A call to a mobile phone via the SIP server

A SIP call to a mobile phone is a special case of calls between a browser and a SIP device, when the SIP server either operates as a GSM/PSTN gateway itself or connects to one during the call.

Overview

Supported platforms and browsers

	Chrome	Firefox	Safari 11	Edge
Windows	+	+		+
Mac OS	+	+	+	
Android	+	+		
iOS	-	-	+	

Supported protocols

- WebRTC
- RTP
- SIP

Supported codecs

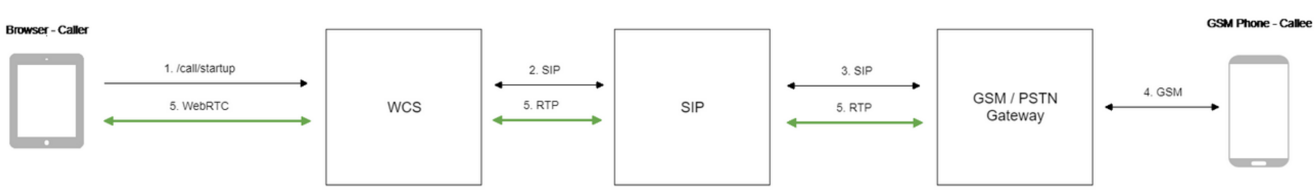
- H.264
- VP8
- G.711
- Speex
- G.729
- Opus

Supported SIP functions

- DTMF
- Holding a call
- Transferring a call

Management of SIP functions is performed using theREST API.

Operation flowchart



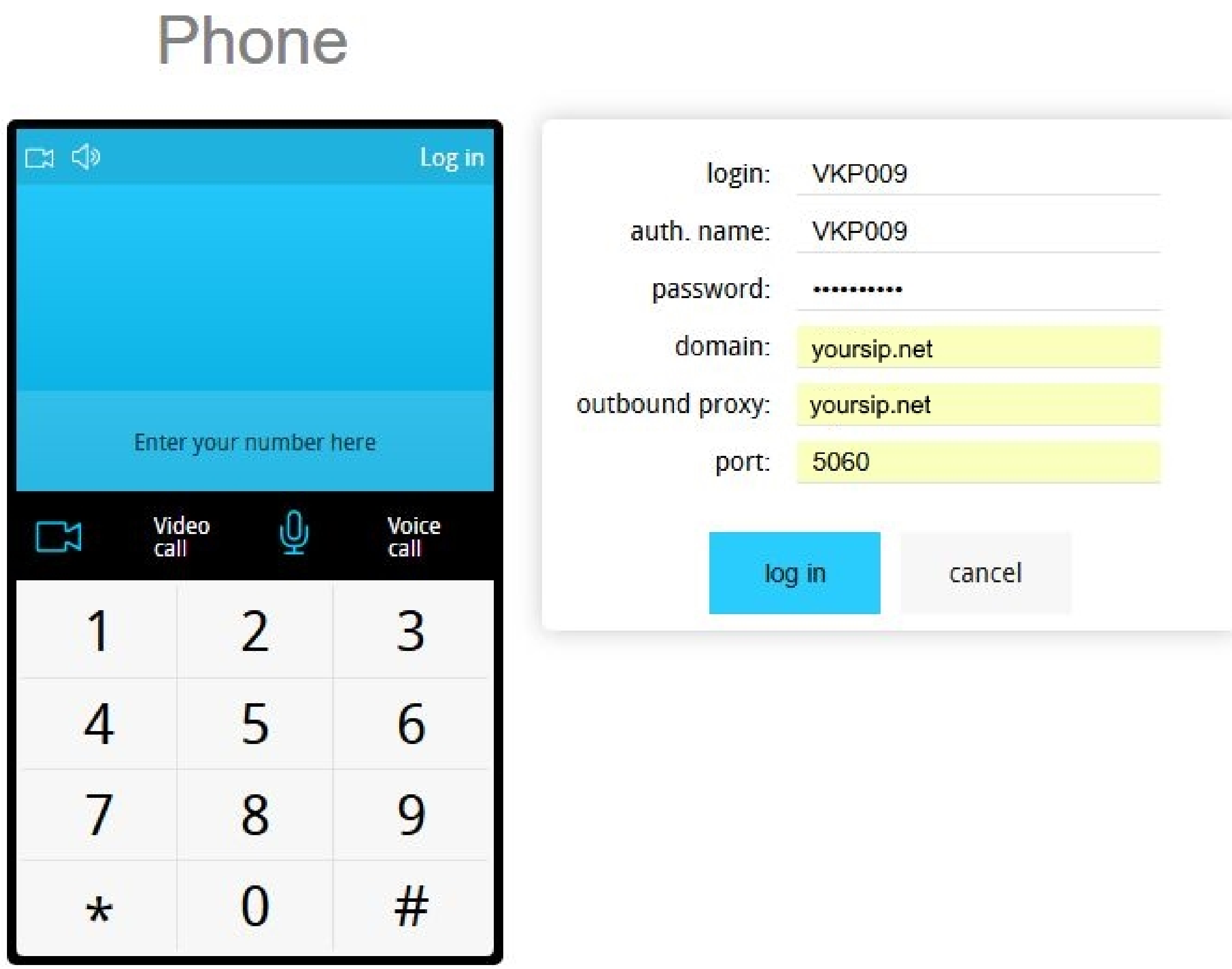
1. The browser begins a call with the /call/startup REST query
2. WCS connects to the SIP server
3. The SIP server connects to the GSM/PSTN gateway
4. The GSM/PSTN gateway connects to the mobile phone
5. The browser and the phone exchange audio streams

