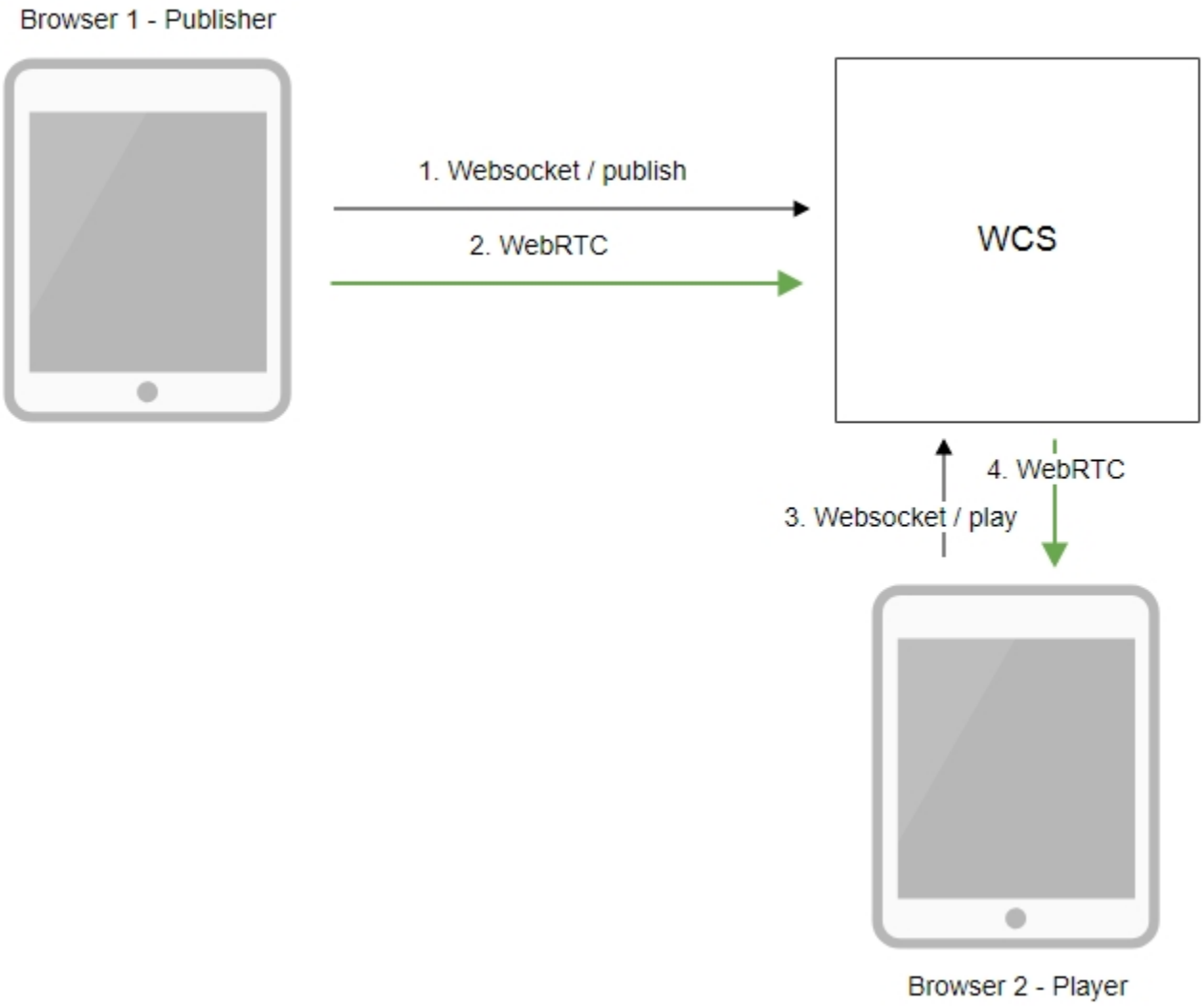
From a web camera in a browser via WebRTC

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 a 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 that stream on the page.

Quick manual on testing

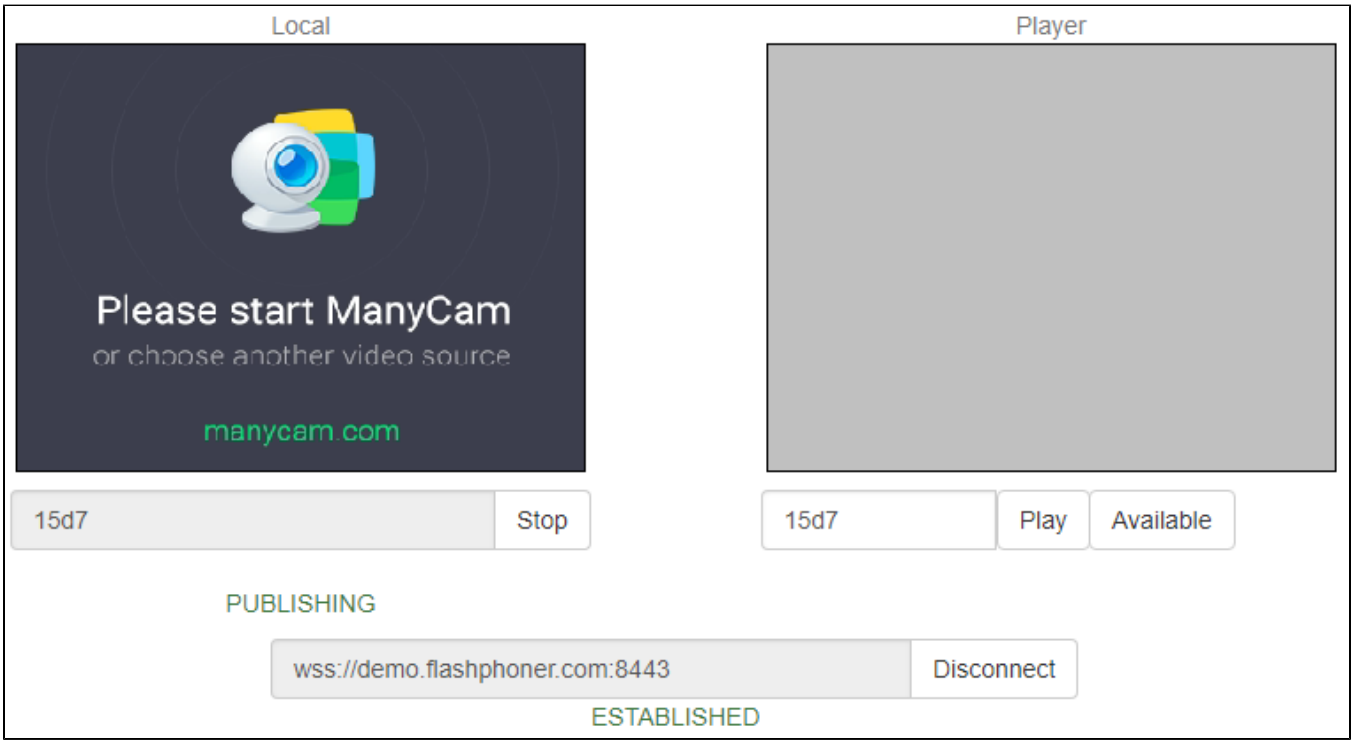
Capturing a video stream from the web camera and preparing for publishing

