

Phone Video

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Example of web phone for video calls

Phone Video

Connection

WCS URL

ws://localhost:8080

SIP Login

10006

SIP Auth Name

10006

SIP Password

SIP Domain

sip.flashphoner.com

SIP Outbound Proxy

sip.flashphoner.com

SIP Port

5060

Register required

☒

REGISTERED

Disconnect



Video size

320

240

Video FPS

30

Mute Audio

off

Mute Video

off

SDP replace

0

with

0

Statistics Video

Bytes sent

439775

Packets sent

545

Statistics Audio

Bytes sent

88193

Packets sent

1379

10008

Hangup

Code of the example

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

/usr/local/FlashphonerWebCallServer/client/examples/demo/sip/phone-video

phone-video.css - file with styles
phone-video.html - page of the web phone
call-fieldset.html - form with fields required for connection
call-controls.html - HTML code for call controls
phone-video.js - script providing functionality for the web phone

This example can be tested using the following address:

https://host:8888/client/examples/demo/sip/phone-video/phone-video.html

Here host is the address of the WCS server.

Analyzing the code

To analyze the code, let's take the version of file phone-video.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();
```

2. Connection to server.

Flashphoner.createSession() [code](#)

Object with connection options is passed to the method

- urlServer - URL for WebSocket connection to WCS server
- sipOptions - object with parameters for SIP connection

```
var url = $('#urlServer').val();
var registerRequired = $("#sipRegisterRequired").is(':checked');

var sipOptions = {
    login: $("#sipLogin").val(),
    authenticationName: $("#sipAuthenticationName").val(),
    password: $("#sipPassword").val(),
    domain: $("#sipDomain").val(),
    outboundProxy: $("#sipOutboundProxy").val(),
    port: $("#sipPort").val(),
    registerRequired: registerRequired
};

var connectionOptions = {
    urlServer: url,
    sipOptions: sipOptions
};

//create session
console.log("Create new session with url " + url);
Flashphoner.createSession(connectionOptions).on(SESSION_STATUS.ESTABLISHED, function(session){
    ...
});
```

3.Receiving the event confirming successful connection

ConnectionStatusEvent ESTABLISHED [code](#)

```
Flashphoner.createSession(connectionOptions).on(SESSION_STATUS.ESTABLISHED, function(session){
    setStatus("#regStatus", SESSION_STATUS.ESTABLISHED);
    onConnected(session);
    if (!registerRequired) {
        disableOutgoing(false);
    }
}).on(SESSION_STATUS.REGISTERED, function(session){
    ...
}).on(SESSION_STATUS.DISCONNECTED, function(){
    ...
}).on(SESSION_STATUS.FAILED, function(){
    ...
}).on(SESSION_STATUS.INCOMING_CALL, function(call){
    ...
});
```

4.Receiving the event confirming successful registration on SIP server

ConnectionStatusEvent REGISTERED [code](#)

```
Flashphoner.createSession(connectionOptions).on(SESSION_STATUS.ESTABLISHED, function(session){
    ...
}).on(SESSION_STATUS.REGISTERED, function(session){
    setStatus("#regStatus", SESSION_STATUS.REGISTERED);
    onConnected(session);
    if (registerRequired) {
        disableOutgoing(false);
    }
}).on(SESSION_STATUS.DISCONNECTED, function(){
    ...
}).on(SESSION_STATUS.FAILED, function(){
    ...
}).on(SESSION_STATUS.INCOMING_CALL, function(call){
    ...
});
```

5.Receiving the event on incoming call

ConnectionStatusEvent INCOMING_CALL [code](#)

```
Flashphoner.createSession(connectionOptions).on(SESSION_STATUS.ESTABLISHED, function(session){
    ...
}).on(SESSION_STATUS.REGISTERED, function(session){
    ...
}).on(SESSION_STATUS.DISCONNECTED, function(){
    ...
}).on(SESSION_STATUS.FAILED, function(){
    ...
}).on(SESSION_STATUS.INCOMING_CALL, function(call){
    call.on(CALL_STATUS.RING, function(){
        ...
    });
    onIncomingCall(call);
});
```

6. Outgoing call.

session.createCall(), call.call() [code](#)

The following parameters are passed when call is created

- callee - callee SIP username
- visibleName - display name
- localVideoDisplay - <div> element, in which video from camera will be displayed
- remoteVideoDisplay - <div> element, in which video from the other party will be displayed

