Tieline
The Codec Company

ViA

Codec User Manual

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Table of Contents

1 Warnings & Safety Information

1. The power cable and battery must be removed from the device for Power Disconnection.
2. Remove phone or ISDN cables from the codec before removing a module or servicing.

THUNDERSTORM AND LIGHTNING WARNING:

DO NOT USE Tieline codecs during thunderstorms and lightning. You may suffer an injury using a phone, Tieline codec, or any device connected to a phone during a thunderstorm. This can lead to personal injury and in extreme cases may be fatal. Protective devices can be fitted to the line, however, due to the extremely high voltages and energy levels involved in lightning strikes, these devices may not offer protection to the users, or the Tieline codec and equipment connected to the codec.

Secondary strikes can occur. These secondary strikes are induced by lightning strikes and also produce dangerously high currents and energy levels. You only need to be near an object struck by lightning to lead to personal injury or damage to equipment. e.g. if you are located near a lighting tower at a sports facility, water features and drains on golf courses, you may be affected by these secondary strikes.

Damage to personnel and Tieline codecs may occur during thunderstorm, even if the codec is turned off but remains connected to the phone or ISDN system, LAN or the power.

ANY DAMAGE TO A TIELINE PRODUCT CAUSED BY LIGHTNING or an ELECTRICAL STORM WILL VOID THE WARRANTY.

DIGITAL PHONE SYSTEM WARNING:

DO NOT CONNECT THE ANALOG POTS MODULE TO A DIGITAL PHONE SYSTEM. PERMANENT DAMAGE MAY OCCUR! If you are unfamiliar with any facility, check that the line you are using is NOT a digital line. If the Tieline codec becomes faulty due to the use of a digital phone system, the WARRANTY WILL BE VOID.

SAFETY PRECAUTION:

- Any procedures that involve opening panels or changing components must be performed by qualified service personnel only.

SERVICING WARNINGS:

- Do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so.
- All work should be carried out by suitably qualified personnel.

LINE VOLTAGE:

Before connecting the AC adapter to the power line, make sure the voltage of the power source matches the requirements of the device. Refer to the device Specifications for information about the correct power rating for the unit.

WARNING: To Reduce the Risk of Electrical Shock and Fire

1. All servicing must be undertaken only by qualified service personnel. There are no user serviceable parts inside the unit.
2. DO NOT plug in, turn on or attempt to operate an obviously damaged unit.
3. Ensure that the chassis ventilation slots/holes in the unit are NOT COVERED OR BLOCKED.
4. Do not operate the device in a location where the maximum ambient temperature exceeds 40°C (104°F), or is below 0°C (32°F).

LITHIUM-ION BATTERY WARNINGS:

1. Please read the RRC2057 battery user manual shipped with this product before use. It includes very important safety, charging, operational and disposal information. This user manual can also be downloaded at http://www.rrc-ps.com/.
2. For safety reasons, the battery is prevented from discharging (i.e. from powering the codec) if the internal temperature reaches a pre-set threshold. If a battery temperature warning is displayed, the battery should be removed from the codec and allowed to cool.
3. For safety reasons, the battery is prevented from charging if the internal temperature reaches a pre-set threshold. Move the codec to a cooler location to allow the battery to continue charging.
4. If a battery is installed and the “battery unavailable” icon appears (or the battery icon doesn’t appear at all). The battery should immediately be removed from the codec. Please contact Tieline if this situation persists.
5. The battery may continue to charge when external power is applied to the codec even when the codec is off.
6. When external power is not being applied to the codec, the battery will discharge slowly even if the codec is off. To avoid depleting the battery it should be removed from the codec when not in use.

BATTERY TRANSPORTATION

1. This device includes a Lithium-ion battery and it is the owner’s responsibility to ship this device in full compliance with all of the latest applicable transportation regulations. For air transport, refer to current IATA and FAA regulations, as appropriate, and to your carrier for air transport compliance information. For worldwide sea transportation compliance information refer to the IMO-IMDG code (special provision 188). For European road transportation compliance information see ADR (special provision 188).
2. When the codec is first shipped from Tieline to the customer the battery pack is delivered in shipping-mode (status display off, no measurable voltage at the connector).
3. Please request the RRC2057 Material Safety Data Sheet from RRC at http://www.rrc-ps.com/ for additional transportation and regulatory information.

GENERAL WARNINGS:

1. Do not operate the codec on a hot surface.
2. Only operate the codec within the specified environmental conditions. The codec is considered to be in an operational state when external power is being supplied or the battery is installed, even if the codec is off.
3. If the environmental conditions exceed the specified values, the codec should be switched off, external power should be removed and the battery should be removed from the codec.
4. Do not operate or store the codec in direct sunlight for an extended period of time. Heat build-up due to sunlight exposure can cause permanent damage to the codec that is not covered under warranty.

RADIO FREQUENCY (RF) SAFETY INFORMATION:

IMPORTANT: To satisfy radio frequency exposure compliance requirements, the antenna and transmitter in the ViA codec, or cellular module antennas/transmitters, must be at least 20 cm from all persons and must not be used in conjunction with any other antennas or transmitters. Only use the product with the supplied antenna/s. Failure to adhere this may result in the product exceeding RF exposure limits and may void the user’s authorization to operate the equipment.

The device has an internal Wi-Fi antenna which is located at the rear of the unit. For optimum performance with minimum power consumption do not shield the device or cover with any object. Covering the antenna affects signal quality, may cause the product to operate at a higher power level than needed, and may shorten battery life.

Due to the possibility of radio frequency (RF) interference, it is important that you follow any special regulations regarding the use of radio equipment. Follow the safety advice:

- Operating your device close to other electronic equipment may cause interference if the equipment is inadequately protected. Observe any warning signs and manufacturers' recommendations.
- Different industries and businesses restrict the use of cellular devices. Respect restrictions on the use of radio equipment in fuel depots, chemical plants, or where blasting operations are in process. Follow restrictions for any environment where you operate the device.
- Do not place the antenna outdoors.
- Switch OFF your wireless device when in an aircraft. Using portable electronic devices in an aircraft may endanger aircraft operation, disrupt the cellular network, and is illegal. Failing to observe this restriction may lead to suspension or denial of cellular services to the offender, legal action, or both.
- Switch OFF your wireless device when around gasoline or diesel-fuel pumps and before filling your vehicle with fuel.
- Switch OFF your wireless device in hospitals and any other place where medical equipment may be in use.

Sécurité relative aux appareils à radiofréquence (RF)

À cause du risque d'interférences de radiofréquence (RF), il est important de respecter toutes les réglementations spéciales relatives aux équipements radio. Suivez les conseils de sécurité ci-dessous.

- Utiliser l'appareil à proximité d'autres équipements électroniques peut causer des interférences si les équipements ne sont pas bien protégés. Respectez tous les panneaux d'avertissement et les recommandations du fabricant.
- Certains secteurs industriels et certaines entreprises limitent l'utilisation des appareils cellulaires. Respectez ces restrictions relatives aux équipements radio dans les dépôts de carburant, dans les usines de produits

chimiques, ou dans les zones où des dynamitages sont en cours. Suivez les restrictions relatives à chaque type d'environnement où vous utiliserez l'appareil.

- Ne placez pas l'antenne en extérieur.
- Éteignez votre appareil sans fil dans les avions. L'utilisation d'appareils électroniques portables en avion est illégale: elle peut fortement perturber le fonctionnement de l'appareil et désactiver le réseau cellulaire. S'il ne respecte pas cette consigne, le responsable peut voir son accès aux services cellulaires suspendu ou interdit, peut être poursuivi en justice, ou les deux.
- Éteignez votre appareil sans fil à proximité des pompes à essence ou de diesel avant de remplir le réservoir de votre véhicule de carburant.
- Éteignez votre appareil sans fil dans les hôpitaux ou dans toutes les zones où des appareils médicaux sont susceptibles d'être utilisés.

Interference with Pacemakers and Other Medical Devices

Potential interference:

Radio frequency energy (RF) from cellular devices can interact with some electronic devices. This is electromagnetic interference (EMI). The FDA helped develop a detailed test method to measure EMI of implanted cardiac pacemakers and defibrillators from cellular devices. This test method is part of the Association for the Advancement of Medical Instrumentation (AAMI) standard. This standard allows manufacturers to ensure that cardiac pacemakers and defibrillators are safe from cellular device EMI. The FDA continues to monitor cellular devices for interactions with other medical devices. If harmful interference occurs, the FDA will assess the interference and work to resolve the problem.

Precautions for pacemaker wearers:

If EMI occurs, it could affect a pacemaker in one of three ways:

- Stop the pacemaker from delivering the stimulating pulses that regulate the heart's rhythm.
- Cause the pacemaker to deliver the pulses irregularly.

- Cause the pacemaker to ignore the heart's own rhythm and deliver pulses at a fixed rate.

Based on current research, cellular devices do not pose a significant health problem for most pacemaker wearers. However, people with pacemakers may want to take simple precautions to be sure that their device doesn't cause a problem.

- Keep the device on the opposite side of the body from the pacemaker to add extra distance between the pacemaker and the device.
- Avoid placing a turned-on device next to the pacemaker (for example, don’t carry the device in a shirt or jacket pocket directly over the pacemaker).

Vehicle Safety

When using your device in a vehicle:

- Do not use this device while driving.
- Respect national regulations on the use of cellular devices in vehicles.
- If incorrectly installed in a vehicle, operating the wireless device could interfere with the vehicle’s electronics. To avoid such problems, use qualified personnel to install the device. The installer should verify the vehicle electronics are protected from interference.
- Using an alert device to operate a vehicle’s lights or horn is not permitted on public roads.
- UL evaluated this device for use in ordinary locations only. UL did NOT evaluate this device for installation in a vehicle or other outdoor locations. UL Certification does not apply or extend to use vehicles or outdoor applications or in ambient temperatures above 40° C.

Device Maintenance

When maintaining your device:

- Do not attempt to disassemble the device. There are no user serviceable parts inside.
- Do not expose your device to any extreme environment where the temperature or humidity is high.
- Do not expose the device to water, rain, or spilled beverages. It is not waterproof.
- Do not place the device alongside computer discs, credit or travel cards, or other magnetic media. The information contained on discs or cards may be affected by the device.
- Using accessories, such as antennas, that Tieline has not authorized or that are not compliant with Tieline's accessory specifications may invalidate the warranty.

SAFE LISTENING GUIDANCE

WARNING: LISTENING TO AUDIO AT EXCESSIVE VOLUMES CAN CAUSE PERMANENT HEARING DAMAGE. USE AS LOW A VOLUME AS POSSIBLE.

Over exposure to excessive sound levels can damage your ears resulting in permanent noise-induced hearing loss (NIHL). Please use applicable health and safety authority guidelines on maximum exposure limits. As a rule of thumb, avoid extended periods listening to sound pressure levels (SPLs) of 85dBA or higher.

CHINESE SAFETY WARNINGS:

此设备仅限于非热带地区使用

This device must only be used in not-tropical climate regions.

此设备只限在海拔高度低于 2000 米处使用。

This device must only be used at altitude not exceeding 2000 meters.

JAPANESE SAFETY WARNINGS:
Statement for Class A VCCI-certified Equipment:

この装置は，クラスA情報技術装置です。この装置を家庭環境で使用すると電波妨害を引き起こすことがあります。この場合には使用者が適切な対策を講ずるよう要求されることがあります。　VCCI－A

Translation of previous Class A VCCI Statement: This is a Class A product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this equipment is used in a domestic environment, radio disturbance may occur, in which case, the user may be required to take corrective action.

Special Notices for North American Users:
For North American power connection, select a power supply cord that is UL Listed and CSA Certified 3 - conductor, [18 AWG], terminated in a molded on plug cap rated 125 V, [5 A], with a minimum length of 1.5m [six feet] but no longer than 4.5m.

Special Notices for European Users:
For European connection, select a power supply cord that is internationally harmonized and marked “<HAR>”, 3 - conductor, 0,75 mm2 minimum mm2 wire, rated 300 V, with a PVC insulated jacket. The cord must have a molded on plug cap rated 250 V, 3 A.

Interconnection Cabling:

Cables for connecting to the unit's RS232 and Ethernet Interfaces must be UL certified type DP-1 or DP-2. (Note: when residing in non-LPS circuit)

Overcurrent Protection:

A readily accessible listed branch-circuit over current protective device rated 15 A must be incorporated in the building wiring for the power input.

Replaceable Batteries:

The equipment is provided with replaceable batteries, and if replaced by an incorrect battery type, then an explosion may occur.

CAUTION: RISK OF EXPLOSION IF BATTERY IS REPLACED BY AN INCORRECT BATTERY TYPE. DISPOSE OF USED BATTERIES ACCORDING TO THE INSTRUCTIONS.

This equipment is provided with a long life replaceable Panasonic CR2032 model 3V manganese dioxide lithium coin battery. Service personnel should only replace this battery with the same brand and type of battery. If this is replaced by an incorrect battery type, then an explosion may occur. Contact the manufacturer to view the Material Safety Data Sheet for this battery.

This equipment is provided with a replaceable RRC2057 Lithium-ion battery. Service personnel should only replace this battery with the same brand and type of battery. If this is replaced by an incorrect battery type, then an explosion may occur. Contact the manufacturer to view the Material Safety Data Sheet for this battery.

End of Life Statement

Tieline hereby declares that all materials, components and products supplied are in full compliance with RoHS & WEE directives. This product must be disposed of according to local laws and regulations. Because the product contains a battery it must be disposed of separately from household waste. Do not incinerate, but take it to a recycling facility.

Warranty and Disclaimer

This equipment manufactured by Tieline is warranted by Tieline against defects in material and workmanship for two years from the date of original purchase. During the warranty period, we will repair or, at our option, replace at no charge a product that proves to be defective, provided you obtain return authorization from Tieline and return the product, shipping prepaid, to Tieline. For return authorization, contact Tieline's US or Australian office (see www.tieline.com).

This Warranty does not apply if the product has been damaged by accident or misuse or as the result of service or modification performed by anyone other than Tieline. With the exception of the warranties set forth above, Tieline makes no other warranties, expressed or implied or statutory, including but not limited to warranties of merchantability and fitness for a particular purpose, which are hereby expressly disclaimed. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at www.tieline.com before using this product.

In no event will Tieline, its directors, officers, employees, agents, owners, consultants or advisers (its “Affiliates”), or authorized dealers or their respective Affiliates, be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

Whilst every effort has been made to ensure the accuracy of this manual we are not responsible for any errors or omissions within it. The product specifications and descriptions within this manual will be subject to improvements and modifications over time without notice, as changes to software and hardware are implemented. This codec can provide high voltages on inputs and suitable broadcast equipment must be used at all times. Tieline takes no responsibility for any damage to equipment attached to the codec.

Battery Warranty

Tieline expressly disclaims any and all implied warranties on the RRC Li-ion Smart Battery Pack RRC2057. The manufacturer's warranty applies. Contact the battery manufacturer for any warranty claims. To contact the battery manufacturer visit their website at http://www.rrc-ps.com/.

2 How to Use the Documentation

Manual Conventions

Warnings: Instructions that, if ignored, could result in death or serious personal injury caused by dangerous voltages or incorrect operation of the equipment. These must be observed for safe operation.

Cautions: Instructions warning against potential hazards, or to detail practices that must be observed for safe operation and to prevent damage to equipment or personnel.

Important Note: Information you should know to connect and operate your codec successfully.

Information specific to IP connections.

Information specific to ISDN connections.

Information specific to POTS connections.

Typographic Conventions

- Codec software elements are in Arial bold, e.g. **Contacts**
- Codec hardware elements are in bold Capitals, e.g. **KEYPAD**
- Codec button states are in bold capitals and surrounded by square brackets, e.g. **[ON]** or **[OFF]**

Codec Configuration Descriptions

Codec configuration descriptions primarily focus on tapping the **TOUCH SCREEN** because this is generally simpler than using the **NAVIGATION** buttons. However, most settings and operations can be configured by either using the **TOUCH SCREEN**, or **NAVIGATION** buttons and the **OK** button. Toolbox Web-GUI configuration for the codec is included in separate sections within this user manual.

3 Glossary of Terms

AES/EBU	Digital audio standard used to carry digital audio signals between devices
AES3	Official term for the audio standard referred to often as AES/EBU
AES42	AES standard for acoustics - Digital interface for microphones
APN	A gateway between a cellular network and another computer network.
BRI	Basic Rate Interface for ISDN services
CCC	Cloud Codec Controller
CSRF	Cross-Site Request Forgery (CSRF) is an attack that forces a user to execute unwanted actions on a web application in which they are currently authenticated.
DHCP	A network protocol enabling a server to automatically assign an IP address to a device from a defined range of numbers
DN	Directory Number for ISDN
DNS	The Domain Name System (DNS) is used to assign domain names to IP addresses over the World-Wide Web
Domain	A group of computers or devices on a network which are administered with common rules and procedures. Devices sharing a common part of the IP address are said to be in the same domain
DSCP	The Differentiated Services Code Point is a field in an IP packet header for prioritizing data when traversing IP networks
Failover	Method of switching to an alternative backup audio stream if the primary connection is lost.
Fuse-IP	Tieline bonding of IP interfaces to aggregate data
GPIO	General-purpose Input/Output
GUI	Graphical User Interface
HTML5	A markup language used for structuring and presenting content on the internet. It is the fifth major version of the HTML standard.
IFB	Interrupted Foldback/Interruptible Foldback: an intercom circuit consisting of a mix-minus program feed sent to talent, which can be interrupted and replaced by a producer's or director's intercom microphone
ISDN	Integrated Services Digital Network
ISP	Internet Service Providers (ISPs) are companies that offer customers access to the internet
IP	Internet Protocol; used for sending data across packet-switched networks
LAN	Local Area Network; a group of computers and associated devices sharing a common communications link
Latency	Delay associated with IP networks and caused by algorithmic, transport and buffering delays
LIO	Logic Input/Output
MIB	A management information base (MIB) is a database used for managing the entities in a communications network. This term is associated with the Simple Network Management Protocol (SNMP).
Multicast	Efficient one to many streaming of IP audio using multicast IP addressing
Multi-unicast	A multi-unicast program (also known as multiple unicast) can transmit a single audio stream with common connection settings to a number of different destinations.
MSN	Multiple Subscriber Number for ISDN
NAT	Network Address Translation is a system for forwarding data packets to different private IP network addresses that reside behind a single public IP address.

Packet	A formatted unit of data carried over packet-switched networks.
PAT	Port Address Translation is related to NAT; a feature of a network device that allows IP packets to be routed to specific ports of devices communicating between public and private IP networks
POTS	Plain old telephone system: copper phone network infrastructure
PSTN	Public switched telephone network which is another term for POTS (see previous)
PSU	Power Supply Unit
QoS	Quality of Service priority is given to different users or data flows across managed IP networks. This generally requires a Service Level Agreement (SLA) with a Telco or ISP
RTP	A standardized packet format for sending audio and video data streams and ensures consistency in the delivery order of voice data packets
Runtime (edits)	Configuration changes which have not yet been saved, e.g. Matrix Editor edits.
SDP	Session Description Protocol defines the type of audio coding used within an RTP media stream. It works with a number of other protocols to establishes a device's location, determines its availability, negotiates call features and participants and adjusts session management features
SIM	Subscriber Identity Module is an integrated circuit used to identify and authenticate subscribers on mobile cellular devices.
SIP	Session Initiation Protocol is a common protocol which works with a myriad of other protocols to establish connections with other devices to provide interoperability
SLA	Service Level Agreements (SLAs) a contractual agreement between an ISP and a customer defining expected performance levels over a network
SmartStream PLUS	Tieline implementation of redundant IP streaming.
SNMP	Simple Network Management Protocol: Simple Network Management Protocol: a protocol used mostly in network management systems to monitor devices for conditions that warrant administrative attention.
SPID	Service Profile ID for identifying devices over ISDN networks
SPL	Sound pressure level
SSL	Secure Sockets Layer is a security protocol for establishing encrypted links between a web server and a browser for online communication
STL	Studio-to-transmitter link for program audio feeds
STS	Studio-to-studio audio link
STUN	The STUN protocol (Simple Traversal of UDP through NATs) assists devices behind a NAT firewall or router with packet routing. A STUN client generates STUN requests and a STUN server, attached to the public internet, receives STUN requests and sends responses.
TCP	Transmission Control Protocol ensures reliable in-order delivery of data packets between a sender and a receiver
TieLink	Traversal Server used to add Tieline codecs to a TieServer Domain and centralize codec contact list management, by providing self-discovery of codecs within call-groups, and NAT traversal to simplify connections.
TieServer	Centralized servers providing domain management facilities for Tieline applications including the TieServer Console, Cloud Codec Controller and TieLink Traversal Server.
TieServer Domain	A high-level group, associated with a particular broadcaster/customer, that is used to securely demarcate their Tieline assets from other broadcasters/customers. It applies to usage and management of Tieline

	codecs and Report-IT users when using Tieline applications including the TieServer Console, Cloud Codec Controller and TieLink Traversal Server.
TLS	Transport Layer Security is an updated version of SSL.
TTL	Time-to-Live is the setting used in multicast servers to ensure data packets have a finite life and don't cause congestion over networks
UDP	User Datagram Protocol: the most commonly used protocol for sending internet audio and video streams. UDP packets include information which allows them to travel independently of previous or future packets in a data stream
Unicast	Broadcasting of a single stream of data between two points
VLAN	Virtual Local Area Network: partitioning of a single layer-2 network to create multiple distinct broadcast domains
WAN	Wide Area Network; a computer network spanning regions and/or countries to connect separate LANs
WheatNet-IP	Network system that utilizes Internet Protocol to enable audio to be intelligently distributed to devices across scalable networks

4 Items Shipped with ViA

Your new ViA codec is shipped with the following items:

1. ViA codec.
2. 12VDC power supply.
3. Rechargeable Li-ion battery pack RRC2057.
4. Protective case.

Immediately contact Tieline or your dealer if any of these items are missing or damaged.

5 Introduction

ViA connects to, and is compatible with, any Tieline codec supporting IP, ISDN and POTS connections. It is designed to seamlessly integrate with Tieline's Merlin and Merlin PLUS audio codecs to transmit high fidelity, full duplex stereo program audio with a separate bidirectional IFB circuit. ViA connection options include:

1. IP over fiber optic networks, LANs, WANs, the internet and satellite IP.
2. Cellular wireless networks using USB modems or an optional cellular module.
3. Internal Wi-Fi .
4. ISDN using an optional ViA ISDN module.
5. POTS using an optional ViA POTS module.

ViA has an long list of features, some of which include:

- Dual Gigabit (10/100/1000) Ethernet ports with automatic switching for redundancy.
- Dual USB ports for wireless modems and an external USB keyboard.
- Fuse-IP bonding technology to aggregate data from multiple IP interfaces.
- SmartStream PLUS redundant streaming for high reliability over IP networks without Quality of Service.
- Record and playback capability.
- IPv4 & IPv6 compatible and ready.
- Matrix Editor with crosspoint input/output routing.
- Fast charging internal battery and external DC power supply.
- Uncompressed linear PCM audio plus the low-delay, cascade resilient aptX® Enhanced algorithm.
- Other popular algorithms including LC-AAC, HE-AAC v1 and v2, AAC-LD, AAC-ELD, Opus, MPEG-1 Layer II and III, Tieline Music and MusicPLUS, G.722 and G.711.
- Asymmetric encoding.
- SNMP and integrated alarm management.
- HTML5 Toolbox GUI enables remote codec control over WANs.
- Low latency in-band RS-232 auxiliary data channel.
- 4 x Control Port in/outs plus WheatNet-IP LIO compatibility.
- AES3 input (stereo) and support for AES 42 Mode 1 and Mode 2 microphones in input 1.
- Configurable software rules engine via a GUI for Control Port functions.
- Simple, user-friendly touch screen menus.
- Support for multiple languages*.

* Supported in later releases.

Compatibility

ViA connects over IP to any compatible codec brand that supports the EBU N/ACIP tech 3326 standard using SIP and SDP protocols. The codec also connects to competitor ISDN codecs in 'sessionless' mode. ViA is also POTS-compatible with Comrex® Vector, Matrix® or BlueBox® codecs. ViA supports a wide range of commonly used algorithms such as Opus, AAC, MPEG Layer 2, G.722, G.711, aptX® Enhanced and many, many more.

Please see the Connection Guide for instructions on getting connected for the first time.

6 Battery Use and Power Management

ViA has an internal battery slot on the rear panel designed for high performance RRC2057 Lithium-ion batteries. Only use this battery in the codec.

Caution:

1. Please read the important safety and user information in the manufacturer user manuals for both the battery and any external charger purchased separately before use.
2. The internal **BATTERY** is delivered in shipping-mode (status display off, no measurable voltage at the connector). Attach the codec power supply to the **POWER SOCKET** to charge the battery. When you start the charge cycle the **BATTERY** will be activated. Charge fully before first use.
3. If the codec is off the **BATTERY** continues to charge when external power is applied to the codec.

Inserting the Battery

1. Push down on the **BATTERY CASE CLIP** to open the lid to the battery compartment.
2. Insert the **BATTERY** carefully and ensure the grooves at the bottom of the **BATTERY** line up correctly with the bottom of the **BATTERY COMPARTMENT**. Note: It should slide in smoothly.
3. Replace the **BATTERY CASE CLIP** carefully by lining up the two protruding plastic lugs with the base of the compartment, then push the center of the **BATTERY CASE CLIP** to close it fully.

Removing the Battery

1. Push down on the **BATTERY CASE CLIP** to open the lid for the **BATTERY COMPARTMENT**.
2. Pinch the tag on the **BATTERY** to pull it slowly out of the **BATTERY COMPARTMENT**.

Charging the Battery

When the power supply is connected to the codec's **POWER SOCKET** it will charge the internal **BATTERY** in less than 4 hrs.

Battery Indications

The battery has a charge status button to verify its charge state when it is **[ON]**. The **POWER** ⏻ button is green when the power supply is connected to the codec's power socket. When the codec is operating on battery power it provides the following battery level indications:

1. The **Battery symbol** and charge remaining is visible in the **Status Bar** in the top right corner of the **TOUCH SCREEN**.
2. When operating on battery power the **POWER** ⏻ button indications are as follows:

	LED Indication	Battery State
	GREEN LED (Solid)	Battery level is between 21 - 100 %
	ORANGE LED (Solid)	Battery level is between 11 - 20 %
	RED LED (Solid)	Battery level is between 6 - 10 %
	RED LED (Flashing)	Battery level is 5% or lower

Low Battery Headphone Alarm Tones

A low battery alarm is audible in the headphones when the battery level reaches 20%, 10% and 5%.

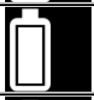
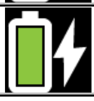
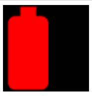
1. Alarm tones are audible in the left headphone output of **HP 1-3**.
2. A warning dialog is displayed on the **TOUCH SCREEN** until acknowledged.
3. The **ALARM LED** flashes until acknowledged, and then turns solid red.

Tap the **TOUCH SCREEN** or touch any codec controls to acknowledge the alarm and stop the alarm tones.

Touch Screen Battery Indications

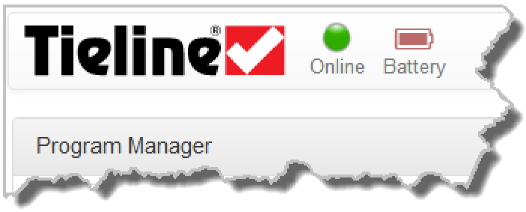
The percentage of battery charge remaining is displayed next to the battery symbol in the **Status Bar** on the **TOUCH SCREEN**. In addition:

- When power is attached to the codec the **BATTERY** symbol is green and the white **Power**  symbol is displayed.
- When operating on battery power the **BATTERY** symbol is white.

Symbol	Battery State
	Power is attached to the codec and the battery is charging.
	The battery is fully charged and no power is attached to the codec.
	The battery is fully charged and the external power supply is in use.
	Unknown battery error while the external power supply is attached. The battery should immediately be removed from the codec. Please contact Tieline if this situation persists.
	Unknown battery error. The battery should immediately be removed from the codec. Please contact Tieline if this situation persists.
	The battery is too hot or too cold. Move the codec to a cooler or warmer location as required.
	The battery has overheated or is faulty. For safety reasons the battery will be prevented from discharging and powering the codec. Plug in external power to prevent imminent shutdown. The battery **MUST** be removed from the codec and allowed to cool.
	An external power supply is attached and the battery is too hot or too cold and not charging. For safety reasons, the battery is prevented from charging if the internal temperature is above or below pre-set thresholds. Move the codec to a cooler or warmer location as required, to allow the battery to continue charging.
	An external power supply is attached and the battery has overheated or is faulty. For safety reasons the battery is prevented from charging, or discharging and powering the codec. The battery **MUST** be removed from the codec and allowed to cool.

HTML5 Toolbox Web-GUI Low Battery Indications

A low battery icon is displayed in the HTML5 Toolbox Web-GUI when the battery level is 20 percent or lower. This flashes until it is acknowledged on the unit itself.



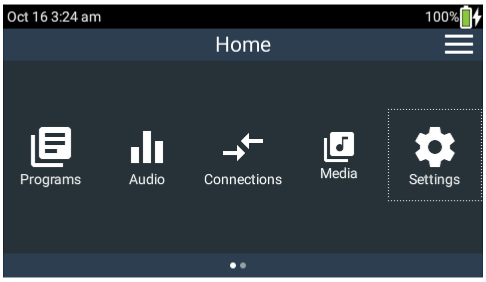
Touch Screen Power Management

By default the **TOUCH SCREEN** has **Screen Sleep** mode enabled to preserve battery power. The default **Screen Timeout** setting is **1 minute**. The **TOUCH SCREEN** will "reawaken" from sleep mode when a button or rotary encoder is adjusted, or the **TOUCH SCREEN** is tapped or swiped. It is also possible to:

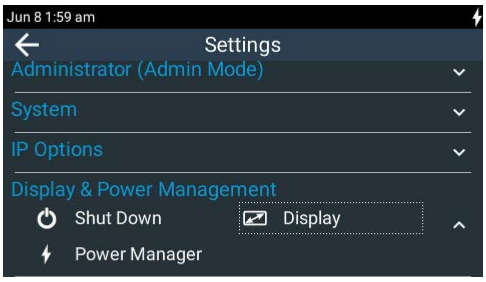
- Adjust **Screen Brightness**.
- Enable **Adaptive Brightness Adjustment**.

To adjust these settings:

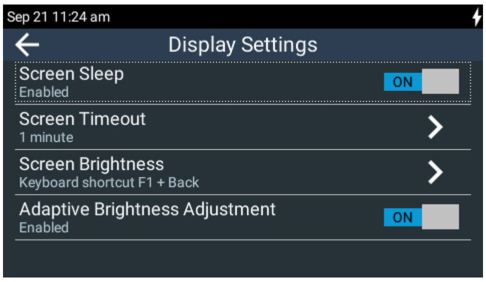
1. Press the **HOME** button to return to the **Home screen**, then tap **Settings**.



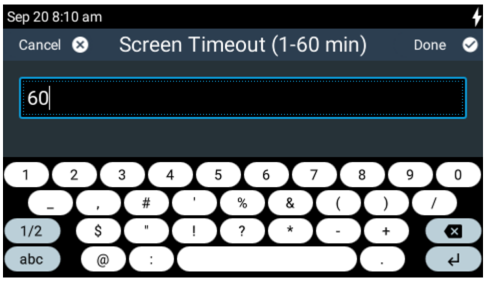
2. Tap to expand the **Display and Power Management** menu and then tap **Display**.



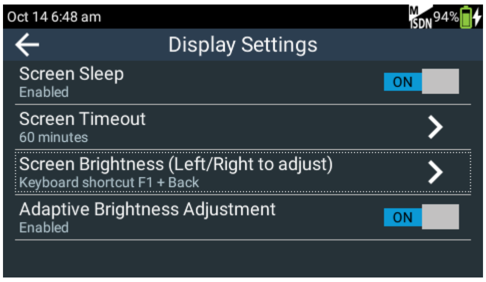
3. Tap the **On/Off** button to toggle between enabling and disabling **Screen Sleep** (default setting **On**).



4. Tap **Screen Timeout** to adjust the number of minutes, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



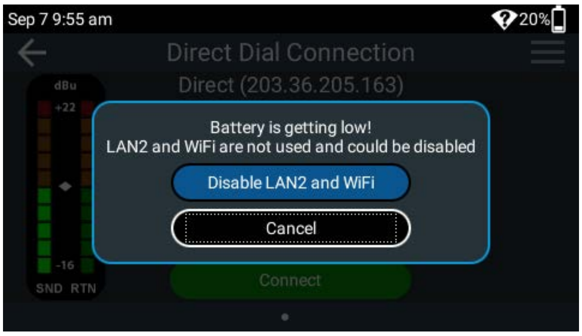
5. Tap **Screen Brightness** and then use the left ◀ and right ▶ **NAVIGATION** buttons to adjust brightness.



6. Tap the **On/Off** button to toggle between enabling and disabling **Adaptive Brightness Adjustment** (default setting **On**). Note: The codec has an ambient light sensor and is able to adjust optimum screen brightness automatically when this feature is enabled.

Advanced Power Management

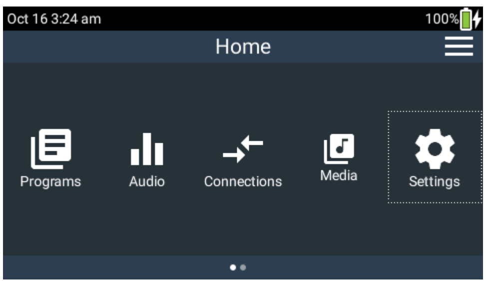
Other ways to conserve battery power when operating without a power supply include the following:

Power Management Mode	Explanation
Audio Power Setting (default setting **High**)	Power consumption can be reduced by lowering the audio output threshold from +22dBU to +16dBU. Tap **Audio Power Setting** to make this adjustment
Low Battery Screen Power Saving (default setting **On**)	This feature is activated when the battery charge level is 15%. The **TOUCH SCREEN** brightness is reduced and 'sleep mode' is enabled with a 1 minute time-out. This setting overrides the **Screen Timeout** setting in the **Display** menu. Note: When power is applied to the codec this feature is overridden.
Low Battery Network Power Saving (default setting **On**)	This feature is activated when the battery charge level is 20%. The codec launches a dialog suggesting any unused network interfaces be disabled to conserve power.  If **LAN2** is being used and **LAN1** is inactive, a dialog will appear and suggest switching to **LAN1** as this consumes less power.

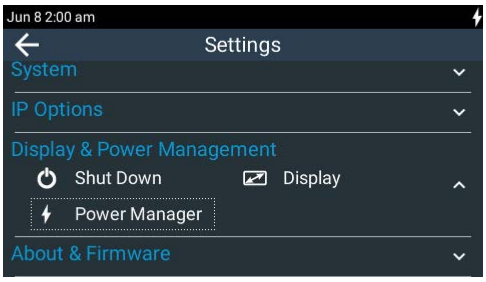
Caution: Low power mode lowers the threshold at which audio will distort. Ensure audio levels do not exceed -6dBFS (+16dBU) in this mode of operation.

To adjust these settings:

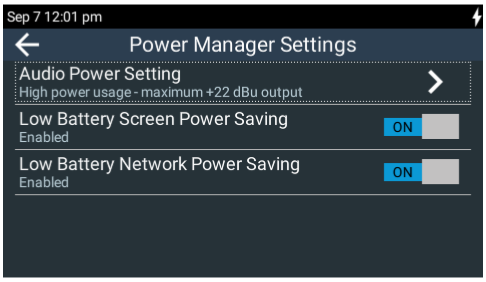
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.



2. Tap to expand the **Display and Power Management** menu and then tap **Power Manager** .



3. Tap to adjust each option.



7 Inserting and Removing Modules

A single module slot is available on the codec rear panel for inserting an optional cellular, ISDN or POTS module into the codec.

ViA codec with an ISDN module installed

Inserting or Removing a Module

Ensure the codec is **[OFF]** when inserting or removing modules. Where possible use anti-static precautions to help minimize the chance of static charges damaging the highly sensitive circuitry. Do not force a module into the codec. Modules should be installed slowly and gently.

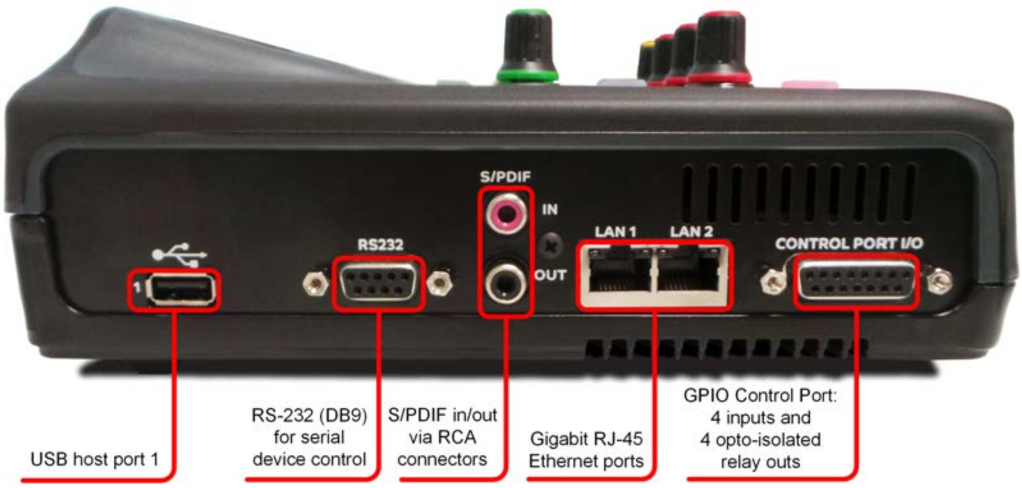
1. Press the **POWER** button and tap **Shut Down** to turn the codec **[OFF]**.
2. Remove the 4 screws from the blanking panel or module installed in the codec.
3. Carefully slide the new module into the module slot and ensure the base of the module remains flat during insertion, to ensure it lines up correctly with the module connector in the codec.
4. Reinsert the 4 screws to hold the module firmly in place.
5. Press the **POWER** button to power up the codec.
6. Press the **HOME** button to return to the **Home** screen and tap **Settings**.
7. Tap **Transport Interfaces** to expand the menu and tap **ISDN Module**, **POTS Module** or **Cellular > Cellular Module** to configure module settings.

8 External Connections

Front Panel Connections

	Connection	Details
1	Analog XLR, AES3 and AES42 Inputs	XLR inputs 1-3 are all balanced mic/line inputs. Input 1 can also be used as an AES3 (AES/EBU) digital input, which accepts both mono and stereo digital AES3 signals. Alternatively, input 1 also supports AES42 Mode 1 and Mode 2 mics. AES input modes can be selected via codec menus. ⚠ **VOLTAGE WARNING:** DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** mode is selected, or equipment may be damaged by high voltages.
2	Headphone Outputs	The codec has three 6.35mm (1/4") RTS stereo headphone outputs labeled as **HP 1 - 3**. These are designed to be used with 600 ohm headphones.

Left Side Panel Connections



	Connection	Details
1	USB Port 1	USB host port 1 for attachment of a supported cellular modem for data connections. Also supports attachment of a USB keyboard.
2	RS232	Nine pin female **RS232** serial data connection for local and remote control of equipment at either end of the link.
3	S/PDIF In/Out (Auxiliary input)	**S/PDIF IN/OUT** auxiliary input and output via RCA connectors.
4	LAN1 and LAN2 Ports	The codec features two Gigabit (10/100/1000) RJ-45 Ethernet ports for IP connections. By default, the codec assumes **LAN1** is the primary LAN connection and **LAN2** is the backup (secondary) LAN connection when in use. LED indications allow identification of LAN speeds as follows: • 1000 Mbps: Green = Blinking / Orange = Off • 100 Mbps: Green = Off / Orange = Blinking • 10Mbps: Green = Blinking / Orange = Blinking
5	Control Port I/O	Four relay inputs and four opto-isolated outputs for machine control via the DB15 **CONTROL PORT I/O** connector.

Right Side Panel Connections

	Connection	Details
1	Stereo Line In	**STEREO LINE IN** is an auxiliary line input option via a 3.5mm (1/8") TRS connector.
2	Outputs 1 and 2	**OUTPUT 1** and **2** are balanced analog XLR line outputs.
3	USB Audio I/O	**USB AUDIO I/O** is an auxiliary input/output option via a micro USB Type-B connector.
4	USB Port 2	USB host port 2 for attachment of a supported cellular modem for data connections. Also supports attachment of a USB keyboard.

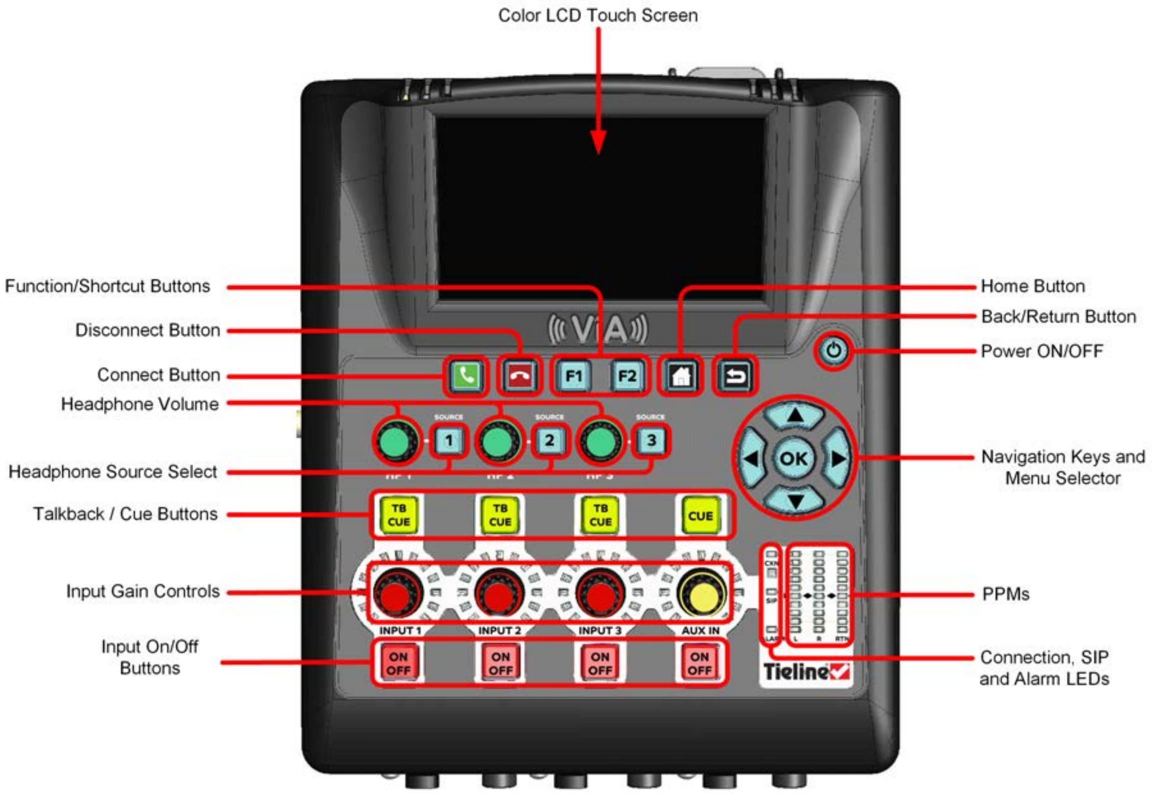
Important Note: Only one auxiliary input option can be selected at a time.

Rear Panel Connections

	Connection	Details
1	Battery slot	Battery slot for RRC2057 rechargeable Li-ion battery pack.
2	SD Card slot	SD card slot for record and playback of audio files, firmware upgrades, codec backup and restore functions, and installation of SSL certificates.
3	Module slot	Module slot for inserting an optional ViA cellular, ViA POTS or ViA ISDN module.
4	XLR Power Socket	4-pin male XLR on the rear panel to attach an external 12VDC 3A power supply.

9 Codec Controls

The top of the codec features input and headphone controls, as well as navigation buttons, a color LCD **TOUCH SCREEN** display and PPM meters.



Navigation Buttons

The codec has four arrow-shaped **NAVIGATION** buttons and the **OK** button buttons for navigating codec menus and adjusting levels and settings. The **OK** button is used to select menu items.

Operation Button Descriptions

	Features	Operation Button Descriptions
	Connect Button	Press to dial/connect
	Disconnect Button	Press to disconnect a connection
F1	Function Button 1	Press to activate codec user functions
F2	Function Button 2	Press to activate codec user functions.
	Home Button	Press to return to the **Home** screen
	Back/Return Button	Press to move back through menus & delete characters
	Headphone Source Select Button	Press to adjust headphone send/return balance, or select headphone monitoring sources
	Headphone Volume	Adjusts the headphone level for each headphone output
	TB/Cue Button	Supports user talkback, or cue source monitoring modes. Note: Cue only on auxiliary input.
	Input Gain Control	Adjusts mic/line input gain for inputs 1 - 3 and the auxiliary input
	Input On/Off	Press to turn each input on or off
	Power On	Press to turn the codec on; press and tap **Shut Down** to turn the codec **[OFF]**, **Reboot** or enter **Screen Sleep** mode.

Front Panel LED Descriptions

LED	LED Description
CXN (Connection)	When dialing a connection the **CXN LED** flashes until the codec connects. Note: It also flashes until all streams/transports are connected when multiple streams/transports are connecting.
SIP	The **SIP LED** flashes orange when registering to an active SIP server account. It illuminates solid orange when the codec has been registered to an active SIP account successfully.
Alarm	The **ALARM LED** flashes red when an alarm is active in the codec. It stops flashing and illuminates solid red after an alarm has been acknowledged.

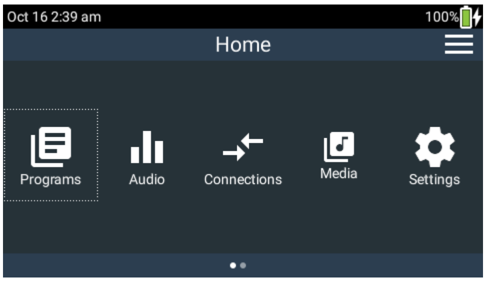
Touch Screen Sleep Mode

By default the codec **TOUCH SCREEN** will enter 'sleep mode' after a minute of inactivity. To adjust this setting go to the **Home screen** and tap **Settings** **> Display and Power Management > Display**. See Battery Use and Power Management for more info.

10 Menu Navigation

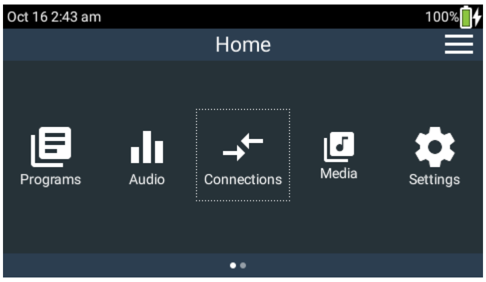
Home Screen Navigation

All main codec menus can be launched from the **Home** screen.



	Features	Home Screen Elements
1	Programs	Tap to view, edit and load Program configurations
2	Audio	Tap to configure audio input settings and input processing
3	Connections	Tap to view the **Connections screen** and create new programs with multiple audio streams and connections; also manage connection/disconnection of audio streams
4	Media	Tap to configure record and playback settings
5	Settings	Tap to configure a range of codec settings

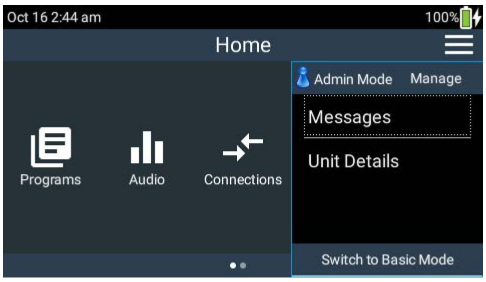
Use the **NAVIGATION** buttons to navigate through menus and press the **OK** button to select a menu item. Currently selected menu items are highlighted with a dotted 'focus' outline.



Press the **HOME** button to return to the **Home** screen from any menu. Press the **BACK/RETURN** button to navigate backwards through menus or delete characters in fields.

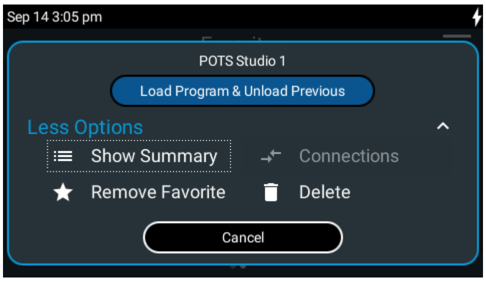
Hamburger Menu

The 'Hamburger Menu' ☰ is used to display menu items relevant to the current screen. For example, tap **Menu** ☰ on the **Home** screen and you are presented with a shortcut to view **Messages**, **Unit Details** and administrator management options.



Navigation Shortcuts

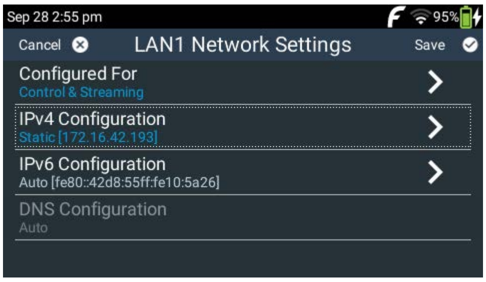
1. For IP and POTS connections negotiate higher bit rates by pressing the **F2** button and then the **NAVIGATE UP** ▲ button while viewing the **Statistics** screen; for lower bit rates press the **F2** button and then the **NAVIGATE DOWN** ▼ button.
2. Press the **F1** button and the **Return** button to activate screen brightness controls. Use the **NAVIGATE LEFT** ◀ and **NAVIGATE RIGHT** ▶ buttons to adjust the level of brightness.
3. Touch and hold a program favorite to view program editing features when [operating in Admin Mode](). This shortcut saves having to navigate to the **Programs** menu to edit program attributes.



Saving Multiple Config Settings

Some menus require all changes to be saved in one go to ensure correct operation of the item being configured. Examples of menus using this approach include **Settings** menus like **LAN1**, **LAN2**, **VLANs**, **SIP Interfaces**, **SIP Accounts**, and **Basic Mode Settings**. When changes are made to these menus, items with unsaved changes are highlighted blue.

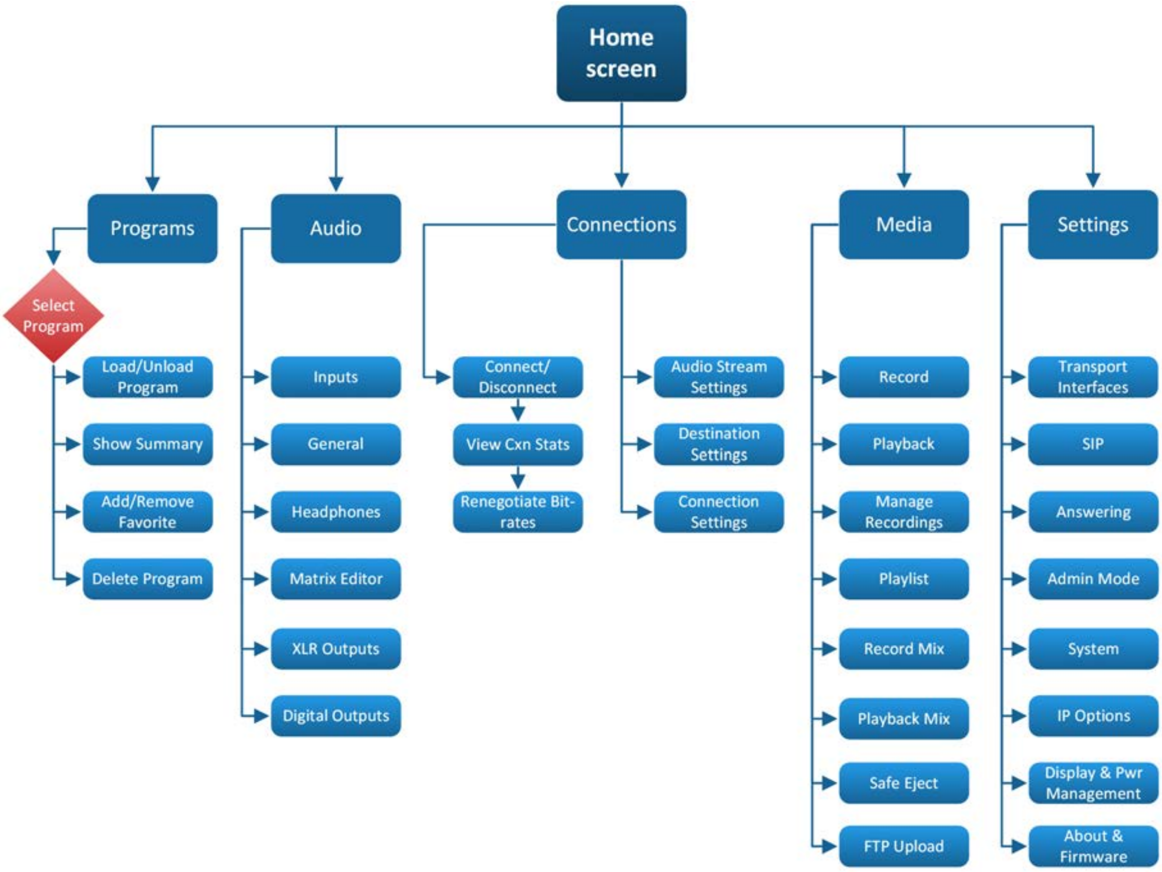
In the following example, **LAN1** has been configured to allow **Control and Streaming** and a static IP address has been configured in the **IPv4 Configuration** menu. To save these pending settings tap **Save** in the top right-hand corner of the **TOUCH SCREEN**.



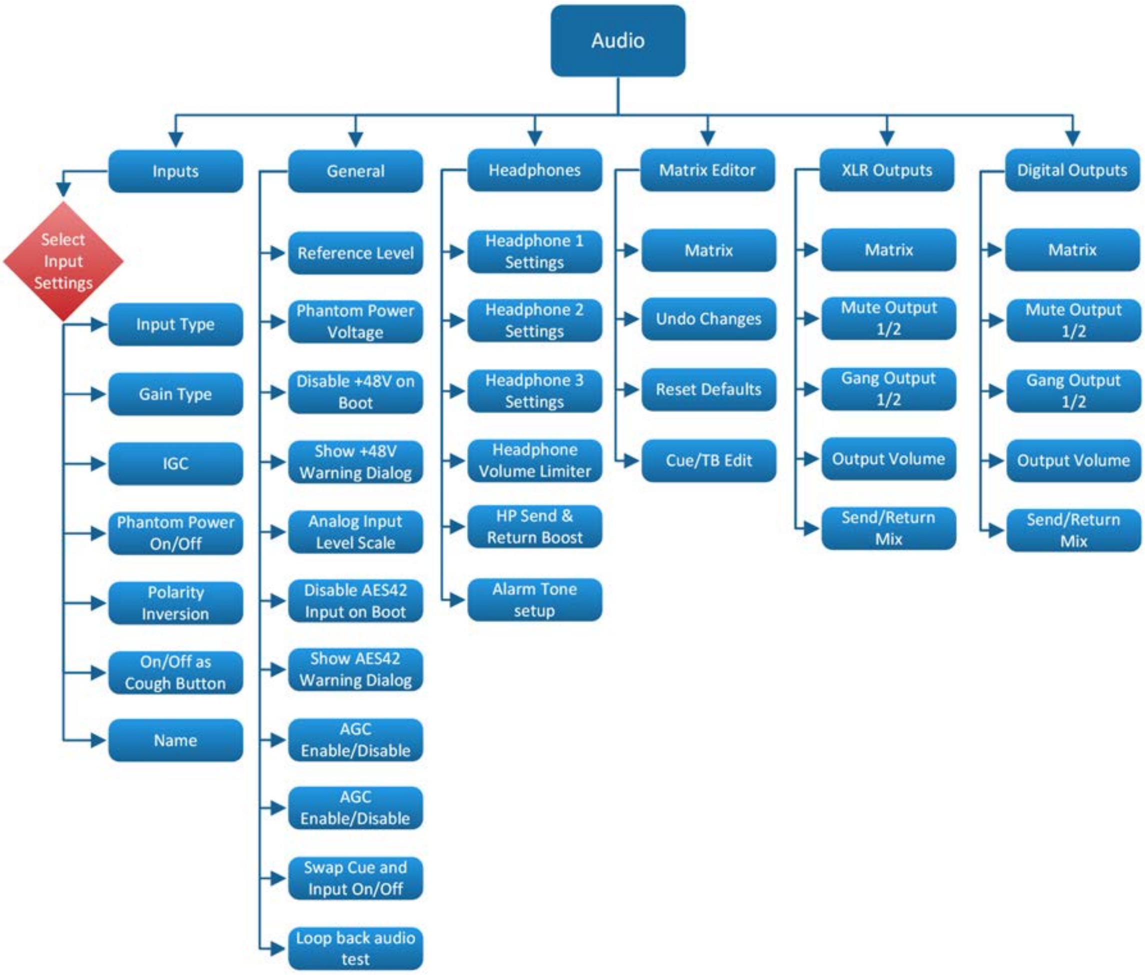
11 Codec Menus

Following is an overview of the menu structures in the codec.

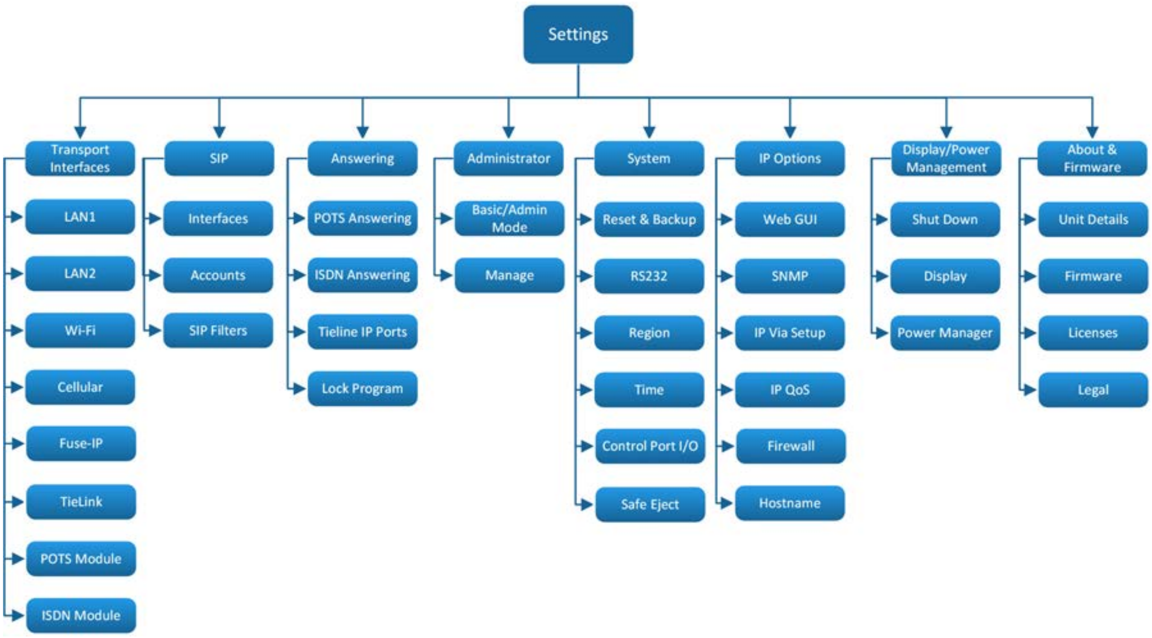
Home Screen Menu Overview



Audio Menu Overview



Settings Menu Overview



12 Wireless IP Connection Options

The codec has multiple IP interface connection options, including:

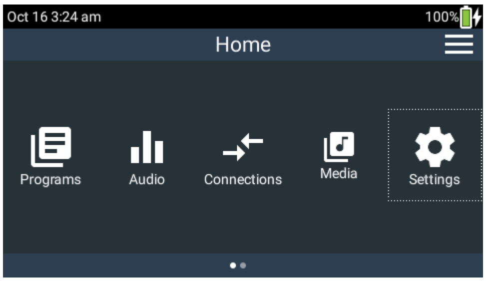
1. **LAN1** Ethernet port (default **Primary** Via interface)
2. **LAN2** Ethernet port (default **Secondary** Via interface)
3. Internal Wi-Fi (default **Tertiary** Via interface)
4. External **USB PORT 1**. Note: for use with a USB modem or tethered smartphone.
5. External **USB PORT 2**. Note: for use with a USB modem or tethered smartphone.
6. Optional Internal Single SIM LTE Module. Note: two antennas are shipped with this module to support diversity.
7. Optional Internal Dual Active SIM LTE Module: Note: Requires firmware v2.20.xx or higher and four antennas are shipped with this module to support diversity.

For more information about prioritizing the interface used when dialing a connection see Configuring Via Interfaces.

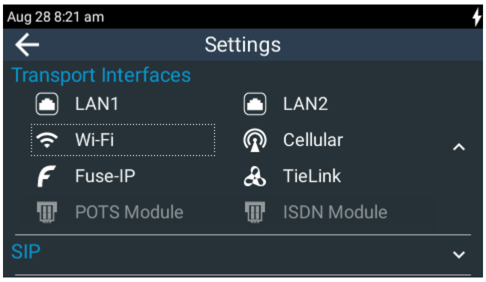
Connecting a Wi-Fi Access Point

To connect the codec to a Wi-Fi network access point:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.