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

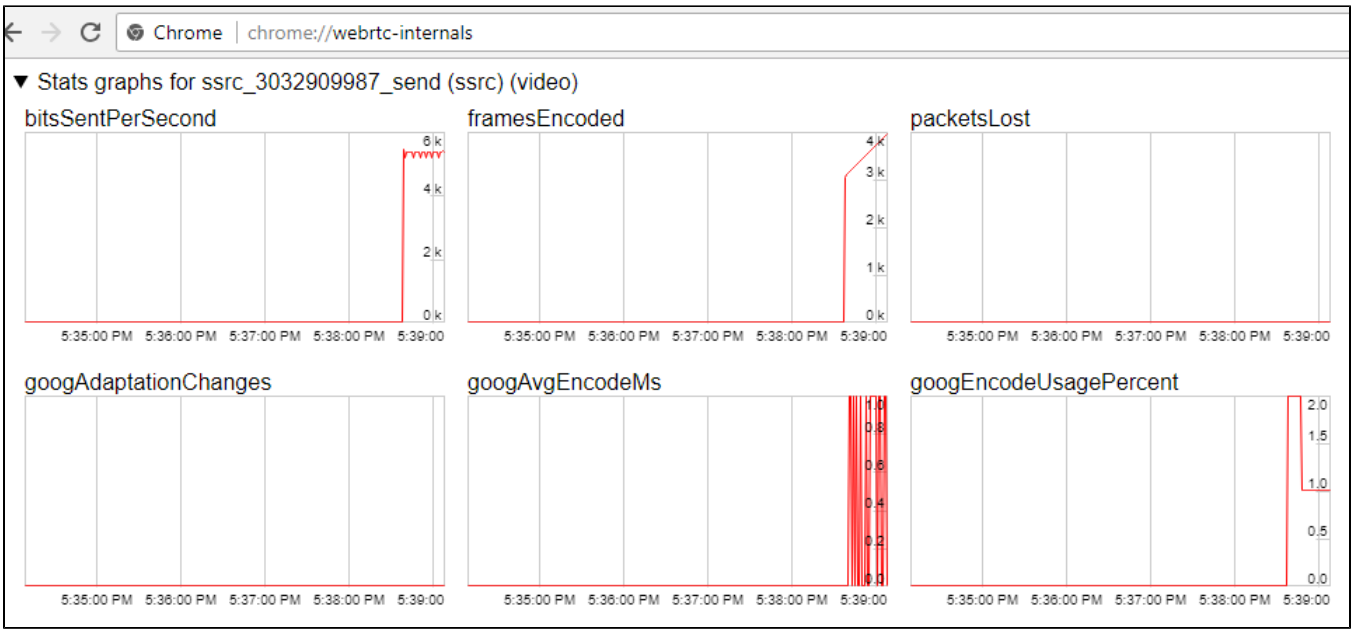
https://demo.flashphoner.com/client2/examples/demo/streaming/two_way_streaming/two_way_streaming.html

2. Establish a connection with the server by clicking the Connect button

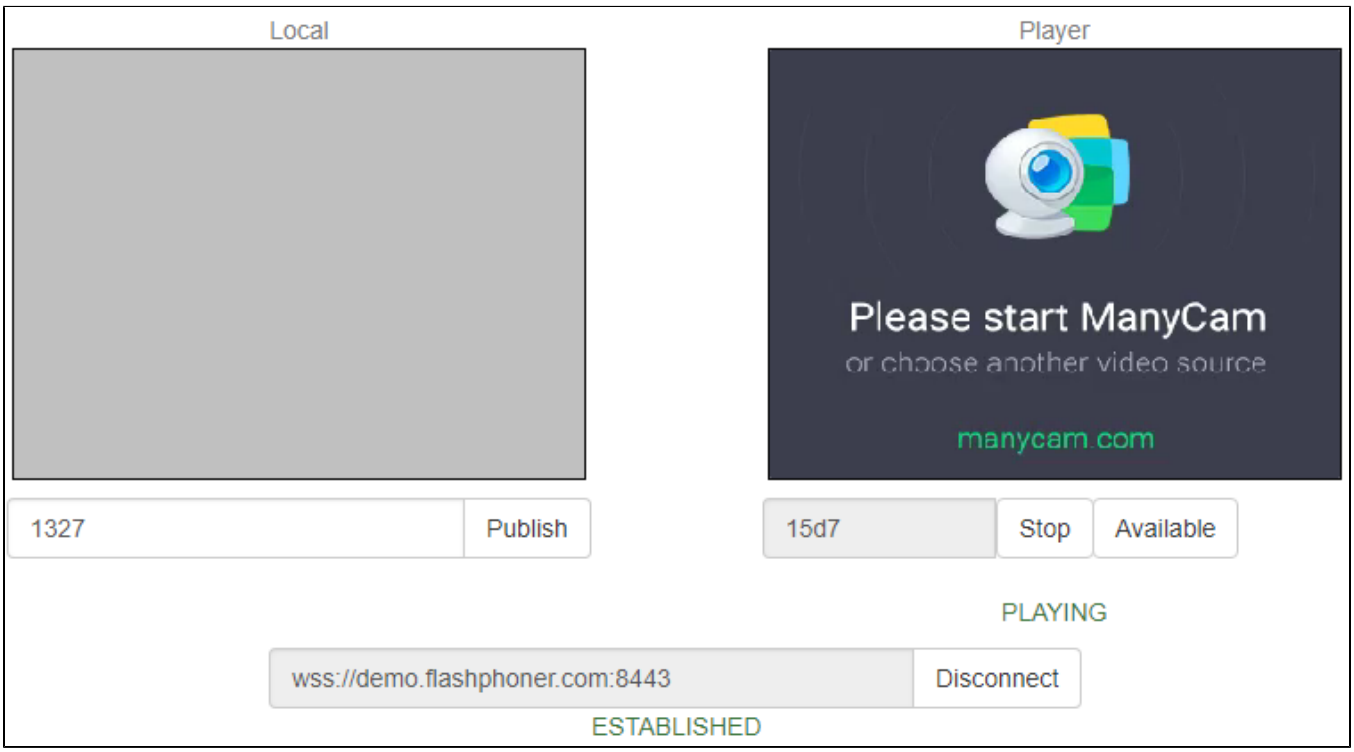
3. Click Publish. The browser captures the camera and sends the stream to the server.



