‹ACCESS›

COMREX

Integrating the Android CSipSimple app for use with Comrex ACCESS codecs

Introduction

For several years, Comrex ACCESS IP codecs have supported interworking with SIP based hard and soft phones utilizing a range of audio codec algorithms, some of which provide wideband audio transfer.

Several free, open source apps are available for Android that offer VoIP services using the SIP protocol. There's even a native dialer app built in to later versions of Android that interworks with ACCESS, although at this point no wideband codecs are supported with that dialer.

One particular offering, CSipSimple, is especially powerful and configurable. Recently, the app has integrated the AAC-ELD codec as one of its audio coding choices. This is quite interesting because it creates a free, open source app that utilizes a pro-grade broadcast-capable codec. Previous apps that integrated AAC family codecs sell for hundreds of dollars. With AAC-ELD integration, CSipSimple has enabled the use of true broadcast quality, low delay interactive audio transfer on smartphones for free.

About Licenses

The AAC family of codecs, while regarded as the absolute best technology in its class, has seen implementations burdened by high patent licensing costs. The AAC family is protected by a pool of patents from dozens of rights-holders, and most free implementations are infeasible due to these restrictions.

This changed with the advent of Android 4.1, which natively contains elements of the AAC codec family. The patent and copyright licenses to AAC are paid by the device manufacturer when Android 4.1 is applied, so use of the codec on the device is covered. Free open source apps are now able to leverage the codec inherent in the phone.

About CSipSimple

This app is designed to allow Android smartphone users to make VoIP calls. It can be very simple to set up for average users, but contains several "expert" modes that allows deeper access to functions. Because it's not designed as a broadcast codec app, it can be less than intuitive for radio contribution. But there are ways to simplify it. In this note we'll show step-by-step how to configure the app for "one touch dial" to your ACCESS codec.

ACCESS Settings

Before using any SIP-based app with ACCESS, you'll need to configure it to make and receive SIP calls in addition to the "BRIC normal" calls it makes by default. This is done via the web-based interface. We'll assume this process is happening on ACCESS Rack, but the process is similar for ACCESS Portable. Unfortunately, the Comrex BRIC-Link does not currently support the AAC-ELD codec required, but it will work using the G.722 codec also contained within CSipSimple.

The ACCESS must be running firmware 2.7.1 or higher. Firmware upgrades are available for free on the Comrex website, along with instructions and videos on how to perform these upgrades. Comrex tech support can also help upgrade hardware codecs.

In addition, if the ACCESS has a serial number below AR2869, it did not ship enabled for AAC by default. If no AAC upgrade was purchased previously, you must purchase the AAC upgrade license key for your

ACCESS. Contact Comrex about pricing. G.722 mode will work without an additional license.

Once the studio-side codec is running the correct firmware and license, there are some settings to make in the user interface for that codec. Consult the codec manual on how to access the user interface
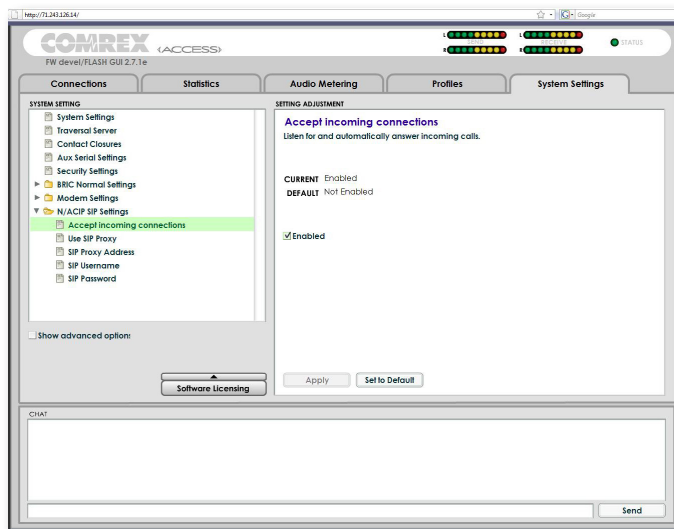

Figure 1

Figure 1 shows the **Systems Settings** tab on the Comrex ACCESS codec . Under the section titled **N/ACIP SIP** settings, find the entry for **Accept Incoming Connections**. Click the **Enabled** box and press **Apply**.

The other settings in this section are only required if you plan to register this codec with a SIP service. See the hardware codec manual on how to set and test these correctly. In this note, we will configure both ends to connect in **Unregistered mode**, meaning we will not be using an external SIP server on either end. The use of unregistered mode has the advantage that no other servers are required in the link, and data flows from the devices in the most direct way possible. **Unregistered mode** may not be possible to ACCESS units that are located behind firewalls and routers.