Quick manual on testing

1. For the test we use:

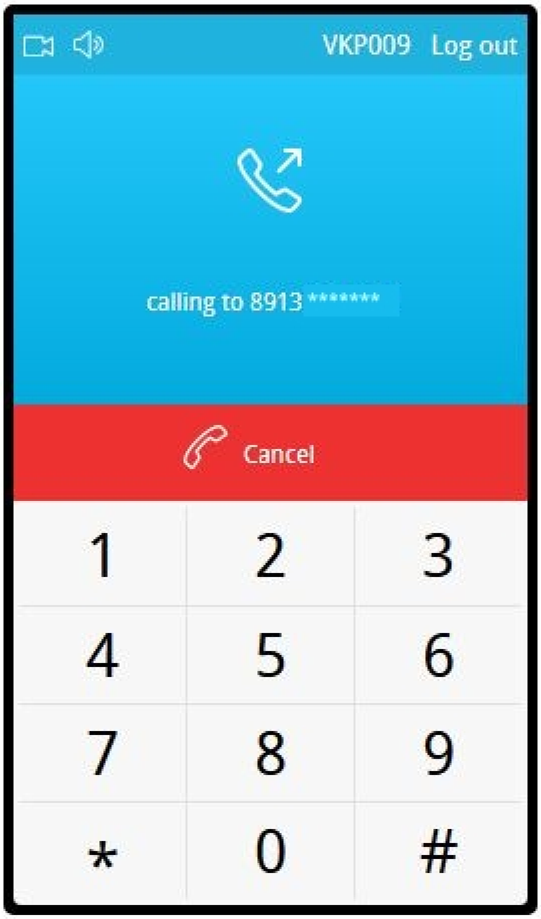
- two SIP accounts;
- thePhone UIweb application to make a call;
- a mobile phone to answer the call.

