

Media Devices

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Example of streamer with access to media devices

This streamer can be used to publish or playback WebRTC streams on Web Call Server and allows to select media devices and parameters for the published video

- camera
- microphone
- FPS (Frames Per Second)
- resolution (width, height)

On the screenshot below a stream is being published from the client.


Media Devices

Video stats
Codec: H264
Codec Rate: 90000
Packets Sent: 1185
Bytes Sent: 841051
Fir Count: 0
Pli Count: 2
Nack Count: 0
Height: 360
Width: 640
Bitrate: 891392

Audio stats
Codec: opus
Codec Rate: 48000
Packets Sent: 699
Bytes Sent: 112699
Nack Count: 0
Bitrate: 64496

Connection

Local




640x360

test Stop

PUBLISHING

Player



640x360

test Stop

PLAYING

ws://localhost:8080 Disconnect

Timeout 1000 msec

ESTABLISHED

Video stats
Codec: H264
Codec Rate: 90000
Packets Lost: 0
Packets Received: 518
Bytes Received: 541528
Jitter Buffer Emitted Count: 188
Fir Count: 0
Pli Count: 4
Nack Count: 0
Height: 360
Width: 640
Bitrate: 879744

Audio stats
Codec: opus
Codec Rate: 48000
Packets Lost: 0
Packets Discarded: 0
Packets Received: 346
Bytes Received: 55866
Jitter Buffer Emitted Count: 330240
Audio Level: 0.022462
Bitrate: 64704

Connection

Two video elements are displayed on the page

- 'Local' - video from the camera
- 'Player' - the video received from the server

Code of the example

The path to the source code of the example on WCS server is:

`/usr/local/FlashphonerWebCallServer/client/examples/demo/streaming/media_devices_manager`

manager.css - file with styles
media_device_manager.html - page of the streamer
manager.js - script providing functionality for the streamer

This example can be tested using the following address:

`https://host:8888/client/examples/demo/streaming/media_devices_manager/media_device_manager.html`

Here host is the address of the WCS server.

Analyzing the code

To analyze the code, let's take the version of file manager.js with hash ecbadc3, which is available [here](#) and can be downloaded with corresponding build [2.0.212](#).

1. Initialization of the API.

Flashphoner.init() [code](#)

```
Flashphoner.init({
  screenSharingExtensionId: extensionId,
  mediaProvidersReadyCallback: function (mediaProviders) {
    if (Flashphoner.isUsingTemasys()) {
      $("#audioInputForm").hide();
      $("#videoInputForm").hide();
    }
  }
})
```

2. List available input media devices.

Flashphoner.getMediaDevices() [code](#)

When input media devices are listed, drop-down lists of microphones and cameras on client page are filled.

```
Flashphoner.getMediaDevices(null, true).then(function (list) {
  list.audio.forEach(function (device) {
    ...
  });
  list.video.forEach(function (device) {
    ...
  });
  ...
}).catch(function (error) {
  $("#notifyFlash").text("Failed to get media devices");
});
```

3. List available output media devices

Flashphoner.getMediaDevices() [code](#)

When output media devices are listed, drop-down lists of speakers and headphones on client page are filled.

```
Flashphoner.getMediaDevices(null, true, MEDIA_DEVICE_KIND.OUTPUT).then(function (list) {
  list.audio.forEach(function (device) {
    ...
  });
  ...
}).catch(function (error) {
  $("#notifyFlash").text("Failed to get media devices");
});
```

4. Get audio and video publishing constraints from client page

getConstraints() [code](#)

Publishing sources:

- camera (sendVideo)
- microphone (sendAudio)

```
constraints = {
  audio: $("#sendAudio").is(':checked'),
  video: $("#sendVideo").is(':checked'),
};
```

Audio constraints:

- microphone choice (deviceId)
- error correction for Opus codec (fec)