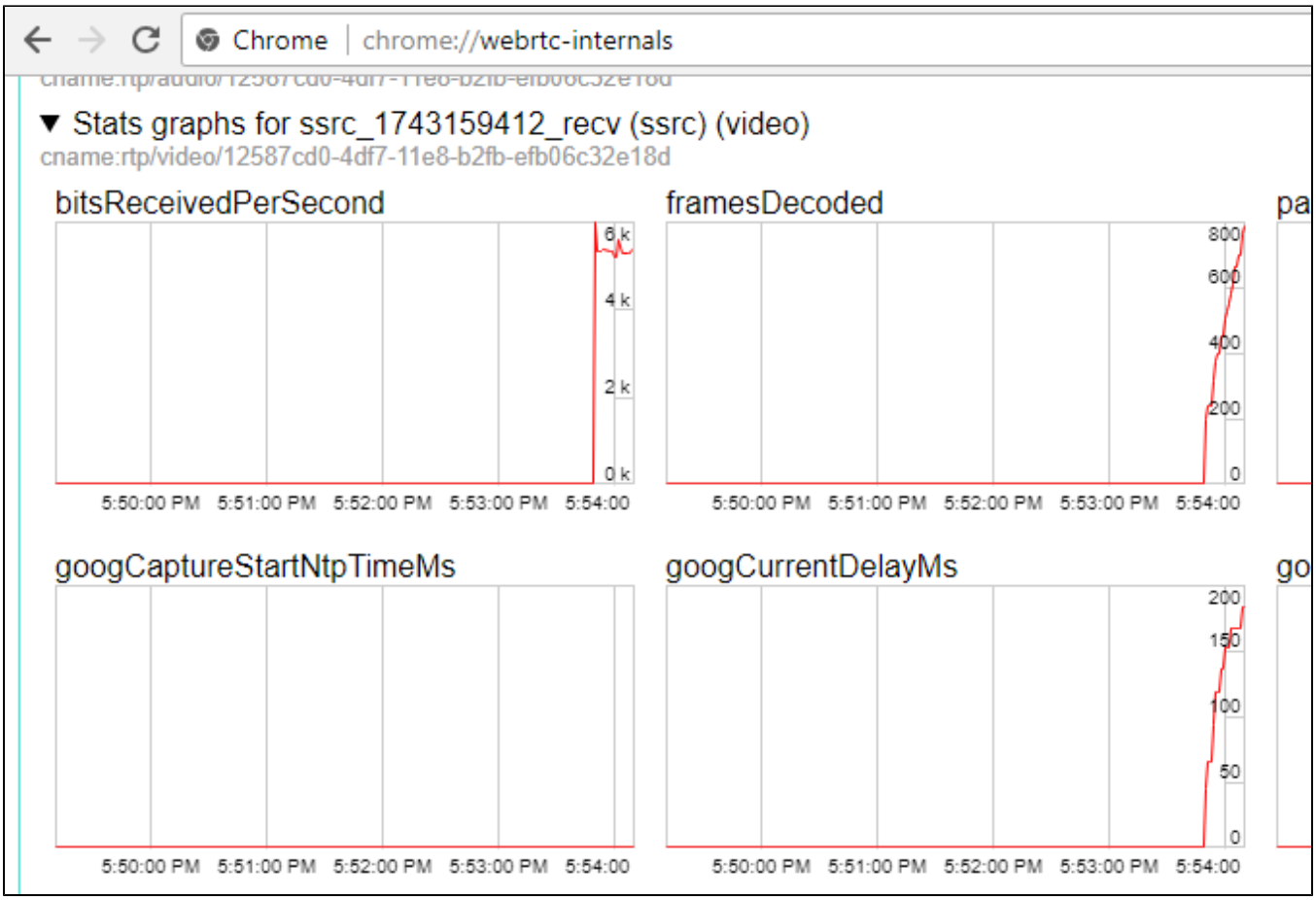
4. Make sure the stream us sent to the server and the system operates normally by opening chrome://webrtc-internals