2. Open the Phone UI web application. Click Log in and enter the data of the SIP account:

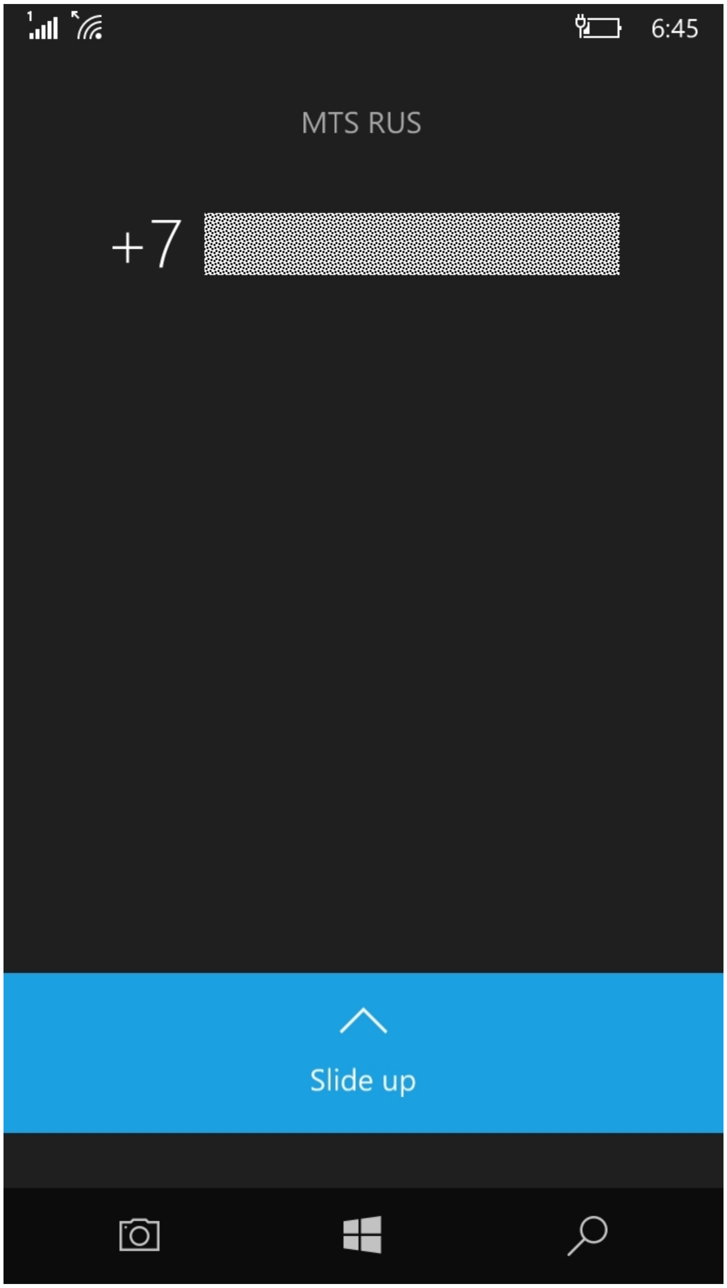


3. Enter the mobile phone number and click Voice call. Dialing starts:

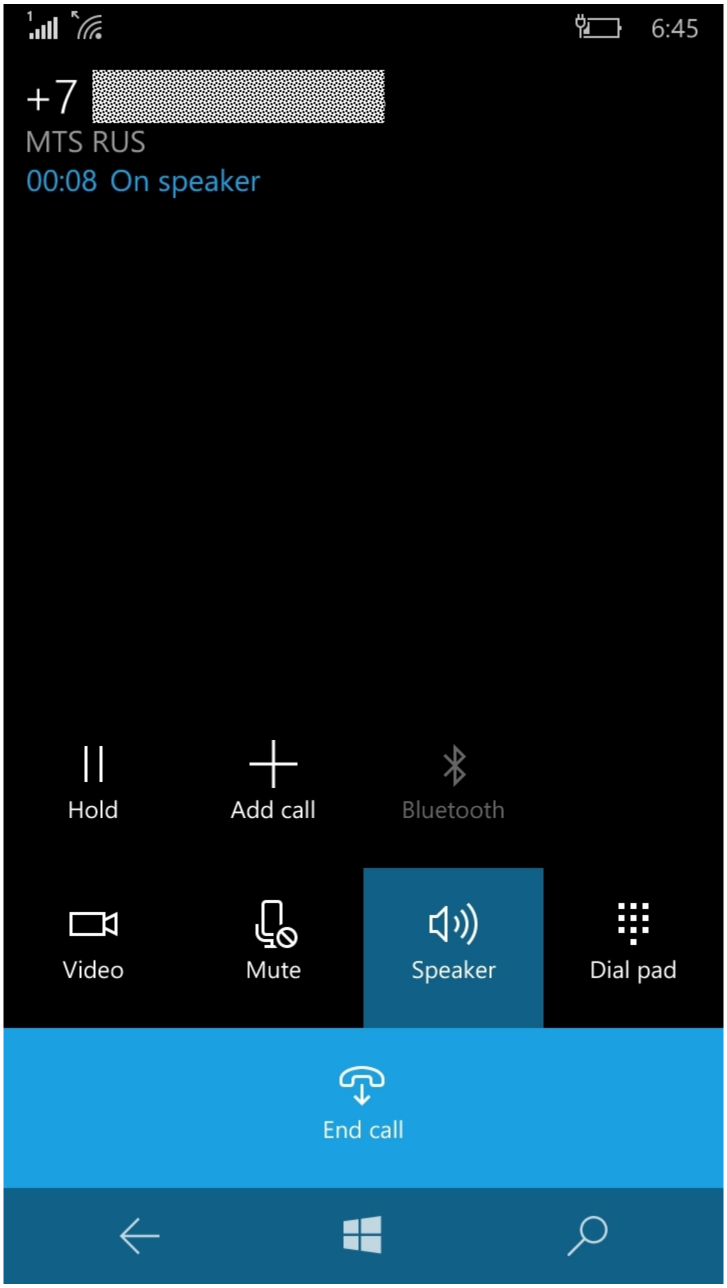
Phone



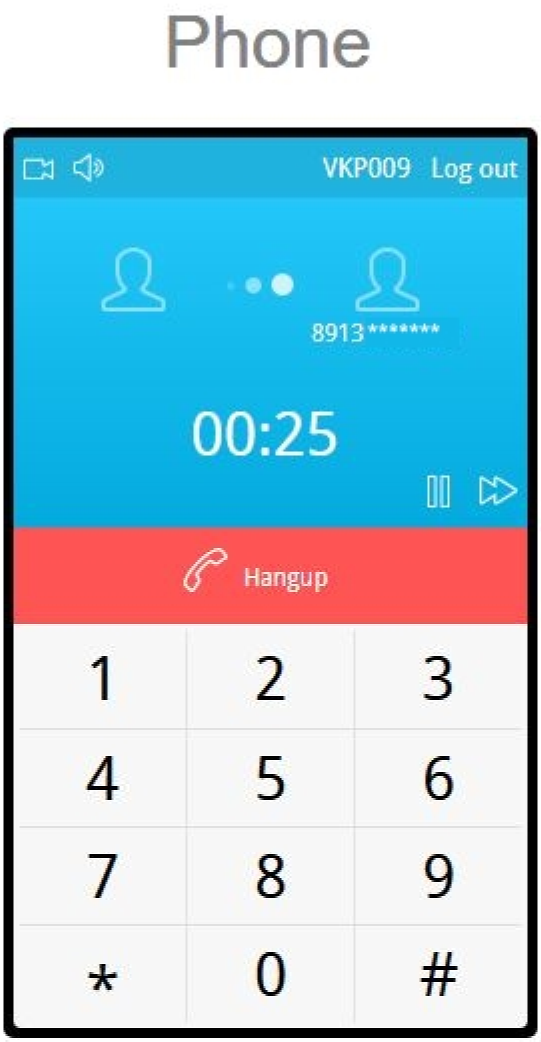
4. The mobile phone displays an incoming call on the screen:



5. Answer the call on the mobile phone:



6. The browser also shows that the connection is established.



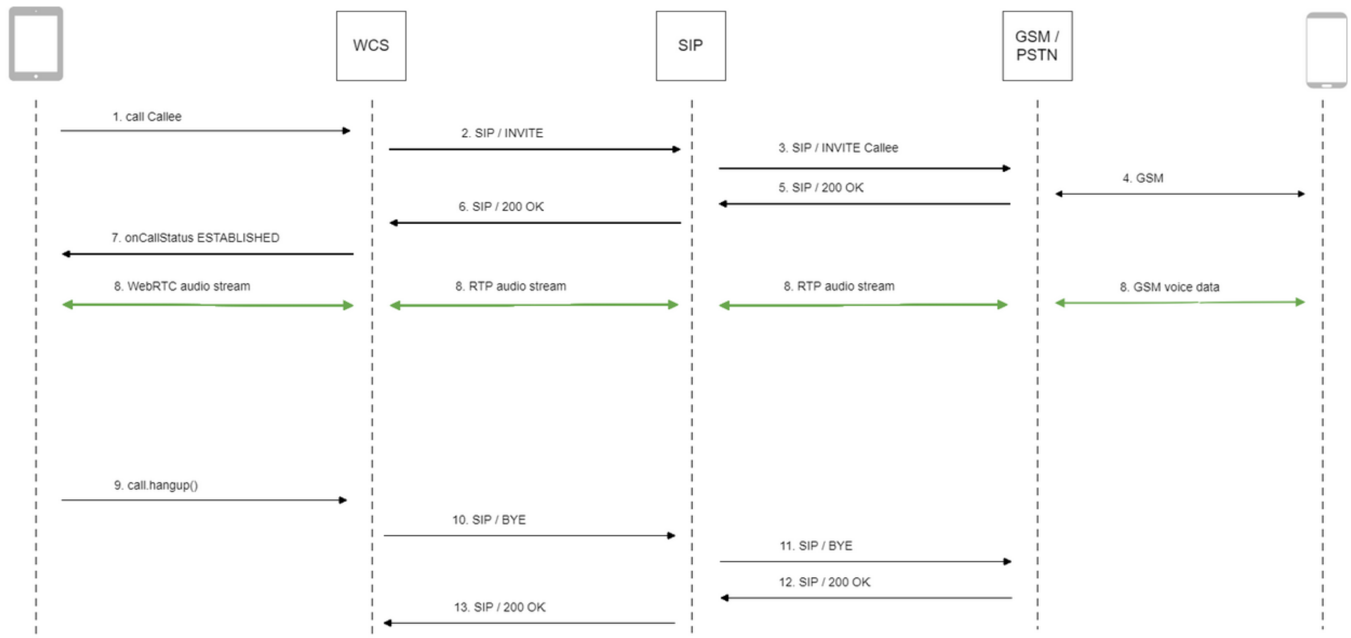
7. To terminate the call, click the "Hangup" button.

Call flow

Below is the call flow when using the Phone example to create a call.

Phone.html

Phone.js



1. Creating a call:

session.createCall(), call.call()code

```
    var outCall = this.session.createCall({
        callee: callee,
        visibleName: this.sipOptions.login,
        localVideoDisplay: this.localVideo,
        remoteVideoDisplay: this.remoteVideo,
        constraints: constraints
        ...
    });

    outCall.call();
```

2. Establishing a connection to the SIP server

3. Establishing a connection to the GSM/PSTN gateway

4. Establishing a connection to the mobile terminal

5. Receiving a confirmation from the GSM/PSTN gateway

6. Receiving a confirmation from the SIP server

7. Receiving from the server an event confirming successful connection.

CallStatusEvent ESTABLISHEDcode

```
    var outCall = this.session.createCall({
        callee: callee,
        visibleName: this.sipOptions.login,
        localVideoDisplay: this.localVideo,
        remoteVideoDisplay: this.remoteVideo,
        constraints: constraints
        ...
    }).on(CALL_STATUS.ESTABLISHED, function(call){
        me.callStatusListener(call);
        ...
    });

    outCall.call();
```

8. Participants of the call exchange audio streams

9. Terminating the call

call.hangup()code

```
Phone.prototype.hangup = function () {
    trace("Phone - hangup " + this.currentCall.id() + " status " + this.currentCall.status());
    this.hideFlashAccess();
    if (this.currentCall.status() == CALL_STATUS.PENDING) {
        this.callStatusListener(this.currentCall);
    } else {
        this.currentCall.hangup();
    }
    this.flashphonerListener.onHangup();
};
```

10. Sending the command to the SIP server

11. Sending the command to the GSM/PSTN gateway

12. Receiving a confirmation from the GSM/PSTN gateway

13. Receiving a confirmation from the SIP server