

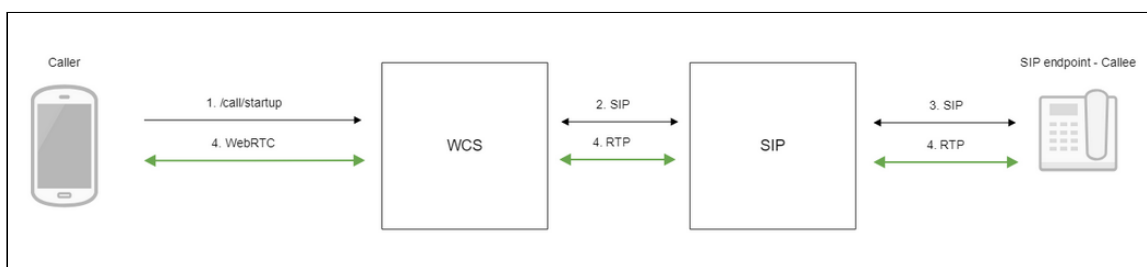
SIP calls using iOS SDK

Overview

SIP call on iOS devices can be made both [from a browser](#) and using the [iOS SDK](#).

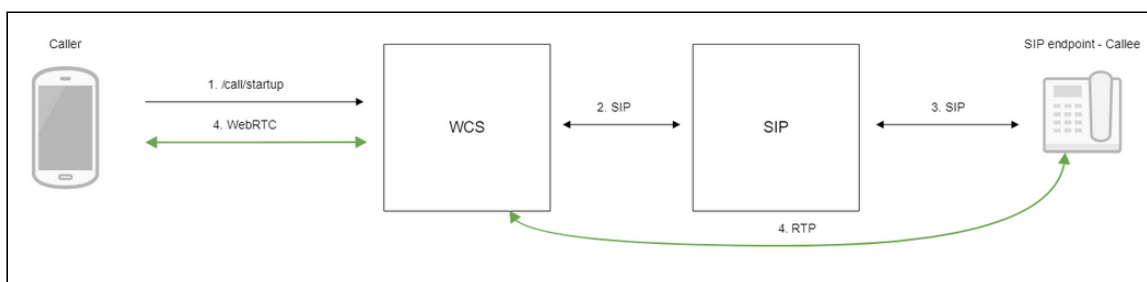
Operation flowchart

1. SIP server as a proxy server to transfer calls and RTP media



1. The iOS device begins a call
2. WCS connects to the SIP server
3. The SIP server connects to the SIP device that receives the call
4. The iOS device and the SIP device exchange audio and video streams

2. SIP server as a server to transfer calls only



1. The iOS device begins a call
2. WCS connects to the SIP server
3. The SIP server connects to the SIP device that receives the call
4. The iOS device and the SIP device exchange audio and video streams

Testing

Making an outgoing call from iOS to a SIP device

1. For the test we use:
2. two SIP accounts;
3. the [Phone](#) application to make a call;
4. a software phone to answer the call.
5. Build and install the Phone app to the iOS device. Start the app, enter the URL of the WCS server to connect to it via Websocket and the data of the SIP account making a call:

No SIM

16:50

40 %

wss://wcs5-eu.flashphoner.com:8443

Sip Login

1000

Sip Auth Name

1000

Sip Password

1234

Sip Domain

192.168.0.1

Sip Outbound Proxy

192.168.0.1

Sip Port

5060

Sip Register Required

☒

NO STATUS

6. Run the softphone, enter the data of the SIP account that receives the call:

Account Voicemail Topology Presence Transport Advanced

Account name:

Protocol:

Allow this account for

☒ Call

☒ IM / Presence

User Details

* User ID:

* Domain:

Password:

Display name:

Authorization name:

Domain Proxy

☒ Register with domain and receive calls

Send outbound via:

☒ Domain

☐ Proxy Address:

7. Tap the **Connect** button in the app, a connection will be established to the server. Then enter the identifier of the SIP account that receives the call and click the **Call** button:

No SIM

87.226.225.56

07:37

86 %

Sip Port

5060

Sip Register Required

REGISTERED

DISCONNECT

Invite Parameters

Callee

10005

RING

HANGUP

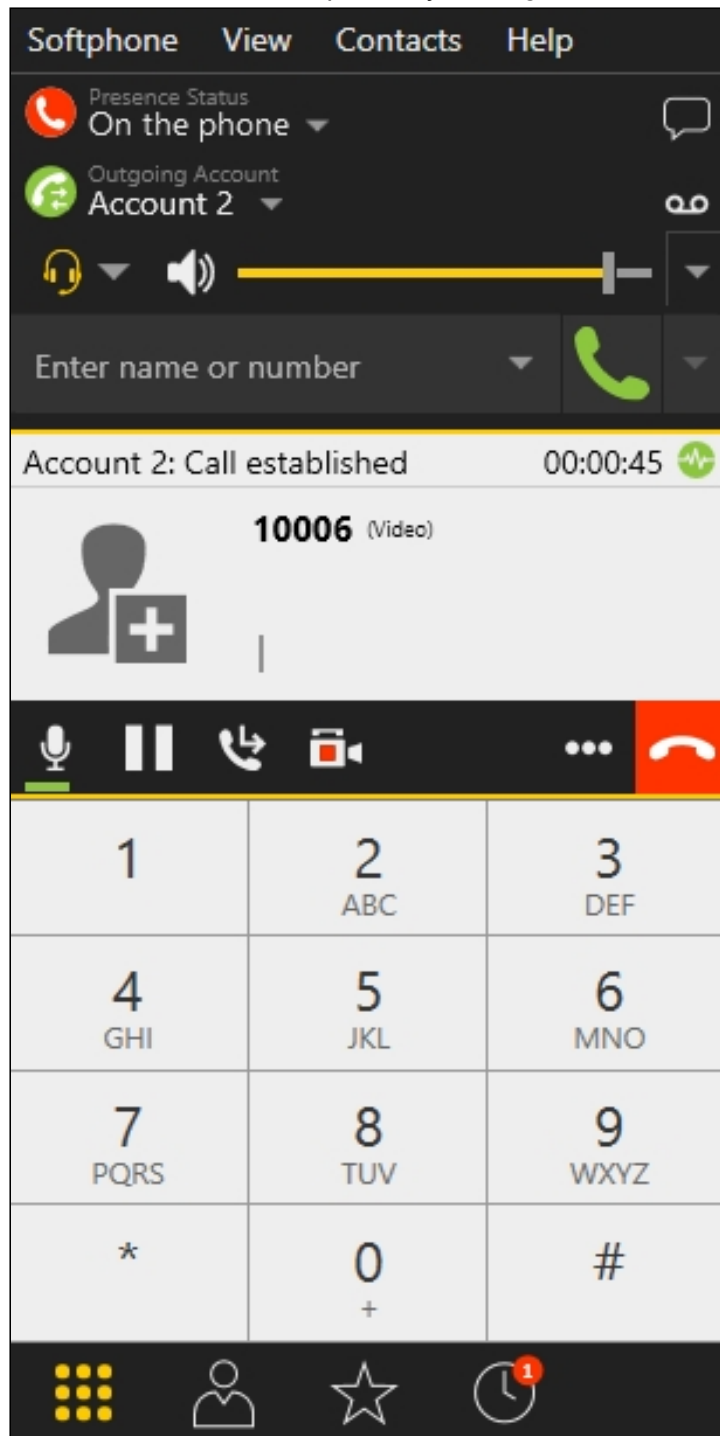
HOLD

DTMF

1

DTMF

8. Answer the call in the softphone by clicking the answer button:



9. To terminate the call, tap the **Hangup** button in the application, or click the end call button in the softphone.

Receiving an outgoing call from a SIP device to iOS

1. For the test we use:
2. two SIP accounts;

3. a softphone to make a call;
4. the [Phone](#) application to answer the call.
5. Build and install the Phone app to the iOS device. Start the app, enter the URL of the WCS server to connect via Secure Websocket and the data of the SIP account that receives the call:

The screenshot shows the configuration interface of the Phone app on an iOS device. The status bar at the top indicates 'No SIM', signal strength, time '16:50', Bluetooth, 40% battery, and a charging icon. The configuration fields are as follows:

Field	Value
URL	wss://wcs5-eu.flashphoner.com:8443
Sip Login	1000
Sip Auth Name	1000
Sip Password	1234
Sip Domain	192.168.0.1
Sip Outbound Proxy	192.168.0.1
Sip Port	5060
Sip Register Required	Enabled (Toggle)
Status	NO STATUS

Tap the [Connect](#) button in the app to establish a connection to the WCS server

6. Run the software phone and enter the data of the SIP account making the call:

Account Voicemail Topology Presence Transport Advanced

Account name:

Protocol:

Allow this account for

☒ Call

☒ IM / Presence

User Details

* User ID:

* Domain:

Password:

Display name:

Authorization name:

Domain Proxy

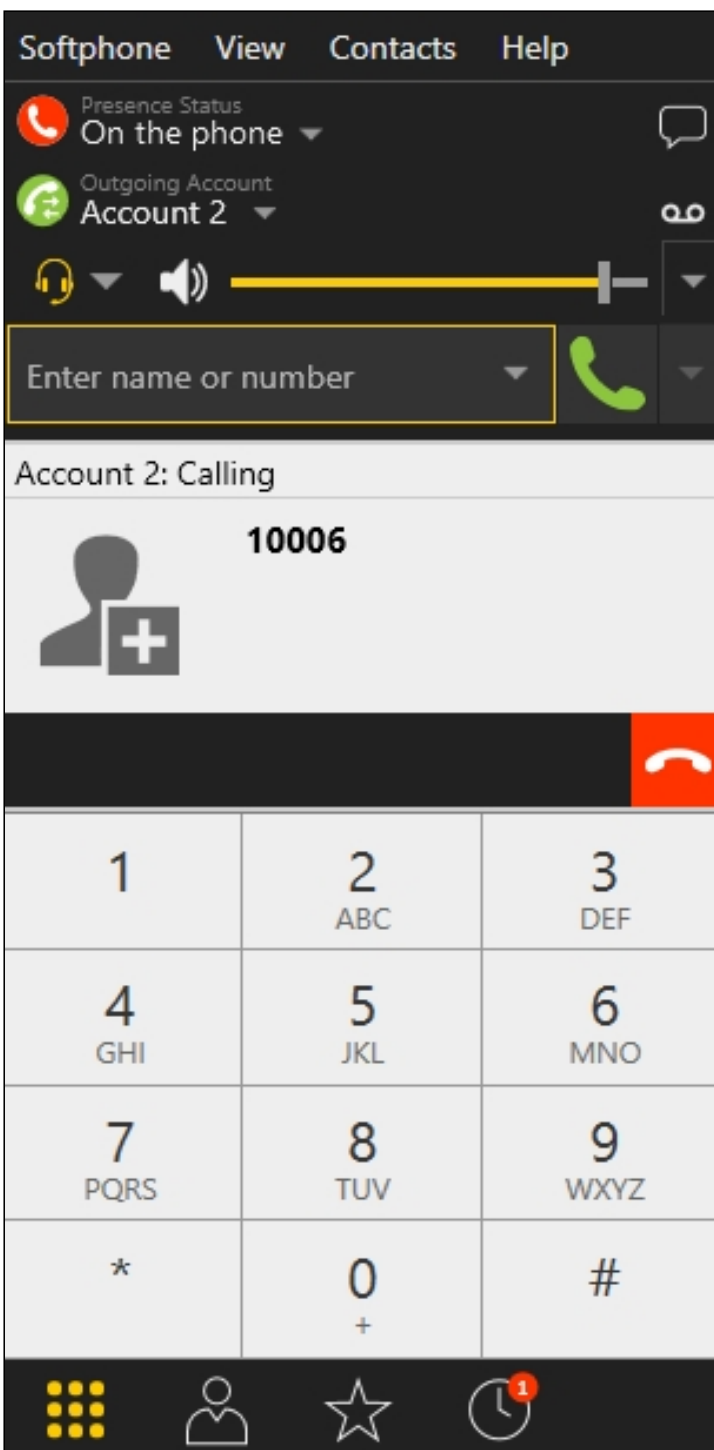
☒ Register with domain and receive calls

