Comrex FieldTap Instructions (for use with codecs)

About FieldTap

FieldTap is a smartphone app that allows users to make live, high quality audio connections to Comrex codec hardware, currently ACCESS and BRIC-Link products. **FieldTap** is designed to be very simple to set up and use. It delivers studio quality sound in each direction of the call. **FieldTap** uses the SIP protocol and the Opus audio encoder. It's a free download on the **Apple App Store** for **iOS** and the **Google Play Store** for **Android**.

About SIP

When connecting Comrex codecs between each other, the SIP protocol is normally not used. For this reason, you must enable SIP modes on your Comrex codec and, if necessary, make sure several additional ports are open to the codec in your router or firewall (if you have one).

SIP connections can be made two different ways—with or without the help of a SIP cloud server.

For connections with a server, one or both ends will be pre-registered with the server and the server will help with the handshake at the beginning of the call. In some cases, the server will also proxy the call audio between the two points. We call this "registered mode" and it can help make connections where there are challenges with routers and firewalls in the mix. This is shown in **Figure 1**.
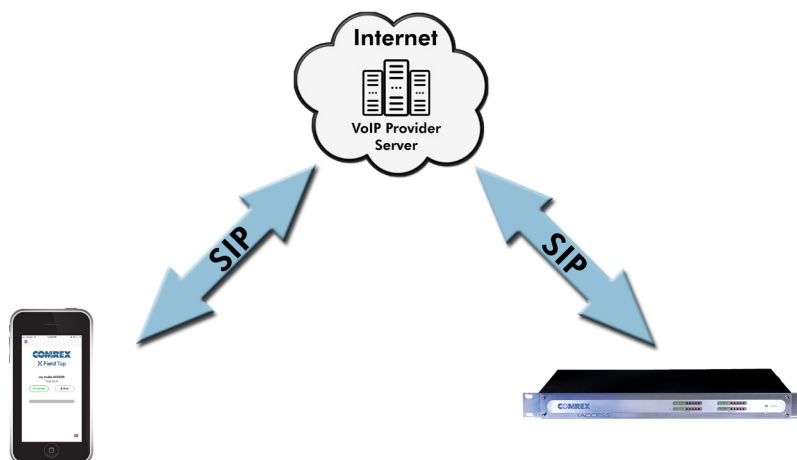

FIGURE 1

A simpler connection mode allows **FieldTap** to make an Internet call directly to the IP address of the studio codec. We call this "unregistered" or "peer-to-peer" mode. This method is quick and simple so it is the recommended connection mode for **FieldTap**. It does require that the proper ports be open on the studio side, but that's a one-time configuration. Unregistered mode is shown in **Figure 2**.
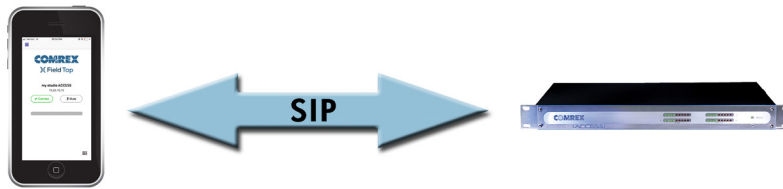
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Quick Start in Unregistered Mode

Assuming your Comrex hardware and firewall are set up for accepting SIP calls, **FieldTap** is very easy to use. Download and install it from your app store, and launch the app. The first screen shown prompts you to input the details for your first "contact" or codec destination. Choose the "**+**" icon and enter information into the first two fields:

1. **Name** - A familiar name for this destination

2. **Number/SIP URI** - The IP address of the Comrex codec

Ignore the "**send PIN on connection**", and make sure "**Make Active Contact**" is chosen. Save the new contact. Now select the menu button in the upper left corner and choose the "**Home**" screen. It will look like **Figure 3**.
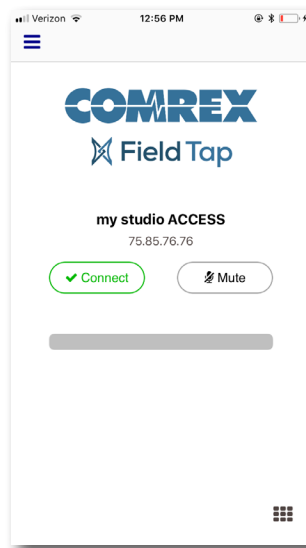
FieldTap offers two buttons: "**Connect**" and "**Mute**". Pressing "Connect" will attempt a call to your codec. Once connected, the mic level bar under the buttons will become active, and the "**Connect**" button will change to "**Stop**". Press "**Mute**" to temporarily mute the mic, or press "**Stop**" to end the call.

Unregistered Mode Advanced Options

Two options in the "**Settings**" menu may be relevant for unregistered mode. By navigating (via the **Main Menu** button on the top left) to "**Settings**", then "**Advanced**", you'll see an option labeled "**Caller Name**" as shown in **Figure 4**.
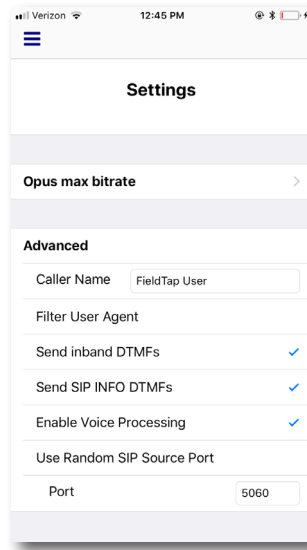

FIGURE 4

Put your name here (or alias of your choice) and this name will appear in the incoming entry on the codec's Web-based Interface as shown in **Figure 5**.

FIGURE 5

Also in **Figure 4**, you'll see the bottom option called "**use random SIP source port**". By disabling this option, you can specify the **UDP/IP** port used by the phone to make the outgoing SIP connection. Default is **5060**, as this is the "well known" SIP port. Unless you have specific firewall issues, you should leave this setting on.

Registered Mode

Registered Mode involves registering the **FieldTap** app with a cloud-based VoIP provider or a PBX. Lots of free SIP providers exist, including **Linphone**, **Callcentric**, and **OnSIP**.

You'll need registration credentials to input into **FieldTap** from your provider. These usually consist of a **User Name**, **password**, and **domain URL**. On some providers (e.g. **OnSIP**), an **Auth Username** and **outbound proxy** address are also needed.

The credentials are entered into **FieldTap** in the **SIP Settings** menu, found in the main selection menu in the upper left corner of any screen. If advanced entries are required, they can be found under the "**Advanced**" option as in **Figure 6**.
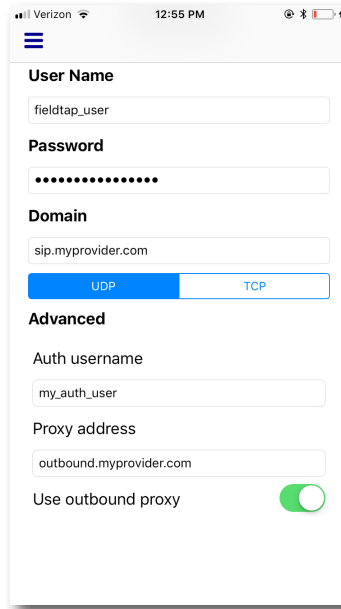
Once these entries are made, **FieldTap** will attempt to register with your provider. The status of the registration will be displayed in a bar at the bottom of the home screen as shown in **Figure 7**.
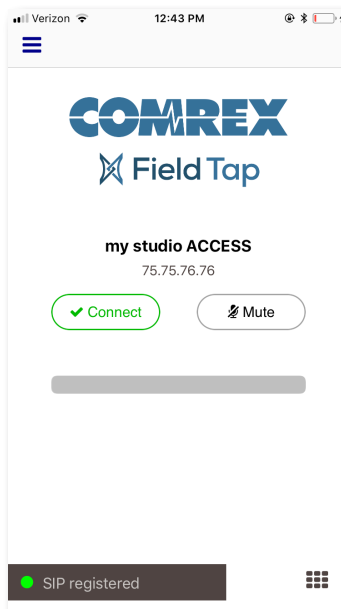
For registered mode to be useful, the codec in the studio should also be registered to a provider. This way, rather than calling via an IP address, a contact can be created with the URI of the studio codec. SIP URIs look like email addresses. E.g., **mycodec@callcentric.com**.