- stereo mode (stereo)
- audio bitrate (bitrate)

```
if (constraints.audio) {
    constraints.audio = {
        deviceId: $('#audioInput').val()
    };
    if ($("#fec").is(':checked'))
        constraints.audio.fec = $("#fec").is(':checked');
    if ($("#sendStereoAudio").is(':checked'))
        constraints.audio.stereo = ($("#sendStereoAudio").is(':checked'));
    if (parseInt($('#sendAudioBitrate').val()) > 0)
        constraints.audio.bitrate = parseInt($('#sendAudioBitrate').val());
}
```

Video constraints:

- camera choose (deviceId)
- publishing video size (width, height)
- minimal and maximal video bitrate (minBitrate, maxBitrate)
- FPS (frameRate)

```
constraints.video = {
    deviceId: {exact: $('#videoInput').val()},
    width: parseInt($('#sendWidth').val()),
    height: parseInt($('#sendHeight').val())
};
if (Browser.isSafariWebRTC() && Browser.isiOS() && Flashponer.getMediaProviders()[0] === "WebRTC")
{
    constraints.video.deviceId = {exact: $('#videoInput').val()};
}
if (parseInt($('#sendVideoMinBitrate').val()) > 0)
    constraints.video.minBitrate = parseInt($('#sendVideoMinBitrate').val());
if (parseInt($('#sendVideoMaxBitrate').val()) > 0)
    constraints.video.maxBitrate = parseInt($('#sendVideoMaxBitrate').val());
if (parseInt($('#fps').val()) > 0)
    constraints.video.frameRate = parseInt($('#fps').val());
```

5. Get access to media devices for local test

Flashponer.getMediaAccess() [code](#)

Audio and video constraints and <div>-element to display captured video are passed to the method.

```
Flashponer.getMediaAccess(getConstraints(), localVideo).then(function (disp) {
    $("#testBtn").text("Release").off('click').click(function () {
        $(this).prop('disabled', true);
        stopTest();
    }).prop('disabled', false);
    ...
    testStarted = true;
}).catch(function (error) {
    $("#testBtn").prop('disabled', false);
    testStarted = false;
});
```

6. Connecting to the server

Flashponer.createSession() [code](#)

```
Flashponer.createSession({urlServer: url, timeout: tm}).on(SESSION_STATUS.ESTABLISHED, function (session) {
    ...
}).on(SESSION_STATUS.DISCONNECTED, function () {
    ...
}).on(SESSION_STATUS.FAILED, function () {
    ...
});
```

7. Receiving the event confirming successful connection

ConnectionStatusEvent ESTABLISHED [code](#)

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

8. Stream publishing

session.createStream(), publishStream.publish() [code](#)

```
publishStream = session.createStream({
    name: streamName,
    display: localVideo,
    cacheLocalResources: true,
    constraints: constraints,
    mediaConnectionConstraints: mediaConnectionConstraints,
    sdpHook: rewriteSdp,
    transport: transportInput,
    cvoExtension: cvo,
    stripCodecs: strippedCodecs,
    videoContentHint: contentHint
    ...
});
publishStream.publish();
```

9. Receiving the event confirming successful streaming

StreamStatusEvent PUBLISHING [code](#)

```
publishStream = session.createStream({
    ...
}).on(STREAM_STATUS.PUBLISHING, function (stream) {
    $("#testBtn").prop('disabled', true);
    var video = document.getElementById(stream.id());
    //resize local if resolution is available
    if (video.videoWidth > 0 && video.videoHeight > 0) {
        resizeLocalVideo({target: video});
    }
    enablePublishToggles(true);
    if ($("#muteVideoToggle").is(":checked")) {
        muteVideo();
    }
    if ($("#muteAudioToggle").is(":checked")) {
        muteAudio();
    }
    //remove resize listener in case this video was cached earlier
    video.removeEventListener('resize', resizeLocalVideo);
    video.addEventListener('resize', resizeLocalVideo);
    publishStream.setMicrophoneGain(currentGainValue);
    setStatus("#publishStatus", STREAM_STATUS.PUBLISHING);
    onPublishing(stream);
}).on(STREAM_STATUS.UNPUBLISHED, function () {
    ...
}).on(STREAM_STATUS.FAILED, function () {
    ...
});
publishStream.publish();
```

