

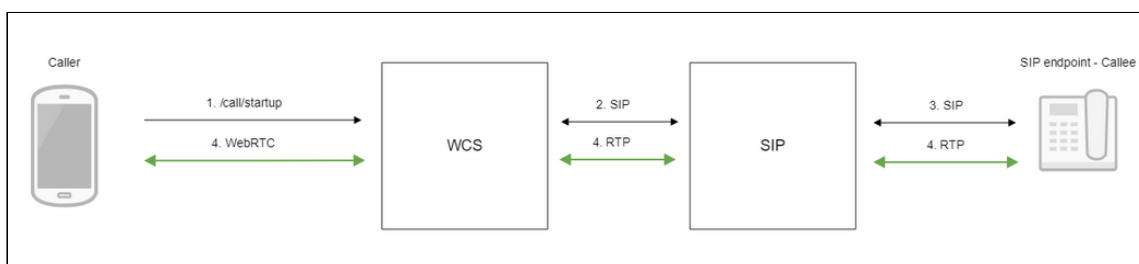
SIP calls using Android SDK

Overview

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

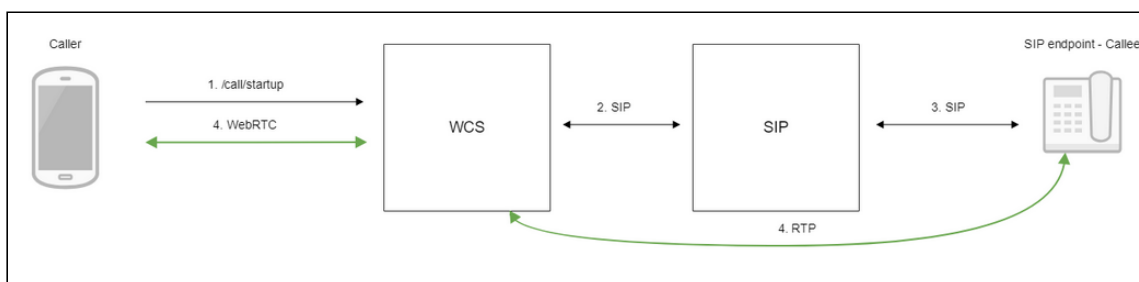
Operation flowchart

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



1. The Android 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 Android device and the SIP device exchange audio and video streams