Registered mode keeps a channel open between **FieldTap** and your provider. It's also the only way to receive incoming calls from studio codecs to **FieldTap**. If **FieldTap** is registered and detects an incoming call, it will deliver an audible ring and show a call accept screen as shown in **Figure 8**.
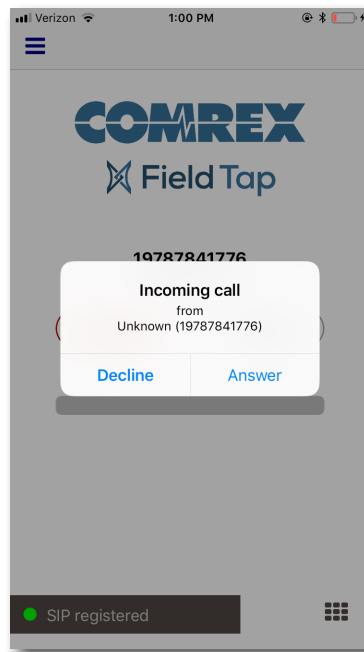
Only one set of SIP credentials can be applied to **FieldTap**. To revert to unregistered mode, remove all credentials from the SIP Settings entries.

FieldTap Behavior on Incoming Mobile Calls

If **FieldTap** is connected while an incoming mobile phone call is received (to the phone's main mobile number) it will continue operating while the phone's call acceptance screen is displayed. The phone will also ring normally. If the new call is accepted, **FieldTap** will end the current call. If it is declined, **FieldTap** will continue its connection until terminated.

Setting up Comrex Codecs to Work with FieldTap

1) Firmware

It's highly recommended that you update ACCESS Rack or BRIC-Link to firmware **4.0p17** or higher to work with **FieldTap**. This release has special options that allow the codecs to accept SIP calls from the Internet at large without being subjected to random "robo-calls" that may tie up your hardware for a short time.

2) Routers and Firewalls

For those users that have ACCESS or BRIC-Link on a local LAN, and not tied directly to a public IP address, you'll probably need to make some changes to your network infrastructure.
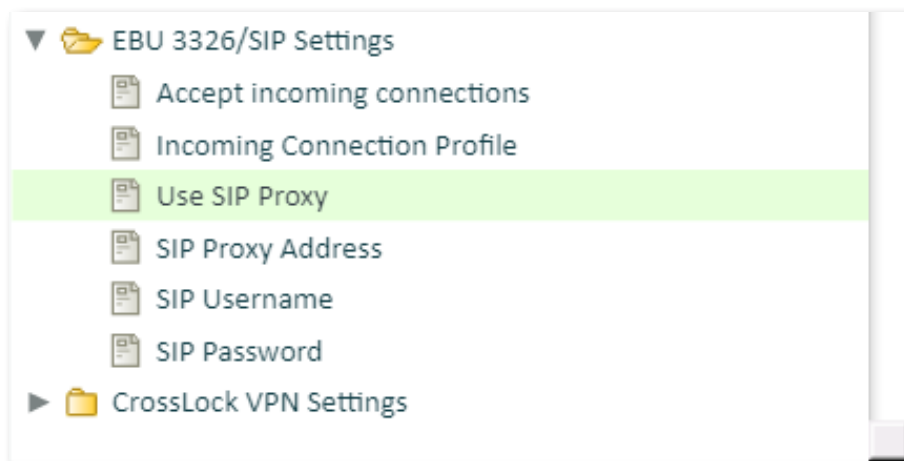
If allowed, you should enable the "**SIP ALG**" function on your router or firewall. This will allow your router to detect SIP traffic and open the correct ports to your codec. If you are running your studio side codec in "registered" mode (registering with a VoIP provider), this is often all that is necessary.

