In a browser via WebRTC

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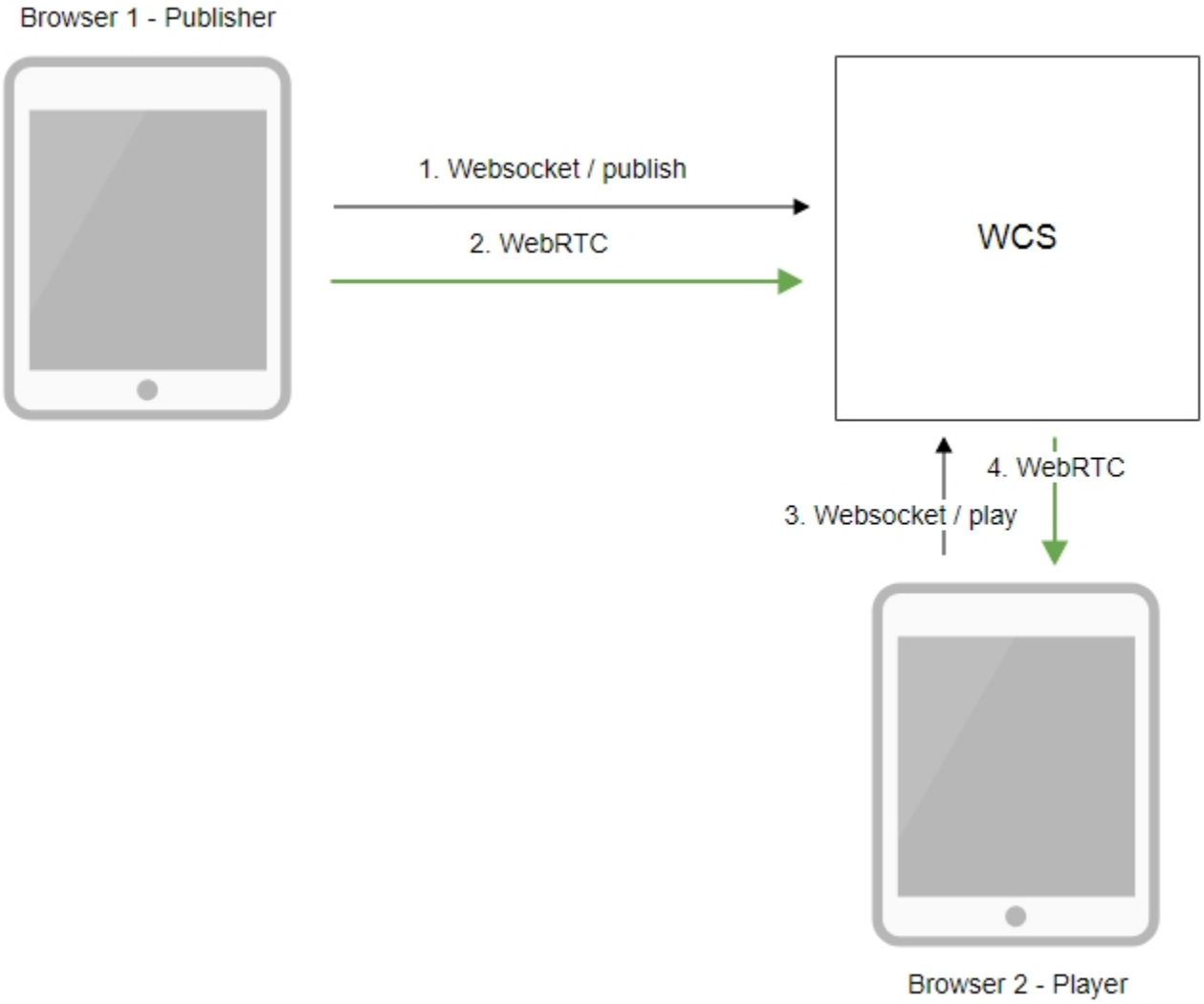
Overview

Supported platforms and browsers

	Chrome	Firefox	Safari 11	Edge
Windows	+	+		+
Mac OS	+	+	+	
Android	+	+		
iOS	+(iOS 14.4)	+(iOS 14.4)	+	

Operation flowchart



1. The browser connects to the server via the Websocket protocol and sends the publish command.
2. The browser captures the microphone and the camera and sends the WebRTC stream to the server.
3. The second browser establishes a connection also via Websocket and sends the play command.
4. The second browser receives the WebRTC stream and plays this stream on the page.

Quick manual on testing

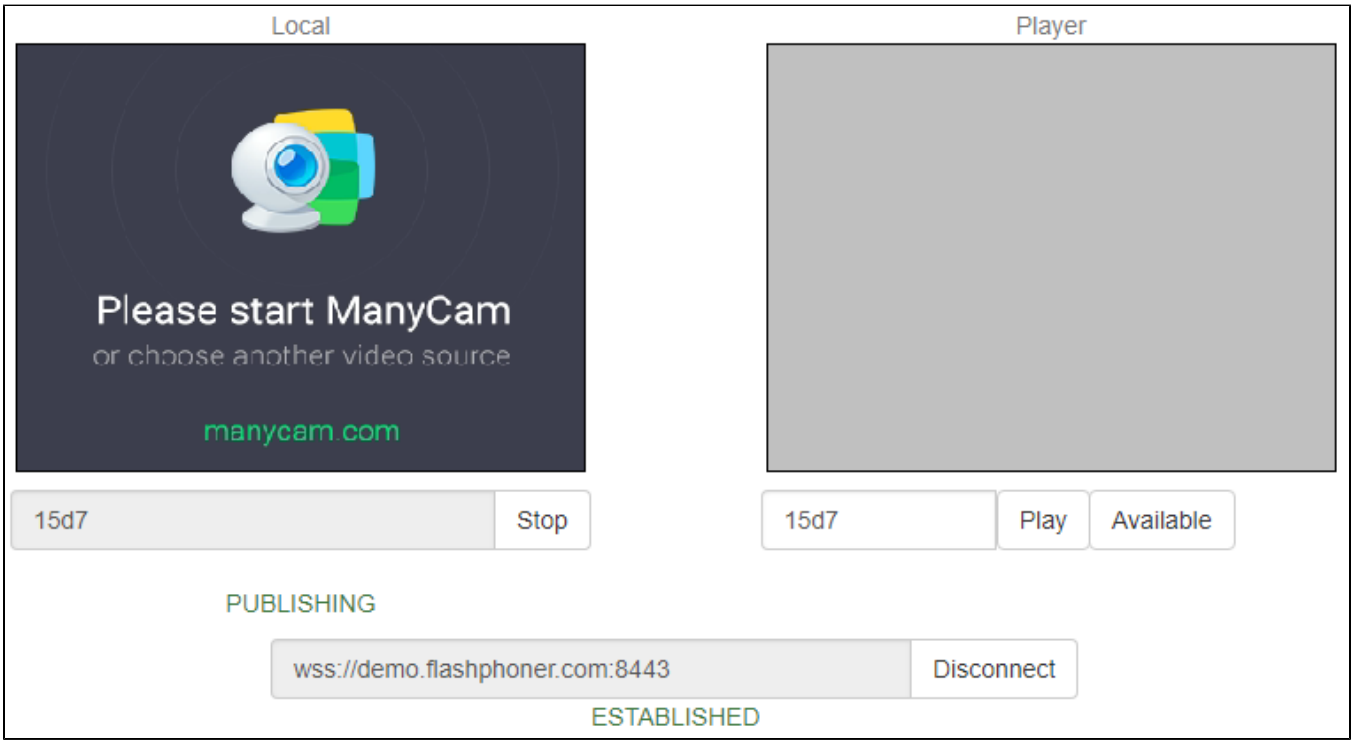
Publishing a video stream on the server and playing it via WebRTC in a browser

