MPEG-TS RTP stream publishing

Overview

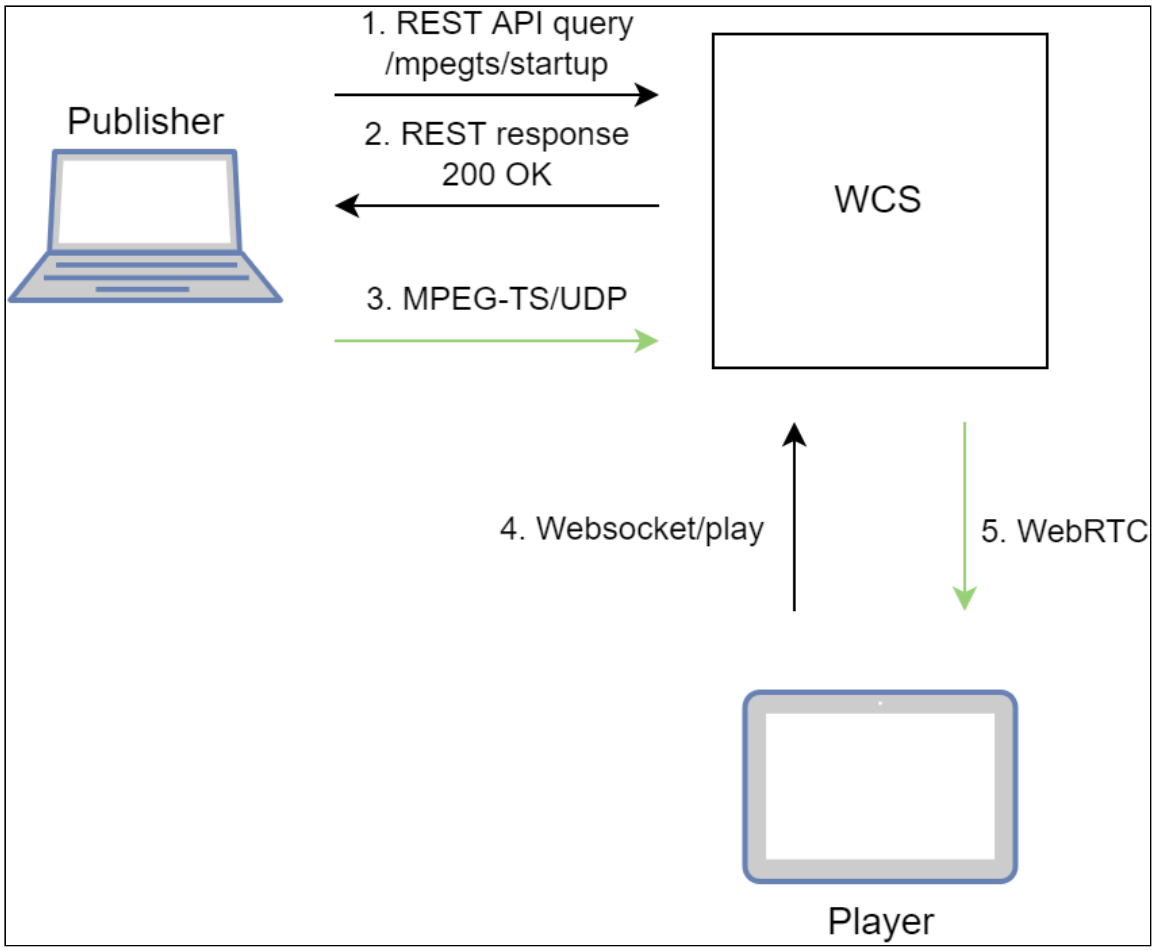
Since WCS build 5.2.1193 it is possible to publish MPEG-TS RTP stream via UDP to WCS, and since build 5.2.1253 MPEG-TS stream may be published via SRT. The feature can be used to publish H264+AAC stream from software or hardware encoder supporting MPEG-TS. Since build 5.2.1577 H265+AAC stream publishing is also allowed.

SRT protocol is more reliable than UDP, so it is recommended to use SRT for MPEG-TS publishing if possible.

Codecs supported

- Video: H264, H265 (since build 5.2.1577)
- Audio: AAC

Operation flowchart



1. Publisher sends REST API query `/mpegts/startup`
2. Publisher receives `200 OK` with URI to publish
3. Stream is publishing to WCS using URI
4. Browser establishes Websocket connestion and sends `playStream` command.
5. Browser receives WebRTC stream and plays it on web page.