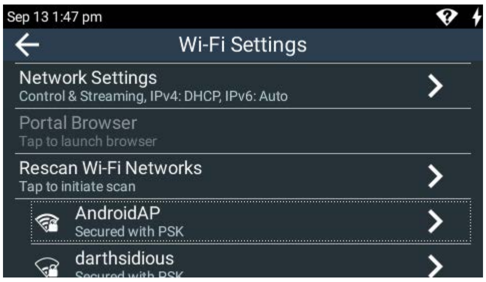


2. Tap to select **Transport Interfaces** and then tap **Wi-Fi**.

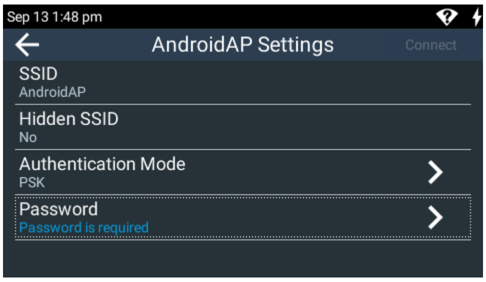


3. Ensure the Wi-Fi **State** is **Enabled** and tap **Rescan Access Points** if the required Wi-Fi network is not populated in the access point list.

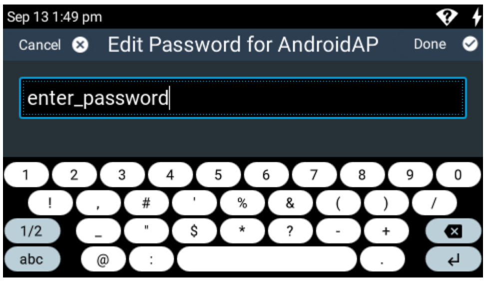
4. Tap to select the Wi-Fi access point to which you are connecting, in this example **AndroidAP**.



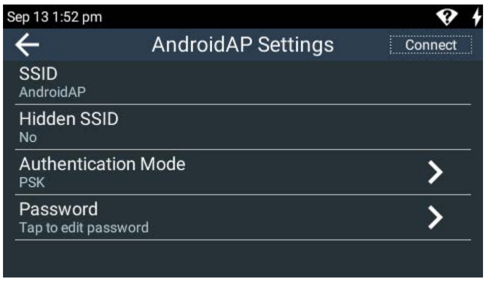
5. Tap to select the preferred **Authentication Mode** and then tap **Password** to enter the network password.



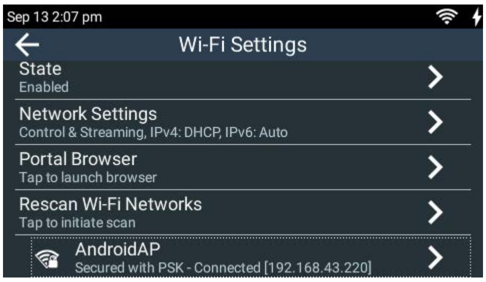
6. Enter the password, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



7. Tap **Connect** in the top right-hand corner of the **TOUCH SCREEN** to connect to the Wi-Fi network.



8. Verify the Wi-Fi  symbol is visible in the **Status Bar** to confirm the codec has connected to the Wi-Fi access point.



Important Note: ViA supports IEEE 802.11 a/b/g/n Wi-Fi with dual band connectivity (2.4 and 5 GHz). For increased security Tieline has implemented the WPA2-PSK authentication protocol because standard WEP encryption is less secure.

Wi-Fi Indications

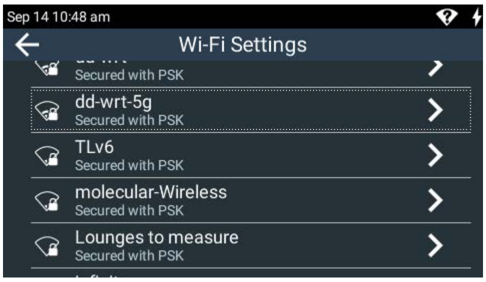
The following Wi-Fi indications are displayed in the **Status Bar** on the codec screen.

	Symbol	Description of Status
1	No symbol displayed	Wi-Fi is disabled in the codec
2		• Wi-Fi is enabled in the codec, but it is out of range of a Wi-Fi network, or • The codec is within range of a Wi-Fi network, but is either not connected or is in the process of connecting
3		The codec is connected to a Wi-Fi network and signal strength is displayed

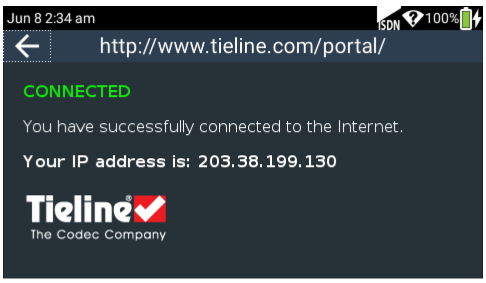
Browser Wi-Fi Login

Some Wi-Fi networks require a browser to log in and connect to the access point, e.g. hotel Wi-Fi networks. To log in to this type of network:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings** > **Transport Interfaces > Wi-Fi**.
2. Tap to select the network to which you are connecting.



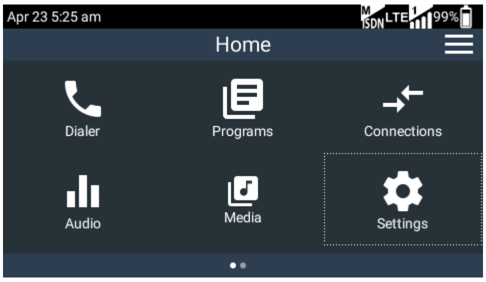
3. Tap to select **Portal Browser** to launch a web-browser on the **TOUCH SCREEN**. Then enter credentials and/or accept terms as required in the web-browser.



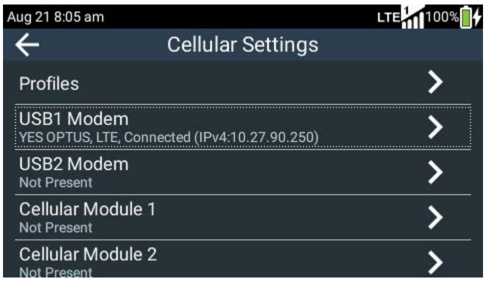
Helpful Hint: Use fingers on the **TOUCH SCREEN** to pinch zoom in to web-browser content, or zoom out.

Connecting over Cellular Wireless via USB Modems

1. Attach a supported USB Modem to either **USB PORT 1** or **USB PORT 2** on the codec. When the modem is detected by the codec, the network symbol and signal strength is displayed in the **Status Bar** on the **TOUCH SCREEN**. Note: It may take up to 90 seconds for the modem to be detected by the codec and connect to the network.

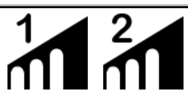


2. From the **Home** screen tap **Settings** > **Transport Interfaces** > **Cellular** > **USB1/2 Modem** to view menus and cellular modem details.



Important Notes:
- Sometimes a cellular network will be detected and a USB modem will connect automatically after it is inserted. Tieline recommends always using the **Overwrite APN** (Access Point Number) menu setting and entering the correct APN for your Telco. This ensures use of the correct data APN. It is usually simple to search the internet for the correct APNs used by each Telco. See Adding Access Points and a SIM PIN for more details.
- It may be necessary to enter a SIM PIN if the codec cannot connect automatically to the network.
- To safely remove a USB modem press the **HOME** button to return to the **Home** screen, then tap **Settings** **> System > Safe Eject**.

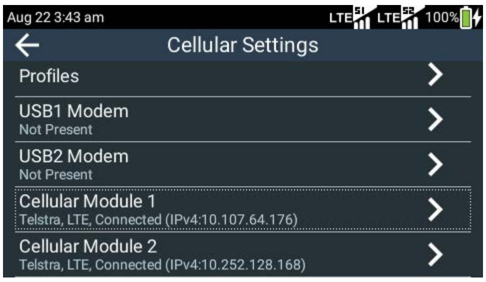
USB Cellular Modem Indications

	Symbol	Description of Status
1	No symbol	• A cellular modem is not attached to the codec. • A modem is attached but not ready yet; it can take up to a minute for the modem to be detected. • The modem has a SIM card issue (locked or not present). • The modem is not supported by the codec.
2		Each USB port is identified by the number 1 or 2 in the top-left corner of the symbol.

3		A cellular modem is attached to the codec but there is an error. Check that: • The SIM card does not have a PIN code enabled. • The modem has data enabled. • The modem is compatible with the network to which you are attempting to connect.
4		The cellular modem is connected to the network and signal strength is displayed. Note: when first attached the symbol is greyed out while the modem is connecting to the cellular network. It will remain greyed out if: • Data is not enabled. • No APN is selected and the modem doesn't contain a correct APN setting from a factory default or previous connection. • An incorrect APN is selected. When using different SIMs they may require different APNs, even if they are from the same Telco. • The correct APN is selected but data has run out. • A custom APN has been added but either the APN info, or the Authentication Type, is incorrect.

Connecting using a Cellular Module

1. Turn the codec off and insert a cellular module into the module slot.
2. Power up the codec and press the **HOME** button to return to the **Home** screen, then tap **Settings** > **Transport Interfaces** > **Cellular** > **Cellular Module 1 / 2** to view cellular module menus and details.

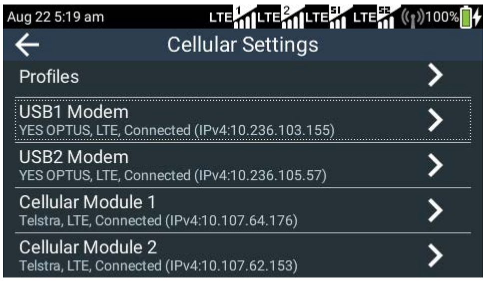


Important Notes:

- Sometimes a cellular network will be detected and a module will connect automatically after it is inserted. Tieline recommends always using the **Overwrite APN** (Access Point Number) menu setting and entering the correct APN for your Telco. This ensures use of the correct data APN. It is usually simple to search the internet for the correct APNs used by each Telco. See Adding Access Points and a SIM PIN for more details.
- It may be necessary to enter a SIM PIN if the codec cannot connect automatically to the network.
- Cellular radio transmissions can be turned off without removing the module. Select **Settings** > **Transport Interfaces** > **Cellular** > **Cellular Module 1 / 2** > **Enable Radio [OFF]**

SIM LTE Indications

Indications displayed in the **Status Bar** on the **TOUCH SCREEN** for an internal cellular module are similar to USB modem cellular network indications. The only difference is that **S1** and **S2** is displayed to identify SIM card 1 and 2. In the following image there are two USB modems and two SIM cards displayed in the **Status Bar**.

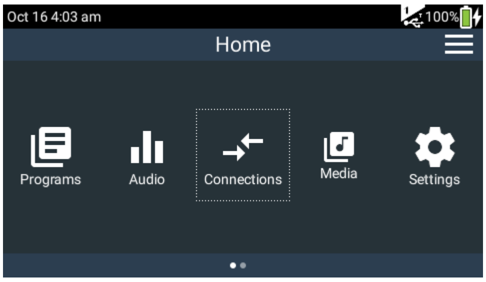


Tethering a Smartphone

Important Note: Only tethering to an Android phone is supported.

It is possible to tether a phone to the codec with a USB cable and use cellular or Wi-Fi data from the connected device.

1. Enable USB tethering in the smartphone to allow sharing of the phone's internet connection. Note: This will probably need to be enabled each time the phone is connected to the codec.
2. Attach a USB cable to the phone and then attach the USB cable to one of the two host **USB PORTs** on the codec.
3. The codec should detect the smartphone connection and acquire an IP address from the phone. When it connects successfully the USB tethering symbol changes from greyed out to illuminated in the **Status Bar** at the top of the **TOUCH SCREEN**.



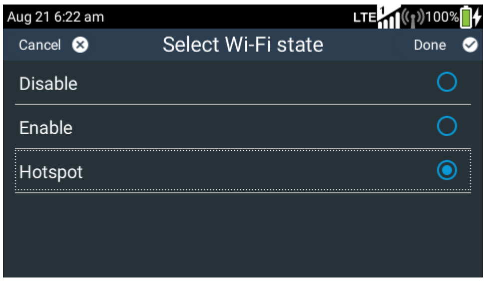
4. Select **Any**, or choose **USB1** or **USB2** when specifying a **Via** interface to use when connecting.

Wi-Fi Hotspot Configuration

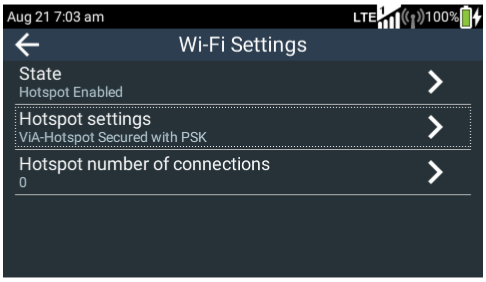
The codec supports Wi-Fi hotspot data sharing, which allows external devices to share data.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings** **> Transport Interfaces > Wi-Fi** **> State**.

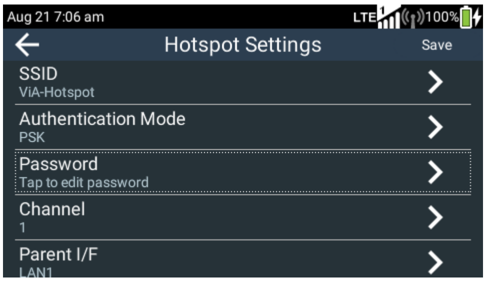
2. Tap **Hotspot** to enable the Wi-Fi hotspot.

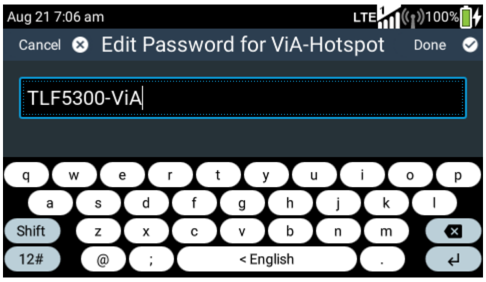


3. Tap **Hotspot Settings** to adjust Wi-Fi hotspot settings.

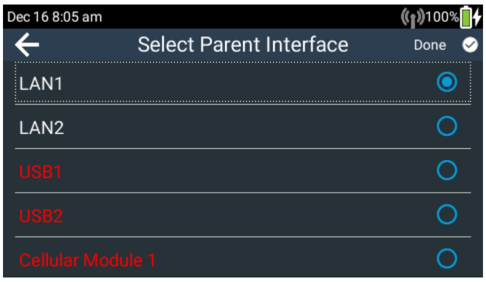


4. Tap **Password** to edit the default Wi-Fi hotspot password.





5. Tap **Parent Interface** to choose the interface from which data will be shared using the Wi-Fi hotspot.



6. When a device connects successfully the Hotspot symbol changes from greyed out to illuminated in the Status Bar in the top right corner of the **TOUCH SCREEN**.

Helpful Hint: Note the Wi-Fi IP address on the screen when you set up a hotspot and then enter this IP address in a browser on a device connected to the hotspot, e.g. a tablet, to view the Toolbox HTML web-GUI for the codec and control the codec.

Important Note: Exercise extreme caution when using the Wi-Fi hotspot feature if streaming live IP audio simultaneously while this mode is enabled. Sharing data may lead to less reliable IP connections due to reduced bandwidth being available.

12.1 Cellular Activation in the USA

LTE Activation Notice

Before using a Tieline ViA LTE module in the USA it is necessary to set up a wireless account with your Telco. Follow the steps in the tech note at the following link to set up a wireless account: https://www.multitech.com/documents/publications/activation-guides/lte-device-activation-steps-s000620.pdf

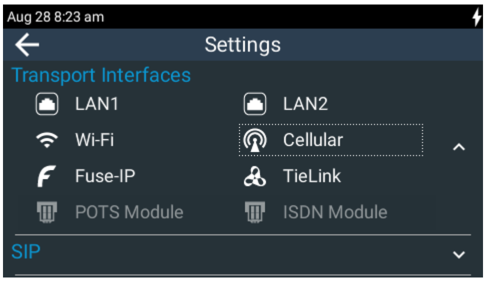
13 Adding Access Points and a SIM PIN

It may be necessary to add a custom access point number (APN) profile if a USB modem or cellular module does not connect to a cellular network. Often this is necessary when you are using different SIM cards in an unlocked USB modem. E.g. When using SIM cards from various carriers in different countries. As a rule of thumb, Tieline recommends always using the **Overwrite APN** menu setting and entering the correct APN for your Telco. This ensures use of the correct data APN. It may also be necessary to enter a SIM PIN unlock code.

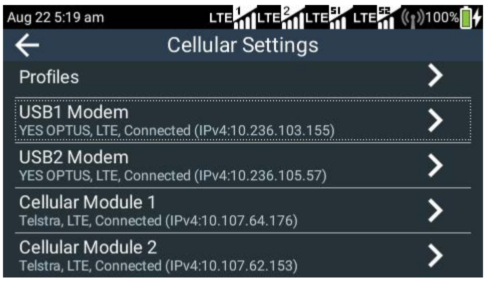
Up to 10 custom access point profiles can be added to the codec. Before configuring the custom access point you need to obtain the access point details from the cellular network provider; this is normally found on a Telco's website. Usually a Telco will list internet and MMS APN information and you need to enter the internet APN details in the codec, as well as the correct authentication type.

Entering a SIM PIN Unlock Code

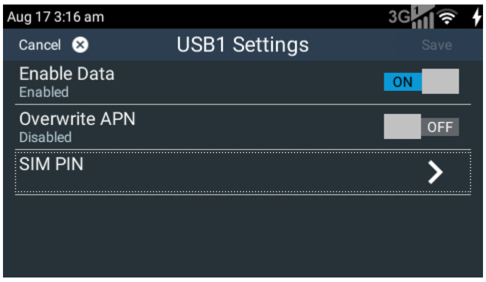
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **Cellular**.



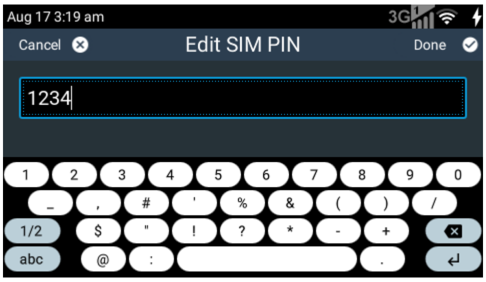
3. Tap to select the device you are configuring.



4. Tap to select **SIM PIN**.



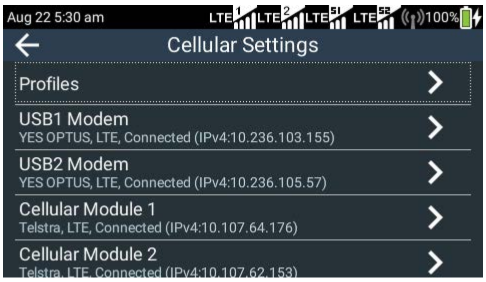
5. Enter the **SIM PIN** and tap **Done**.



6. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to apply new settings.

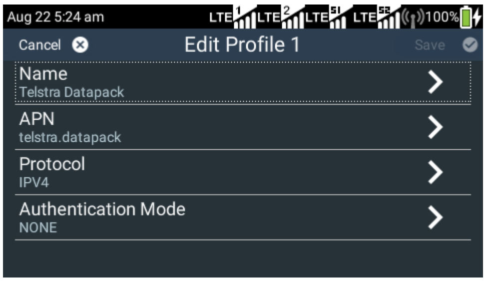
Add a Custom APN Profile

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **Cellular**.
3. Tap **Profiles** to add a new APN profile.



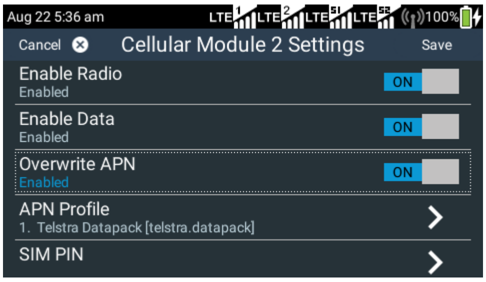
4. Tap to configure a profile with the custom APN settings.

5. Tap **Name** to name the profile and then tap to enter each access point setting. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** when all settings are configured.

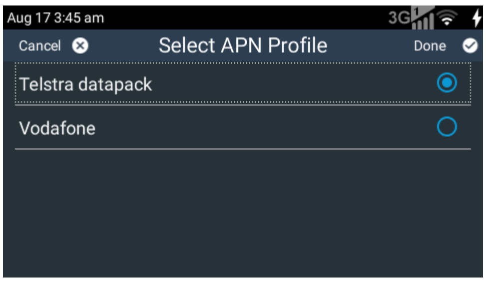


Select a Custom APN Profile

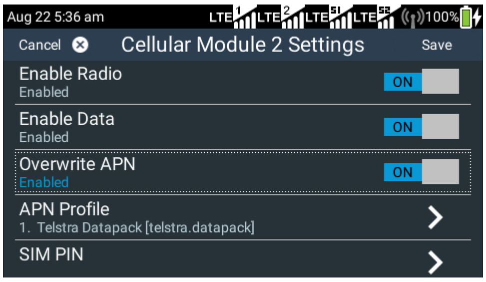
1. After adding the custom profile and APN, tap to select the interface you want to configure. (Navigate to **Home** screen **> Settings** **> Cellular** **> Select a modem or module**).
2. Tap to toggle the **On/Off** button for **Enable Data** and **Overwrite APN** to enable these settings.



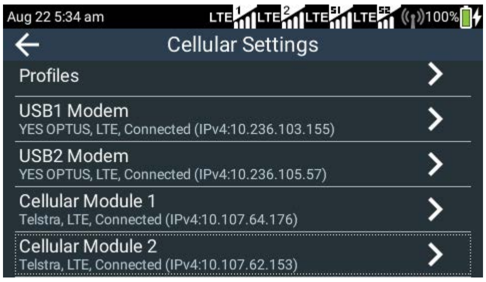
3. Tap to select the APN profile you want to apply to the device, or tap **Done** to keep the existing setting.



4. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to apply all settings.

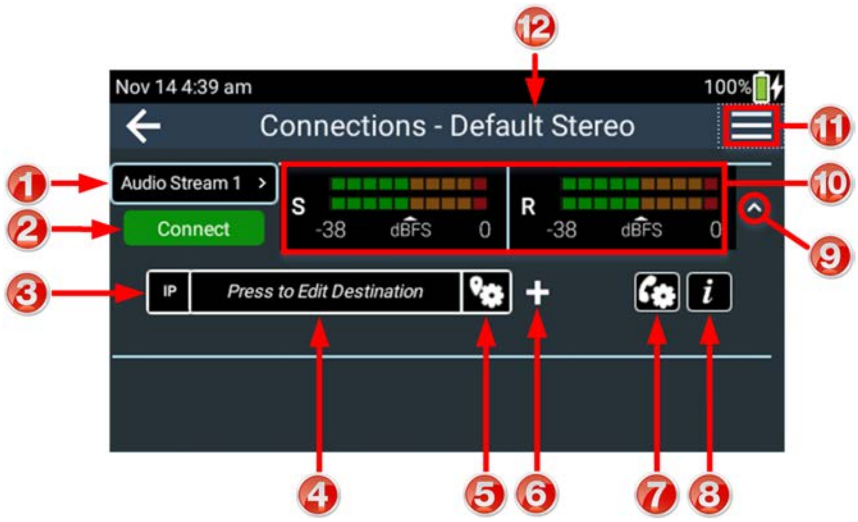


5. The device should connect to the network as displayed in the following image.



14 Connection Guide

This quick start guide will get you connected with ViA in mono or stereo in just a few minutes. Configuration features on the **Connections screen** are as follows:



	Feature	Description
1	Audio Stream name / button	Displays the audio stream name; tap to edit audio stream settings, including Stream Name, Caller ID, Routing Type, G3 Profile Type, G3 Channel.
2	Connect button	Tap the **Connect button** to dial a connection.
3	Connection type / TieLink selector	Tap to toggle between selecting whether to dial a configured IP/SIP, ISDN or POTS connection, or a TieLink Address Book contact.
4	Dialing destination	Tap to enter an IP/SIP address or ISDN/POTS number, or select a TieLink Contact list contact if TieLink is configured.
5	Destination settings	Tap to configure destination settings, e.g. dialing interface and IP ports.
6	Add SmartStream PLUS	Tap the **Plus symbol** to add a SmartStream PLUS dialing connection. By default, the last configured IP address is inserted in a new SmartStream PLUS connection.
7	Connection settings	Tap to adjust connection-related settings like Transport, Algorithm, Auto Reconnect, Jitter Buffer, FEC, Direction, Auxiliary Data, Connection Name.
8	Information (statistics)	Tap to view audio stream connection statistics, e.g. Link quality, packet statistics. Swipe in this screen to view other active audio stream statistics.
9	Expand / Collapse	Tap to expand or collapse the audio stream information displayed.
10	PPMs	Audio stream PPMs for send and return audio in dBFS by default.
11	'Hamburger Menu'	Tap to load a default or custom program, or create a new program and save it as a custom program. Switch between Basic and Admin modes of operation.
12	Program name	The name of the loaded program.

The codec is configured to connect in mono over IP by default when shipped, or when factory default settings are restored.

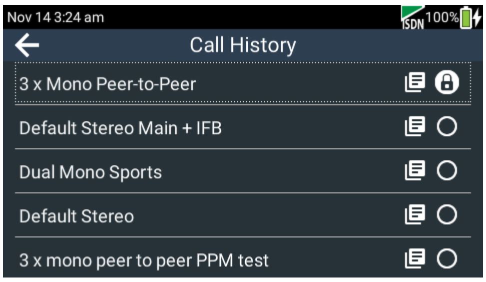
1. Ensure the codec is not powered up when inserting or removing modules.

2. Attach the 12VDC power supply to the codec or insert the charged **BATTERY**. Note: the **BATTERY** is automatically charged inside the codec when the power supply is connected.
3. Attach headphones and microphones to the codec.
4. Press the **POWER** button to power up the codec.
5. Press the **INPUT ON/OFF** button to turn each input on and adjust the **INPUT GAIN** rotary encoder for each microphone. Note: Navigate to **Home screen > Audio > Inputs > Input Type** > **Gain Type** to adjust coarse gain levels for each input source.
6. Follow the procedure for connecting over IP, SIP, ISDN or POTS.

CAUTION: DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** mode is selected, or equipment may be damaged by high voltages.

Important Note:
- Unused inputs should be turned **[OFF]** to avoid introducing noise and improve audio quality.
- To load a saved program and dial, press the **HOME** button to return to the **Home screen** and then tap **Programs**. Tap to select the program you want to use and then tap **Load Program** in the dialog. Tap **Connect** or press the **CONNECT** button to dial using the newly loaded program.
- It is also possible to load a program from the **Connections screen**. From the **Home screen** tap **Connections** > **Menu** > **Load Program [tap to select a custom program]**, or tap **Default Program** and select a program from the default configurations listed.

Helpful Hints:
- Swipe left once from the home screen to view programs selected as **Favorites**. To learn more about configuring programs as **Favorites** see Load, Connect and Manage Programs.
- The codec remembers recent calls just like a cell-phone. To view **Call History** press the **CONNECT** button in any screen, except when you have selected **Programs** from the **Home** screen.



- If you don't rename and save a program during configuration, the default program is displayed in the **Call History** list if you want to redial the program.

14.1 Connecting over IP

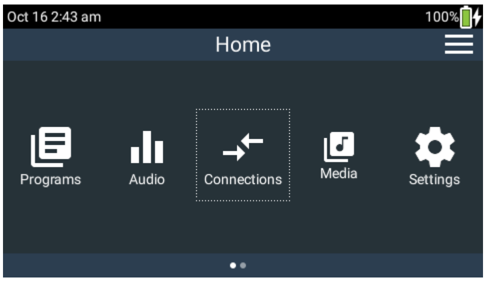
The following procedure will use a default 'program' to connect using the codec's **TOUCH SCREEN**. For IP connections, attach an RJ45 Ethernet cable to one of the **LAN** ports on the codec's left side panel, or use one of the codec's wireless IP connection options.

Important Notes:
- See Using the HTML5 Toolbox Web-GUI for details on configuring connections remotely via a computer.
- See Installing the Codec at the Studio for valuable information about installing your codec, negotiating firewalls and port forwarding.

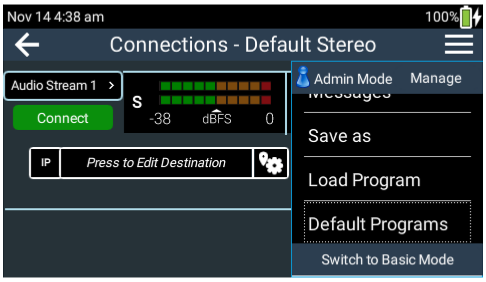
- See Tips for Creating Reliable IP Connections for a range of IP information to assist with setting up IP services for your codecs.
- See Testing IP Network Connections to learn how you can test and verify the reliability of your IP connection.

The following example uses Tieline default settings to create a stereo IP connection:

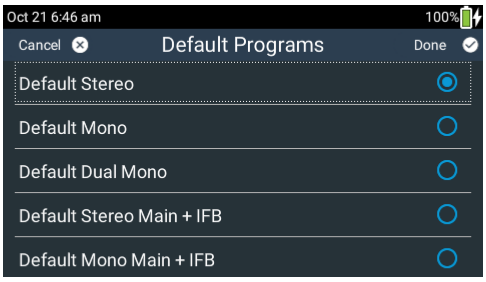
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.



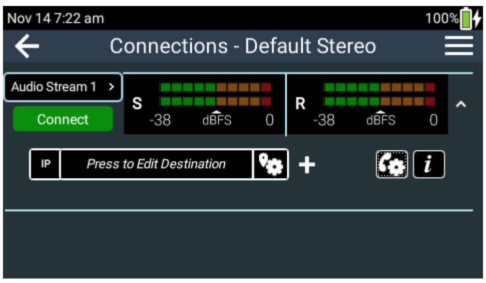
2. Tap **Menu** and then tap **Default Programs** to view default program options. Note: Tap **Load Program** to load preconfigured custom programs. Default programs cannot be locked in the codec. Save a program as a custom program if a program needs to be locked.



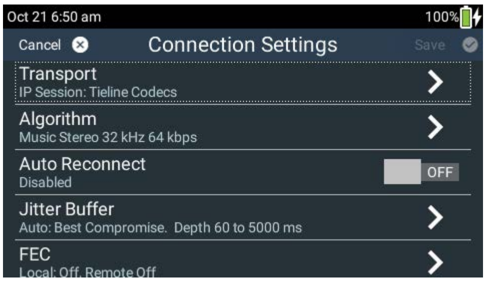
3. Tap to select the preferred program connection option, e.g. **Default Mono** or **Default Stereo**, and then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

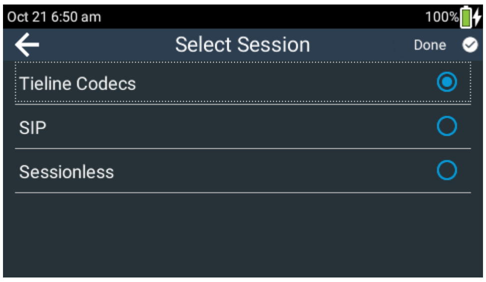


4. With factory default settings the codec is configured to connect over IP. To adjust connection settings tap the **Connection Settings** symbol.



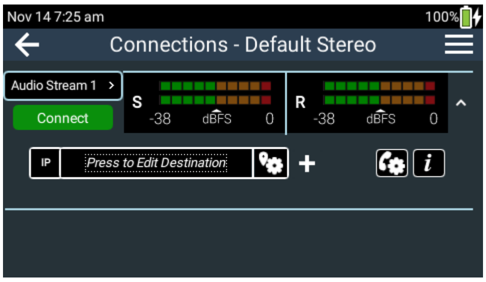
5. If required, tap **Transport** and select **IP > Tieline Codecs** if ISDN or POTS is the currently configured transport.





Note: Select **SIP** instead of **Tieline Codecs** when dialing non-Tieline codecs and see Dialing SIP Peer-to-Peer to configure SIP connections. Select **Sessionless** to dial a sessionless IP connection over networks supporting this connection method.

6. To connect very simply using default settings navigate to the **Connections screen**.

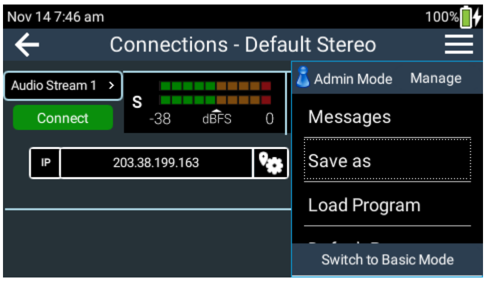


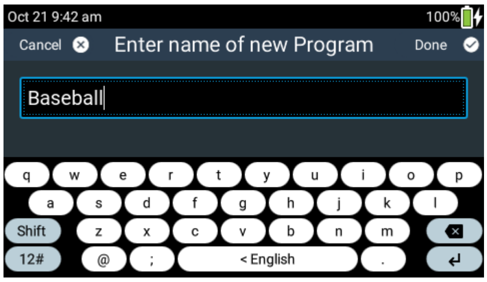
7. Tap **Press to Edit Destination** and then enter the IP address of the destination codec being dialed, or tap the **Dial History** symbol to select a previously dialed number. Then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

Note: Tap **IP** to toggle and display the **Address Book** symbol, then tap to select a TieLink Traversal Server contact list destination if this is available. For more details on TieLink see TieLink Configuration.

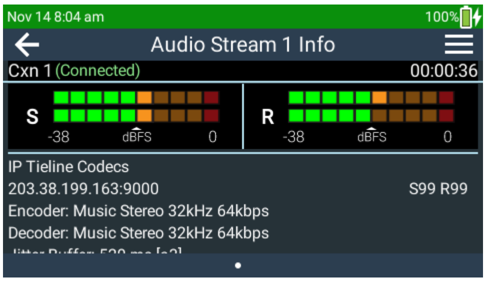
8. Tap **Connect** Connect or press the **CONNECT** button to dial the destination codec. The **Status Bar** turns green when the codec is connected.

Helpful Hint: To save the program as a new custom program for later use tap **Menu** and then tap **Save as**, then name the program before connecting.

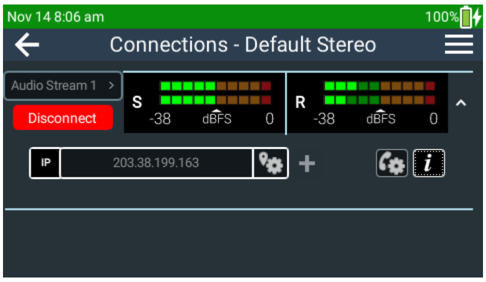


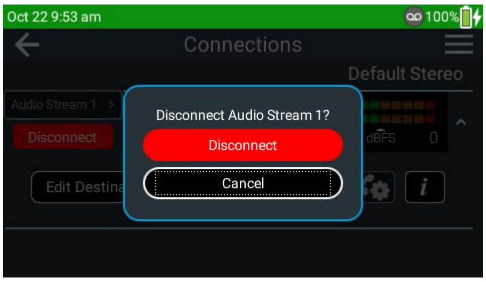


9. Tap the **Info** button to view audio stream connection details and packet statistics while connected. Note: Incrementally renegotiate higher connection bit rates by pressing the **F2** button and then the **NAVIGATE UP** button while viewing the **Audio Stream Info** screen; for lower bit rates press the **F2** button and then the **NAVIGATE DOWN** button.



10. Tap **Disconnect** Disconnect or press the **DISCONNECT** button to hang up the connection.

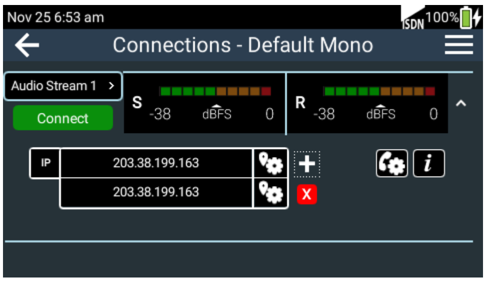




Adding/Removing a SmartStream PLUS Connection

To add a SmartStream PLUS IP connection for redundancy:

1. Tap the **Plus symbol** to add a SmartStream PLUS connection.
2. A new connection will appear below the primary connection. Note: By default, each new SmartStream PLUS connection uses the same IP address as the last configured connection.



3. Tap the IP address field to edit the IP address of the destination codec being dialed, then tap **Done**. Note: Tap **IP** to display the **Address Book** symbol and then tap to select a TieLink Traversal Server contact list destination if this is available.
4. Ports are allocated automatically and tap the **Destination Settings** symbol to select the interface over which to stream the connection, e.g. LAN, Wi-Fi or Cellular. See Other IP Connection Settings for more details.
5. The SmartStream PLUS connection is now configured.
6. Tap the **Red Cross symbol** to delete a SmartStream PLUS connection.

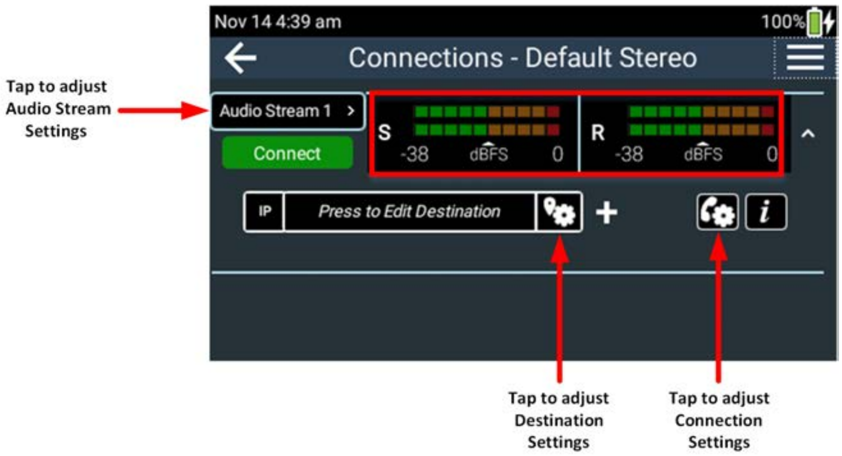
Important Notes:

- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.

14.2 Advanced IP Connection Settings

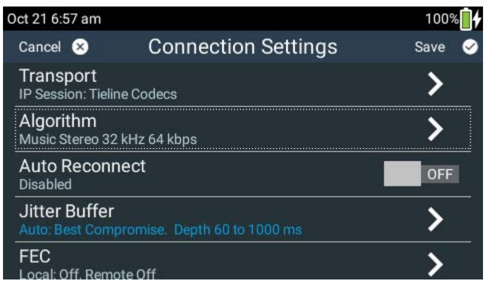
To adjust other IP connection settings there are three options:

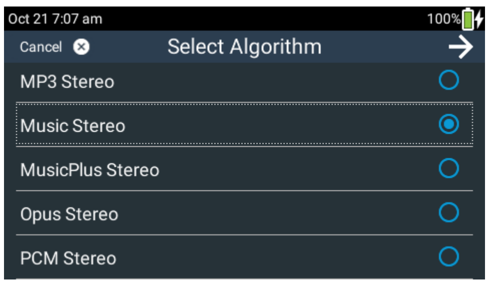
- Connection settings: Algorithm, jitter, FEC, auto reconnect, auxiliary data, encoding/decoding settings and connection name.
- Destination settings: Stream level settings like selection of IP interface (Via) and local/remote audio and session ports.
- Audio stream settings: High level settings for Caller ID, call routing, G3 Profile and Channel settings, and adjust the stream name.



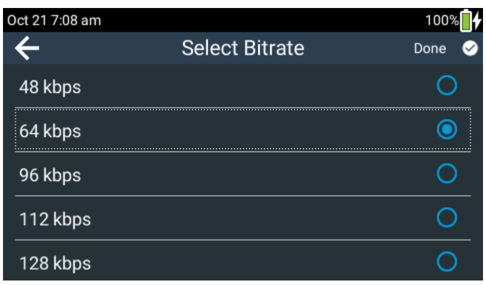
Adjusting IP Connection Settings

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap **Algorithm**, then tap to select an algorithm from those listed. Note: In most situations Tieline Music is the best algorithm to use and this is configured by default. See Selecting an Algorithm for more details on algorithm options.

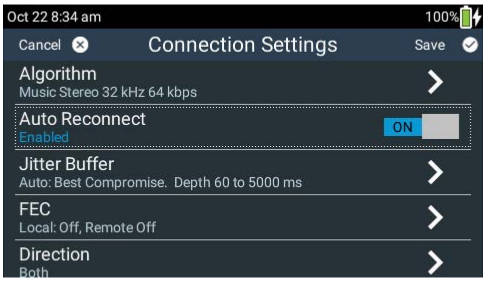




4. Tap to select a sample rate. Note: multiple sample rate options are available for algorithms like MP2, MP3, AAC and aptX Enhanced.
5. Tap to select the initial connection bit rate.

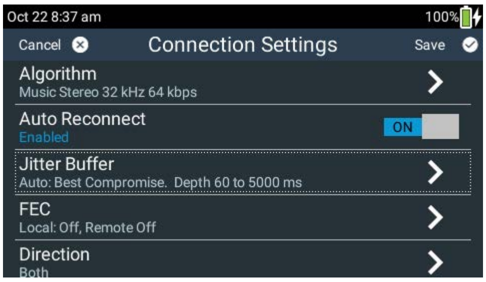


6. Tap to toggle the **On/Off** button for **Auto reconnect** if you want the codec to automatically reconnect when the connection is temporarily lost.

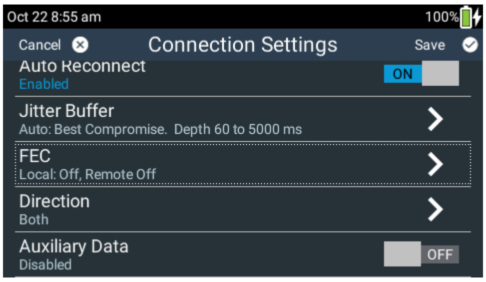


7. Tap **Jitter Buffer** to select a different automatic jitter buffer setting for your connection, or to enter a fixed buffer setting in milliseconds (maximum 5000 ms). The default **Auto, Best Compromise** setting is a good starting point for most internet connections. Note: The **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000ms** is a good starting point for most networks. Recommended maximum fixed jitter limits are as follows:

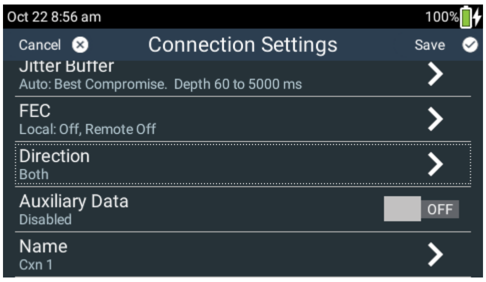
- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.
- 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS. See Configuring the Jitter Buffer for more details.



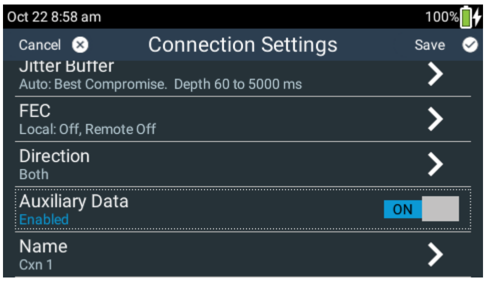
8. Tap **FEC** to configure local and remote forward error correction settings. See Configuring Forward Error Correction for more details.



9. Tap **Direction** if you want to save data and configure the codec to either **Encode Only** or **Decode Only**. Note: this can be helpful if connection bandwidth is limited.



10. Tap to toggle the **On/Off** button for **Auxiliary Data** to activate **CONTROL PORT I/O** operation and RS232 data in the codec.

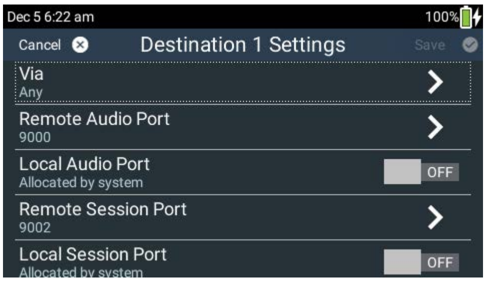


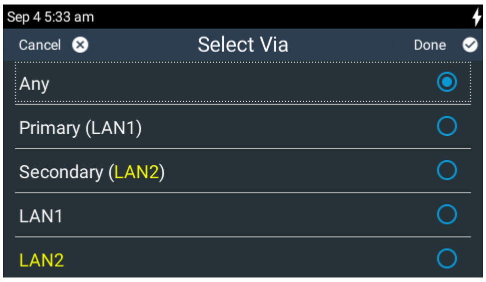
11. Tap **Name** to rename the connection. When all settings have been configured tap **Save** in the top right-hand corner of the **TOUCH SCREEN**.

Destination Settings

1. Press the **HOME**  button to return to the **Home** screen, then tap **Connections** .
2. Tap the **Destination Settings**  symbol.
3. Tap **Via** to select a specific dialing interface in the **Select Via** screen, or use the default **Any** setting. Tap **Save** in the top right-hand corner of the screen to dial using the default **Any** setting. This setting uses the first available interface based on the following default priorities.

- Primary = LAN1.
- Secondary = LAN2.
- Tertiary = Wi-Fi.





You can also select a specific interface, e.g. **Primary**, **Secondary**, **LAN1** or **LAN2**, as displayed in the preceding image. A LAN interface displayed in yellow text is enabled, but a network cable is not attached. Interfaces are not displayed if they are unavailable, e.g. if Wi-Fi is turned off or there is no cellular modem attached to **USB PORT 1** or **USB PORT 2**. To reconfigure the default **Primary**, **Secondary** and **Tertiary** settings see Configuring Via Interfaces.

4. If required:

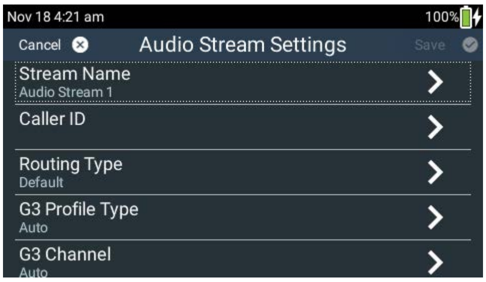
- Tap **Remote Session Port** to enter the session port number of the remote codec (to which you are dialing).
- Tap **Remote Audio Port** to enter the audio port number of the remote codec (to which you are dialing).

Important Note: The **Local Audio Port** and **Local Session Port** can also be configured. In most situations this is configured by the codec arbitrarily.

5. Tap Save in the top right-hand corner of the **TOUCH SCREEN** when these settings are configured .

Audio Stream Settings

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Audio Stream name** to display stream settings.



3. Tap **Stream Name** to edit the default name.
4. Tap **Caller ID** to enter an identifier for inbound calls on Tieline codecs.
5. Tap **Routing Type** to configure routing options as per the table below:

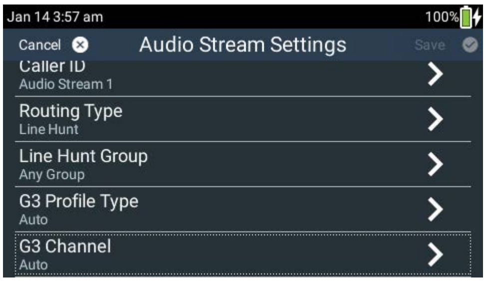
Routing Type Options:	
Default ISDN POTS	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

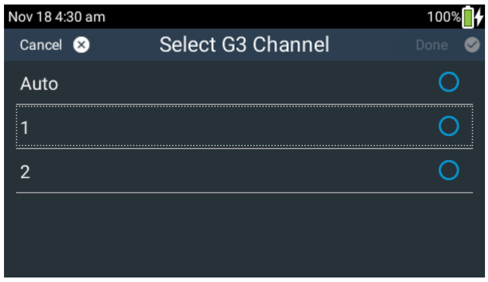
6. Tap **G3 Profile** to configure profile settings when dialing a Tieline G3 Codec.

Important Note: The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. Note: This is overridden in a ViA codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

7. Tap **G3 Channel** when connecting to a G3 codec in dual mono mode. This setting lets you configure which G3 channel (encoder) is used when the G3 codec receives a call from this codec. E.g. **Channel 1** will route the incoming stream to Encoder 1 on the G3 codec and **Channel 2** will route the incoming stream to Encoder 2 on the G3 codec.



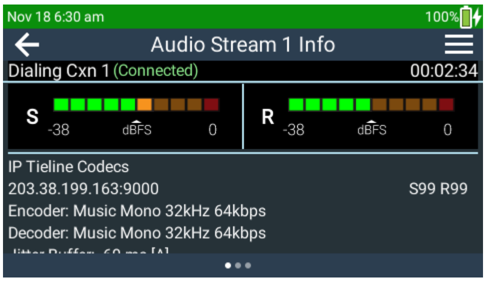


8. When configuration is complete, tap **Save** in the top right-hand corner of the **TOUCH SCREEN**.

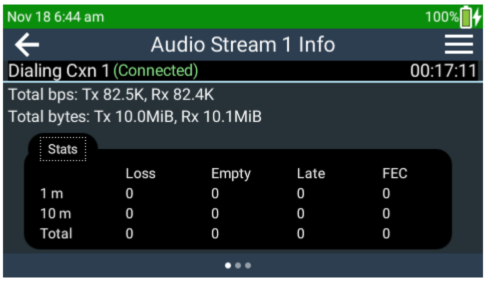
14.3 Monitoring IP Connections

To monitor active connections:

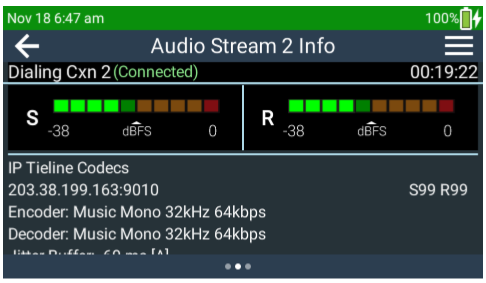
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap **Info** to view audio stream connection details and statistics.



Scroll down to view jitter and IP packet arrival statistics. Analysis is historic and assessed over 1 minute and 10 minutes of connection time.

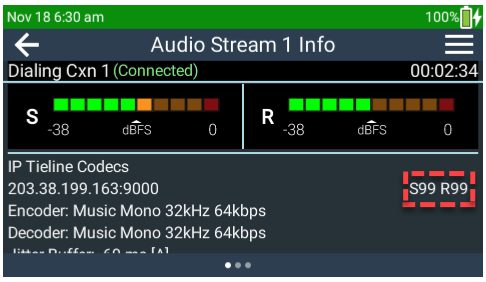


With multiple stream programs swipe left to view the connection statistics for other audio streams.



Send/Return Link Quality

Send and **Return** link quality numbers in the **Audio Screen Info** screen assist in determining if there is a problem at either end of a connection.



For example, on an IP connection the **Return** indication represents the audio being received from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the **Send** indication represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** reading.

Important Note:
- The **Return** link quality reading is the same as the Local (**L**) setting displayed on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed on a G3 codec.

	Feature	Description
1	Lost Packets	Packets sent that failed to arrive
2	Empty (Jitter Buffer)	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio
3	Late Packets	The number of packets that arrive late, i.e. after audio play out
4	FEC Packets	Indicates the number of forward error correction (FEC) packets that have been replaced if it is enabled in the codec
5	1 minute	Statistics listed for the last minute of network activity
6	10 minutes	Statistics for the last 10 minutes of network activity

Important Note:
- Increasing the minimum jitter buffer depth may address the issue of packets arriving late.
- If the jitter buffer, FEC or the connection bit rate is changed, we recommend assessing a minute of recent connection performance in preference to 10 minutes of historical connection performance. 10 minutes of data will include connection settings which may no longer be relevant. 'Packet arrival history' is cleared when you hang up a connection.

Following is a packet arrival analysis table with solutions for any noticeable packet loss statistics displayed on the screen.

Packet Analysis	Displays	Possible Causes	Possible Solutions
Loss	Packets failed to arrive.	• LAN/WAN congestion • Unreliable ISPs • Unreliable networks • Unreliable IP hardware	• Renegotiate connection bit rate downwards • If link quality good, add or increase FEC as required • Assess ISPs QoS if very bad performance
Empty	Indicates how often the jitter buffer 'reservoir' empties causing loss of audio.	• High number of packets being lost or arriving late • Signal dropouts using cell-phone networks • Renegotiation causes the jitter buffer reservoir to empty	• Once could be an anomaly – assess lost & late packets • If many lost packets and network is unreliable – renegotiate bit rate and /or FEC down • If many late packets, increase jitter buffer maximum and minimum depth
Late	The number of packets that arrive late and after audio play out.	• Network congestion • Jitter Buffer depth is too low	• Auto-jitter buffer will adjust automatically • For manual jitter buffer settings, increase jitter buffer depth 50-100 ms & reassess (if only a few packets arrive late over time; audio repairs will be automatic and may not require buffer changes).
FECd	Indicates the number of FEC repaired packets if FEC active.	• Packets have been lost or corrupted over the network	• Assess audio quality & the number of FEC repairs – if many packets are being 'lost', perhaps reduce FEC &/or renegotiate bit rate down.

14.4 Connecting over SIP

Important Notes:

1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **Wi-Fi** by default, which is mapped to the **Tertiary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN1** or **LAN2**.
5. **SIP1** and **SIP2** can each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface has been enabled.
7. Enter a public IP address in the **Public IP** menu if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.
8. It is not possible to renegotiate the connection bit rate over a SIP connection.
9. When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. From the **Home** screen navigate to **Audio** > **General** > **Reference Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
   - Select either a mono or stereo profile
   - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
   - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

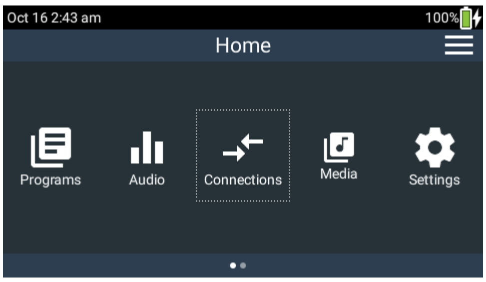
Dialing SIP Connections

SIP can be used to make direct peer-to-peer or SIP account calls to IP codecs configured with public IP addresses, or between two codecs over a LAN. Peer-to-peer SIP calls are often used to connect to other brands of codecs and perform call and session management tasks.

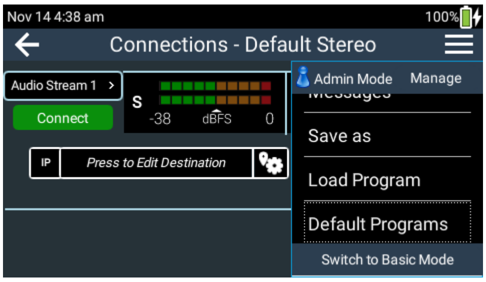
To make a peer-to-peer SIP call between codecs we recommend both codecs use public IP addresses. Find out the IP address of the codec being dialed and configure each codec with a compatible algorithm and sample rate etc. If the remote codec has a private IP address then it should be configured for port forwarding and should dial the public IP address at the studio.

To dial using a SIP account it is necessary to add a SIP account and register it to the codec. See Configuring SIP Accounts for more info.

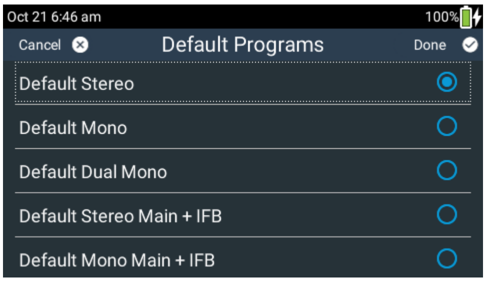
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.



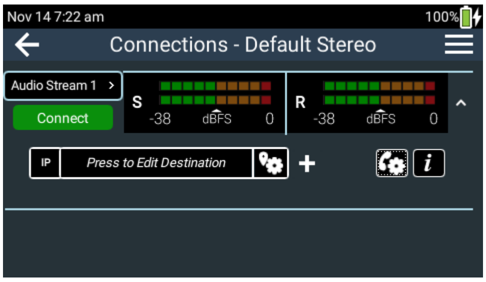
2. Tap **Menu** and then tap **Default Programs** to view default program connection options. Note: Tap **Load Program** to load preconfigured custom programs. Default programs cannot be locked in the codec. Save a program as a custom program if a program needs to be locked.



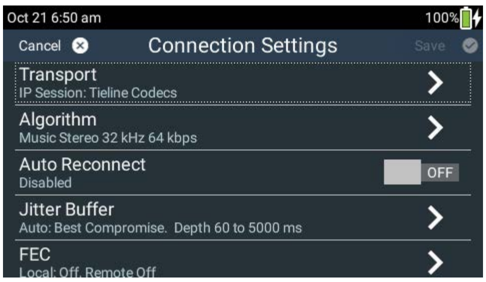
3. Tap to select the preferred program connection option, e.g. **Default Mono** or **Default Stereo**, and then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

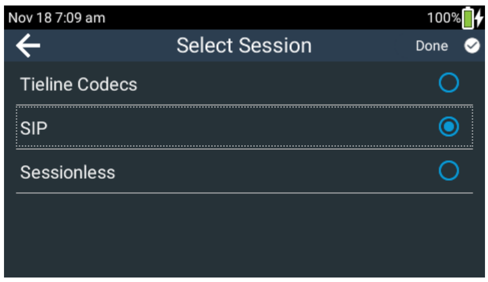


4. To adjust connection settings tap the **Connection Settings**  symbol.

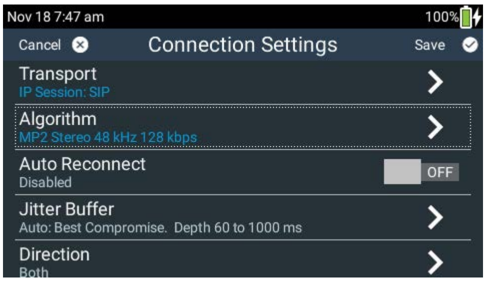


5. Tap **Transport** and select **IP > Tieline Codecs** if ISDN or POTS is the currently configured transport, then tap **SIP**.

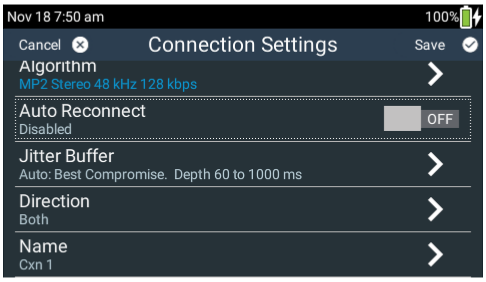




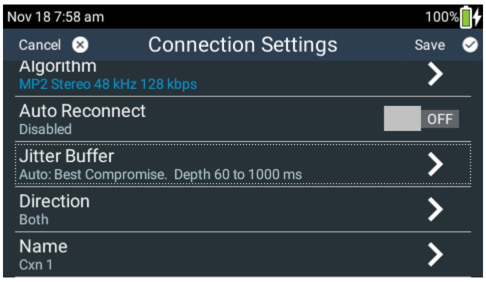
6. Tap **Algorithm** to select the preferred algorithm, sample rate and bit rate. G.711 is configured by default in mono.



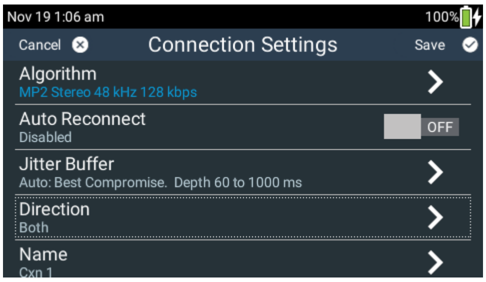
7. Tap to toggle the **On/Off** button for **Auto reconnect** if the codec should automatically reconnect if the connection is temporarily lost.



8. Tap **Jitter Buffer** to select a different automatic jitter buffer setting for your connection, or to enter a fixed buffer setting in milliseconds (maximum 5000 ms). The default **Auto, Best Compromise** setting is a good starting point for most internet connections. Note: The **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000ms** is a good starting point for most networks. Recommended maximum fixed jitter limits are as follows:
   - 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
   - 2,500ms for AAC ELD, AAC LD.
   - 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS. See Configuring the Jitter Buffer for more details.

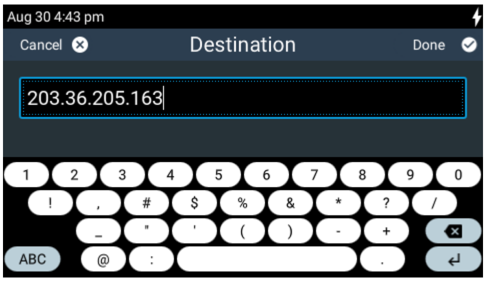


9. Tap **Direction** if you want to save data and configure the codec to either **Encode Only** or **Decode Only**. Note: this can be helpful if connection bandwidth is limited.