1. For this test we use the demo server at demo.flashphoner.com and the Two Way Streaming web application

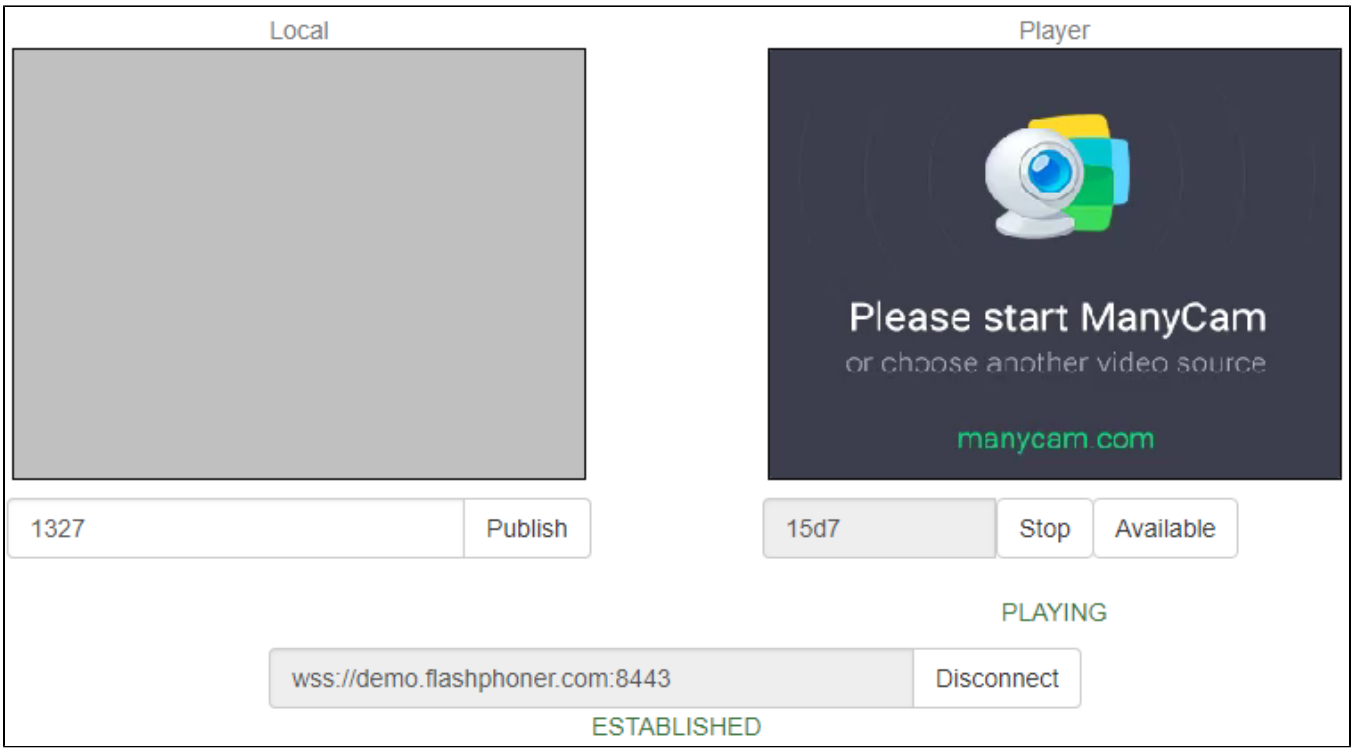
https://demo.flashphoner.com/client2/examples/demo/streaming/two_way_streaming/two_way_streaming.html

2. Establish a connection to the server using the Connect button

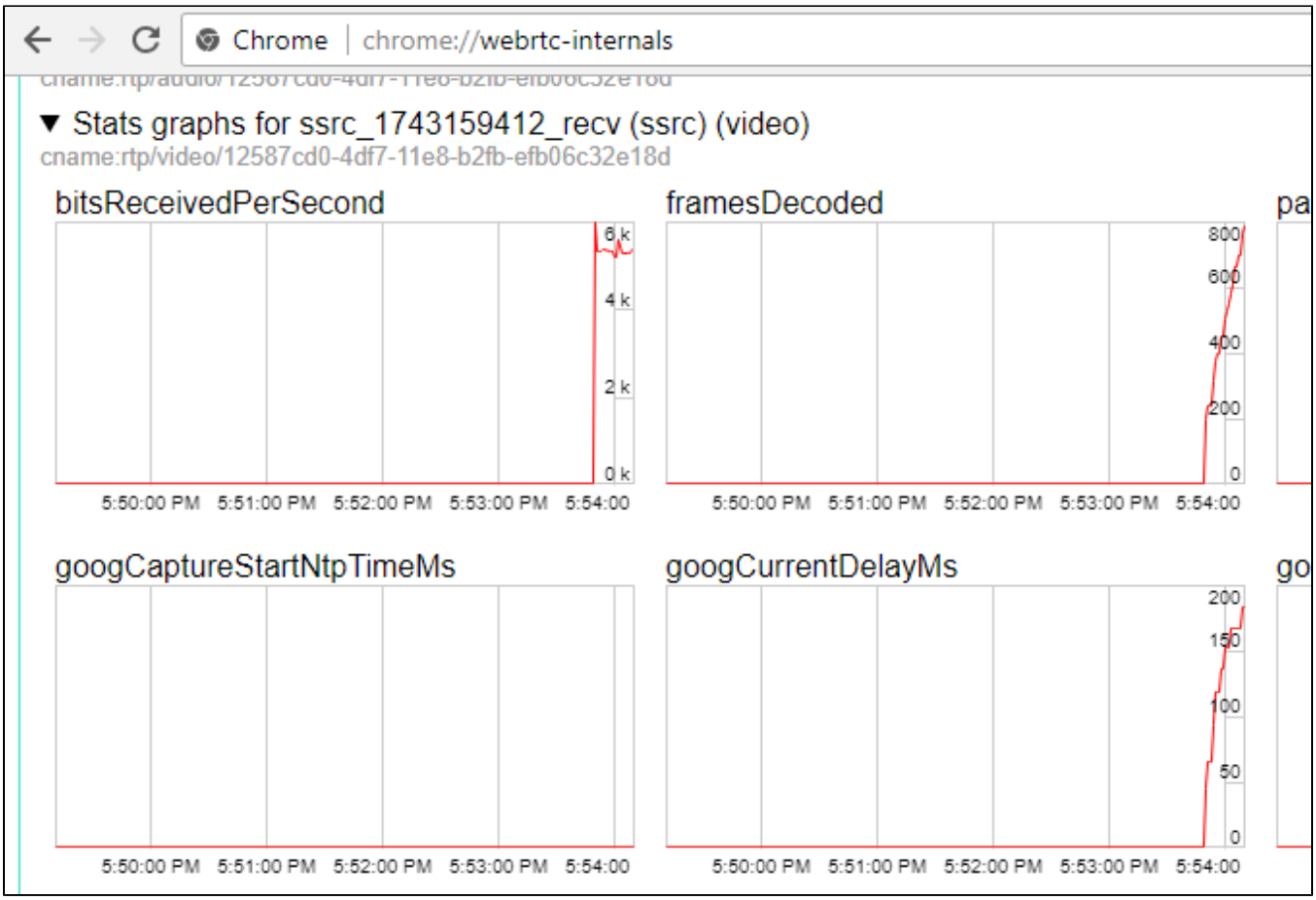
3. Click Publish. The browser captures the camera and sends the stream to the server.