Testing

1. For test we use:
2. WCS server
3. ffmpeg to publish MPEG-TS stream
4. Player web application in Chrome browser to play the stream
5. Send `/mpegts/startup` query with stream name `test`
   SRT:

```
curl -H "Content-Type: application/json" -X POST
http://test1.flashphoner.com:8081/rest-api/mpegts/startup -d
'{"localStreamName":"test","transport":"srt"}'
```

UDP:

```
curl -H "Content-Type: application/json" -X POST
http://test1.flashphoner.com:8081/rest-api/mpegts/startup -d
'{"localStreamName":"test","transport":"udp"}'
```

Where `test1.flashphoner.com` - WCS server address

6. Receive `200 OK` response
   SRT:

```
{
  "localMediaSessionId": "32ec1a8e-7df4-4484-9a95-e7eddc45c508",
  "localStreamName": "test",
  "uri": "srt://test1.flashphoner.com:31014",
  "status": "CONNECTED",
  "hasAudio": false,
  "hasVideo": false,
  "record": false,
  "transport": "SRT",
  "cdn": false,
  "timeout": 90000,
  "maxTimestampDiff": 1,
  "allowedList": []
}
```

   UDP:

```
{
  "localMediaSessionId": "32ec1a8e-7df4-4484-9a95-e7eddc45c508",
  "localStreamName": "test",
  "uri": "udp://test1.flashphoner.com:31014",
  "status": "CONNECTED",
  "hasAudio": false,
  "hasVideo": false,
  "record": false,
  "transport": "UDP",
  "cdn": false,
  "timeout": 90000,
  "maxTimestampDiff": 1,
  "allowedList": []
}
```

7. Publish MPEG-TS stream using URI from the response
   SRT:

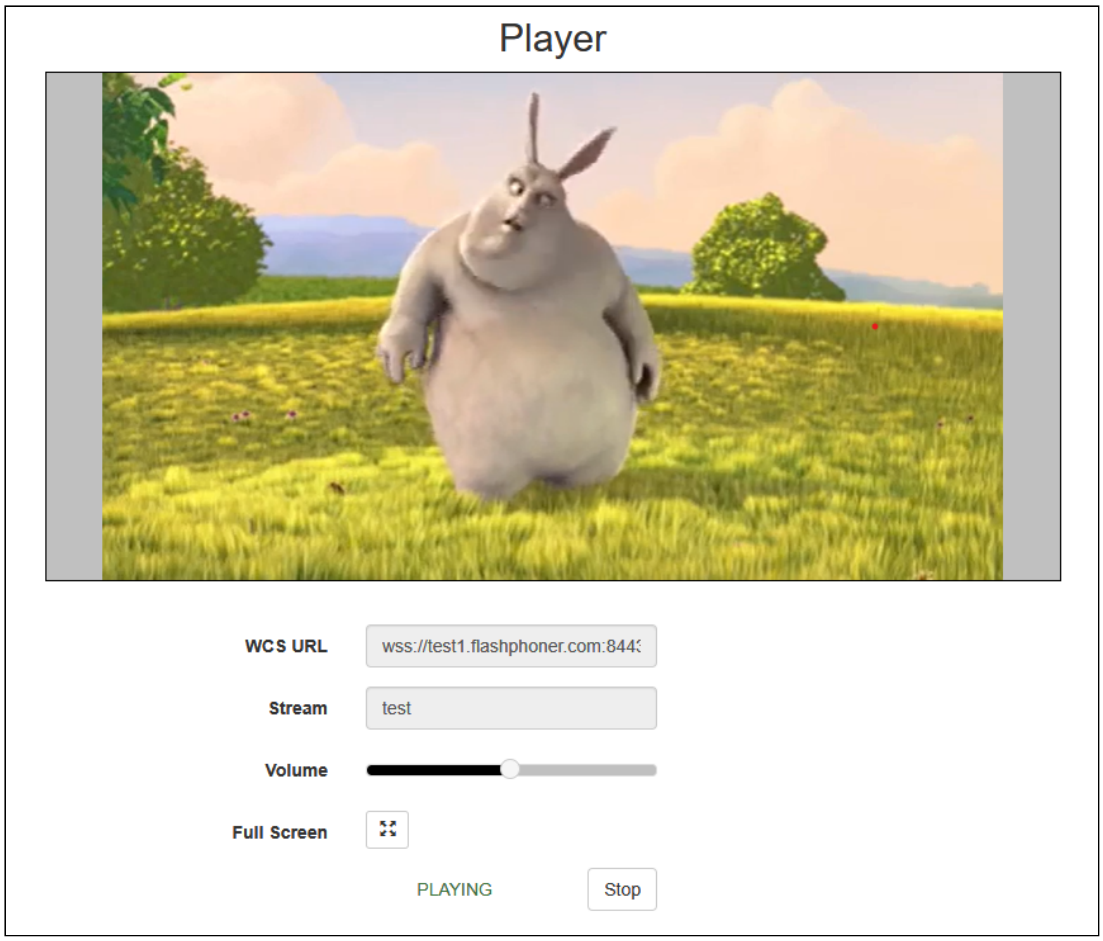
```
ffmpeg -re -i bunny360p.mp4 -c:v libx264 -c:a aac -b:a 160k -bsf:v
h264_mp4toannexb -keyint_min 60 -profile:v baseline -preset veryfast -f
mpegts "srt://test1.flashphoner.com:31014"
```