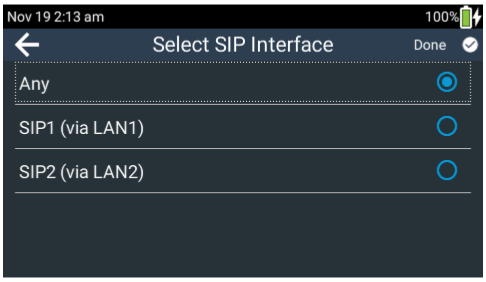


10. Tap **Name** to rename the connection. When all settings have been configured tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

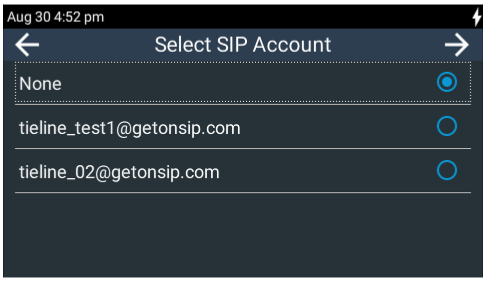
11. Tap **Press to Edit Destination** and then enter the IP address, or alphanumeric characters in the SIP URI for the codec you want to dial, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN** to confirm all settings.



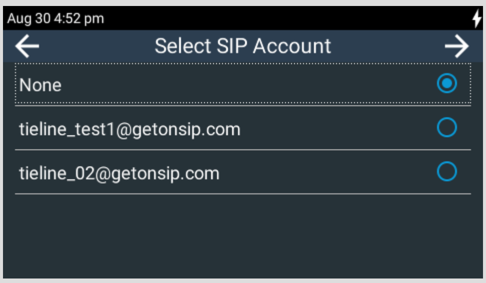
12. Tap the **Destination Settings** symbol and then tap Via to select a specific dialing interface in the **Select SIP Interface** screen. For a peer-to-peer SIP connection select **Any** as the interface with which to dial if you have an Ethernet cable connected to **LAN1** or **LAN2**.



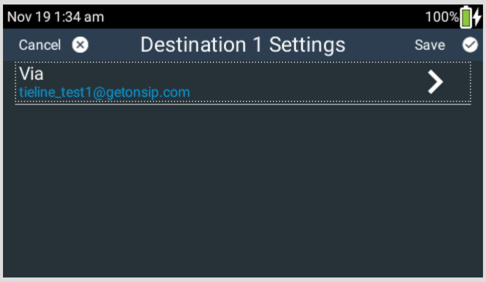
13. If a SIP Account is registered to the codec, tap to select **None** as the SIP account when dialing peer-to-peer, or select a registered SIP account if you are using a SIP server to establish a connection.



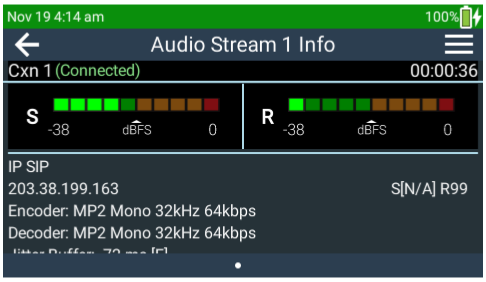
Important Note: Any valid SIP server accounts registered successfully to the codec will be displayed in the **Select SIP Account** screen. Tap to select an account if you want to use a SIP server to establish the connection.



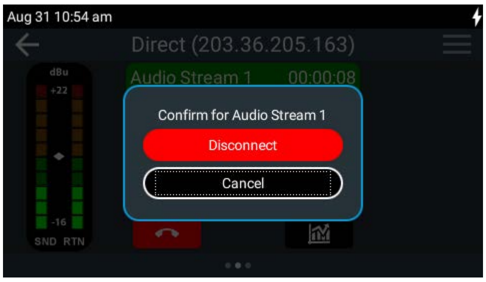
When you select a SIP account this will complete configuration. Simply tap **Save** in the top right-hand corner of the **Destination Settings** screen to return to the **Connections screen**.



14. When configuration is complete, tap **Connect** Connect or press the **CONNECT** button to dial the destination codec. The **Status Bar** turns green when the codec is connected.
15. Tap **Info** to view audio stream connection details and statistics.



16. Tap **Disconnect** Disconnect or press the **DISCONNECT** button to hang up the connection.



Important Notes:
- See Configuring SIP Accounts for instructions on entering SIP account details into the codec. If your codec is registered with same SIP registrar as the destination codec then you only need to enter the SIP user name to dial successfully.
- It is also possible to configure SIP programs using the Toolbox web-GUI. See the section titled Configure SIP Peer-to-Peer Programs for more information.

14.5 Connecting with Fuse-IP

Fuse-IP is a proprietary Tieline IP bonding technology allowing a user to aggregate data by bonding multiple IP interfaces (peers) and establishing a “tunnel” between two Tieline codecs. A streaming connection can be established after the tunnel is created. Fuse-IP automatically distributes data over any two bonded interfaces, which may include using any of 7 possible IP interfaces, including:

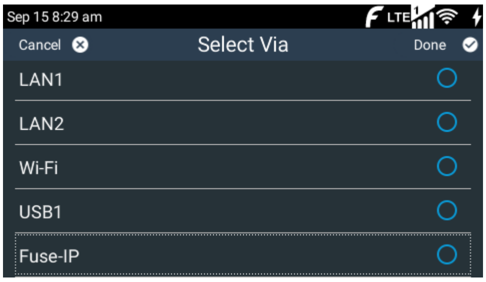
- 2 USB modems.
- Cellular connections via Dual Active SIM LTE module.
- Wi-Fi.
- Dual Ethernet LAN ports.

There are several benefits in using Fuse-IP to aggregate data from multiple IP interfaces, including:

- The ability to create more stable connections with higher overall data bandwidth.
- Greater choice of encoding algorithms because of higher available bandwidth.
- Redundancy in case one IP connection is lost.

How does Fuse-IP work?

Fuse-IP is another selectable **Via** interface over which to dial, similar to selecting a LAN port or Wi-Fi.



Fuse-IP requires one codec to be a server and the other codec a client. Normally the remote codec is configured as the client and the studio codec is the server, because it's easier to dial static IP addresses configured at the studio than cellular or Wi-Fi interfaces at the remote site. Like with SmartStream PLUS redundant streaming, configure two IP interfaces at the studio for additional redundancy.

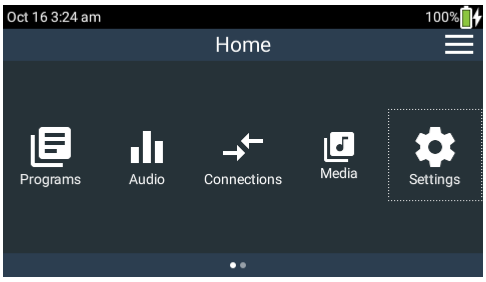
Prerequisites

ViA codecs using Fuse-IP can connect to the Tieline Genie, Merlin and Bridge-IT codec families. Version 2.16.xx firmware is required to use Fuse-IP in these codecs. Before configuring Fuse-IP you need:

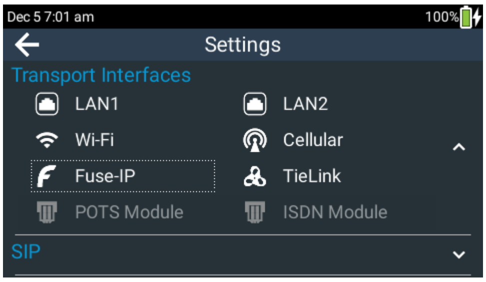
- The IP address (or addresses) for the codec acting as the server at the studio.
- The serial number of the server codec to which you are connecting using Fuse-IP.

Configuring Fuse-IP

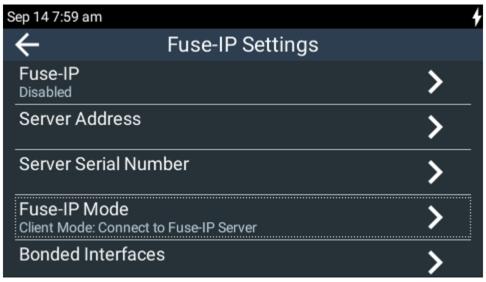
1. Press the **HOME** button to return to the **Home** screen and tap **Settings**.

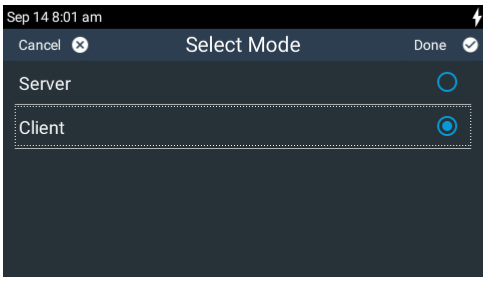


2. Tap **Transport Interfaces** to expand the menu if it is not displayed, then tap **Fuse-IP** to configure module settings.

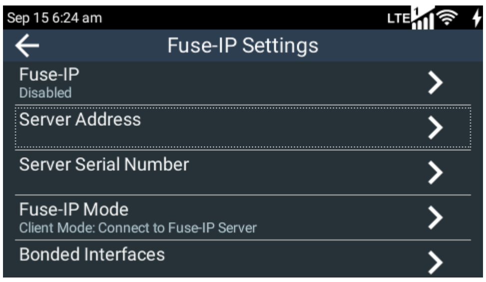


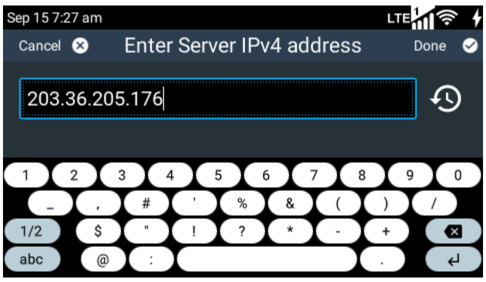
3. Configure the **Fuse-IP Mode** as **Client**.



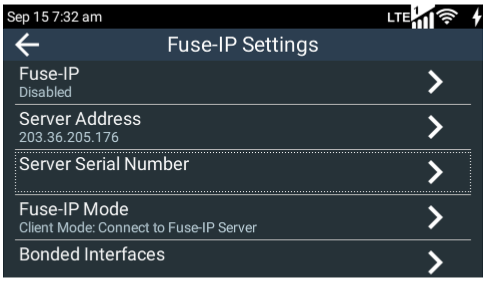


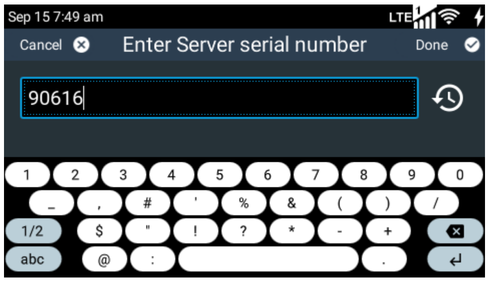
4. Tap **Server Address** to enter a public static IP address associated with the bonded interfaces at the studio, then tap **Done** in the top-right hand corner of the **TOUCH SCREEN**. Note: if the bonded interfaces have private addresses behind a firewall then port forwarding needs to be configured. See Installing the Codec at the Studio for more details on port forwarding.



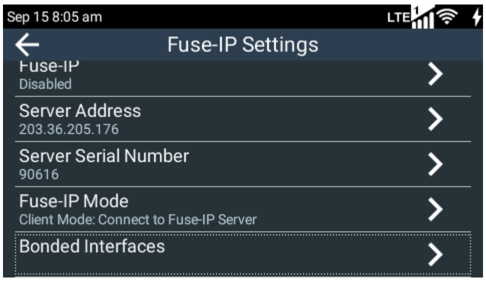


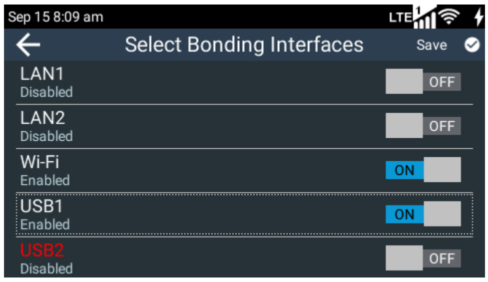
5. Tap **Server Serial Number** and enter the serial number of the server codec to which you are connecting, then tap **Done** in the top-right hand corner of the **TOUCH SCREEN**.



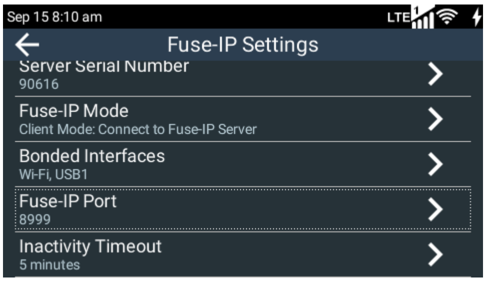


6. Tap **Bonded Interfaces** and then tap the **On/Off** button for the interfaces you want to bond. Next, tap **Save** in the top-right hand corner of the **TOUCH SCREEN**.

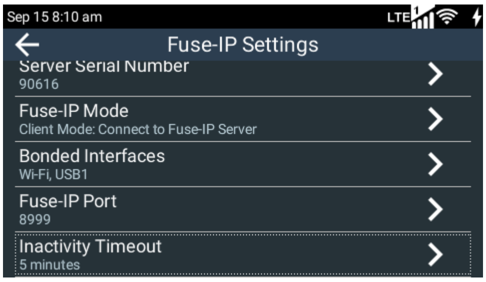




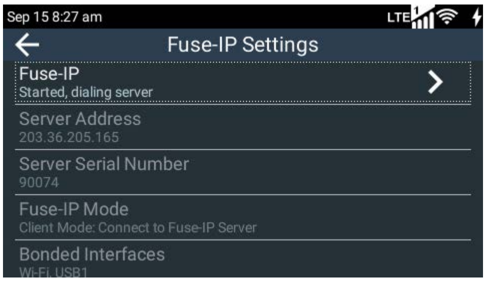
7. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.

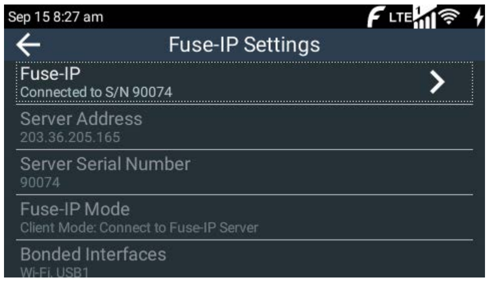


8. Tap **Inactivity Timeout** if you want to turn the Fuse-IP tunnel off after a predetermined time period to save data, then click **Save**. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter **0** to disable the timeout.

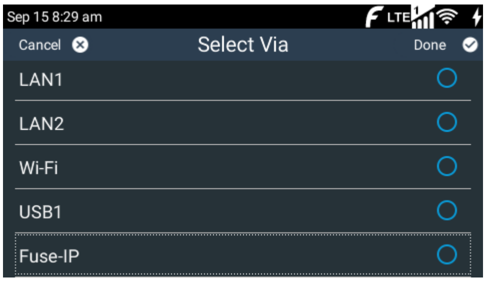


9. Tap **Fuse-IP** to create a Fuse-IP tunnel between the server and client codecs. The Fuse-IP symbol  is displayed in the **Status Bar** to confirm the two codecs have created an IP tunnel. Remember Fuse-IP must be enabled on both codecs. Please note: double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.





10. Select **Fuse-IP** as the **Via** interface with which to dial when creating a program in the HTML5 Toolbox Web-GUI **Program Manager panel** or the **Dialer** .



Important Notes:

- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, the codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- The codec remembers the Fuse-IP enabled/disabled state on power up
- For additional stability it is recommended that a fixed jitter buffer is configured when streaming using Fuse-IP. The actual jitter buffer depth should account for the difference in delay between the interfaces and the maximum jitter experienced. To determine the jitter over each link you can connect and stream audio over each interface separately and look at the jitter reading displayed on the **Connection Statistics** screen.
- Use a dotted quad IPv4 address when configuring the Fuse-IP Server Address.
- Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary **Via**.

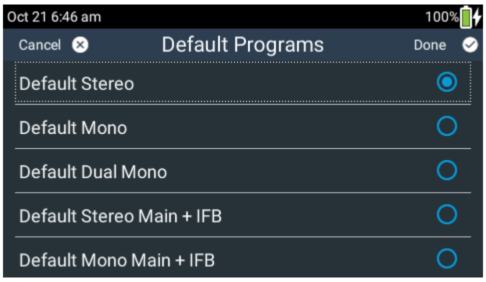
14.6 Connecting with ISDN

The following procedure will create a custom peer-to-peer ISDN connection program using the codec's **TOUCH SCREEN**. Attach an ISDN cable to the ISDN module installed in your codec. The codec displays ISDN line sync status in the **Status Bar** at the top of the **TOUCH SCREEN**.

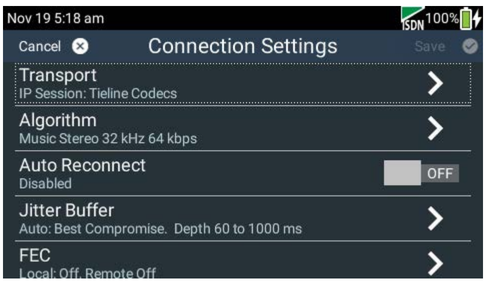
	Symbol	Description of Status
1	No ISDN Symbol	No ISDN module is installed in the codec.
2		An ISDN module is installed in the codec and either: • No ISDN line is attached to the codec, or • ISDN line is attached but line sync hasn't been detected by the module
3		An ISDN module is installed in the codec and line sync has been detected.

Important Notes:
- See Testing ISDN Connections for valuable information about setting up and maintaining reliable ISDN connections.
- See ISDN Module Configuration for details on module settings.
- See ISDN Answering Configuration for details on ISDN answering settings.
- Click here for details about configuring connections using the HTML5 Toolbox Web-GUI.

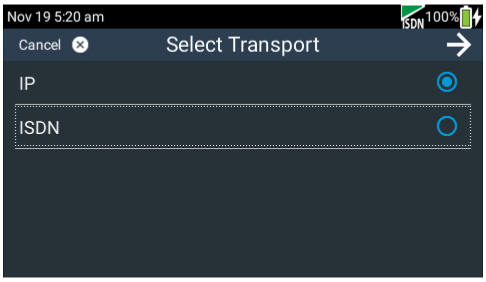
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap **Menu** and then tap **Default Programs** to view program connection options. Note: Default programs cannot be locked in the codec. Save a program as a custom program if a program needs to be locked.
3. Tap to select the preferred program connection option, e.g. **Default Mono** or **Default Stereo**, and then tap **Done**. Note: Tap **Load Program** to load preconfigured custom programs.



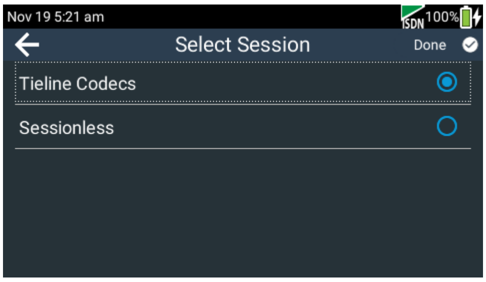
4. To adjust connection settings tap the **Connection Settings** symbol.
5. Tap **Transport** to adjust this setting if the current selection is IP.



6. Tap to select **ISDN** as the preferred connection type.

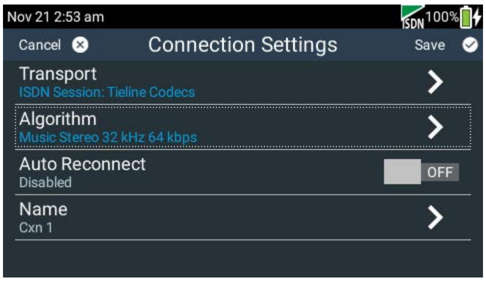


7. When dialing a Tieline codec tap to select the default **Tieline Codecs** Session Data setting, or tap **Sessionless** if dialing a non-Tieline codec.

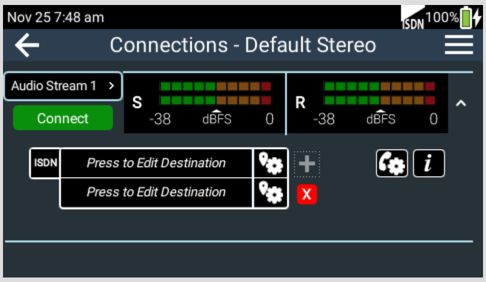


Important Note: By default, when Tieline codecs dial they send configuration settings to the remote codec using Tieline Session Data. This configures the codec receiving the call with matching algorithm, sample rate and bit rate settings. Non-Tieline devices don't support this feature and **Sessionless** must be selected to provide compatibility. See www.tieline.com for more compatibility information when dialing non-Tieline codecs.

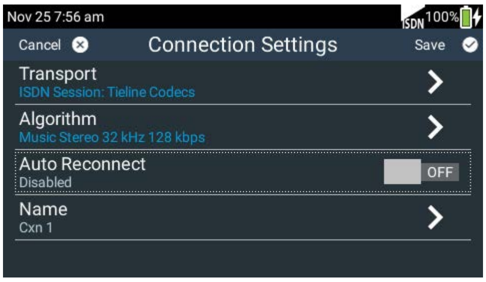
8. Tap **Algorithm** to reconfigure the selected algorithm and adjust the sample rate.



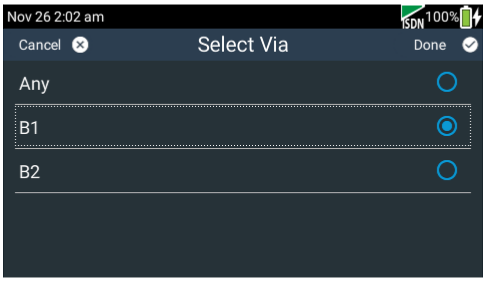
Important Note: When a stereo program template is selected, only supported algorithms and bit rates are displayed. To display 128kbps algorithm options, e.g. MP2 stereo at 128kbps, tap the **Plus symbol** on the **Connections screen** to add a second bonded 64kbps B channel connection.



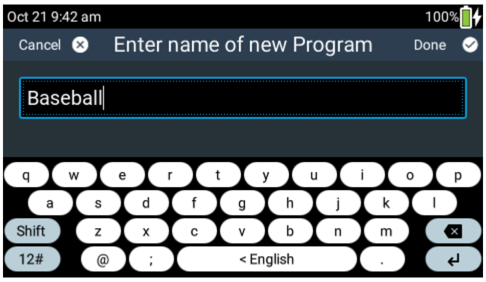
9. Tap to enable or disable **Audio Reconnect** as required and tap **Name** to edit the connection name. Tap **Save** to store all connection settings.



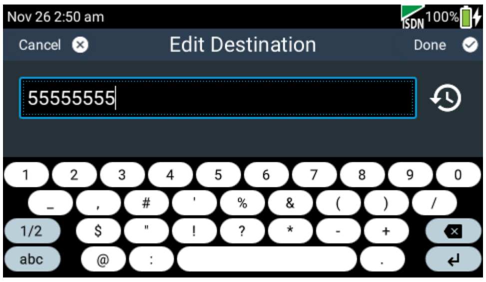
10. Tap the **Connection Settings** symbol on the **Connections screen** to specify a B channel dialing interface for each connection when more than one B channel is in use. Then tap **Done**. Note: **Any** is the default setting.



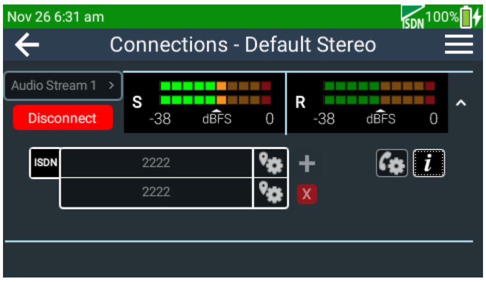
Helpful Hint: To save the program as a new custom program for later use tap **Menu** and then tap **Save as**, then name the program before connecting.



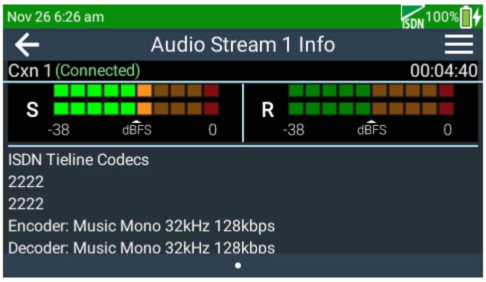
11. Tap **Press to Edit Destination** to enter the number for the codec being dialed, or tap the **Dial History** symbol to select a previously dialed number. Then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**. Note: If you are dialing over multiple B channels you will need to enter a **Destination** for each B-channel.



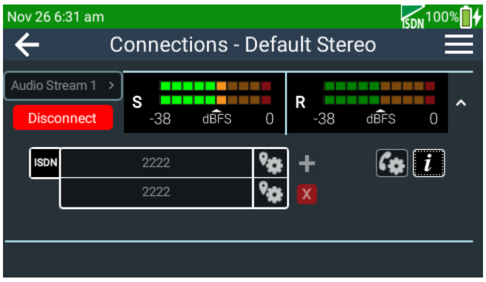
12. Tap **Connect** Connect or press the **CONNECT** button to dial the destination codec. The **Status Bar** turns green when the codec is connected.

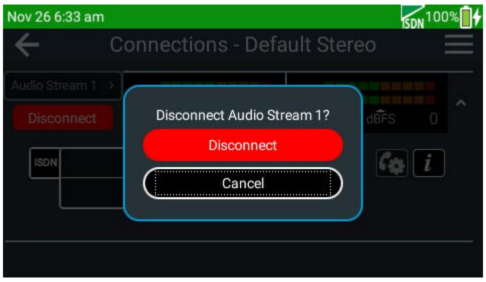


13. Tap the **Info** button to view audio stream connection details.



14. Tap **Disconnect** Disconnect or press the **DISCONNECT** button to hang up the connection.





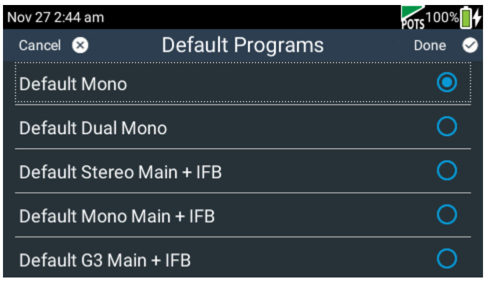
14.7 Connecting POTS

The following procedure will create a custom peer-to-peer POTS connection program using the codec's **TOUCH SCREEN**. Attach a POTS line to the module installed in your codec. The codec displays POTS line status in the **Status Bar** at the top of the **TOUCH SCREEN**.

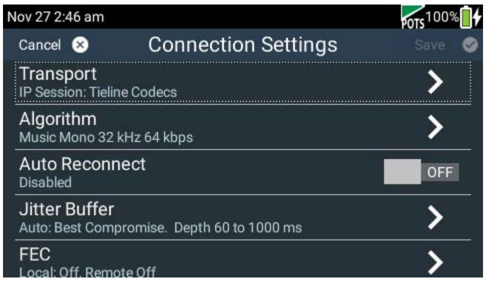
	Symbol	Description of Status
1	No POTS Symbol	No POTS module is installed in the codec.
2		A POTS module is installed in the codec but no POTS line is attached to the codec
3		A POTS module is installed in the codec and line voltage has been detected

Important Notes:
- Configure the correct country setting for connections over POTS to adjust the ViA POTS G5 module for varying ring tones and line impedances in different countries.
- See POTS Connection Tips and Precautions for valuable information about setting up and maintaining reliable POTS connections.
- See POTS Module Settings for details on module settings.
- See POTS Answering Configuration for details on POTS answering settings (required for answering calls from non-Tieline POTS codecs)
- See Configuring POTS for details on configuring codec connections via a computer.
- The **Local** and **Remote** line quality displayed for **POTS Codec** connections is related to the actual POTS line quality at either end of the link. This reading affects the maximum allowable bit rate when the codec is training and negotiating a connection.
- DO NOT connect the analog POTS module to a digital phone system. PERMANENT DAMAGE MAY OCCUR!

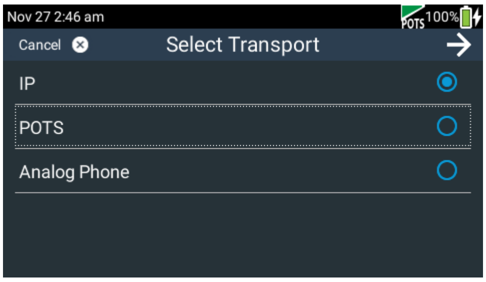
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap **Menu** and then tap **Default Programs** to view program connection options. Note: Default programs cannot be locked in the codec. Save a program as a custom program if a program needs to be locked.
3. Tap to select the preferred program connection option, e.g. **Default Mono**, and then tap **Done**. Note: Tap **Load Program** to load preconfigured custom programs.



4. To adjust connection settings tap the **Connection Settings** symbol on the **Connections screen**.
5. Tap **Transport** to adjust this setting if the current selection is IP.

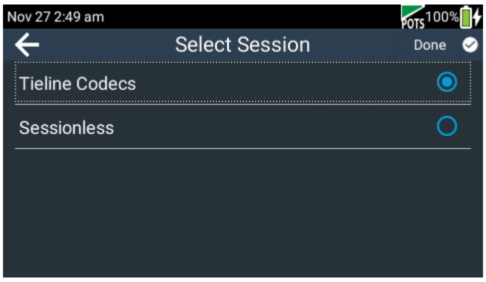


6. Tap to select **POTS** as the preferred connection type, or tap **Analog Phone** to configure a standard analog phone call.



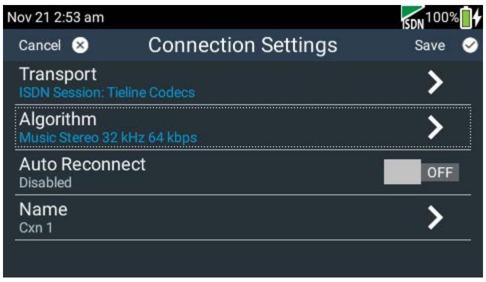
Important Note: When **Analog Phone** is configured, the codec displays a simplified **Connection Settings** menu with applicable settings.

7. When dialing a Tieline codec tap to select the default **Tieline Codecs** Session Data setting, or tap **Sessionless** if dialing a non-Tieline codec.



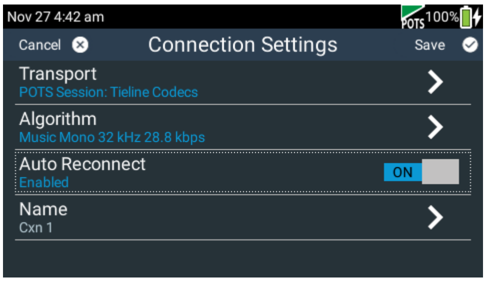
Important Note: By default, when Tieline codecs dial they send configuration settings to the remote codec using Tieline Session Data. This configures the codec receiving the call with matching algorithm, sample rate and bit rate settings. Non-Tieline devices don't support this feature and **Sessionless** must be selected to provide compatibility. See www.tieline.com for more compatibility information when dialing non-Tieline codecs.

8. Tap **Algorithm** to reconfigure the selected algorithm.



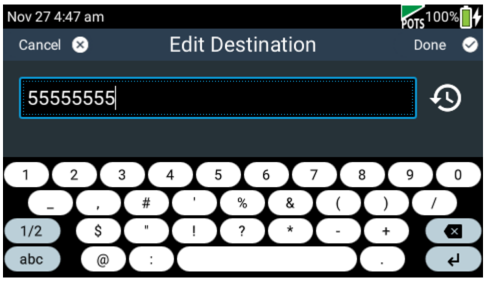
Important Notes: Tieline Music Mono is the algorithm automatically configured for **Tieline Codecs**. The **Other** algorithm is automatically configured when **Sessionless** is selected to support dialing a Comrex® Vector, Matrix® or BlueBox® codec. 9.6kbps connections are not supported by Comrex codecs.

9. Tap the **On/Off** button for **Auto reconnect** to toggle between enabling and disabling this feature. Note: See [Configuring Auto Reconnect] for more details.



10. Tap **Name** to adjust the connection name and then tap **Save** to store all settings and return to the **Connections screen**.

11. Tap **Press to Edit Destination** to enter the number for the codec being dialed, or tap the **Dial History** symbol to select a previously dialed number. Then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



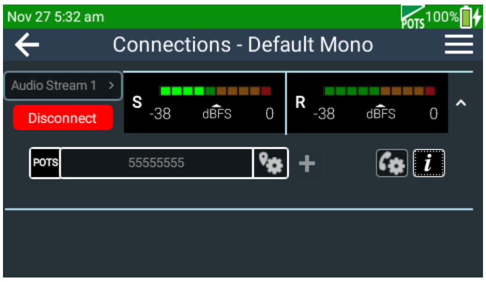
Important Note on Dial Pause Time:
When dialing out through a PBX the codec must find a line on two occasions:

- At the beginning of the call, and
- When the call goes into the POTS/PSTN network from the PBX.

The PSTN takes time to prepare the line for an outgoing call, i.e. disabling the incoming call circuitry. A pause is inserted by typing a comma "," in the dial string. The length of the pause may need to be adjusted when dialing through older PBX systems that have latency in connecting to the PSTN. Pauses can be set from zero up to two seconds in 250 millisecond increments. Each comma you insert will insert a pause of 250ms.

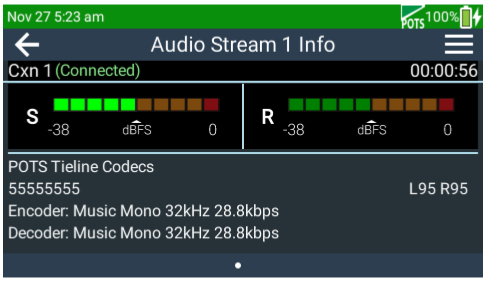
It is also usually necessary to add a pause when making long distance or international calls e.g. dialing Tieline 1,3178458000.

12. Tap **Connect** Connect or press the **CONNECT** button to dial the destination codec. Note:
- The **CXN LED** on the front of the unit flashes green when dialing and negotiating; it turns solid green when connected.
- The **Status Bar** turns green after modem negotiation when the connection stabilizes.

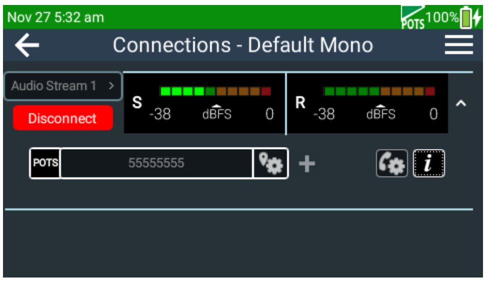


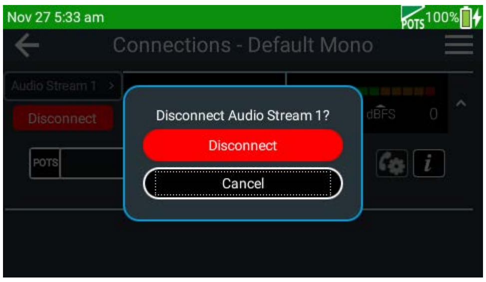
13. Tap the **Info** button to view audio stream connection details and **Local** and **Remote** line quality. The **Local** and **Remote** line quality displayed for **POTS Codec** connections is related to the actual POTS line quality at either end of the link. This reading affects the maximum allowable bit rate when the codec is training and negotiating a connection. It also indicates the stability of the connection when a call has been connected for a long period of time. If the line quality starts drop quite low after being connected for a long period, we recommend you retrain the connection to improve the line quality and avoid loss of audio.

Note: Incrementally negotiate higher connection bit rates by pressing the **F2** button and then the **NAVIGATE UP** ▲ button while viewing the **Audio Stream Info** screen; for lower bit rates press the **F2** button and then the **NAVIGATE DOWN** ▼ button.



14. Tap **Disconnect** Disconnect or press the **DISCONNECT** button to hang up the connection.





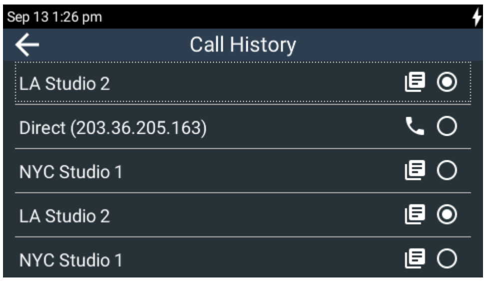
It is also possible to select a **Dial Route** if required. When routing multiple audio streams over transports like ISDN or POTS, you can use **Dial** and **Answer Routes** to configure deterministic routing of audio streams. See [POTS Answering Configuration] for more information.

Important Notes About Using POTS Codecs over Non-Standard Technologies:
Traditional copper POTS lines are being removed or replaced in many regions and some broadcasters find themselves trying to broadcast from areas where these traditional phone lines have been replaced by alternative solutions using fiber infrastructure. Migrating to these services with traditional legacy telephones or POTS codecs may be successful in some situations. However, unfortunately these voice services come in many different flavors around the world and there is no "one size fits all" solution. In our experience, if a service supports a traditional fax machine, then it is likely to support a Tieline codec connecting

and encoding over POTS. In some circumstances these services do not support data bit rates sufficient for high quality audio broadcasts.

15 Redialing a Connection

Press the **CONNECT** button in any screen except the **Connections screen** to view **Call History** and redial previous connections.



Manually dialed connections are saved as programs and retain all dialing and configuration information. Previous connections are identified in the **Call History** screen by their program name, or by the directly dialed number or address used.

16 Load, Connect and Manage Programs

By default, all Tieline codecs transmit proprietary session data when connecting to each other in order to establish, manage and terminate connections. When a connection between two codecs is established:

1. The dialing codec sends information about how the codec receiving the call should be configured.
2. After the answering codec receives session data successfully, it sends an acknowledgment to the dialing codec and streaming can commence.

What is a Program?

Tieline ViA, Genie, Merlin and Bridge-IT codecs also use 'programs' to connect to each other. A program configures a Tieline codec to send or receive one or more **Audio Streams** based upon the particular application the codec is being used for at any given time. The attributes of the audio stream and associated connections are embodied within a program when it is created including the configuration, dialing and answering parameters. Each audio stream within a program is defined separately and contains settings relating to the number of connections (e.g. primary and failover) and the number of destinations to which each audio stream is distributed.

Creating Programs

Default programs of all types can be created using the codec touch screen (see Connection Guide). If you know the IP address of the codec being dialed, simply enter this into the codec, select your preferred connection settings and tap **Connect**. These default programs can be named and saved as custom programs.

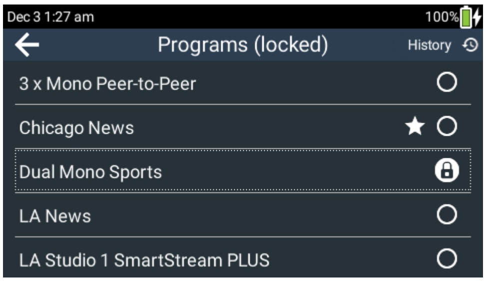
HTML5 Toolbox Web-GUI Programs

The HTML5 Toolbox web-GUI contains a **Program Manager panel** with a wizard for configuring program settings and backup connections for all supported program types. Connections can also be configured using the **Connections panel**. Edit settings easily at the touch of a button and use existing programs as templates for creating other programs. These can be saved and loaded onto a codec. Supported programs include:

- Mono program
- Stereo program
- Mono program with IFB
- Stereo program with IFB
- Dual mono program: includes two mono audio streams. Each audio stream includes a separate peer-to-peer connection to a different destination, which can also be configured with different transport, audio and backup settings.
- Triple mono program: includes three mono audio streams.

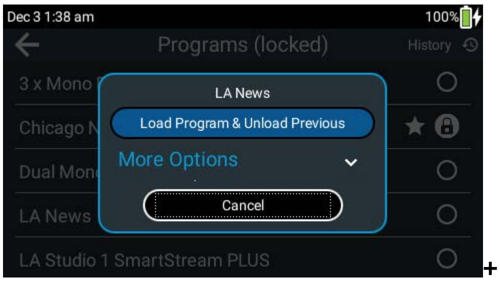
Load and Connect a Custom Program

1. Press the **HOME** button to return to the **Home** screen, then tap **Programs**.
2. Tap to select a saved program in the codec.

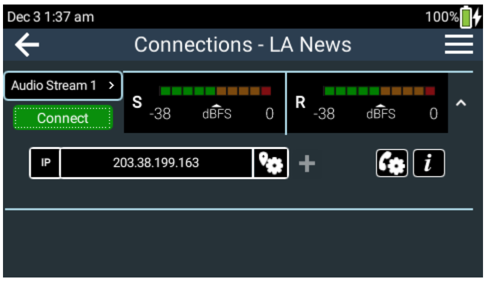


Important Note: It is possible to lock custom programs in a codec to ensure they are not unloaded by incoming calls. This is useful to support answering of multiple audio stream programs, or if you want to configure the codec to always use a particular jitter buffer or FEC setting. The **Lock symbol** is displayed in **Programs** to indicate a loaded program is locked. To learn more see Lock or Unlock a Program in the Codec.

3. Tap **Load Program & Unload Previous** to load a saved program and unload the current program. Note: Saved programs may have been configured using the HTML5 Toolbox Web-GUI, or locally via the **TOUCH SCREEN**.

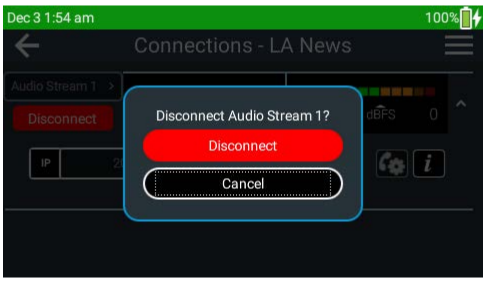


The **Connections** screen is displayed when you load a new program. Tap **Connect** to dial the program.



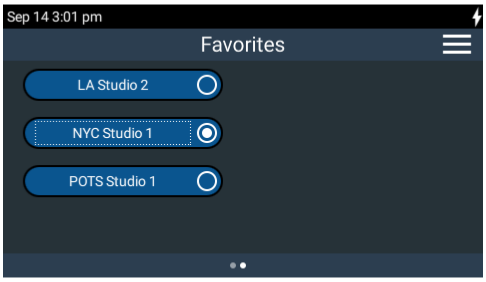
Disconnect a Program

1. Press the red **DISCONNECT** button at any time to hangup a connection.
2. Tap **Disconnect** or press the **DISCONNECT** button and then confirm hangup when the confirmation dialog appears.



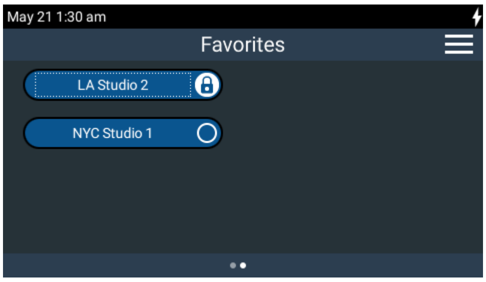
Add Programs to Favorites

Add programs as **Favorites** to view and connect them easily by swiping left from the **Home screen**. This makes it simple for non-technical users to connect and go live with preconfigured dialing configurations. See Administrator and Basic Operation Modes for more details about powering up on the **Favorites** screen by default, to make it simple for users to connect. In the following image the radio button of the currently loaded program is selected.

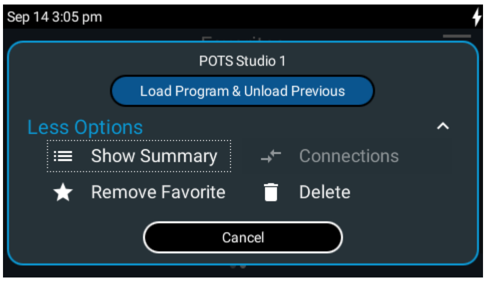


Favorites: NYC Studio1 Loaded

If the **Lock Program** feature is enabled, the **Lock symbol** is displayed for the currently loaded program.

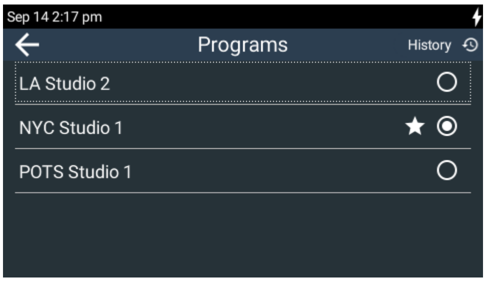


Helpful Hint: Touch and hold a favorite to view program editing features when operating in Admin Mode. This shortcut saves having to navigate to the **Programs** menu to edit program attributes.

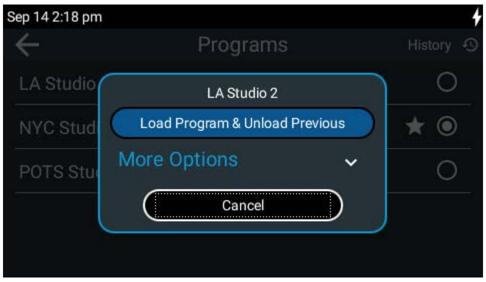


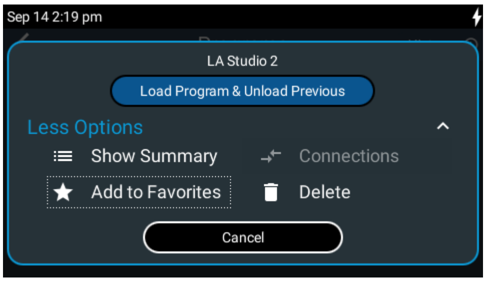
To add a program to **Favorites**:

1. Press the **HOME** button to return to the **Home** screen, then tap **Programs**.
2. Tap to select a program which is not a favorite, i.e. there is no **Star** symbol next to the radio button.

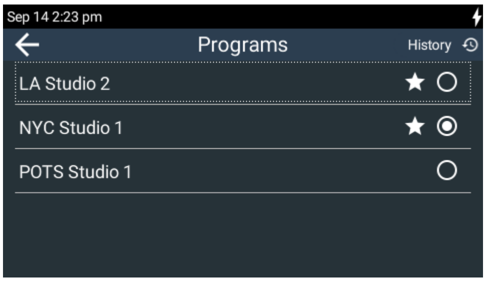


3. Tap **More Options** and then tap **Add to Favorites**.



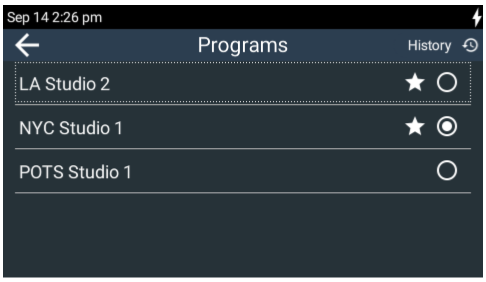


4. In this example, the program **LA Studio 2** appears in the **Programs** list with a **Star** symbol next to the radio button, indicating it has been added to the list of **Favorites**.

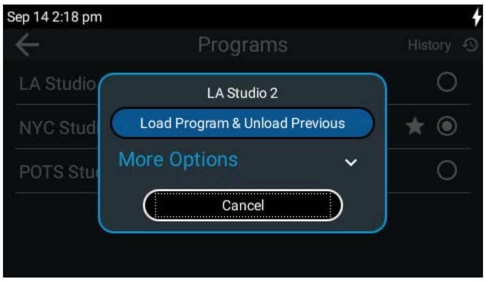


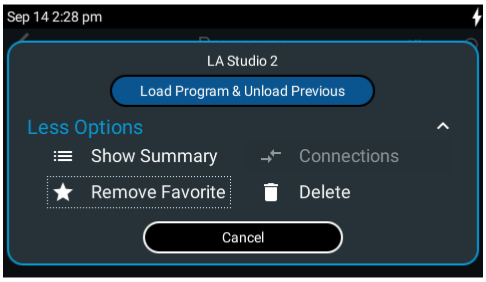
Remove Programs from Favorites

1. Press the **HOME** button to return to the **Home** screen, then tap **Programs**.
2. Tap to select a program which is a favorite, i.e. there is a **Star** symbol next to the radio button.

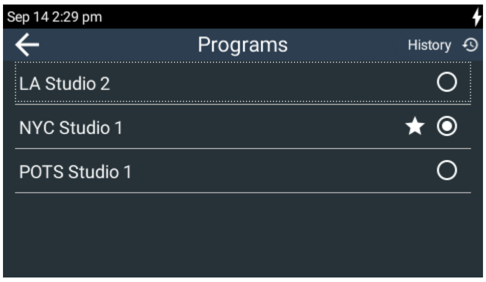


3. Tap **More Options** and then tap **Remove Favorite**.



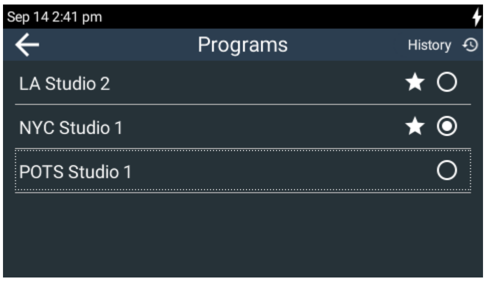


4. The example program **LA Studio 2** appears in the **Programs** list without a **Star** symbol next to the radio button, indicating it has been removed from **Favorites**.

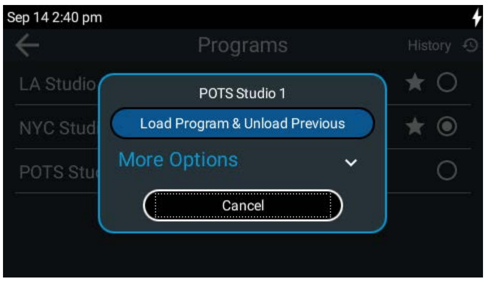


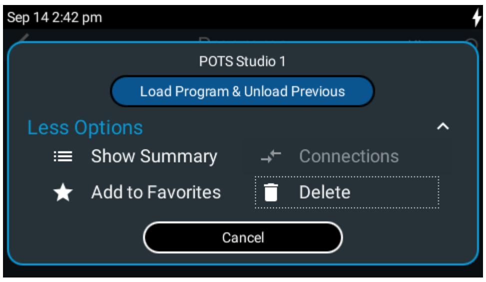
Delete a Program

1. Press the **HOME** button to return to the **Home** screen, then tap **Programs**.
2. Tap to select the program you want to delete.

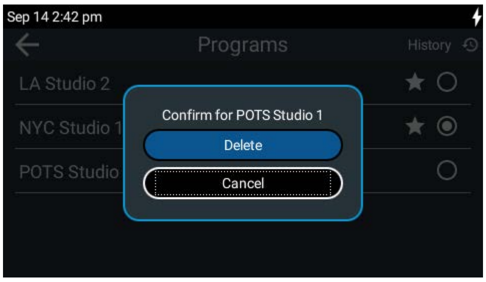


3. Tap **More Options** and then tap **Delete**.





4. Tap **Delete** in the confirmation dialog.



17 ViA Headphone Controls

WARNING: LISTENING TO AUDIO AT EXCESSIVE VOLUMES CAN CAUSE PERMANENT HEARING DAMAGE. REDUCE VOLUME AS LOW AS POSSIBLE.

A headphone volume limiter can be employed to protect hearing when monitoring loud sources and/or when using low impedance headphones. To configure this:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio**.
2. Tap **Headphones**.
3. Tap **Headphone Volume Limiter** to toggle between enabling and disabling this feature.

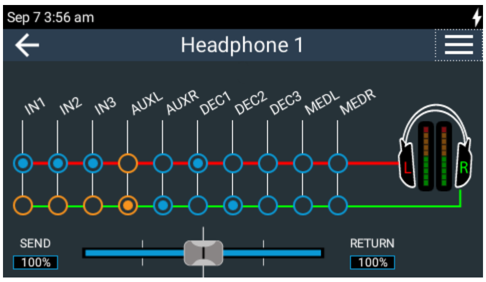
Adjust Headphone Levels

The codec has three 6.35mm (1/4") RTS stereo headphone outputs (**HP 1-3**) for monitoring inputs and return program audio. Use the green headphone **HP 1-3** rotary encoders to adjust the headphone level for each headphone output.

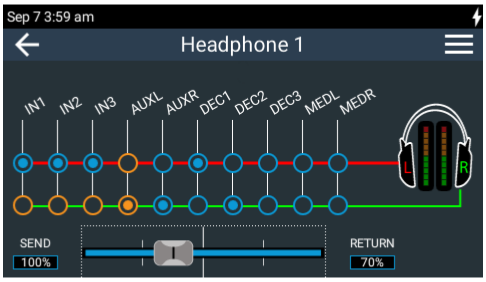
Adjust Headphone Monitoring Settings

1. Press the **SOURCE** button on inputs 1-3 to display the headphone monitoring screen for each headphone output.

2. To monitor different audio sources tap on the **TOUCH SCREEN** to select or deselect an audio crosspoint, or use the **NAVIGATION** buttons to focus on a crosspoint and press the **OK** button to select or deselect an audio crosspoint. Changes not yet saved as a custom headphone mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted. These routing changes are also reflected in the **Matrix Editor**, which displays all input/output routing in the codec.

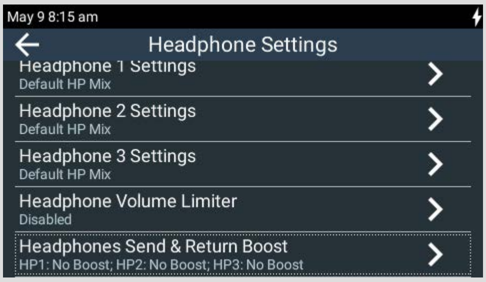


3. Adjust the **Send & Return Balance** slider to change the **Send** and **Return** audio headphone balance. Each headphone output has individual controls and fine adjustment in 2% increments is possible using the left and right arrow-shaped **NAVIGATION** buttons. This doesn't affect the level of the audio transmitted or received, only the local headphone monitoring level.

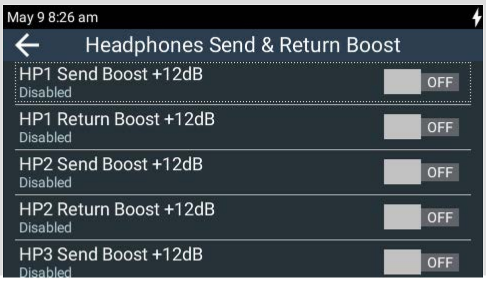


4. Swipe left on the **TOUCH SCREEN** to adjust the audio level of each individual headphone source. This allows the audio levels of feeds to be adjusted in relation to each other and tailored for each headphone output.

Important Notes: When the audio menu is enabled in administrator mode it is possible to configure the headphone mix via the **Headphone** menu. Select **Home** screen **> Audio** **> Headphones**. Then tap **Headphone Settings** for each headphone output and select the preferred headphone mix.



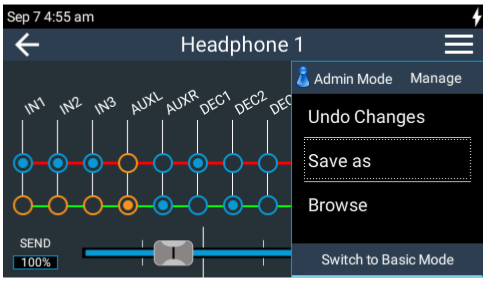
Tap **Headphones Send & Return Boost** to boost either the Send or Return audio for Headphones 1-3 by 12dB. This can be helpful when attempting to balance audio in noisy environments, or if incoming return audio is low.



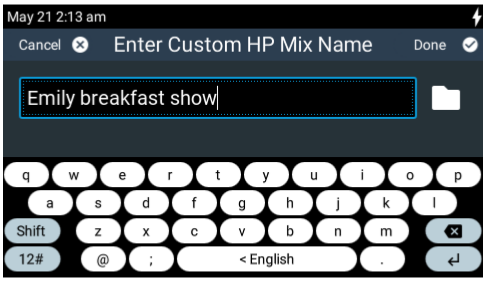
Save & Edit a Custom Headphone Mix

The codec supports saving and recalling custom headphone mixes configured for specific users and events. A custom headphone mix includes matrix crosspoint routing and **Send** and **Return** balance settings.To save edits to current headphone settings, or a custom headphone mix:

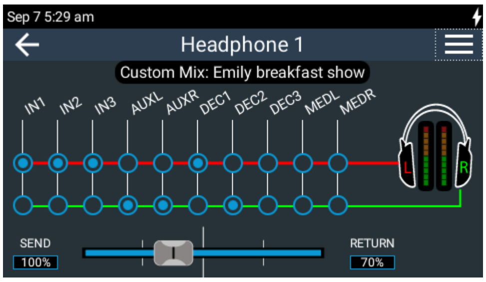
1. Press the **SOURCE** button on inputs 1-3 to display the headphone monitoring screen for a headphone output.
2. Tap **Menu** and then **Save as**.



3. Enter a custom headphone mix name and tap **Done**, or tap the file symbol to select an existing Custom HP Mix to either edit or overwrite it.

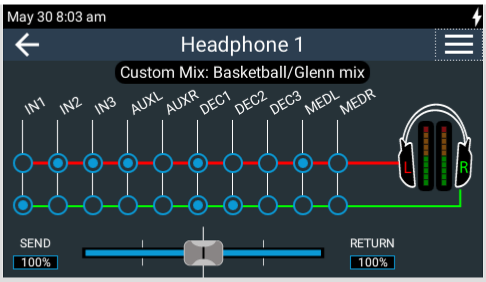


4. The custom mix will be saved and loaded and is displayed in the **Headphone** screen.



Important Notes:

- The mix displayed for **Headphone 1-3** in the panel is reflected in the loaded mix in the **Matrix Editor**. If you create custom headphone mixes and load them, and then save all routing in the **Matrix Editor** as a custom matrix, all the custom headphone mixes and their **Send/Return** balance settings are also saved. In essence, custom headphone mixes are "attached" to a **Matrix Editor** matrix when it is saved. This is displayed in the codec as per the following image. In this example, "Basketball" is the name of the custom **Matrix Editor** matrix and "Glenn mix" is the name of the custom headphone mix attached to this matrix.

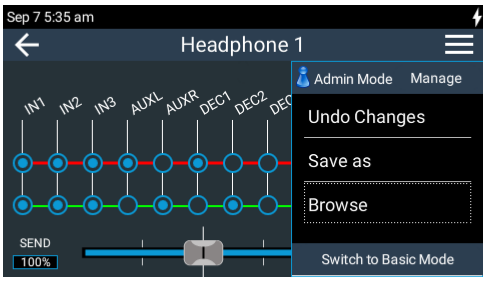


- If a custom headphone mix is saved with a custom matrix and the headphone mix is subsequently deleted, the custom headphone mix settings are retained in the saved custom matrix.
- Individual headphone source level adjustments for each headphone output are not saved with a **Matrix Editor** custom matrix. These individual source levels are accessed by swiping left in the **Headphone Mix** view on the **TOUCH SCREEN**.

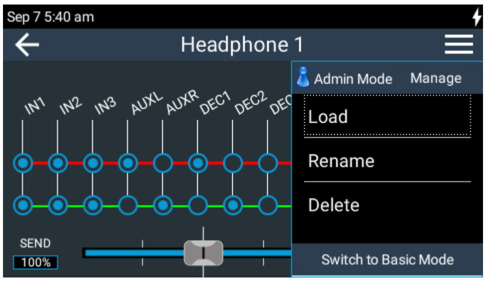
Load a Custom Headphone Mix

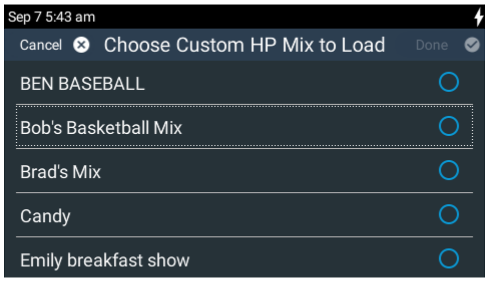
To load a saved custom headphone mix:

1. Press the **SOURCE** button to display the headphone monitoring screen.
2. Tap **Menu** ☰ and then tap **Browse**.



3. Tap **Load** and then tap to select a custom mix to load. Note: When you load a custom headphone mix the **Matrix Editor** will be adjusted based on the headphone mix loaded. Any edits to the **Matrix Editor** are displayed in orange.

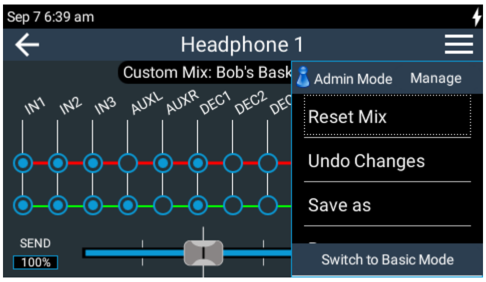




Reset Headphone Mix Settings

There are two options for resetting headphone matrix settings.

1. Press the **SOURCE** button to display the headphone monitoring screen.
2. Tap **Menu** ☰ and then either:
   a. Tap **Reset Mix** to reset routing to the matrix editor's headphone mix settings. This will unload a custom headphone mix and clear any runtime changes.
   b. Tap **Undo changes** to reset any runtime changes made to matrix editor headphone settings, or a custom headphone mix.



