

Redirecting an audio file to a SIP call using `/call/inject_sound`

1. For the test we use:

- two SIP accounts;
- a softphone to answer the call;
- the [REST client](#) in the Chrome browser.

2. On the WCS server create a directory: `/usr/local/FlashphonerWebCallServer/media`.
Put a file in the RIFF WAV format there, for example `test.wav`.

3. Open the [REST client](#). Send the `/call/startup` query to the WCS server and specify in its parameters:

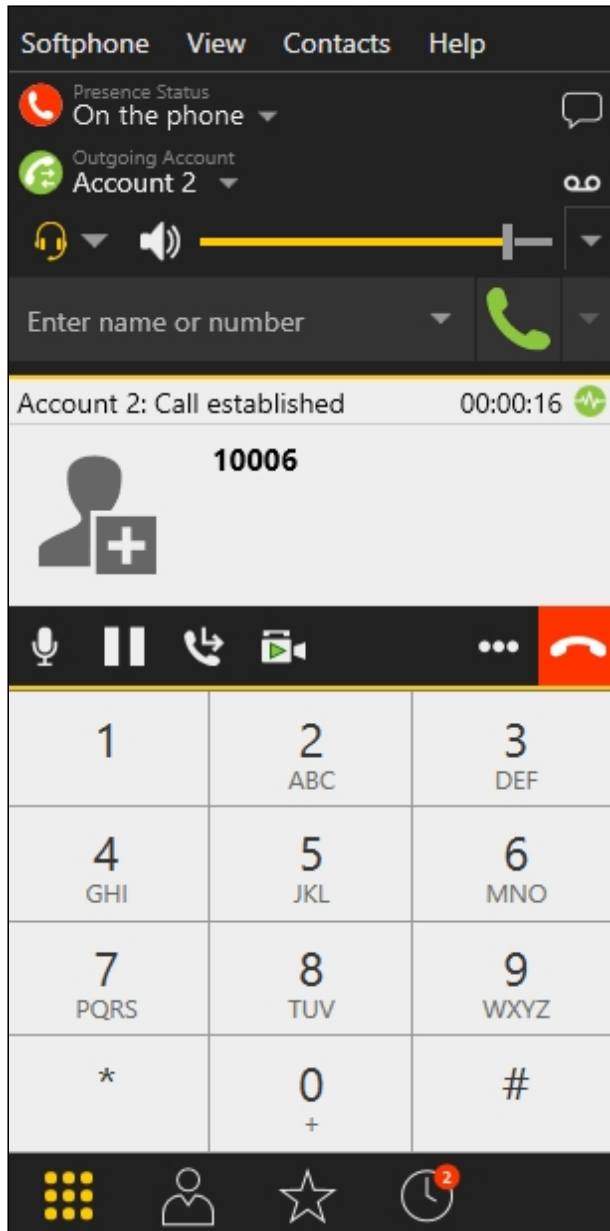
- parameters of your SIP account the call is made from
- the name of your second SIP account the call is made to

The screenshot shows a REST client interface with the following details:

- Method:** POST
- Request URL:** `http://test2.flashphoner.com:9091/rest-api/call/startup`
- Parameters:** Expanded view showing Headers, Body, and Variables tabs.
- Body content type:** application/json
- Editor view:** Raw input
- JSON Body:**

```
{
  "callId": "12345678910",
  "callee": "10005",
  "hasAudio": "true",
  "hasVideo": "true",
  "sipLogin": "10006",
  "sipAuthenticationName": "10006",
  "sipPassword": "*****",
  "sipDomain": "yourdomain.net",
  "sipOutboundProxy": "yourdomain.net",
  "sipPort": "5060",
  "appKey": "defaultApp",
  "sipRegisterRequired": "true"
}
```

4. Receive the call in the softphone:



5. From the REST-client send the `/call/inject_sound` query to the WCS server and specify in the query's parameters:

- the identifier of the call

- the name of the applied audio file `test.wav`

The screenshot shows a REST client interface with the following details:

- Method:** POST
- Request URL:** `http://test2.flashphoner.com:9091/rest-api/call/inject_sound`
- Parameters:** A tabbed interface with 'Headers', 'Body', and 'Variables' tabs. The 'Body' tab is selected.
- Body content type:** application/json
- Editor view:** Raw input
- JSON Body:**

```
{  "callId": "12345678910",  "fileName": "test.wav"}
```
- Buttons:** 'SEND' and a menu icon (three dots) are located at the top right.

6. Make sure the softphone plays the test file.
7. To terminate the call, click the corresponding button in the softphone.

Known issues

1. There is no sound when injecting file to a call stream



Symptoms

REST API query was correct with response code 200 OK, but there is no sound from file in the stream.

✓ **Solution**

a) in `flashphoner.properties` file set the following parameter

```
generate_av_for_ua=all
```

b) in softphone settings specify a STUN server address, for example

`stun.l.google.com:19302` on the appropriate page of SIP account settings

The screenshot shows the 'Topology' tab of a SIP account settings window. The 'Firewall Traversal' section is active, displaying four radio button options: 'Auto-detect firewall traversal method using ICE (recommended)' (selected), 'Discover public IP address (STUN)', 'Use media relay (TURN)', and 'None'. Below these options, the 'Server address' field is populated with 'stun.l.google.com:19302'. The 'User name' and 'Password' fields are empty. The 'Port Ranges' section below contains three rows of checkboxes and input fields. The first row, 'Range of ports used for signaling', has an unchecked checkbox and two input fields both containing '0'. The second row, 'Range of ports used for RTP', has an unchecked checkbox and two input fields both containing '0'. The third row, 'Video', also has an unchecked checkbox and two input fields both containing '0'.

Account	Voicemail	Topology	Presence	Transport	Advanced
Firewall Traversal					
Firewall traversal method:					
<input checked="" type="radio"/> Auto-detect firewall traversal method using ICE (recommended)					
<input type="radio"/> Discover public IP address (STUN)					
<input type="radio"/> Use media relay (TURN)					
<input type="radio"/> None					
Server address: <input type="text" value="stun.l.google.com:19302"/>					
User name: <input type="text"/>					
Password: <input type="text"/>					
Port Ranges					
<input type="checkbox"/> Range of ports used for signaling <input type="text" value="0"/> - <input type="text" value="0"/>					
<input type="checkbox"/> Range of ports used for RTP Audio: <input type="text" value="0"/> - <input type="text" value="0"/>					
Video: <input type="text" value="0"/> - <input type="text" value="0"/>					