   UDP:

```
ffmpeg -re -i bunny360p.mp4 -c:v libx264 -c:a aac -b:a 160k -bsf:v
h264_mp4toannexb -keyint_min 60 -profile:v baseline -preset veryfast -f
mpegts "udp://test1.flashphoner.com:31014?pkt_size=1316"
```

```
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[0x1](eng): Video: h264 (High) (avc1 / 0x31637661), yuv420p(progressive), 640x360, 499 kb/s, 24 fps, 24 tbr, 12288 tbn (default)
    Metadata:
      handler_name    : VideoHandler
      vendor_id       : [0][0][0][0]
  Stream #0:1[0x2](eng): Audio: aac (LC) (mp4a / 0x6134706D), 48000 Hz, stereo, fltp, 128 kb/s (default)
    Metadata:
      handler_name    : SoundHandler
      vendor_id       : [0][0][0][0]
Stream mapping:
  Stream #0:0 -> #0:0 (h264 (native) -> h264 (libx264))
  Stream #0:1 -> #0:1 (aac (native) -> aac (native))
Press [q] to stop, [?] for help
[libx264 @ 00000249853ac540] using cpu capabilities: MMX2 SSE2Fast SSSE3 SSE4.2 AVX FMA3 BMI2 AVX2
[libx264 @ 00000249853ac540] profile Constrained Baseline, level 3.0, 4:2:0, 8-bit
Output #0, mpegts, to 'udp://95.191.130.39:31006?pkt_size=1316':
  Metadata:
    major_brand     : isom
    minor_version   : 512
    compatible_brands: isomiso2avc1mp41
    encoder         : Lavf59.16.100
  Stream #0:0(eng): Video: h264, yuv420p(progressive), 640x360, q=2-31, 24 fps, 90k tbn (default)
    Metadata:
      handler_name    : VideoHandler
      vendor_id       : [0][0][0][0]
      encoder         : Lavc59.18.100 libx264
    Side data:
      cpb: bitrate max/min/avg: 0/0/0 buffer size: 0 vbv_delay: N/A
  Stream #0:1(eng): Audio: aac (LC), 48000 Hz, stereo, fltp, 160 kb/s (default)
    Metadata:
      handler_name    : SoundHandler
      vendor_id       : [0][0][0][0]
      encoder         : Lavc59.18.100 aac
frame= 5553 fps= 24 q=28.0 size=   18251kB time=00:03:51.65 bitrate= 645.4kbits/s speed=   1x
```

8. Open Player web application. Set the stream name `test` to `Stream name` field and click `Start` button. Stream playback will start



Configuration

Stop stream publishing if there are no media data

By default, MPEG-TS stream publishing will stop at server side if server does not receive any media data from publisher in 90 seconds. The timeout is set in milliseconds by the following papameter

```
mpegts_stream_timeout=90000
```

Close subscribers sessions if publisher stops sending media data

If publisher stopped sending media data by some reason, then started again (for example, ffmpeg was restarted), the stream frame timestamps sequence is corrupting. T.he stream cannot be played via WebRTC correctky in this case. As workaround, all the subscribers sessions will be closed if stream timestamps sequence corruption occurs, then all the subscribers should connect to the stream again. A maximum timestamp difference is set in seconds by the following parameter

```
mpegts_max_pts_diff=1
```