2. SIP server as a server to transfer calls only

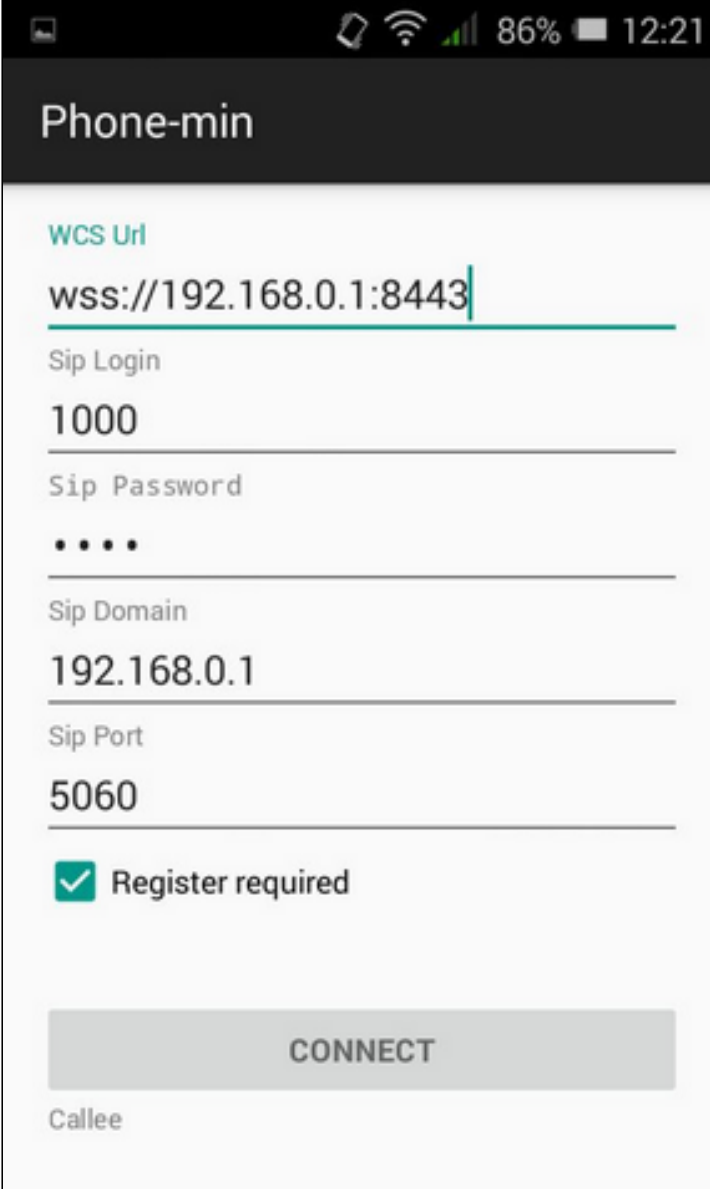


1. The Android device begins a call
2. WCS connects to the SIP server
3. The SIP server connects to the SIP device that receives the call
4. The Android device and the SIP device exchange audio and video streams

Testing

Making an outgoing call from Android 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. Install the [Phone](#) app to the Android 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:



The screenshot shows the 'Phone-min' application interface on an Android device. The status bar at the top indicates 86% battery and the time 12:21. The app title 'Phone-min' is displayed in a dark header. Below the header, the configuration fields are as follows:

- WCS Url:** wss://192.168.0.1:8443
- Sip Login:** 1000
- Sip Password:** (masked with four dots)
- Sip Domain:** 192.168.0.1
- Sip Port:** 5060
- Register required:**

A large grey button labeled 'CONNECT' is positioned below the configuration fields. At the bottom of the screen, the text 'Callee' is visible.

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

Account	Voicemail	Topology	Presence	Transport	Advanced
Account name:	<input type="text" value="Account 2"/>				
Protocol:	<input type="text" value="SIP"/>				
Allow this account for					
<input checked="" type="checkbox"/>	Call				
<input checked="" type="checkbox"/>	IM / Presence				
User Details					
* User ID:	<input type="text" value="10005"/>				
* Domain:	<input type="text" value="yuordomain.net"/>				
Password:	<input type="password" value="•••••"/>				
Display name:	<input type="text" value="10005"/>				
Authorization name:	<input type="text" value="10005"/>				
Domain Proxy					
<input checked="" type="checkbox"/>	Register with domain and receive calls				
Send outbound via:					
<input checked="" type="radio"/>	Domain				
<input type="radio"/>	Proxy Address: <input type="text"/>				

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:

Callee

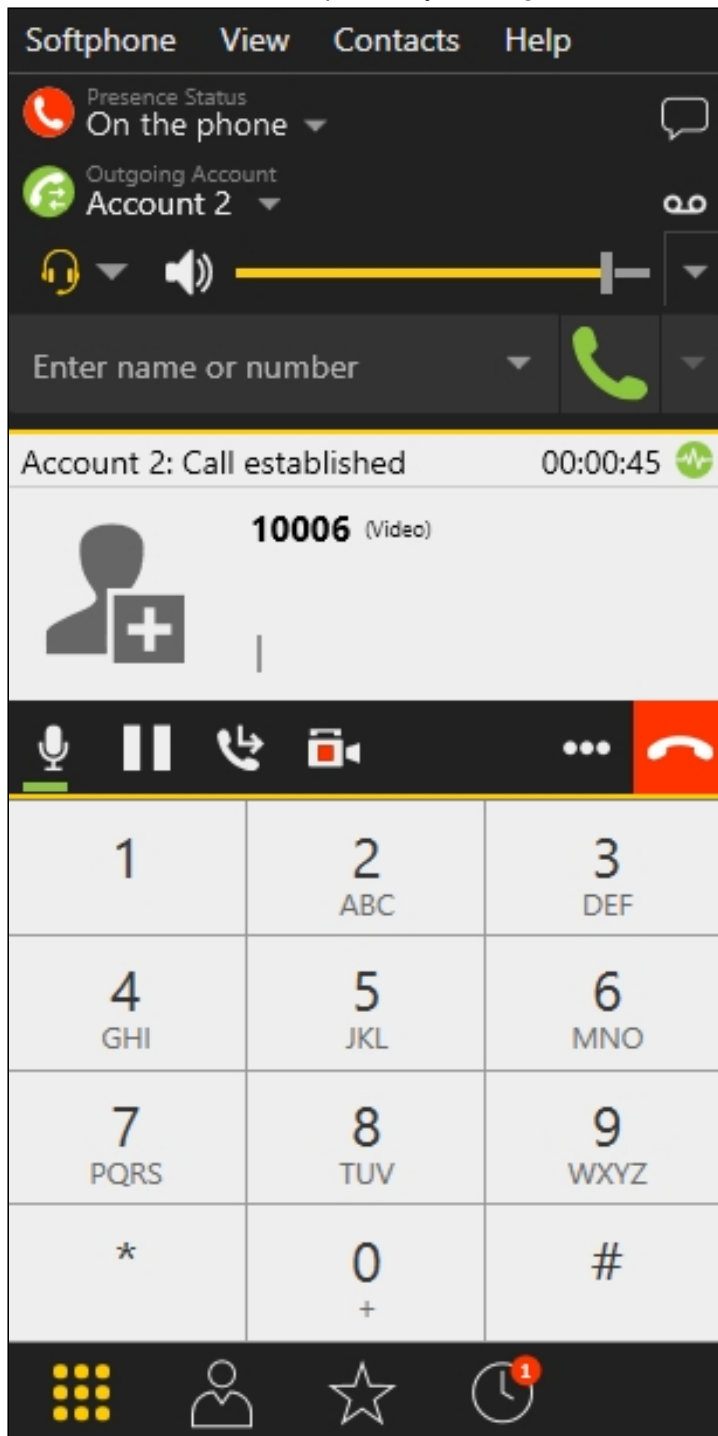
10005

ESTABLISHED

HANGUP

HOLD

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 incoming call from a SIP device to Android

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. Install the [Phone](#) app to the Android device. Start the app, enter the URL of the WCS server to connect via Websocket and the data of the SIP account that receives the call:

The screenshot shows the 'Phone-min' app interface on an Android device. The status bar at the top indicates 86% battery and the time 12:21. The app title 'Phone-min' is displayed in a dark header. Below the header, the configuration fields are as follows:

- WCS Url:** wss://192.168.0.1:8443
- Sip Login:** 1000
- Sip Password:** masked with four dots
- Sip Domain:** 192.168.0.1
- Sip Port:** 5060
- Register required:** checked (indicated by a green checkmark)

A large grey button labeled 'CONNECT' is positioned below the configuration fields. At the bottom of the screen, the text 'Callee' is visible.

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:	<input type="text" value="Account 2"/>				
Protocol:	<input type="text" value="SIP"/>				
Allow this account for	<input type="checkbox"/> Call				
	<input checked="" type="checkbox"/> IM / Presence				
User Details					
* User ID:	<input type="text" value="10005"/>				
* Domain:	<input type="text" value="yuordomain.net"/>				
Password:	<input type="password" value="•••••"/>				
Display name:	<input type="text" value="10005"/>				
Authorization name:	<input type="text" value="10005"/>				
Domain Proxy					
<input checked="" type="checkbox"/>	Register with domain and receive calls				
Send outbound via:					
<input checked="" type="radio"/>	Domain				
<input type="radio"/>	Proxy Address: <input type="text"/>				

