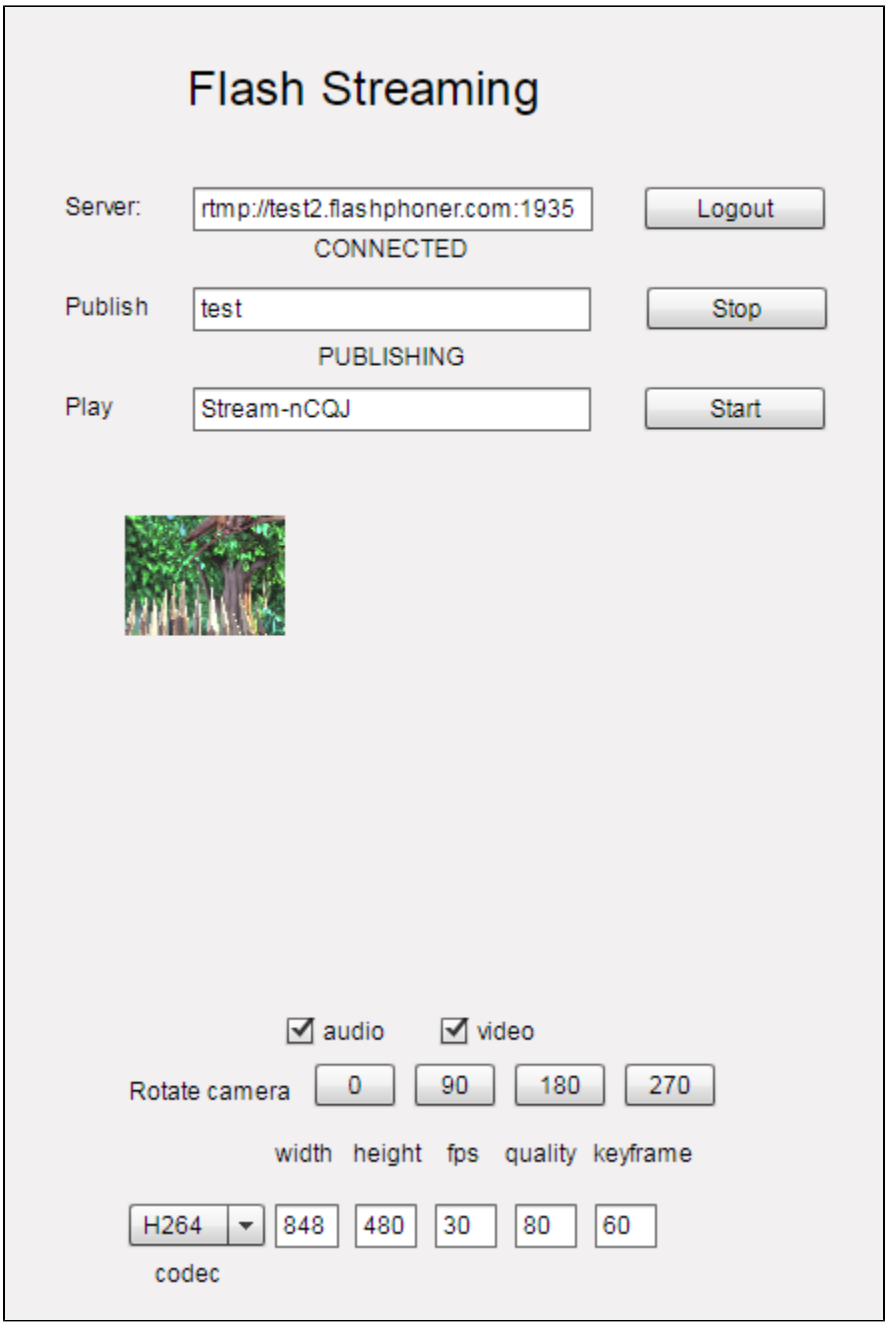
1. For test we use:

- WCS server with picture rotation enabled
- Flash Streamingapplication to publish and rotate the stream
- Playerapplication to play the stream