4. Open Two Way Streaming in a separate window, click Connect and provide the identifier of the stream, then click Play.



5. Playback diagrams in chrome://webrtc-internals

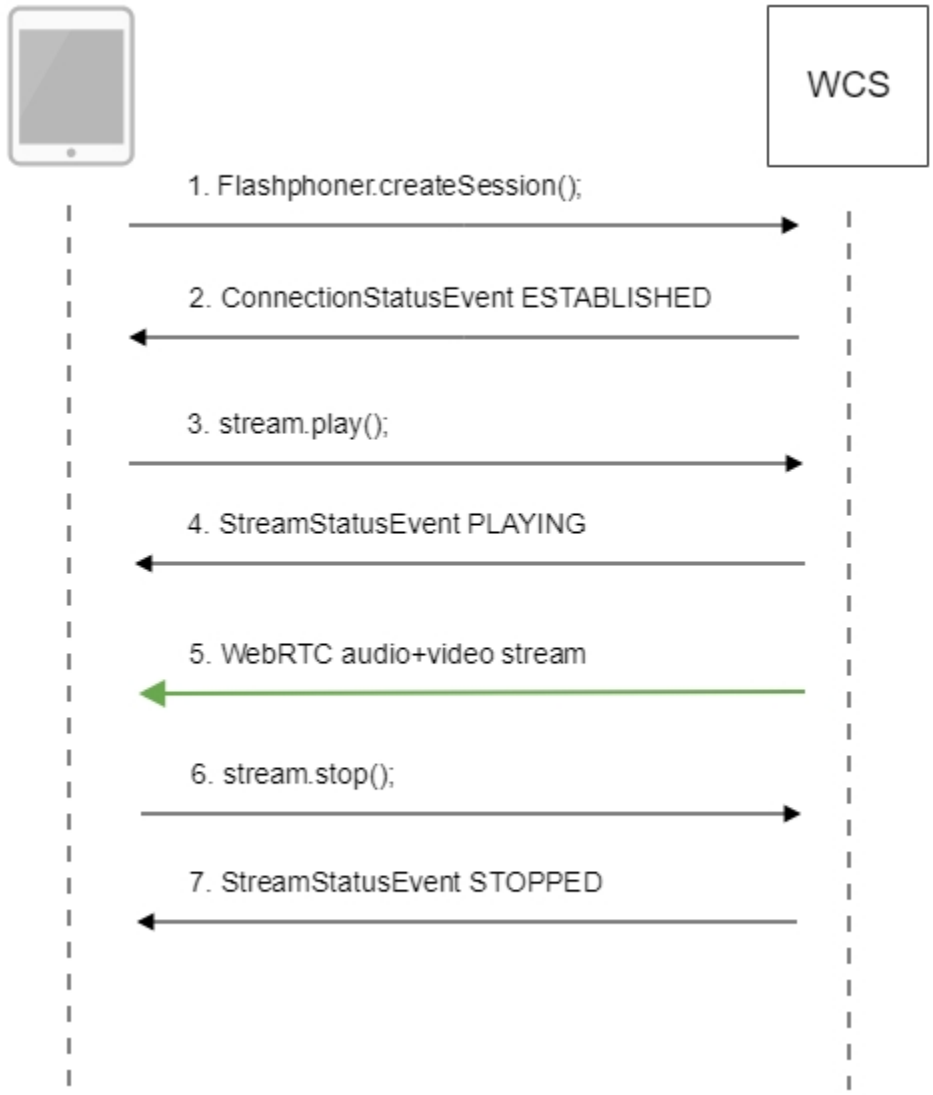


Call flow

Below is the call flow when using the Two Way Streaming example to play the stream

two_way_streaming.html

two_way_streaming.js



1. Establishing a connection to the server.

Flashphoner.createSession();code

```
Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function (session) {
    setStatus("#connectStatus", session.status());
    onConnected(session);
}).on(SESSION_STATUS.DISCONNECTED, function () {
    setStatus("#connectStatus", SESSION_STATUS.DISCONNECTED);
    onDisconnected();
}).on(SESSION_STATUS.FAILED, function () {
    setStatus("#connectStatus", SESSION_STATUS.FAILED);
    onDisconnected();
});
```

2. Receiving from the server an event confirming successful connection.

ConnectionStatusEvent ESTABLISHEDcode

```
    Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function (session) {
        setStatus("#connectStatus", session.status());
        onConnected(session);
    }).on(SESSION_STATUS.DISCONNECTED, function () {
        ...
    }).on(SESSION_STATUS.FAILED, function () {
        ...
    });
```

3. Playing the stream.

stream.play();code

```
    session.createStream({
        name: streamName,
        display: remoteVideo
        ...
    }).play();
```

4. Receiving from the server an event confirming successful playing of the stream.

StreamStatusEvent, status PLAYINGcode

```
    session.createStream({
        name: streamName,
        display: remoteVideo
    }).on(STREAM_STATUS.PENDING, function(stream) {
        ...
    }).on(STREAM_STATUS.PLAYING, function (stream) {
        setStatus("#playStatus", stream.status());
        onPlaying(stream);
    }).on(STREAM_STATUS.STOPPED, function () {
        ...
    }).on(STREAM_STATUS.FAILED, function (stream) {
        ...
    }).play();
```

5. Receiving the audio and video stream via WebRTC

6. Stopping playing the stream.

stream.stop();code

```
function onPlaying(stream) {
    $("#playBtn").text("Stop").off('click').click(function () {
        $(this).prop('disabled', true);
        stream.stop();
    }).prop('disabled', false);
    $("#playInfo").text("");
}
```

7. Receiving from the server an event confirming the playback is stopped.

StreamStatusEvent, status STOPPEDcode

```
session.createStream({
    name: streamName,
    display: remoteVideo
}).on(STREAM_STATUS.PENDING, function(stream) {
    ...
}).on(STREAM_STATUS.PLAYING, function (stream) {
    ...
}).on(STREAM_STATUS.STOPPED, function () {
    setStatus("#playStatus", STREAM_STATUS.STOPPED);
    onStopped();
}).on(STREAM_STATUS.FAILED, function (stream) {
    ...
}).play();
```