Configuring CSipSimple

CSipSimple requires two separate apps to be installed on your phone in order to support wideband codecs. As of this writing, neither of the versions available on the Play Store have the required features to support AAC-ELD. Users of G.722 can download the apps directly from the Play Store.

You'll need to understand how to install an .apk file under Android. There are many references available on line on how to do this.

The main application .apk file must be downloaded from the CSipSimple project site at

http://code.google.com/p/csipsimple/wiki/ HowToInstallDevVersion

Use the latest nightly release available.

The codec pack can be downloaded from Comrex at

ftp://ftp.comrex.com/CSipSimple/ CSipSimpleCodecPack.apk

This is a special version of the codec pack with AAC-ELD capability. The codec pack on the CSipSimple site or on the Play Store does not have this mode.

The goal here is not to use many of the features of the app, but rather to use the app to allow for "one touch" outgoing only calls to a specific ACCESS codec. That way, once configured, the app can be used by anyone at any technical level.

Once both programs are installed, launch the CSipSimple app on your phone or tablet. The **Easy Configuration Wizard** launches by default. Here we uncheck the **Integrate with Android** box, and change the **Availability Profile** to *Only for Outgoing*. Finally, we check the **I'm allowed to use mobile** box on the bottom of the wizard and press **Save**.
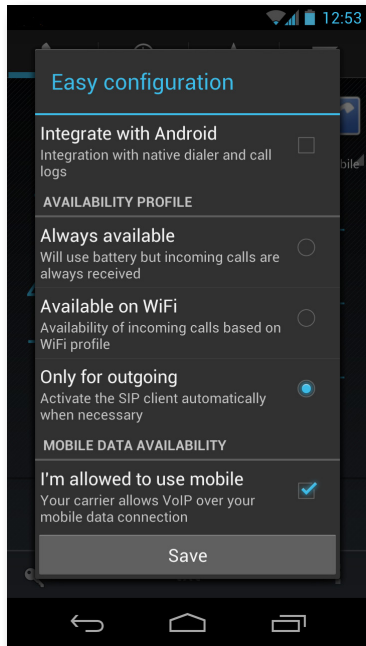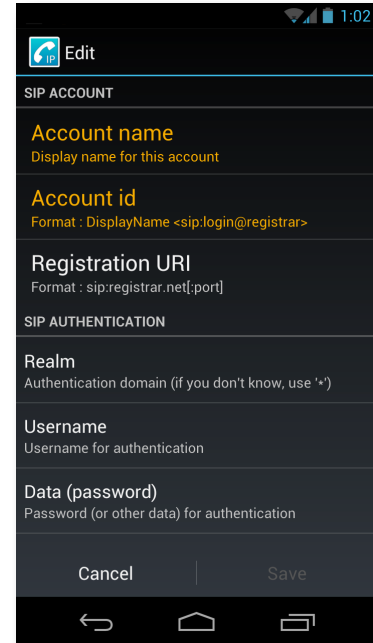
Figure 2

Figure 3

Next you'll be taken to the **Account Entry** screen. We don't intend to actually use a SIP account (we'll dial the ACCESS IP address directly) but we'll need to "spoof" CSipSimple into thinking it has an actual account to allow outgoing calls to work.

So go ahead and click **Add Account** so we can build our dummy account. You're next given a choice of Account Wizards (this is a really nice feature of the app for real VoIP use, but not terribly useful to us). Scroll down to find the **Expert** Wizard and start it up.

You'll need to enter a couple of dummy values in the first two entries. Anything will do in the first entry, including the word **dummy**. The second field requires an Account ID in a particular format to be accepted. We used the value **<sip:dummy@dummy>** (including the leading and trailing symbols) to make the app happy.

At this point you can press **Save** and CSipSimple will "activate" your dummy account (in reality, since you've left several essential fields empty, it won't use the account even though it shows **Registered**) and allow you to place calls.

Next press the **Back** button to return to the main dialing screen. Choose the **Menu** button (on newer phones, this is three vertical dots in the lower right corner, on older phones it's a dedicated key) and select **Settings**.

Choose the **Media** option and deselect the option labeled **Codec Priority list per bandwidth** box. Next select the **Codecs** entry.

Figure 4

Figure 5

Here's the codec priority list. For simplicity, deselect every option except the AAC-ELD codec (our dev version has this listed as mpeg4-generic 48KHz). Alternately, choose G.722 which is compatible with BRIC-Link. Press the **Back** button twice to return to the main settings menu.

