Professional IP Audio Codec Compatibility

By the Codec Answer Guy

Q: Hey CAG, can I connect Comrex IP audio codecs to other brands?

A: Yes. But there's a lot to know! Interconnecting codecs is very complicated, and it takes some work.

Q: Wait, there isn't a simple, hassle-free and reliable way to interconnect brands?

A: It feels like there should be, right? But due to a range of factors, there isn't, and it's probably not going to happen any time soon.

Here's why:

1. Cooperation is great, but we live in a world where manufacturers are competing for business. Manufacturers need to maintain a technical edge over their competition, and that means keeping secrets. It would be unreasonable to ask a manufacturer to disclose technical details of its products, including the way it transports packets over the network.

2. Given point 1, the best way to achieve compatibility is to establish an independent standard for manufacturers to follow. This independent standard would be included on codecs as an alternate mode to the manufacturer's proprietary transport. But for every manufacturer to be able to meet the standard, they would need to compromise and reduce features.

3. There are some universal standards that exist today! But they rely on protocols (like SIP) that require extra configuration from an IT perspective, and special settings within the codec itself. They're neither simple nor hassle-free.

Q: Well, that's a bummer.

A: Even though we won't see a simple way to interconnect any time soon, it is still possible with a little bit of time and effort!

Q: I've heard a little bit about the interconnect standard - what is that?

A: EBU Tech 3326, a joint effort by the European Broadcast Union and manufacturers, defines the way codecs should interoperate. It standardizes the encoders, transport protocols and signaling required for interoperation.

Q: What doesn't it do?

A: It does not address "premium" features added to codecs like presence, dual-network streaming and error-correction. So none of these features are supported by manufacturers in "compatible mode".

Q: Does everyone support it?

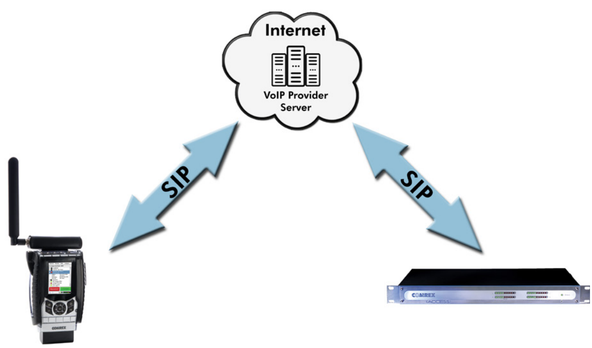
A: Market forces have been successful in driving nearly every manufacturer to support Tech 3326 in some form. Tech 3326 also has the added benefit of relying on the SIP protocol for signaling, which is used almost universally by Voice-over-IP products, often making them compatible with your codecs.

Q: What about SIP? Why doesn't everyone just use that?

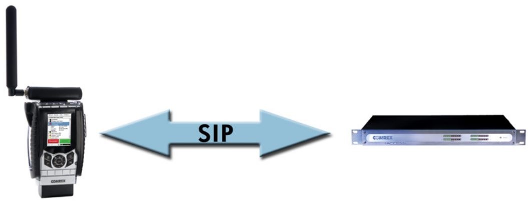
A: SIP/EBU Tech 3326 isn't perfect. It requires that the user open multiple incoming ports on their network, and these are different ports than codec brands will use when connecting between themselves. And because it's an alternate mode to the "default" of most codecs, you usually have to configure the parameters of outgoing calls to use it. And you can't use presence or traversal servers (like Comrex Switchboard) to make connections. But under most circumstances it can be made to work in some way.

Q: How can I learn more about SIP?

A: Comrex has published a primer on the topic and it's available on our website (www.comrex.com). The main difference between VoIP/SIP and Tech 3326/SIP is the use of a SIP server. This is almost always part of a VoIP arrangement—the VoIP device will register with a cloud server (or PBX) at startup and maintain a communications channel.



For most codec users, the server will be skipped, as it makes things more complex. SIP devices have the ability to connect directly together without a server, and since this process is very much like the way codecs already connect, it makes for the easiest transition.



Q: Cool! How can I get started?

A: Well, I can describe the Comrex side of things. The "other brand" side may have differences, but the idea is the same.

Whether you intend to make or receive calls from the Comrex codec, you'll need to forward some ports (unless your codec is already out on the Internet without any router or firewall). The codec will expect some unsolicited data from the Internet, and your router will need to know where to send that data. Unlike with Comrex "BRIC-Normal" calls (the protocol used between Comrex devices), even outgoing callers need ports open. This is because SIP uses different ports for signaling and for caller audio.

Q: I can do that! Which ports do I need to forward?

A: The ports to open on the Comrex side are in the following table. All ports are **UDP** only. **5060** is a "well known" SIP port and should be universal among codecs. The RTP ports are custom to each manufacturer, and probably differ (but that's OK). As noted, it is not usually necessary to forward **UDP 5060** if only outgoing calls are desired.

Port	Outgoing	Incoming	Function
5060 UDP	No	Yes	SIP signaling
6014 UDP	Yes*	Yes*	RTP
6015 UDP	Yes*	Yes*	RTCP

*One additional point on port forwarding—the RTP and RTCP ports might not have to be forwarded if your router employs a SIP ALG (Application Level Gateway) and it's enabled. In this case, the router is smart enough to "sniff" the SIP signaling channel and forward these for you on demand.

Q: I'm not sure how to forward ports. Any help?