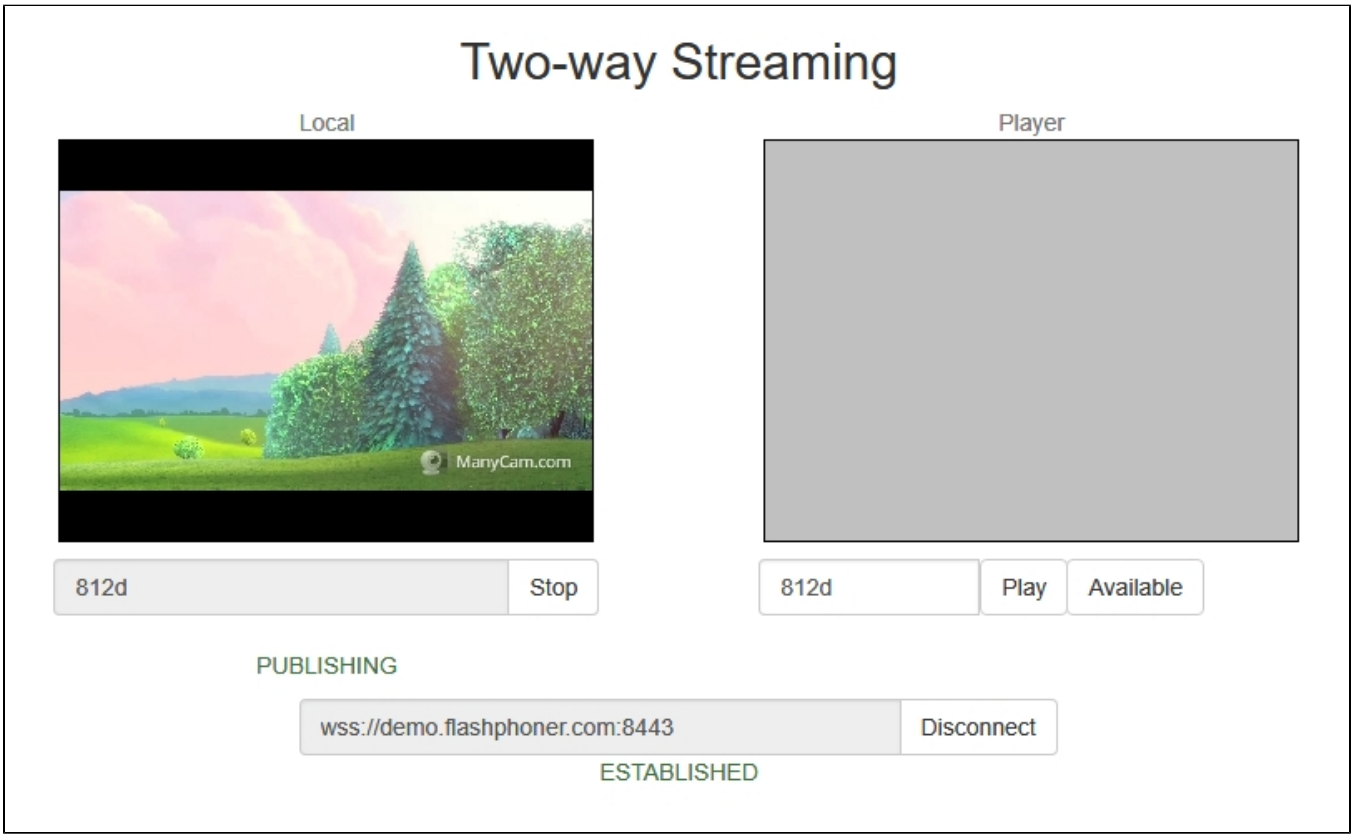
Playing two or more streams on the same page

WCS provides possibility to play two or more streams on the same page. In the context offlowchartandcall flowplaying multiple streams is no different from playing just one.

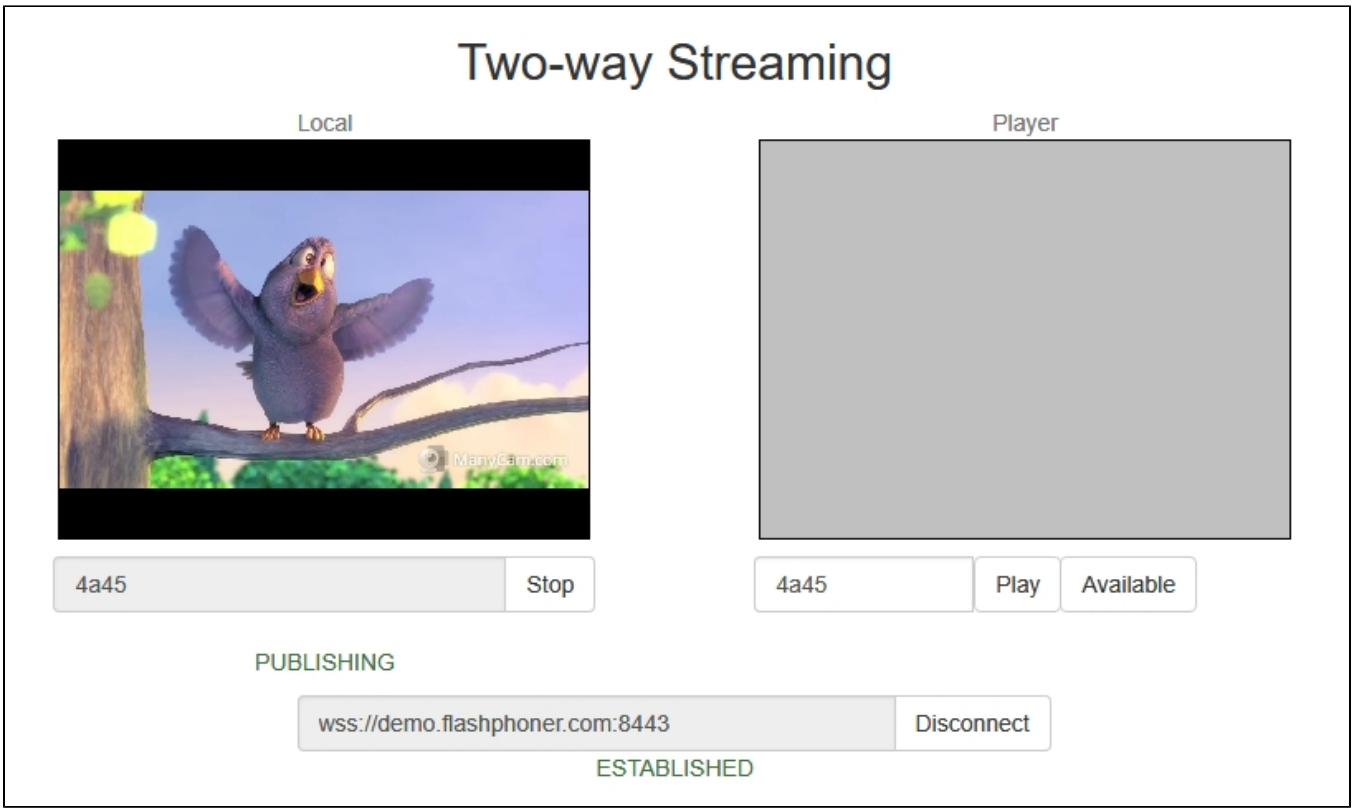
1. For the test we use:

- the demo server at demo.flashphoner.com;
- theTwo Way Streamingweb application to publish streams
- the2 Playersweb application to play streams

2. Open the Two Way Streaming web application, click Connect, then Publish. Copy the identifier of the first stream from the Play window:

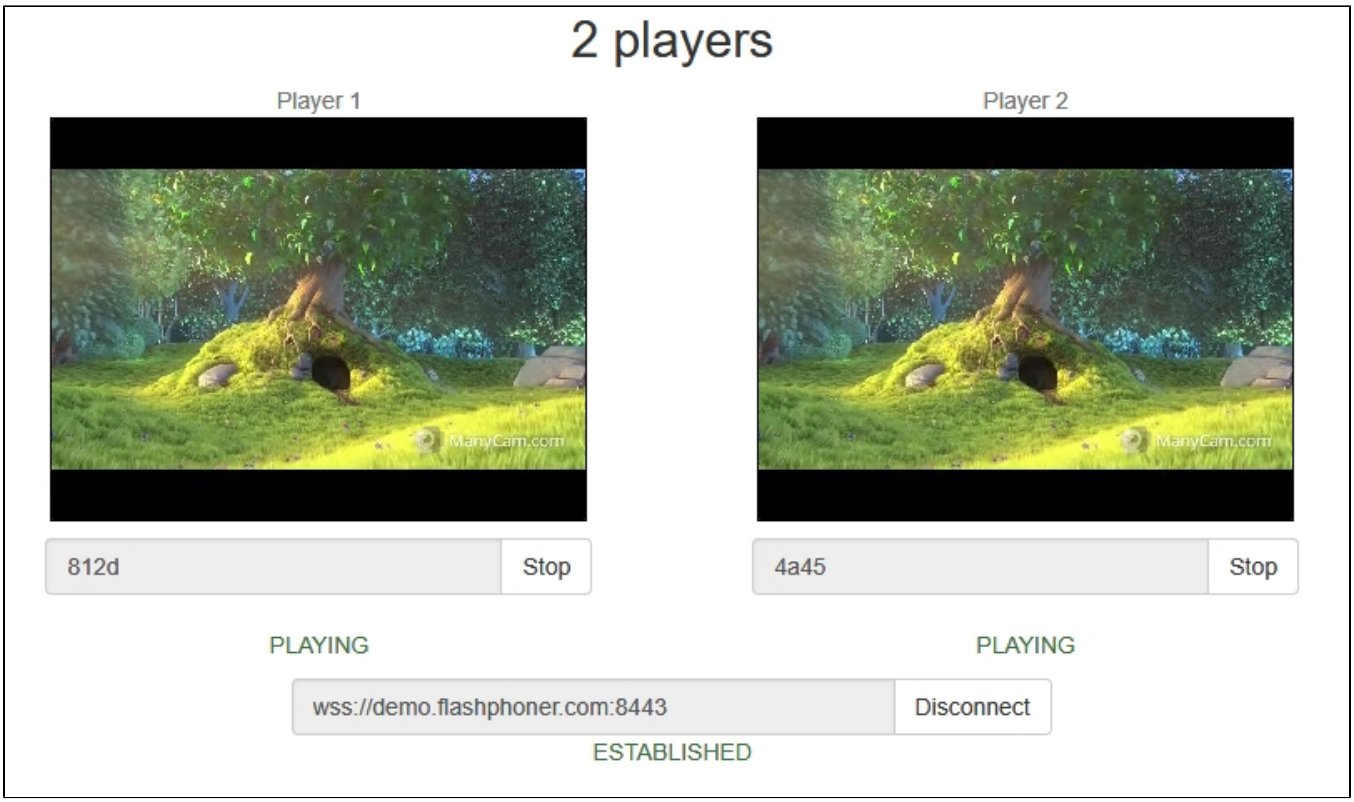


3. In another tab, open the Two Way Streaming web application, click Connect, then Publish. Copy the identifier of the second stream from the Play window:

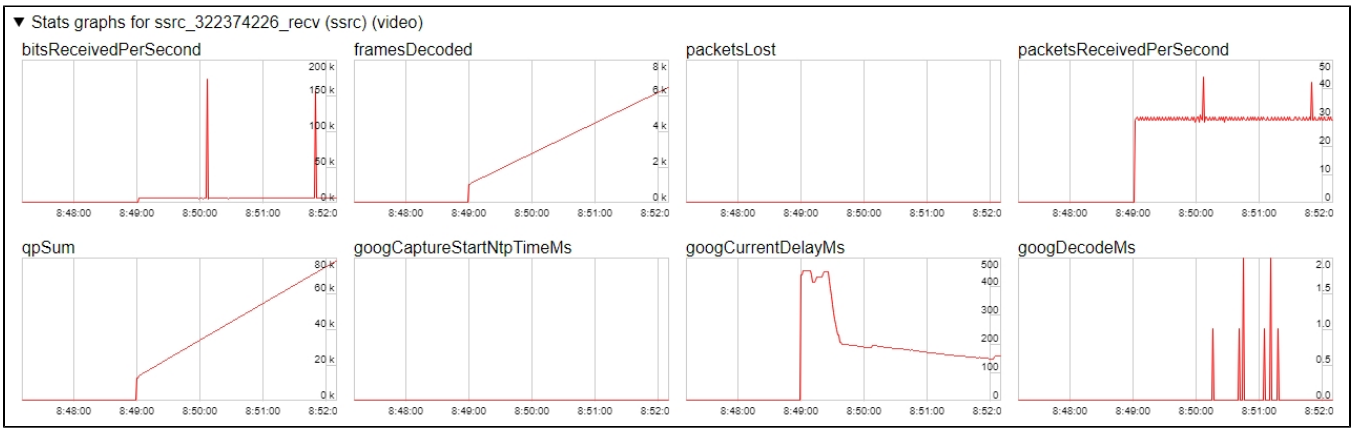


4. Open the 2 Players web application and specify identifiers of the first (left) and the second (right) streams:

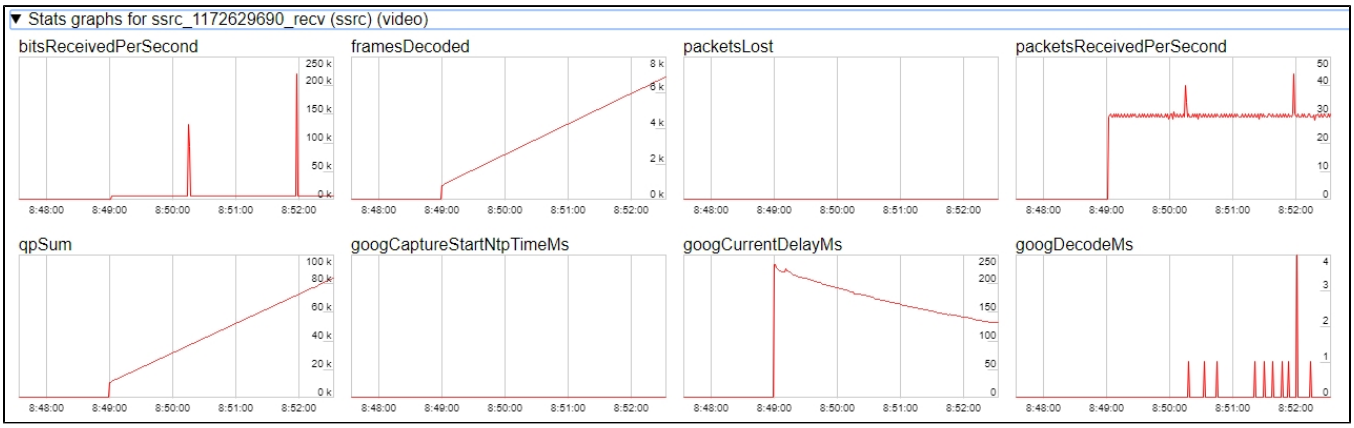
5. Click Play below right and left players:



6. Diagrams in chrome://webrtc-internals for the first stream:



7. Diagrams in chrome://webrtc-internals for the second stream:



Maximum number of streams to play on the same page simultaneously

Maximum number of streams to play on the same page simultaneously with acceptable quality depends on the following parameters:

- a single stream parameters (resolution amd bitrate)
- channel bandwidth from server to client
- a transport used (UDP or TCP)
- client device performance

For example, the following maximum values are experimentally obtained for the stream 1920x1080 with 2 Mbps bitrate using TCP transport on channel bandwidth 30-35 Mbps:

- Intel Core i5 8 gen and newer based PC, from 8 Gb RAM: up to 15 audio+video streams, or up to 6 audio+video and 14 audio only streams
- A flagship Android/iOS device of year 2018 and newer (Samsung S series, Apple iPhone Pro): up to 15 audio+video streams, or up to 6 audio+video and 14 audio only streams
- A middle ol lower class device, or obsoleted Android/iOS device (Nokia 5, Apple iPhone 7): up to 6 audio+video streams, or audio only streams

Thus, for the stream 1920x1080 with 2 Mbps bitrate seems optimal to play no more than 6 streams on the same page for any client can play them.

Let's test a webinar case: one desktop stream 1920x1080 with 2 Mbps bitrate and a number of webcam streams 640x360 with 500 kbps bitrate. Under the same channel conditions:

- Intel Core i5 8 gen and newer based PC, from 8 Gb RAM: up to 25 audio+video streams, or up to 6 audio+video and 25 audio only streams
- A flagship Android/iOS device of year 2018 and newer (Samsung S series, Apple iPhone Pro): up to 20 audio+video streams, or up to 6 audio+video and 25 audio only streams
- A middle ol lower class device, or obsoleted Android/iOS device of year 2017 and newer: up to 10 audio+video streams, or up to 6 audio+video and 15 audio only streams

Thus, for webinar case with one desktop stream and a number of webcam streams seems optimal to play no more than 10 streams on the same page for any client can play them.