3. Tap **OK** in the confirmation dialog to reset the headphone matrix settings.

Important Note: The following scenarios lead to the same result:
1. **Reset Mix**.
2. **Undo Changes** when no custom headphones are loaded.

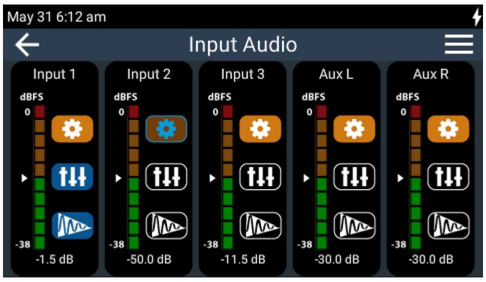
18 Input Levels and Input Settings

Important Notes:
- See Configuring AES3 and AES42 Input Audio for more info on digital input sources. Input audio functions can also be configured using the HTML5 Toolbox Web-GUI (see Configuring Input/Output Settings).
- Turn off all unused inputs to avoid additional noise in program audio.
- Audio signal processing occurs in the following order: IGC/Limiter (Intelligent Gain Control limiting on the input) > Input Compression > Input EQ > Mixer > Output AGC (Automatic Gain Control limiter).

The **Input Audio** screen has symbols to select input, EQ and compressor configuration settings. Different colored symbols make is simple to verify whether EQ or compression is enabled on each input.

Symbol	Explanation
	Input Settings Symbol
	Input Settings Symbol focus (when last selected)
	Equalizer Symbol
	Equalizer Symbol (EQ enabled)
	Compressor Symbol
	Compressor Symbol (Compressor enabled)

In the following image, the blue EQ and compression symbols indicate these features are enabled on **Input 1**.



Adjusting Analog Input Levels

Analog inputs 1-3 can be adjusted in two ways:

1. Coarse input gain via the **Input Config** screen.
2. Finer input level control via the **INPUT GAIN** rotary encoder.

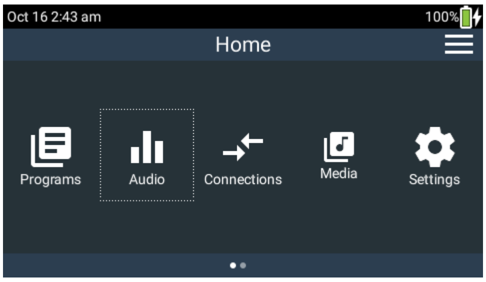
Important Note: Input 1 on the codec supports a mic or line level analog source, or AES3 (AES/EBU) format digital audio, or an AES42 digital microphone.

VOLTAGE WARNINGS:

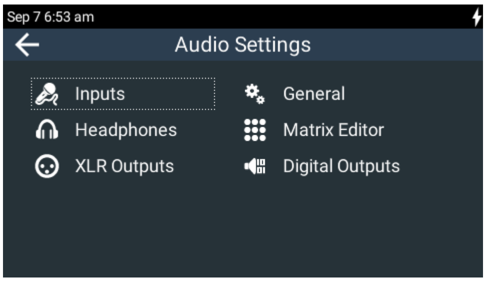
1. DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** input mode is selected, or equipment may be damaged by high voltages supplied in this mode. See Configuring AES3 and AES42 Input Audio for more info.
2. Do not attach an unbalanced input source (e.g. smartphone) to an analog XLR input if phantom power is enabled, or it may damage the connected device. Tieline recommends using an unbalanced to balanced converter to avoid damaging connected equipment.

To adjust coarse input gain:

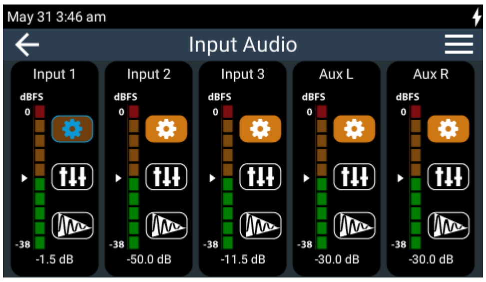
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio**.



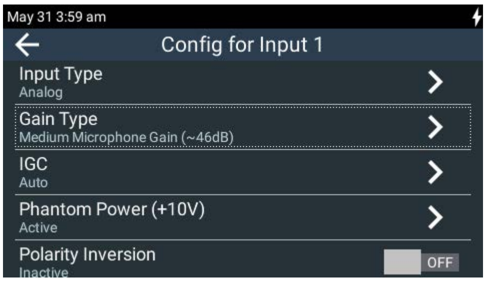
2. Tap **Inputs**.



3. Tap the **Settings symbol** on the input you want to adjust.



4. Tap **Input Type** and select **Analog**, then tap to select **Gain Type**.



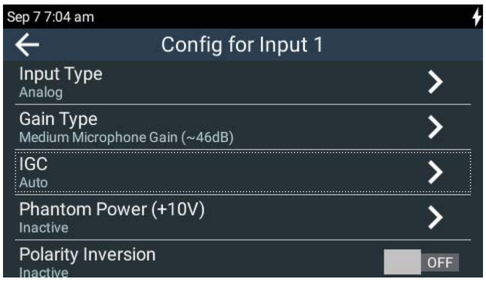
5. Tap to select the correct gain setting, or tap **Save** to keep the existing setting. There are 5 mic level settings as well as **Unbalanced** and **Line** level menu options.

Use the **INPUT GAIN** rotary encoders to make fine adjustments to input levels. Levels are indicated on **PPM LEDs** surrounding each **INPUT GAIN** rotary encoder. The current gain setting is indicated by a single **PPM LED** illuminated around the **INPUT GAIN** rotary encoder while it is being adjusted. More precise input audio levels are also indicated at the bottom of the **TOUCH SCREEN**. See PPM Meters and Analog Audio Outputs for more info.

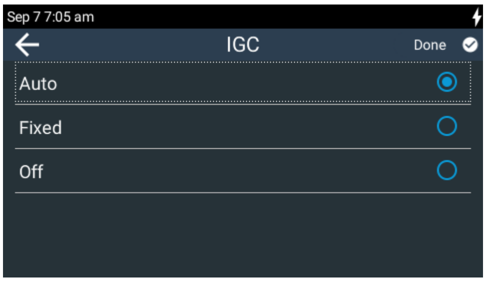
IGC (Intelligent Gain Control)

When the broadcast action really starts to heat up, the codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. **IGC** (Intelligent Gain Control) is enabled by default and is automatically activated at +20 dBu (G5 audio scale) and +14dBu (G3 audio scale) to prevent audio clipping.

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** **> Settings symbol** **[tap to select an input] > IGC**.



2. Tap to select **IGC**. There are three settings; **Auto**, **Fixed** and **Off**.



In the default **Auto** mode, the codec detects when incoming audio levels have reduced sufficiently after IGC has been activated and automatically returns input levels to the gain setting prior to IGC being activated. The codec takes just 250 milliseconds to detect audio levels have returned to normal after **IGC** has been initiated, and returns levels to the previous setting within half a second. This response is linear. In addition, the **INPUT ON/OFF** button will flash while IGC auto leveling is activated.

In **Fixed** mode, audio levels are automatically adjusted downwards until they are acceptable. The input level remains at this point until the **INPUT GAIN** rotary encoder is readjusted by the user. In addition, the **INPUT ON/OFF** button will flash until the rotary encoder is adjusted. AGC is also available on codec outputs. See General Audio Settings for more details.

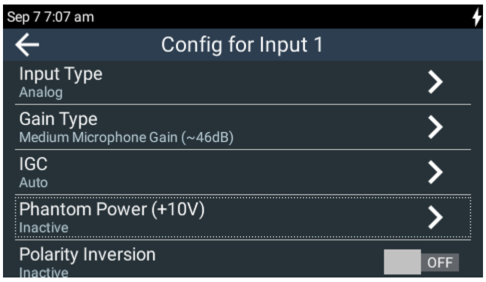
Phantom Power Settings

Phantom power can be enabled or disabled when inputs 1-3 are in analog mic level input mode (default setting disabled). Phantom power of 10V or 48V is supplied to all inputs when enabled. The default setting is 10V and the currently configured voltage is displayed in brackets next to **Phantom Power** in the **Input Config** screen.

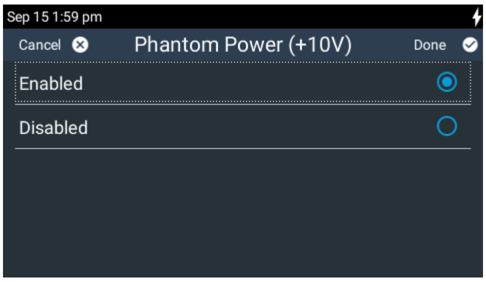
⚠ **VOLTAGE WARNING:** Check the specifications of all microphones attached to the codec to ensure they will not be damaged by either 10V or 48V phantom power supplied by the codec inputs.

Enabling Phantom Power

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio > Inputs > Settings symbol [tap to select an input] > Phantom Power**.

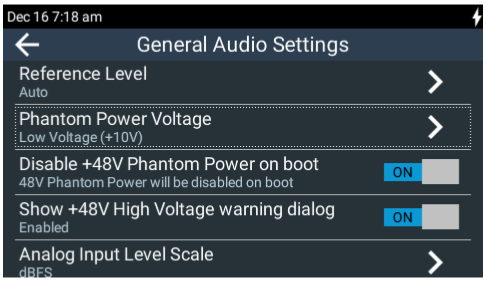


2. Tap **Enabled** to activate phantom power on the selected input.



Adjusting Phantom Power Voltage

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> General** **> Phantom Power Voltage**.



2. Tap to select the correct voltage to suit the microphone you are using.

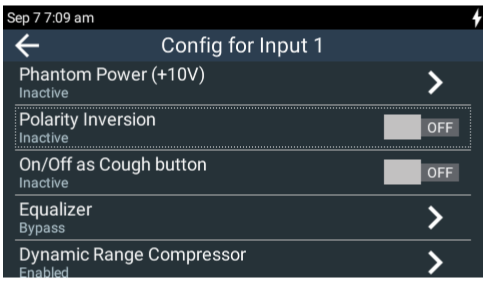
Important Note: The same phantom power voltage is applied across all compatible inputs. See General Audio Settings for more information about:

- Adjusting **Phantom Power Voltage**.
- Disabling phantom power each time the codec reboots (default **On**).
- Displaying phantom power warning dialogs (default **On**).

Polarity Inversion

Polarity Inversion inverts the polarity of an input when it is enabled.

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** **> Settings symbol** **[tap to select an input] > Polarity Inversion**.



2. Tap the **On/Off** button to toggle between enabling and disabling **Polarity Inversion** (default setting **Off**).

On/Off as Cough Button

Each **INPUT ON/OFF** button can be configured as a cough button. When **On/Off as Cough Button** is enabled on an input, press the **INPUT ON/OFF** button to mute the input for the duration of the button press.

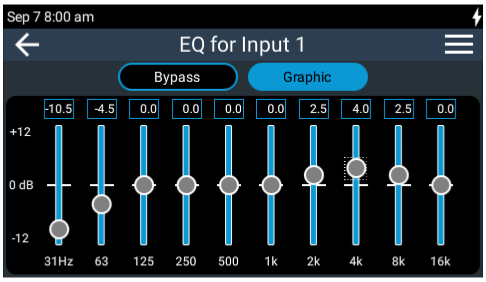
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** **> Settings symbol** **[tap to select an input] > On/Off as Cough Button**.

2. Tap the **On/Off** button to toggle between enabling and disabling **On/Off as Cough Button** (default setting **Off**).

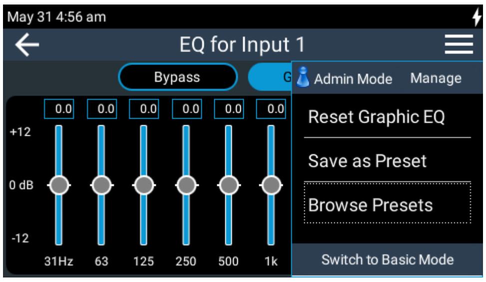
Equalizer

Each input has a 10 band graphic equalizer for adjusting EQ settings. To adjust EQ:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** **> Equalizer symbol** **[tap to select an input]**.



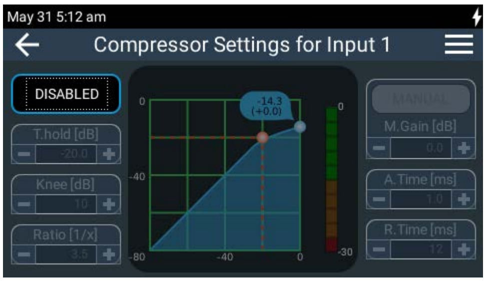
2. Tap a slider and swipe up and down the **TOUCH SCREEN** to boost or cut EQ at each frequency, or use the **NAVIGATION** buttons to select and boost or cut EQ in gradual increments.
3. Tap **Menu** to select from available options, which include: **Reset Graphic EQ** (to defaults); **Save as Preset**; or **Browse Presets** from a saved EQ Preset configuration.



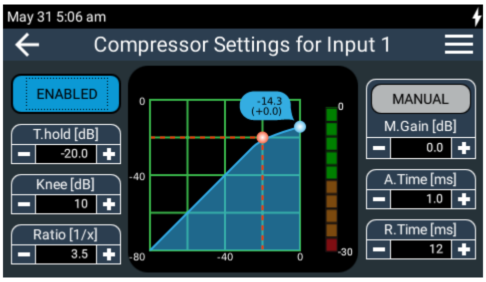
Compressor

Each input has a configurable dynamic range compressor.

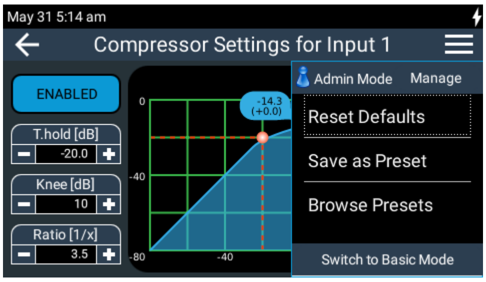
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** **> Compressor symbol** .



2. Tap the **Disabled** button to toggle it to **Enabled**.

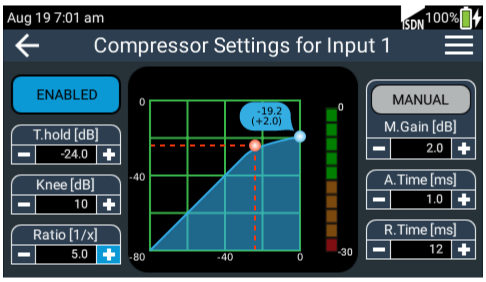


3. Tap to select and enter the value for each setting as required, or tap and drag to adjust parameters using the **TOUCH SCREEN**. Tap **Menu** ☰ to select from available options, which include: **Reset Defaults**; **Save as preset**; Browse **Presets** from a saved file.



Example of Input Compression Settings

The following settings can be used as a starting point for a sports broadcast with fairly consistent audio input levels and moderate fluctuations in dynamic range, with occasional louder audio due to excited play-by-play commentary.

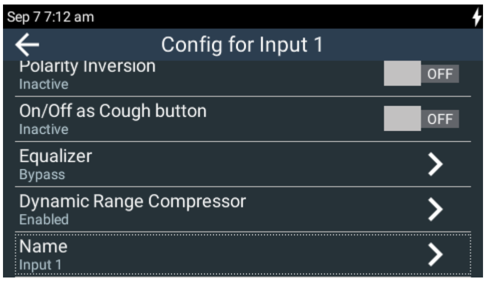


A higher compression **Ratio** can be used for broadcasts with very loud commentary calls.

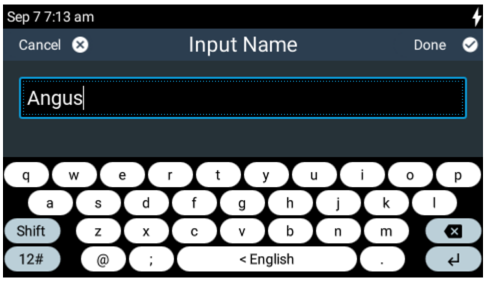
Rename an Input

Inputs 1-3 and the auxiliary input can be renamed as required.

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** **> Settings symbol** **[tap to select an input] > Name**.



2. Use the on-screen keyboard to enter the new input name, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



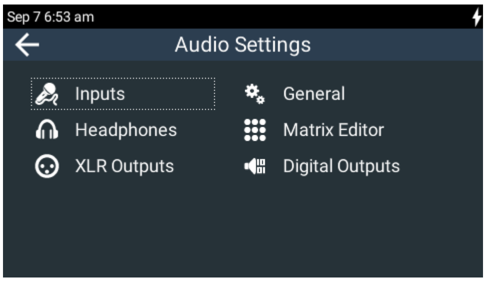
Auxiliary Input/Output Options

There are 3 stereo auxiliary input and 2 stereo auxiliary output options available in the codec:

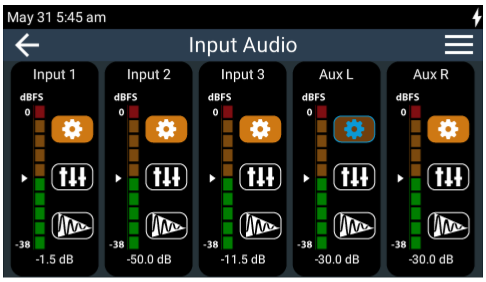
- An analog stereo line input via 1/8" (3.5mm) TRS jack.
- A stereo S/PDIF input and output via RCA connectors, supporting 32kHz, 44.1kHz and 48kHz sample rates.
- USB audio input and output via a micro USB connector.

To select an input option:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> Inputs** .

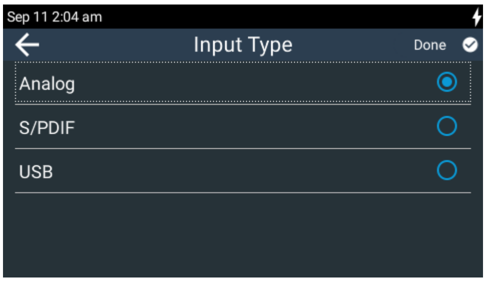


2. Tap the **Settings symbol** on the **Aux** input (left or right) you want to adjust.



3. Tap to select **Input Type**.

4. Tap to select your preferred auxiliary **Input Type**, or tap **Done** to keep the existing setting.



Important Notes:

- After selecting the type of auxiliary input being used, e.g. Analog, USB or S/PDIF, audio routing needs to be configured via the **Home** screen, then tap **Audio** and use either the **Matrix Editor** or **XLR Outputs** matrix, or **Digital Outputs** matrix to adjust signal routing.
- Analog, S/PDIF or USB auxiliary output audio is only available if the auxiliary input is also selected as that source. E.g. Select **Analog** as the Aux **Input Type** to enable analog auxiliary output audio, or select **USB** as the Aux **Input Type** to enable USB auxiliary output audio.
- There is a maximum of 6dB of additional gain available when adjusting a digital auxiliary input.
- Press the **INPUT ON/OFF** button to turn the auxiliary input **Off** if you are not using it. This reduces any unwanted additional noise in program audio.
- The codec should be detected by a Windows PC and drivers should update automatically when attached to a USB port for audio playback. If not, select **Start button > Control Panel > Device Manager > Universal Serial Bus Controllers**. Then right-click **USB Composite Device** and select **Update Driver Software**.
- On Mac OS X El Capitan v10.11 or macOS Sierra v10.12: Open "System Preferences", then select "Sound", then select "Output." Select "Tieline G5 ViA Digital Audio" as the output source.

Other Auxiliary Input Options

Other input options available include:

1. **Polarity Inversion**: Tap the **On/Off** button to toggle inversion of the selected input (default **Off**).
2. **Name:** Customize the auxiliary input name displayed.

19 PPM Meters and Analog Audio Outputs

A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the Tieline G5 reference scale setting. For more information on reference scales see General Audio Settings.

Important Notes:

- Front panel LED **PPM L** displays the Encoder 1 mix.
- Front panel LED **PPM R** displays the Encoder 2 mix.
- Front panel LED **PPM RTN** displays a mix of Decoders 1 & 2.
- The PPM examples which follow are based on default program routing of inputs to encoders and return audio to decoders. To learn more about adjusting encoder and decoder routing see Using the Matrix Editor.
- Please use the software PPMs on the **TOUCH SCREEN** or in the HTML5 Toolbox Web-GUI to view PPMs for all encoders and decoders.

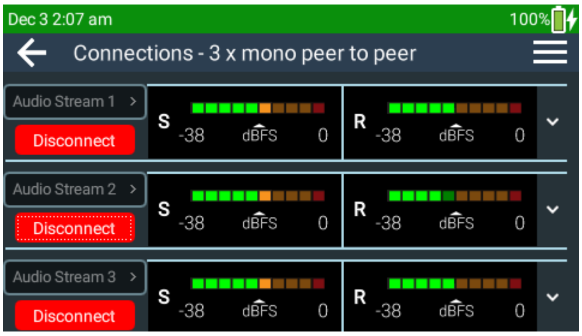
Setting Levels with PPM Meters

The **INPUT PPM METERS** surrounding the **INPUT GAIN** rotary encoders, and the PPM meters on the **TOUCH SCREEN**, use the dBFS audio scale. Set levels so that audio peaks average around the first **ORANGE LED**.

When adjusting input levels, a **GREEN** or **ORANGE LED** surrounding each **INPUT GAIN** rotary encoder displays the amount of gain wound into the input. This disappears after a few seconds if audio is present and normal PPM metering resumes.

Set mixed audio levels so that audio peaks average around the first **ORANGE LED** on the **LEFT**, **RIGHT** and **RETURN PPMs** on the front of the codec, or the PPM meters on the **Connections** screen. This represents nominal 0 VU at -18dBFS (+4dBU). Audio peaks can safely reach 0dBFS (+22 dBu) without clipping, providing 18dB of headroom.

PPMs on the Codec

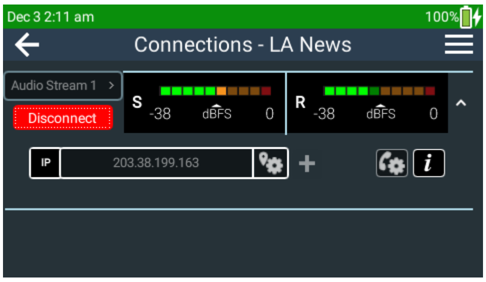


PPMs on the Connections screen

PPM Meters for Mono Peer-to-Peer Programs

When a mono program is connected the codec displays a mix of all inputs on the **LEFT PPM** and mono return audio on the **RETURN PPM**. By default, all inputs are mixed in the outgoing mono program audio stream.

The **Connections** screen PPM meters show program audio on the **SEND PPM** and return audio on the **RETURN PPM**.

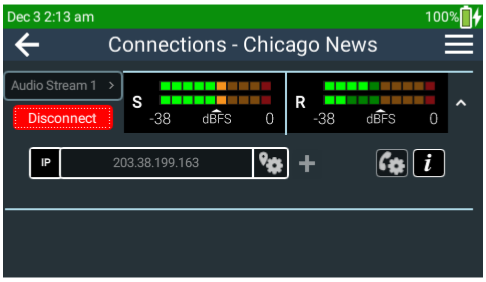


By default, mono incoming return audio from decoder 1 is routed to both analog **OUTPUT 1** and **OUTPUT 2**.

PPM Meters for Stereo Peer-to-Peer Programs

When a stereo program is connected the codec displays outgoing program on the **LEFT PPM** and **RIGHT PPM**. A mono mix of stereo return audio is displayed on the **RETURN PPM**. Inputs 1-3 are routed to both encoders 1 and 2. To provide stereo from the auxiliary input and for audio file playback, the left channel is routed to encoder 1 and the right channel is routed to encoder 2.

The **Connections** screen PPM meters display outgoing stereo program audio on the left and right **SEND PPMs**. Return audio is displayed on the left and right **RETURN PPMs**.

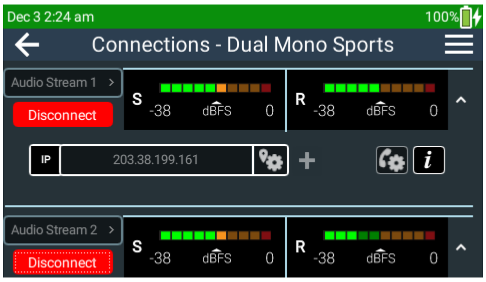


By default, stereo return audio is routed to analog **OUTPUT 1** (decoder 1) and **OUTPUT 2** (decoder 2).

PPM Meters for 2 x Mono Peer-to-Peer Programs

The codec is capable of creating two independent mono peer-to-peer connections simultaneously. By default, all inputs are mixed in mono for both outgoing audio streams. The codec displays a mix of all inputs on the **LEFT PPM** for the first audio stream, and a mix of all inputs on the **RIGHT PPM** for the second audio stream. The sum of both mono return audio channels is displayed on the **RETURN PPM**.

The left hand **Connections** screen PPM meters display outgoing program audio on the **SEND PPM** and return audio on the **RETURN PPM** for both connections.

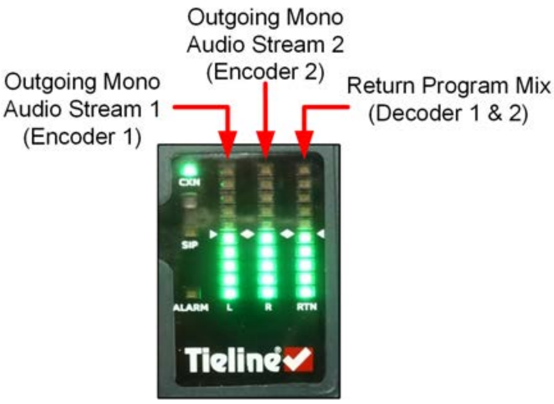


By default, return audio from the first audio stream (decoder 1) is routed to analog **OUTPUT 1**. Return audio from the second audio stream (decoder 2) is routed to **OUTPUT 2**.

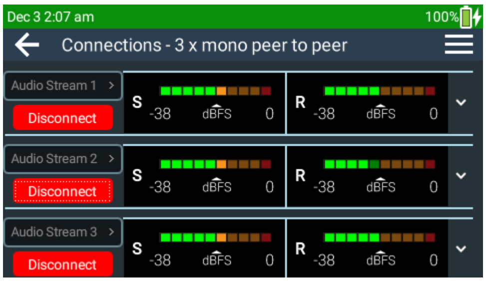
PPM Meters for 3 x Mono Peer-to-Peer Programs

The codec is capable of creating three independent mono peer-to-peer connections simultaneously. By default, input 1 is routed to encoder 1, input 2 is routed to encoder 2, and input 3 is routed to encoder 3.

The codec displays a mix of all inputs for the first audio stream on the **LEFT PPM** and a mix of all inputs for the second audio stream on the **RIGHT PPM**. There is no PPM mix displayed for the third audio stream. The **RETURN PPM** displays the sum of return audio for the first and second audio streams.



The left hand **Connections** screen PPM meters display outgoing program audio on the **SEND PPM** and return audio on the **RETURN PPM** for all three connections.

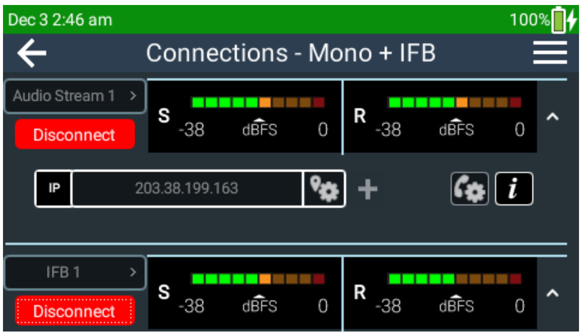


PPMs on the Connections screen

PPM Meters for Mono + IFB Programs

The codec is capable of broadcasting a bidirectional mono peer-to-peer audio stream and a separate bidirectional IFB (talkback/communications) audio stream. By default, all inputs are mixed into the outgoing mono program audio stream. The codec displays a program audio mix of all inputs on the **LEFT PPM** and mono return audio on the **RETURN PPM**. Outgoing talkback audio is displayed on the **RIGHT PPM**.

The left hand **Connections** screen PPM meters display outgoing program audio and outgoing talkback (IFB) audio on the **SEND PPMs**. The right-hand PPMs display return program audio and incoming communications audio on the **RETURN PPMs**.

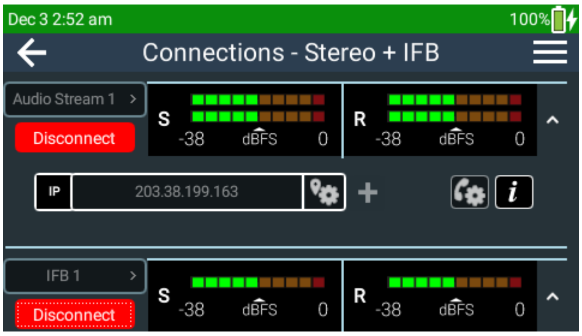


By default, return audio from the program audio stream (first connection) is routed to both analog **OUTPUT 1** (decoder 1) and **OUTPUT 2** (decoder 2).

PPM Meters for Stereo + IFB Programs

The codec is capable of broadcasting a bidirectional stereo peer-to-peer audio stream and a separate bidirectional IFB (talkback/communications) audio stream. Inputs 1-3 are routed to both encoders 1 and 2. To provide stereo from the auxiliary input and audio file playback, the left channel is routed to encoder 1 and the right channel is routed to encoder 2. By default, outgoing stereo program audio is displayed on the **LEFT PPM** and **RIGHT PPM**. A mono mix of stereo return audio is displayed on the **RETURN PPM**.

The left hand **Connections** screen PPM meters display outgoing stereo program audio and outgoing talkback (IFB) audio on the **SEND PPMs**. The right-hand PPMs display return stereo program audio and incoming communications audio on the **RETURN PPMs**.



By default, stereo return audio is routed to analog **OUTPUT 1** (decoder 1) and **OUTPUT 2** (decoder 2).

Important Note: To learn more about the Matrix Editor see Using the Matrix Editor. To learn more about how Cue and Talkback functions operate see Cue and Talkback Operation.

20 Using the Matrix Editor

The matrix editor in the codec allows any input to be routed to any output. Default routing settings are configured for each program type and these default matrices can be edited and saved as custom matrices. Custom matrices can be recalled as required. All saved custom matrices are available if a compatible program is loaded. If a matrix is not compatible with a program type it will not be visible in the menu, e.g. a saved stereo matrix is not visible when a mono program is loaded.

Custom matrices can be created and saved and then backed up with program and scheduler data. This allows them to be copied between codecs by using the Backup and Restore feature.

Important Notes:

- If you create custom headphone mixes and load them, and then save all routing in the **Matrix Editor** as a custom matrix, all the custom headphone mixes and their **Send/Return** balance settings are also saved. In essence, custom headphone mixes are "attached" to a **Matrix Editor** matrix when it is saved.
- Only crosspoint routing settings for XLR and Digital Outputs are stored in a custom matrix. The send/return balance, ganging, output mute and output level settings are not saved.

Viewing the Matrix Editor

To view matrices open the **Matrix Editor** by selecting **HOME** **> Audio** **> Matrix Editor**. Inputs are listed on the left and outputs are displayed at the top of the matrix.

Tap **Menu** to view matrix editor options. These options include:

1. **Undo Changes**: Clears any changes that have not been saved.
2. **Reset Defaults**: Resets the matrix to defaults for the currently loaded program.
3. **Cue/TB Edit**: Opens the Cue/TB Settings screen.
4. **Save as**: Save the current matrix settings as a new **Custom Mix** with a unique name (includes headphone and Cue/TB Mix matrices).
5. **Browse**: View saved Custom Mixes and load them.

Matrix Editor Editing

Routing can be adjusted very simply in the matrix editor. When a crosspoint is selected audio is routed from the input on the left to the output at the top. To edit default matrix settings, either:

1. Tap a crosspoint on the **TOUCH SCREEN** to select or deselect an audio crosspoint, or
2. Use the **NAVIGATION** buttons to focus on a crosspoint and press the **OK** button to select or deselect an audio crosspoint.

Any edits take effect immediately in:

1. The Headphone mix accessed by pressing the **SOURCE** button.
2. The **XLR Outputs** and **Digital Outputs** accessed via **HOME** **> Audio** **> XLR Outputs** **/ Digital Outputs**.

In the following example, the auxiliary input (**AUXL** and **AUXR**) have been deselected from the encoder outputs (**ENC1** and **ENC2**) and have instead been routed to the analog XLR outputs (**OUT1** and **OUT2**). Note: Changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.

Edits can be saved as a custom mix, which also includes any edits to **Headphone**, **XLR Output**, **Digital Output** or **Cue/TB Mix** matrices. To save a custom mix:

1. Tap **Menu** and then **Save as**.

2. Enter a new **Mix Name** and tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

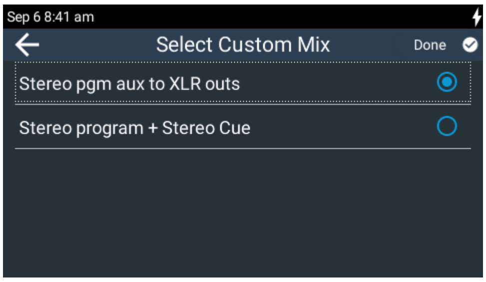
3. The new custom mix stays loaded in the codec until a new mix is loaded, an incompatible program is loaded, or program defaults are restored via **Menu** **> Reset Defaults**.

Important Notes:
- If a new program is loaded and it is compatible with the current custom mix then it will remain loaded. If an incompatible program type is loaded, the last compatible mix to suit the new program type is loaded, including any runtime changes made previously. E.g. If a custom stereo mix is loaded and then a mono program is loaded, the last mono mix used will be loaded.
- If you make runtime matrix edits to a loaded mono program, then load a stereo program, and subsequently reload the original mono program, the runtime matrix edits are recalled.
- Edits to default program matrices can be saved as a custom matrix if required. It is possible to save edits to a custom program without renaming it.

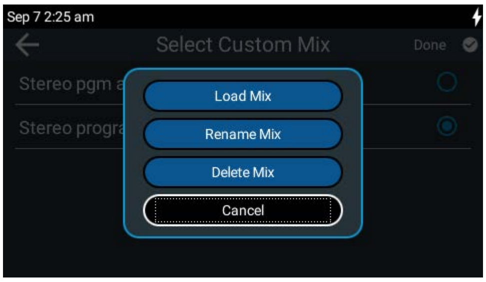
Load, Rename and Delete a Custom Matrix Editor Mix

When a new program is loaded it may be necessary to load a new custom mix. It is also possible to rename or delete mixes that are no longer required.

1. Select **HOME** > **Audio** > **Matrix Editor** .
2. Tap **Menu** and then **Browse**.
3. Tap to select a saved **Custom Mix**.



4. Tap to load, rename or delete a **Custom Mix**.



21 Cue and Talkback Configuration

The codec supports both cue and talkback functions. Cue provides:

- Offline communications between local announcers via the headphone jack outputs when the **TB CUE** button is pressed on inputs 1-3.
- Offline monitoring of the auxiliary input via headphone output 1 (**HP1** only by default).

Talkback communication is available on inputs 1-3 between the remote codec and the studio. This requires a dedicated IFB connection when using a:

- Mono Peer-to-Peer + IFB program, or
- Stereo Peer-to-Peer + IFB program.

It is also possible to broadcast mono program and use a separate communications channel, with either the Tieline Music or Music PLUS algorithm, over a stereo program using the **G3 Mono Peer-to-Peer + IFB** program template.

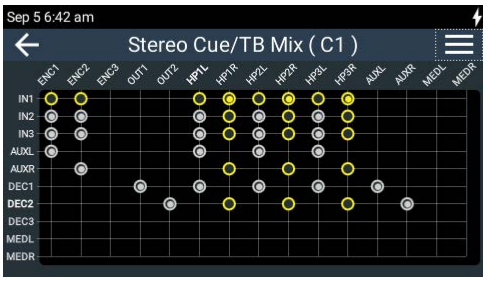
Configuring Cue and Talkback Matrix Routing

There is a separate Cue/TB matrix for each headphone output and for each program type, e.g. mono, stereo, Mono + IFB, Stereo + IFB. To view these matrices open the **Matrix Editor** by selecting **HOME** > **Audio** > **Matrix Editor**. Inputs are listed on the left and outputs are displayed at the top of the matrix.

There are three ways to view the **Cue/TB Mix** screen from the **Matrix Editor** screen:

1. To momentarily view **Cue/TB Mix** routing, briefly press the **TB CUE** button on inputs 1-3, or the **CUE** button on the stereo auxiliary input.
2. To view the **Cue/TB Mix** screen and edit routing press and hold the **TB CUE** button on inputs 1-3, or the **CUE** button on the stereo auxiliary input for 3 seconds. Note: it is possible to overlay more than one **Cue/TB Mix** on the screen simultaneously. The **TB CUE** button is illuminated if a mix is active. Cue and talkback audio is also audible.
3. View and edit the **Cue/TB Mix** screen via **Menu** > **Cue/TB Edit** > **Cue 1-3/Aux** > **Edit**.

Cue and talkback crosspoint routing is overlaid in the matrix and indicated in yellow. In the following default stereo cue matrix example, when the **TB CUE** button for input 1 is pressed, audio from input one is audible locally in the right side of headphones 1-3. In other words, the **Cue/TB Mix** overrides **Matrix Editor** audio routing while the button is pressed.

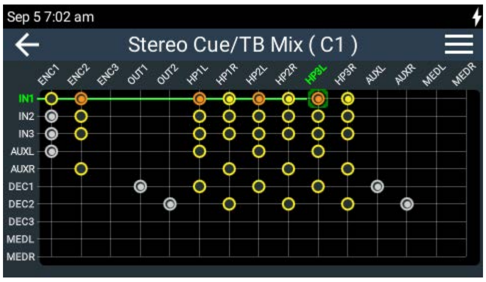


Editing Cue and Talkback Routing

To edit the routing of cue and talkback audio, either:

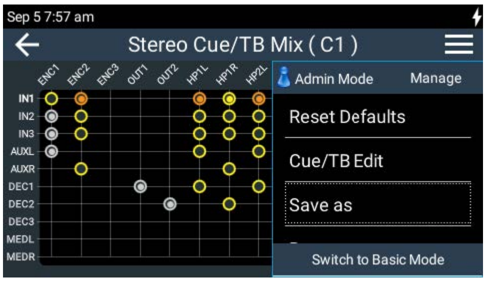
1. Tap a crosspoint on the **TOUCH SCREEN** to select or deselect an audio crosspoint, or
2. Use the **NAVIGATION** buttons to focus on a crosspoint and press the **OK** button to select or deselect an audio crosspoint.

In the following example, audio routing edits have been applied to select encoder 2 (**ENC2**) and the left side of headphones 1-3. Note: Changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.



Edits to the **Cue/TB Mix** for each headphone output can be saved as a **Custom Mix** with all the other routing displayed in the **Matrix Editor**. To save all routing edits:

1. Tap **Menu** ☰ and then **Save as**.



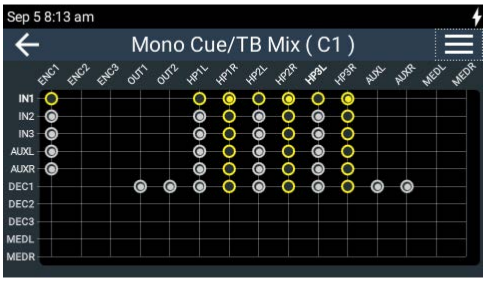
2. Enter a new **Mix Name** and tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

Understanding Cue Mode

Typically, local cue intercom is configured for commentators to talk to each other offline. In cue mode, offline communications audio from inputs 1-3 is routed to the right side of all local headphone outputs (**HP 1-3**) when the **TB CUE** button is pressed. By default this is the same for:

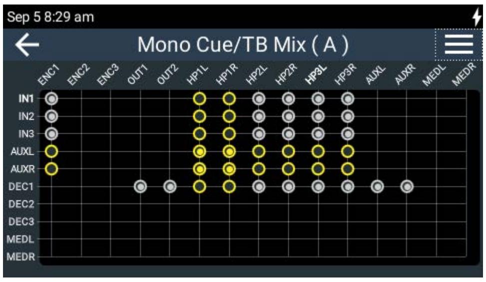
- Mono Peer-to-Peer programs.
- Stereo Peer-to-Peer programs.
- 2 x Mono Peer-to-Peer programs.

In the following matrix routing example, the **CUE TB** button for input 1 has been pressed and a mono program is loaded in the codec. Cue audio is audible in the right side of **HP 1-3**.

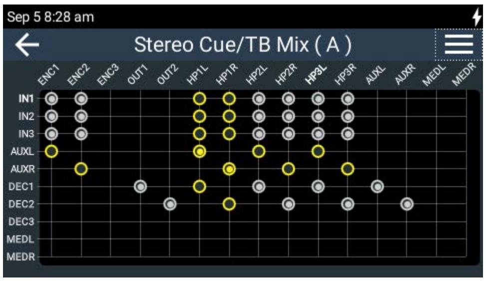


Auxiliary Input Cue

The codec supports offline cueing of external analog and digital sources attached to the stereo auxiliary input. By default, offline cue monitoring of the auxiliary input is only available via headphone output 1 (**HP1**). This routing can be adjusted as required.



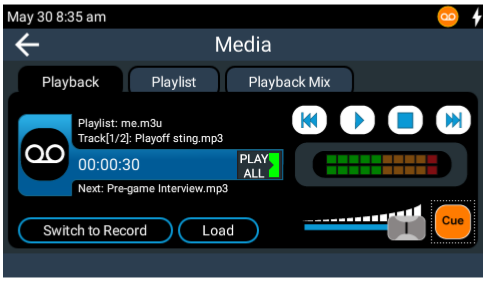
Mono Program Aux Cue Defaults



Stereo Program Aux Cue Defaults

Playback Cue

The **Playback Cue** matrix allows configuration of where offline audio is routed when the **Cue button** is enabled on the **Playback** screen. To enter offline cue mode for file playback select **HOME** > **Media** > **Playback** and then tap the **Cue** button on the **Playback** screen. An orange **Cue symbol** is displayed in the **Status Bar** at the top of the **TOUCH SCREEN** to notify the user that offline cue monitoring is enabled.



To adjust settings:

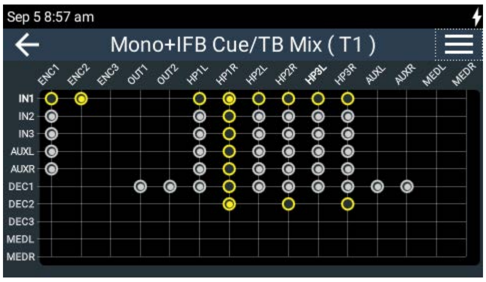
1. Press the **HOME** button to return to the **Home** screen.
2. Tap **Audio** > **Matrix Editor** > **Menu** > **Cue/TB Edit > Playback Cue > Edit.**

Alternatively, enable playback cue in the **Playback screen** and tap **HOME** > **Audio** > **Matrix Editor** to edit the highlighted **Playback Cue** matrix settings.

Understanding Talkback Mode

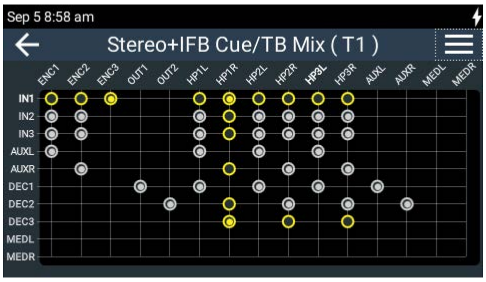
When you load a **Mono Peer-to-Peer + IFB** program, or a **Stereo Peer-to-Peer + IFB** program, two independent audio streams are configured to stream program and communications audio separately. When an announcer presses an input's **TB CUE** button, audio is routed to the IFB audio stream encoder and outgoing audio is simultaneously monitored in the right side of the selected input's headphone output. Return IFB audio is also audible in the right side of the headphones. In other words, only the announcer using the **CUE TB** button hears incoming and outgoing communications audio.

In the following matrix routing example a **Mono Peer-to-Peer + IFB** program is loaded in the codec. When the **CUE TB** button for input 1 is pressed, outgoing talkback audio is sent to encoder 2 (**ENC2**) and it is monitored in the right side of **HP 1**. Return IFB from the studio is also audible in the right side of **HP 1**.



Default Mono + IFB TB Routing

In the next matrix routing example a **Stereo Peer-to-Peer + IFB** program is loaded in the codec. When the **CUE TB** button for input 1 is pressed, outgoing talkback audio is sent to encoder 3 (**ENC3**) and it is monitored in the right side of **HP 1**. Return IFB from the studio is also audible in the right side of **HP 1**.



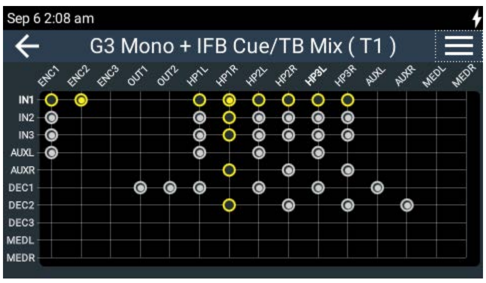
Default Stereo + IFB TB Routing

Using G3 Mono Peer-to-Peer + IFB Mode with a Stereo Connection in ViA

It is also possible to connect a stereo program and use one channel for talkback communications. This is preconfigured in ViA if you create a program using the **G3 Mono Peer-to-Peer + IFB** program template.

Important Note: This program template will only work if you are using Tieline Music or Music PLUS algorithms.

The following image displays the talkback routing matrix defaults for a **G3 Mono Peer-to-Peer + IFB** program. When the **CUE TB** button is pressed on inputs 1-3 the codec will deliver discreet talkback between local announcers and a remote codec using encoder 2 (**ENC2**). Incoming IFB audio is monitored in the right channel of the headphones by default.



22 Analog and Digital Outputs

The codec has two XLR outputs and also supports S/PDIF or micro USB digital outputs. The **XLR Output Settings** and the **Digital Output Settings** screens support:

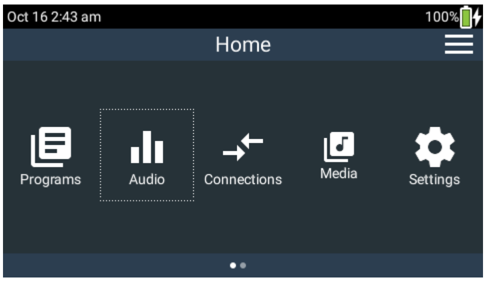
- Selection of input sources to route to outputs.
- Output mute and output level controls.
- Adjustment of the Send/Return balance between input sources and decoder audio sent to the outputs.

Important Note: Analog audio can always be fed from the **XLR OUTPUTS**. However, to feed audio from either the digital **USB AUDIO I/O** (via micro USB Type-B) or **S/PDIF OUT** it is necessary to configure the auxiliary input to be the same as the output you are using. E.g. From the **Home** screen tap **Audio > Inputs** **> Aux Input > Settings symbol** **> Input Type > USB** on the aux input when using the micro USB output. Note: It is not possible to feed audio from both the **USB AUDIO I/O** or **S/PDIF OUT** simultaneously.

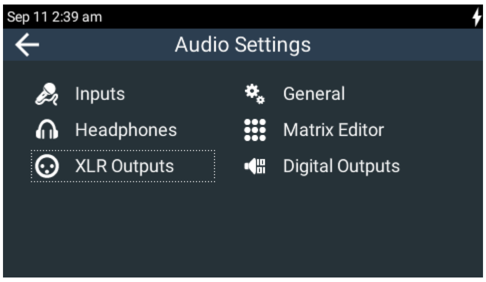
XLR Output Controls

It is simple to control input and decoder audio routed to the XLR outputs.

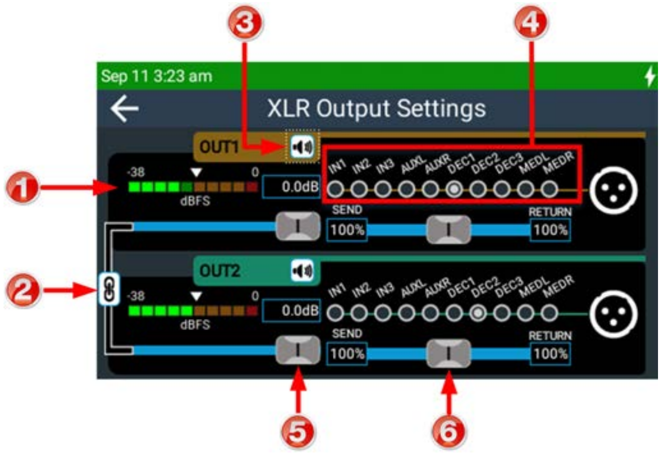
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** .



2. Tap **XLR Outputs** .



3. The **XLR Output Settings** screen is displayed with several controls

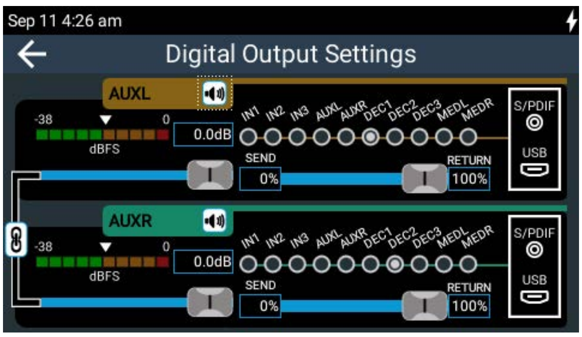


	Feature	Description
1	**PPM**	Output level PPM meter in dBFS.
2	**Gang** button	Tap to select/deselect ganging of the output sliders and the **Input Mute** buttons.
3	**Output mute** button	Mute button is available for each output; this function can be ganged using the **Gang** button.
4	Output matrix	Matrix on each output used to select the sources routed to each output. Input sources and decoders can be selected.
5	Output slider	Slider on each output to adjust output levels.
6	Output **Send/Return** balance slider	Send/Return balance slider on each output adjusts the balance between input sources and decoder audio.

Digital Outputs

The **Digital Output Settings screen** operates in the same way as the **XLR Output Settings** screen. To route input and other audio sources to the USB or S/PDIF digital outputs, and control output muting etc., it is necessary to select the sources in the **Digital Output Settings** screen. To open the screen:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio**.
2. Tap **Digital Outputs**.



3. Tap to select and deselect sources to adjust audio routed from the digital outputs. Note: Adjust the **Send** and **Return** balance slider to adjust the balance between the mix of incoming and outgoing audio sources.

AGC

By default, automatic gain control limiting (AGC) is enabled on all outputs. AGC is independent of Intelligent Gain Control (IGC) which can be enabled on each codec input. The codec also offers adjustable dynamic range compression on each input. For more information about IGC and compression see Input Levels and Input Settings. To enable and disable AGC see General Audio Settings.

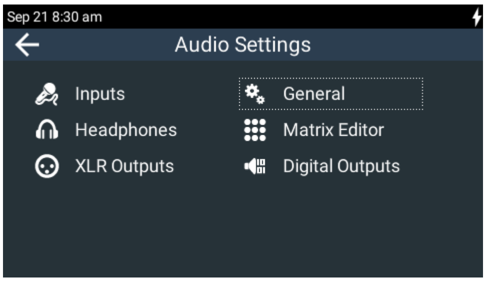
23 General Audio Settings

General audio settings include the following options:

- Audio reference levels.
- Phantom power voltage settings for inputs 1-3.
- Disabling +48V Phantom Power on boot.
- Show/hide +48V High Voltage warning dialog.
- Analog Input Level Scale adjustment.
- Disable AES42 Mic Input on boot.
- Show AES42 Mic Input warning dialog.
- Output AGC enable/disable.
- Swap Cue and Input On/Off Buttons.
- Loopback audio test.

To adjust these settings:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio**.
2. Tap **General**.



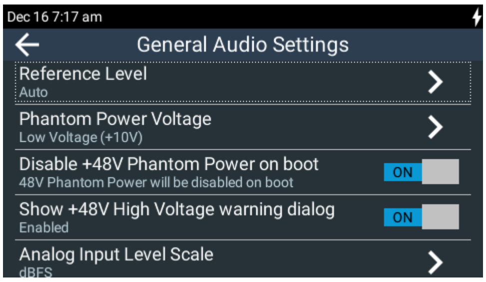
Audio Reference Levels

By default, codec the **PPM METERS** on the front of the codec, on the **TOUCH SCREEN**, or the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels. The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** reference scale setting.

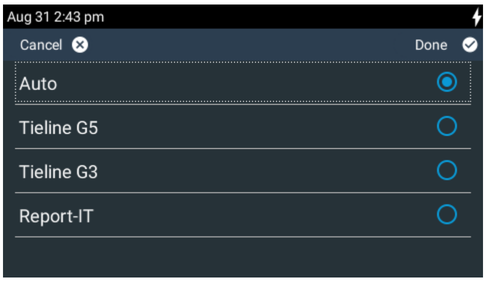
The default Tieline G5 audio reference scale displayed on the PPMs when you connect to a Tieline G5 codec is -38dBFS to 0dBFS (e.g. Merlin, Genie, Bridge-IT and ViA codec families). Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 18dB of headroom from the nominal 0vu point. The comparison table below outlines the reference scales for G5 and G3 codecs and Report-IT in dBFS, as well as the equivalent dBU scale.

	Reference Setting	Description	dBu	dBFS
1	Tieline G5	PPM meter low point	-16dBu	-38dBFS
		Nominal 0vu reference level	+4dBu	-18dBFS
		Level at which audio will clip/distort	+22dBu	0dBFS
2	Tieline G3	PPM meter low point	-11dBu	-29dBFS
		Nominal 0vu reference level	+4dBu	-14dBFS
		Level at which audio will clip/distort	+18dBu	0dBFS
3	Report-IT	PPM meter low point	-9dBu	-23dBFS
		Nominal 0vu reference level	+4dBu	-10dBFS
		Level at which audio will clip/distort	+14dBu	0dBFS

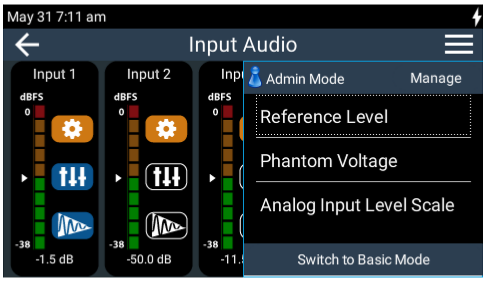
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** **> General** **> Reference Level**.



2. Tap to select the appropriate reference scale to suit the codec to which you are connecting.



Helpful Hint: As a shortcut you can tap the **Hamburger** menu in the **Input Audio** screen to adjust this setting.



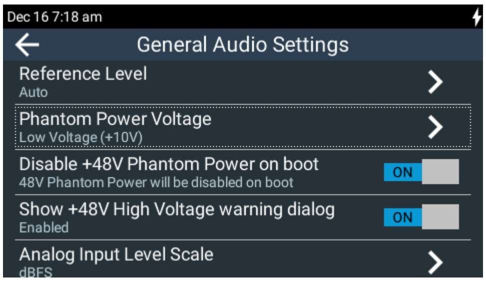
Important Note: When dialing a multistream program to both G3 and G5 codecs, by default the Audio Reference Level will be configured for the compatibility of the codec that connects first. I.e. if you dial a G3 codec first then the G3 Audio Reference Level will be configured for all connections.

Phantom Power Voltage

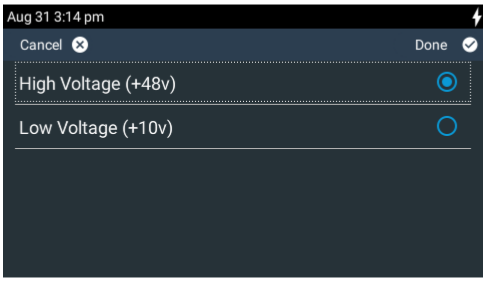
VOLTAGE WARNING: Microphones and other ancillary audio equipment may be very sensitive to phantom power voltages and should not be attached to the codec inputs when phantom power is enabled, or equipment may be damaged these voltages.

The phantom power voltage setting applies across inputs 1-3. However, it is possible to enable or disable phantom power on individual inputs.

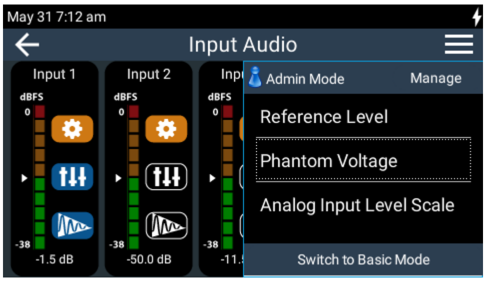
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **General** **> Phantom Power Voltage**.



2. Tap to select the preferred voltage setting.



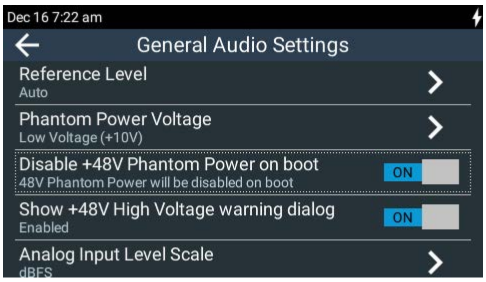
Helpful Hint: As a shortcut you can tap the **Hamburger** menu in the **Input Audio** screen to adjust this setting.



Disable +48V Phantom Power on Boot

By default, 48V phantom power is disabled when the codec is rebooted. This protects users from attaching a non-phantom device to a phantom-enabled codec input and damaging equipment. This feature can be disabled if required:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **General > Disable +48V Phantom Power on Boot**.



2. Tap the **On/Off** button to enable and disable this feature.

Show +48V High Voltage Warning Dialog

By default, a 48V phantom power warning dialog is displayed when phantom power is enabled in the codec. This feature can be disabled if required:

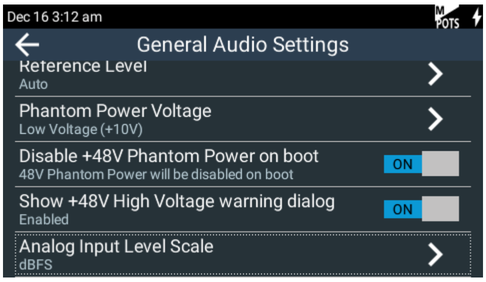
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **General** **> Show +48V High Voltage warning dialog**.

2. Tap the **On/Off** button to enable and disable this feature.

Analog Input Level Scale

It is possible to change the input PPM meter scale used by the codec from dBFS (default) to dBU. This is most useful when viewing PPM meters in the **Inputs panel** in the Toolbox web-GUI. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **General** **> Analog Input Level Scale**.



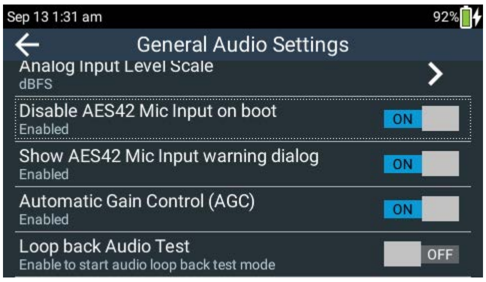
2. Tap **Analog Input Level Scale** to toggle between selecting dBFS and dBU input PPM scales.

Disable AES42 Mic Input on Boot

⚠ **VOLTAGE WARNING:** DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** input mode is selected, or equipment may be damaged by high voltages.

By default, the **AES42** mic **Input Type** option on input 1 is disabled when the codec is rebooted. This protects users from attaching a non-AES42 device to input 1 and damaging it. This feature can be disabled if required:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **General** **> Disable AES42 Mic Input on boot**.



2. Tap the **On/Off** button to enable and disable this feature.

Show AES42 Mic Input Warning Dialog

By default, an AES42 warning dialog is displayed when **AES42** is selected as the **Input Type** on input 1. This feature can be disabled if required:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **General** **> Show AES42 Mic Input warning dialog**.

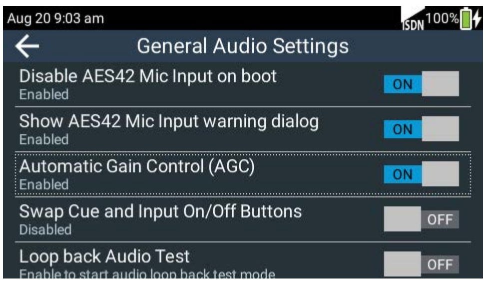
2. Tap the **On/Off** button to enable and disable this feature.

Output AGC (Automatic Gain Control)

AGC is enabled by default on all outputs. AGC is independent of IGC (Intelligent Gain Control) on each input.

