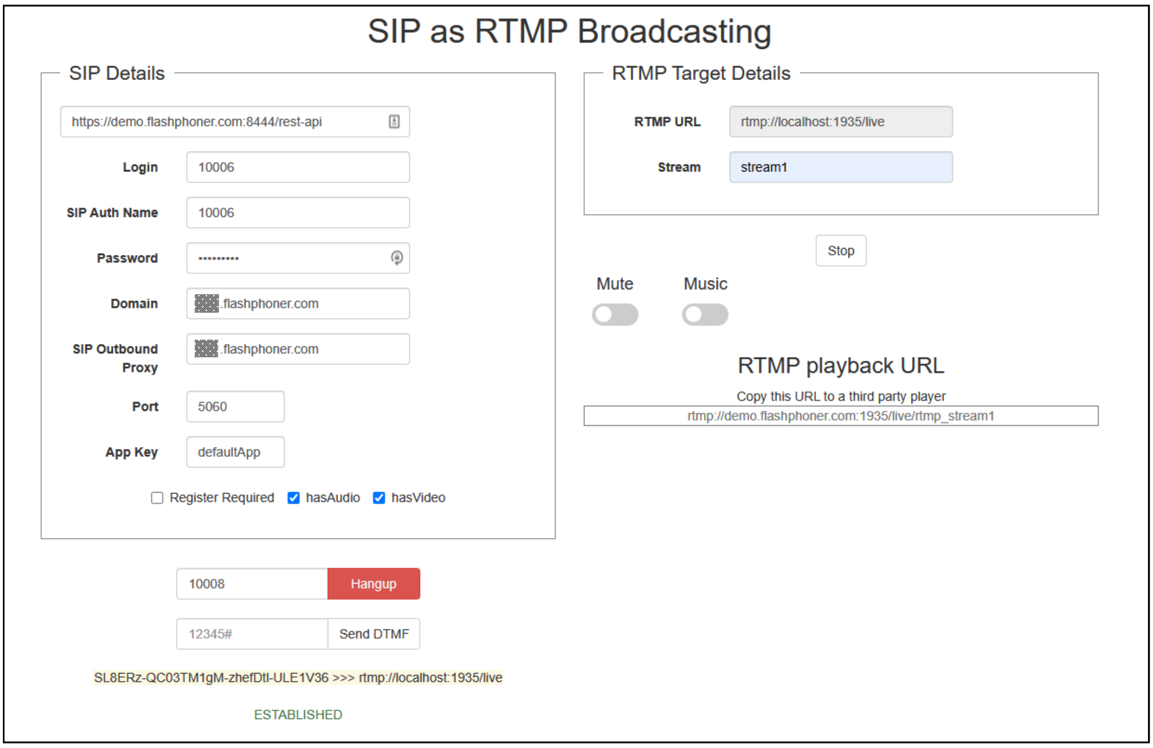
SIP as RTMP 2

Example of video stream republishing from SIP call to RTMP server with sound injection to the stream

This example demonstrates how to make a call to SIP, receive audio and video traffic from SIP in response, inject sound from file to the received video stream and then redirect the stream to a third-party RTMP server for further broadcasting



The code of the example

This example is a simple REST client written on JavaScript, available at:

/usr/local/FlashphonerWebCallServer/client2/examples/demo/sip/sip-as-rtmp-2

- sip-as-rtmp-2.js - a script dealing with REST queries to the WCS server
- sip-as-rtmp-2.html - example page

The example may be tested at this URL:

https://host:8888/client2/examples/demo/sip/sip-as-rtmp-2/sip-as-rtmp-2.html where host is WCS server address.

Analyzing the code

To analyze the code get `sip-as-rtmp-2.js` file version with hash `ecbadc3` that can be found here and is availabe to download in build 2.0.212.

1. REST / HTTP queries sending

code

Sending is done using POST method with `ContentType: application/json` by AJAX query using jQuery framework.

```
function sendREST(url, data, successHandler, errorHandler) {
    console.info("url: " + url);
    console.info("data: " + data);
    $.ajax({
        url: url,
        beforeSend: function ( xhr ) {
            xhr.overrideMimeType( "text/plain;" );
        },
        type: 'POST',
        contentType: 'application/json',
        data: data,
        success: (successHandler === undefined) ? handleAjaxSuccess :
successHandler,
        error: (errorHandler === undefined) ? handleAjaxError : errorHandler
    });
}
```

2. Making outgoing call with REST-request `/call/startup`

code

Call data (`RESTCall`) are collected from the boxes on page

```
var url = field("restUrl") + "/call/startup";
callId = generateCallID();
...

var RESTCall = {};
RESTCall.toStream = field("rtmpStream");
RESTCall.hasAudio = field("hasAudio");
RESTCall.hasVideo = field("hasVideo");
RESTCall.callId = callId;
RESTCall.sipLogin = field("sipLogin");
RESTCall.sipAuthenticationName = field("sipAuthenticationName");
RESTCall.sipPassword = field("sipPassword");
RESTCall.sipPort = field("sipPort");
```

```
RESTCall.sipDomain = field("sipDomain");
RESTCall.sipOutboundProxy = field("sipOutboundProxy");
RESTCall.appKey = field("appKey");
RESTCall.sipRegisterRequired = field("sipRegisterRequired");

for (var key in RESTCall) {
  setCookie(key, RESTCall[key]);
}

RESTCall.callee = field("callee");

var data = JSON.stringify(RESTCall);

sendREST(url, data);
startCheckCallStatus();
```