10. Stream playback

session.createStream(), previewStream.play() [code](#)

```

previewStream = session.createStream({
    name: streamName,
    display: remoteVideo,
    constraints: constraints,
    transport: transportOutput,
    stripCodecs: strippedCodecs
    ...
});
previewStream.play();

```

11.Receiving the event confirming successful playback

StreamStatusEvent PLAYING [code](#)

```

previewStream = session.createStream({
    ...
}).on(STREAM_STATUS.PLAYING, function (stream) {
    playConnectionQualityStat.connectionQualityUpdateTimestamp = new Date().valueOf();
    setStatus("#playStatus", stream.status());
    onPlaying(stream);
    document.getElementById(stream.id()).addEventListener('resize', function (event) {
        $("#playResolution").text(event.target.videoWidth + "x" + event.target.videoHeight);
        resizeVideo(event.target);
    });
    //wait for incoming stream
    if (Flashphoner.getMediaProviders()[0] == "WebRTC") {
        setTimeout(function () {
            if (Browser.isChrome()) {
                detectSpeechChrome(stream);
            } else {
                detectSpeech(stream);
            }
        }, 3000);
    }
    ...
});
previewStream.play();

```

12. Stop stream playback

stream.stop() [code](#)

```

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

```

13.Receiving the event confirming successful playback stop

StreamStatusEvent STOPPED [code](#)

```

previewStream = session.createStream({
    ...
}).on(STREAM_STATUS.STOPPED, function () {
    setStatus("#playStatus", STREAM_STATUS.STOPPED);
    onStopped();
    ...
});
previewStream.play();

```

14. Stop stream publishing

stream.stop() [code](#)

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

15. Receiving the event confirming successful publishing stop

StreamStatusEvent UNPUBLISHED [code](#)

```
publishStream = session.createStream({
    ...
}).on(STREAM_STATUS.UNPUBLISHED, function () {
    setStatus("#publishStatus", STREAM_STATUS.UNPUBLISHED);
    onUnpublished();
    ...
});
publishStream.publish();
```

16. Mute publisher audio

stream.muteAudio() [code](#):

```
function muteAudio() {
    if (publishStream) {
        publishStream.muteAudio();
    }
}
```

17. Mute publisher video

stream.muteVideo() [code](#):

```
function muteVideo() {
    if (publishStream) {
        publishStream.muteVideo();
    }
}
```

18. Show WebRTC stream publishing statistics

stream.getStats() [code](#):

```

publishStream.getStats(function (stats) {
  if (stats && stats.outboundStream) {
    if (stats.outboundStream.video) {
      showStat(stats.outboundStream.video, "outVideoStat");
      let vBitrate = (stats.outboundStream.video.bytesSent - videoBytesSent) * 8;
      if ($('#outVideoStatBitrate').length == 0) {
        let html = "<div>Bitrate: " + "<span id='outVideoStatBitrate' style='font-weight:
normal'>" + vBitrate + "</span>" + "</div>";
        $('#outVideoStat').append(html);
      } else {
        $('#outVideoStatBitrate').text(vBitrate);
      }
      videoBytesSent = stats.outboundStream.video.bytesSent;
      ...
    }

    if (stats.outboundStream.audio) {
      showStat(stats.outboundStream.audio, "outAudioStat");
      let aBitrate = (stats.outboundStream.audio.bytesSent - audioBytesSent) * 8;
      if ($('#outAudioStatBitrate').length == 0) {
        let html = "<div>Bitrate: " + "<span id='outAudioStatBitrate' style='font-weight:
normal'>" + aBitrate + "</span>" + "</div>";
        $('#outAudioStat').append(html);
      } else {
        $('#outAudioStatBitrate').text(aBitrate);
      }
      audioBytesSent = stats.outboundStream.audio.bytesSent;
    }
  }
  ...
});

