# RTP stream publishing via RTSP



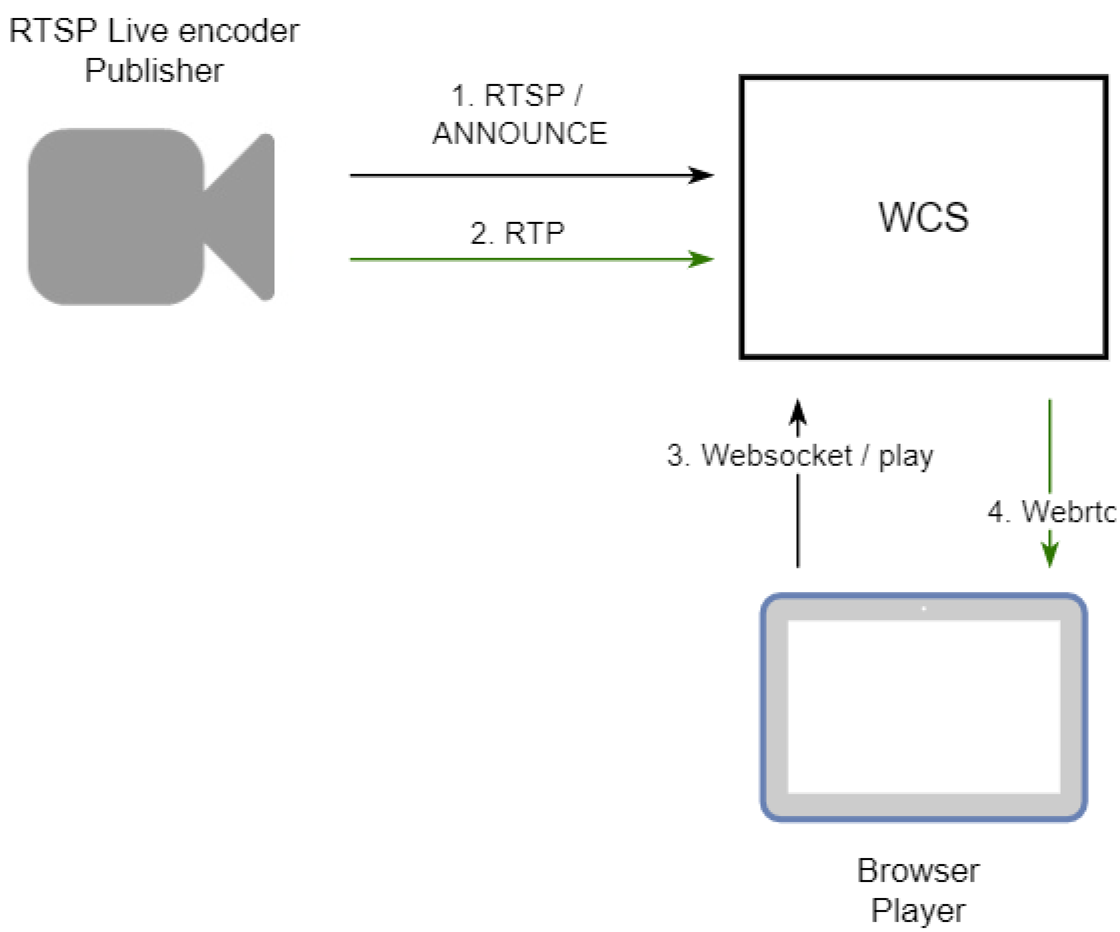
## Overview

Since build5.2.902 it is possible to publish RTP stream via RTSP directly to WCS. Both TCP interleaved and UDP transports are supported. This can be useful to publish H264+Opus streams from hardware or software encoder supporting RTSP. Since build 5.2.1584, H265 codec is also supported for publishing.

## Codecs supported

- H264
- H265 (since build 5.2.1584)
- VP8
- AAC
- Opus

## Operation flowchart




1. Live Encoder connects to the server via RTSP and sends ANNOUNCE command.
2. Live Encoder sends RTP 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:

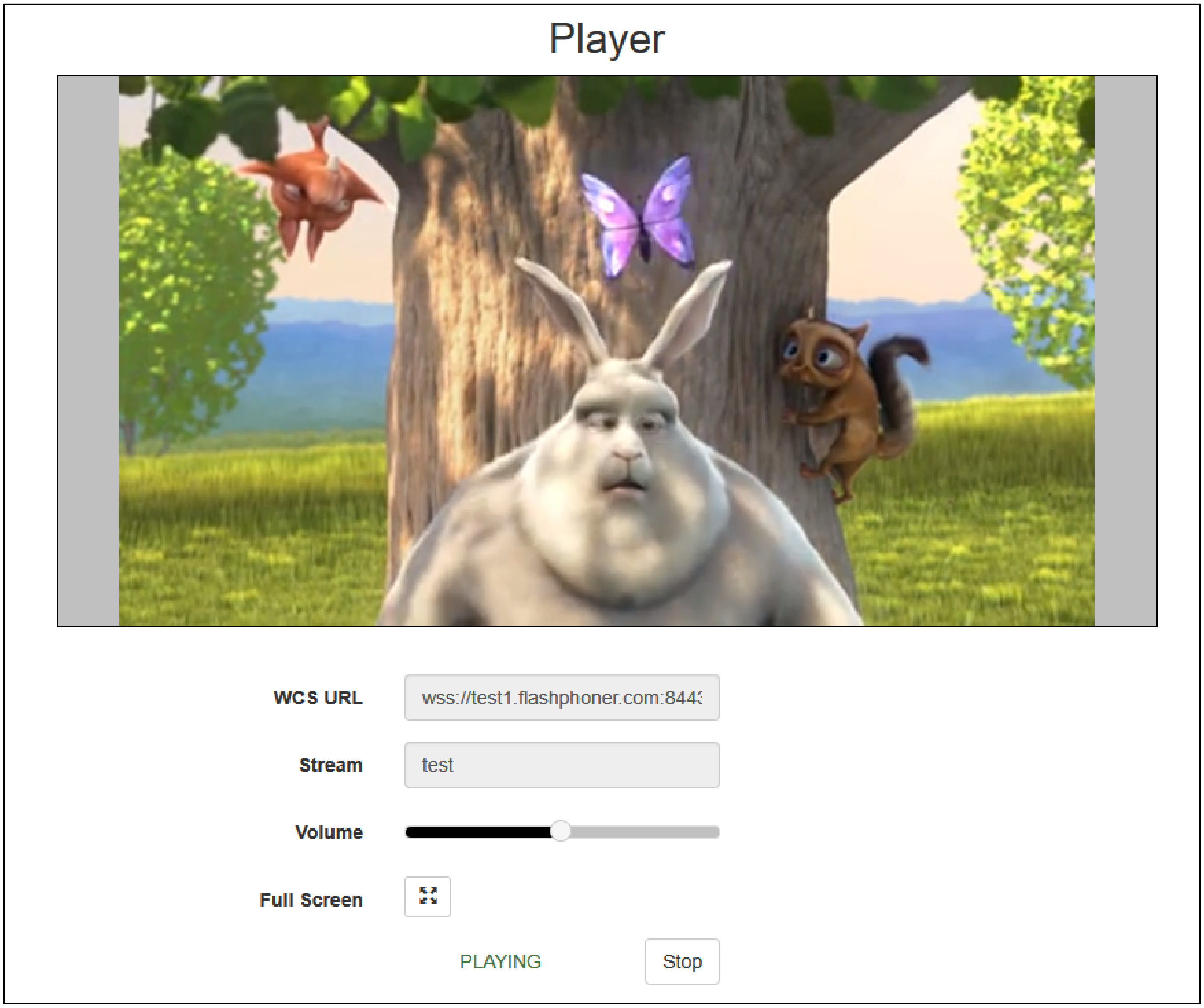
- WCS server
- ffmpeg
- Playerweb application in Chromebrowser to play the stream

2. Start RTSP H264+Opus stream publishing using ffmpeg

```
ffmpeg -stream_loop -1 -re -i bunny360p.mp4 -c:a libopus -ac 2 -ar 48000 -c:v copy -b:a 96K -b:v 500K -f rtsp -
rtsp_transport tcp rtsp://test1.flashphoner.com:554/test
```

```
Input #0, mov,mp4,m4a,3gp,3g2,mj2, from 'bunny360p.mp4':
  Metadata:
    major_brand     : isom
    minor_version   : 512
    compatible_brands: isomiso2avc1mp41
    encoder         : Lavf58.12.100
  Duration: 00:09:56.46, start: 0.000000, bitrate: 631 kb/s
    Stream #0:0(eng): Video: h264 (High) (avc1 / 0x31637661), yuv420p, 640x360, 499 kb/s, 24 fps, 24 tbr, 12288 tbn, 48 tbc (default)
    Metadata:
      handler_name    : VideoHandler
    Stream #0:1(eng): Audio: aac (LC) (mp4a / 0x6134706D), 48000 Hz, stereo, fltp, 128 kb/s (default)
    Metadata:
      handler_name    : SoundHandler
Stream mapping:
  Stream #0:0 -> #0:0 (copy)
  Stream #0:1 -> #0:1 (aac (native) -> opus (libopus))
Press [q] to stop, [?] for help
Output #0, rtsp, to 'rtsp://test1.flashphoner.com:554/test':
  Metadata:
    major_brand     : isom
    minor_version   : 512
    compatible_brands: isomiso2avc1mp41
    encoder         : Lavf58.45.100
    Stream #0:0(eng): Video: h264 (High) (avc1 / 0x31637661), yuv420p, 640x360, q=2-31, 500 kb/s, 24 fps, 24 tbr, 90k tbn, 24 tbc (default)
    Metadata:
      handler_name    : VideoHandler
    Stream #0:1(eng): Audio: opus (libopus), 48000 Hz, stereo, flt, 96 kb/s (default)
    Metadata:
      handler_name    : SoundHandler
      encoder         : Lavc58.91.100 libopus
frame=  170 fps= 24 q=-1.0 size=N/A time=00:00:07.05 bitrate=N/A speed=0.998x
```

3.Open Player application.Set stream name to "Stream" field and click "Start".The stream captured playback begins.




## H265 publishing

Since build 5.2.1584, RTP stream in H265 codec may be published via RTSP

```
ffmpeg -re -i source.mp4 -c:v libx265 -c:a aac -b:a 160k -bsf:v hevc_mp4toannexb -keyint_min 60 -profile:v main
-preset veryfast -x265-params crf=23:bframes=0 -f rtsp -rtsp_transport tcp rtsp://test1.flashphoner.com:554/test
```

To do this, H265 should be added to codecs supported 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
```

H265 stream will be transcoded to H264 or VP8 to play from server!

## RTSP port setup

TCP port 554 is used by default to publish RTP stream via RTSP

```
rtsp.port=554
```

However, the port is privileged and is available to listen to applications launched by `root` only. If WCS is starting from`flashphoner` user (the default launch mode), the port should be changed, for example

```
rtsp.port=5554
```