REST API

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

- HTTP: `http://test.flashphoner.com:8081/rest-api/mpegts/startup`
- HTTPS: `https://test.flashphoner.com:8444/rest-api/mpegts/startup`

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
- `/mpegts/startup` - REST mathod to use

REST methods and responses

/mpegts/startup

Start MPEG-TS publishing

REQUEST EXAMPLE

```
POST /rest-api/mpegts/startup HTTP/1.1
Host: localhost:8081
Content-Type: application/json

{
    "localStreamName":"test",
    "transport":"srt",
    "hasAudio": true,
    "hasVideo": true
}
```

RESPONSE EXAMPLE

```
HTTP/1.1 200 OK
Access-Control-Allow-Origin: *
Content-Type: application/json

{
    "localMediaSessionId": "32ec1a8e-7df4-4484-9a95-e7eddc45c508",
    "localStreamName": "test",
    "uri": "srt://192.168.1.39:31014",
    "status": "CONNECTED",
    "hasAudio": false,
```

```
    "hasVideo": false,
    "record": false,
    "transport": "SRT",
    "cdn": false,
    "timeout": 90000,
    "maxTimestampDiff": 1,
    "allowedList": []
}
```

RETURN CODES

Code	Reason
200	OK
409	Conflict
500	Internal error

/mpegts/find

Find the MPEG-TS stream by criteria

REQUEST EXAMPLE

```
POST /rest-api/mpegts/find HTTP/1.1
Host: localhost:8081
Content-Type: application/json

{
    "localStreamName":"test",
    "uri": "srt://192.168.1.39:31014"
}
```

RESPONSE EXAMPLE

```
HTTP/1.1 200 OK
Access-Control-Allow-Origin: *
Content-Type: application/json

[
    {
        "localMediaSessionId": "32ec1a8e-7df4-4484-9a95-e7eddc45c508",
        "localStreamName": "test",
        "uri": "srt://192.168.1.39:31014",
        "status": "PROCESSED_LOCAL",
        "hasAudio": false,
        "hasVideo": false,
        "record": false,
        "transport": "SRT",
        "cdn": false,
        "timeout": 90000,
        "maxTimestampDiff": 1,
        "allowedList": []
```

```
    }
]
```

RETURN CODES

Code	Reason
200	OK
404	Not found
500	Internal error

/mpegts/find_all

Find all MPEG-TS streams

REQUEST EXAMPLE

```
POST /rest-api/mpegts/find_all HTTP/1.1
Host: localhost:8081
Content-Type: application/json
```

RESPONSE EXAMPLE

```
HTTP/1.1 200 OK
Access-Control-Allow-Origin: *
Content-Type: application/json

[
    {
        "localMediaSessionId": "32ec1a8e-7df4-4484-9a95-e7eddc45c508",
        "localStreamName": "test",
        "uri": "srt://192.168.1.39:31014",
        "status": "PROCESSED_LOCAL",
        "hasAudio": false,
        "hasVideo": false,
        "record": false,
        "transport": "SRT",
        "cdn": false,
        "timeout": 90000,
        "maxTimestampDiff": 1,
        "allowedList": []
    }
]
```

RETURN CODES

Code	Reason
200	OK
404	Not found

Code	Reason
500	Internal error

/mpegts/terminate

Stop MPEG-TS stream

REQUEST EXAMPLE

```
POST /rest-api/mpegts/find_all HTTP/1.1
Host: localhost:8081
Content-Type: application/json

{
    "localStreamName":"test"
}
```

RESPONSE EXAMPLE

```
HTTP/1.1 200 OK
Access-Control-Allow-Origin: *
Content-Type: application/json
```

RETURN CODES

Code	Reason
200	OK
404	Not found
500	Internal error

Parameters

Name	Description	Example
localStreamName	Name to set to the stream on server	`test`
transport	Transport to use	`srt`
uri	Endpoint URI to publish the stream	`srt://192.168.1.39:31014`
localMediaSessionId	Stream media session Id	`32ec1a8e-7df4-4484-9a95-e7eddc45c508`
status	Stream status	`CONNECTED`

Name	Description	Example
hasAudio	Stream has audio track	`true`
hasVideo	Stream has video track	`true`
record	Stream is recording	`false`
timeout	Maximum media data receiving timeout, ms	`90000`
maxTimestampDiff	Maximum stream timestamps difference, s	`1`
allowedList	Client addresses list which are allowed to publish the stream	`["192.168.1.0/24"]`