```

19. Show WebRTC stream playback statistics

stream.getStats() [code](#):

```

previewStream.getStats(function (stats) {
  if (stats && stats.inboundStream) {
    if (stats.inboundStream.video) {
      showStat(stats.inboundStream.video, "inVideoStat");
      let vBitrate = (stats.inboundStream.video.bytesReceived - videoBytesReceived) * 8;
      if ($('#inVideoStatBitrate').length == 0) {
        let html = "<div>Bitrate: " + "<span id='inVideoStatBitrate' style='font-weight:
normal'>" + vBitrate + "</span>" + "</div>";
        $('#inVideoStat').append(html);
      } else {
        $('#inVideoStatBitrate').text(vBitrate);
      }
      videoBytesReceived = stats.inboundStream.video.bytesReceived;
      ...
    }

    if (stats.inboundStream.audio) {
      showStat(stats.inboundStream.audio, "inAudioStat");
      let aBitrate = (stats.inboundStream.audio.bytesReceived - audioBytesReceived) * 8;
      if ($('#inAudioStatBitrate').length == 0) {
        let html = "<div style='font-weight: bold'>Bitrate: " + "<span id='inAudioStatBitrate'
style='font-weight: normal'>" + aBitrate + "</span>" + "</div>";
        $('#inAudioStat').append(html);
      } else {
        $('#inAudioStatBitrate').text(aBitrate);
      }
      audioBytesReceived = stats.inboundStream.audio.bytesReceived;
    }
  }
  ...
});

```

20. Speech detection using ScriptProcessor interface (any browser except Chrome)

audioContext.createMediaStreamSource(), audioContext.createScriptProcessor() [code](#)

```
function detectSpeech(stream, level, latency) {
    var mediaStream = document.getElementById(stream.id()).srcObject;
    var source = audioContext.createMediaStreamSource(mediaStream);
    var processor = audioContext.createScriptProcessor(512);
    processor.onaudioprocess = handleAudio;
    processor.connect(audioContext.destination);
    processor.clipping = false;
    processor.lastClip = 0;
    // threshold
    processor.threshold = level || 0.10;
    processor.latency = latency || 750;

    processor.isSpeech =
        function () {
            if (!this.clipping) return false;
            if ((this.lastClip + this.latency) < window.performance.now()) this.clipping = false;
            return this.clipping;
        };

    source.connect(processor);

    // Check speech every 500 ms
    speechIntervalID = setInterval(function () {
        if (processor.isSpeech()) {
            $("#talking").css('background-color', 'green');
        } else {
            $("#talking").css('background-color', 'red');
        }
    }, 500);
}
```

Audio data handler [code](#)

```
function handleAudio(event) {
    var buf = event.inputBuffer.getChannelData(0);
    var bufLength = buf.length;
    var x;
    for (var i = 0; i < bufLength; i++) {
        x = buf[i];
        if (Math.abs(x) >= this.threshold) {
            this.clipping = true;
            this.lastClip = window.performance.now();
        }
    }
}
```

21. Speech detection using incoming audio WebRTC statistics in Chrome browser

stream.getStats() [code](#)


```

function detectSpeechChrome(stream, level, latency) {
  statSpeechDetector.threshold = level || 0.010;
  statSpeechDetector.latency = latency || 750;
  statSpeechDetector.clipping = false;
  statSpeechDetector.lastClip = 0;
  speechIntervalID = setInterval(function() {
    stream.getStats(function(stat) {
      let audioStats = stat.inboundStream.audio;
      if(!audioStats) {
        return;
      }
      // Using audioLevel WebRTC stats parameter
      if (audioStats.audioLevel >= statSpeechDetector.threshold) {
        statSpeechDetector.clipping = true;
        statSpeechDetector.lastClip = window.performance.now();
      }
      if ((statSpeechDetector.lastClip + statSpeechDetector.latency) < window.performance.now()) {
        statSpeechDetector.clipping = false;
      }
      if (statSpeechDetector.clipping) {
        $("#talking").css('background-color', 'green');
      } else {
        $("#talking").css('background-color', 'red');
      }
    });
  }, 500);
}

```