

# RTP stream publishing via RTSP

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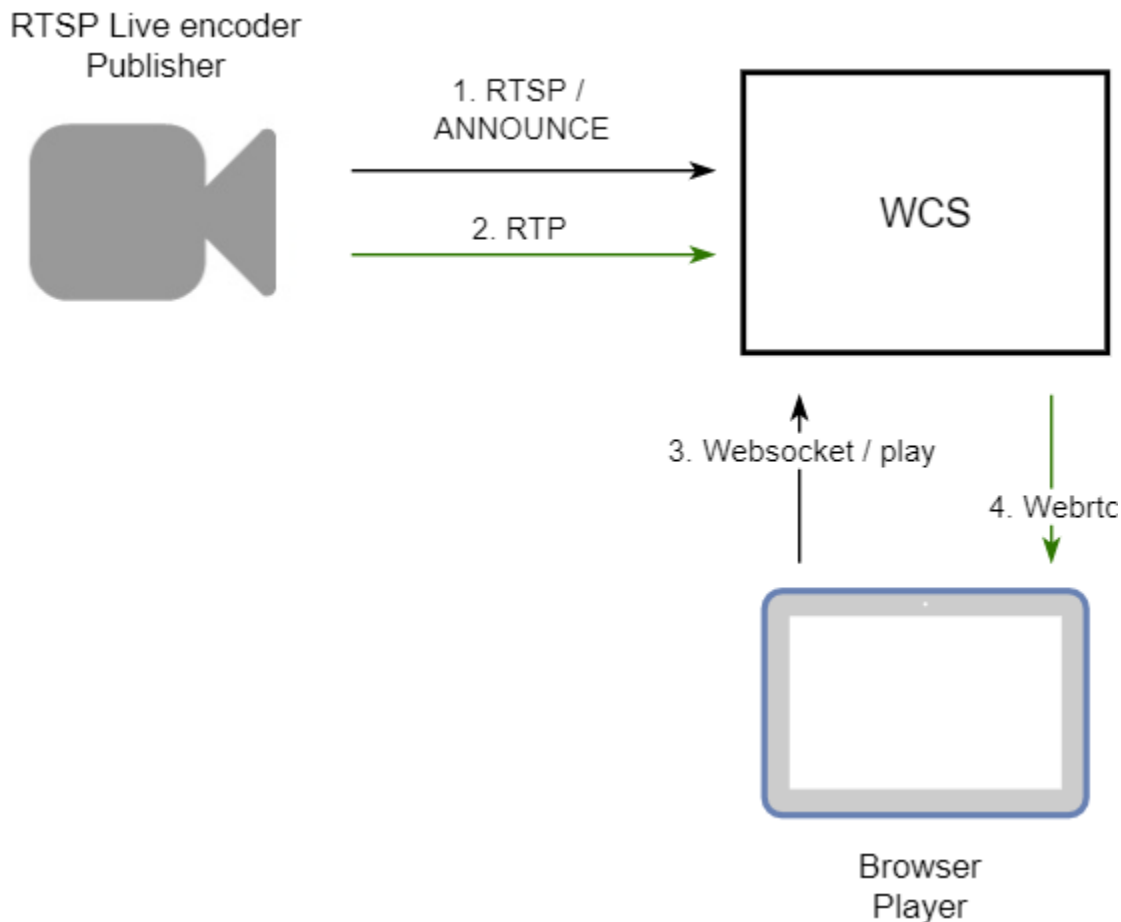
## Overview

Since build [5.2.902](#) 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.

## Codecs supported

- H264
- 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
- [Player](#) web 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.

# Player



WCS URL

wss://test1.flashphoner.com:8446

Stream

test

Volume



Full Screen



PLAYING

Stop