5. Open the Two Way Streaming app in a new window, click Connect and specify the stream ID, then click Play.



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

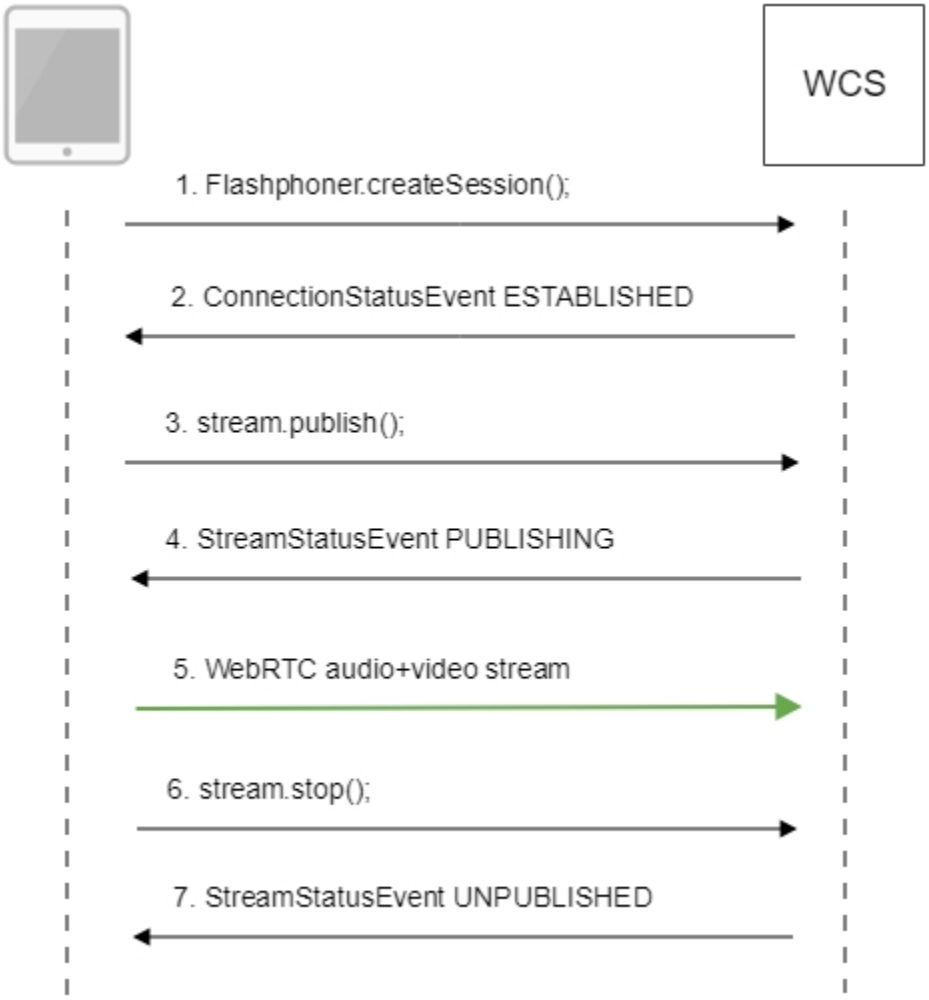


Call flow

Below is the call flow based on the Two Way Streaming example

two_way_streaming.html

two_way_streaming.js



1. Establishing 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 the successful connection status.

ConnectionStatusEvent ESTABLISHEDcode

```
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();
    });
```

3. Publishing the stream.

stream.publish();code

```
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false
    }).on(STREAM_STATUS.PUBLISHING, function (stream) {
        setStatus("#publishStatus", STREAM_STATUS.PUBLISHING);
        onPublishing(stream);
    }).on(STREAM_STATUS.UNPUBLISHED, function () {
        setStatus("#publishStatus", STREAM_STATUS.UNPUBLISHED);
        onUnpublished();
    }).on(STREAM_STATUS.FAILED, function () {
        setStatus("#publishStatus", STREAM_STATUS.FAILED);
        onUnpublished();
    }).publish();
```

4. Receiving from the server the successful publishing status.

StreamStatusEvent, статус PUBLISHINGcode

```
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false
    }).on(STREAM_STATUS.PUBLISHING, function (stream) {
        setStatus("#publishStatus", STREAM_STATUS.PUBLISHING);
        onPublishing(stream);
    ...
    }).publish();
```

5. Sending audio-video stream via WebRTC

6. Stopping publishing the stream.

stream.stop();code

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

7. Receiving from the server an even confirming successful unpublishing.

StreamStatusEvent, статус UNPUBLISHEDcode

```
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false
    ...
    }).on(STREAM_STATUS.UNPUBLISHED, function () {
        setStatus("#publishStatus", STREAM_STATUS.UNPUBLISHED);
        onUnpublished();
    ...
    }).publish();
```