AGC is triggered if mixed audio signals approach -2dBFS.This is to prevent hard clipping, which would introduce excessive distortion when the summed input signals exceed the full-scale range. AGC has a very fast reaction time and will decrease high audio levels by approximately 1dB every few milliseconds, up to a maximum of 34 dB of attenuation. AGC tries to return levels to normal once the output dips below -7dBFs. This will gradually increase the levels by approximately 1dB after a predetermined time-frame measured in milliseconds. To enable or disable AGC:

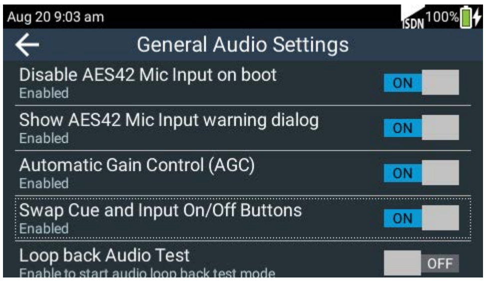
1. Press the **HOME** button to return to the **Home** screen, then tap **Audio**.
2. Tap **General** .
3. Tap the **On/Off** button to toggle **AGC** on and off.



Swap Cue and Input On/Off Buttons

Some broadcasters prefer **TB/CUE** button operation to be available using the **ON/OFF** button for each input.

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** .
2. Tap **General** .
3. Tap to enable **Swap Cue and Input On/Off Buttons**.
4. In this mode the **TB/CUE** buttons on inputs 1-3 are illuminated. When an input **ON/OFF** button is pressed the button illuminates red, and the **TB/CUE** button is not illuminated, to indicate **TB/CUE** mode is in use.



Loopback Audio Test

Loopback Audio Test mode is used by the codec to perform an input/output loopback test of audio. E.g. **Input 1** is routed to analog **Output 1**, **Input 2** is routed to **Output 2**. The test remains active until it is disabled, or a connection is dialed or answered, or a new program is loaded.

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** .
2. Tap **General** .
3. Tap the **On/Off** button to toggle the **Loopback Audio Test** setting on and off.

The following table outlines the default matrix settings for how inputs are routed to analog outputs in loopback audio test mode.

Inputs	Analog Output 1	Analog Output 2
1	✓	
2		✓
3	✓	✓
Aux In Left	✓	
Aux in Right		✓

24 Configuring AES3 and AES42 Input Audio

INPUT 1 (**MIC/LINE/AES**) on the codec supports:

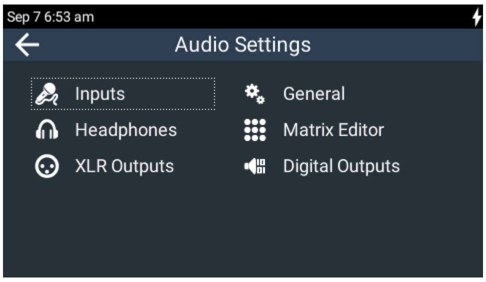
- A mic or line level analog source, or
- AES3 (AES/EBU) format digital audio, or
- An AES42 digital microphone.

VOLTAGE WARNING: DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** input mode is selected, or equipment may be damaged by high voltages.

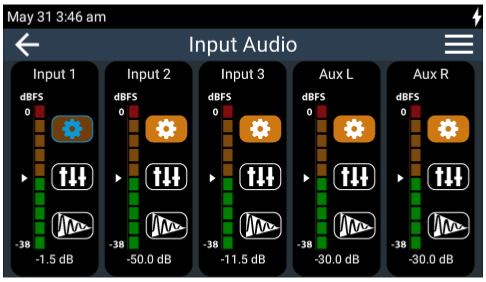
AES3 Input Source

Both **INPUT 1** and **INPUT 2** are configured as AES inputs when the **Input Type** for either input is configured as **AES3**. The audio level for both inputs is controlled using **INPUT 1**. AES operates effectively over distances of up to 100 meters. The sample rate converter in the codec converts the AES input to the codec's sample clock rate.

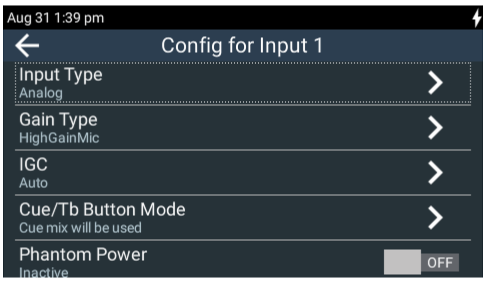
1. Press the **HOME** button to return to the **Home** screen and tap **Audio**.
2. Tap **Inputs** to adjust audio input configuration settings.



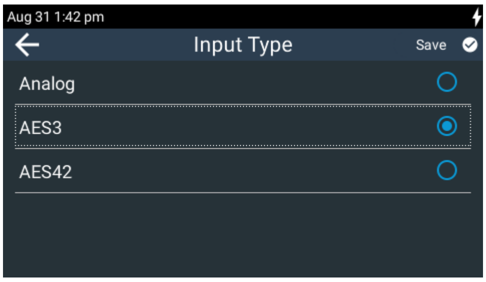
3. Tap the **Settings symbol** on **INPUT 1** or **INPUT 2**.



4. Tap **Input Type**.



5. Tap **AES3**.



6. **AES3** settings are confirmed in the **Input Type** field on the screen.

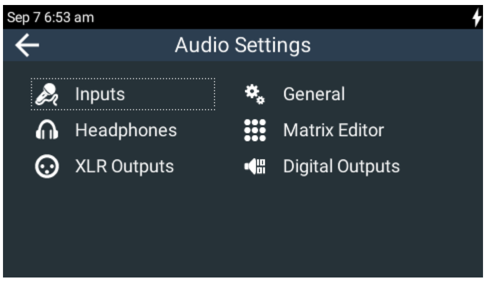
Important Note: There is a maximum of 6dB of additional gain available when adjusting an AES input source.

AES42 Digital Microphone Input

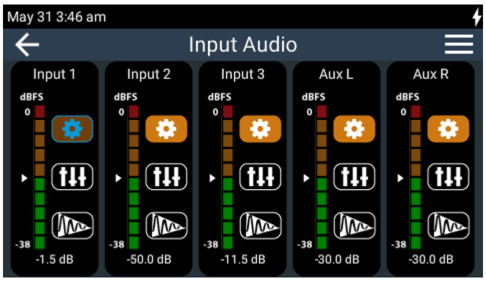
VOLTAGE WARNING: DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** input mode is selected, or equipment may be damaged by high voltages. As a safety measure, by default the codec will reboot and configure input 1 for an analog microphone after an AES42 mic has been used. This default setting and AES42 warning settings can be adjusted. See General Audio Settings for more information.

Select **AES42** as the **Input Type** for **INPUT 1** to attach an AES42 digital microphone. The codec supports mode 1 synchronization and if the mic sample rate is asynchronous to the codec sample rate, a sample rate converter is used to convert between them. In AES42 mode **INPUT 1** supplies the 10V power required to operate mode 1 digital microphones. It also supports the remote control of features such as pre-attenuation and low-cut filters in AES42 microphones.

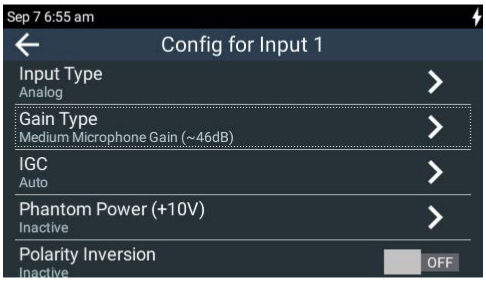
1. Press the **HOME** button to return to the **Home** screen and tap **Audio**.
2. Tap **Inputs** to adjust audio input configuration settings.



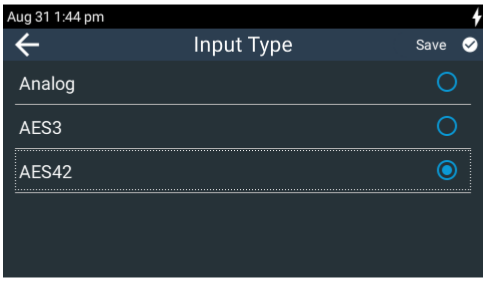
3. Tap the **Settings symbol** on **INPUT 1**.



4. Tap **Input Type**.

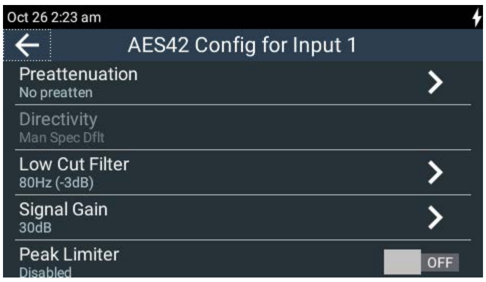


5. Tap **AES42.**



6. **AES42** settings are confirmed in the **Input Type** field on the screen and an **AES42 Settings** menu appears.

7. Tap **AE42 Settings** to adjust mic parameters as required.



25 Backup and Redundancy Options

Fuse-IP and SmartStream PLUS

Tieline's proprietary Fuse-IP data aggregation technology lets you bond any IP interfaces you choose. Imagine the peace of mind knowing you can bond two cellular data links from different Telcos and let Tieline's Fuse-IP technology automatically manage the data capability of each link! Flexibility is paramount for remotes, so you can even bond a USB modem with a Wi-Fi connection, or bond two Ethernet connections.

ViA also includes Tieline's SmartStream PLUS redundant streaming software, which has set the benchmark for redundant IP streaming over the public internet. Tieline codecs feature highly advanced backup and redundancy options to maintain reliable audio codec streaming. These include the options outlined in the following table:

Tieline Audio Codec Backup Features			
Backup Option	**Transport: IP, ISDN or POTS**	**Time Required to Respond**	**How to Enable**
SmartStream PLUS	IP Only (Note: redundant packet stream sent; codec detects IP packet loss or delayed packets)	No time delay - simultaneous redundant streaming	Enabled in dialing codec program; configures local decoding, or remote decoding via session data
Fuse-IP	IP Only: multiple IP interfaces are bonded and data is shared across interfaces	No time delay if interface is lost; immediately adjusts to existing network data capacity	Configure multiple IP interfaces and create a Fuse-IP "Tunnel" prior to dialing destination codec
On-demand (cold) Failover	All transports (Note: codec detects loss of data or connection and redials the backup connection)	User configurable detection parameters during program configuration*. Delay is equal to detection time plus the time required to dial the alternative connection	Dialing codec program monitors streaming and manages failover
FEC (Tieline FEC or RFC compliant FEC)	Decoding codec detects IP packet loss or delayed packets. Note: Only Tieline Music and Music PLUS can be used for Tieline FEC.	No failover time delay as packet replacement occurs in real-time. Note: RFC compliant FEC can be configured to send 100% FEC at a specified delay if desired.	• Dialing codec using Tieline session data configures local and remote FEC settings via session data transfer when connecting, or • In sessionless mode FEC configuration is as per RFC compliant settings configured in the local and remote codecs.
Auto Reconnect	All transports; codec will redial continuously to try and reconnect	Immediately redials after loss of IP stream detected	Enabled in dialing codec program

* Note: POTS can take up to 60 seconds to connect successfully.

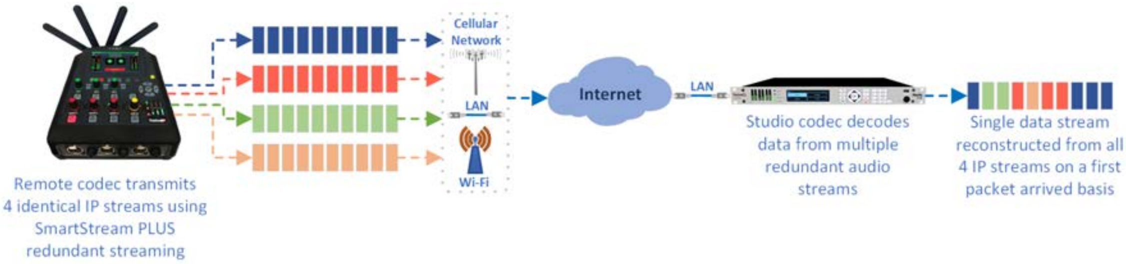
Important Note: Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.

SmartStream PLUS Redundant IP Streaming

Tieline's proprietary SmartStream PLUS IP technology ensures you're always on the air. The codec features up to 7 different IP options for redundant streaming. There are three levels to SmartStream PLUS IP streaming.

1. The codec can stream simultaneous redundant data streams over separate IP interfaces and deliver seamless redundancy by switching back and forth, without loss of audio, from the nominated primary data link to the backup link if one fails and then subsequently recovers. Use IP links from different IP network providers for optimal redundancy over mission critical connections.
2. Second, when multiple redundant audio streams are sent, the decoding codec automatically reconstructs audio into a single stream on a first packet arrived basis, to minimize program latency and ensure audio integrity.
3. Third, SmartStream PLUS features automated jitter buffer management and Forward Error Correction (FEC) and these advanced network management tools deliver uncompromising audio quality, while dynamically responding to variable conditions over unmanaged IP networks like the internet.

These combined measures ensure Tieline is capable of offering a rock solid IP audio solution for broadcasting IP audio economically and efficiently across broadcast networks. See the procedures for configuring different programs using the web-GUI for more configuration details.



On-Demand Failover (IP, ISDN or POTS)

On-demand failover requires configuration of a primary connection and an on-demand 'cold' backup connection. On-demand failover is activated when the dialing codec program detects the loss of the primary connection, or if audio streaming ceases. The backup connection is then dialed to replace the primary connection.

The codec can be configured to switch to a backup connection over IP, ISDN or POTS as required. For example, you can create a program with IP as the primary connection and also create a backup ISDN connection, or a backup POTS connection in the same program. Note: ViA has a single module slot and can connect over POTS or ISDN if a ViA POTS or ViA ISDN module is installed. For details on configuring backup connections using failover see Configuring Mono or Stereo Peer-to-Peer Programs.

Forward Error Correction (FEC)

FEC transmits a secondary stream of audio data packets over a single connection. If packets are lost or corrupted over the connection then replacement FEC data packets can be substituted to replace them.

Note: FEC should not be confused with SmartStream PLUS. FEC packets are sent over a single data stream connection, whereas SmartStream PLUS redundant streaming transmits completely redundant audio data streams. FEC is also a subset of features within SmartStream PLUS, which means you can configure SmartStream PLUS redundant data streams and also configure FEC on each of these data streams. For more info on FEC see Configuring Forward Error Correction.

Auto Reconnect

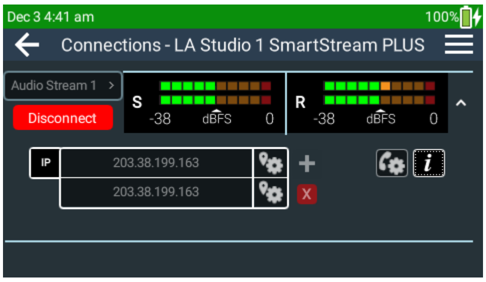
Auto Reconnect is the simplest form of connection backup whereby the codec will redial a lost connection continuously until it is either:

- Re-established, or
- Dialing is manually stopped.

Auto reconnect can be enabled when configuring a codec program designed to dial another codec or codecs. See the procedures for configuring different programs using the web-GUI for more configuration details.

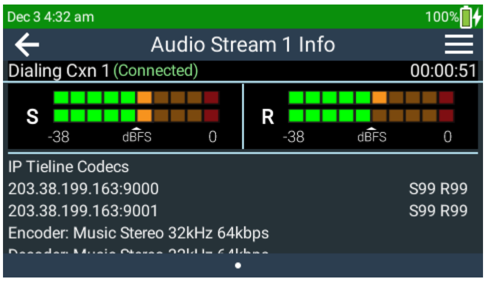
25.1 Monitoring SmartStream PLUS

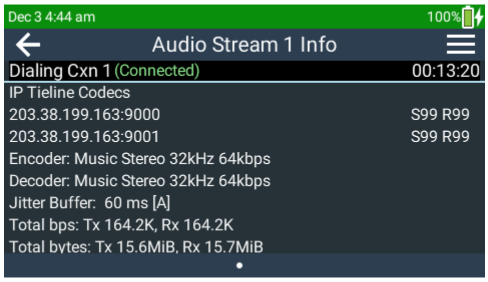
When you configure a SmartStream PLUS connection it appears like a regular stereo connection on the **Connections Screen**. The additional audio stream connection below the primary audio stream is for a single SmartStream PLUS connection. Up to 3 SmartStream PLUS connections may be configured, depending on the algorithm selected.



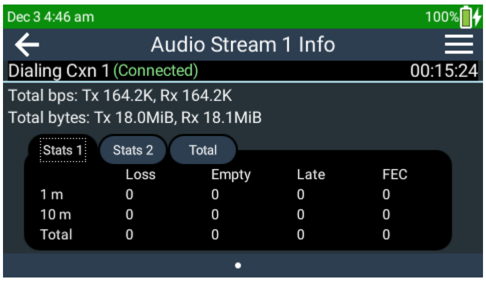
Important Notes:
- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

Tap the **Info** button and scroll down to view stream info as well as **Jitter** and **Send/Return** values and packet statistics.





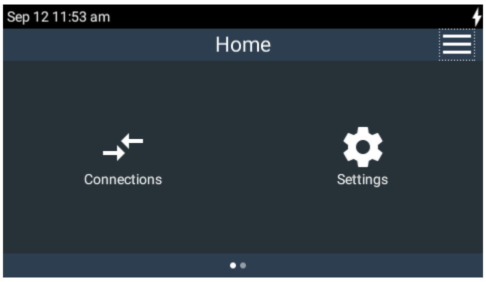
The **Stats 1** tab displays IP packet statistics for the primary audio stream. The **Stats 2** tab displays IP packet statistics for the SmartStream PLUS redundant audio stream. The **Total** tab shows the lost and empty packets for the aggregated audio stream, i.e. the audio stream that is made up using data from the primary and redundant audio stream. Packet losses or empty events in this audio stream may result in audible audio artifacts.



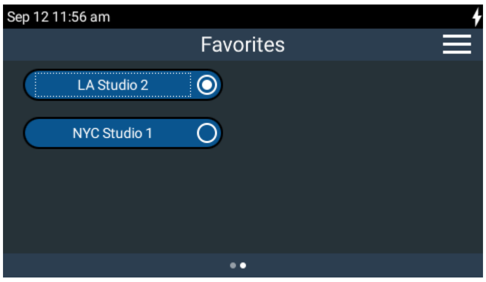
Note: Incrementally renegotiate higher connection bit rates on both connections by pressing the **F2** button and then the **NAVIGATE UP** ▲ button while viewing the **Audio Stream Info** screen; for lower bit rates press the **F2** button and then the **NAVIGATE DOWN** ▼ button.

26 Administrator and Basic Operation Modes

The codec can be operated in either **Admin Mode** or **Basic Mode**. When shipped the codec is configured in **Admin Mode** with access to all features and menus. An administrator can also configure the codec to present a subset of menus for non-technical users in **Basic Mode** to simplify operation. As an example, the **Dialer, Favorites** and **Audio** menus can be removed from the **Home** screen, as displayed in the following image.

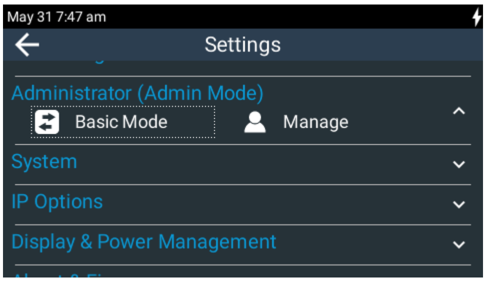


The default start-up screen can also be configured to make it easier for non-technical users to get connected. In the following example, an administrator has configured **Favorites** as the default screen to make it easier to connect using programs with preconfigured settings. See [Load, Connect and Manage Programs] for more details on configuring **Favorites**.



Viewing Admin/User Mode Status

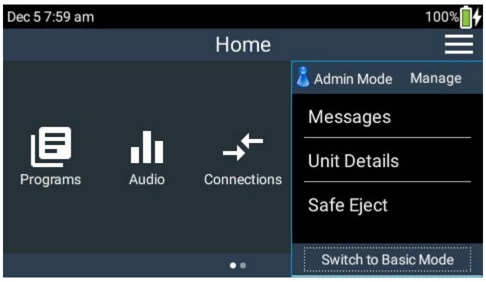
Navigate to the **Home** screen **> Settings** **> Administrator** to verify and manage the currently configured mode. The current mode is displayed in brackets next to the **Administrator** menu.



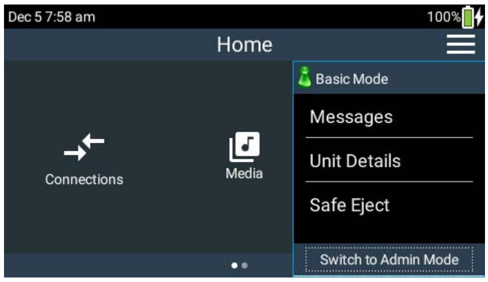
Admin Mode Active

Basic Mode Active

It is also possible to verify and manage the currently configured mode by tapping **Menu** from the **Home** screen and some other screens like the **Input Audio** screen.



Admin Mode Active

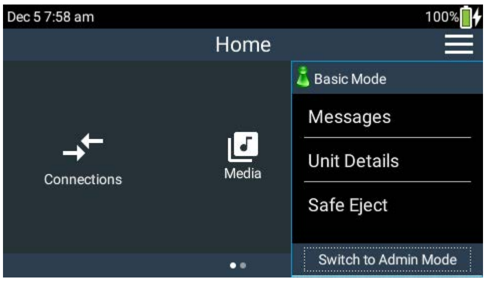


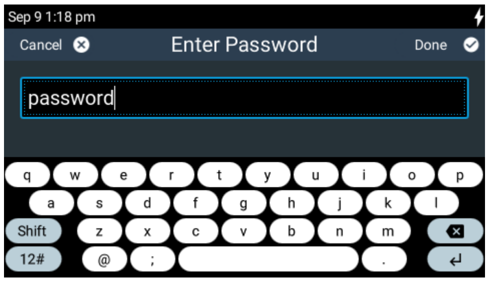
Basic Mode Active

Admin Mode Login

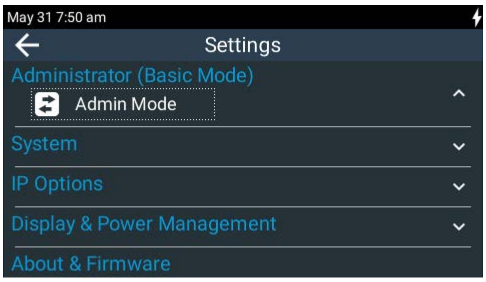
There are two ways to log in as an administrator when the codec is in **Basic Mode**:

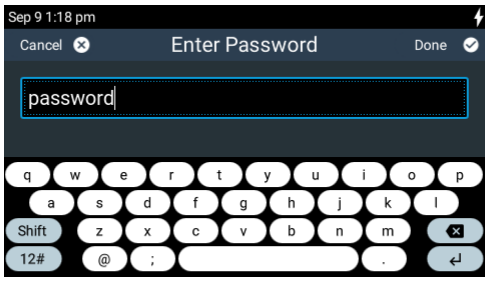
1. Tap **Menu** on the **Home** screen and then tap **Switch to Admin Mode** to log in as an administrator. To login for the first time enter the default password "**password**", then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.





2. From the **Home** screen tap **Settings** **> Administrator > Admin Mode** and enter the password, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



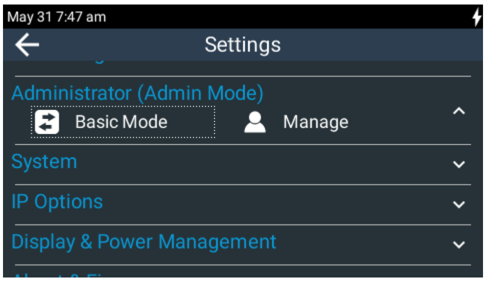


Important Note: The **Admin Mode** password is separate to the Web-GUI password, which is used to log into the codec using the HTML5 Toolbox Web-GUI. To adjust the Web-GUI password press the **HOME** button to return to the **Home** screen and tap **Settings** **> System > Web GUI** . See Configuring Web-GUI Settings.

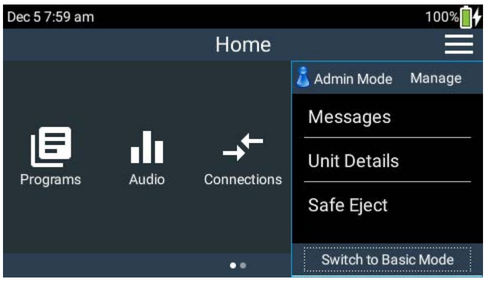
Switch from Admin Mode to Basic Mode

There are two ways to change from **Admin Mode** to **Basic Mode**.

1. Navigate to **Home** screen **> Settings** **> Administrator**, then tap **Basic Mode**.



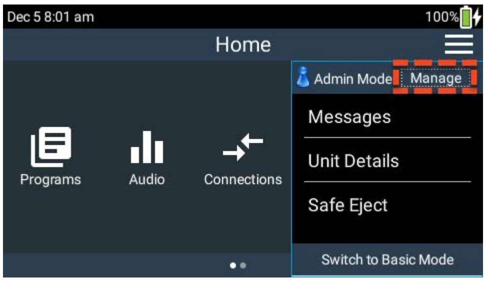
2. Tap **Menu** on the **Home** screen and then tap **Switch to Basic Mode**.

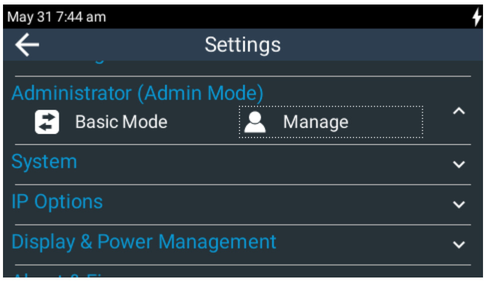


Configure Basic Mode Settings and Passwords

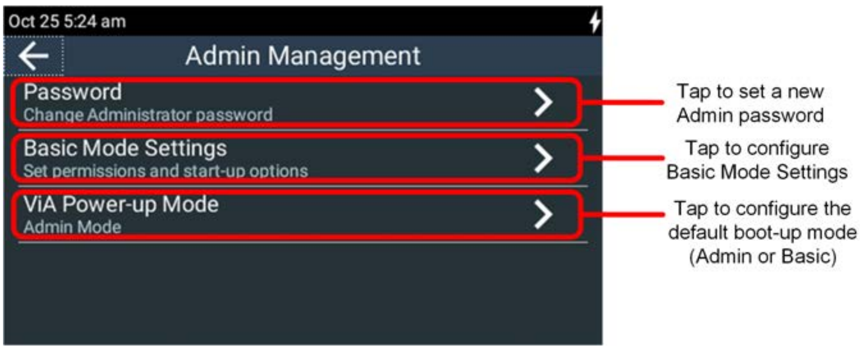
An administrator can adjust the subset of settings visible to a user when the codec is in **Basic Mode**. To adjust these settings and the login **Password**:

1. Log in as an administrator and tap **Manage** via the **Menu** button, or navigate to **Home** screen **> Settings** **> Administrator > Manage**.



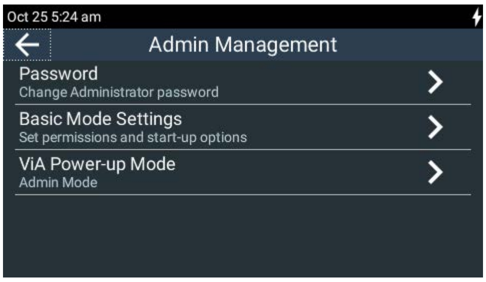


2. Administrator management options are presented.

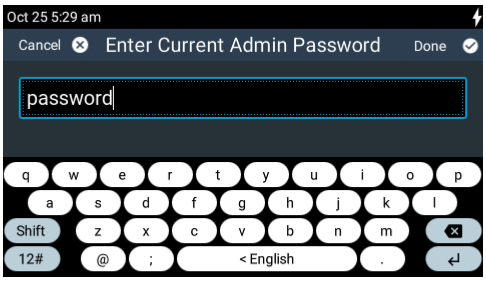


Change the Administrator Password

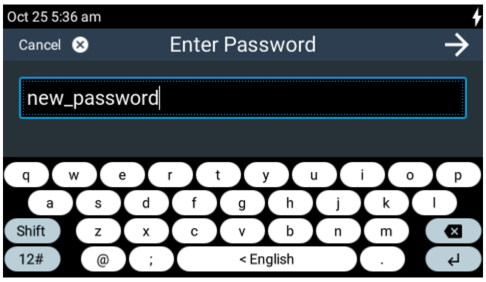
1. Tap **Password**.



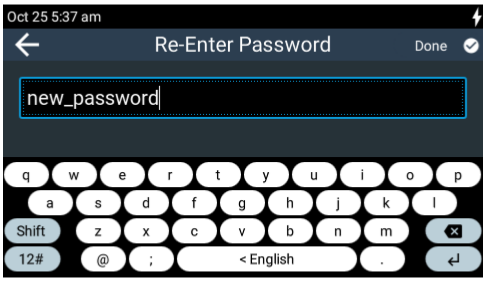
2. Enter the current admin password, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



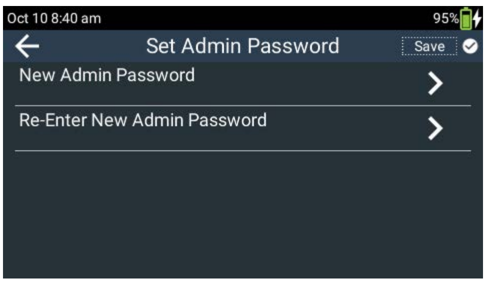
3. Tap **New Admin Password** to enter the new password, then tap the **Right Arrow** in the top right-hand corner of the **TOUCH SCREEN**.



4. Re-enter the new password to confirm it, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

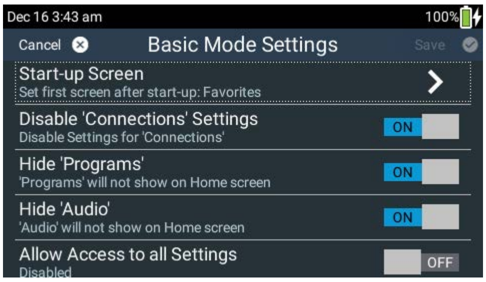


5. Tap **Save** in the right-hand corner of the **TOUCH SCREEN** to confirm the changed password.

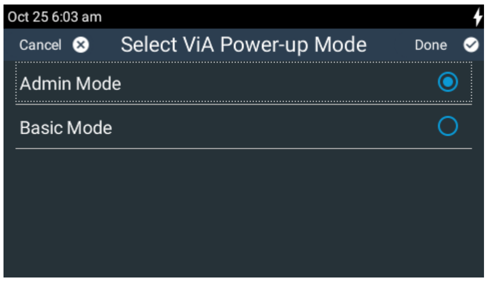


Configure Basic Mode Settings

Tap **Basic Mode Settings** to configure which menus and screens are visible in the codec in **Basic Mode**.



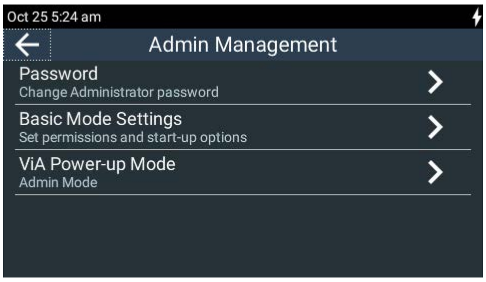
Basic Mode Settings screen

Basic Mode Setting	Explanation
Start-up Screen (default setting **Favorites**)	Tap to select the default screen displayed on start-up. The default setting displays the **Favorites** screen. This allows an administrator to create programs, add them as favorites, and then display them automatically on start-up. This makes it easy for users to load and connect different programs.  Other options allow you to select the **Home**, **Dialer** or **Connections** screen as the default on start-up.
Disable 'Connections' Settings (default setting **Off**)	Tap to toggle the **On/Off** button for **Disable 'Connections' Settings** to enable/disable this feature. When disabled, most settings in the **Connections screen** cannot be edited. IP addresses can still be edited.
Hide 'Programs' (default setting **Off**)	Tap to toggle the **On/Off** button for **Hide 'Programs'** to enable/disable this feature. **Programs** is not displayed on the **Home** screen when this feature is enabled in **Basic Mode**.
Hide 'Audio' (default setting **Off**)	Tap to toggle the **On/Off** button for **Hide 'Audio'** to enable/disable this feature. **Audio** is not displayed on the **Home** screen when this feature is enabled in **Basic Mode**.
Allow access to all Settings (default setting **On**)	Tap to toggle the **On/Off** button for **Allow Access to all Settings** to enable/disable this feature. If enabled, a user can change any setting under the **Settings** menu, e.g. LAN, Wi-Fi, etc. If disabled, in Basic Mode many items are not shown, e.g. **System** menus. Some other menu items are read only, e.g. Wi-Fi access point information.

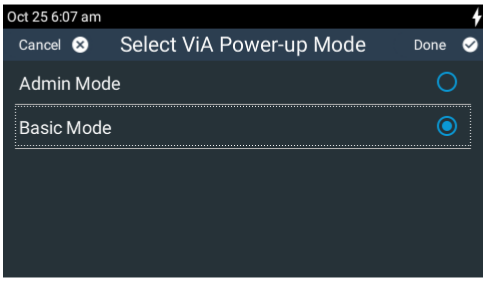
Configure ViA Power-up Mode

The codec can be configured to always boot in either **Admin Mode** or **Basic Mode**.

1. Tap **ViA Power-up Mode**.



2. Tap to select the preferred option.

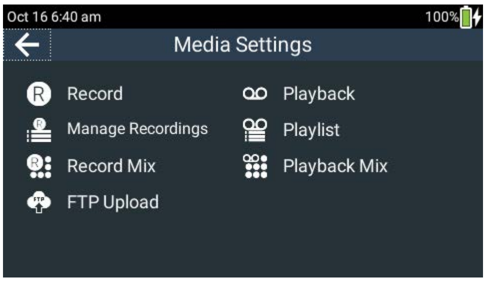


27 Record, Playback and FTP

The codec supports recording and playing back audio files, including the ability to use the **TOUCH SCREEN** to:

- Select and record any input, return audio and file playback audio.
- View and manage recordings.
- Create playlists of local recordings and imported files.
- Control playback routing to specified codec encoders, as well as analog and digital outputs or record media.
- Use offline playback cue monitoring.
- Record entire programs ready to upload as podcasts.
- Upload audio files using FTP.

To access record and playback functionality, press the **HOME** button to return to the **Home** screen, then tap **Media** to view record and playback options.



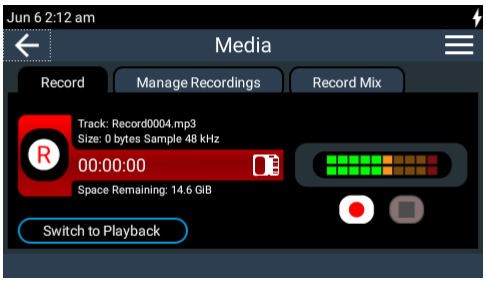
Record and Playback Options Screen

27.1 Record and Manage Audio Files

Checklist Before Recording

Before you start recording:

1. Insert a FAT-formatted SDHC card into the **SD CARD SLOT** on the rear panel of the codec.
2. Verify enough storage space is available to support the length of time you will be recording. Note: Storage space remaining is displayed on the **Record screen**, or tap **Menu** ☰ on the **Record screen** and then tap **Storage**.
3. Ensure PPM meter levels on the **Record screen** are not over-modulating or recordings may be distorted. Note: Ensure recordings are not recorded at very low audio levels or there may not be enough gain on playback. Aim for audio peaks to average around the first orange PPM indications on the screen, as displayed in the following image.



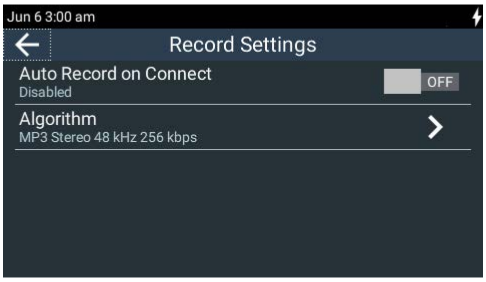
Important Notes:
- The ViA codec supports SDHC cards which have a physical capacity of up to 32GB.
- The codec supports using SD cards with a write speed of 2MB/second or higher.
- The maximum file size that can be recorded is 2GB. Note: Recording will stop when the file size reaches 2GB.

- MP2, MP3 and WAV (PCM) files are supported.
- Ensure MP3 recordings are not variable bit rate files.
- When creating audio files please ensure that audio levels match the audio scale used by the codec and peaks average at the correct levels.

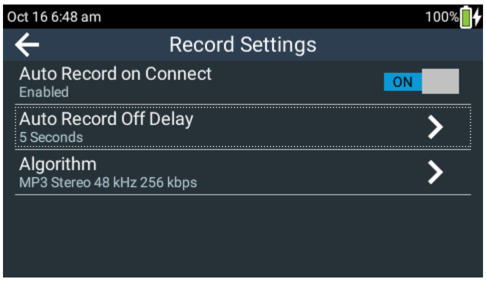
Adjust Record Settings

1. Navigate to **HOME** > **Media** > **Record** .
2. Next tap **Menu** on the **Record screen** and then tap **Record Settings**.

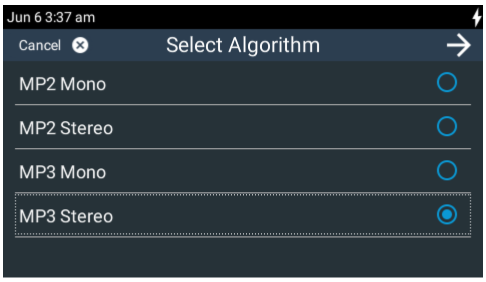
3. Tap to select the options available and adjust record settings.



4. If **Auto Record on Connect** is enabled, it is possible to adjust the **Auto Record Off Delay** from the default setting of 5 seconds. This setting allows a recording to continue for a certain period of time in the event of a disconnection, to allow for a codec reconnecting after a temporary disconnection during a broadcast. This setting has a time limit of 9,999 seconds.

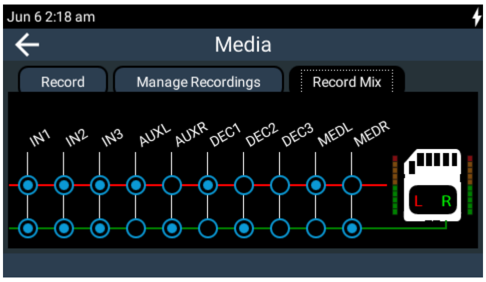


5. Tap **Algorithm** to select a recording format and whether a mono or stereo recording is preferred. Note: Default setting is **MP3 Stereo**.

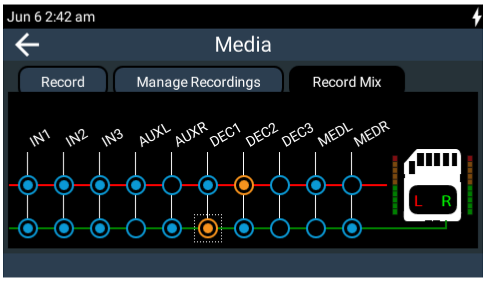


Adjusting the Record Mix

From the **Record screen** tap **Record Mix** and use the **TOUCH SCREEN** to edit the sources from which audio can be recorded. For example, this lets you record individual input sources, or create split track recordings as required. It is also possible to record file playback audio simultaneously while recording, and incoming decoder audio, which means you can record incoming and outgoing audio for podcasts or similar applications. Edits to the default settings are displayed in orange.



Default Stereo Record Mix

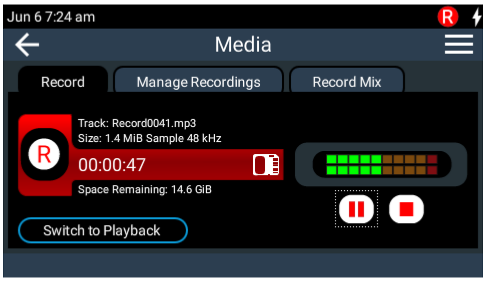


Record Mix with Edits

Recording a File

To start a new recording:

1. From the **Record screen** tap the **Record Start** button and recording will commence. The **Recording symbol** will appear and flash in the **Status Bar** and recording details update dynamically.



2. Tap the **Record-Pause** button to pause a recording. The **Paused symbol** will appear in the **Status Bar**.

3. Tap the **Record-Paused** button to recommence recording, or press the **Stop** button to end recording.
4. Tap **Manage Recordings** to manage and lock the recording, or tap **Switch to Playback** to:
   - Load and listen to the file.
   - Adjust the **Playback Mix**.
   - Manage a playlist of tracks.

See Playing Audio Files for more details on playback features.

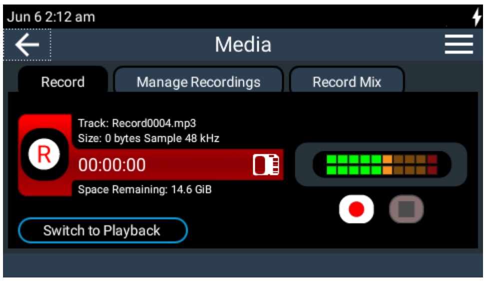
Managing Recordings

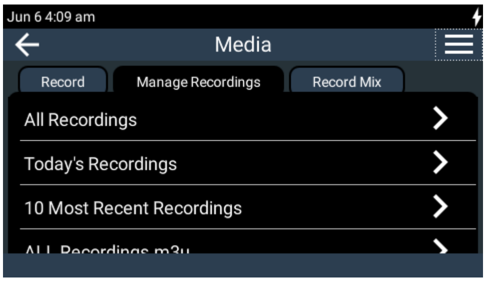
Manage Recordings can be used to:

- View an individual recording's settings (e.g. record format and bit rate), rename or lock a recording, or add metadata such as the city or description.
- Delete recordings.
- Create, rename or delete a Playlist.

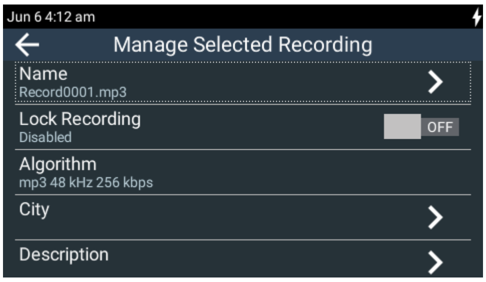
Managing Individual Recording Settings

1. From the **Record screen** tap **Manage Recordings**.





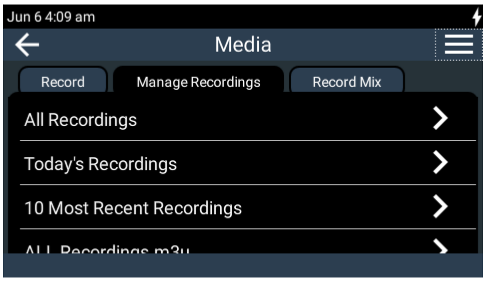
2. Navigate to a recording and tap to select it.



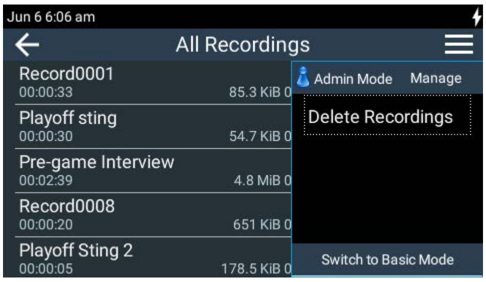
3. Adjust settings and add metadata as required.

Deleting Recordings

1. From the **Record screen** tap **Manage Recordings**, then navigate to a list of recordings.



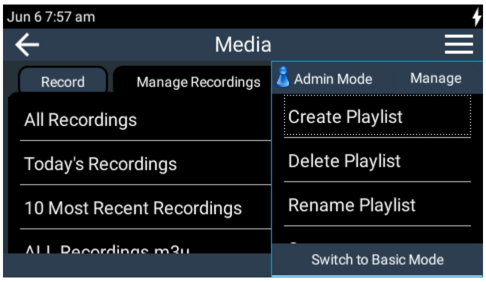
2. Tap **Menu** ≡ in the top left-hand corner of the **TOUCH SCREEN**, then tap **Delete Recordings**.



3. Tap to select the recordings to be deleted, then tap **Done**.

Creating a Playlist

Tap **Menu** in the top left-hand corner of the **Manage Recordings screen** to create and manage playlists of recordings. These features are also available in the **Playlist screen** in **Playback** mode. For more information on creating playlists see Playing Audio Files.

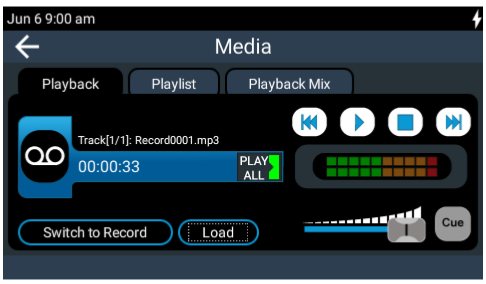


27.2 Playing Audio Files

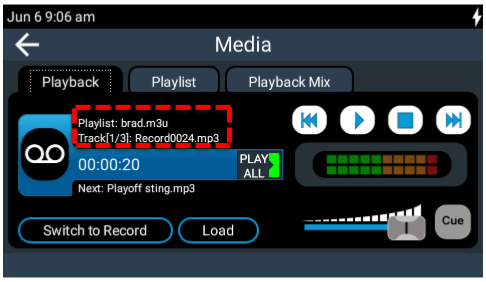
Before playing audio files ensure an SDHC card with a compatible audio file or files is inserted into the **SD CARD SLOT** on the rear panel of the codec.

Load a File or Playlist

1. Navigate to **HOME** > **Media** > **Playback**.
2. Tap the **Load** button.



3. Navigate to a group of recordings and tap to select a file, or a .m3u playlist file to load it. The new file or playlist is displayed on the screen to confirm it has been loaded.



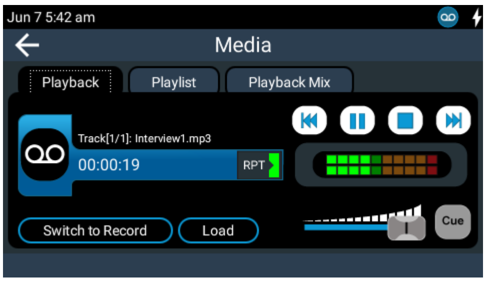
File Playback Controls and Playback Modes

Button	Playback Control
	Play button plays the currently loaded audio file.
	Pause button pauses a file being played.
	Stop button stops file playback and re-cues the file at the start.
	Skips backward to a previous file in a playlist when a file is not playing. Restarts playback at the beginning of a file when a file is playing.
	Skips forward to the next file in a playlist.

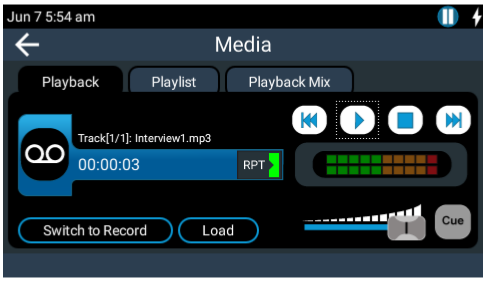
Button	Playback Mode
PLAY ALL	All files in a playlist will be played in sequence.
RPT	A specific track will repeat until stopped.
P.All RPT	All files in a playlist will be played in sequence and then the playlist will repeat playing all tracks in sequence until stopped.
RPT	The selected track is played once only.

Play a File

1. In the **Playback screen** tap **Play** to play an audio file. The **Playback symbol** will appear and flash in the **Status Bar** and a countdown timer is displayed to indicate playback time remaining for the file being played.



2. Tap the **Pause** button to pause playback. The **Paused symbol** appears in the **Status Bar** when file playback is paused.

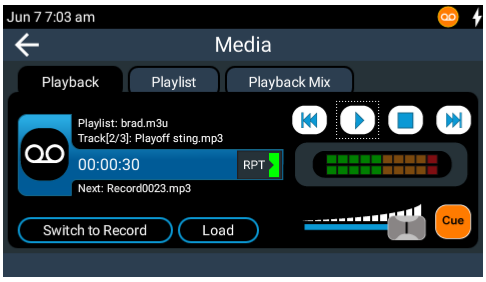


3. Press the **Stop** button to stop and re-cue a file. Use the **Skip Forward** button and **Skip Backward** button to navigate to different files in a playlist.

Monitoring a File in Cue Mode

Playback Cue mode allows a user to listen to a file offline to verify it is the correct file, or cue it up at a particular point.

1. In the **Playback screen** tap the **Cue button** to enable cue mode. The **Cue button** changes from grey to orange to indicate Playback Cue mode has been enabled.



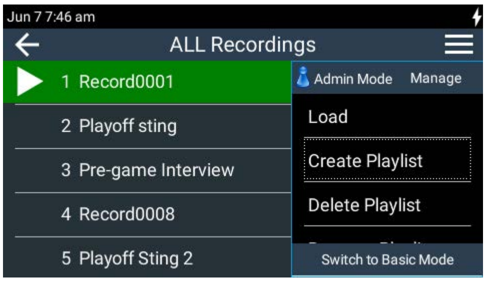
2. Use the playback controls to listen to a file offline.
3. Tap the orange **Cue button** to disable Playback Cue mode and revert to online file playback mode. This is confirmed by the **Cue button** turning grey.

Button/Symbol	Cue Playback Indication
	Cue button indicates Cue mode is disabled.
	Cue button indicates Cue mode is enabled.
	The **Cue symbol** appears in the **Status Bar** when Cue mode is enabled; the **Cue symbol** flashes in the **Status Bar** to indicate a file is playing while Playback Cue mode is enabled.
	The **Cue-pause symbol** in the **Status Bar** indicates file playback has been paused while in Playback Cue mode.

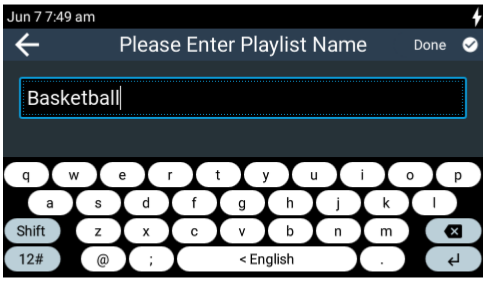
Creating a New Playlist

1. In the **Playback screen** select **Playlist** to display the **Playlist screen**. By default, the folder **All Recordings** is visible.

2. Tap **Menu** ☰ and select **Create Playlist**.



3. Enter a playlist name and tap **Done**.



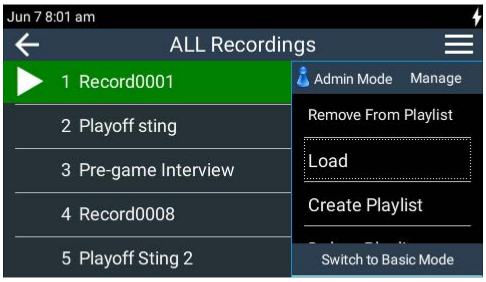
4. Tap to select recordings to add to the new playlist, then tap **Done**.

Importing Files

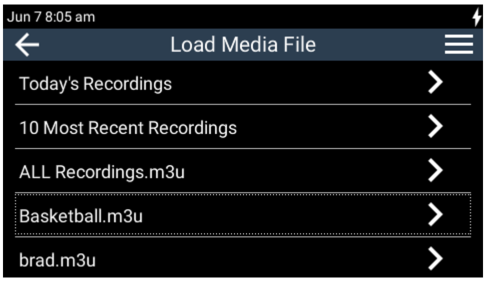
It is also possible to import individual files on an SD card for file playback in the codec. See Importing Audio Files for more details.

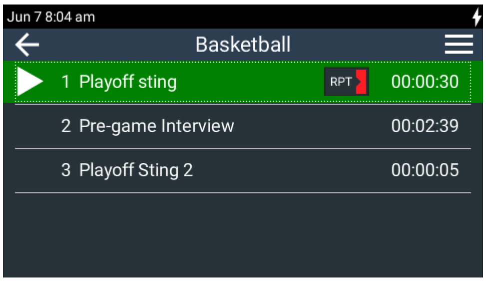
Loading a Playlist

1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.
2. Tap **Menu** ☰ and select **Load**.



3. Navigate to a playlist and tap to select and load it.





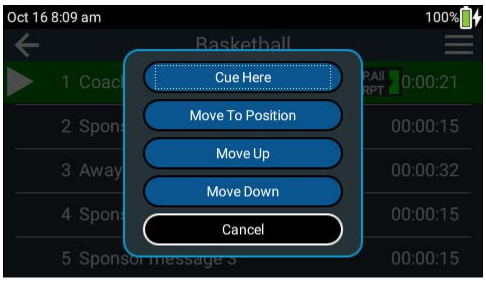
Managing Files in Playlists

Playing Files in the Playlist Screen

Tap the **Play symbol** or **Pause symbol** in a highlighted file to manage file playback from the **Playback screen**.

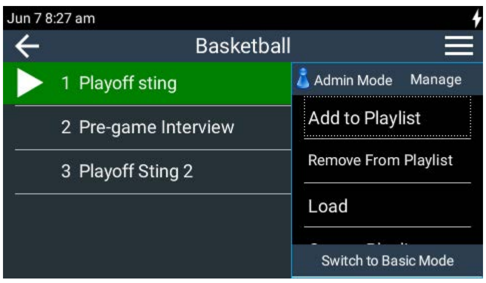
Playing Files Out of Order

Tap and hold a file in the playlist to select it and display file options, then tap **Cue Here** to cue the file for playback.

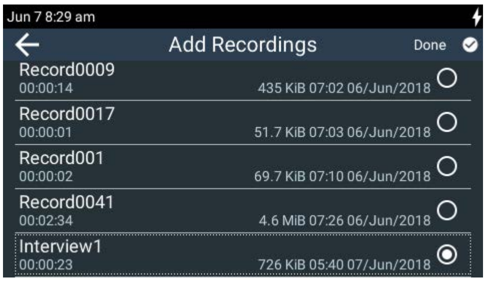


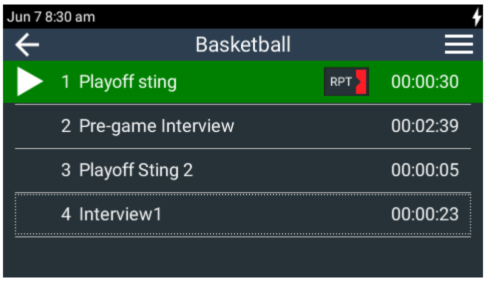
Adding Files to a Loaded Playlist

1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.
2. Tap **Menu** and select **Add to Playlist**.



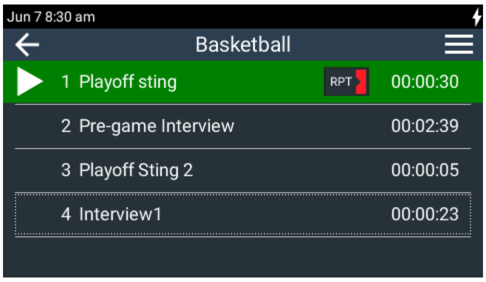
3. Tap to select the file or files to add to a playlist, then tap **Done** to add the selections to the currently loaded playlist.





Reorder a Playlist File

1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.



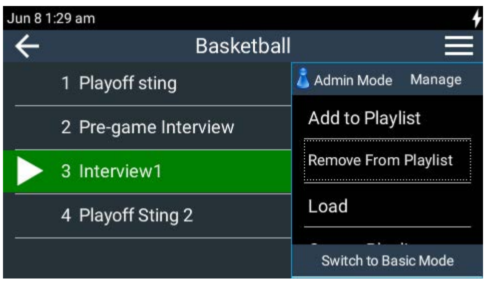
2. Tap and hold on a file to view reordering options.

3. Tap **Move Up** or **Move Down** to shift the selected file up or down in the playlist order. Tap **Move To Position** to move a file to a specified position in the playlist.

Remove a File from a Playlist

To remove a file or files from a playlist:

1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.
2. Tap **Menu** ☰ and select **Remove From Playlist**.

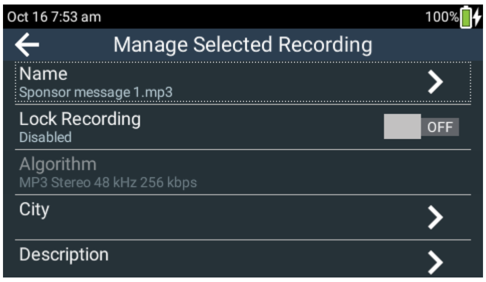


3. Tap to select the files to be removed from the playlist, then tap **Done**.

Rename a Recording

To rename a file in the **Playlist screen**:

1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.
2. Tap the file to be renamed to view file options.
3. Tap **Name** and then rename the recording, then tap **Done**.

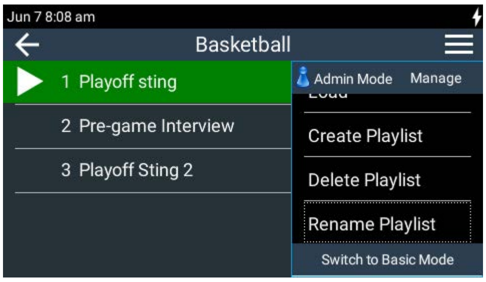


It is also possible to rename a file from the **Manage Recordings** screen. See [Record and Manage Audio Files] for more details.

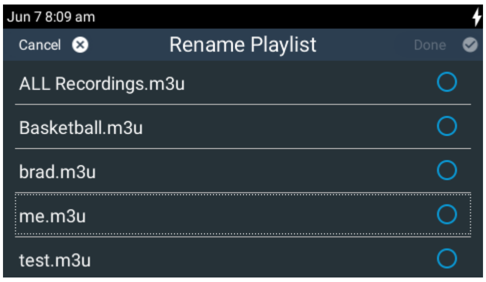
Rename a Playlist

Important Note: A playlist needs to be unloaded before it can be renamed.

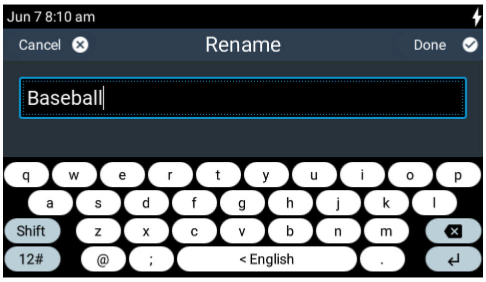
1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.
2. Tap **Menu** ☰ and select **Rename Playlist**.



3. Tap to select the playlist being renamed.

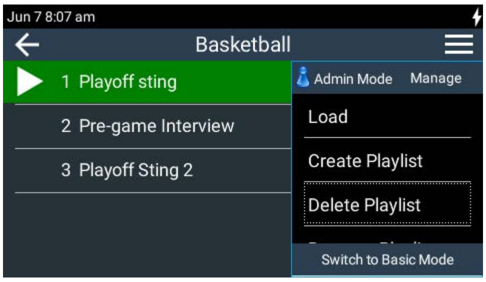


4. Enter the new playlist name and tap **Done**.

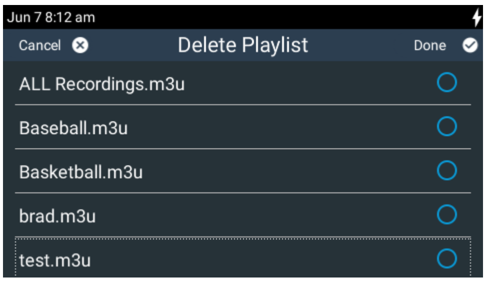


Delete a Playlist

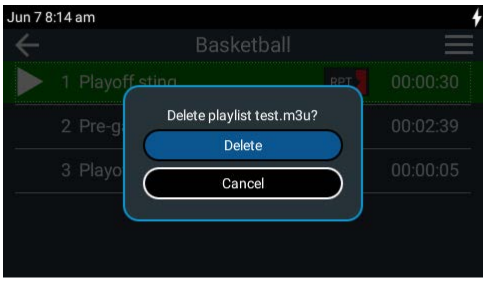
1. In the **Playback screen** select **Playlist** to display the **Playlist screen**.
2. Tap **Menu** ☰ and select **Delete Playlist**.



3. Navigate to the playlist to be deleted and tap to select it.



4. Tap **Delete** in the confirmation dialog to delete the playlist.



27.3 Importing Audio Files

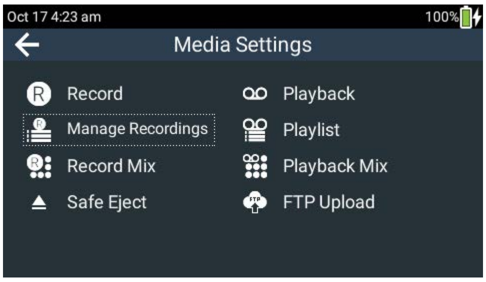
Import individual files on an SD card for file playback in the codec.

Importing Audio Files

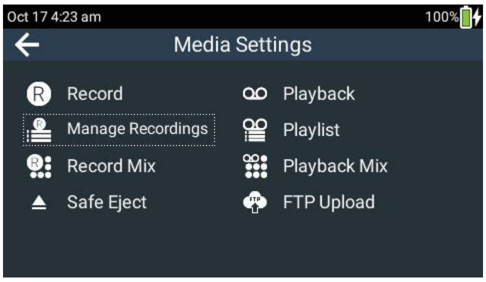
Imported files must reside in the same folder as files recorded on the ViA codec. Remove the SD card from the codec and save files in the **Recordings** folder available via **Tieline > Media > Recordings**. If using the record feature for the first time, create this folder structure on the SD card and copy all audio files before inserting the SD card into the codec.