Send outbound via:

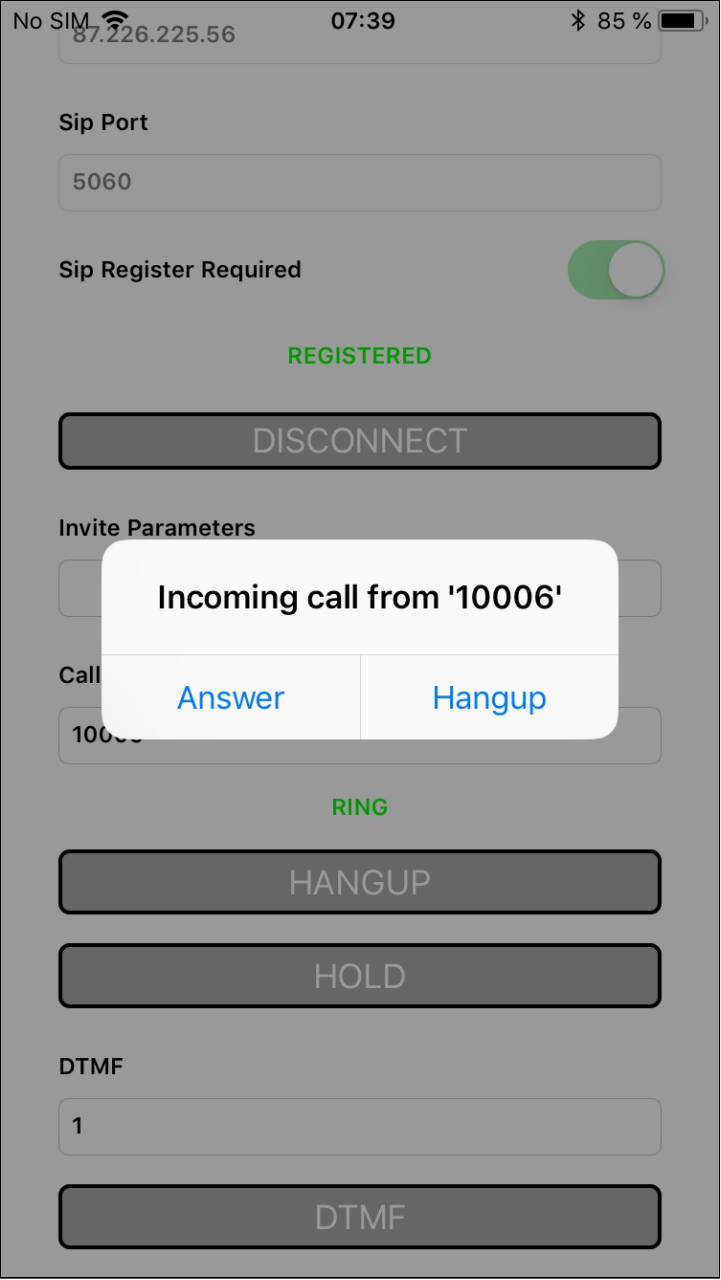
☒ Domain

☐ Proxy Address:

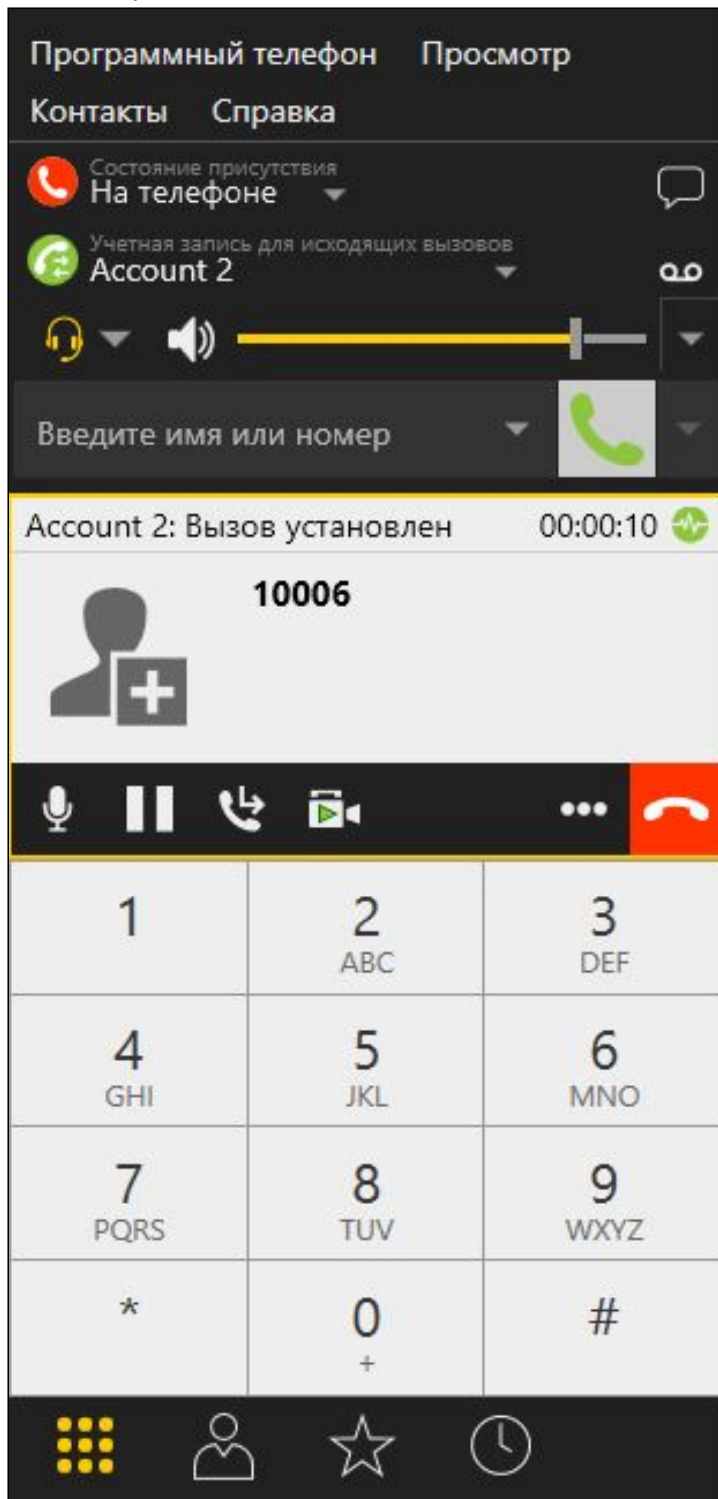
7. In the softphone enter the identifier of the SIP account that receives the call and click the call button:



8. Answer the call in the application by tapping **Answer** :



9. In the softphone make sure the call has started:

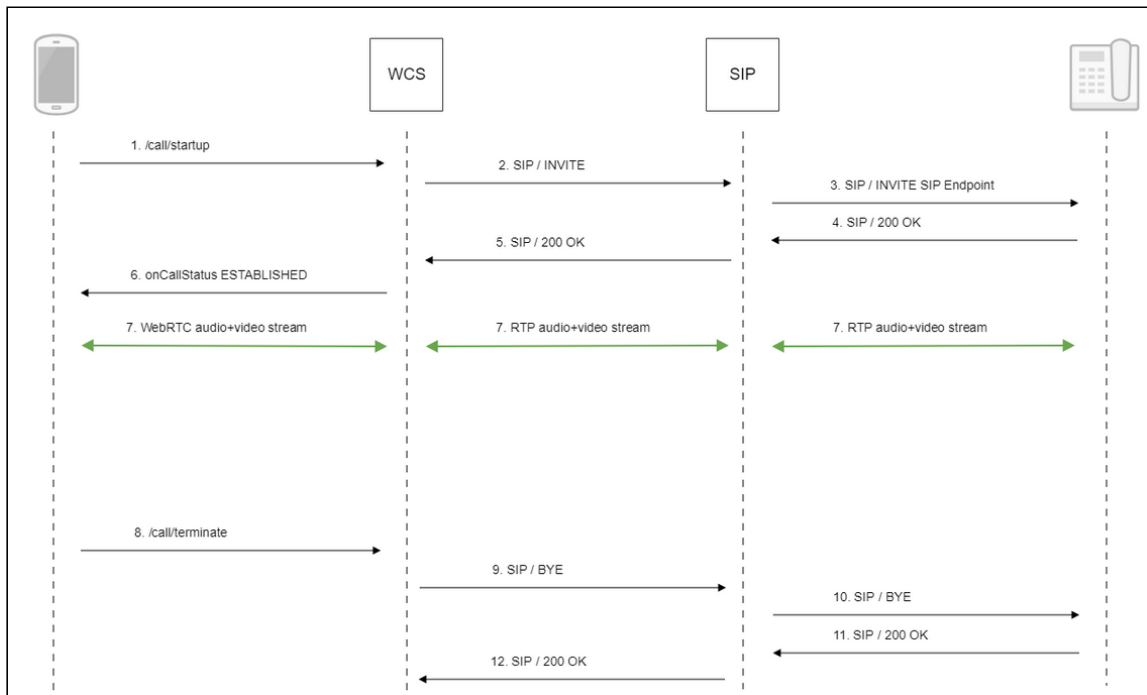


10. To terminate the call, tap the Hangup button in the app, or click the end call button in the softphone.

Call flow

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

View class for the main view of the application: ViewController (header file [ViewController.h](#); implementation file [ViewController.m](#))



1. Creating a call

`FPWCSEApi2Session.createCall()`, `FPWCSEApi2Call.call()` [code](#)

```

- (FPWCSEApi2Call *)call {
    FPWCSEApi2Session *session = [FPWCSEApi2 getSessions][0];
    FPWCSEApi2CallOptions *options = [[FPWCSEApi2CallOptions alloc] init];
    NSString *parameters = _inviteParameters.input.text;
    if (parameters && [parameters length] > 0) {
        NSError* err = nil;
        parameters = [parameters stringByReplacingOccurrencesOfString:@" "
withString:@"\""];
        NSMutableDictionary *dictionary = [NSJSONSerialization
JSONObjectWithData:[parameters dataUsingEncoding:NSUTF8StringEncoding]
options:0 error:&err];
        if (err) {
            NSLog(@"Error converting JSON Invite parameters to dictionary
%@", JSON %@", err, parameters);
        } else {
            options.inviteParameters = dictionary;
        }
    }
    options.callee = _callee.input.text;
    ...
    NSError *error;
    call = [session createCall:options error:&error];
    ...
    [call call];
    return call;
}

```

2. Sending **SIP INVITE** to the SIP server
3. Sending **SIP INVITE** to the SIP device
4. Receiving a confirmation from the SIP device
5. Receiving a confirmation from the SIP server
6. Receiving from the server an event confirming successful connection
7. The caller and the callee exchange audio and video streams
8. Terminating the call

`FPWCSSApi2Call.hangup()` [code](#)

```
- (void)callButton:(UIButton *)button {
    [self changeViewState:button enabled:NO];
    if ([button.titleLabel.text isEqualToString:@"HANGUP"]) {
        if ([FPWCSSApi2 getSessions].count) {
            [call hangup];
        } else {
            [self toCallState];
        }
        ...
    }
}
```

9. Sending **SIP BYE** to the SIP server
10. Sending **SIP BYE** to the SIP device
11. Receiving a confirmation from the SIP device
12. Receiving a confirmation from the SIP server

Known issues

1. It's impossible to make a SIP call if **SIP Login** and **SIP Authentication name** fields contain inappropriate characters



Symptoms

SIP call sticks in **PENDING** state

✓ Solution

According to [RFC3261](#), `SIP Login` and `SIP Authentication name` should not contain any of unescaped spaces and special symbols and should not be enclosed in angle brackets `<>`.

For example, this is not allowed by the specification

```
sipLogin='Ralf C12441@host.com'  
sipAuthenticationName='Ralf C'  
sipPassword='demo'  
sipVisibleName='null'
```

and this is allowed

```
sipLogin='Ralf_C12441'  
sipAuthenticationName='Ralf_C'  
sipPassword='demo'  
sipVisibleName='Ralf C'
```