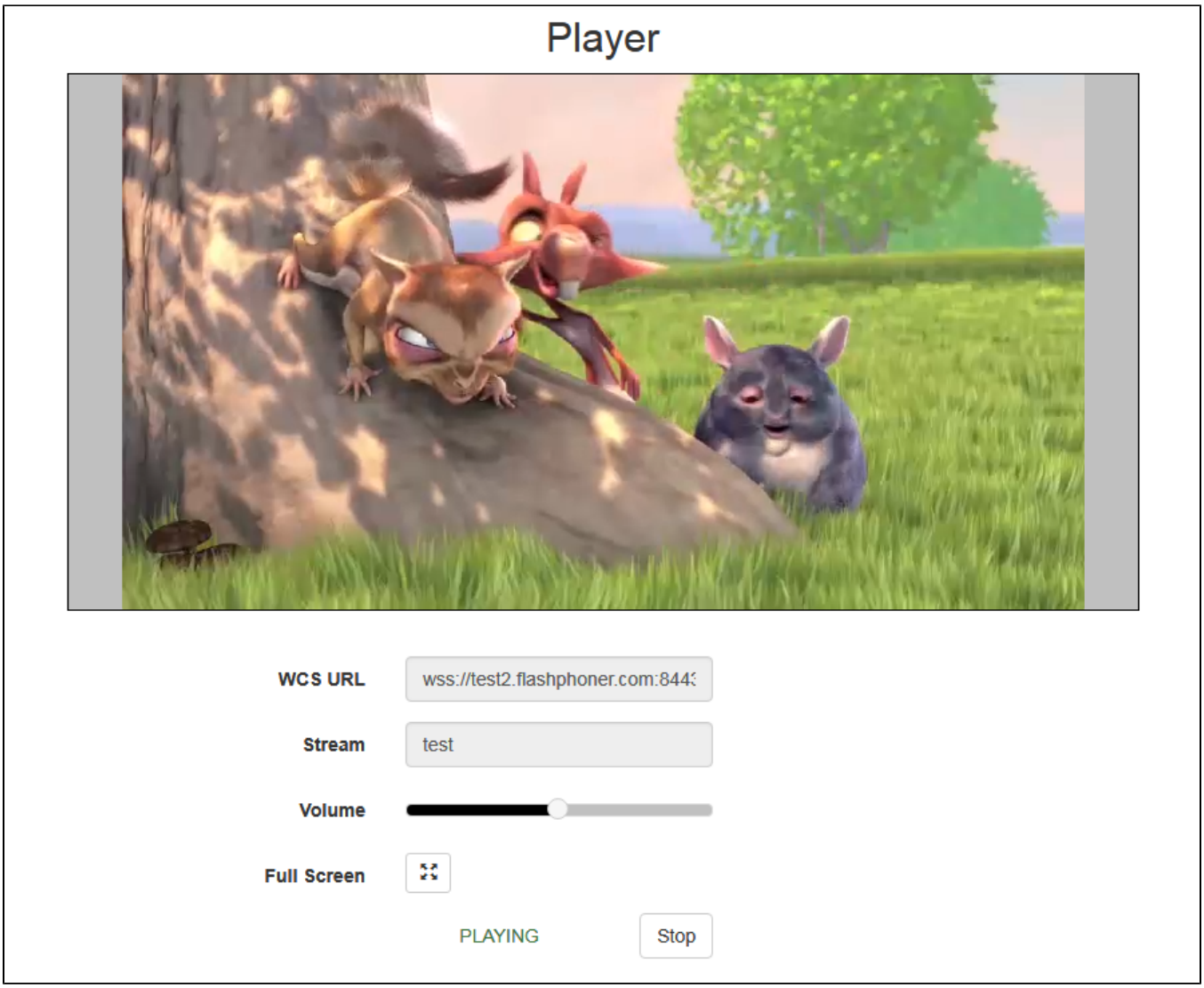
2. Open Flash Streaming application. Set stream name test and desired stream publishing parameters:



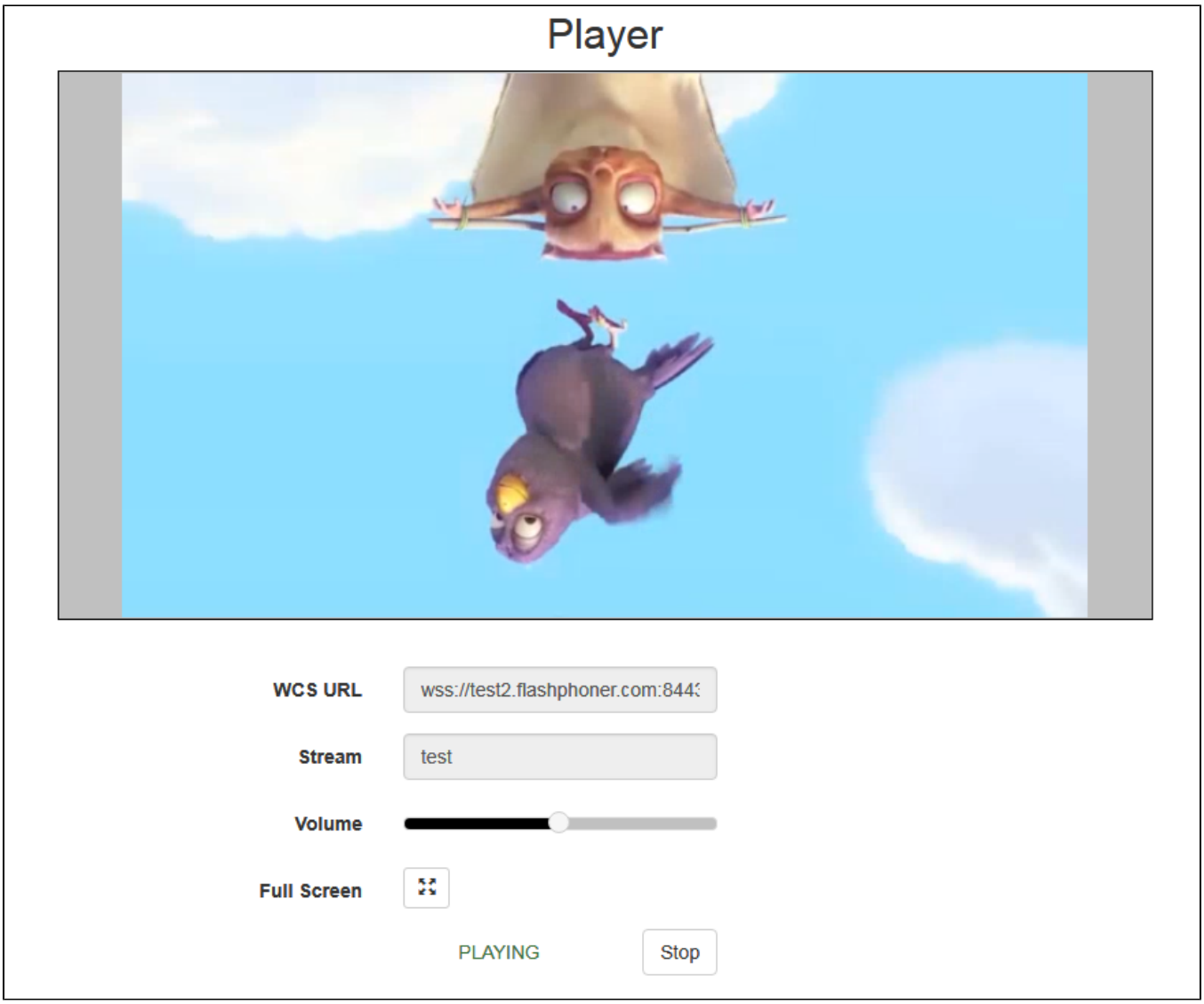
3. Click Login then Start. Stream publishing begins



4. Open Player application in anoter tab or browser, play the stream named test



5. In Flash Streaming application click 180 button in Rotate camera section. The stream rotated to 180 degrees clockwise will be displayed in Player application



To developer

Stream orientation metadata sending is implemented in Flash Streaming application as follows:

code

```
                    private function rotate(degree:Number):void {
                        var metaDataObj:Object = new Object();
                        switch(degree) {
                case 0:
                    Logger.info("rotate camera to 0");
                    metaDataObj.orientation = 0;
                    break;
                case 90:
                    Logger.info("rotate camera to 90");
                    metaDataObj.orientation = 1;
                    break;
                case 180:
                    Logger.info("rotate camera to 180");
                    metaDataObj.orientation = 2;
                    break;
                case 270:
                    Logger.info("rotate camera to 270");
                    metaDataObj.orientation = 3;
                    break;
                default:
                    metaDataObj.orientation = 0;
                    break;
                        }
                        sendMetaData(metaDataObj);
                    }

                    private function sendMetaData(data:Object):void{
                        if (publishStream != null) {
                            publishStream.send("@setDataFrame", "onMetaData", data);
                        }
                    }
```

Note that orientation value should be sent but not angle itself.

How to rotate stream published from ffmpeg

ffmpeg RTMP encoder allows to send orientation metadata to WCS server using command line switches:

```
ffmpeg -i input.mp4 -metadata:s:v rotate=90 -vcodec copy -acodec copy -strict -2 -f flv rtmp://test1.
flashphoner.com:1935/live/stream_ffmpeg
```

Note that ffmpeg sends orientattion value but notangle itself.

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).

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.

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

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

5. Some RTMP-encoders does not support KeepAlive.

Symptoms: disconnection occurs often while stream publishing with RTMP-encoder.

Solution: switch KeepAlive off for RTMP on the server using the following parameter inflashphoner.propertiesfile

```
keep_alive.enabled=websocket,rtmfp
```

7. When stream published with RTMP encoder is played as HLS, freezes may occur if GOP is not multiple of FPS of file published

Symptoms: freezes occur when RTMP stream is played as HLS

Solution: in RTMP encoder settings, assign GOP to value equal or multiple of FPS of file published. For example, when publishing file with FPS 25 set GOP to 50.