To import audio files:

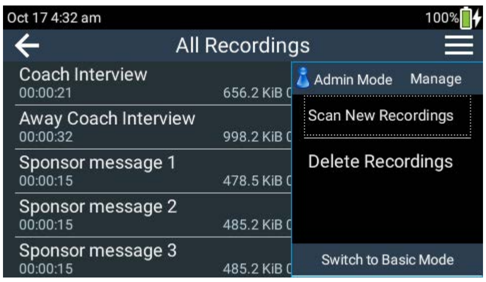
1. Navigate to **HOME** **> Media** > Manage Recordings .



2. Select **All Recordings**.



3. Tap **Menu** on the **All Recordings** screen and then tap **Scan New Recordings** to discover all imported files on the SD card.



4. The imported files will be added into the list of available recordings.

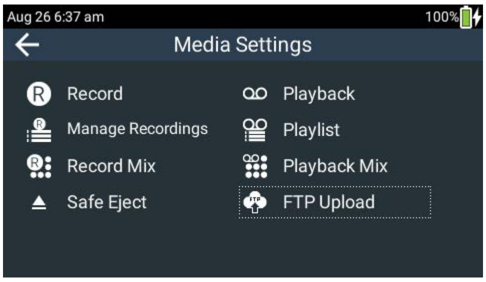
27.4 FTP File Uploads

The ViA can operate as an FTP client allowing file transfers to an FTP server from the **TOUCH SCREEN**.

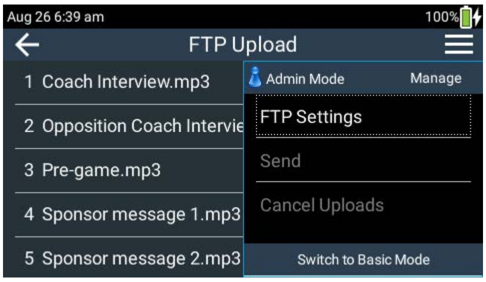
Important Notes: FTP file uploads are only supported using files on an SD card.

Configure FTP Settings

1. Navigate to **HOME** > **Media** > **FTP Upload**.



2. Tap **Menu** on the **FTP Upload** screen and then tap **FTP Settings** to enter server settings.

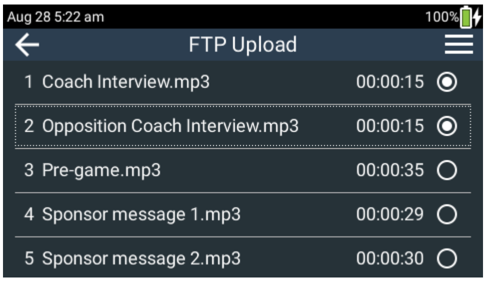


3. Tap to enter each FTP setting as required.

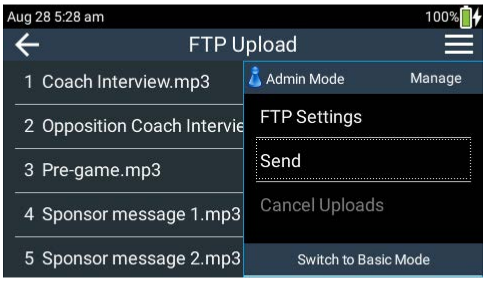
Uploading Files

Individual files or multiple files can be uploaded using FTP.

1. Navigate to **HOME** > **Media** > **FTP Upload**.
2. Tap to select files that are to be uploaded.

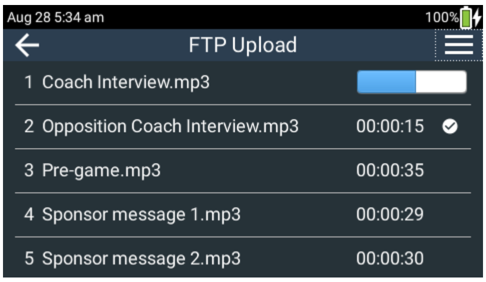


3. Tap **Menu** ≡ on the **FTP Upload** screen and then tap **Send** to upload all selected files.



4. Tap **Start Sending** in the confirmation dialog to commence FTP uploads.

5. A progress bar is displayed to show upload progress and a tick appears next to each successful upload.



28 About ISDN Modules

ISDN stands for Integrated Services Digital Network. The Basic Rate Interface (BRI) of ISDN consists of 2 bearer (B) channels at 64 kbps each and 1 data (D) channel at 16 kbps, i.e. (2B +D). This can be provided over a 2 wire facility and the two B channels can be bonded together to form a single 128kbps channel. The B channel can carry user information such as voice, video or data. The D channel carries signaling information between a user and the network.

The codec has a single module slot supporting 2 B channels using the ViA USA ISDN module. This delivers high quality mono or stereo audio over a single B channel using the Tieline Music algorithm. If you have 2 B channels you can use one as a standby link, or configure higher bandwidth mono or stereo connections using algorithms such as MusicPLUS, aptX Enhanced, or MPEG.

Important Considerations

There are several considerations when using your codec in ISDN mode, including:

- Will you operate within North America or other countries?
- Will you use a single B channel or 2 B channels?
- Which network are you using?
- Is your ISDN line Point-to-Point or Point-to-Multipoint?
- What are your directory numbers (DN)?
- If you are in the US, what are your Service Profile ID (SPID) numbers?
- What is your Multiple Subscriber Number (MSN) if you need to enter this outside North America?

ISDN configuration is influenced by the country in which you operate. For example, a SPID does not need to be entered into a Tieline codec for operation within Europe, but it does in North America.

U and S/T ISDN Interfaces

In North America the telephone company provides its BRI customers with a U interface. The U interface is a two-wire (single pair) interface from the phone switch. It supports full-duplex data transfer over a single pair of wires, therefore only a single device can be connected to a U interface.

The situation is different in Europe, the UK, most of Asia, Australia, Africa and parts of the Middle East, where the phone company is allowed to supply the NT-1 and the customer is given an S/T interface. The NT-1 is a relatively simple device that converts a 2-wire U interface into the 4-wire S/T interface.

If you have an NT-1 device connected to the U interface line then you will require a Tieline ViA (ST) ISDN module (S/T interface - model: TLISDNEUROVIA). If you don't have an NT-1 device installed then the Tieline ViA (U) USA ISDN module (U interface - model: TLISDNUSAVIA) will be required. You can ring your telecommunications provider to ask if you're not sure. Note: In Japan use the Tieline ViA (ST) ISDN module.

Important Note: Tieline S/T ViA ISDN modules do not have internal terminating resistors. When you connect terminating equipment such as a Tieline codec to an NT-1, 100 ohm termination resistors must be connected between pins 3 and 6 and between pins 4 and 5 at the last socket on the ISDN line. Check your NT-1 device user manual as this may be supported. Suppliers of electronic components sell suitable plugs with termination resistors when required. Please note: U interface ISDN terminations do not require terminating resistors.

How to Configure ISDN G5 Modules

To configure the codec to dial using ISDN for the first time:

1. Ensure that the correct country setting is configured in your codec via **Home screen > Settings > System > Country**.
2. If you are dialing between two Tieline codecs configure an ISDN dialing program via **Home screen > Dialer**.

More advanced module and answering settings can also be configured:

a. Select **Home screen > Settings > Transport Interfaces> ISDN Module** to adjust ISDN module settings specific to your codec site.

i. See [Configure ISDN Module Settings] for more information about configuration using the **TOUCH SCREEN**.
ii. See [Configuring ISDN Modules] to adjust settings using the HTML5 Toolbox Web-GUI.

b. ISDN answering can be configured to suit:
- Hardware available in the codec, i.e. the number of B channels available.
- Expected dialing behaviors, e.g. if B channels should bond or not, and whether audio streams need to use **Route** tags.
- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (sessionless calls).

To adjust answering configuration using the **TOUCH SCREEN** select **Home screen > Settings** **> Answering > ISDN Answering** and see ISDN Answering Configuration for more information.

To configure answering settings with the HTML5 Toolbox Web-GUI see Configuring ISDN Answering.

28.1 ISDN Module Settings

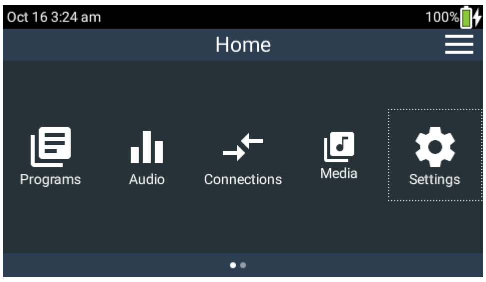
ISDN Module settings in the **Transport Interfaces** menu determine how an ISDN module operates at a particular site. Other answering-related settings are available in the **Transport** menu via **Answering > [Select ISDN Config**]. The codec displays ISDN line sync status in the **Status Bar** at the top of the **TOUCH SCREEN**.

	Symbol	Description of Status
1	No ISDN Symbol	No ISDN module is installed in the codec.
2		An ISDN module is installed in the codec and either: • No ISDN line is attached to the codec, or • ISDN line is attached but line sync hasn't been detected by the module
3		An ISDN module is installed in the codec and line sync has been detected.

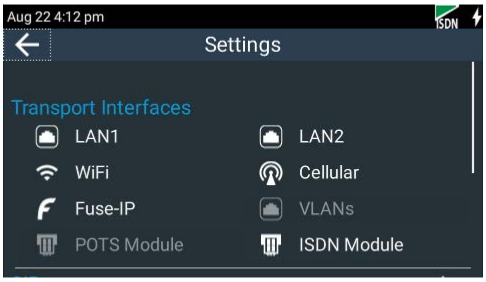
Configuring ViA ISDN Modules

To adjust ISDN module settings:

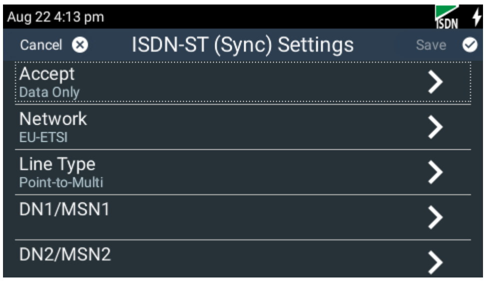
1. Press the **HOME** button to return to the **Home** screen and tap **Settings**.



2. Tap **Transport Interfaces** to expand the menu if it is not displayed, then tap **ISDN Module** to configure module settings.

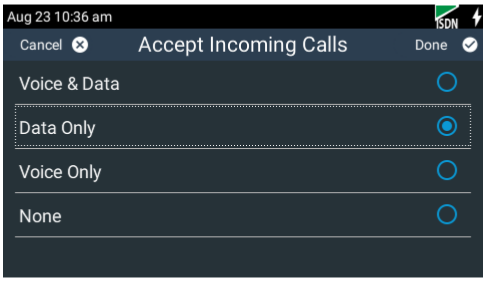


3. Tap **Accept** to adjust the type of calls the codec will accept.



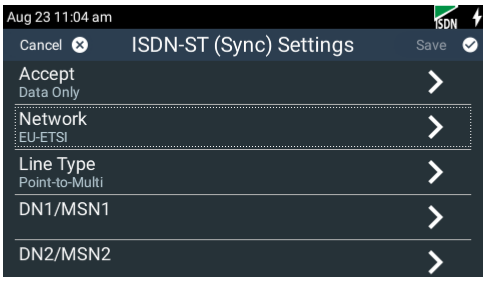
Important Note: ISDN **Sync** should be displayed when an ISDN line is connected to the codec. This appears regardless of whether you have configured the 'ISDN Line Type' correctly.

4. This menu is a call filter to allow or deny voice or data calls according to your preferences. By default **Data Only** is selected. Tap to accept or disallow voice or data calls according to your preferences.

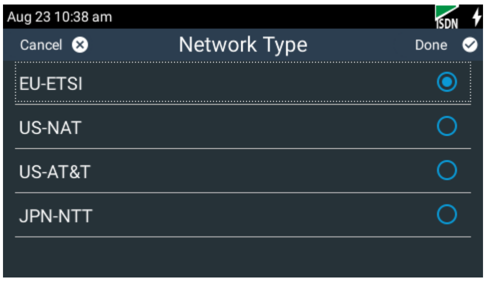


Important Note: G.711 is the default algorithm for incoming connections when **Voice Only** is selected. There are two G.711 algorithms and the one used by the codec depends on the country setting in the codec. The µ-law algorithm is used in the USA, Japan and Canada, whereas the A-law algorithm is used in other countries.

5. Tap **Network** to view **Network Type** options.

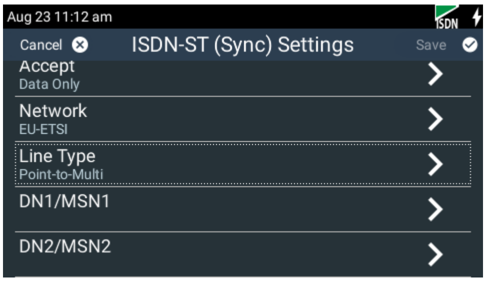


6. Tap to select the correct **Network Type** for the region in which you are using the codec.

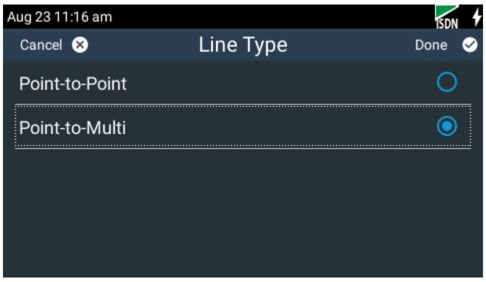


Networks	Region
EU-ETSI	If Switch Type is ETSI (UK, Europe, Australia and most other countries)
US-Nat	If switch type is National ISDN-1 and 2
US-AT&T	If switch Type is AT&T 5ESS
JPN-NTT	If you are in the Japan and your network is NTT

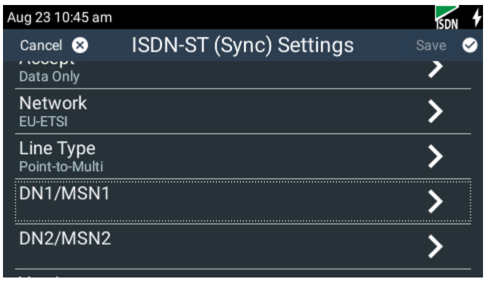
7. Tap **Line Type** to select whether your ISDN line is Point-to-Point or Point-to-Multipoint. This can be verified by asking your Telco.



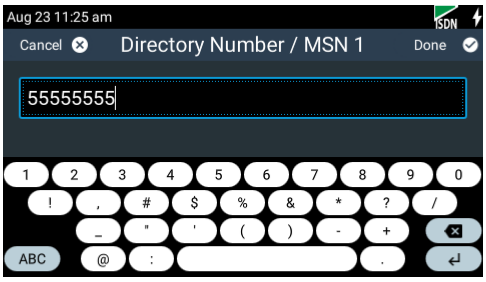
8. By default select **Point-to-Multipoint**, unless your switch type is point-to-point, your Telco says the line is point-to-point, or you are connected to a PABX system. Most PABX systems are point-to-point.



9. Tap to select each **SPID**, **DN** or **MSN**. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required. Note: **SPID1** and **SPID2** are only visible when a ViA (U) USA ISDN module is installed in the codec.



10. Use the on-screen keyboard to enter each number, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN** to add each number.



11. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to confirm and store all entered numbers.

Important Notes:

Directory Numbers and Multiple Subscriber Numbers
Directory Numbers (DN) in North America and Multiple Subscriber Numbers (MSN) in the rest of the world are simply phone numbers associated with an ISDN B channel, like lines listed in a typical phone directory. Your Telco will normally supply 2 DN/MSN numbers for each pair of B channels. However, these numbers may or may not be associated with a specific B channel.

Often broadcasters prefer to predict which B channel will answer an incoming call to ensure audio routing is consistent. However, if a DN or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number. Programming DN/MSN numbers for each B channel allows the codec to ignore calls without matching DN/MSN numbers. This is the best way to answer calls from codecs in a predictable manner.

SPID Numbers in North America
ISDN relies on an initialization procedure for associating Service Profiles with specific terminating equipment (e.g. your audio codec) rather than lines. In the US Telcos assign a Service Profile ID (SPID) number which assists in identifying different ISDN services across the network. Your Telco must provide a SPID for each B channel you order when connecting over US-Nat or US-AT&T networks in the US. A SPID is not required when using the AT&T PTP protocol.

Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch. The switch then associates the SPID with a specific Service Profile and directory number.

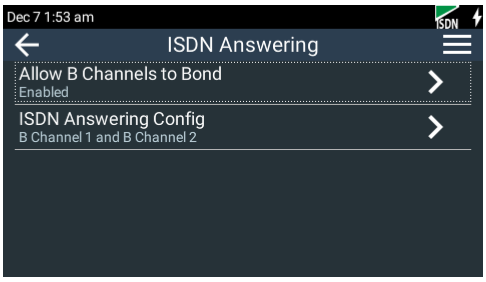
Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

28.2 ISDN Answering Configuration

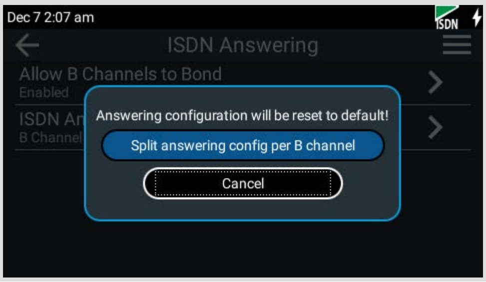
Important Note: For more detailed information about configuring **ISDN Answer Config** parameters using the HTML5 Toolbox Web-GUI, including bonding and 'route' configuration, see Configuring ISDN Answering.

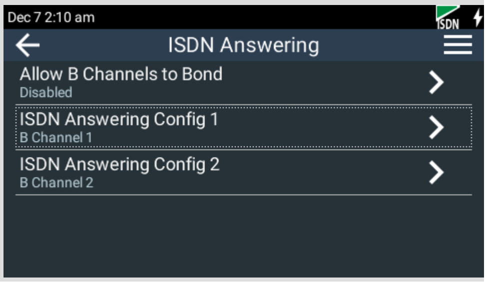
ISDN Answering Configs determine how the codec will behave when answering ISDN calls at a particular site. ISDN module settings may need to be adjusted depending on your country and network requirements. Note: You can copy similar programs between codecs installed at different locations and configure site-specific settings for how the module should connect.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Navigate to **Answering** and then tap **ISDN Answering**.

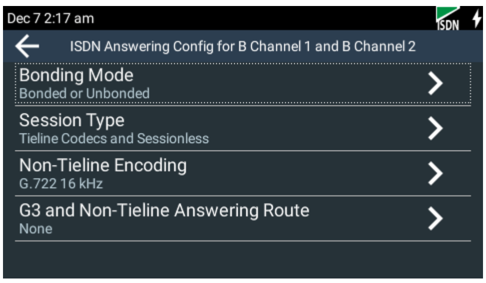


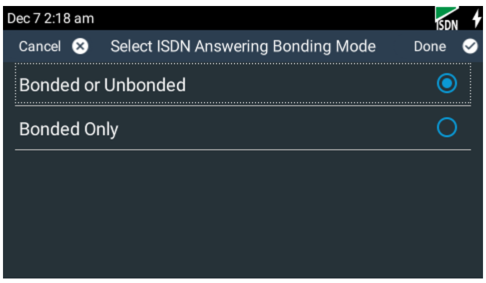
Important Note: By default, the codec is configured to allow bonding of 2B channels. In this mode the codec will accept a 1B call, two 1B calls, or a bonded 2B connection. Tap **Allow B Channels to Bond** to configure B channels separately and then tap **Split answering config per B channel**. This will allow configuration of each B channel individually and two **ISDN Answering Configs** are displayed.



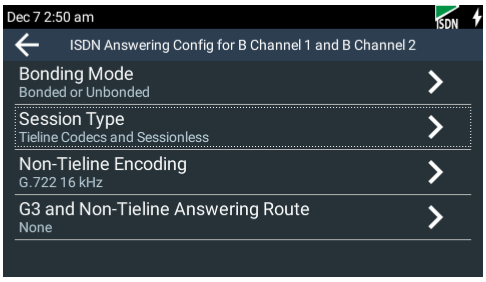


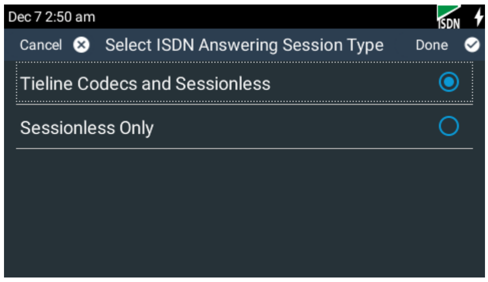
3. Tap **Bonding Mode** when using a single config for two B channels to adjust the bonding method.





4. Tap **Session Type** to adjust the session data setting.

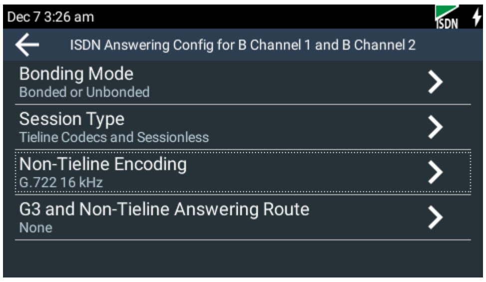


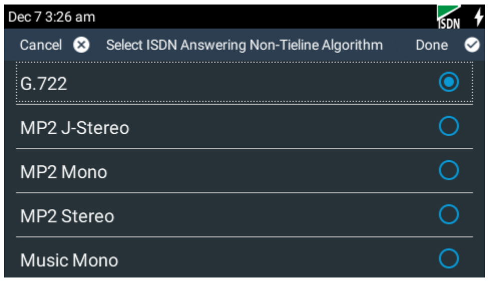


Important Note: By default, calls from Tieline codecs contain session data information, which includes algorithm, bit rate, route and other settings. Calls from non-Tieline codecs are sessionless.

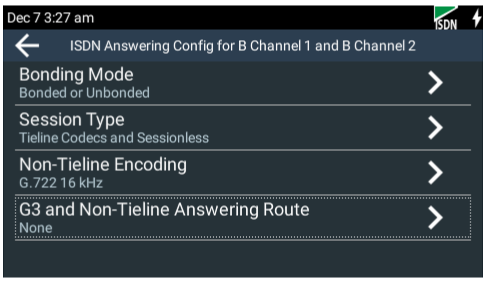
1. Select **Tieline Codecs and Sessionless** if answering calls from both Tieline and non-Tieline codecs.
2. Select **Sessionless Only** if answering calls from non-Tieline codecs only. Note: When **Sessionless Only** is configured, the codec will not wait to receive Tieline session data and this reduces the time taken to answer a sessionless call. **Non-Tieline Encoding** and answer route settings are used to answer sessionless calls.

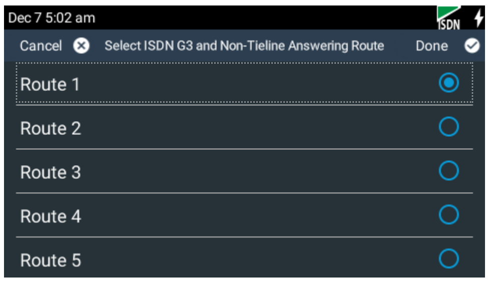
5. Tap **Non-Tieline Encoding** and then tap to select the default algorithm used when receiving a call from a non-Tieline codec.





6. Tap **G3 and Non-Tieline Answering Route** to configure an answer route when receiving a call from a non-Tieline or Tieline G3 codec, then tap to select a **Route**.





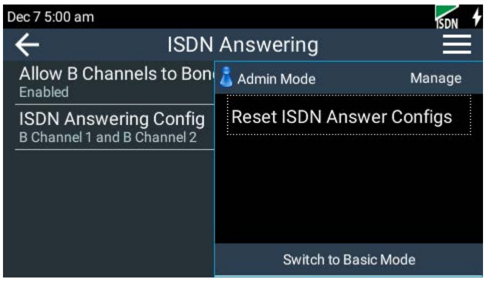
Important Note: Routing configuration is unsupported on Tieline G3 and non-Tieline codecs. Configure an ISDN Answer Route to direct incoming calls to a matching Answer program route. Note: If there is a mismatch between the ISDN Answer Route and an Answer program route, the call can still connect and routes are ignored.

Reset ISDN Answering Settings

To reset ISDN answering settings to factory defaults:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.

2. Navigate to **Answering** and then tap **ISDN Answering** .
3. Tap **Menu** in the top left corner of the **TOUCH SCREEN** and then tap **Reset ISDN Answer Config**.



4. Tap in the confirmation dialog to revert to factory default settings.

29 About POTS Modules

The ViA POTS G5 module is essentially the same as other G5 POTS modules in Genie and Merlin codecs. It can be used in the codec to stream high quality audio over a POTS (PSTN) phone line. The Tieline Music algorithm can deliver 15 kHz quality bi-directional audio at bit rates as low as 24kbps over a POTS connection.

Modem Negotiation and Line Quality

The codec can send and receive high-speed digital information over a standard POTS telephone line via the modem in the ViA POTS module.

ViA POTS module modems initially attempt to establish a link at the lowest **Maximum Bit rate** setting configured in the **POTS Module** settings. If the POTS line doesn't support a selected bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link. The modem then performs a process called ‘training’, during which the codecs at each end of the link analyze the line. The codecs will then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%.

The ViA POTS module contains a SmartDAA™ (Smart Data Access Arrangement) line interface, which isolates the modem from voltages on phone lines. It is important to select the correct country in the codec from which you are dialing. This allows the SmartDAA to automatically adjust for varying ring tones and line impedances in different countries.

Connecting to G3 Codecs using POTS

The codec will successfully connect to Tieline Commander G3 and i-Mix G3 codecs, as well as the Genie and Merlin codec families over POTS. Tieline G3 codecs may use:

- **POTS** modules (older superseded G3 codec version)
- **POTS G3** modules (current G3 codec version)

Connecting to POTS G3 Modules

POTS G3 modules operate in the same way as all G5 POTS modules (Genie, Merlin and ViA) when connecting, e.g. they establish a link at the default bit rate of 28.8kbps and then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%.

Connecting to Legacy POTS Modules in G3 Codecs

These modules have slightly different characteristics when connecting. When dialing from a G5 POTS module to these older POTS modules the codecs will attempt to connect initially at 19.2kbps. If line quality is above 80% at this bit rate then the codec will 'retrain' the connection up to a maximum of 28,800bps (depending on modem handshaking). The codec will then renegotiate the link downwards to the highest possible bit rate where line quality is greater than 70%.

How to Configure ViA POTS G5 Codec Connections

To configure the codec to dial using POTS for the first time:

1. Ensure that the correct country setting is configured in your codec via **Home screen >** **> System > Country**. This ensures the correct settings are used by the codec when making POTS connections.
2. When dialing between two Tieline codecs use the **Dialer** to create a POTS dialing program.

Other more advanced settings can also be configured:

1. Select **Home screen > Settings** **> Transport Interfaces > POTS Module** to adjust POTS module settings specific to your codec site.
   a. See Configure POTS Module Settings for more information about configuration using the **TOUCH SCREEN**.
   b. See Configuring POTS Modules to adjust settings using the HTML5 Toolbox Web-GUI.
2. If you are connecting to non-Tieline codecs you may need to create an answering "Config", which determines the settings used when answering a non-Tieline POTS call:
   a. To adjust answering configuration using the **TOUCH SCREEN** select **Home screen > Settings** **> Answering > POTS Answering** and see POTS Answering Configuration for more information.
   b. See Configuring POTS Answering to adjust settings using the HTML5 Toolbox Web-GUI.

Important Note: The phone input is used to **Monitor Modem Tones** (default setting **On**) and for monitoring and adjusting audio in **Analog Phone** mode (default is **POTS Codec** mode). These settings are configured via **Settings** **> Transport Interfaces > POTS Module** **> Answer Mode**.

Making Analog Phone (Voice) Calls

The ViA POTS G5 module is capable of making analog voice calls. This may be useful to dial a telephone hybrid, or to use for communications, or when there is no Tieline codec at the other end of the link. Remember analog voice calls are only 3 kHz audio quality. To configure analog phone answering mode navigate to **Settings** **> Transport Interfaces > POTS Module** **> Answer Mode > Analog phone**.

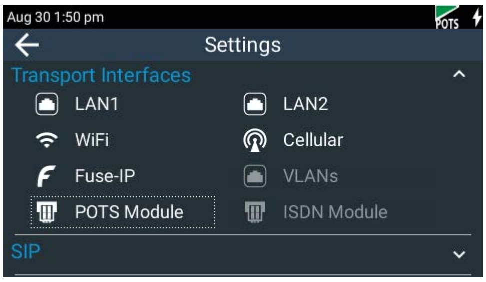
29.1 POTS Module Settings

Navigate to **Settings** > **Transport Interfaces > POTS Module** to configure how the codec operates at a particular site when using POTS. Other POTS answering-related settings are available in the **Answering** menu via **Settings** > **Answering > POTS Answering** if you are connecting to non-Tieline codecs. The codec displays POTS line status in the **Status Bar** at the top of the **TOUCH SCREEN**.

	Symbol	Description of Status
1	No POTS Symbol	No POTS module is installed in the codec.
2		A POTS module is installed in the codec but no POTS line is attached to the codec
3		A POTS module is installed in the codec and line voltage has been detected

Configure ViA POTS Modules

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings** .
2. Navigate to **Transport Interfaces** and then tap **POTS Module** .



3. Complete configuration changes as per the following options and then tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to confirm all settings.

Module (Site) Settings

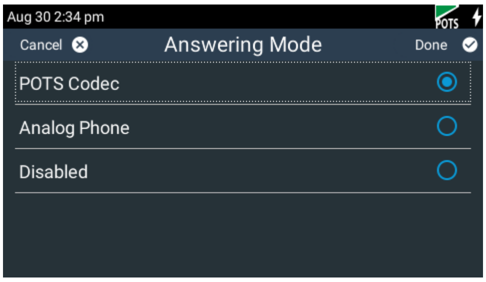
Country

This displays the current country setting in the codec. To adjust this setting select **Settings** **> System > Country** .

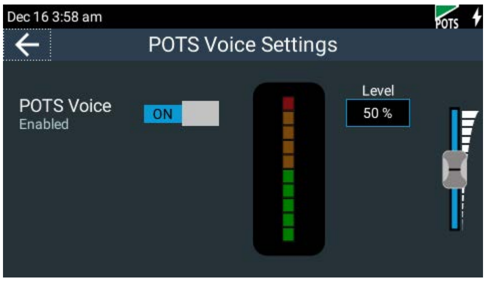
Answer Mode (Affects Answering Only)

Answer Mode determines how the selected module in the codec will be able to answer incoming POTS line calls. Options include:

- **POTS Codec:** allows the module to receive encoded audio data over a POTS line.
- **Analog Phone:** configures the module to receive a standard analog phone call.
- **Disabled:** disables the module from receiving a **POTS Codec** or **Analog Phone** call.



In **Analog Phone** mode the **Phone** input is visible on the **Input Audio** menu, which is accessed via **Settings** > **Audio** > **Inputs**. Tap the **Phone** input to adjust the level of the phone input and to turn it **On** or **Off**.



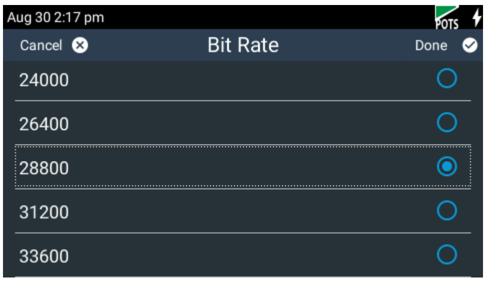
Other call answering settings are accessible via **Settings** > **Answering** > **POTS Answering**. Adjustments to these settings are not normally necessary when connecting between Tieline codecs. Default session and algorithm settings may need to be adjusted when connecting to non-Tieline codecs over POTS. To configure answering settings in the HTML5 Toolbox Web-GUI see Configuring POTS Modules.

Maximum Bit rate (Affects Dialing and Answering)

Configure connection bit rates from 9.6kbps to 33.6kbps. The default **Max Bitrate** setting is **28800** (28.8kbps) and this only affects **POTS Codec** calls. Even if the line supports a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted. Reducing this value can improve connection reliability on poor quality lines.

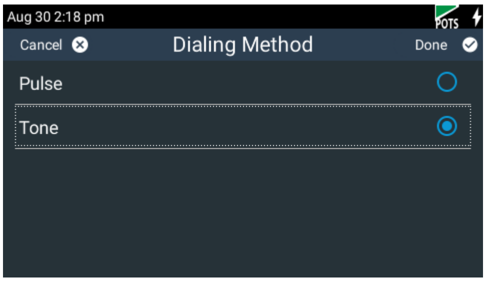
ViA POTS G5 modules initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

In the initial connection phase, the modems perform a process called ‘training’, to analyze the line and compensate for frequency and phase response. This also cancels out any echo that may be present. The codec will then 'renegotiate' the link downwards to the highest possible bit rate where line quality is greater than 70%. Negotiation is the process of bit rate adjustment.



Dialing Method (Affects Dialing only)

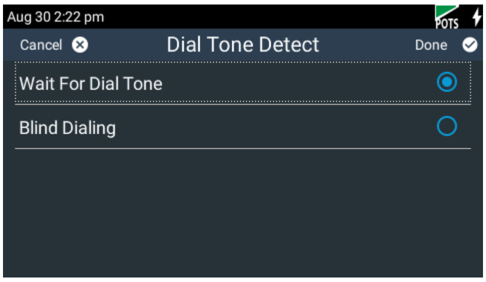
Use this menu to select **Tone** (DTMF) or **Pulse** dialing over **POTS Codec** connections. Tone dialing (default) is used always when the **Answer Mode** is **Analog Phone**.



Dial Tone Detect (Affects Dialing only)

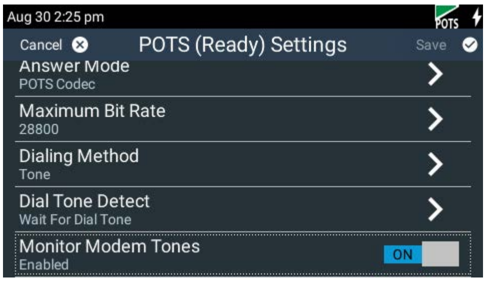
There are two settings in this menu:

- **Wait For Dial Tone**: The module will only dial when a dial tone is present on the line (default).
- **Blind Dialing**: Allows the module to dial when no dial tone is present.



Monitor Modem Tone (Affects Dialing and Answering)

Tap the **On/Off** button to **Enable** (default) or **Disable** this feature. If enabled the module will allow audio monitoring of modem tones via the phone input during **POTS Codec** mode connections.



Important Notes:

- POTS dial and modem tones are always audible in the left and right sides of the headphone output when connecting.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.

29.2 POTS Answering Configuration

POTS answering can be configured to suit:

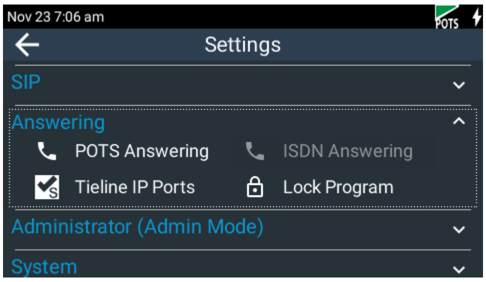
- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (Sessionless).
- Expected dialing behaviors and encoding, e.g. whether audio streams use **Route** tags and which algorithm is used.

By default, Tieline codecs communicate information via session data transferred when a connection is established. If you answer a call from a non-Tieline codec (or a sessionless call from a Tieline codec), **POTS Answering** settings determine the algorithm used when connecting. An **Answering Route** can also be configured, which routes a call to a specific audio stream on the answering codec.

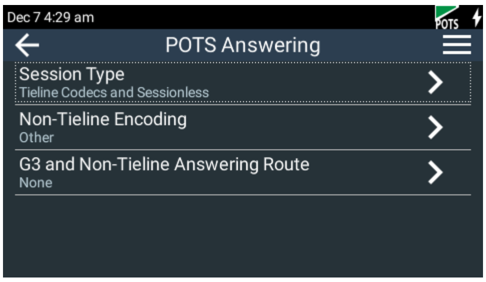
Important Notes:

- **POTS Answering** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When call is received from a Tieline codec sending session data (i.e. not a **Sessionless** connection), the algorithm setting sent from the dialing codec overrides the algorithm setting in the **POTS Answering** menu.
- The default **POTS Answering** settings will accept a call from an incoming Tieline codec with session data enabled, or a sessionless call from a Comrex POTS codec using the **Other** algorithm.
- For more information about POTS answering parameters, including 'route' configuration, see Configuring POTS Answering in the HTML5 Toolbox Web-GUI manual.

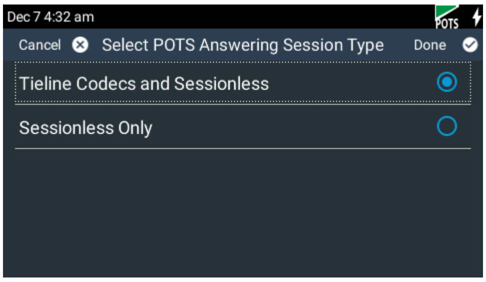
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Navigate to **Answering** and then tap **POTS Answering**.



3. Tap **Session Type** to adjust whether to accept Tieline session calls or sessionless calls only.



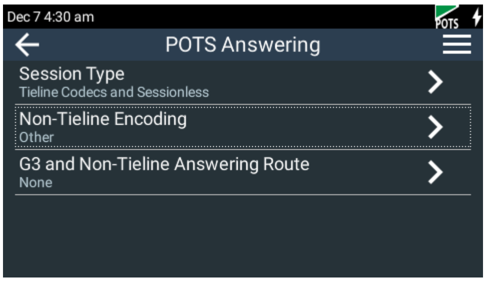
4. Tap to select **Tieline Codecs and Sessionless** or **Sessionless Only** if connecting to non-Tieline codecs only.

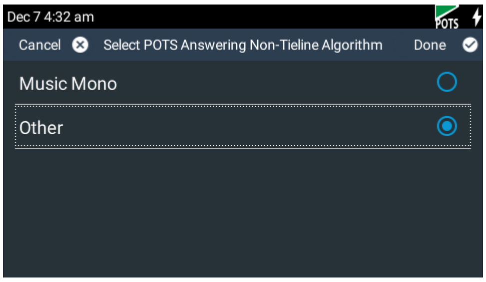


Important Note: By default, calls from Tieline codecs contain session data information, which includes algorithm, bit rate, route and other settings. Calls from non-Tieline codecs are sessionless.

1. Select **Tieline Codecs and Sessionless** if answering calls from both Tieline and non-Tieline codecs.
2. Select **Sessionless Only** if answering calls from non-Tieline codecs only. Note: When **Sessionless Only** is configured, the codec will not wait to receive Tieline session data and this reduces the time taken to answer a sessionless call. **Non-Tieline Encoding** and answer route settings are used to answer sessionless calls.

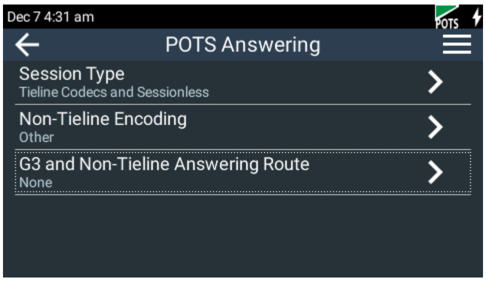
5. Tap **Non-Tieline Encoding**, then tap **Other** when connecting to Comrex® Vector, Matrix® and BlueBox® codecs.

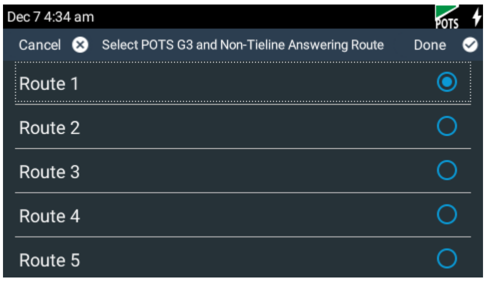




Important Note: On the Comrex codec select the Comrex "Music" algorithm. Please note that 9.6kbps connections are not supported by Comrex codecs.

6. Tap **G3 and Non-Tieline Answering Route** to specify the audio stream **Route** when answering a sessionless call. Tap to select a specific **Route**. For more information on Answering routes see Configuring POTS Answering.



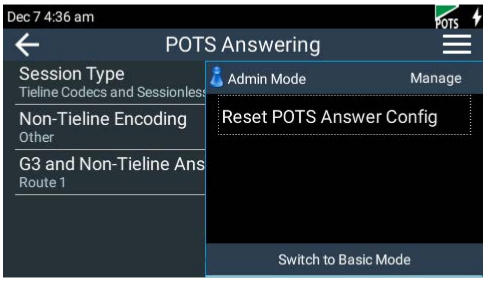


Important Note: Routing configuration is unsupported on Tieline G3 and non-Tieline codecs. Configure a POTS Answer Route to direct incoming calls to a matching Answer program route. Note: If there is a mismatch between the POTS Answer Route and an Answer program route, the call can still connect and routes are ignored.

Reset POTS Answering Settings

To reset POTS answering settings to factory defaults:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Navigate to **Answering** and then tap **POTS Answering**.
3. Tap **Menu** in the top left corner of the **TOUCH SCREEN** and then tap **Reset POTS Answer Config**.



4. Tap in the confirmation dialog to revert to factory default settings.

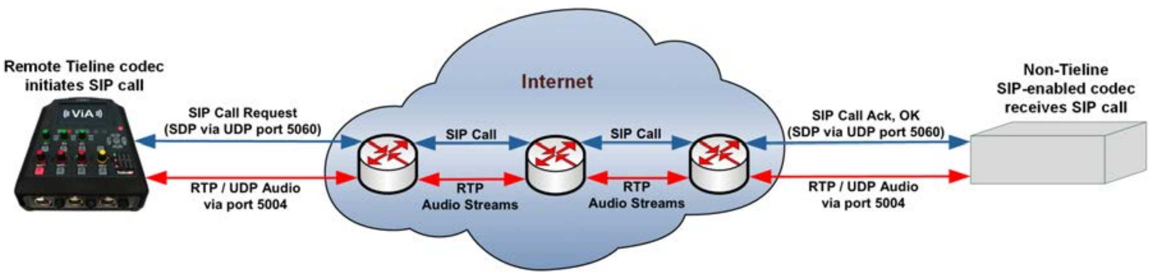
30 About SIP

To dial between Tieline and non-Tieline codecs over IP it is necessary to configure all codecs to connect in SIP mode. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. Tieline IP codecs are EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs.

SIP is also a useful way of dialing another device and locating it easily. This task is usually performed by SIP servers, which communicate between SIP-compliant devices to set up a call. SIP connections can be made in two ways; registered or unregistered.

Unregistered Peer-to-Peer SIP Connections

Codecs don't need to be registered to a SIP server to dial peer-to-peer SIP connections. An unregistered SIP peer-to-peer connection involves two codecs connecting to each other directly using an IP address, as you would for a standard Tieline IP call. This is simpler and much like the way codecs normally connect. The difference is that a Tieline IP call uses proprietary Tieline session data to negotiate call parameters (e.g. algorithm and bit rate) when a call is established, whereas a peer-to-peer SIP connection uses Session Description Protocol (SDP) for this purpose. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting dissimilar devices and is used when connecting Tieline codecs to non-Tieline devices.

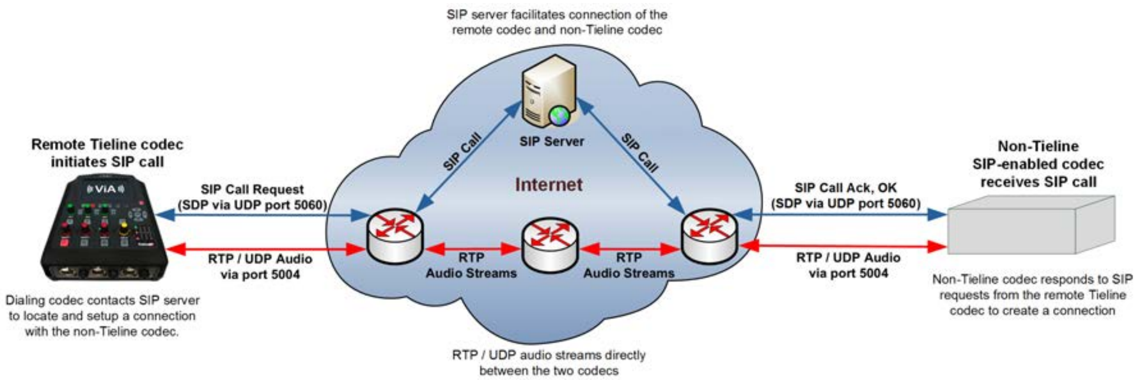


There are two very distinct parts to a call when dialing over IP. The initial stage is the call setup stage and this is what SIP and SDP is used for. The second stage is when data transfer occurs and this is left to the other protocols such as RTP/UDP to stream audio data. SDP works with a number of other protocols, to deliver the following functions when connecting devices over SIP:

- Establish a codec's location.
- Determine the availability of a codec.
- Negotiate the features to be used during a call, e.g. the algorithm and bit rate.
- Provide call management of participants.
- Adjust session management features while a call is in progress (e.g. termination and transfer of calls).

SIP Server Connections

The benefit of using a SIP server to connect is that any device can be 'discovered' via its SIP server registration. This is particularly useful if a codec is being used in multiple locations with IP addresses that are DHCP assigned. These DHCP addresses are unreliable and are not recommended for live broadcast connections. As long as your codec and the device you are dialing are both registered to a SIP server you can connect by simply dialing the destination SIP address.



Some SIP servers route RTP audio through the SIP server as well and Tieline recommends avoiding this type of server whenever possible. Otherwise you will be reliant on the SIP server for streaming broadcast audio packets and most servers are not designed for mission critical packet streaming.

To dial a codec via a SIP server requires:

1. Both devices to be registered with separate SIP accounts.
2. Both codecs configured to operate in SIP mode.
3. The IP address of the SIP server.
4. An IT administrator to open UDP port 5060 to enable SIP traffic, as well as UDP audio port 5004.

A SIP server administrator should be able to provide the following details to enable SIP registration of a device:

- Authenticating username
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port

Advantages and Disadvantages of Using SIP

Advantages of SIP

1. SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices, e.g. Session Description Protocol (SDP).
2. EBU N/ACIP Tech 3326 provides a minimum set of requirements necessary to ensure interoperability between equipment intended for the transport of audio over IP networks. It standardizes the use of ports, encoders, transport protocols and signaling to ensure codecs, and other SIP-enabled devices like smartphones and VoIP phones, can connect successfully.

Disadvantages of SIP

1. SIP requires both codecs to be configured with the same settings before connecting or the connection will fail. When dialing Tieline to Tieline the dialing codec provides all connection information to the answering codec. This is not possible when using SDP in SIP mode.
2. Not all professional codec manufacturers are fully compliant with all requirements for interoperability.
3. SIP connections are more complex to configure.
4. SIP does not support advanced software enhancements which deliver redundancy and rock solid reliability over IP, e.g. failover connections, SmartStream PLUS redundant streaming, Fuse-IP bonding, plus error concealment strategies.
5. Codecs using SIP cannot use the TieLink Traversal Server for presence and connections. In addition port forwarding is usually required.
6. Some ISPs and/or cellular networks may block SIP traffic.

SIP Security

Tools such as Shodan make it easy for anyone to easily locate devices connected to the internet around the world. Therefore it is critical that security measures are in place for all IP and SIP connections over the public internet.

Managing Unwanted SIP Calls

Hackers and other nefarious net-bound characters look for networks with easy access in which to ply their trade. As a starting point they look for networks with open gateways and platforms using default passwords.

Maintaining Codec Network Security

Adequate security is a major factor in ensuring your codecs and your broadcast network remain secure. There are several layers of security available in Tieline codecs to maintain secure connections. These include:

1. Immediately change the default password when you commission and install your codecs (see instructions which follow). Create a strong password which includes both capital and lower case letters, symbols and numbers (up to 15 characters can be entered). Password managers can be useful when managing multiple passwords within organizations.
2. Ensure your codec is behind a firewall and only open the TCP and UDP ports required to transmit session and audio data between your codecs. Using non-standard ports instead of Tieline default ports can also ensure the codec is more difficult to discover by external parties.
3. Ports 80 and 8080 are commonly used to access the Tieline codec web server. You can add an additional layer of security by translating these ports on the WAN side of your network

into non-standard port numbers. Adjust ports using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Configuring TCP/UDP Ports.
4. By default SIP interfaces are disabled to avoid unwanted traffic. The **SIP Filter Lists panel** in the Toolbox HTML5 Web-GUI allows filtering of SIP URIs and User Agents to provide greater security when using SIP. See Configure SIP White and Blacklists for more information.
5. An SSL security certificate can be installed on each codec in your network to ensure it is a trusted device within your network. See Installing a Security Certificate for more information.
6. Firewall settings to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Adjust settings using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Firewall Configuration.
7. Implementation of CSRF protection (Cross-Site Request Forgery). Enable and disable this setting using the **Options panel** in the Toolbox HTML5 Web-GU, or see Enabling CSRF Security for more info.

Be sure to document any port changes because this information will be required if you need to contact Tieline or other online support services.

Understanding SIP Terminology

What is a URI?

A Uniform Resource Identifier (URI) is a string of characters identifying a name or resource on the internet. A Uniform Resource Locator (URL) is a URI specifying where an identified resource is available (e.g. network location) and the mechanism for retrieving it. E.g. http://twitter.com, where http:// is the protocol used to retrieve information from "twitter.com" which is the network location.

What is a SIP URI?

SIP URIs can be used to whitelist or blacklist devices from connecting to a codec. A SIP URI is the address or characters used to call another person via SIP. A SIP URI is essentially a user's 'contact address' and is used by VoIP phones and other devices to call using SIP. An example of a SIP URI is: "sip:tieline_test1@getonsip.com". When dialing using Tieline codecs "sip:" is automatically placed in the dial string and does not need to be entered. SIP URIs can be used to whitelist or blacklist devices from connecting to a codec.

Important Notes:
- The default UDP audio port when using SIP for a peer-to-peer connection is 5004 in Tieline codecs.
- Enter the IP address or SIP URI, then a full colon and the session port number to change the session port from the default setting of UDP port 5060, e.g. "sip:tieline_test1@getonsip.com:5070"
- Tieline codecs automatically add "**sip:**" to the address entered in the **Address** field when dialing, so it's not necessary to add this.
- To only allow a predefined list of codecs to connect, add them to the **URI Whitelist** and add a wildcard (asterisk) * to the **URI Blacklist**: all incoming calls will be blocked except for codecs in the Whitelist.

What is a SIP User Agent?

A user agent is a software agent acting on behalf of a user. Hackers use SIP user agents or 'botnets' to scan for open ports on the internet. When they locate open ports they can force their way into SIP servers and scan for valid accounts to use for fraudulent purposes, like making free international calls. Known user agents like "sipvicious" and "friendly-scanner" can be added to a User Agent Blacklist in Tieline codecs to stop them from accessing them.

How do I find my Codec's User Agent

It may be necessary to discover a codec user agent as some codec manufacturers allow whitelisting of calls by 'User Agent.' Therefore, it may be necessary to enter a Tieline codec user agent into a non-Tieline codec to allow it to connect to a Tieline SIP codec.

- From firmware v2.16.xx and later releases the user agent in a Tieline G5 codec is configured as "Tieline <ProductName> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68.

- In Tieline G3 codecs the user agent is configured as "Tieline <ProductName> <Serial Number>". E.g. Tieline TLR350 8972. The model numbers for Tieline G3 codecs are as follows:

  - Commander G3 Rack TLR300 = Model Number TLR300

  - Commander G3 Rack TLR300B = Model Number TLR350

  - Commander G3 Field TLF300 = Model Number TLF300

  - i-Mix G3 TLM600 = Model Number TLM600

Troubleshooting SIP Connections

Manufacturers of professional IP codecs in most cases do not use the standard SIP ports (UDP 5060 and UDP 5004) for making IP connections. Therefore you will need to reconfigure each codec to use the standard SIP ports in most instances.

Most of the time EBU N/ACIP Tech 3326 SIP compliant codecs should connect when using the same encoders and connection settings at both ends. However, if one-way audio is encountered, it is highly likely to be a port forwarding issue. Port forwarding configuration instructions for most popular routers are available at www.portforward.com.

Some manufacturers support a subset of the EBU N/ACIP Tech 3326 recommendations so not all algorithms are supported by all manufacturers. G.722 is supported by most codec manufacturers, so if you encounter connection issues, default to using this algorithm for most troubleshooting scenarios.

Getting Started with SIP

To dial over SIP peer-to-peer without using a SIP server see Dialing SIP Peer-to-Peer. To dial over SIP using a SIP Server you will need to:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: This **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. LAN1 or LAN2.

Common SIP settings Tieline recommends when connecting to non-Tieline codecs like Comrex are as follows:

- Profile: Mono
- Bit rate: 64kbps
- Algorithm: G.722
- Session port: UDP 5060
- Audio port: UDP 5004

Important Notes:

- The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**.
- SIP dialing is only supported over point-to-point connections, not multi-unicast connections.
- The codec supports a SIP call being placed on-hold. Note: there are several different implementations of "on-hold" by various SIP providers. Some will stop streaming when a call is placed on-hold and others will replace live streaming with on-hold messages or music.
- Tieline supports RFC5109 and RFC2733 compliant FEC over SIP from firmware v2.18.xx.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- Tieline G3 codecs do not support connections using algorithms like AAC, aptX Enhanced and Opus and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level in the codec. To do this select **Audio** > **General** > **Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

30.1 Configuring SIP Interfaces

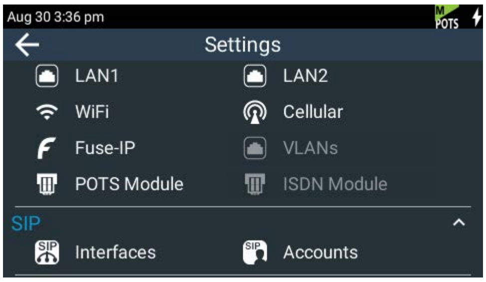
The codec supports dialing over two SIP interfaces simultaneously.

Important Notes:

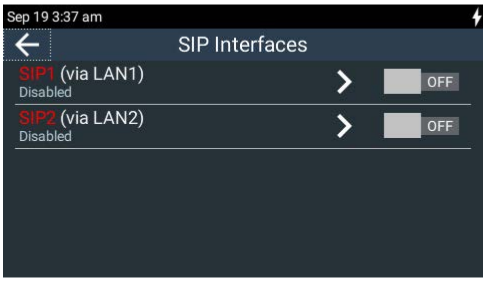
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configure to use **Wi-Fi** by default, which is mapped to the **Tertiary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN1** or **LAN2**.
5. **SIP1** and **SIP2** can however each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface is enabled.
7. Enter a public IP address in the **Public IP** menu if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.

To configure **SIP1** or **SIP2**:

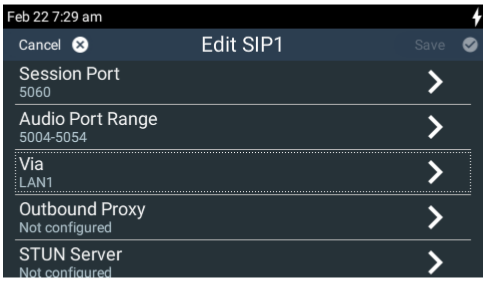
1. Press the **HOME** button to return to the **Home** screen and tap **Settings**.
2. Tap **SIP** to expand the menu and then tap **Interfaces**.



3. Tap **SIP1** to configure the first available SIP interface.



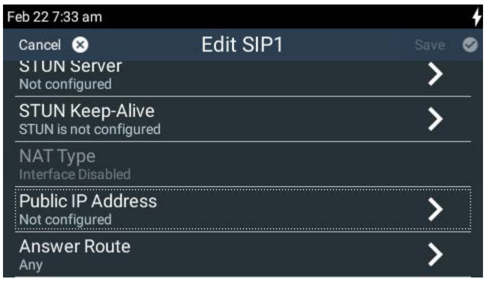
4. Tap to select and edit any setting, then tap to select the **Via**, i.e. the interface used to dial over **SIP1** or **SIP2**.



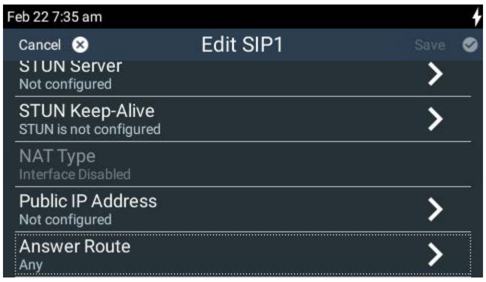
5. Tap **STUN Server** to enter server details if using STUN (Session Traversal of User Datagram Protocol). Note: UDP Port 3478 is the default port assigned.

6. The **NAT Type** used by the firewall is displayed after STUN has been configured and the interface has been enabled.

7. Tap **Public IP Address** if you want to dial over SIP from behind a firewall. Enter the public IP address and then configure port forwarding to route traffic to the codec's local IP address behind your firewall.

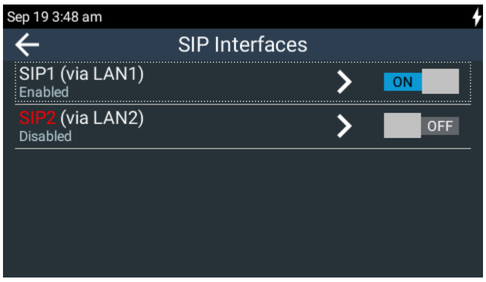


8. Tap **Answer Route** to route calls using this SIP interface to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value **Any** is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.



9. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to save all the configured SIP interface settings.

10. Tap to enable the interface you have just configured so that it can be used to make a SIP call.



30.2 Configuring SIP Accounts

Getting Started

Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, select the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. When you register an account with a SIP server you will be provided with:

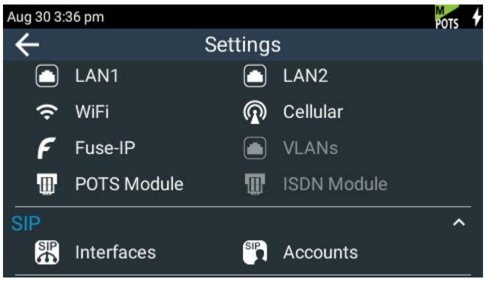
- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port
- Password
- Registration Timeout (this shouldn't need to be adjusted from the default setting).

Important Notes:

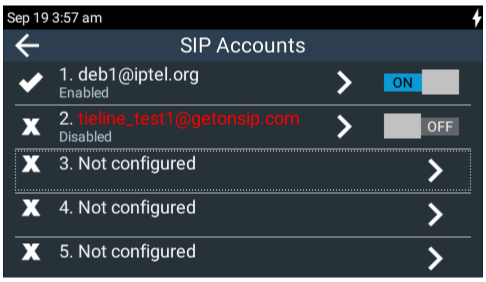
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** or **SIP 2**.
- Up to 6 SIP accounts can be added to the codec.
- It is also possible to add and register a SIP account to your codec using the HTML5 Toolbox Web-GUI.

Adding a SIP Account

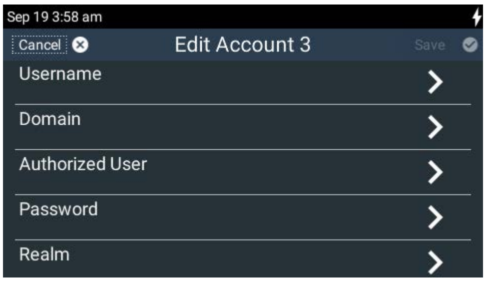
1. Press the **HOME** button to return to the **Home** screen and tap **Settings**.
2. Tap **SIP** to expand the menu and tap **Accounts**.



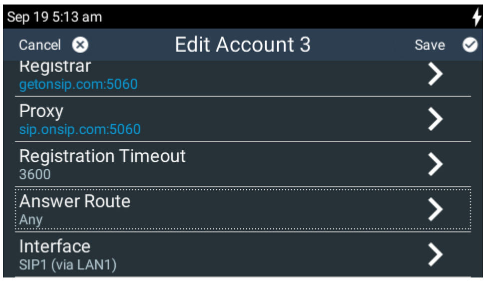
3. Tap **Accounts** to view the **SIP Accounts** screen. To add another SIP account, tap an account which is **Not configured**.



4. Tap each field in turn to enter SIP account credentials. Tap **Done** to confirm the details for each entry.

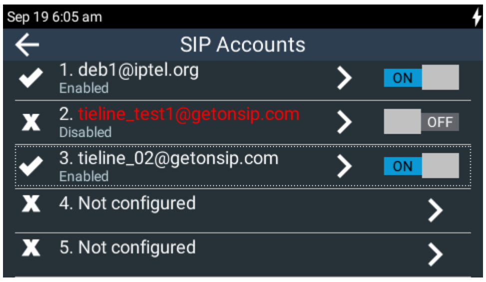


5. Select **Answer Route** to route incoming calls to this SIP account to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded program. If the default value **Any** is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.

