RTP stream publishing via RTSP

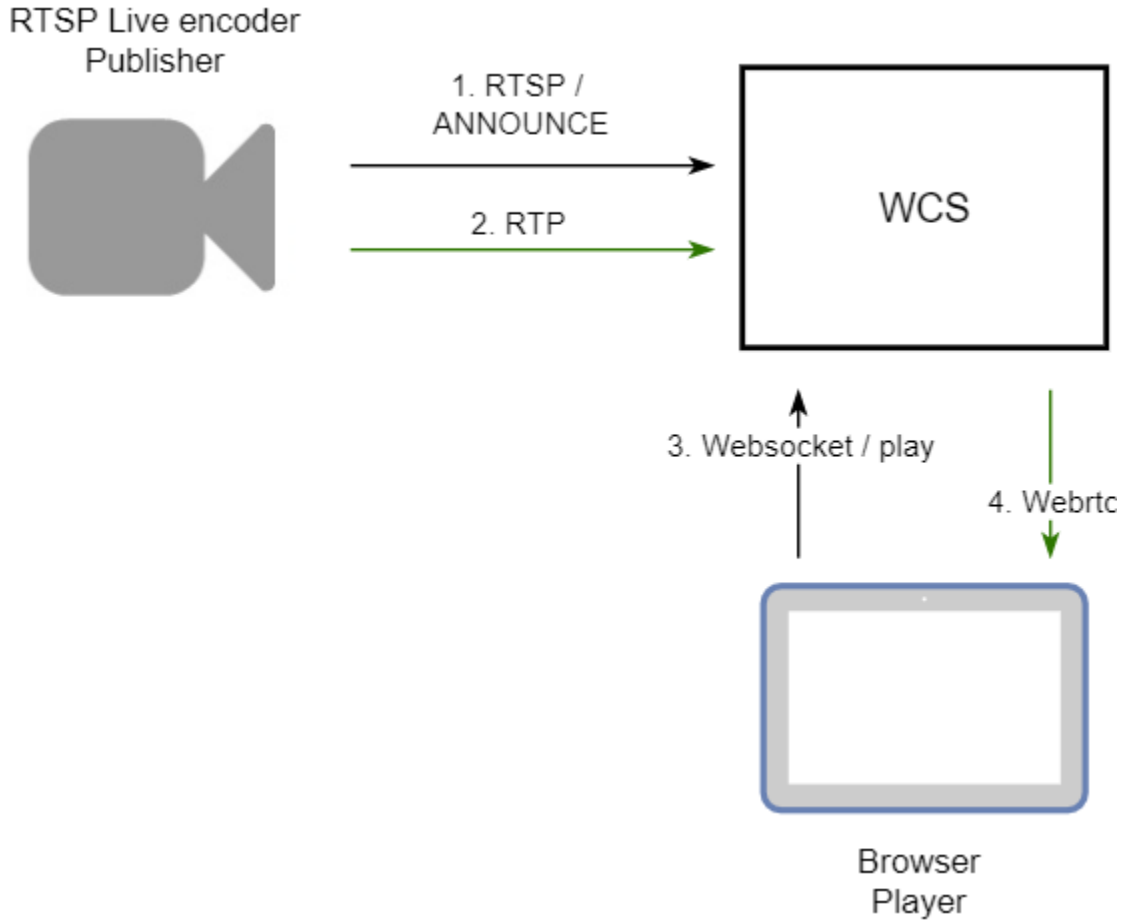
Overview

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:

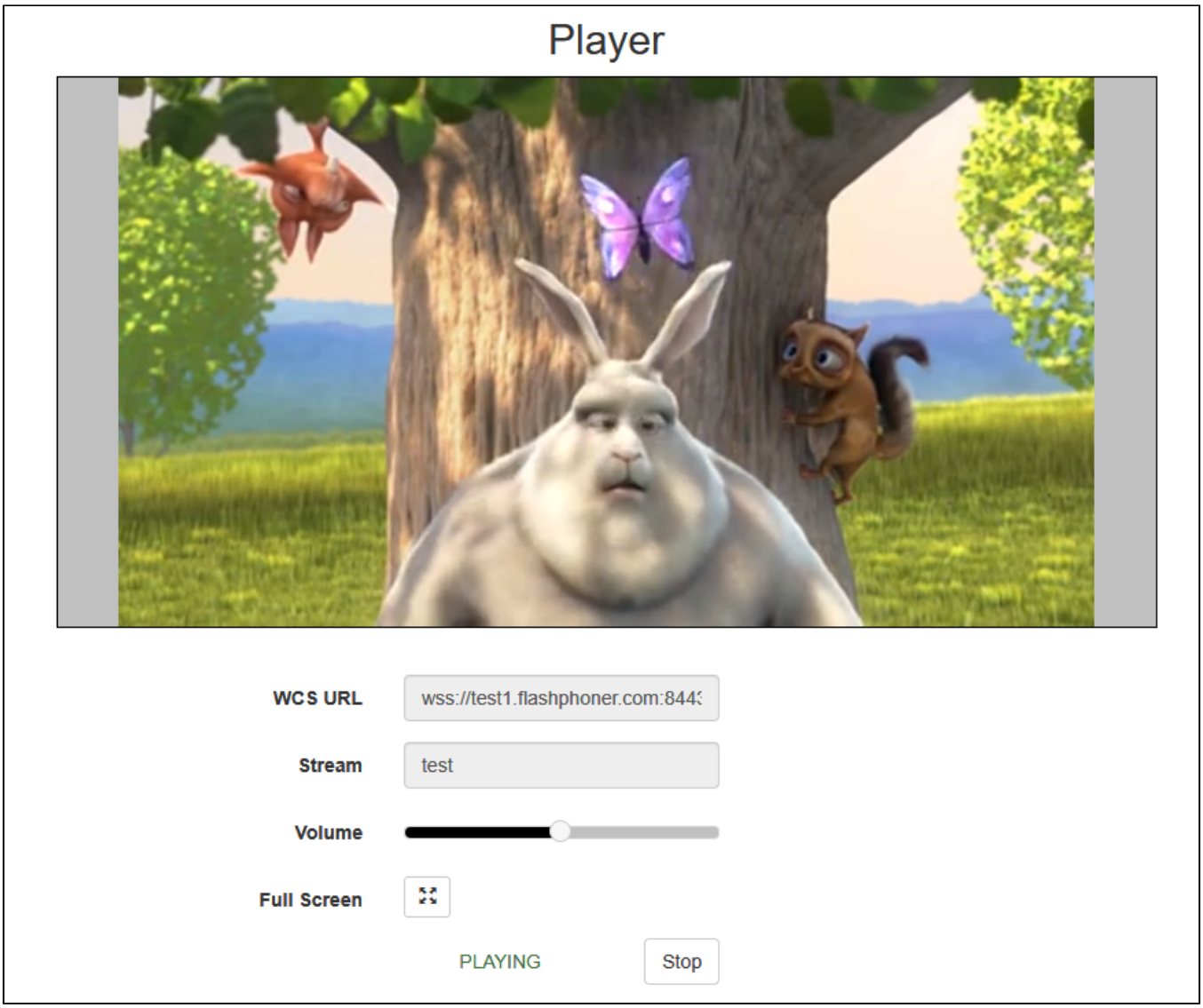
- WCS server
- ffmpeg
- Playerweb application in Chromebrowser to play the stream

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
```