3. Getting the SIP call status with `/call/find` REST query.

code

```
function getStatus() {
    var url = field("restUrl") + "/call/find";
    currentCallId = { callId: callId };
    $("#callTrace").text(callId + " >>> " + field("rtmpUrl"));
    var data = JSON.stringify(currentCallId);
    sendREST(url, data);
}
```

4. Sending DTMF signal with `/call/send_dtmf` REST query.

code

```
function sendDTMF(value) {
    var url = field("restUrl") + "/call/send_dtmf";
    var data = {};
    data.callId = callId;
    data.dtmf = value;
    data.type = "RFC2833";
    data = JSON.stringify(data);
    sendREST(url, data);
}
```

5. Re-publishing the SIP call stream to an RTMP server with sound file injecting to the stream by `/push/startup` REST query

code

```
function startRtmpStream() {
    if (!rtmpStreamStarted) {
        rtmpStreamStarted = true;
```

```
        var url = field("restUrl") + "/push/startup";
        var RESTObj = {};
        var options = {};
        if ($("#mute").is(':checked')) {
            options.action = "mute";
        } else if ($("#music").is(':checked')) {
            options.action = "sound_on";
            options.soundFile = "sample.wav";
        }
        RESTObj.streamName = field("rtmpStream");
        RESTObj.rtmpUrl = field("rtmpUrl");
        RESTObj.options = options;
        sendREST(url, JSON.stringify(RESTObj), startupRtmpSuccessHandler,
startupRtmpErrorHandler);
        sendDataToPlayer();
        startCheckTransponderStatus();
    }
}
```

6. Getting RTMP stream status with `/push/find` REST query

code

```
function getTransponderStatus() {
    var url = field("restUrl") + "/push/find";
    var RESTObj = {};
    // By default transponder's stream name will contain prefix "rtmp_"
    RESTObj.streamName = "rtmp_" + field("rtmpStream");
    RESTObj.rtmpUrl = field("rtmpUrl");
    sendREST(url, JSON.stringify(RESTObj), transponderStatusSuccessHandler,
transponderStatusErrorHandler);
}
```

7. Mute/unmute stream sound

Mute sound with `/push/mute` code

```
function mute() {
    if (rtmpStreamStarted) {
        $("#mute").prop('disabled', true);
        var RESTObj = {};
        RESTObj.mediaSessionId = rtmpMediaSessionId;
        var url = field("restUrl") + "/push/mute";
        sendREST(url, JSON.stringify(RESTObj), muteSuccessHandler,
muteErrorHandler);
    }
}
```

Unmute sound with `/push/unmute` code

```
function unmute() {
    if (rtmpStreamStarted) {
```

```
        $("#mute").prop('disabled', true);
        var RESTObj = {};
        RESTObj.mediaSessionId = rtmpMediaSessionId;
        var url = field("restUrl") + "/push/unmute";
        sendREST(url, JSON.stringify(RESTObj), muteSuccessHandler,
muteErrorHandler);
    }
}
```

8. Injecting additional sound to RTMP stream

Injecting sound from file with `/push/sound_on` code

```
function soundOn() {
    if (rtmpStreamStarted) {
        $("#music").prop('disabled', true);
        var RESTObj = {};
        RESTObj.mediaSessionId = rtmpMediaSessionId;
        RESTObj.soundFile = "sample.wav";
        RESTObj.loop = false;
        var url = field("restUrl") + "/push/sound_on";
        sendREST(url, JSON.stringify(RESTObj), injectSoundSuccessHandler,
injectSoundErrorHandler);
    }
}
```

Stop injecting sound from file with `/push/sound_off` code

```
function soundOff() {
    if (rtmpStreamStarted) {
        $("#music").prop('disabled', true);
        var RESTObj = {};
        RESTObj.mediaSessionId = rtmpMediaSessionId;
        var url = field("restUrl") + "/push/sound_off";
        sendREST(url, JSON.stringify(RESTObj), injectSoundSuccessHandler,
injectSoundErrorHandler);
    }
}
```

9. Hangup the SIP call with `/call/terminate` REST query.

code

```
function hangup() {
    var url = field("restUrl") + "/call/terminate";
    var currentCallId = { callId: callId };
    var data = JSON.stringify(currentCallId);
    sendREST(url, data);
}
```

10. RTMP URL displaying on the page to copy to a third party player

code

```
function sendDataToPlayer() {
    var host = field("rtmpUrl")
        .replace("localhost", window.location.hostname)
        .replace("127.0.0.1", window.location.hostname);

    var rtmpStreamPrefix = "rtmp_";
    var url = host + "/" + rtmpStreamPrefix + field("rtmpStream");
    $("#player").text(url);
}
```