Stream publishing with local video playback in Delight Player

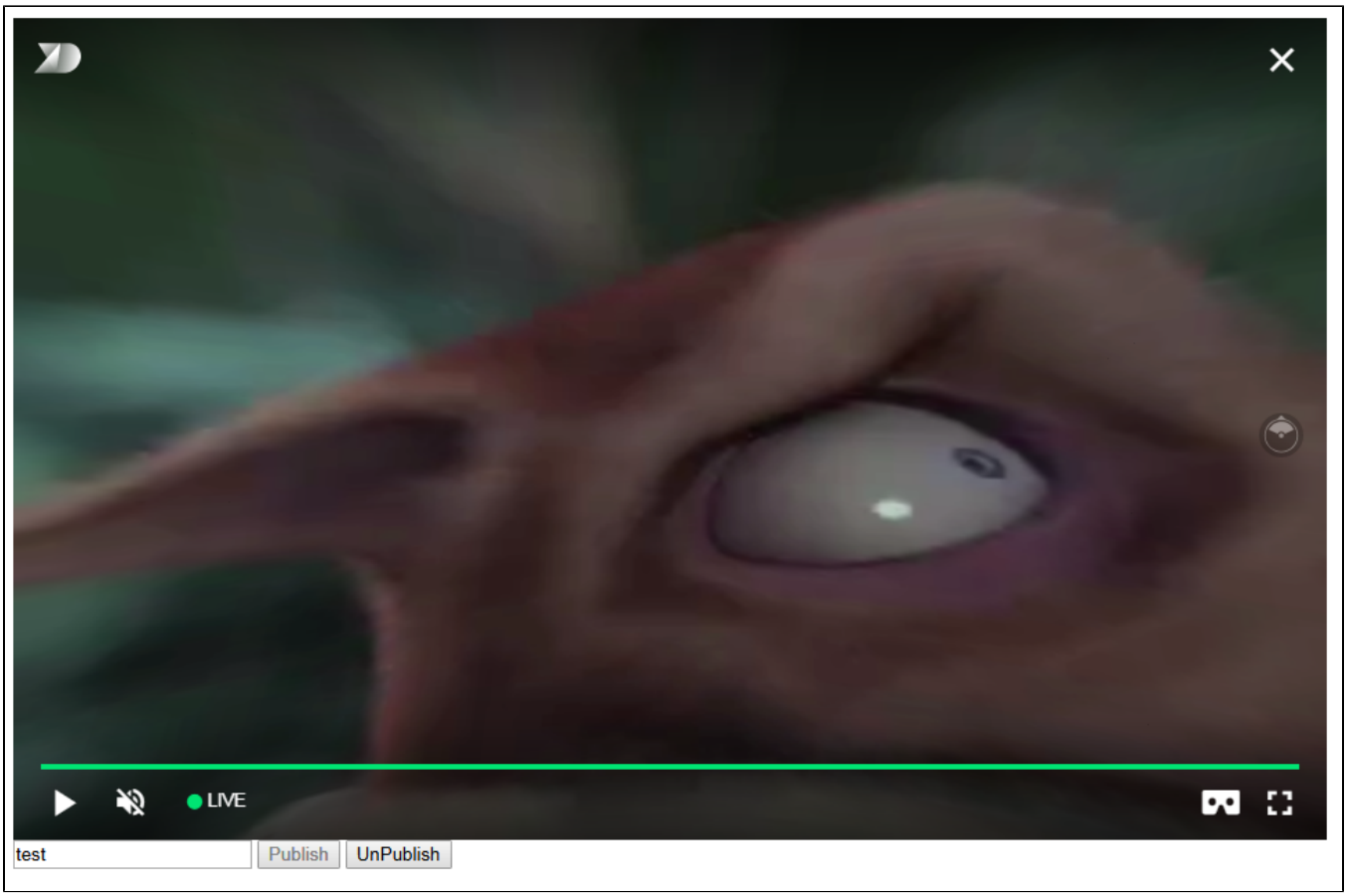
When stream is captured from webcamera it is possible to play a local videoin a browser-based VR player,Delight Playerfor example. This way stream can be played in virtual and mixed reality devices if one of browsers supporteds work on this device. To integrate a custom player JavaScript and HTML5 features are used.

Testing

1.For test we use:

- WCS server
- test page with DelightVR player to play stream while publishing

2. Set stream name test and press Publish. Stream published is played in Delight player



Player page example code

1.Declaration of video element to play the stream, stream name input field and Publish/Unpublish buttons

```
        <div style="width: 50%;">
            <dl8-live-video id="remoteVideo" format="STEREO_TERPON" muted="true">
                <source>
            </dl8-live-video>
        </div>
                <input class="form-control" type="text" id="streamName" placeholder="Stream Name">
                <button id="publishBtn" type="button" class="btn btn-default" disabled>Publish</button>
                <button id="unpublishBtn" type="button" class="btn btn-default" disabled>UnPublish</button>
```

2.Player readiness event handling

```
            document.addEventListener('x-dl8-evt-ready', function () {
                                dl8video = $('#remoteVideo').get(0);
                                $('#publishBtn').prop('disabled', false).click(function() {
                                        publishStream();
                                });
            });
```

3.Creating mock elements to play a stream

```
                        var mockLocalDisplay = $('<div></div>');
                        var mockLocalVideo = $('<video></video>',{id:'mock-LOCAL_CACHED_VIDEO'});
                        mockLocalDisplay.append(mockLocalVideo);
```

4.Establishing connection to the server 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: $('#streamName').val(),
                                display: mockLocalDisplay.get(0)
                        }).on(STREAM_STATUS.PUBLISHING, function (stream) {
                ...
                                }).publish();
                                })
```

5. Publishing stream, playback start in VR player and Unpublish button handling

```
                        ...
                        }).on(STREAM_STATUS.PUBLISHING, function (stream) {
                                var srcObject = mockLocalVideo.get(0).srcObject;
                                video.srcObject = srcObject;
                                dl8video.start();
                                mockLocalVideo.get(0).pause();
                                mockLocalVideo.get(0).srcObject = null;
                                $('#unpublishBtn').prop('disabled', false).click(function() {
                                        stream.stop();
                                        $('#publishBtn').prop('disabled', false);
                                        $('#unpublishBtn').prop('disabled', true);
                                        dl8video.exit();
                                });
                        }).publish();
```

Full source code of the sample VR player page

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%;">
            <dl8-live-video id="remoteVideo" format="STEREO_TERPON" muted="true">
                <source>
            </dl8-live-video>
        </div>
                <input class="form-control" type="text" id="streamName" placeholder="Stream Name">
                <button id="publishBtn" type="button" class="btn btn-default" disabled>Publish</button>
                <button id="unpublishBtn" type="button" class="btn btn-default" disabled>UnPublish</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 publishBtn = $('#publishBtn').get(0);
                        var dl8video = null;
                        var url = setURL();
            document.addEventListener('x-dl8-evt-ready', function () {
                                dl8video = $('#remoteVideo').get(0);
                                $('#publishBtn').prop('disabled', false).click(function() {
                                        publishStream();
                                });
            });
                        var mockLocalDisplay = $('<div></div>');
                        var mockLocalVideo = $('<video></video>',{id:'mock-LOCAL_CACHED_VIDEO'});
                        mockLocalDisplay.append(mockLocalVideo);
                        function publishStream() {
                        $('#publishBtn').prop('disabled', true);
                        $('#unpublishBtn').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: $('#streamName').val(),
                                display: mockLocalDisplay.get(0)
                        }).on(STREAM_STATUS.PUBLISHING, function (stream) {
                                var srcObject = mockLocalVideo.get(0).srcObject;
                                video.srcObject = srcObject;
                                dl8video.start();
                                mockLocalVideo.get(0).pause();
                                mockLocalVideo.get(0).srcObject = null;
                                $('#unpublishBtn').prop('disabled', false).click(function() {
                                        stream.stop();
                                        $('#publishBtn').prop('disabled', false);
                                        $('#unpublishBtn').prop('disabled', true);
                                        dl8video.exit();
                                });
                        }).publish();
                        })
                        }
        </script>
    </body>
</html>
```

Audio and video tracks activity checking

Before build5.2.533media traffic presence in a stream can be controlled by the following parameter inflashphoner.properties

```
rtp_activity_detecting=true,60
```

By default, RTP activity checking is enabled. If no media packets from browser in 60 seconds, publisher connection will be closed with the message to log "Failed by RTP activity".