```
var outCall = session.createCall({
    callee: $("#callee").val(),
    visibleName: $("#sipLogin").val(),
    localVideoDisplay: localVideo,
    remoteVideoDisplay: remoteVideo,
    localVideoDisplay: localVideo,
    constraints: constraints,
    sdpHook: rewriteSdp,
    stripCodecs: "SILK"
    ...
});

outCall.call();
```

7. Answering incoming call.

call.answer() [code](#)

Object with answer options is passed to the method

- localVideoDisplay - <div> element, in which video from camera will be displayed
- remoteVideoDisplay - <div> element, in which video from the other party will be displayed

```

$("#answerBtn").off('click').click(function(){
    $(this).prop('disabled', true);
    inCall.answer({
        localVideoDisplay: localVideo,
        remoteVideoDisplay: remoteVideo,
        constraints: constraints,
        sdpHook: rewriteSdp,
        stripCodecs: "SILK"
    });
    showAnswered();
}).prop('disabled', false);

```

8. Outgoing call hangup.

call.hangup() [code](#)

```

$("#callBtn").text("Hangup").off('click').click(function(){
    $(this).prop('disabled', true);
    outCall.hangup();
}).prop('disabled', false);

```

9. Incoming call hangup

call.hangup() [code](#)

```

$("#hangupBtn").off('click').click(function(){
    $(this).prop('disabled', true);
    $("#answerBtn").prop('disabled', true);
    inCall.hangup();
}).prop('disabled', false);

```

10. Call hangup on session disconnection

call.hangup() [code](#)

```

function onConnected(session) {
    $("#connectBtn").text("Disconnect").off('click').click(function(){
        $(this).prop('disabled', true);
        if (currentCall) {
            showOutgoing();
            disableOutgoing(true);
            setStatus("#callStatus", "");
            currentCall.hangup();
        }
        session.disconnect();
    }).prop('disabled', false);
}

```

11. Mute/unmute

currentCall.muteAudio(), currentCall.muteAudio(), currentCall.muteVideo(), currentCall.unmuteVideo() [code](#)

```

// Mute audio in the call
function mute() {
    if (currentCall) {
        currentCall.muteAudio();
    }
}

// Unmute audio in the call
function unmute() {
    if (currentCall) {
        currentCall.unmuteAudio();
    }
}

// Mute video in the call
function muteVideo() {
    if (currentCall) {
        currentCall.muteVideo();
    }
}

// Unmute video in the call
function unmuteVideo() {
    if (currentCall) {
        currentCall.unmuteVideo();
    }
}

```

12. WebRTC statistics collection during the call

call.getStats() [code](#)

```

function loadStats() {
    if (currentCall) {
        // Stats should be collected for active calls only #WCS-3260
        let status = currentCall.status();
        if (status !== CALL_STATUS.ESTABLISHED && status !== CALL_STATUS.HOLD) {
            return;
        }
        currentCall.getStats(function (stats) {
            if (stats && stats.outboundStream) {
                if (stats.outboundStream.video) {
                    $('#videoStatBytesSent').text(stats.outboundStream.video.bytesSent);
                    $('#videoStatPacketsSent').text(stats.outboundStream.video.packetsSent);
                } else {
                    $('#videoStatBytesSent').text(0);
                    $('#videoStatPacketsSent').text(0);
                }

                if (stats.outboundStream.audio) {
                    $('#audioStatBytesSent').text(stats.outboundStream.audio.bytesSent);
                    $('#audioStatPacketsSent').text(stats.outboundStream.audio.packetsSent);
                } else {
                    $('#audioStatBytesSent').text(0);
                    $('#audioStatPacketsSent').text(0);
                }
            }
        });
    }
}

```