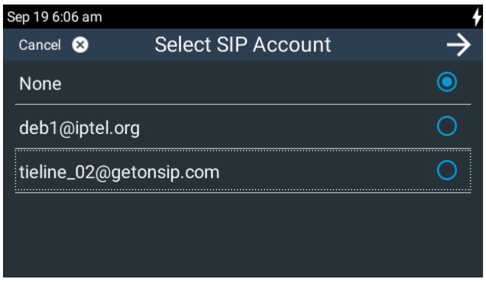


6. Tap **Save** in the top right-hand corner of the screen to confirm the account details for the new account.

7. The new account appears in the list of **SIP Accounts.** Tap the switch to enable or disable the newly created SIP account.



8. Once enabled, the SIP account can be selected when creating a new SIP connection via the **Home screen > Dialer > Destination.**



Confirming Account Registration

There are three symbols displayed in the codec next to an account which indicate SIP account registration status.

Symbol	Description
	Cross symbol indicates the account is not yet registered.
	Hourglass symbol indicates account registration is currently being attempted.
	Tick symbol indicates the account is registered to a SIP server.

The **SIP LED** adjacent to the **PPM METERS** on the front of the codec also displays the following SIP account registration states:

SIP LED State	Description
Not Illuminated	Indicates the account is not yet registered.
Orange Flashes	Codec is in the process of registering to an active SIP server account.
Solid Orange Illumination	The codec has been successfully registered to an active SIP account.

Troubleshooting SIP Registration

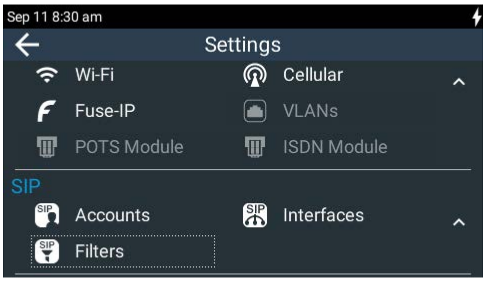
If a SIP account is not being registered please check the following:

1. Confirm all account registration information has been entered correctly.
2. Confirm the SIP **Interface** (**SIP1** or **SIP2**) configured in the account is enabled.
3. Verify that the **Via** selected in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **LAN1**, **LAN2** or **Wi-Fi**.

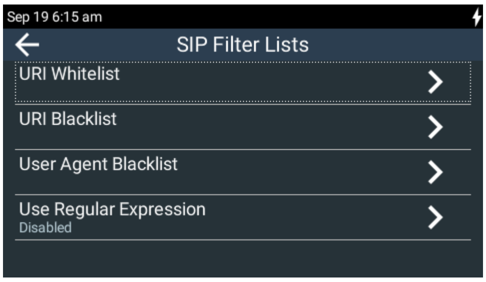
30.3 Adding SIP White and Black Lists

The SIP **Filters** menu allows filtering of SIP URIs and User Agents to provide greater security for your codec connections. For example, add trusted network codecs to the **URI Whitelist** in this panel and only codecs using these SIP URIs will be able to connect. It is also possible to add SIP URIs to the **URI Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec. These blacklists also filter unwanted traffic and increase the likelihood of rejecting unwanted traffic.

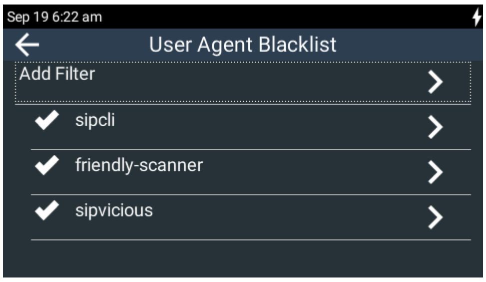
1. Press the **HOME** button to return to the **Home** screen and tap **Settings**.
2. Tap **SIP** to expand the menu and tap **Filters**.



3. Tap to select **URI Whitelist**, **URI Blacklist** or **User Agent Blacklist**..



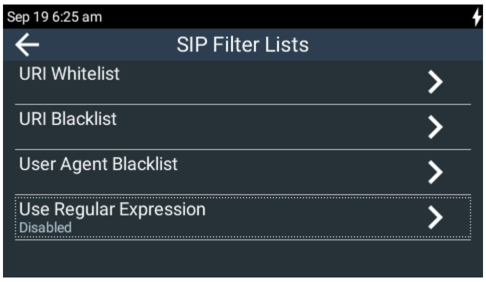
4. Tap **Add Filter** to add a new filter. Any filters previously added are also listed.



Important Note: Some codec manufacturers allow whitelisting of calls by 'User Agent.' Therefore, it may be necessary to enter a Tieline codec user agent into a non-Tieline codec to allow it to connect to a Tieline SIP-enabled codec. From firmware v2.16.xx the user agent in the codec is configured as "Tieline <Product Name> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68

Using Regular Expressions

To filter using regular expressions tap **Use Regular Expression**. Tap to toggle enabling and disabling this feature.



Important Note: Regular expressions should not use ^ and $ anchors because searches implicitly try to match anywhere in the line.

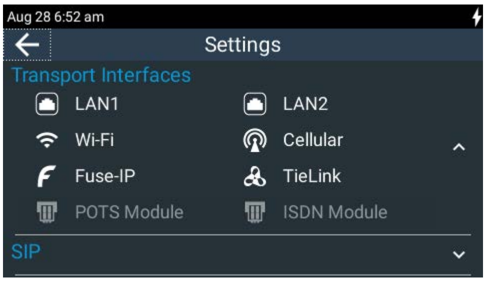
31 Other Touch Screen Configuration Tasks

The following sections explain how to configure a wide range of codec settings using the **TOUCH SCREEN**.

31.1 Configuring LAN/VLAN Settings

Verifying IP Address Details in the Codec

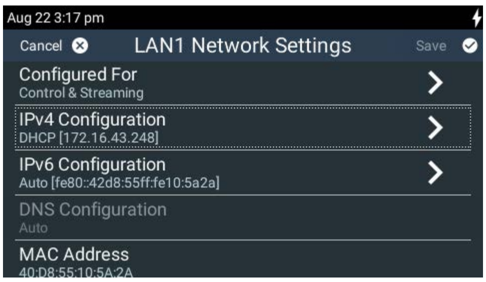
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then tap **LAN1** or **LAN2** .



3. Tap to select **Network Settings**.

Important Note: It is possible to turn off **LAN1** and/or **LAN2** to conserve power consumption when using the internal battery. If **LAN 2** is disabled, the codec needs to be rebooted to turn it back on.

4. IP address details and other unit details are listed.



Important Note: See Configuring IP Settings for more details about configuring IP connections using the HTML5 Toolbox Web-GUI. For further assistance with IPv4 or IPv6 network settings contact your IT Administrator.

Ethernet and VLAN Configuration Options

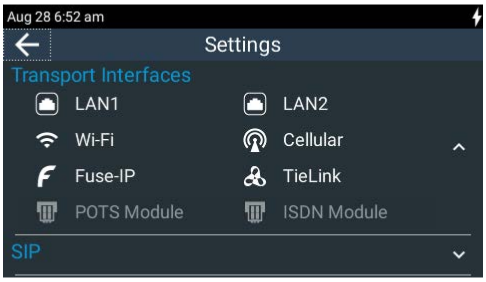
The codec features two physical Ethernet port interfaces (**LAN1** and **LAN2**) and up to four additional VLAN interfaces. VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for VLANs to be supported in your codec. As an example, if only one physical Ethernet interface is available, VLANs can be used to operate SmartStream PLUS or separate codec 'control' and 'streaming' functions if required.

Following are a range of options which can be configured in the **LAN** menus. After completing configuration tap **Save** in the top right-hand corner of the screen to apply the new settings.

Configure an IPv4 DHCP Address

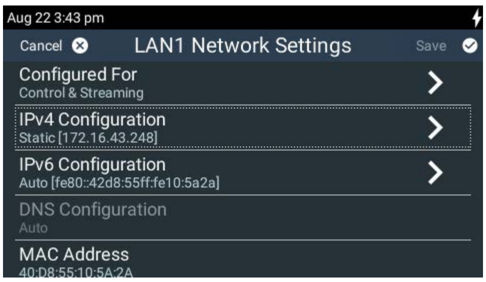
By default the codec is configured for DHCP-assigned IP addresses. DHCP addresses are automatically assigned and can change each time you connect to your Internet Service Provider, or by a router on your local area network (LAN).

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **LAN1** or **LAN2**. Note: VLANs need to be configured using the Toolbox HTML Web-GUI.

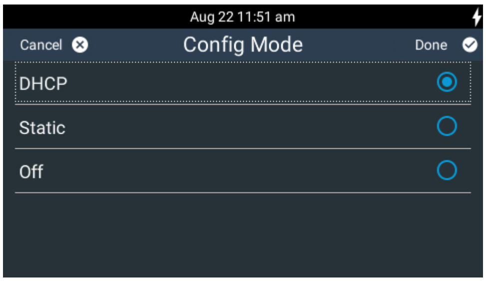


3. Tap **Network Settings**.

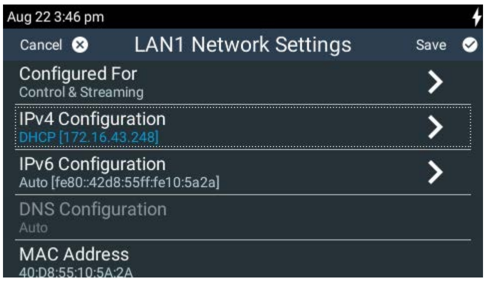
4. Tap **IPv4 Configuration**.



5. Tap to select **DHCP**.



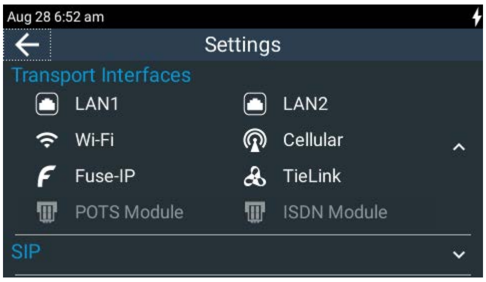
6. Tap **Save** in the top right-hand corner to reconfigure the codec for DHCP IP address configuration.



Configure a Static IPv4 Address

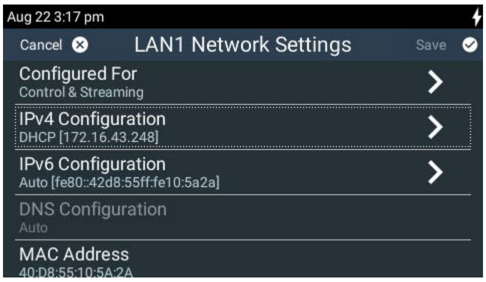
Static IP addresses are fixed addresses which are normally recommended for studio installations. Using a static IP address ensures remote codecs can connect reliably using the same IP address over time.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **LAN1** or **LAN2** or a **VLAN** interface.

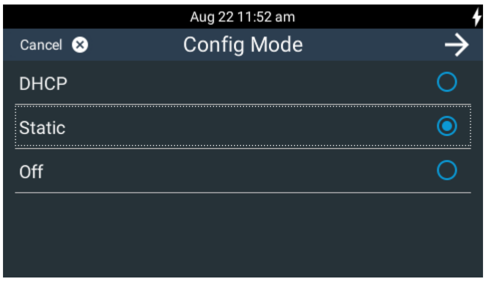


3. Tap **Network Settings**.

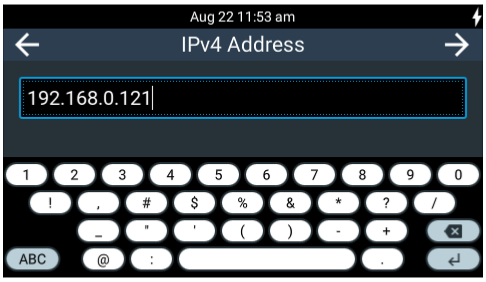
4. Tap **IPv4 configuration**.



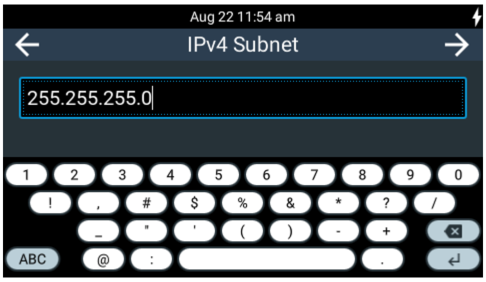
5. Tap to select **Static**.



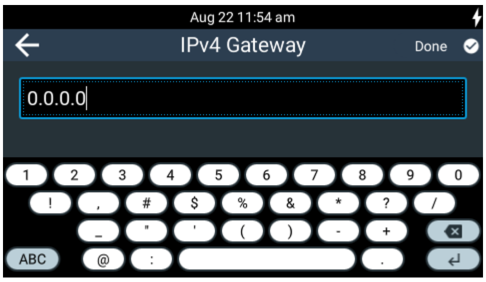
6. Use the on-screen keyboard to enter the static IP address into the codec, and then tap the **Right Arrow** in the top-right hand corner of the **TOUCH SCREEN**.



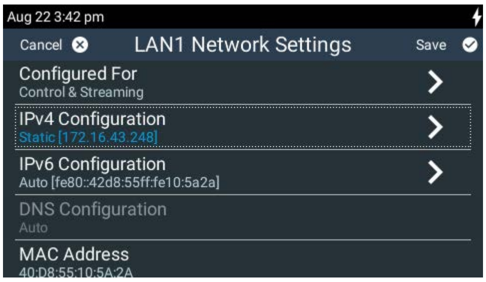
7. Enter the **IPv4 Subnet** mask into the codec, and then tap the **Right Arrow**.



8. Enter the **IPv4 Gateway** details as required, then tap **Done** in the top-right hand corner of the **TOUCH SCREEN**.



9. Tap **Save** in the top right-hand corner to reconfigure the codec for static IP address configuration.



IPv6 Address Assignment

There are three IPv6 settings available for each LAN port on the codec and any VLANs which are configured.

1. **Auto**: By default, an address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. **Manual**: Select to manually enter IPv6 address details.
3. **Off**: Select to ignore IPv6 address details.

Important Note: Select **Off** if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

By default the codec is configured to automatically receive IPv6 address information from an IPv6 enabled router. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **LAN1**, **LAN2** or a **VLAN** interface.
3. Tap **Network Settings**.
4. Tap **IPv6 Configuration**.
5. Tap **Auto**, **Manual** or **Off**.

Manual IPv6 Address Assignment

Select **Manual** using the previous procedure and enter information into the **IPv6 Address**, **IPv6 Prefix Size** and **IPv6 Gateway** fields to manually configure address details.

DNS Server

When a static IP address is specified it is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains** section within the Web-GUI. This feature can be turned on or off in the codec menu.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **LAN1**, **LAN2** or a **VLAN** interface.
3. Tap **Network Settings**.
4. Tap **DNS Settings**.
5. Tap the **On/Off** switch to toggle between **Manual** and **Auto** modes.
6. Tap to enter preferred DNS and domain details.

Link Mode Configuration

It is possible to configure the Ethernet, VLAN or Wi-Fi link speed (10/100/1000/Auto) and whether the an interface will operate in Full-Duplex or Half-Duplex modes.

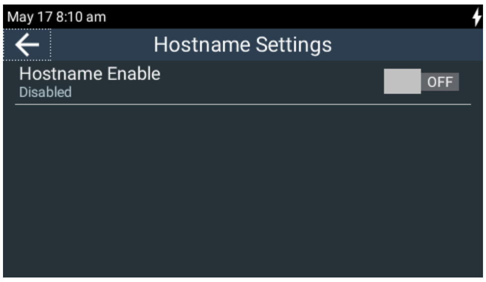
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **LAN1**, **LAN2** or **Wi-Fi**.
3. Tap **Network Settings**.
4. Tap **Ethernet Link Mode**.
5. Tap to select a preferred setting. Note: Default setting is **Auto**.

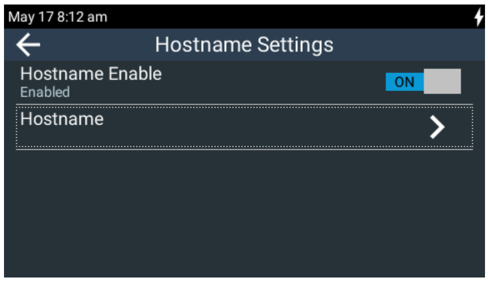
31.2 Configuring a Hostname

It is possible to assign a hostname to the codec to provide a flexible way of identifying the codec on a network.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **IP Options** menu and then tap **Hostname**.

3. Tap the **On/Off** button to enable **Hostname**.





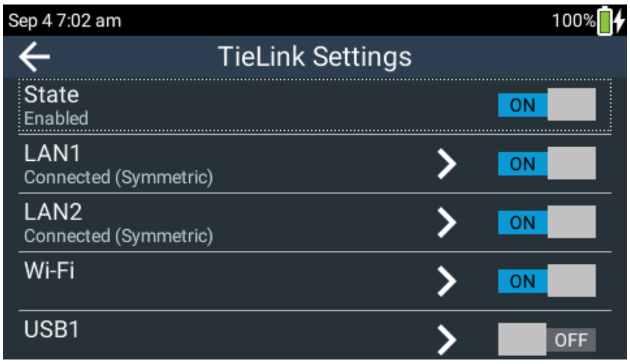
4. Enter the **Hostname** and tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

Important Note:
- Modifying hostname settings requires a codec restart before they take effect
- In the **Hostname** only enter the characters a-z, A-Z, 0-9 and - and the first or last character cannot be a hyphen/dash.

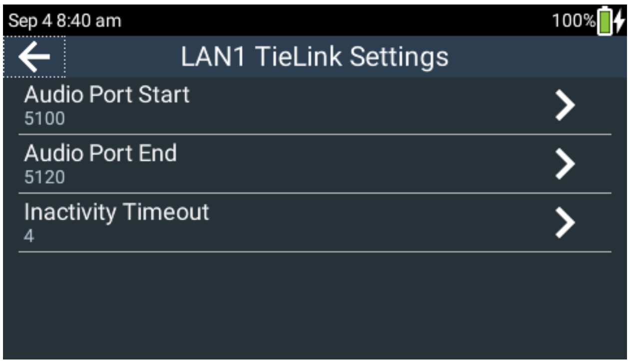
31.3 TieLink Configuration

TieLink settings can be adjusted in the codec using the **TOUCH SCREEN**.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then tap **TieLink**.
3. Connectivity to the TieLink Traversal Server and individual interfaces can be enabled in the TieLink Settings screen.



4. Tap to select each interface and then tap to adjust port and timeout settings.



5. Navigate to the bottom of the screen to adjust STUN server settings if required.

For more detailed information about TieLink configuration see Configuring TieLink Settings.

31.4 Selecting an Algorithm

The codec offers uncompressed linear PCM audio as well as aptX® Enhanced, LC-AAC, HE-AAC v.1 and HE-AAC v.2, AAC-LD, AAC-ELD, AAC-ELDv2, MPEG Layer 2, G.711 and G.722, Tieline Music and MusicPLUS algorithms.

Overview of Tieline Algorithms

1. The Tieline Music algorithm is optimized for audio bit rates as low as 19.2kbps with only a 20 millisecond encode delay. It offers 15 kHz mono from 24kbps to 48kbps.
2. Tieline MusicPLUS delivers up to 20 kHz mono from 48kbps upwards. It can also deliver up to 20 kHz stereo from 96kbps upwards, offering huge savings on your IP data bills and outstanding audio quality.

Overview of AAC Algorithms

AAC-LC

LC-AAC is optimized for audio bit rates of 64kbps per channel or higher using a sample rate of 48kHz. Tieline recommends using LC-AAC instead of HE-AAC if bandwidth of 64kbps or higher per channel is available, to optimize audio quality. If lower bandwidth than 64kbps is available consider using HE-AAC, Tieline Music or Tieline MusicPLUS.

AAC-HE

Codecs include both HE-AAC v.1 and HE-AAC v.2, which are optimized for low bit rate connections. Selection of HE-AAC v.1 and v.2 is automatically managed within the codec, so only **AAC-HE** is displayed on the screen. When used for mono connections, HE-AAC v.1 performs best at bit rates of 24kbps per channel or higher. HE-AAC v.1 is also used for stereo connections when audio connection bandwidth is 48kbps or higher.

HE-AAC v.2 is used for stereo connections when audio connection bandwidth is below 48kbps and is capable of delivering 15kHz quality stereo audio at audio bit rates as low as 24kbps.

A sample rate of 32kHz is used in the codec's default profiles to achieve ultra-low bit rate connections, but this is adjustable to 44.1kHz or 48kHz if required.

AAC-LD

AAC-LD (Low Delay AAC), AAC-ELD (Enhanced Low Delay AAC) and AAC-ELD v 2 are optimized for low latency real-time communication. AAC-LD is suited to bit rates of 96kbps or higher for stereo audio.

AAC-ELD

AAC-ELD is optimized for high quality stereo connections from 48 - 96kbps and performs better at these bit rates when compared with AAC-LD.

AAC-ELD v 2

For stereo connections below 48kbps AAC-ELD v2 will deliver better performance than AAC-ELD down to 24kbps.

Overview of aptX Enhanced Audio Coding

aptX® Enhanced audio coding is used by thousands of radio stations to deliver very low delay audio for IP broadcasts and is ideal for high quality studio-to-transmitter links and audio distribution. It delivers outstanding audio quality with exceptionally low delay across a range of IP networks.

32kHz or 48kHz sample rates are available at either 16 bit or 24 bits per sample. aptX Enhanced has a minimum connection bit rate of 128kbps per channel and offers 10Hz to 24kHz frequency response. 24 bit, 48kHz aptX Enhanced at the maximum bit rate of 576kbps delivers >120dB of dynamic range.

Mono 16 bit, 32 kHz aptX® Enhanced is supported at 128kbps over ISDN using 2 B channels.

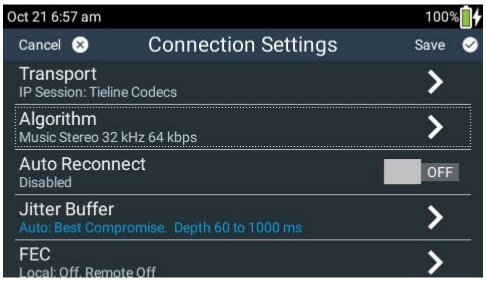
Overview of Opus Algorithm

Opus is a highly versatile open source audio coding algorithm. It incorporates technology from the well-known SILK and CELT codecs to create a low latency speech and audio codec. It is a variable bit rate algorithm ideal for live broadcast situations because of its capacity to deliver high quality, real-time Audio over IP (AoIP) at low bit rates. Visit http://www.opus-codec.org for more info. There are three Opus algorithm configurations available:

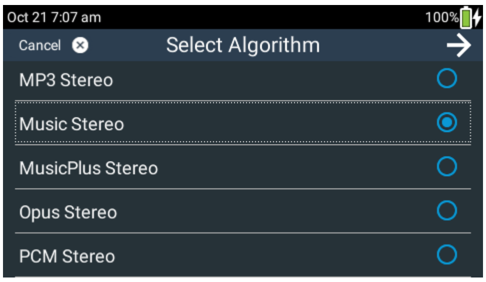
Algorithm	Recommended connection for on-air use
Opus Voice	High quality low bit rate remotes (9.6kbps -64kbps)
Opus Mono	Very high quality mono remotes, STLs and audio distribution (48kbps - 128kbps)
Opus Stereo	Very high quality stereo remotes, STLs and audio distribution (64kbps - 256kbps)

Configuring an Algorithm in the Codec

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap **Algorithm** to reconfigure the algorithm for the selected **Transport** (IP, SIP, ISDN, POTS).



4. Tap to select an algorithm, sample rate (some algorithms like G.722, MP2 and aptX Enhanced) and bit-rate, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**. Note: In most situations Tieline Music is the best algorithm to select and this is configured by default.



How do I choose the right algorithm?

The algorithm you select will not only affect the quality of the broadcast but it will also contribute to the amount of latency or delay introduced. For example, if MP2 algorithms are used, program delays will be much longer than when using Tieline Music or MusicPLUS algorithms. This is due to the additional inherent encoding delays involved when using MP2 algorithms. This can be a major consideration for live applications that integrate remotes into a broadcast. The algorithm you select to connect with will also depend upon:

- The codecs to which you are connecting (Tieline versus non-Tieline)
- Whether you are connecting using SIP or not.
- The uplink bandwidth capability of your broadband connection.

Important Notes: Music and MusicPLUS algorithms cannot be used over SIP connections. Use MP2 algorithms at 64kbps mono or 128kbps stereo for high quality connections when using SIP, or use G.711 and G.722 if required. Tieline G3 codecs do not support connections using AAC, aptX Enhanced and Opus algorithms and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.

It can be a good idea to listen to the quality of your program signal using each algorithm and to see how it sounds when it is sent at different connection bit rates (as well as different FEC and jitter-buffer millisecond settings). This will assist you to determine which is the best algorithm setting for the connection you are setting up. Please see the following table for details on the connection requirements of the different algorithms available.

Algor-ithm	**Audio Band-width**	**Algor-ithmic Delay**	**IP bit rate per channel**	**IP over-head per connection**	**Audio Quality and Features**	**Recommended applications for on-air use**
Linear/PCM (Uncom-pressed)	16/24 bit up to 45kHz	0ms	sample rate x bits per sample x no. channels; 512kbps minimum	80kbps	• Full bandwidth, perfect audio quality for voice and music • No error concealment/correction or artifacts	• Extremely high quality uncompressed linear PCM audio for STLs and audio distribution. • Ideal for fiber or high bandwidth links.
Tieline Music	Up to 15kHz	20ms	24 kbps minimum	16kbps	• High quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth (e.g. POTS or 3G wireless) • Suitable when bidirectional communication between announcers is required • Deliver 15kHz stereo over 1 x 64kbps ISDN B Channel.
Tieline Music-PLUS	Up to 22kHz	20ms	48 kbps minimum (Optimized for 64kbps per audio channel)	16kbps	• Very high quality voice and music • Very low delay at low to moderate bit-rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel • Suitable when bidirectional

						communication between announcers is required
G.711	3kHz	1ms	64kbps minimum	80kbps	• Low quality 3kHz POTS phone quality audio • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Low quality and used generally for compatibility
G.722	7kHz	1ms	64kbps minimum	80kbps	• Good quality 7kHz voice • Better quality than a standard POTS phone call • Very low delay at moderate bit rates	• Highly compatible with other brands of audio codec • Good voice quality audio for remotes and other voice quality applications
MPEG Layer 2	Up to 22kHz	24 to 36ms	64kbps minimum	8.5 - 13.3kbps	• Very high quality voice and music • Low to moderate delay at moderate to high bit rates	• Highly compatible with other brands of audio codec • Very high quality audio for remotes, STLs and audio distribution
MPEG Layer 3	Up to 15kHz	100ms	64kbps	8.5 - 13.3kbps	• High quality voice and music • Moderate bit rates • High delay	• High quality remotes, STLs and audio distribution • Use when bidirectional communication between announcers is not required
LC-AAC	Up to 15kHz	64ms	64kbps	15kbps	• High quality voice and music at lowest bit rate; better quality at higher bit rates • Moderate delay at moderate to high bit rates	• Voice or music remotes as well as STLs and audio distribution where some delay is tolerable • Tieline Music or MusicPLUS deliver lower delay
HE-AAC v.1	Up to 15kHz	128ms	48kbps	7.4kbps	• High quality voice and music at the lowest bit rate; better quality at higher bit rates • Low to Moderate bit rates • High delay	• Live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth • Use when bidirectional communication between announcers is not required
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16kbps (Mono); 24kbps (stereo)	7.4kbps	• High quality voice and music • Low bit rates • High delay	• Used for DAB+ radio streaming • Ideal for low bit rate remotes • Use when bidirectional communication between announcers is not required
AAC-LD	Up to 20kHz	20ms at 48kHz	48kbps minimum	30kbps	• Very high quality voice and music • Very low delay at low to moderate bit rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel requiring • Suitable when bidirectional communication

						betw een announcers is required
AAC-ELD	Up to 20kHz	15-30ms	24 kbps minimum	15-30kbps	• Very high quality voice and music • Very low delay at low bit rates	• Great for live voice or music remotes • Suitable w hen bidirectional communication betw een announcers is required
AAC-ELDv.2	Up to 20kHz	35ms	Pending release	Pending release	• High quality voice and music • Low delay at low bit rates	• Great for live voice or music remotes w here limited connection bandw idth is available • Suitable w hen bidirectional communication betw een announcers is required
aptX Enhanced	10Hz-24kHz	2.5ms at 48kHz	128kbps minimum (16bit; 32kHz) to 288kbps (24bit;48kHz)	80kbps	• Very high quality voice and music • Extremely low delay at high bit rates • Highly cascade resilient	• Ideal for STLs and audio distribution w here high connection bandw idth is available and very low delay is highly desirable. • Resilient w ith multiple encodes/decodes w hen required
Opus	4Hz-20kHz	20ms	9.6-256kbps	16kbps	• Very high quality voice and music • Very low delay at low bit rates	• "Opus Voice" is ideal for high quality, and low delay voice quality remotes at extremely low bit rates. • "Opus Mono" and "Opus Stereo" are perfect for high fidelity remotes, STLs and audio distribution at higher bit rates

Algorithm Selection Guide

Algorithm	Very Low Delay	Moderate to High Delay	Excellent Performance at Low Bit rates	Preferred for Live Remotes	Preferred for STLs and Audio Distribution	Highly Compatible with other Codecs
Linear/PCM	✓				✓	✓
Opus	✓		✓	✓	✓	✓
Tieline Music	✓		✓	✓		
Tieline MusicPLUS	✓		✓	✓	✓	
aptX Enhanced	✓				✓	✓
LC-AAC		✓			✓	✓
HE-AACv1		✓			✓	
HE-AACv2		✓	✓	✓*		
AAC-LD	✓			✓	✓	
AAC-ELD	✓		✓	✓		
MPEG Layer 2	✓				✓	✓
MPEG Layer 3		✓				✓
G.722	✓					✓
G.711	✓					✓

***** Use with caution for remotes due to high delay; not suitable when bidirectional communications is required.

IP Connection Bit rates Supported

Algorithm	Sample Rate	Genie and Merlin Connection Bit Rates Supported over IP																										
AAC HE Mono	32kHz				16	20		24		32	40	48	56															
AAC HE Mono	44.1kHz							24		32	40	48	56	64														
AAC HE Mono	48kHz							24		32	40	48	56	64														
AAC HE Stereo	32kHz							24		32	40	48																
AAC HE Stereo	44.1kHz										40	48	56	64														
AAC HE Stereo	48kHz										40	48	56	64														
AAC LD Mono	32kHz										40	48	56	64														
AAC LD Mono	44.1kHz												56	64	96	112	128	160		192								
AAC LD Mono	48kHz												56	64	96	112	128	160		192								
AAC LD Stereo	32kHz													64	96	112	128											
AAC LD Stereo	44.1kHz														96	112	128	160		192	256			320		384		
AAC LD Stereo	48kHz														96	112	128	160		192	256			320		384		
AAC ELD Mono	32kHz							24		32	40	48	56	64														
AAC ELD Mono	44.1kHz							24		32	40	48	56	64														
AAC ELD Mono	48kHz							24		32	40	48	56	64														
AAC ELD Stereo	32kHz											48		64	96	112	128											
AAC ELD Stereo	44.1kHz											48	56	64	96	112	128											
AAC ELD Stereo	48kHz											48	56	64	96	112	128											
AAC LC Mono	32kHz									32	40	48																
AAC LC Mono	44.1kHz											48	56	64	96		128	160										
AAC LC Mono	48kHz											48	56	64	96		128	160										
AAC LC Stereo	32kHz													64	96													
AAC LC Stereo	44.1kHz													64	96	112	128	160		192	256			320				
AAC LC Stereo	48kHz													64	96	112	128	160		192	256			320				
MP2 Mono	24kHz													64														
MP2 Mono	32kHz													64	96	112	128	160		192								
MP2 Mono	44.1kHz													64	96	112	128	160		192								
MP2 Mono	48kHz													64	96	112	128	160		192								
MP2 Stereo	32kHz															112	128	160		192	256							
MP2 Stereo	44.1kHz															112	128	160		192	256			320		384		
MP2 Stereo	48kHz															112	128	160		192	256			320		384		
MP2 J-Stereo	32kHz															112	128			192								
MP2 J-Stereo	44.1kHz															112	128			192								
MP2 J-Stereo	48kHz															112	128			192								
MP3 Mono	32kHz													64	96	112	128	160										
MP3 Mono	44.1kHz													64	96	112	128	160										
MP3 Mono	48kHz													64	96	112	128	160										
MP3 Stereo	32kHz															112	128	160		192	256							
MP3 Stereo	44.1kHz																128	160		192	256			320				
MP3 Stereo	48kHz																128	160		192	256			320				
aptX Enhanced Mono	32kHz																128			192								
aptX Enhanced Mono	44.1kHz																		176.4			264.6						
aptX Enhanced Mono	48kHz																			192			288					
aptX Enhanced Stereo	32kHz																				256					384		
aptX Enhanced Stereo	44.1kHz																								352.8		529.2	
aptX Enhanced Stereo	48kHz																									384		576
Opus Voice	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128											
Opus Mono	48kHz											48		64	96	112	128											
Opus Stereo	48kHz													64	96	112	128	160		192	256							
Music Mono	32kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music Stereo	32kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music PLUS Mono	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							
Music PLUS Stereo	48kHz	9.6	12	14.4	16.8	19.2	21.6	24	28.8	33.6	38.4	48		64	96	112	128	160		192	256							

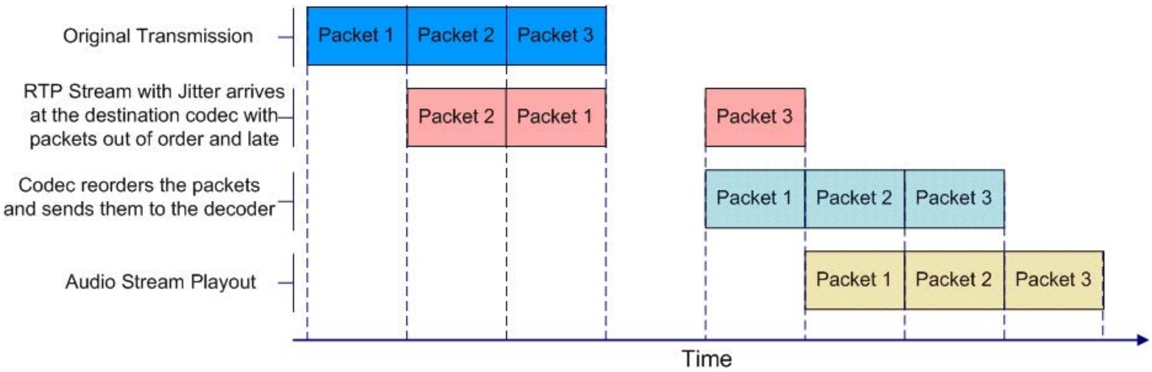
ISDN Encoding Options

The codec supports ISDN connections using the following algorithms and B Channel assignments.

ISDN Encoding	1B	2B
E-AptX Mono 16bit 32KHz		✓
Music Mono	✓	✓
Music Stereo	✓	✓
Music Plus Mono	✓	✓
Music Plus Stereo	✓	✓
MP2 Mono 32KHz	✓	✓
MP2 Stereo 32KHz		✓
MP2 Mono 48KHz	✓	✓
MP2 Stereo 48KHz		✓
MP2 J-Stereo 32KHz		✓
MP2 J-Stereo 48KHz		✓
G.711	✓	
G.722	✓	

31.5 Configuring the Jitter Buffer

Jitter, also known as latency or delay, is the amount of time it takes for a packet of data to get from one point to another. A jitter buffer is a temporary storage buffer used to capture incoming data packets. It is used in packet-based networks to ensure the continuity of audio streams by smoothing out packet arrival times during periods of network congestion. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, i.e. LAN versus wireless networks. The concept of jitter buffering is displayed visually in the following image.



Jitter-buffer management is encompassed within Tieline's SmartStream IP technology which can:

- Remove duplicate packets.
- Re-order packets if they arrive out-of-order.
- Repair the stream in the event of packet loss (error concealment).
- Manage delay dynamically based on current network congestion.
- Manage forward error correction (FEC).

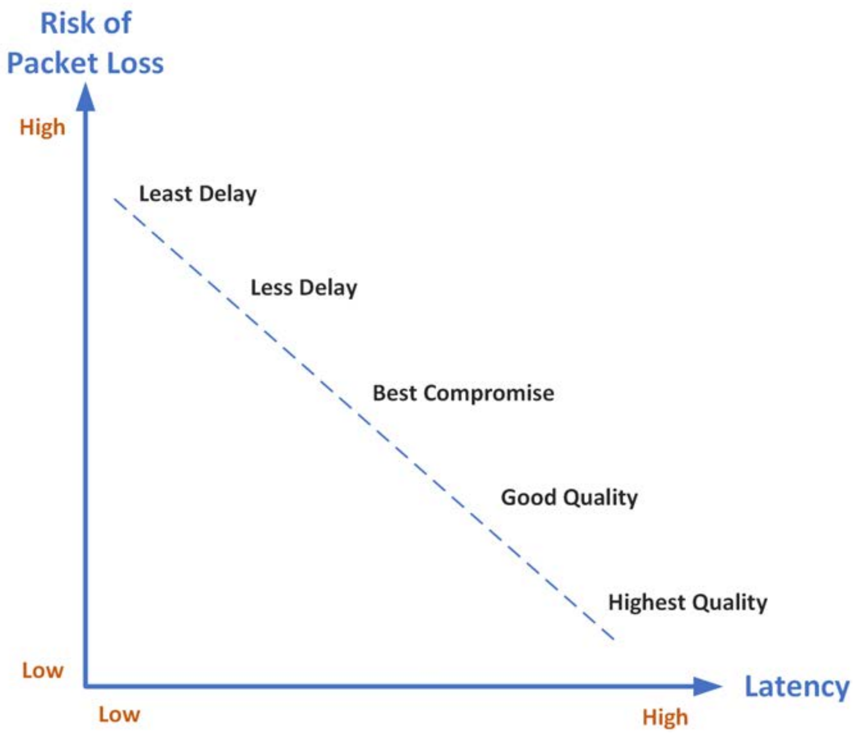
With Tieline codecs you can configure either a fixed or automatic jitter buffer and the settings you use depend on the IP network over which you are connecting. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to prevailing IP network conditions to provide continuity of audio streaming and minimize delay.

A fixed jitter buffer is preferable over satellite or high latency connections to ensure continuity of signals.

CAUTION: If a Tieline codec connects to a device that is using non-compliant RTP streams then the last fixed setting configured in the codec will be enabled (default is 500ms). Non-compliant devices include some other brands of codec, web streams and other devices.

Tieline 'Auto Jitter Buffer' Settings

The following automatic jitter buffer settings range from the most aggressive "Least Delay" setting, which endeavors to minimize delay as much as possible while adapting to prevailing network conditions, to the "Highest Quality" setting, which is the most conservative delay setting to minimize packet loss when packet latency is not as critical. Best compromise is the most popular setting used by most broadcasters as it achieves an excellent balance that minimizes latency for bidirectional communications, as well as minimizing packet loss over most wired and wireless IP networks.



Jitter Buffer Settings and Relationship of Latency and Packet Loss

Least Delay: This setting attempts to reduce the jitter buffer to the lowest possible point, while still trying to capture the majority of data packets and keep audio quality at a high level. This setting is the most aggressive in adapting to prevailing conditions, so the jitter buffer may vary more quickly than with the other settings. It is not recommended in situations where jitter variation is significant, or occurs in bursts. (E.g. cellular/multi-user wireless networks). It is best for stable and reliable links such as dedicated or lightly-loaded WAN/LANs.

Less Delay: This setting lies between "Best Compromise" and "Least Delay". It may assist in reducing latency over a connection without incurring packet loss.

Best Compromise: This default setting is the midpoint between the jitter buffer settings applicable for "Highest Quality" and "Least Delay." It is designed to provide the safest level of good audio quality without introducing too much latency. In most situations it will deliver very high quality and low delay to support live bidirectional communications over cellular and wireless networks.

Good Quality: This setting lies between "Best Compromise" and "Highest Quality." It may assist in achieving higher quality connections without incurring extreme delays in transmission or significant packet loss.

Highest Quality: This setting is the most conservative in terms of adapting to prevailing network conditions to reduce delay. The jitter-buffer will remain higher for a longer period after a jitter spike is detected – just in case there are more spikes to follow. This setting is best to use where audio quality is the most important factor and delay is not as critical. Unless delay is irrelevant, this setting is not recommended over peaky jitter networks (e.g. cellular networks) and is best used on more stable networks where fluctuating jitter bursts are not common.

Jitter Depth

The jitter **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000ms** is a good starting point for most networks. It may be necessary to increase the maximum auto jitter latency setting for networks experiencing higher packet latency, or the minimum depth depending on the reliability of the network.

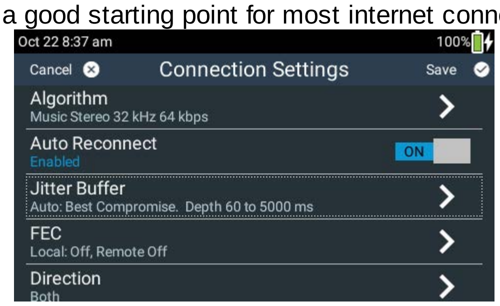
Which Algorithms can use Automatic Jitter Buffering?

The following table provides an overview of which algorithms are capable of using the automatic jitter buffer feature over SIP and non-SIP connections.

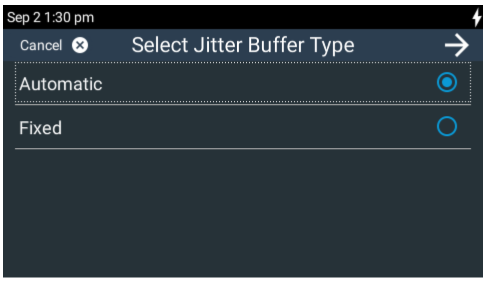
Algorithm	Non-SIP Connections	SIP Connections
PCM Linear (Uncompressed)	✗	✗
Tieline Music	✓	✗
Tieline MusicPLUS	✓	✗
G.711	✗	✓
G.722	✗	✓
MPEG Layer 2	✓	✓
MPEG Layer 3	✓	✗
LC-AAC	✓	✓
HE-AAC v.1	✓	✓
HE-AAC v.2	✓	✓
AAC-LD	✗	✗
AAC-ELD	✗	✗
Opus	✓	✓
aptX Enhanced	✗	✗

Configuring Automatic Jitter Buffering

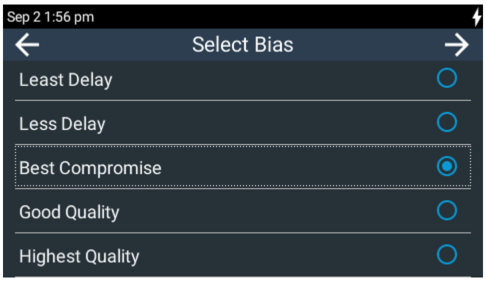
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap **Jitter Buffer** to select a different automatic jitter buffer setting for your connection, or to enter a fixed buffer setting in milliseconds (maximum 5000 ms). The default **Auto, Best Compromise** setting is a good starting point for most internet connections.



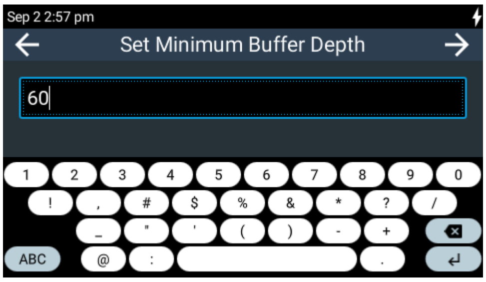
4. Tap **Automatic** to select Automatic Jitter Buffering.



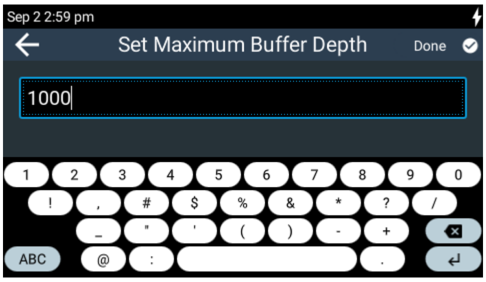
5. Tap to select a preferred setting.



6. Use the on-screen keyboard to set the minimum jitter buffer depth. Note: The **Depth** setting allows you to select predetermined minimum and maximum jitter settings within the auto jitter buffer's minimum and maximum jitter limitations. The default setting of **60 to 1000**ms is a good starting point for most networks. Note: the codec will not allow you to enter an invalid minimum or maximum buffer depth setting.



7. Use the on-screen keyboard to set the maximum jitter buffer depth, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN** to complete configuration.

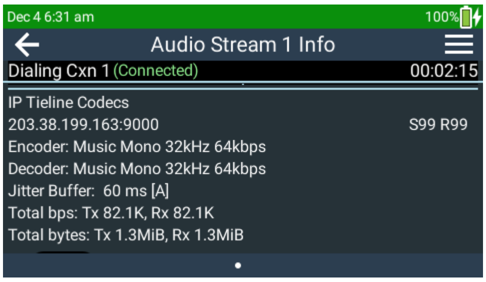


How to get the Best Auto Jitter Buffer Results

When configuring automatic jitter buffer settings, establish the IP connection for a while before 'going live', to allow the codec to evaluate prevailing network conditions. The initial jitter buffer setting when a codec connects is 500ms and it is kept at this level for the first minute of connection (as long as observed delay values are lower than this point).

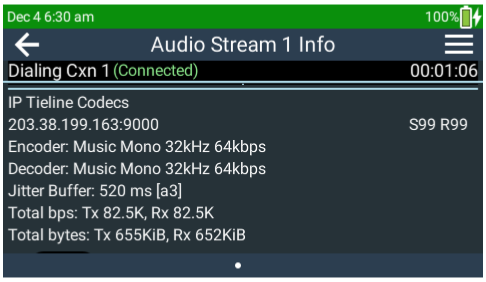
After the initial connection period the jitter buffer is adapted to suit the current network conditions and is usually reduced. Establish a connection for at least 5 minutes prior to broadcasting, so that the codec has been provided with enough jitter history to ensure a reliable connection.

There are five jitter buffer states. Jitter buffer and connection status information can be viewed in an active connection by pressing the **HOME** button to return to the **Home** screen and tap **Connections**. Then swipe left once to view the **Connection Status** screen with **Jitter** and **Send/Return** values.



The first four stages are observed in “auto” jitter buffer mode.

1. **Stabilization period (a1):** A few seconds at the start of a connection where no action is taken at all while the establishment of a stable connection means analysis of jitter data is not valid.
2. **Stage 2 (a2):** A compatibility check to ensure the RTP connection is compliant and RTP clocks are synchronized enough to perform jitter analysis.
3. **Stage 3 (a3):** If the compatibility check is successful, this is the analysis hold-off period. During a minute, the jitter buffer is held at a safe, fixed value of 500ms while enough history is recorded to start jitter buffer adaptation.

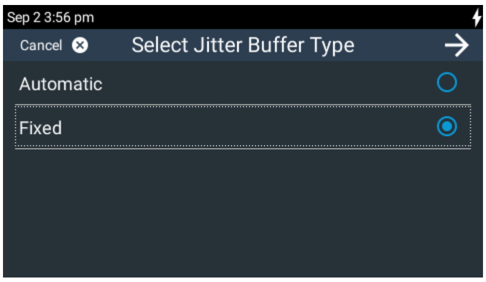


4. **Stage 4 “live” (A)**: This is where the codec determines it is safe enough to start broadcasting using the auto-jitter buffer level. We recommend running the codec for a few more minutes to obtain a more comprehensive history of the connection’s characteristics.
5. **Fixed (F):** This state is displayed if the jitter buffer is fixed.

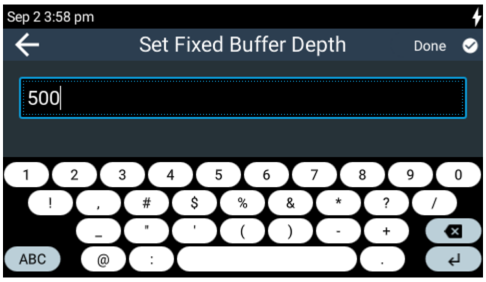
Fixing Jitter Buffer Settings

The default jitter-buffer latency in Tieline codecs is 500 milliseconds. This is a very reliable setting that will work for just about all connections. However, this is quite a long delay and we recommend that when you set up an IP connection you test how low you can configure the jitter-buffer in your codec before the broadcast.

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap **Jitter Buffer**.
4. Tap **Fixed** to select fixed jitter.



5. Use the on-screen keyboard to set the fixed jitter buffer depth, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN** to complete configuration. Note: Recommended maximum fixed jitter limits are as follows:
   - 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
   - 2,500ms for AAC ELD, AAC LD.
   - 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.



Configure the Jitter Buffer on the Answering Codec

Create an answering program to independently configure the jitter buffer settings on an answering codec. This will ensure specific fixed or auto jitter settings can be configured to suit the IP network to which the codec is connected. To do this:

1. Create a new answering program on the answering codec.
2. Configure preferred jitter buffer settings in this answering program.
3. Lock the answering program in the codec.

Please note that with the implementation of EBU N/ACIP 3368 SIP configuration the dialing codec can configure the jitter setting on the answering codec. This will override the jitter buffer settings in a locked and loaded answering program in a Tieline codec.

When manually configuring the jitter-buffer delay in a codec it is necessary to think carefully about the type of connection you will be using. Following is a table displaying rule of thumb settings for configuring jitter-buffer delays into your codec.

Connection	Jitter-Buffer Recommendation
Private LAN	60 milliseconds
Local	100 - 200 milliseconds
National	100 - 300 milliseconds
International	100 – 400 milliseconds
Wireless Network	250 - 750 milliseconds
Satellite IP	500 - 5000 milliseconds

Important Note: The preceding table assumes Tieline Music is the algorithm in use. Do not use PCM (linear uncompressed) audio over highly contended DSL/ADSL connections without enough bandwidth to support the high connection bit rates required.

Relationship between the Jitter Buffer and Forward Error Correction (FEC)

If forward error correction is configured then additional data packets are sent over a connection to replace any lost data packets. There is no need to modify jitter buffer settings if you are sending FEC data, only if you are receiving FEC data.

The jitter buffer depth on the receive codec needs to be increased if FEC is employed. We recommend you add 100ms to the fixed jitter buffer on a codec receiving FEC at a setting of 20% and 20ms at a setting of 100%.

Tieline's auto jitter buffer detects the amount of FEC that is being used and automatically compensates to increase the codec jitter buffer when this feature is enabled.

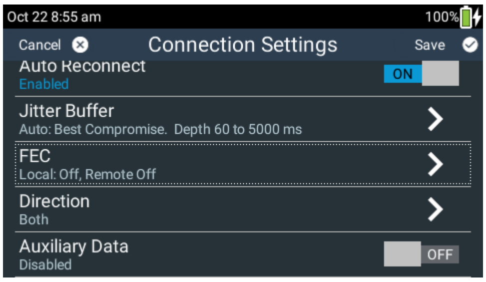
31.6 Configuring Forward Error Correction

Forward Error Correction (FEC) is designed to increase the stability of UDP/IP connections in the event that data packets are lost. FEC works by sending a secondary stream of audio packets over a connection so that if your primary audio stream packets are lost or corrupted, then packets from the secondary stream can be substituted to replace them. The amount of FEC required depends on the number of data packets lost over the IP connection.

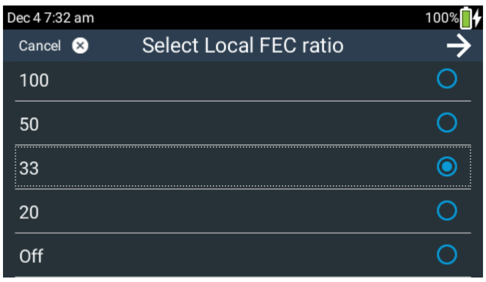
Both the local and remote codec FEC settings can be configured in your codec before dialing. These settings can also be changed 'on the run' while the codecs are connected. FEC should only be used if the Send/Return link quality percentage displayed on the codec is below 99, as it is of no benefit otherwise.

Configuring FEC (Forward Error Correction)

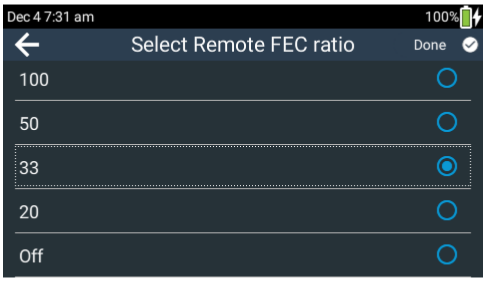
1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap **FEC** to configure local and remote forward error correction settings.



4. Tap to select the local codec's FEC ratio.



5. Tap to select the remote codec's FEC ratio, or tap **Done**.



The four FEC settings are outlined in the following table with their bit rate ratios.

FEC Setting	Bit rate Ratios	Connection Use
100% (Lowest delay)	A simultaneous dual-redundant stream (1:1 ratio) is sent from the codec. Twice the connection bit rate is required to operate the codec using the 100% setting. E.g. if your connection is 14,400kbps, you will require an additional 14,400 kbps of bandwidth to allow for the FEC data stream.	Recommended to be used over wireless and international connections.
50%	Additional data is sent using FEC in a ratio of 2:1.	Recommended for international & national connections
33%	Additional data is sent using FEC in a ratio of 3:1.	Recommended for national and local connections.
20% (Highest delay)	Additional data is sent using FEC in a ratio of 5:1.	Recommended for local and LAN connections.
Off	FEC is off in the codec and the connection bandwidth is equal to the connection bit rate setting in the codec.	Recommended for wired LAN connections & managed T1 & E1 connections for STLs that have connections that aren't shared & have quality of service (QoS).

Important Note: FEC can only be programmed for use with the Music and MusicPLUS algorithms.

How does FEC work?

If you configure a FEC setting of 20% and you are losing one packet in every five sent, the lost packet will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

Note: There is an inverse relationship between FEC settings and the jitter-buffer millisecond setting that you use for IP connections.

So why not use 100% FEC every time? The answer is because you need twice the bit rate to achieve full redundancy and depending on the link conditions, this could potentially cause more dropouts because of network congestion than it fixes. Here is a simple rule to remember: Your maximum uplink speed is all the bandwidth you have to play with. As a rule of thumb, try not to exceed more than 80% of your maximum bandwidth. If your link is shared, be even more conservative.

You should also consider the remote end too. What is their maximum upload speed? Is the connection shared at either end? Your bit rates, FEC settings and buffer rates must be pre-configured at both ends before you connect, so it's always better to set your connection speed and balance your FEC according to the available uplink bandwidth at each end for best performance.

As an example, if you want 15 kHz mono (using the Tieline Music Algorithm) you will need at least a 24kbps connection for audio. Adding 100% FEC will add another 24kbps making your bit rate 48kbps plus some overhead of around 10kbps is required. If you're on a 64kbps uplink, you should consider reducing your FEC to minimize the likelihood of exceeding your bandwidth capacity.

Here is another example, if you want 15 kHz stereo, you need at least 56kbps for the audio. 100% FEC requires at least 112kbps and 50% FEC requires at least 84kbps. If your uplink speed is 256kbps and you're on a shared connection, then choosing a lower FEC setting of 20%-33% may deliver better results.

Conserving Bandwidth with FEC

There is a trade-off between the quality and the reliability of an IP connection – particularly when FEC is activated on your codecs. However, it is possible in certain situations to set different FEC on each codec to match connection bandwidth requirements at either end of the link, conserve bandwidth and create more stable IP connections.

For example, if your broadcast is a one-way broadcast from a remote site, i.e. you are not using the return path from the studio, or only using it for communications purposes, it is possible to reduce or turn off FEC at the studio codec. This effectively reduces the bandwidth required over the return link (communications channel) and increases the overall bandwidth available for the incoming broadcast signal from the remote site.

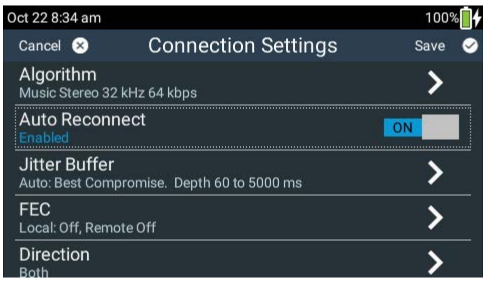
31.7 Configuring Auto Reconnect

Auto Reconnect is disabled by default. When enabled, the dialing codec attempts to reconnect when data is not detected over a connection.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until the **DISCONNECT** button is pressed, **Disconnect** is tapped in the **Connections** screen, or the audio stream is disconnected in the HTML5 Toolbox Web-GUI.

Auto Reconnect using IP, ISDN and POTS

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap the **On/Off** button for **Auto reconnect** to toggle between enabling and disabling **Auto reconnect** (default setting **Off**). This setting configures the codec to automatically reconnect if the connection is temporarily lost.



4. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to save the setting.

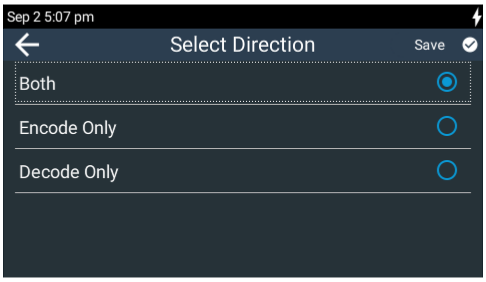
31.8 Configuring Encode/Decode Direction

By default the codec by is configured to both encode and decode data. However, it is possible to configure the codec to either encode or decode audio data only. This is useful for:

- Conserving connection bandwidth when you only need to stream data in one direction.
- Lowering data costs.
- Increasing overall connection reliability over low bit rate connections.

Configure the transmitting codec to encode only and the receive codec to decode only when using this feature. To adjust this setting:

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap **Direction** if you want to save data and configure the codec to either **Encode Only** or **Decode Only**.



4. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to save the setting.

31.9 Headphone Settings

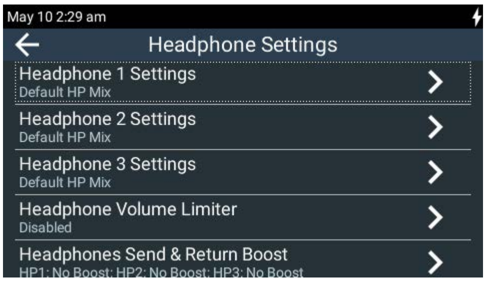
The **Headphone Settings** menu includes configuration settings for the three 1/4 inch (6.35mm) headphone jack outputs. To access this menu press the **HOME** button to return to the **Home** screen, then tap **Audio** > **Headphones**. To adjust headphone matrix settings "on the fly," press the **SOURCE** button on inputs 1-3 to display the headphone monitoring screen for each headphone output. For more information on adjusting headphone monitoring see ViA Headphone Controls.

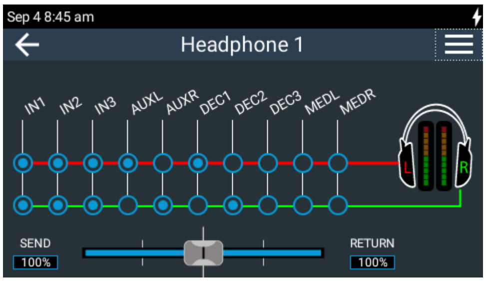
Headphone 1 - 3 Settings

For each headphone output it is possible to:

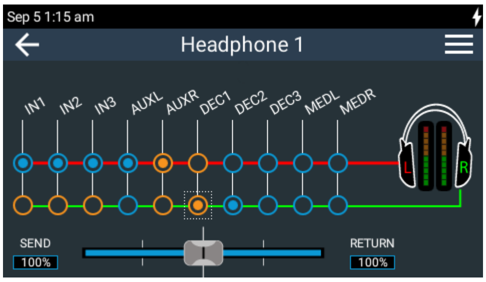
- View the **Matrix Editor** headphone monitoring settings.
- Edit and save a custom headphone mix for an announcer or event.
- Load or rename a custom headphone mix.
- Adjust the **Send** and **Return** headphone balance between outgoing and incoming audio.

Tap **Headphone [1-3] Settings** to display individual headphone output settings.





Tap to select and deselect sources in the left and right headphone output for each individual output. Swipe left on the **TOUCH SCREEN** to adjust the audio level of each individual headphone source.



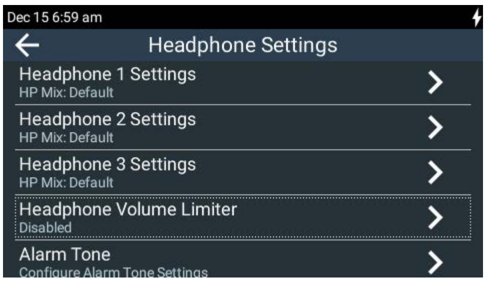
Note: Changes not yet saved as a custom headphone mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.