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

Softphone View Contacts Help

Presence Status
On the phone

Outgoing Account
Account 2

Enter name or number

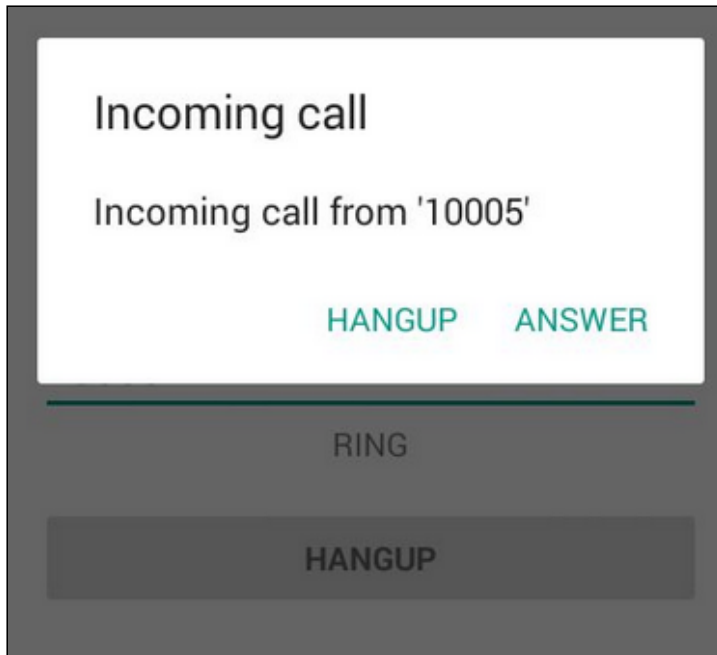
Account 2: Calling

10006

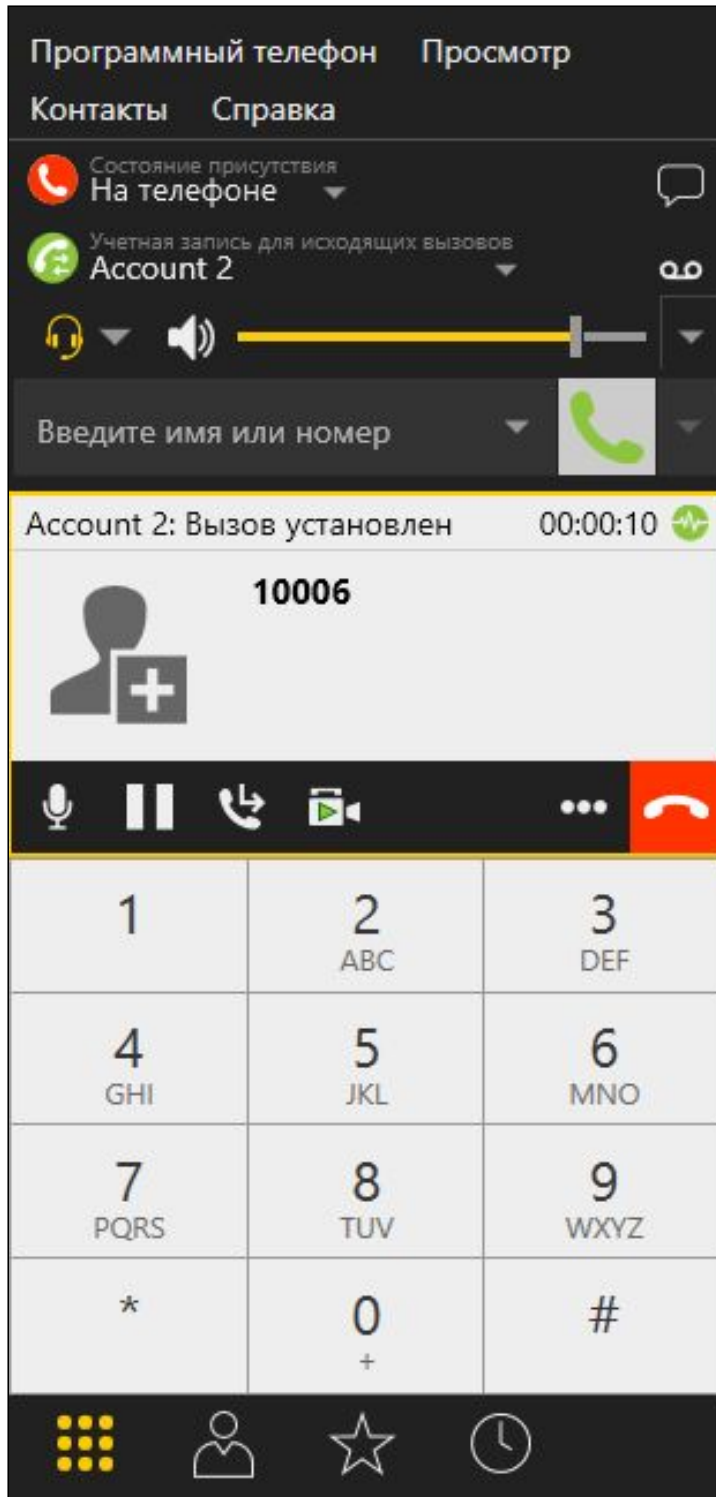
1	2 ABC	3 DEF
4 GHI	5 JKL	6 MNO
7 PQRS	8 TUV	9 WXYZ
*	0 +	#