WebRTC stream playback in custom player

A stream published on WCS server can be played via WebRTC in custom player, for example, in a VR player. To fo this, video page element to play stream should be passed as`remoteVideo`parameter tosession.createStream() WebSDK function

session.ceateStream() code

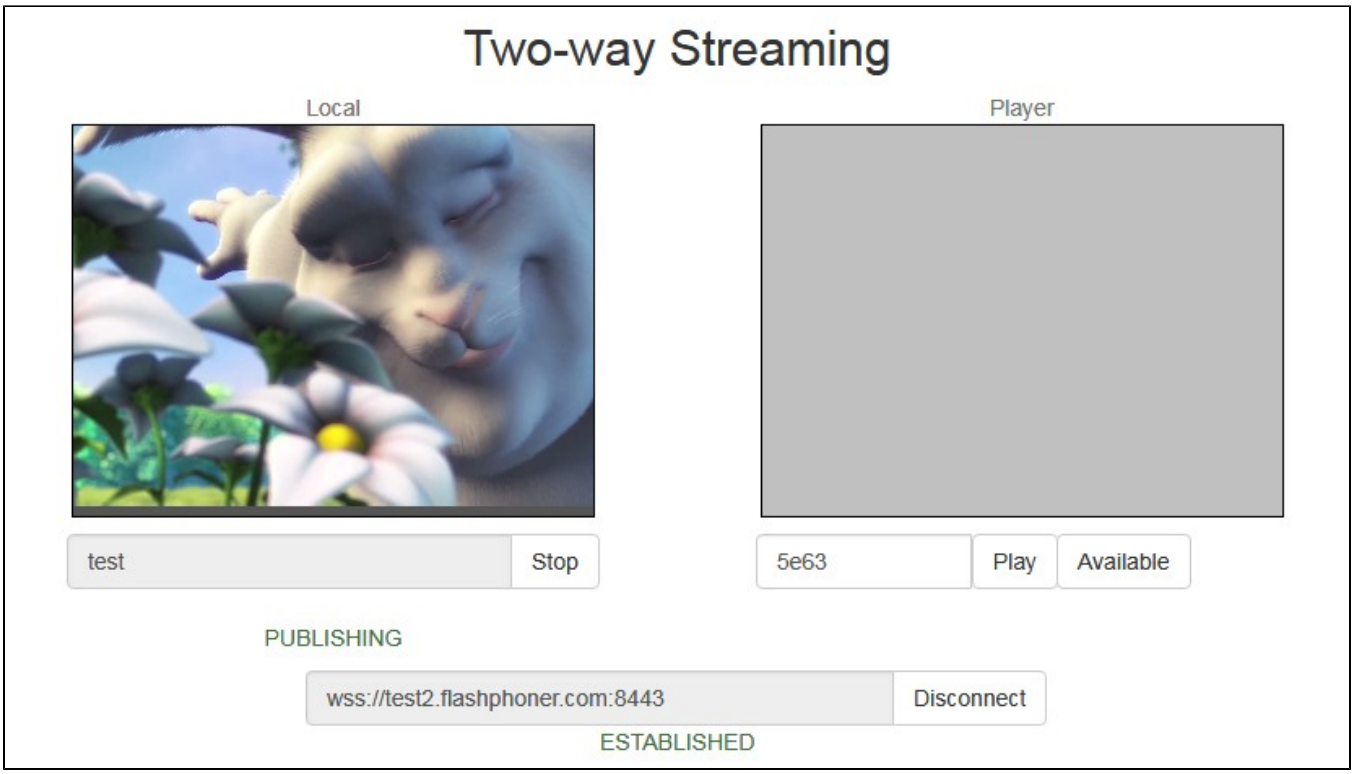
```
session.createStream({
        name: document.getElementById('playStream').value,
        display: display,
        remoteVideo: video
})
...
```

Testing

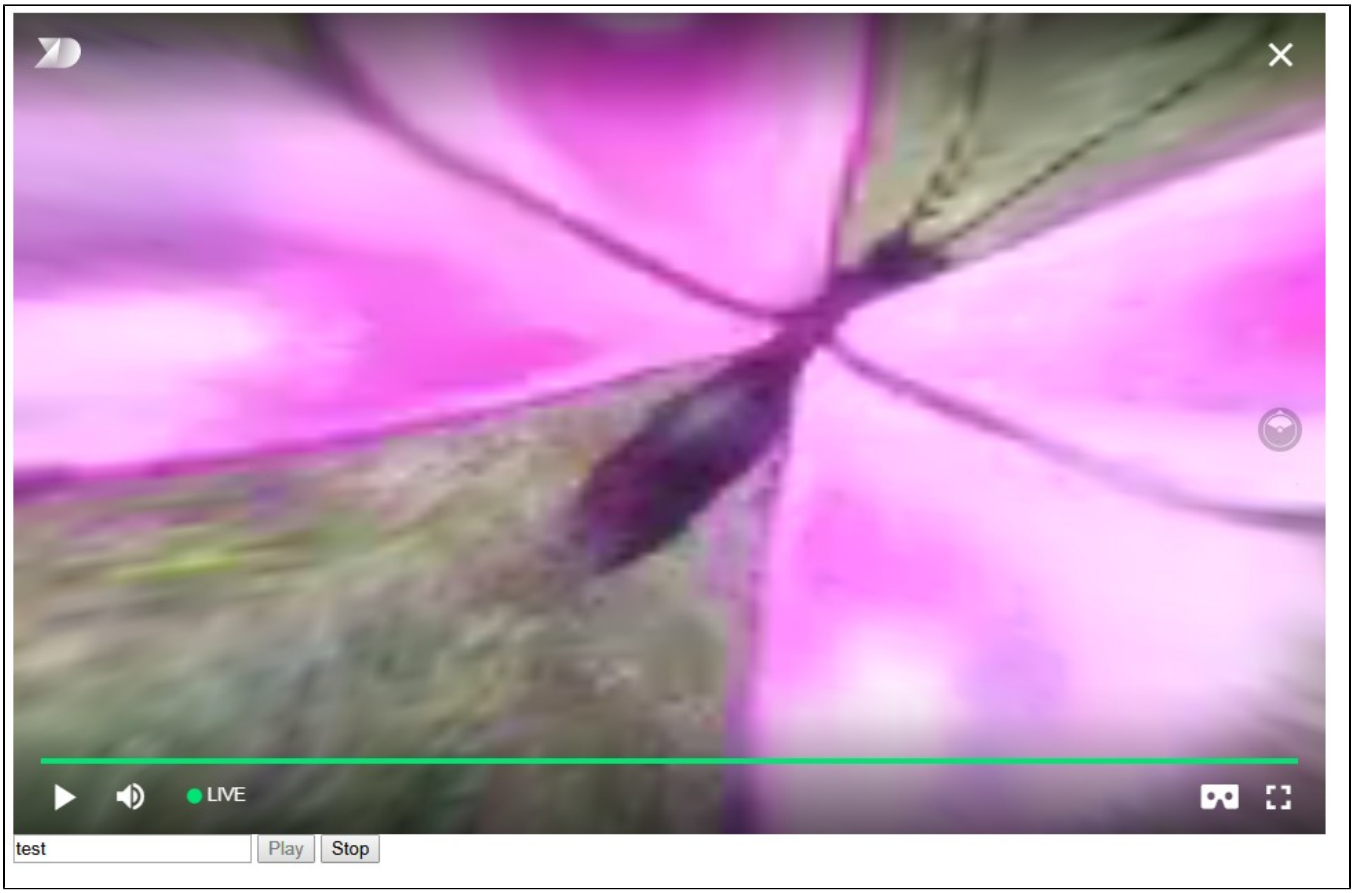
1. For test we use:

