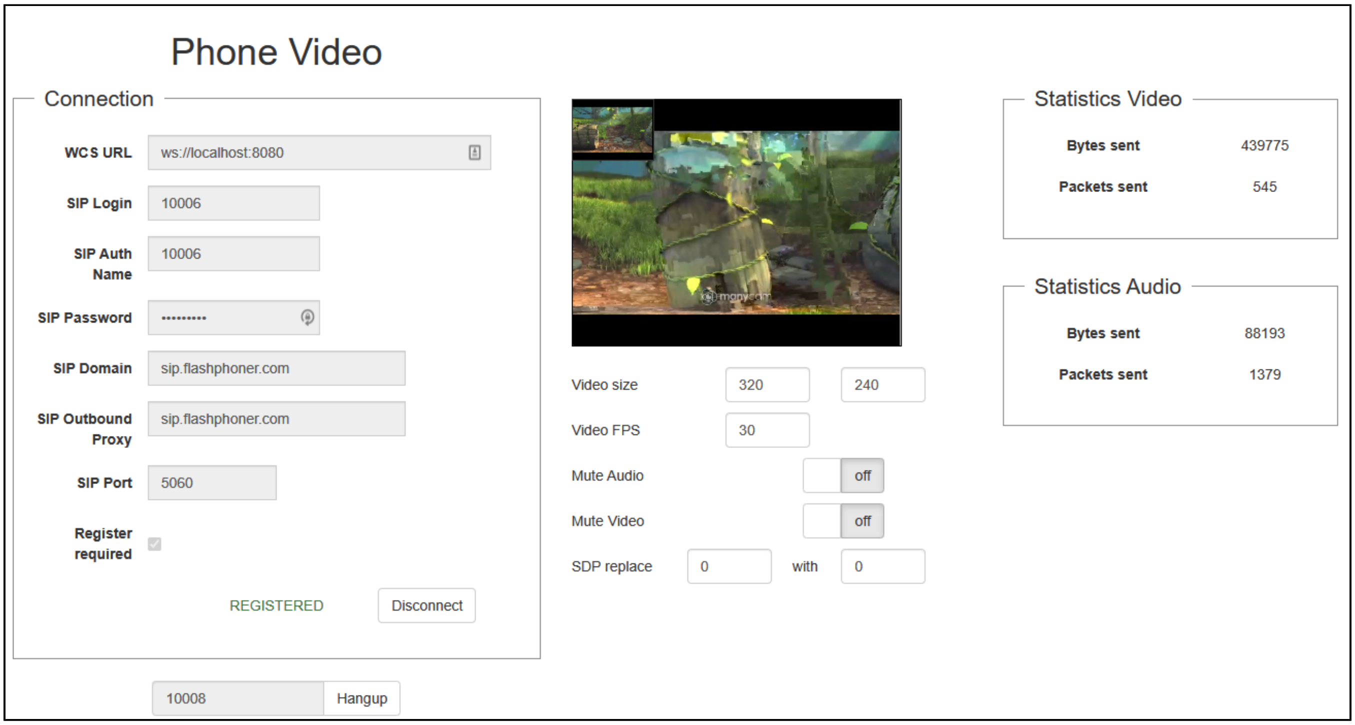
Phone Video

Example of web phone for video calls



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 shoukld 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);
                }
            }
        });
    }
}
```