But even if SIP ALG is enabled, if you're not registering to a SIP provider you'll need to forward port **UDP 5060** to your codec hardware. This is because the SIP calls will be incoming, so your router needs to know where to send incoming connect requests. If no SIP ALG is available, forward all these ports to your codec: **UDP 5060**, **UDP 6014**, and **UDP 6015**. If you are unsure of how to do so, please contact your IT administrator, or **techies@comrex. com**, for suggestions.

3) ACCESS/BRIC-Link Settings in Unregistered Mode

To allow incoming SIP calls to ACCESS and BRIC-Link, you need to set the hardware to accept SIP calls, which are set not to answer by default.

- Log into the Web-based Interface
- Click on **System Settings** tab
- Expand ***EBU 3266/SIP Settings**
- **Accept Incoming Connections**: Set to Enabled
- Click "**Apply**" to save

4) ACCESS/BRIC-Link Settings in Registered Mode

If you wish to register your ACCESS or BRIC-Link with a VoIP provider, obtain the credentials from your provider. These usually consist of a **User Name**, **password**, and **domain URL**. On some providers (e.g. **OnSIP**), an **Auth Username** and **Outbound Proxy Address** are also needed.
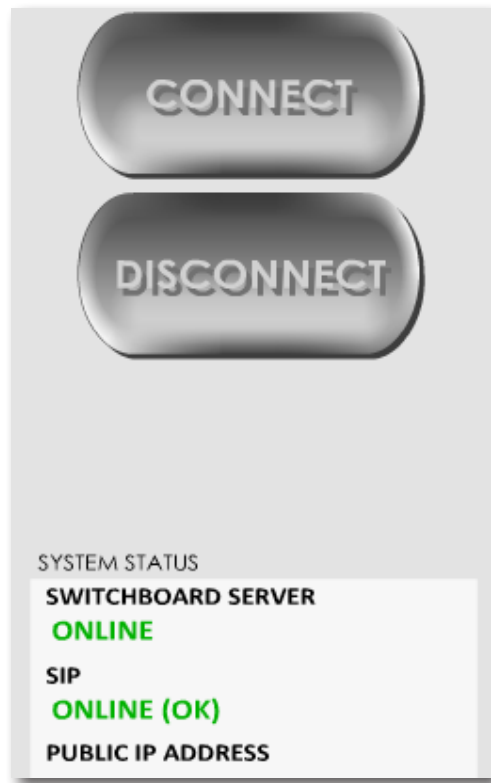
Open the codec's Web-based Interface and follow the instructions for "**unregistered mode**" above. In addition, select "**Use SIP Proxy**" and choose "**enable**" and "**apply**". The fill in the following fields:

- **SIP Proxy Address** (sometimes called "domain")
- **SIP Username**
- **SIP Password**

Clicking "**apply**" after each.

If your provider also requires an **"Auth username"** and/or **"Outbound proxy"**, click the **"Advanced"** box and enter those values in the correct fields.

Once these settings are applied, your codec will attempt to register with the VoIP provider. The status of this registration is now shown in the status box on the **"Connections"** tab of the Web-based Interface.

Once successfully registered, you should be able to call your codec from **FieldTap** by using the URI of the account rather than the IP address. URIs usually consist of the registered **username** and the **proxy/domain address** joined with the @ symbol (e.g. **mycodec@sip.myprovider.com**).