- WCS server
- Two Way Streamingweb application to publish a stream
- DelightVR player to play a stream

2. Publish stream on WCS server



3. Play stream in VR player



Custom player page code sample

1. Video page element, stream name input field and buttons to start and stop playback declaration

```
        <div style="width: 50%;" id="display">
            <dl8-live-video id="remoteVideo" format="STEREO_TERPON">
                <source>
            </dl8-live-video>
        </div>
                <input class="form-control" type="text" id="playStream" placeholder="Stream Name">
                <button id="playBtn" type="button" class="btn btn-default" disabled>Play</button>
                <button id="stopBtn" type="button" class="btn btn-default" disabled>Stop</button>
```

2. Player ready to playback event handling

```
            document.addEventListener('x-dl8-evt-ready', function () {
                                dl8video = document.getElementById('remoteVideo');
                                $('#playBtn').prop('disabled', false).click(function() {
                                        playStream();
                                });
            });
```

3. Connection to WCS server establishing and stream creation

```
                        var video = dl8video.contentElement;
                        Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function
(session) {
                        var session = Flashphoner.getSessions()[0];
                        session.createStream({
                                name: document.getElementById('playStream').value,
                                display: display,
                                remoteVideo: video
                        }).on(STREAM_STATUS.PLAYING, function (stream) {
                ...
                        }).play();
                        })
```

4. Start playback in VR player and stop button click handling

```
                        ...
                        }).on(STREAM_STATUS.PLAYING, function (stream) {
                                dl8video.start();
                                $('#stopBtn').prop('disabled', false).click(function() {
                                        $('#playBtn').prop('disabled', false);
                                        $('#stopBtn').prop('disabled', true);
                                        stream.stop();
                                        dl8video.exit();
                                });
                        }).play();
                        })
```

Full custom player page code sample

Code

```
<!DOCTYPE html>
<html>
    <head>
        <title>WebRTC Delight</title>
        <meta charset="UTF-8">
        <meta name="viewport" content="width=device-width, initial-scale=1.0">
                  <script type="text/javascript" src="../../../../flashphoner.js"></script>
                  <script type="text/javascript" src="../../dependencies/jquery/jquery-1.12.0.js"></script>
                  <script type="text/javascript" src="../../dependencies/js/utils.js"></script>
                  <script src="dl8-66b250447635476d123a44a391c80b09887e831e.js" async></script>
        <meta name="dl8-custom-format" content='{"name": "STEREO_TERPON","base":"STEREO_MESH","params":{"uri":
"03198702.json"}}'>
    </head>
    <body>
        <div style="width: 50%;" id="display">
            <dl8-live-video id="remoteVideo" format="STEREO_TERPON">
                <source>
            </dl8-live-video>
        </div>
                <input class="form-control" type="text" id="playStream" placeholder="Stream Name">
                <button id="playBtn" type="button" class="btn btn-default" disabled>Play</button>
                <button id="stopBtn" type="button" class="btn btn-default" disabled>Stop</button>
        <script>
                        Flashphoner.init({flashMediaProviderSwfLocation: '../../../../media-provider.swf'});
                        var SESSION_STATUS = Flashphoner.constants.SESSION_STATUS;
                        var STREAM_STATUS = Flashphoner.constants.STREAM_STATUS;
                        var STREAM_STATUS_INFO = Flashphoner.constants.STREAM_STATUS_INFO;
                        var playBtn = document.getElementById('playBtn');
                        var display = document.getElementById('display');
                        var dl8video = null;
                        var url = setURL();
            document.addEventListener('x-dl8-evt-ready', function () {
                                dl8video = document.getElementById('remoteVideo');
                                $('#playBtn').prop('disabled', false).click(function() {
                                        playStream();
                                });
            });
                        function playStream() {
                        $('#playBtn').prop('disabled', true);
                        $('#stopBtn').prop('disabled', false);
                        var video = dl8video.contentElement;
                        Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function
(session) {
                        var session = Flashphoner.getSessions()[0];
                        session.createStream({
                                name: document.getElementById('playStream').value,
                                display: display,
                                remoteVideo: video
                        }).on(STREAM_STATUS.PLAYING, function (stream) {
                                dl8video.start();
                                $('#stopBtn').prop('disabled', false).click(function() {
                                        $('#playBtn').prop('disabled', false);
                                        $('#stopBtn').prop('disabled', true);
                                        stream.stop();
                                        dl8video.exit();
                                });
                        }).play();
                        })
                        }
        </script>
    </body>
</html>
```

Automatic stream playback

PlayerandEmbed Playerexamples support automatic stream playback with the following URL parameter

```
autoplay=true
```

for example

```
https://hostname:8888/embed_player?urlServer=wss://hostname:
8443&streamName=stream1&autoplay=true&mediaProviders=WebRTC
```

Where

- hostname is WCS server hostname
- stream1 is a stream name

Autoplay issues in different browsers

Chrome

In latest Chrome versions (71 and higher) content autoplay policy was changed. Now, user has to do something to start video playback on web page, to press a key for example.

The policy change affects also audiocontext creation that is needed to play a sound. Accrding to new policy, audiocontext may only be created as response to some user action.

Therefore, in Chrome 71 and in another Chromium based browsers that support new autoplay policy, video automatic playback starts with muted sound. To enable sound user has to move volume control in Embed Player window.

Firefox and MacOS Safari

As in Chrome browser, autoplay starts with muted sound. Users action is required to unmute.

iOS Safari

Autoplay works since iOS 12.2. Note that autoplay policy as well as in Chrome browser, requires user to move volume control to start sound playback.

In iOS 12.2-12.3 sound playback may not be started even after moving volume control. In this case, video playback should be restarted without reloading the page.

Autoplay does not work in iOS Safari when Low Power Mode is enabled.

Audio playback tuning in iOS Safari

If one video stream is playing and then another video stream is publishing on the same page (videochat case for example) in iOS Safari, the sound level may change for stream played. This can be escaped by the following ways:

1. Query media devices access on session creation before playing a stream

```
    Flashphoner.createSession({urlServer: url}).on(SESSION_STATUS.ESTABLISHED, function (session) {
        ...
        if (Browser.isSafariWebRTC() && Browser.isiOS() && Flashphoner.getMediaProviders()[0] === "WebRTC") {
            Flashphoner.playFirstVideo(localVideo, true, PRELOADER_URL).then(function () {
                Flashphoner.getMediaAccess(null, localVideo).then(function (disp) {
                });
            });
        }
        ...
    });
```

2. 1-1,5 seconds after PLAYING stream status receiving, mute and unmute video and/or sound

```
    session.createStream({
        name: streamName,
        display: remoteVideo
    }).on(STREAM_STATUS.PENDING, function (stream) {
        ...
    }).on(STREAM_STATUS.PLAYING, function (stream) {
        setStatus("#playStatus", stream.status());
        onPlaying(stream);
        if (Browser.isSafariWebRTC() && Browser.isiOS() && Flashphoner.getMediaProviders()[0] === "WebRTC") {
            setTimeout(function () {
                stream.muteRemoteAudio();
                stream.unmuteRemoteAudio();
            }, 1500);
        }
        ...
    }).play();
```

Stereo audio playback in browser

The Opus codec parameters shoul be set on server side to play stereo audio in browser as like as for stream publishing

```
opus_formats = maxaveragebitrate=64000;stereo=1;sprop-stereo=1;
```

In this case Firefox will play stereo audio without additional setup.