Since build5.2.533, video and audio RTP activity checking settings are split

```
rtp_activity_audio=true
rtp_activity_video=true
```

Activity timeout is set by the following parameter

```
rtp_activity_timeout=60
```

Therefore, if video only streams are published to the server, audio RTP activity checking should be disabled

```
rtp_activity_audio=false
```

If audio only streams are published to the server, video RTP activity checking should be disabled

```
rtp_activity_video=false
```

RTP activity can be checked for publishing streams only, not for playing streams.

If Chrome browser sends empty video due to web camera conflict

Some Chrome versions does not return an error if web camera is busy, but publish a stream with empty video (black screen). In this case, stream publishing can be stopped by two ways: using JavaScript and HTML5 on client, or using server settings.

Stopping a stream with empty video on client side

Videotrack that Chrome browsers creates for busy web camera, stops after no more than one second publishing, then stream is send without a videotrack. In this case videotrack state (`readyState`variable) changes to`ended`, and corresponding `onended`event is generated that can be catched by web application. To use this event:

1. Add to web application script the registartion function for onended event handler, in which stream pub;ishing is stopped with`stream.stop()`

```
function addVideoTrackEndedListener(localVideo, stream) {
    var videoTrack = extractVideoTrack(localVideo);
    if (videoTrack && videoTrack.readyState == 'ended') {
        console.error("Video source error. Disconnect...");
        stream.stop();
    } else if (videoTrack) {
        videoTrack.onended = function (event) {
            console.error("Video source error. Disconnect...");
            stream.stop();
        };
    }
}
```

2. Add function to remove event handler when stream is stopped

```
function removeVideoTrackEndedListener(localVideo) {
    var videoTrack = extractVideoTrack(localVideo);
    if(videoTrack) {
        videoTrack.onended = null;
    }
}
```

3. Add function to extract videotrack

```
function extractVideoTrack(localVideo) {
    return localVideo.firstChild.srcObject.getVideoTracks()[0];
}
```

4. Register event handler when publishing a stream

```
    session.createStream({
        name: streamName,
        display: localVideo,
        ...
    }).on(STREAM_STATUS.PUBLISHING, function (stream) {
        addVideoTrackEndedListener(localVideo, stream);
        setStatus("#publishStatus", STREAM_STATUS.PUBLISHING);
        onPublishing(stream);
        ...
    }).publish();
```

5. Remove event handler when stopping a stream

```
function onPublishing(stream) {
    $("#publishBtn").text("Stop").off('click').click(function () {
        $(this).prop('disabled', true);
        removeVideoTrackEndedListener(localVideo);
        stream.stop();
    }).prop('disabled', false);
    $("#publishInfo").text("");
}
```

Videotrack activity checking on server side

Videotrack activity checking for streams published on server is enabled with the following parameters inflashphoner.propertiesfile

```
rtp_activity_video=true
```

In this case, if there is no video in stream, its publishing will be stopped after 60 seconds.

Video only stream publishing with constraints

In some cases, video only stream should be published while microphone is busy, for example video is published while voice phone call. To prevent browser access request to microphone, set the constraints for video on;ly publishing:

```
    session.createStream({
        name: streamName,
        display: localVideo,
        constraints: {video: true, audio: false}
        ...
    }).publish();
```

Audio only stream publishing

In most cases, it is enough to set the constraints to publish audio only stream:

```
    session.createStream({
        name: streamName,
        display: localVideo,
        constraints: {video: false, audio: true}
        ...
    }).publish();
```

Audio only stream publishing in Safari browser

When audio only stream is published from iOS Safari browser with constraints, browser does not send audio packets. To workaround this, a stream should be published with video, then video should be muted:

```
    session.createStream({
        name: streamName,
        display: localVideo,
        constraints: {video: true, audio: true}
        ...
    }).on(STREAM_STATUS.PUBLISHING, function (stream) {
        stream.muteVideo();
        ...
    }).publish();
```

In this case, iOS Safari browser will send emply video packets (blank screen) and audio packets.

Disable resolution constraints normalization in Safari browser

By default, WebSDK normalizes stream publishing resolution constraints set in Safari browser. In this case, if width or height is not set, or equal to 0, then the picture resolution is forced to 320x240 or 640x480. Since WebSDK build0.5.28.2753.109(hash149855cc050bf7512817104fd0104e9cce760ac4), it is possible to disable normalization and pass resolution constarints to the browser as is. for example:

```
publishStream = session.createStream({
    ...
    disableConstraintsNormalization: true,
    constraints {
        video: {
            width: {ideal: 1024},
            height: {ideal: 768}
        },
        audio: true
    }
}).on(STREAM_STATUS.PUBLISHING, function (publishStream) {
    ...
});
publishStream.publish();
```

H264 encoding profiles exclusion

Since build5.2.620some H264 encoding profiles can be excluded from remote SDP which is sent by server to browser. The profiles to exckude should be listed in the following parameter

```
webrtc_sdp_h264_exclude_profiles=4d,64
```

In this case, Main (4d) and High (64) profiles will be excluded, but Baseline (42) still remains:

```
a=rtpmap:102 H264/90000
a=fmtp:102 level-asymmetry-allowed=1;packetization-mode=1;profile-level-id=42001f
a=rtpmap:125 H264/90000
a=fmtp:125 level-asymmetry-allowed=1;packetization-mode=1;profile-level-id=42e01f
a=rtpmap:127 H264/90000
a=fmtp:127 level-asymmetry-allowed=1;packetization-mode=0;profile-level-id=42001f
a=rtpmap:108 H264/90000
a=fmtp:108 level-asymmetry-allowed=1;packetization-mode=0;profile-level-id=42e01f
```

This setting may be useful if some browser encodes B-frames for example using high profiles while hardware acceleration is enabled.