Audio only or video only publishing

Since build 5.2.1253, audio only or video only stream can be published using REST API query `/mpegts/startup` parameters

- video only stream publishing

```
{
    "localStreamName":"mpegts-video-only",
    "transport":"srt",
    "hasAudio": false
}
```

- audio only stream publishing

```
{
    "localStreamName":"mpegts-audio-only",
    "transport":"srt",
    "hasVideo": false
}
```

Publishing audio with various samplerates

By default, the following video and audio parameters are used to publish MPEG-TS stream

```
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=audio 1 RTP/AVP 102
a=rtpmap:102 mpeg4-generic/44100/2
a=sendonly
m=video 1 RTP/AVP 119
a=rtpmap:119 H264/90000
a=sendonly
```

Video track must be published in H264 codec with clock rate 90000 Hz, audio track must be published in AAC with samplerate 44100 Hz, two channels.

An additional samplerates or one channel may be enabled for audio publishing if necessary. Do the following to enable:

1. Create the file `mpegts_agent.sdp` in `/usr/local/FlashphonerWebCallServer/conf` folder

   ```
   sudo touch /usr/local/FlashphonerWebCallServer/conf/mpegts_agent.sdp
   ```

2. Add necessary SDP parameters to the file

   ```
   sudo nano /usr/local/FlashphonerWebCallServer/conf/mpegts_agent.sdp
   ```

   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=audio 1 RTP/AVP 102 103 104
   a=rtpmap:102 mpeg4-generic/44100/2
   a=rtpmap:103 mpeg4-generic/48000/2
   a=rtpmap:104 mpeg4-generic/32000/1
   a=sendonly
   m=video 1 RTP/AVP 119
   a=rtpmap:119 H264/90000
   a=sendonly
   ```

3. Set the necessary permissions and restart WCS to apply changes

   ```
   sudo nano /usr/local/FlashphonerWebCallServer/bin/webcallserver set-permissions
   sudo systemctl restart webcallserver
   ```

Renewing the stream publishing after interruption

A separate UDP port is opened for every MPEG-TS publishing session to accept client connection (for SRT only) and receive media traffic. Due to security reasons, since build

5.2.1299, the stream will be stopped on server if client stops publishing (like WebRTC one), and publisher can't connect and send traffic to the same port. All the stream viewers will receive `STREAM_STATUS.FAILED` in this case. A new REST API query should be used to renew the stream publishing, with the same name if necessary.

Publishers restriction by IP address

Since build 5.2.1314 it is possible to restrict client IP addresses which are allowed to publish MPEG-TS stream using REST API `/mpegts/startup` query parameter

```
{
  "localStreamName":"mpegts-stream",
  "transport":"udp",
  "allowedList": [
    "192.168.0.100",
    "172.16.0.1/24"
  ]
}
```

The list may contain both exact IP adresses and address masks. If REST API query contains a such list, only the clients with IP addresses matching the list can publish the stream.

H265 publishing

Since build 5.2.1577 it is possible to publish MPEG-TS H265+AAC stream. H265 codec should be set in `mpegts_agent.sdp` file:

```
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=audio 1 RTP/AVP 102
a=rtpmap:102 mpeg4-generic/48000/2
a=sendonly
m=video 1 RTP/AVP 119
a=rtpmap:119 H265/90000
a=sendonly
```

H265 must also be added to supported codecs 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 publishing example using ffmpeg

```
ffmpeg -re -i source.mp4 -c:v libx265 -c:a aac -ar 48000 -ac 2 -b:a 160k -
bsf:v hevc_mp4toannexb -keyint_min 120 -profile:v main -preset veryfast -
x265-params crf=23:bframes=0 -f mpegts "srt://test.flashphoner.com:31014"
```

Warinig

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

Known issues

1. Publishing encoder may not know the stream is stopped at server side

When MPEG-TS stream publishing via UDP is stopped at server side via REST API query `/mpegts/terminate`, publishing encoder still sends media data

Symptoms

ffmpeg still sends data via UDP when MPEG-TS stream publishing is stopped on server

Solution

This is normal behaviour for UDP because the protocol itself provides no any methods to let publisher know the UDP port is already closed. Use SRT which handles the case correctly if possible.