When a stream captured from RTMP, RTSP or VOD source is plaing in browser, audio is usually transcoded to Opus codec. By default, Opus encoder is configured to play a speech and monophonic audio. Encoder bitrate should be raised to 60 kbps or higher to play stereo in browser

```
opus.encoder.bitrate=60000
```

Chromium-based browsers

By default, Chrome browser plays WebRTC stream with stereo sound in Opus codec as mono due to engine bug. An additiona client setup is required to workaround this Chrome behaviour depending on client implementation

Using Web SDK

Since Web SDK build 0.5.28.2753.151 the following playback constraint option is available

```
constraints.audio.stereo=true
```

for example

```
    session.createStream({
        name: streamName,
        display: remoteVideo,
        constraints: {
            audio: {
                stereo: true
            }
        }
        ...
    }).play();
```

Using Websocket API

If only Websocket API is used in project, it is necessary to change the Opus codec parameters in offer SDP right after its creation

```
        var connection = new RTCPeerConnection(connectionConfig, connectionConstraints);
        ...
        connection.createOffer(constraints).then(function (offer) {
          offer.sdp = offer.sdp.replace('minptime=10', 'minptime=10;stereo=1;sprop-stereo=1');
          connection.setLocalDescription(offer).then(function () {
              ...
          });
        });
```

Additional video stream playing delay

Sometimes it is necessary to add a certain fixed delay relative to translation while playing a stream. To do this, the option playoutDelay can be used since WebSDK build0.5.28.2753.142shipped withWCS build5.2.708and later:

```
    session.createStream({
        name: streamName,
        display: remoteVideo,
        playoutDelay: 10
    }).on(STREAM_STATUS.PENDING, function (stream) {
        ...
    }).play();
```

The delay is set in seconds.

The option works in Chromium browsers only which support the attribute

```
partial interface RTCRtpReceiver {
  attribute double? playoutDelayHint;
};
```

The delay is not applied to audio tracks in the stream and to audio only streams.

Redundancy support while playing audio

Since build5.2.1969a redundancy is supported while playing audio data (RED,RFC2198). This allows to reduce audio packet loss when using opus codec. The feature is configured as like as redundancy support for publishing audio.

Use the following parameter to enable redundancy for audio playback

```
red_max_encodings_number=2
```

The parameter sets an additional data proportion. WebRTC library in browser usually uses a double redundancy while publishing, therefore it is recommended to set the same value for playback.

Known issues

1. Possible bug in the Safari browser on iOS leads to freezes while playing via WebRTC

Symptoms: video playback stops, while the audio track may continue playing. Recovery needs reloading the page or restarting the browser.

Solution:

a) enable the transcoder on the server by setting the following parameter in flashphoner.properties

```
disable_streaming_proxy=true
```

b) when playing the stream from iOS Safari explicitly specify width and height, for example:

```
session.createStream({constraints:{audio:true,video:{width:320,height:240}}}).play();
```

2. Audiocodec PMCU ia used instead of Opus when sttream is published via RTMP and is played via WebRTC

Symptoms: PMCU codec is shown in chrome://webrtc-internals

Solution: switch Avoid Transcoding Alhorithm off using the following parameter inflashphoner.properties

```
disable_rtc_avoid_transcoding_alg=true
```

3. When RTMP stream is published with Flash Streaming, then it is played in iOS Safari browser via WebRTC, and another stream is published form iOS Safari via WebRTC, sound stops playing in RTMP stream.

Symptoms:

a) The stream1 stream is published from Flash Streaming web application in Chrome browser on Windows
b) The stream1 stream is played in Two Way Streaming web application in iOS Safari browser. Sound and video play normally.
c) The stream2 stream is published from Two Way Streaming web application in iOS Safari browser.Sound stops playing.
d) Stop publishing stream in iOS Safari. Sound of stream1 plays again.

Solution: switch Avoid Transcoding Alhorithm off on the server using the following parameter inflashphoner.propertiesfile

```
disable_rtc_avoid_transcoding_alg=true
```

4. While publishing RTMP stream with Keep Alive disabled for all protocols, this stream playback via WebRTC in browser stops when WebSocket timeout expires

Symptoms: playback of stream published with RTMP encoder stops in browser with no error message

Solution: if Keep Alive is disabled for all protocols with the following parameterinflashphoner.propertiesfile

```
keep_alive.algorithm=NONE
```

it is necessary to switch off WebSocket read timeout with the following parameter

```
ws_read_socket_timeout=false
```

5. G722 codec does not work in Edge browser

Symptoms: WebRTC stream with G722 audio does not play in Edge browser or play without sound and with freezes

Solution: use another codec or another browser. If Edge browser must be used, exclude G722 with the following parameter

```
codecs_exclude_streaming=g722,telephone-event
```

6. Some Chromium based browsers, for example Opera, Yandex, do not support H264 codec depending on browser and OS version

Symptoms: stream publishing does not work, stream playback works partly (audio only) or does not work at all

Solution: enable VP8 on server side

```
codecs=opus,...,h264,vp8,...
```

exclude H264 for publishing or playing on clent side

```
publishStream = session.createStream({
    ...
    stripCodecs: "h264,H264"
}).on(STREAM_STATUS.PUBLISHING, function (publishStream) {
    ...
});
publishStream.publish();
```

Note thatstream transcodingon server is enabled when stream published as H264 is played as VP8 and vice versa.

7. If Flash is enabled in site settings, an error can occur in Chrome 71 and later browser console "Cross-origin content must have visible size large than 400 x 300 pixels, or it will be blocked" while playing WebRTC stream.

Symptoms: "Cross-origin content must have visible size large than 400 x 300 pixels, or it will be blocked" message in browser console while playing WebRTC stream, playback works normally

Solution: use WebSDK without Flash support

```
flashphoner-no-flash.js
```

8.With a large number of subscribers, lags in the playback stream are observed

Symptoms: with a large number of subscribers (more than 200 per 720p stream) video lags and freezes are observed, audio can play normally

Solution: enable mutithreaded frames sending to the clients

```
streaming_distributor_video_proxy_pool_enabled=true
```

Note than the setting affects only the streams which are not transcoded on this server

9. Audio goes to voice speaker by default when playing stream in iOS Safari

Symptoms: low audio while WebRTC is playing in iOS Safari, for example, when iOS user is entering chat room

Solution: mute then unmute sound when playback is started. for example

```
    stream = session.createStream(options).on(STREAM_STATUS.PLAYING, function (stream) {
        stream.muteRemoteAudio();
        stream.unmuteRemoteAudio();
    }).play();
```

10. If JDK 11 is used, server CPU load increases dramatically when iOS Safari subscriber connects to server

Symptoms:server CPU load increases dramatically when iOS Safari subscriber connects to server

Solution: update JDK to the one of recommended versions: 8, 12, 14.

11. When two or more streams are playing on the same page in Chrome browser on some Xiaomi devices with MIUI 12, the first stream pucture may twitch

Symptoms: when two streams are playing on the same page in2 Playersexample,the first stream picture is twitching, the second stream picture flashing over the first one

Solution:

a) use MIUI 11 on Xiaomi device

b) usemixer to play two or more streams on the same page

12. In Safari 16 video can be switched to a full screen mode only if a standard video controls are enabled for HTML5 video element

Symptoms: video is not displaying when switched to full screen mode, but audio still playing, after a couple of subsequents switches the page may hang

Solution: update Web SDK to build 2.0.224 and enable standard controls in player with `useControls` stream option:

```
function playStream(session) {
    var streamName = $('#streamName').val();
    var options = {
        name: streamName,
        display: remoteVideo,
        useControls: true
    };
    ...
    stream = session.createStream(options).on(STREAM_STATUS.PENDING, function (stream) {
        ...
    });
    stream.play();
}
```