Grid icon, Person icon, Star icon, Clock icon (1)

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



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

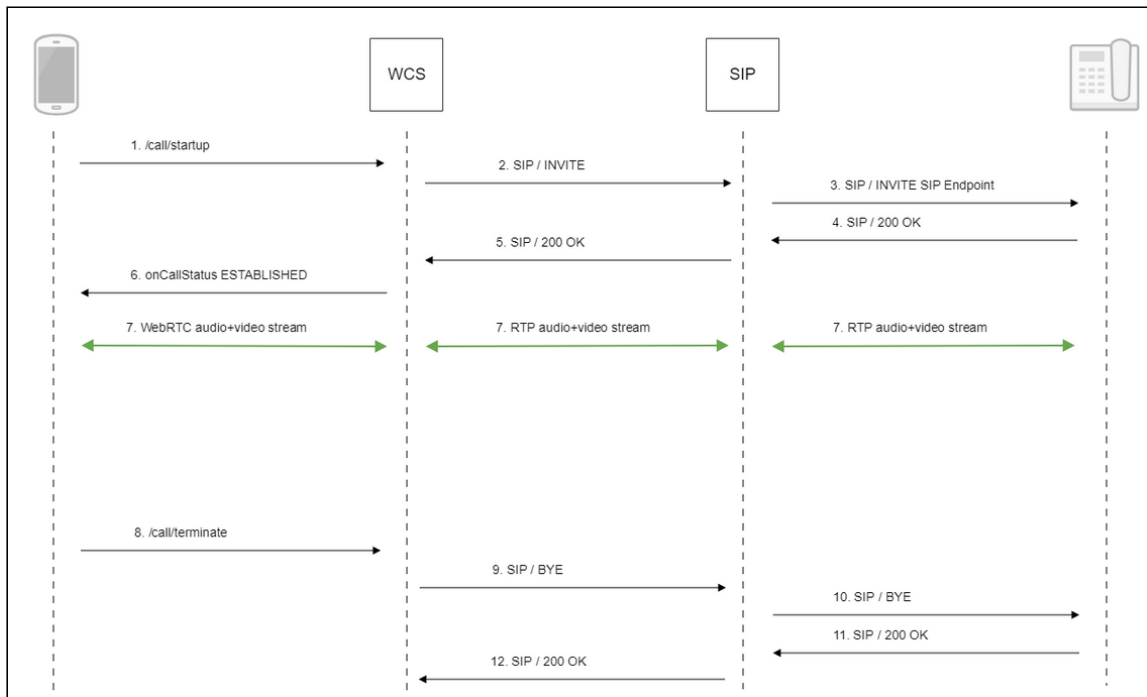


10. To terminate the call, tap the Hangup button in the app, or click the end call button in the softphone.

Call flow

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

PhoneMinActivity.java



1. Creating a call

`Session.createCall()`, `Call.call()` code

```
CallOptions callOptions = new
CallOptions(mCalleeView.getText().toString());
AudioConstraints audioConstraints =
callOptions.getConstraints().getAudioConstraints();
MediaConstraints mediaConstraints =
audioConstraints.getMediaConstraints();
...
call = session.createCall(callOptions);
call.on(callStatusEvent);
/**
 * Make the outgoing call
 */
call.call();
Log.i(TAG, "Permission has been granted by user");
```

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

`Call.hangup()` code

```

if (mCallButton.getTag() == null ||
Integer.valueOf(R.string.action_call).equals(mCallButton.getTag())) {
    if ("".equals(mCalleeView.getText().toString())) {
        return;
    }
    ...
} else {
    mCallButton.setEnabled(false);
    call.hangup();
    call = null;
}

```

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'

```

