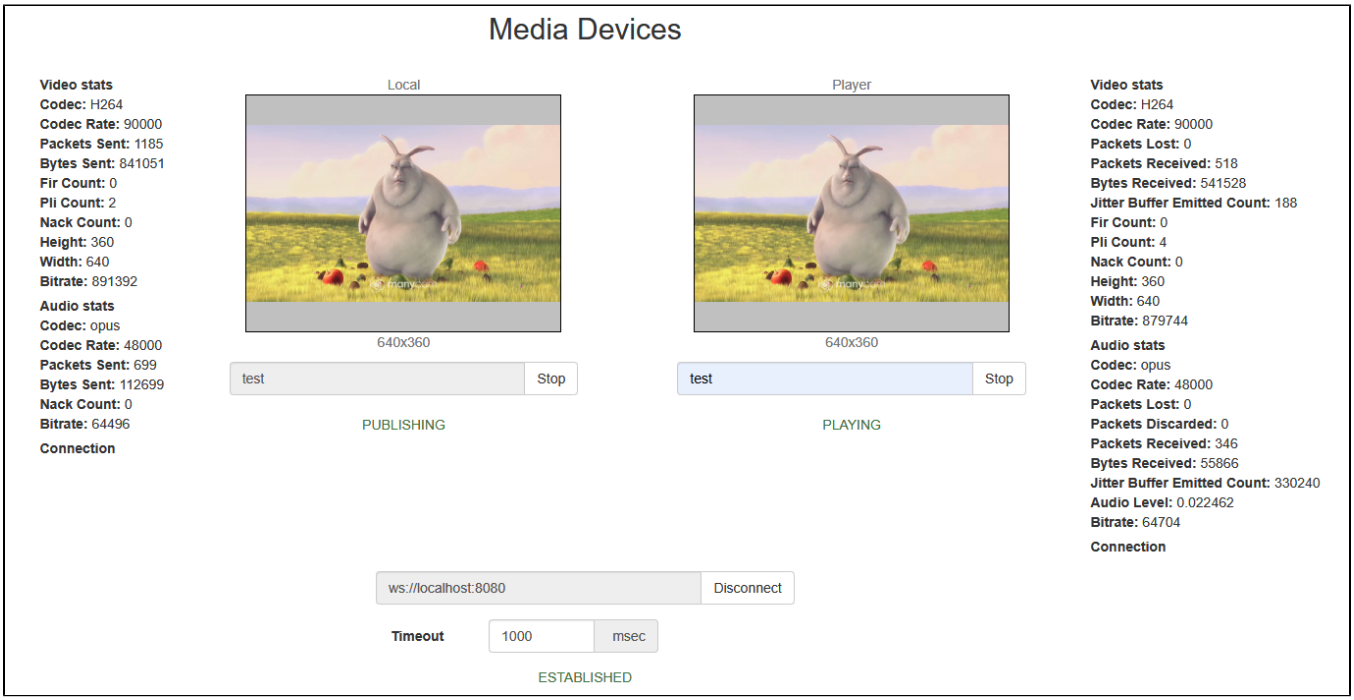
Media Devices

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.



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 whith 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 spakers 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 choise (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 choise (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() && Flashphoner.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

Flashphoner.getMediaAccess() code

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

```
    Flashphoner.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

Flashphoner.createSession() code

```
    Flashphoner.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 publishsing 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);
}
```