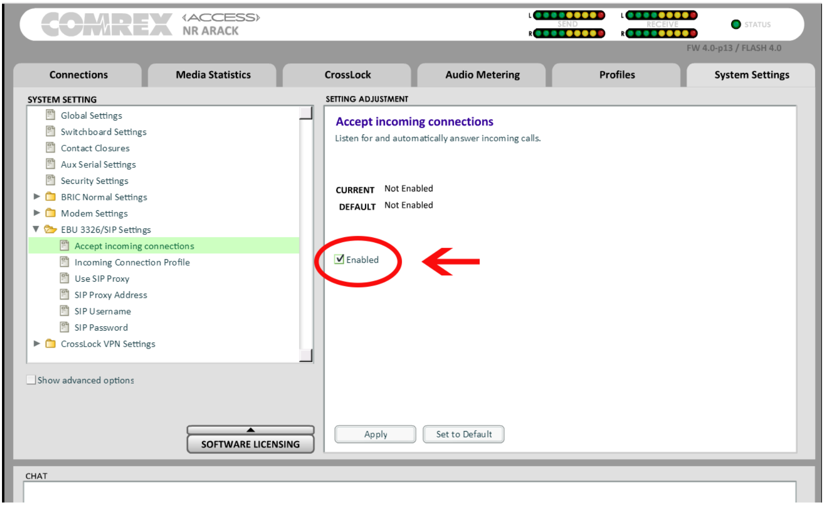
A: Really good information on how to forward ports on popular routers can be found on the website www.portforward.com.

Q: All set! Time to connect?

A: Not quite. In addition to the router settings, there are some settings within your codec you'll need to change from the defaults in order to make or receive SIP connections.

Incoming calls

In the Web-based interface of your ACCESS or BRIC-Link codec (or on the touch screen interface of your portable), go to **System Settings->EBU 3326/SIP Settings->Accept Incoming Connections** and set the value to "**Enabled**".



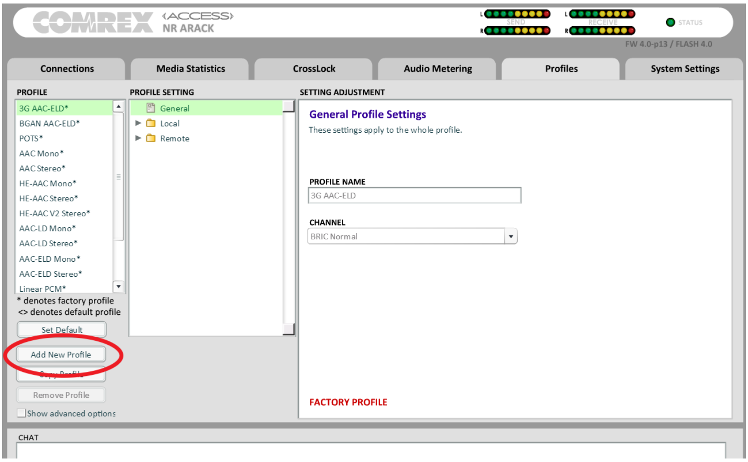
The codec that calls must:

1. Be using the EBU 3326 protocol
2. Have a SIP ALG on their router (or the proper ports open for their codec)
3. Offer an encoder that is supported by Comrex (e.g. AAC family, Opus or G.722)

Outgoing Calls

For outgoing calls on Comrex codecs, you must build an outgoing profile that uses the EBU 3326/SIP transport layer.

New profiles are built on the Comrex user interface by selecting the **Profile** tab and choosing "**Add New Profile**" below the existing profile list.



The important element in creating this profile is under the **General** section of the profile creation tool, with an option labeled **Channel**. By default, this is set to "**BRIC Normal**", which chooses the normal Comrex transport. This must be changed to the "**EBU 3326/SIP"** option to be compatible with other brands. Once this is done, you can enter the **Local** option and choose an encoder that is compatible with the brand being called. It is usually best to leave all other options at their defaults.

Once the profile is created, a new outgoing remote can be created in the usual way on the **Connections** tab. Set the IP address of the target codec and choose the new EBU 3326/SIP profile. Calls should complete if:

1. The target codec is on the open Internet or has the SIP port forwarded to it (usually **UDP 5060**)
2. The target user has a SIP ALG on their router (or the proper RTP ports open to their codec)
3. It can support the encoder offered by the Comrex profile (E.g. AAC family, Opus or G.722)

Q: Have you tested any of this?

A: Somewhat. Here's what we know:

Telos Z/IP One - This codec offers AAC family and G.722 modes. For reasons of backward compatibility, Telos chose to implement their AAC encoder using an MPEG2 protocol rather than the MPEG4 as specified in EBU 3326. For this reason, calls will only connect using G.722 mode at this time. Telos has promised a future update to implement MPEG4.

Tieline - This codec offers G.722, and some models offer AAC family and Opus encoders. G.722 and AAC modes have been tested to work. Tieline opted to implement a subset of the Opus encoder, so we at Comrex implemented compatible encoders on our side. These are found in the profile settings under **Opus CBR** (for constant bit rate).

Others - Most of the time, given compatible encoders and EBU 3326, codecs should interoperate if the proper SIP ports are forwarded on each side. If you experience one-way audio, suspect errors in port forwarding. And feel free to report success and failure to **techies@comrex.com** to help others achieve compatibility!

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