Finally, let's make sure CSipSimple is only enabled for outgoing calls. This will prevent it from interfering with your normal phone usage, and help with the one-touch dialing app.
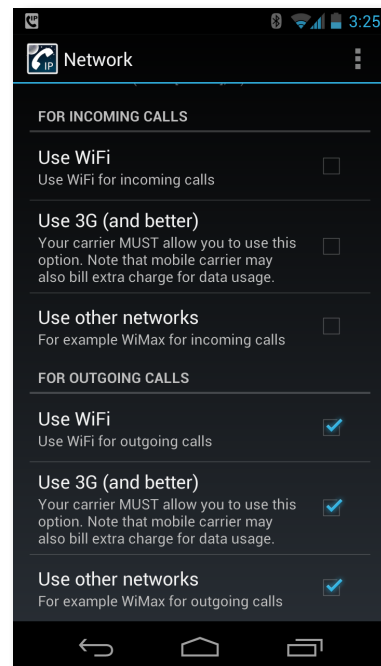
Choose **Network** and scroll down to the section called **For Incoming Calls**. Make sure all the boxes are unchecked. Make sure all the boxes under **For Outgoing Calls** are selected.

Click **Back** a few times to escape and return all the way back to the dialing screen.
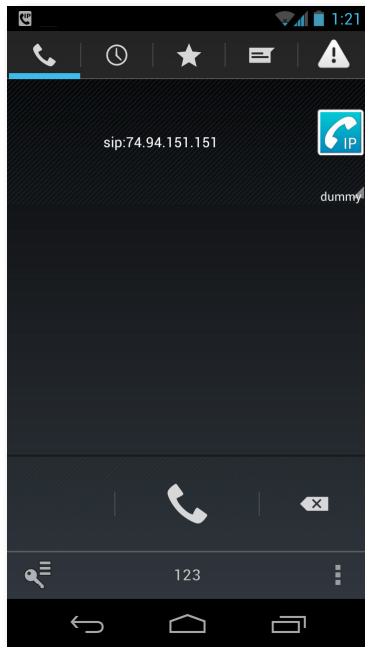
![COMREX]

Figure 6

Figure 7

You can now make a test call to your ACCESS. Select the **txt** option on the bottom bar, then key in the ACCESS info in the following format:

sip:<ip address> (don't include the <>)

Press the **Call** key (or **Dial** icon) and you should complete a call to your ACCESS using the AAC-ELD algorithm. If this call fails, it's time to explore options like STUN and moving your ACCESS to a public IP address.

Making it Simple

The last step is to create a "one button" dial widget on the Android home screen to reach your codec.

Getting back into the **Settings** menu on CSipSimple, choose the **Filters** option and select your dummy account. Choose **Add Filter**.
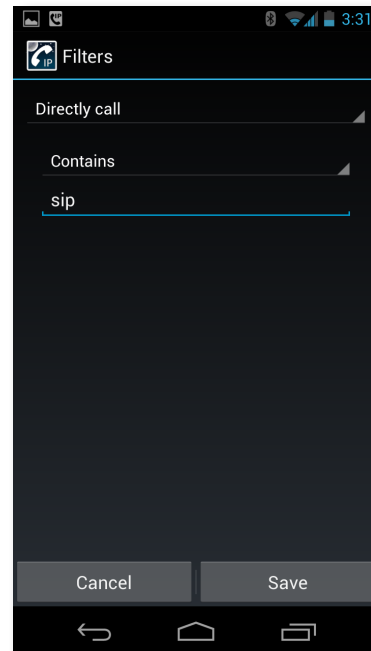
We're going to create a rule within CSipSimple that when asked to dial a SIP call from Android, it will go ahead and do it "no questions asked".

Choose the first field of the filter and set it to **Directly Call**.

Choose the second field of the filter and set it to **Contains**.

In the third field, enter the word *sip*. Save your entry.

You can exit out of CSipSimple completely now (Hint: Do this by repeatedly pressing the **Back** key--pressing the **Home** key leaves the app running in the background, and affects battery life). The CSipSimple Icon appears in the notification bar if the app is still running in the background.

If you're an Android expert, you can now go ahead and create a phonebook entry with your codec's IP address under **Internet Call**. Alternately, there's an additional Comrex Android App that makes this even simpler. It's called Comrex ARC2 (Android Reporter Codec 2) and is available on the Play Store.

ARC2

The first time you launch the ARC2 app, it will prompt you for your codec's IP address and the account to use. Enter the raw IP address without any header or footer, and choose your dummy account.

From now on, whenever the ARC2 app is launched, it will launch CSipSimple behind the scenes and make a call to your codec. When the call is hung up, CSipSimple will end and return your phone to normal use.

In order to change the parameters of ARC2, you'll need to go into your phone's setting, and choose **apps**->**Comrex ARC2**. Clicking **Clear Data** will allow you to re-enter the dialing parameters.

Put Comrex On The Line.

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