By default, no profiles will be excluded from SDP if they are supported by browser

```
webrtc_sdp_h264_exclude_profiles=
```

Content type management while publishing from Chromium based browser

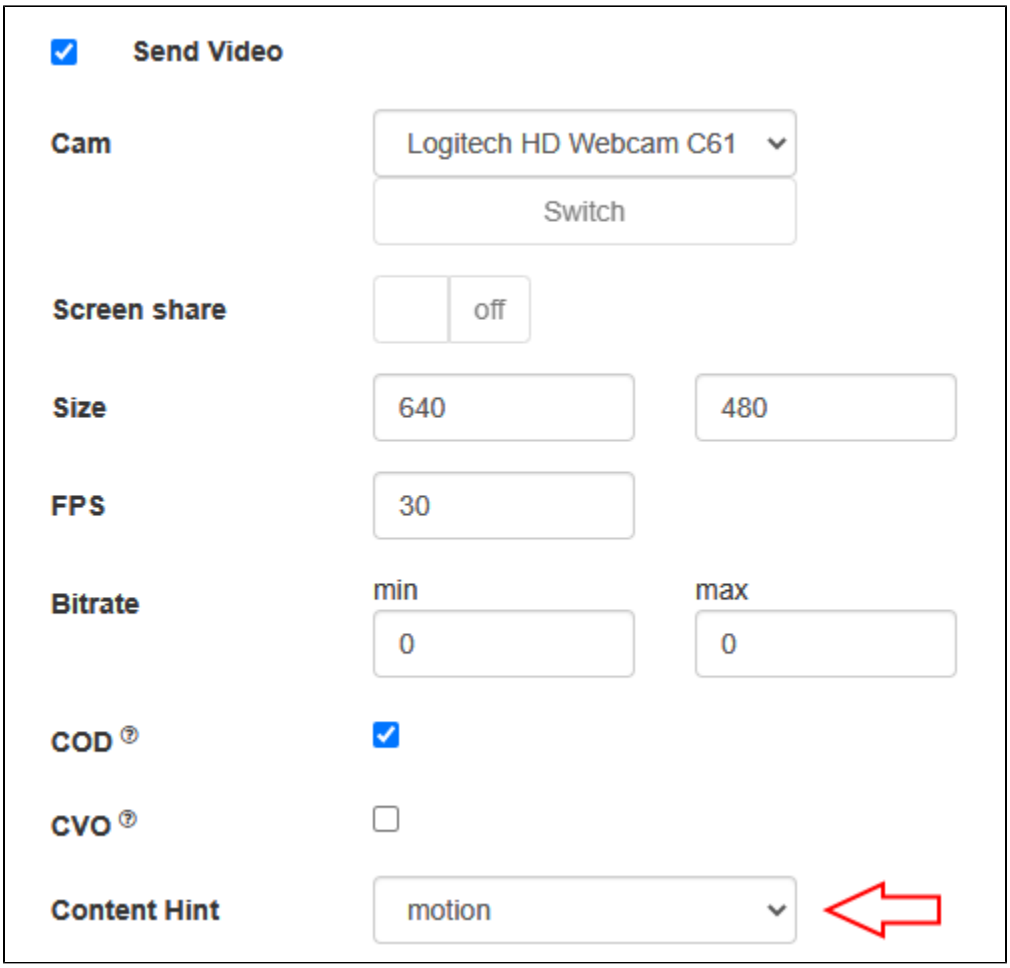
In some cases, most browsers based on Chromium 91 agressively assess a publishing channel quality and drop publishing resolution lower than set in constraints, even if channel is enough to publish 720p or 1080p stream. To workaround this behaviour, since WebSDK build2.0.180videoContentHint option was added:

```
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false,
        videoContentHint: "detail"
        ...
    }).publish();
```

By default, this option is set to`detail` and forces browsers to keep the publishing resolution as set in constraints. However, browser can drop FPS in this case when publishing stream from som USB web cameras. If FPS should be kept mo matter to resolution, the option should be set to`motion`

```
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false,
        videoContentHint: "motion"
        ...
    }).publish();
```

Since WebSDK build 2.0.204 `videoContentHint` selection is available in Media Device example



FPSmanagement in Firefox browser

By default, Firefox is publishing video with maximum FPS shown by web camera driver fo requested resolution. This value is 30 FPS for most of modern web cameras. Publishing FPS can be defined more strictly if necessary. To do this, disable constraints normalization:

```
session.createStream({
    ...
    disableConstraintsNormalization: true,
    constraints: {
        video: {
            width: 640,
            height: 360,
            frameRate: { max: 15 }
        },
        audio: true
    }
}).on(STREAM_STATUS.PUBLISHING, function (publishStream) {
    ...
}).publish();
```

Note that Foirefox can exclude the camera from the list while requesting camera and microphone access if camera driver does not provide a required combination of resolution and FPS. Also, Firefox can change a publishing resolution if there is only one resolution with required FPS in camera driver response.

Stereo audio publishing in browser

Audio bitrate should be more than 60000 bps to publish stereo in Opus codec from browser. This can be done by setting Opus codec parameters on client side

```
session.createStream({
    name: streamName,
    display: remoteVideo,
    constraints: {
        audio: {
            bitrate: 64000
        },
        ...
    }
    ...
}).publish();
```

oron server side

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

In this case, Firefox browser publishes stereo audio without additional setup.

Stereo audio publishing in Chrome based browsers

A certain client setup is required to publish stereo audio from Chrome. Thre is two ways to set this up depending on client implementation

Using Web SDK

If Web SDK is used in project, it is necessary to set the fiollowing constraint option:

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

Using Websocket API

If only Websocket API is used in project, it is necessary to disable echo cancellation

```
    var constraints = {
        audio: {
            echoCancellation: false,
            googEchoCancellation: false
        },
        ...
    };
    ...
    navigator.getUserMedia(constraints, function (stream) {
        ...
    }, reject);
```

If echo cancellation is enabled, Chrome will publish mono audio even if stereo is set in Opus codec options.

Known issues

1.If the web app is inside an iframe element, publishing of the video stream may fail.

Symptoms: IceServer errors in the browser console.

Solution: put the app out of iframe to an individual page.

2. If publishing of the stream goes under Windows 10 or Windows 8 and hardware acceleration is enabled in the Google Chrome browser, bitrate problems are possible.

Symptoms: low quality of the video, muddy picture, bitrate shown in chrome://webrtc-internals is less than 100 kbps.

Solution: turn off hardware acceleration in the browser, switch the browser of the server to use the VP8 codec.

3. Stream publishing with local video playback in Delight Player does not work in MS Edge

Symptoms: when stream is published in MS Edge, local video playback does not start in Delight Player

Solution: use another browser to publish a stream

4. In some cases microphone does not work in Chrome browser while publishing WebRTC stream.

Symptoms: michrophone does not work while publishing WebRTC stream, including example web applications out of the box

Solution: turn off gain node creation in Chrome browser using WebSDK initialization parameter`createMicGainNode: false`

```
        Flashphoner.init({
            flashMediaProviderSwfLocation: '../../../../media-provider.swf',
            createMicGainNode: false
        });
```

Note that microphone gain setting will not work in this case.

5. G722 codec does not work in Edge browser

Symptoms: WebRTC stream with G722 audio does not publish in Edge browser

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 client 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. iOS Safari 12.1 does not send video frames when picture with certain resolution is published

Symptoms: when H264 stream is published from iOS Safari 12.1, subscriber receives audio packets only, publishers WebRTC statistics also shows audio frames only

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.

8. Stream from built-in camera cannot be published in iOS Safari 12 and MacOS Safari 12 in some resolutions

Symptoms: stream publishing from browser fails with error in console

```
Overconstrained error: width
```

Solution:

a) use only resolutions which passesWebRTC Camera Resolutiontest

b) use external camera supporting resolutions as needed in MacOS Safari

c) disable resolution constraints normalization and set width and height as ideal, seeexample above.

9. Non-latin characters in stream name should be encoded

Symptoms: non-latin characters in stream name are replaced to questionmarks on server side

Solution: use JavaScript functionencodeURIComponent() while publishing stream

```
    var streamName = encodeURIComponent($('#publishStream').val());
    session.createStream({
        name: streamName,
        display: localVideo,
        cacheLocalResources: true,
        receiveVideo: false,
        receiveAudio: false
    ...
    }).publish();
```

10. In some cases, server can not parse H264 stream encoded with CABAC

Symptoms: WebRTC H264 stream publishing does not work

Solution:

a) use lower encoding profile

b) 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.

11. When playing WebRTC broadcast in Firefox onmacOS Catalina, displayed system warning and block on playing H264 stream.

Symptoms: when playing WebRTC broadcast in Firefox on macOS Catalina,displayed system warning *"libgmpopenh264.dylib" can`t be opened because it is from an identified developer"*and block of playing H264 stream.

Solution:Firefox uses a third-party library unsigned by the developer to work with H264. In accordance with macOS Catalina security policies, this is prohibited. To add an exception, go to *System Preferences > Security & Privacy > General > Allow apps downloaded from > App Store and identified developers> "libgmpopenh264.dylib" was blocked from opening because it is not from an identified developer" >* tap *Open Anyway*.

12. Since build5.2.672,the parameter

```
ice_keep_alive_enabled=true
```

is not used. ICE keep alive timeout is started automatically if WCS starts sending STUN keep alives first, for example while incoming SIP call or while pushing WebRTC stream to another server

13. MacOS Safari 14.0.2 (MacOS 11) does not publish stream from MacBook FaceTimeHD camera with aspect ratio 4:3

Symptoms: In the examples Two Way Streaming, Stream Recording etc stream publishing starts, then browser stops sending video packets in 10 seconds, the black screen is shown on playres side, publishing fails due to zero video traffic

Solution:

a)Publish stream with aspect ratio 16:9 (for example, 320x180, 640x360 etc)

```
publishStream = session.createStream({
    ...
    constraints: {
        video: {
            width: 640,
            height: 360
        },
        audio: true
    }
}).on(STREAM_STATUS.PUBLISHING, function (publishStream) {
    ...
});
publishStream.publish();
```

b) Update Web SDK to build0.5.28.2753.153, where default resolution for Safari browser is adopted to 16:9

c) Update MacOS to build 11.3.1, Safari to build14.1 (16611.1.21.161.6)

14. Streams published on Origin server in CDN may not be played from Edge if some H264 encoding profiles are excluded

Symptoms: WebRTC H264 stream published on Origin is playing as audio only from Edge with VP* codec shown in stream metrics

Solution: if some H264 encoding profiles are excluded on Origin

```
webrtc_sdp_h264_exclude_profiles=4d,64
```

the allowed H264 profiles should be explicitly set on Egde with the following parameter

```
profiles=42e01f
```

15. MacOS Safari 14.0.* stops sending video packets after video in muted and then unmuted due toWebkit bug

Symptoms: when muteVideo() then unmuteVideo() are applied, the publishing stops after a minute with Failed by Video RTP activity error

Solution: update MacOS to build 11.3.1, Safari to build14.1 (16611.1.21.161.6), in this build theWebkit bugseems to be fixed, the problem is not reproducing

16. When publishing from Google Pixel 3/3XL,the picture is strongly distorted in some resolutions

Symptoms: a local video is displaying normally, but is playing with a strong distortion (transverse stripes)

Solution: avoid the following resolutions while publishing stream from Google Pixel 3/3XL:

- 160x120
- 1920x1080

17. iOS Safari 15.1 requires from another side to enable image orientation extension when publishing H264 stream

Symptoms: webpage crashes in iOS Safari 15.1 when stream publishing is started (Webkit bugs https://bugs.webkit.org/show_bug.cgi?id=232381 and https://bugs.webkit.org/show_bug.cgi?id=231505)

Solution:

a) enable image orientation extension support on client side in iOS Safari

```
session.createStream({
    name: streamName,
    ...
    cvoExtension: true
}).publish();
```

and in WCS builds before5.2.1074disable RTP bundle support

```
rtp_bundle=false
```

Since WCS build5.2.1074 RTP bundle may not be disabled

b) use VP8 to publish a stream

```
session.createStream({
    name: streamName,
    ...
    stripCodecs: "H264"
}).publish();
```