Headphone Volume Limiter

WARNING: LISTENING TO AUDIO AT EXCESSIVE VOLUMES CAN CAUSE PERMANENT HEARING DAMAGE. REDUCE VOLUME AS LOW AS POSSIBLE.

A headphone volume limiter can be employed to protect hearing when monitoring loud sources and/or when using low impedance headphones. To configure this:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio**.
2. Tap **Headphones**.
3. Tap **Headphone Volume Limiter** to toggle between enabling and disabling this feature.



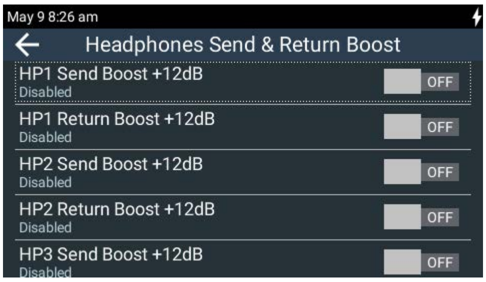
Headphones Send & Return Boost

WARNING: LISTENING TO AUDIO AT EXCESSIVE VOLUMES CAN CAUSE PERMANENT HEARING DAMAGE. REDUCE VOLUME AS LOW AS POSSIBLE.

The **Headphones Send & Return Boost** setting can assist when attempting to balance audio monitoring in noisy environments, or when incoming return audio is low.

1. Select **Home** screen **> Audio > Headphones**.

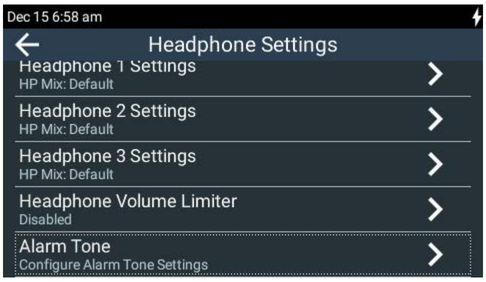
2. Tap **Headphones Send & Return Boost** to boost either the Send or Return audio for Headphones 1-3 by 12dB.



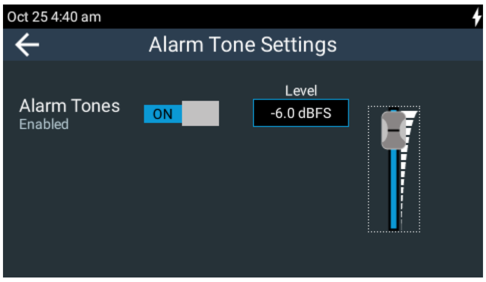
Alarm Tone Settings

By default, alarm tones are enabled in the headphone outputs to deliver low battery level alerts. To disable alarm tones and adjust the level of the alarm tone in the headphones. Acknowledging active alarm tones via the **TOUCH SCREEN** will turn alarm tones off:

1. Press the **HOME** button to return to the **Home** screen, then tap **Audio** > **Headphones** > **Alarm Tone**.



2. Tap the **On/Off** button to toggle **Alarm Tones** on and off and use the **Level** slider to adjust alarm tone levels in the headphone outputs.



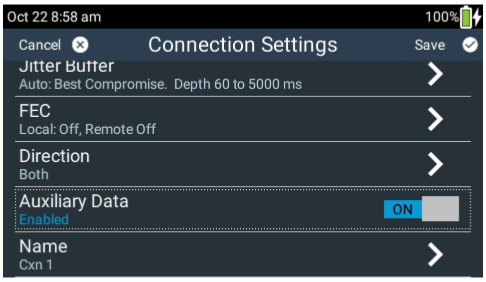
31.10 Enabling Relays & RS232 Data

Data must be enabled to activate contact closure **CONTROL PORT I/O** operation and RS232 data. The codec supports both in-band and out-of-band data depending on the connection transport and algorithm you are using. In-band RPTP data is automatically included within an audio stream when using Tieline Music and Music PLUS algorithms over any transport – IP, ISDN or POTS. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP	ISDN and POTS
Tieline Music and MusicPLUS	• In-band RPTP data is enabled automatically • Synchronized out-of-band data can be enabled and disabled as required • Using out-of-band data with rules between G5 codecs employing relay reflection minimizes latency • These algorithms must be used when connected to G3 codecs as they don't support out-of-band data	• In-band RPTP data is enabled automatically and used for all rules including relay reflection
All other algorithms	• No in-band data available; synchronized out-of-band data can be enabled and disabled	• No in-band or out-of-band data available

Select **Enable Auxiliary Data** when creating a program in the **Programs panel** to enable RS232 data and activate rules employing relay reflection over a connection. This will allow the codec to connect to external devices and send RS232-compatible data via the serial port on the rear panel. Alternatively, enable auxiliary data as follows:

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Connection Settings** symbol.
3. Tap to toggle the **On/Off** button for **Auxiliary Data** to activate **CONTROL PORT I/O** operation and RS232 data in the codec.



4. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to save the setting. Note: Please see [Appendix A for RS232 and Control Port Wiring]() information.

Configuring Control Port Contact Closure Operation

The **Rules panel** in the Web-GUI can be used to configure control port and logic inputs and outputs. Codec 'rules' configure events based on specific codec actions. Typically rules are based on a change in the state of a physical GPIO control port, or a Tieline or WheatNet-IP logic IO, or a codec program being connected or disconnected. Rules can only be created with the Web-GUI while the codec is disconnected. There are two ways to create rules in the HTML5 Toolbox Web-GUI:

1. The **Rules panel**, or
2. The **Program Manager panel**.

Rules for connecting or disconnecting a program are configured in the **Rules panel**. Rules for connecting or disconnecting an audio stream are configured in the **Program Manager panel**.

Important Note: A non-WheatNet-IP Tieline codec can be configured to trigger a logic IO in a Tieline Genie Distribution and Merlin PLUS WheatNet-IP codec, as well as 4 physical **CONTROL PORT** GPIOs. The codec has:

- 4 physical **CONTROL PORT** GPIOs.
- 7 Tieline and WheatNet virtual inputs (1-7). Note: Virtual inputs 5-7 can be activated by pressing the F1 button and **SOURCE** buttons 1-3.
- 64 Tieline virtual logic outputs. Note: Tieline logic IOs (LIOs) are only supported between Genie, Merlin, ViA and Bridge-IT IP codecs.
- 64 virtual WheatNet logic outputs available in Genie Distribution and Merlin PLUS WheatNet-IP codecs. These allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.

See the Creating Rules section for more information.

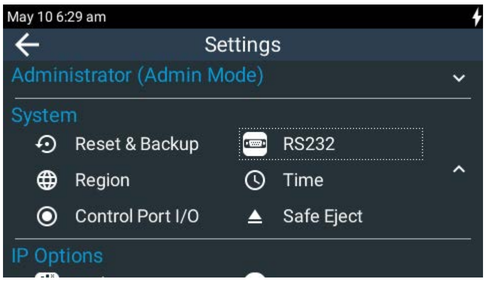
Configuring RS232 Data

When **Data** is enabled the codec can be connected to external devices and transport RS232 data using the **CONTROL PORT I/O** on the left side panel of the codec. It is important to configure two settings:

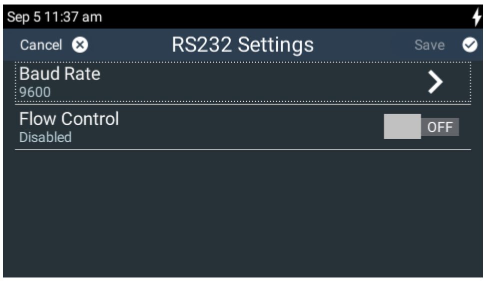
1. Enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, unregulated data will flow and some may be lost.
2. Match the serial port baud rate to the rate used by the device connected to the codec **CONTROL PORT I/O**. Ideally the settings on both codecs should match, or you could encounter data overflow issues.

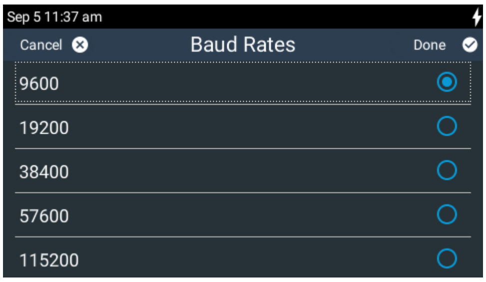
To adjust these settings:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Navigate to **System** and then tap **RS232**.

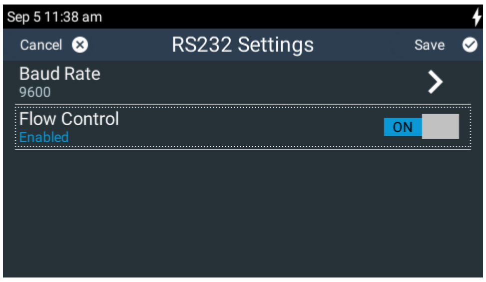


3. Tap **Baud Rate** to select from the baud rate options.





4. Tap the **Flow Control On/Off** button to toggle between enabling and disabling this feature. Tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to confirm the new setting. Note: Flow control is **Disabled** by default.

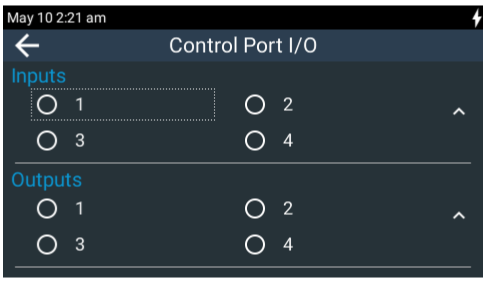


Important Notes:
- When connecting to G3 codecs over IP, ISDN or POTS only in-band data is available via the Music and MusicPLUS algorithms. See RS232 Data Configuration for more details.
- Only the dialing codec needs to be configured to send RS232 data to another Tieline G5 codec. Session data sent from the dialing codec configures the answering codec.

31.11 Monitor Control Port I/O Status

It is possible to monitor the status of the four relay inputs and four opto-isolated outputs available using the DB15 **CONTROL PORT I/O** connector. To monitor status:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Control Port I/O**.



3. Tap on an output to toggle the state from **Off** to **On**. Note: Input states cannot be changed.

Tieline and WheatNet virtual logic inputs 5-7 can be activated by pressing the F1 button and the **SOURCE** buttons on inputs 1-3. E.g. F1 + **SOURCE 1**

31.12 Configuring TCP/UDP Ports

In TCP and UDP networks the codec port is the endpoint of your connection. These network ports are essentially the doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address and unique port numbers can be used to route audio to each codec.

For two codecs to connect, they need to be configured with matching port numbers. In most situations you should consult your organization’s IT professional if there is a need to adjust codec port settings.

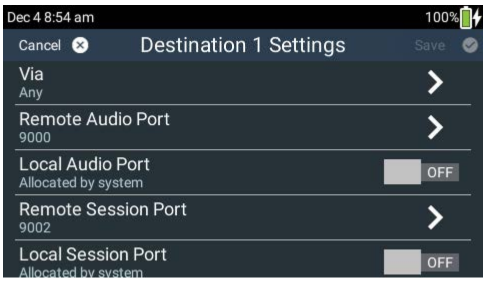
Adjusting Port Numbers

Reasons for adjusting the port setting on your codec include:

- Creating a safe path through an organization's gateway and firewall configuration.
- Another IP device is already using a codec’s port number.
- More than one studio codec is in use and each codec requires a different port number.

Tieline Codec Default Port Settings

By default, the codec uses a TCP session port to send session data and a UDP port to send audio for each connection. The session port is configured to use the TCP protocol because it is the most likely protocol to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably. The same session port can be shared for multiple connections to one codec but a unique audio port needs to be assigned for each IP connection.



Default remote session and audio ports

The default session and audio port settings in Tieline codecs are outlined in the section Installing the Codec at the Studio. This section also contains useful information for configuring port forwarding and troubleshooting IP connections.

The following sections describe how to adjust port settings for Tieline session data and audio ports. To configure SIP ports see Configuring SIP Interfaces and to configure Fuse-IP port settings see Connecting with Fuse-IP.

Configuring Tieline Session and Audio Ports

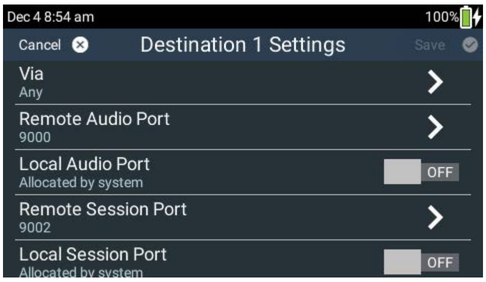
When using **Tieline Codec** session data for establishing connections you can adjust the default audio ports if required.

Port	Default Port	Available Port Range
Remote's Session Port	9002	2000 to 65535
Remote's Audio Port	9000	2000 to 65535
Local Audio Port Allocation	Automatic	2000 to 65535

Note: When the **Local Audio Port Allocation** setting is **Automatic** the codec assigns the return port value and forwards this information to the codec to which you are dialing.

To adjust the session or audio ports for IP connections:

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Destination Settings** symbol.
3. Tap **Remote Audio Port** or **Remote Session Port** to change the dialing program ports.



4. By default, the local audio and session ports are automatically allocated. To change this setting, tap the **On/Off** button for **Local Audio Port** or **Local Session Port** to enable manual configuration. Then configure the ports manually and tap **Save** in the top right-hand corner of the **TOUCH SCREEN** to save all settings.

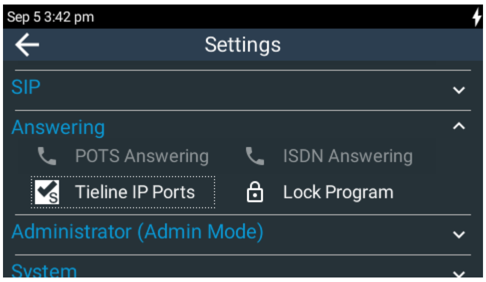
Important Note: The codec will not allow the entry of an invalid port number.

Changing Tieline Session Ports when Answering

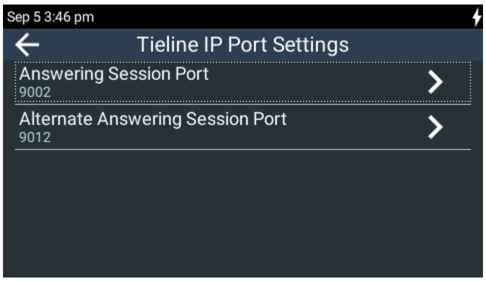
The codec is automatically configured to "listen" for Tieline session calls using ports 9002 and 9012.

Tieline G5 codecs (Bridge-IT, Genie, Merlin and ViA codec families) and Tieline Commander and i-Mix codecs dialing over IP1 use session port 9002 by default. Tieline G3 codecs dialing over IP2 use session port 9012 and the codec can answer calls from these codecs automatically using the **Alternate Answering Session Port**. To adjust the local Tieline session data ports:

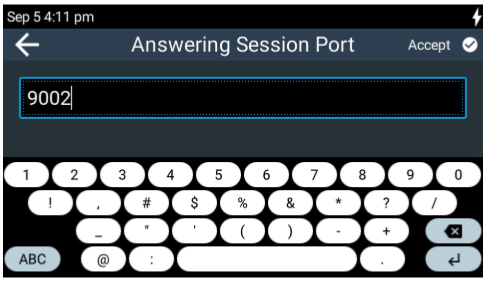
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **Answering** menu and then tap **Tieline IP Ports**.



3. Tap **Answering Session Port** to change the port.



4. Enter the port number, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



5. Tap **Alternate Answering Session Port** to change the port, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

31.13 Configuring QoS for IP Packets

It is possible for IP networks to prioritize and differentiate between data packets transmitted through routers across networks. This is useful because in data networks many different IP services like email, voice, web pages, streaming video and music coexist within the same network infrastructure.

Prioritizing IP Data Packets

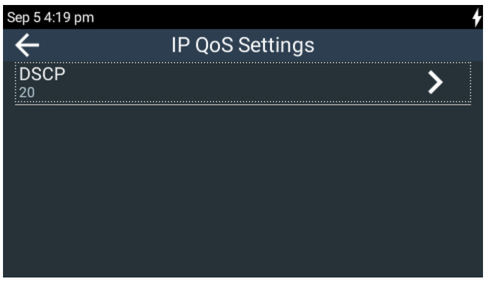
IP audio data packets can be configured for expedited or assured forwarding (Quality of Service, or QoS) when traversing different networks. Routers can also be configured to ignore these forwarding priorities, so they are not assured across all networks.

The codec can be configured to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted over the network. Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

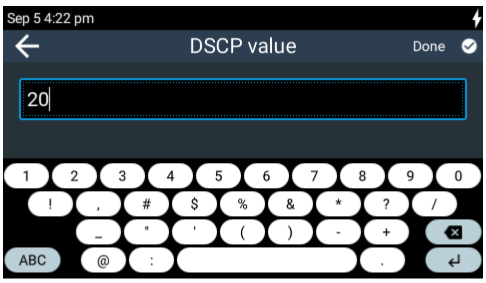
Configuring QoS

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **IP Options** menu and then tap **IP QoS**.

3. Tap to select **DSCP**.



4. Enter the **DSCP value** recommended by your IT administrator, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



Important Note: To ensure the continuous and regular flow of tagged data packets along the path from point-to-point, all routers and switching equipment must allow the QoS DSCP setting. Any bandwidth partitioning schemes should partition over a small interval to ensure the codec jitter buffer does not empty and audio remains continuous.

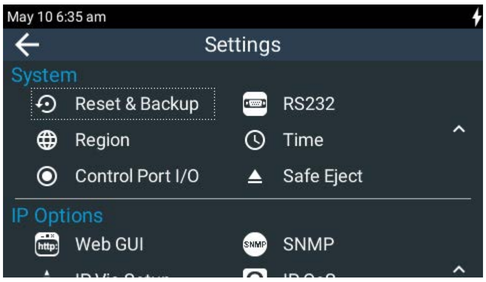
31.14 Reset Defaults and Backup/Restore Settings

There are several options in the **Reset & Backup** menu which allow you to backup, restore or reset a variety of factory default settings within the codec.

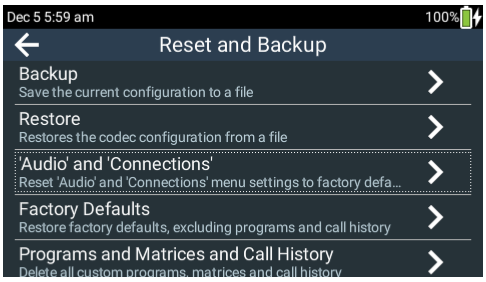
	Function	Description
1	Backup	Save the current configuration to a file
2	Restore	Restores codec configuration from a saved file. Restore Programs and Scheduler settings only, or system settings only, or restore all configuration
3	Audio and Connections	Tap to restore factory default settings for Audio and Connections menu settings.
4	Factory Defaults	Tap to restore factory default settings, excluding custom programs, scheduler data and call history
5	Programs & Matrices and Call History	Tap to delete custom programs, matrices and recent calls in the codec.
6	Codec Logs	Tap to clear codec event and log history. Note: This should only be performed if instructed to by Tieline support staff.

Important Note: After restoring factory defaults the codec will automatically reboot.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Reset & Backup**.



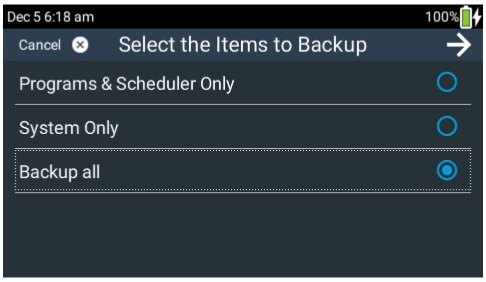
3. Tap to select the preferred reset option.



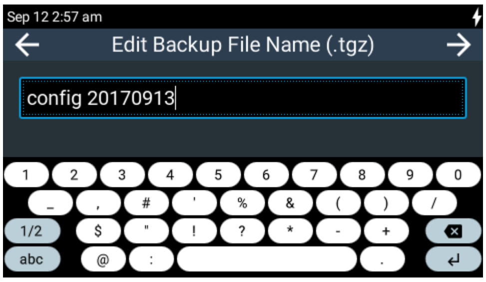
Important Note: A confirmation dialog is presented for each reset function.

Backup Settings

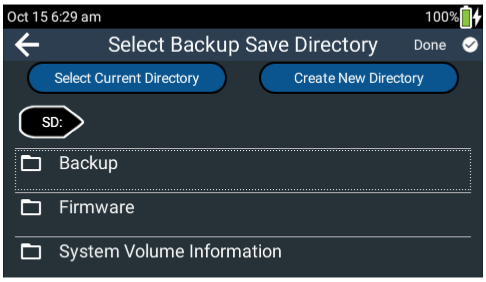
1. Insert an SD card into the SD card slot on the rear panel of the codec. Note: A single partition FAT32 formatted SD card must be used.
2. Press the **HOME** button **> Settings > System > Reset & Backup > Backup**.
3. Tap to select the items to backup.



4. Name the file and tap the arrow in the top right-hand corner of the **TOUCH SCREEN**.



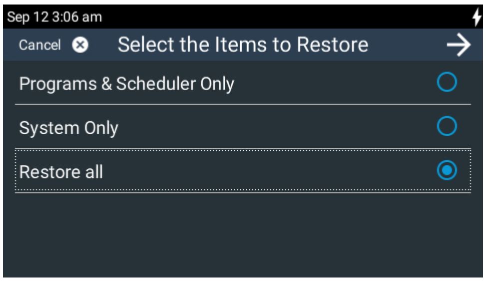
5. Select the directory in which to save the backup file, or create a new directory, then tap **Done**.



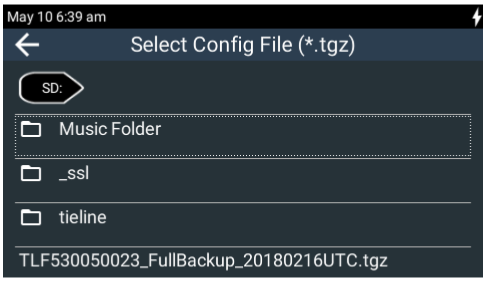
6. A dialog confirms the backup has been successful. Press the **HOME** button, then tap **Settings > System > Safe Eject** to safely remove the SD card.

Restore Settings

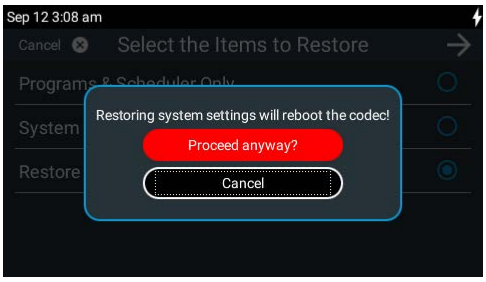
1. Insert the SD card with the backup configuration file into the SD card slot on the rear panel of the codec.
2. Press the **HOME** button > **Settings > System > Reset & Backup > Restore**.
3. Tap to select the items to restore.



4. Tap to select the config file to restore on the codec and commence the system restore process.



5. Tap **Proceed Anyway** in the confirmation dialog to continue.



5. Press the **HOME** button, then tap **Settings** > **System** > **Safe Eject** to safely remove the SD card after the codec has rebooted.

Reset and Restore Defaults using the Toolbox Web-GUI

To use the HTML5 Toolbox Web-GUI for reset and backup functions see Reset Factory Default Settings.

31.15 Configuring Via Interfaces

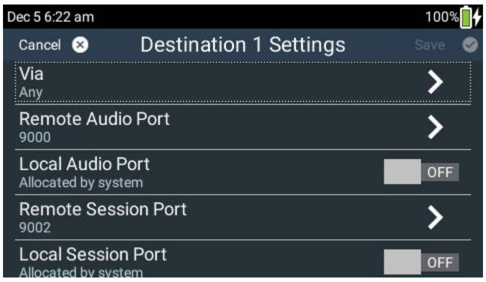
When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface will be used to dial a connection. The default **Via** interfaces in order of use when available are:

1. **LAN1** Ethernet port (default **Primary** Via interface)
2. **LAN2** Ethernet port (default **Secondary** Via interface)
3. Internal Wi-Fi (default **Tertiary** Via interface)

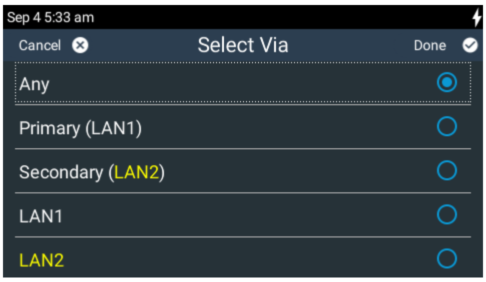
Important Note:
- If an interface is not available it is not listed in the **Via** interface selection screen. E.g. Wi-Fi is not enabled or a USB modem is not attached.
- VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.
- Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary **Via**.

To change the default **Via** interface setting:

1. Press the **HOME** button to return to the **Home** screen, then tap **Connections**.
2. Tap the **Destination Settings** symbol.
3. Tap **Via**.



4. All available interfaces are displayed in the menu and you can adjust the selected setting. Note: A LAN interface displayed in yellow text is enabled but a network cable is not attached. An interface in red text is disabled.

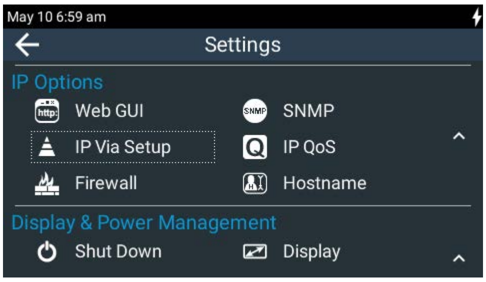


Configure Default Primary, Secondary and Tertiary Interfaces

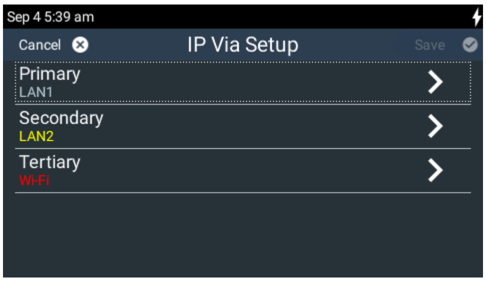
It is possible to reconfigure the default **Primary** (LAN1), **Secondary** (LAN2) and **Tertiary** (Wi-Fi) interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual primary interface used at each location can vary for each codec. For one codec it may be a LAN interface and for another it may be a USB cellular interface. This allows you to configure site-specific settings to suit the available network interfaces at different remote locations.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.

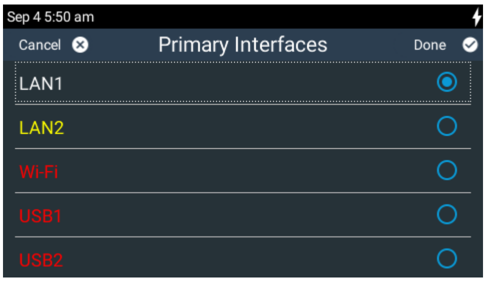
2. Tap to expand the **IP Options** menu and then tap **IP Via Setup** .



3. Tap to select an interface to configure. Note: A LAN interface displayed in yellow text is enabled, but a network cable is not attached. An interface in red text is disabled, e.g. **Wi-Fi** in the following image.



4. Tap to select the default interface to use.



31.16 Configuring SNMP Settings

The codec supports Simple Network Management Protocol (SNMP) for managing devices on IP networks.

Description of SNMP Settings in the Codec

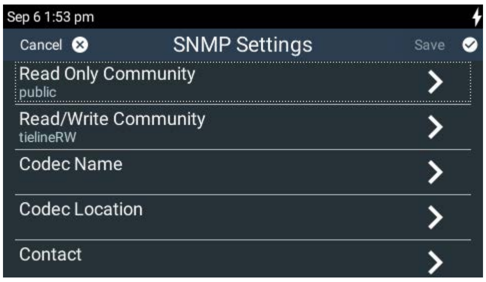
Features	Operation Button Descriptions
Read Only Community	SNMP provides two types of access, namely Read-Only access and Read-Write access. The **Read Only Community** identifier allows Read Only level access.
Read/Write Community	The **Read/Write Community** identifier allows Read/Write level access.
Codec Name	A user-specified alphanumeric identifier which may be used by third-party SNMP software to identify a device. The device name corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysName" SNMP attribute and is completely independent of DNS, NIS, WINS or other device naming and identification schemes, though convention is to use the device's fully-qualified domain name.

Codec Location	A user-specified alphanumeric string which may be used by third-party SNMP software to identify a device. Device location corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysLocation" SNMP attribute.
Contact	A text identifier for the contact person for this managed node, together with information on how to contact this person.

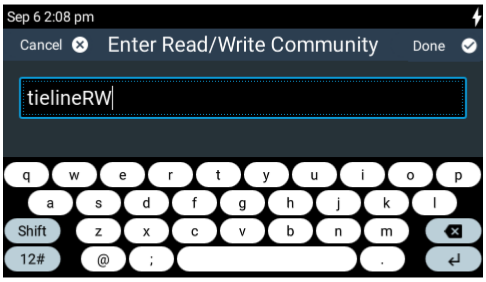
To configure SNMP device settings:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **IP Options** menu and then tap **SNMP**.

3. Tap to adjust each setting in turn.



4. Tap **Done** in the top right-hand corner of the **TOUCH SCREEN** to save each new entry.



Important Note: For more information on configuring SNMP codec settings using the HTML5 Web-GUI see Configuring SNMP in the Codec.

31.17 Configuring Regional Settings

The **Region** menu allows configuration of **Country** and **Language** settings.

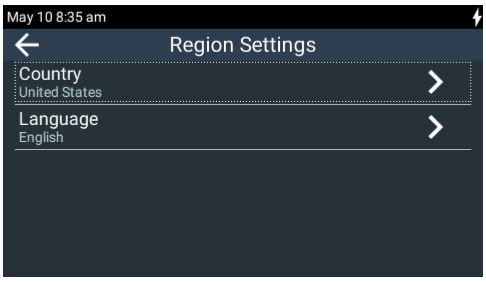
Configure Country Settings

The **Country** setting in the codec configures country-specific settings like:

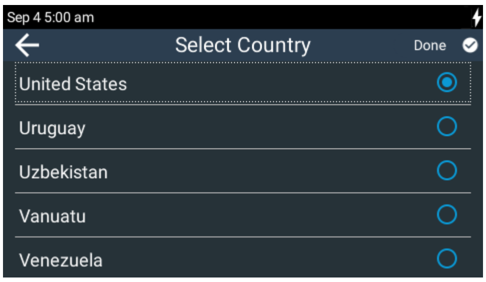
- POTS line ring tones.
- POTS line impedance.
- Use of G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world (e.g. Europe/Australasia), when the G.711 algorithm is used for IP/SIP or ISDN connections.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Region**.

3. Tap **Country**.



4. Swipe up or down to navigate to the correct country, then tap to select and save the new setting.

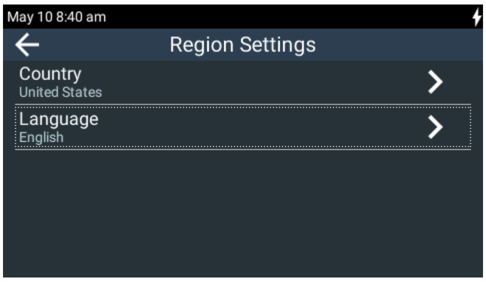


Configure the Language Displayed

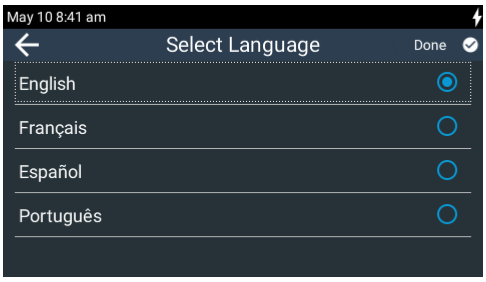
The codec offers support for several languages.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Region**.

3. Tap **Language**.

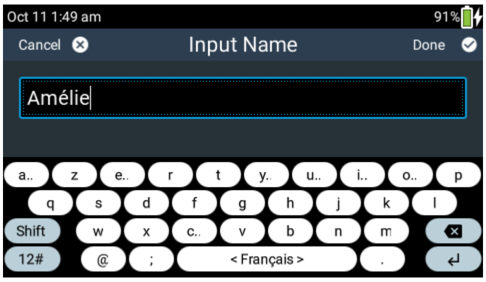


4. Select the preferred language to display in the codec.



31.18 Typing and Language Options

The onscreen keyboard supports multiple languages including English, French, Spanish and Portuguese.

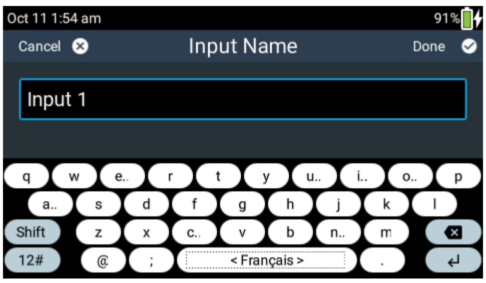


French onscreen keyboard

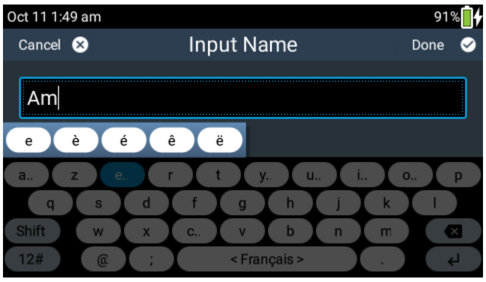
The keyboard can be used in three different ways to enter and edit alphanumeric text.

Touch Screen Typing

With your finger swipe left and right on the space bar to select a different language.



Tap and hold a key to access multiple accent options, then tap to select the preferred option.



Type using Navigation Buttons

Use the **NAVIGATION** buttons to select the space bar, then use the down ▼ **NAVIGATION** button to scroll through language options.

Space bar selected with dotted 'focus'

Use the **NAVIGATION** buttons to select characters and then press the **OK** button to add each character.

Using an External USB Keyboard

Attach an external USB keyboard to a USB port on the codec. Use the arrow keys to navigate to the space bar, then use the down arrow key to scroll through language options.

Use the arrow keys to navigate to each letter or number and press the Enter key to select each character.

Shift Key Functions

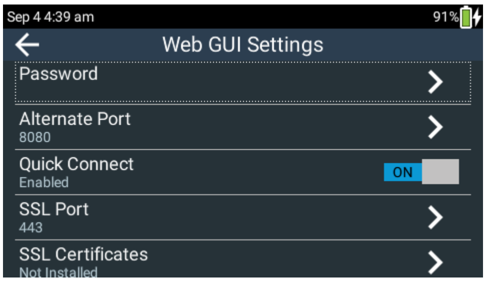
Tap the **Shift key** once in the onscreen keyboard to capitalize a single letter. Tap the **Shift key** twice, or press and hold the **Shift key** to turn the cap lock on. Tap the **Shift key** once to unlock the cap lock. This functionality is the same when an external USB keyboard is attached to the codec.

31.19 Configuring Web-GUI Settings

Use the **Web GUI** menu to adjust the Toolbox Web-GUI password, port settings and enable/disable the HTML5 Quick Connect Web-GUI.

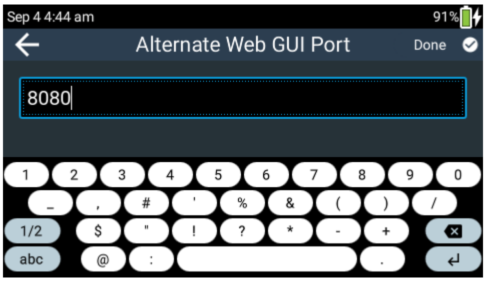
1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **IP Options** menu and then tap **Web GUI**.

3. Tap **Password** to change the Web-GUI login password.

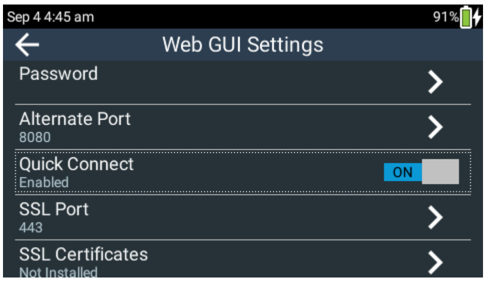


4. Enter a new password and then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

5. Tap **Alternate Port** to change the alternate port used to access the Web-GUI, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.



6. Tap the **On/Off** button to toggle between enabling and disabling the **Quick Connect** Web-GUI (default setting **On**). Note: See Using the HTML5 Toolbox Quick Connect for more information.



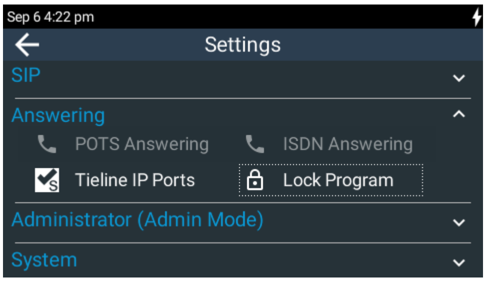
31.20 Lock or Unlock a Program in the Codec

Most remote codecs dial from the remote broadcast site back to the studio. However, if the remote codec is going to answer incoming calls, it is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. mono peer-to-peer plus IFB, cannot be unloaded by a codec dialing in with a different program type like peer-to-peer mono.

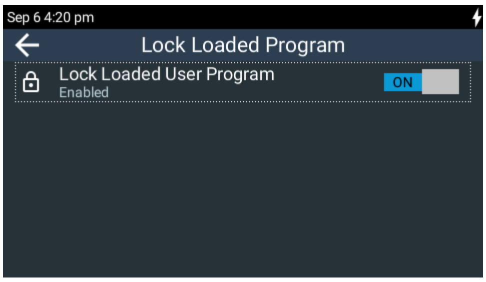
For example, if the codec should always connect using mono peer-to-peer plus IFB, simply load and lock this program in the codec. Another reason for locking a program in the answering codec is to always use a particular jitter buffer or FEC setting.

Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected. A compatible program type can still connect and specify different connection parameters such as the algorithm and bit rate via session data.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **Answering** menu and then tap **Lock Program**.



3. Tap the **On/Off** button to toggle between enabling and disabling **Lock Loaded User Program** (default setting **Off**).

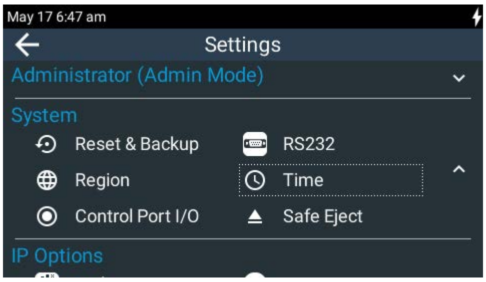


Important Note: It is only possible to lock custom programs in a codec. If **Lock Loaded User Program** is enabled and you load a new custom program in the codec, the feature remains enabled and locks the most recently loaded custom program.

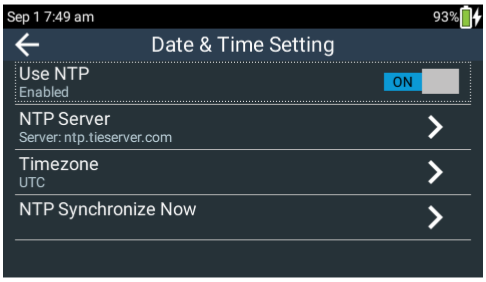
31.21 Adjusting Time Settings

To adjust time settings in the codec:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Time**.



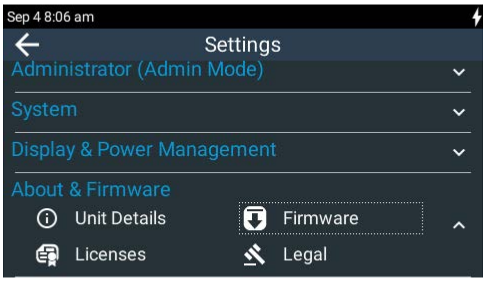
3. Tap the **On/Off** button to enable or disable NTP. Tap to adjust the **NTP Server** or **Timezone** settings. Tap **NTP Synchronize Now** to synchronize time with the NTP Server.



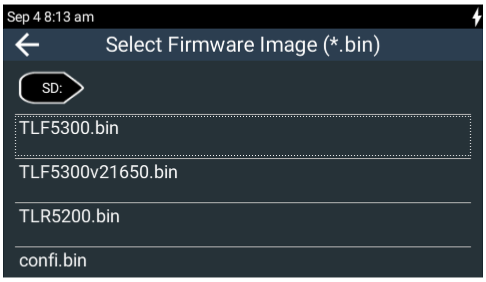
31.22 Upgrading Firmware via SD Card

To download the latest codec firmware visit www.tieline.com. Copy the firmware file onto a USB stick and then use the following procedure to perform a firmware upgrade. Note: A single partition FAT32 formatted SD card must be used. To safely remove an SD card after the upgrade, press the **HOME** button to return to the **Home** screen, then tap **Settings** **> System > Safe Eject**.

1. Insert an SD card with the latest firmware into the SD card slot on the rear panel of the codec.
2. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
3. Swipe up and tap to expand the **About & Firmware** menu, then tap **Firmware**.



4. Navigate to the firmware file on the SD card, then press the button.



5. Tap to confirm and commence the firmware upgrade. **IMPORTANT**: The codec will reboot automatically after the firmware upgrade. DO NOT remove power or reboot the codec before the update has completed and the codec has rebooted itself.

6. Safely remove the SD card from the codec after performing the upgrade. From the **Home** screen tap **Settings** **> System > Safe Eject**.

Important Note: We recommend clearing your browser cache after the upgrade is complete when using the HTML5 Toolbox web-browser GUI to control codec functions. The short cuts for this are:

- Google Chrome: shift+Ctrl+delete
- Mozilla Firefox: Ctrl+shift+delete
- Internet Explorer: Ctrl+shift+delete
- Safari: Ctrl+alt+e

31.23 Safely Remove USB Sticks and SD Cards

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings** .
2. Swipe up and tap to expand the **System** menu, then tap **Safe Eject** .

31.24 Install and Manage Security Certificates

Tieline codecs support the installation of TLS/SSL (hereafter referred to as SSL) security certificates to deliver an additional layer of security when connecting to IP networks. The digital SSL security certificate authenticates the codec and provides more secure encrypted HTTPS browser connections. The codec supports installing a private key as well as an intermediate and SSL certificate.

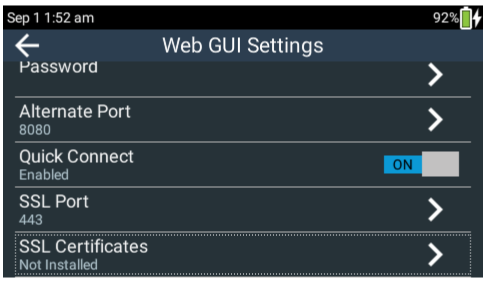
This section outlines certificate installation, removal and updates. It also outlines how to change the default SSL Port. Note: A single partition FAT32 formatted SD card must be used. To safely remove an SD card after installing certificates, press the **HOME** button to return to the **Home** screen, then tap **Settings** **> System > Safe Eject** .

Certificate Installation

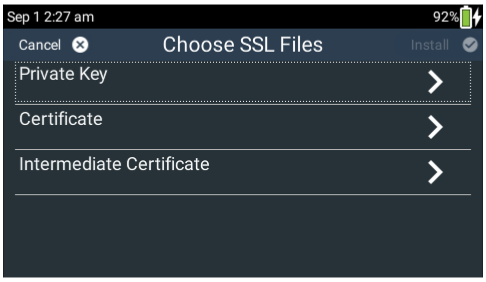
To install certificates purchased from a reputable vendor:

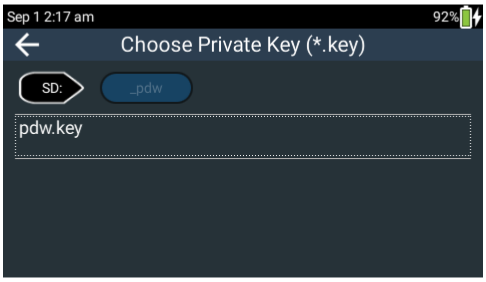
1. Insert an SD card into the SD card slot on the rear panel of the codec. Ensure the Private Key, digital SSL Certificate and Intermediate Certificate (if required) are copied onto the card.
2. Press the **HOME** button to return to the **Home** screen, then tap **Settings** .
3. Swipe up and tap to expand the **IP Options** menu, then tap **Web GUI** .

4. Tap **SSL Certificates**.

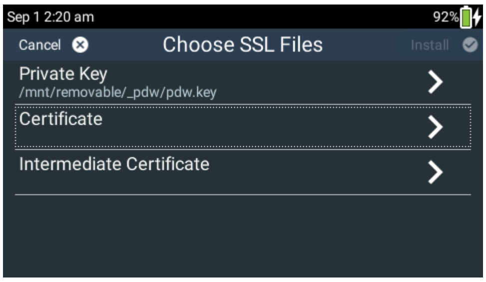


5. Select **Private Key** and navigate to the correct directory and .key (Private Key) file to install from the SD card. Tap to select it.

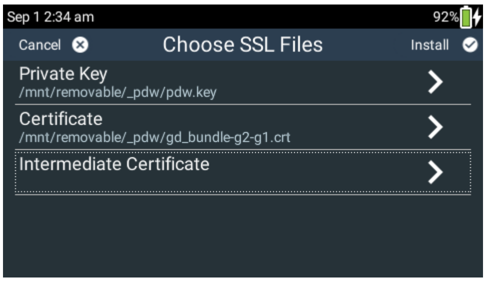


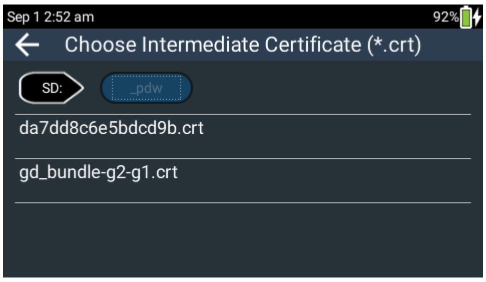


6. Select **Certificate** and navigate to the SSL Certificate (.crt) file on the SD card. Tap to select it.

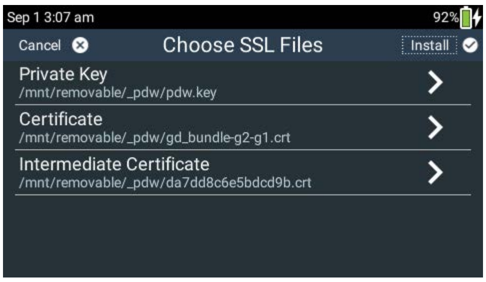


7. If an Intermediate Certificate has been supplied, select **Intermediate** and navigate to the Intermediate Certificate (.crt) file on the SD card. Tap to select it.

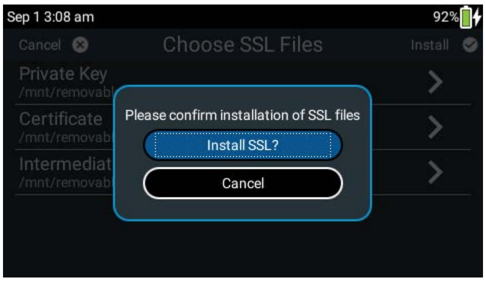


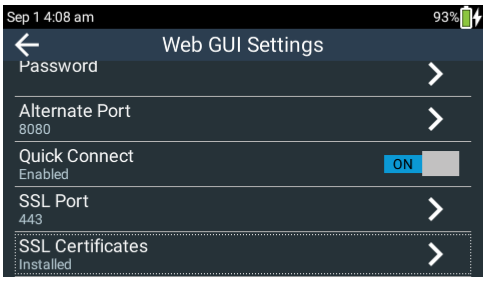


8. After adding the private key, SSL certificate, and intermediate certificate (if supplied), tap **Install** in the top right-hand corner of the **TOUCH SCREEN**.



9. Tap **Install SSL** in the confirmation dialog to complete installation. **Installed** should now be visible in the **SSL Certificates** menu.

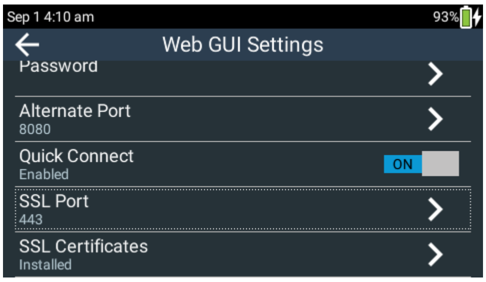




10. To access a codec via the HTML5 Toolbox Web-GUI in a browser after installing SSL security certificates ensure you type "https://" before the codec IP address. For example, https://172.16.0.100.

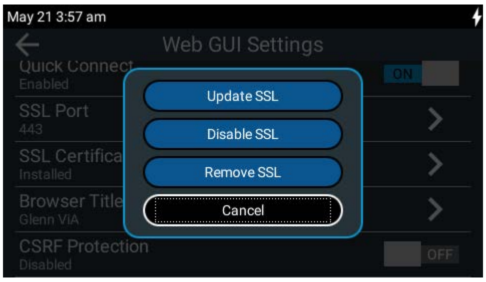
Changing the Default SSL Port

The codec uses the standard TCP port 443 for SSL communications. To adjust this setting navigate to **Settings** > **Web GUI** > **SSL Port**.



Update, Remove & Disable SSL Certificates

To update, remove or disable installed certificates navigate to **Settings** > **Web GUI** >**SSL Certificates**. Tap to **Update SSL**, **Disable SSL** or **Remove SSL** or **Update SSL** certificates.



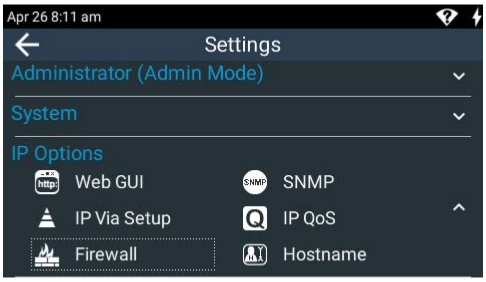
Troubleshooting Certificate Installation

If you type "https://" before the codec IP address and can't open the Toolbox web-GUI, first uninstall the certificates and then reinstall them. Also double-check you are installing the correct certificates. If you continue to have issues, contact your certificate vendor to ensure the certificate is valid.

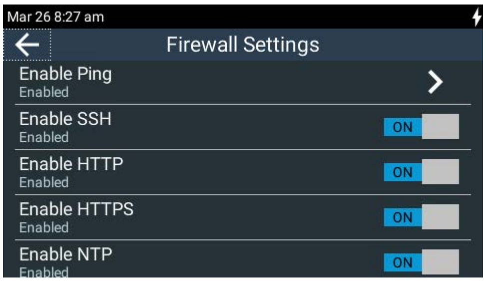
31.25 Firewall Configuration

The **Firewall** menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Swipe up and tap to expand the **IP Options** menu, then tap **Firewall**.



3. Tap to select and configure **Ping**, **SSH**, **HTTP**, **HTTPS**, **NTP** and **SNMP** firewall options.



31.26 Enabling CSRF Security

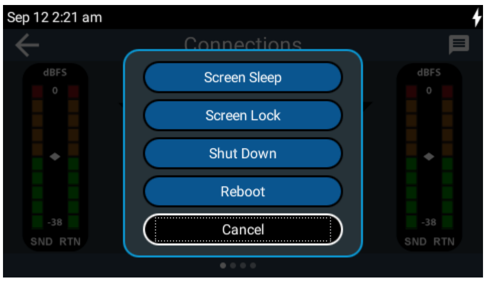
CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **IP Options** menu and then tap **Web GUI**.

3. Swipe up and navigate to **CSRF Protection** and tap the **On/Off** button to enable or disable this feature.

31.27 Shut Down and Screen Lock Options

To view power shut down options press the **POWER** button. The following screen is displayed.



Tap to select an option. Note: a confirmation dialog is displayed for **Shut Down** and **Reboot** functions.

Shut Down Option	Battery State
Screen Sleep	Turns the screen off to conserve battery power.
Screen Lock	Turns the screen off to conserve battery power and simultaneously locks the codec controls. Press and hold the **HOME** button to unlock the codec **TOUCH SCREEN** and all controls.
Shut Down	The codec is turned off.
Reboot	Reboots the codec. Note: this may be required to enable **LAN1** and/or **LAN2** if they have been turned off to conserve power when using the internal **BATTERY**.

Important Notes:

- **Shut Down** options are also available when navigating from the **Home** screen to **Settings** > **Display & Power Management** > **Shut Down** .
- Press and hold the **POWER** button for 5 seconds to turn the codec **[OFF]**. **This is not recommended** and is like turning off your PC by pressing and holding the power button.

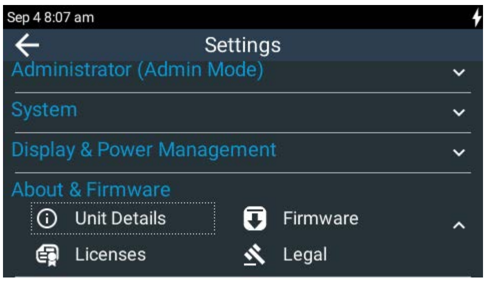
Lock/Unlock Codec Screen and Controls

1. Press and hold the **HOME** button to lock the codec **TOUCH SCREEN** and all controls.
2. When locked, press and hold the **HOME** button to unlock the codec **TOUCH SCREEN** and all controls.

31.28 About Unit Details

The **Unit Details** menu lists details like the codec firmware version, IP addresses and the serial number.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings** .
2. Tap to expand the **About & Firmware** menu and then tap **Unit Details** .



31.29 Alarm Notifications

Low Battery Alarm

When the battery level reaches 20%, 10% and 5%:

1. The **ALARM LED** flashes until acknowledged, and then turns solid red.
2. Alarm tones are audible in the left headphone output of **HP 1-3**.

Tap the **TOUCH SCREEN** or touch any codec controls to acknowledge the alarm and stop the alarm tones. For more information see Battery Use and Power Management.

32 Connecting to the ToolBox Web-GUI

There are two graphical user interface (GUI) options for configuring and connecting Tieline G5 codecs:

1. HTML5 Toolbox Web-GUI: fully configure codec settings, create dialing programs and dial, hangup and monitor connections.
2. HTML5 Toolbox Quick Connect Web-GUI: designed for dialing and managing simple peer-to-peer connections; ideal for non-technical users.

About the HTML5 Toolbox Web-GUI

The HTML5 Toolbox Web-GUI improves the user experience with G5 codec command and control and runs seamlessly on modern browsers. The HTML5 Toolbox Web-GUI will run on Mac, Windows and Linux computers.

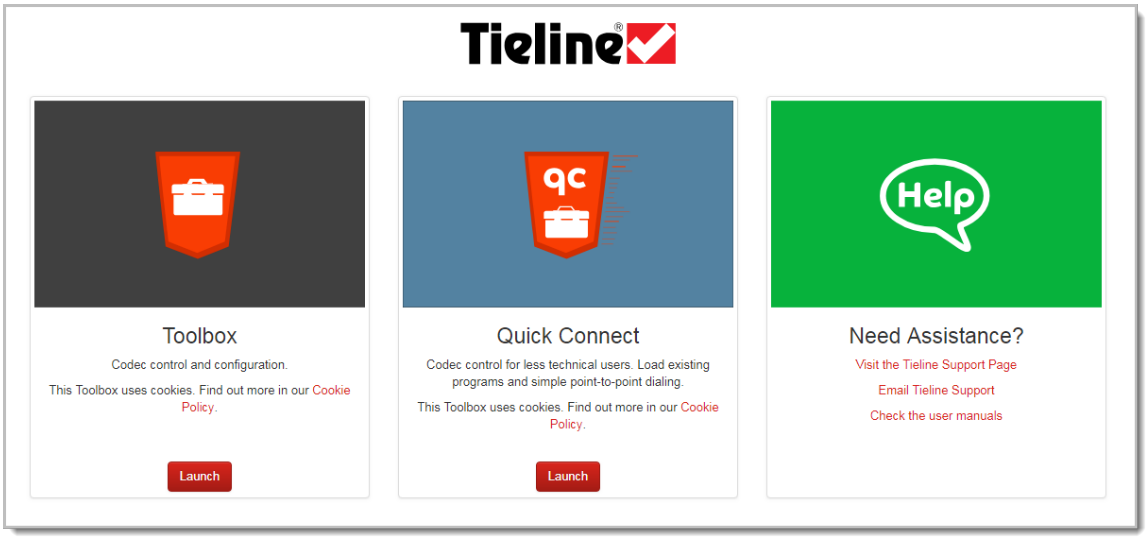
About the HTML5 Toolbox Quick Connect

The HTML5 Toolbox Quick Connect Web-GUI has a reduced feature-set and allows non-technical users to load existing programs and dial via the **Quick Connect panel**. Users can dial a simple peer-to-peer connection over POTS, ISDN or IP. By default, the HTML5 Quick Connect panel is enabled in each codec. To disable this feature:

1. Navigate to **Settings** in the main HTML5 Toolbox web-GUI and click to open the **Options panel**.
2. Navigate to **Quick Connect Enabled** and deselect the check-box.

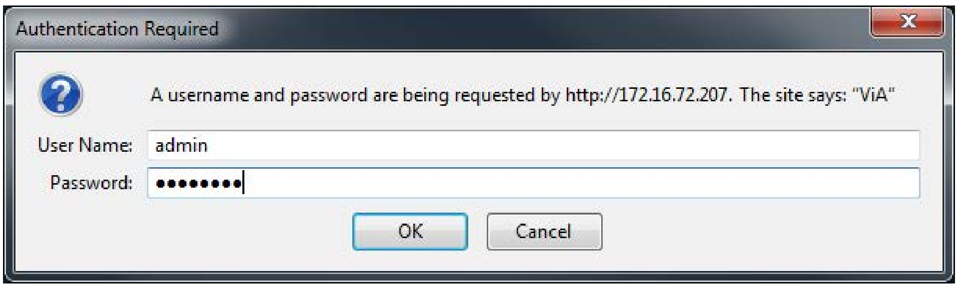
32.1 Opening the HTML5 Web-GUI & Login

1. Attach an Ethernet cable to the **LAN1** port on the codec.
2. Press the **HOME** button to return to the **Home** screen and tap **Settings > LAN1** to display the IP address in your codec.
3. Ensure your PC is connected to the same LAN.
4. Open your web browser and type the IP address of your codec into the address bar of your browser, e.g. **http://192.168.0.xxx** (the last digits are the private address details unique to your codec over a private LAN).
5. Refresh the browser and the Web-GUI landing page will display the various command and control options.



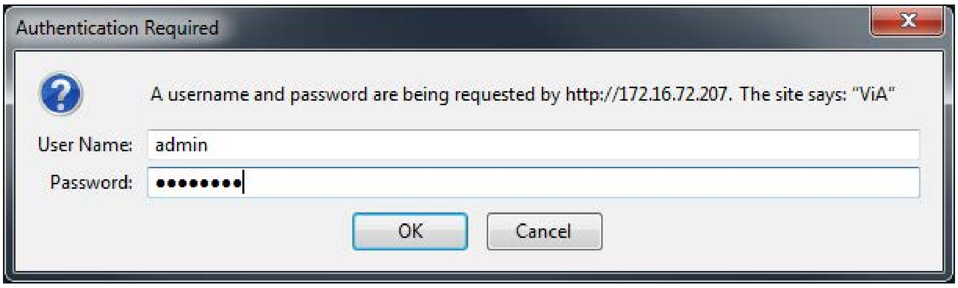
Launching the HTML5 Toolbox Web-GUI

1. Click to launch the **HTML5 Toolbox Web-GUI**.
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline highly recommends you change the password (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.



Launching the HTML5 Toolbox Quick Connect

1. Click to launch the **HTML5 Toolbox Quick Connect** Web-GUI.
2. When you launch Toolbox for the first time an authentication dialog prompts you to enter the user name "**admin**" and password "**password**" to login, then click the **OK** button. Tieline highly recommends you change the password (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.



Using the Web-GUI over the Internet

If your codec is connected over the internet via a public static IP address it is possible to connect and configure it from any PC also connected to the internet. If you have multiple browsers open on a PC for different codecs it is possible to customize the browser title for simple identification. To configure this:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **IP Config** menu and then tap **Web GUI**.
3. Tap **Browser Title** to enter the custom title, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

To configure this setting using the HTML5 Toolbox Web-GUI, click **Settings** at the top of the screen, then click **Options** to display the **Options panel**. Enter the **Browser Title** and then click **Save**.

LAN Troubleshooting

PC LAN Settings

Check the LAN settings on your PC if it is connected to a LAN and is having trouble opening the Toolbox Web-GUI in a web-browser.

1. Open Internet Explorer.
2. Click **Tools > Internet Options > Connections**.
3. Click the **LAN settings** button.
4. If the PC is using a proxy server over the LAN you may need to select the **Bypass proxy server for local addresses** check-box.
5. If you still can't connect, click the **Advanced** button in the **Local Area Network Settings** dialog and ask your IT administrator to assist you with entering the IP address of the codec into the **Exceptions** pane of the **Proxy Settings** dialog.

Port Selection

By default port 80 is used by your PC to communicate with the codec and launch the web-GUI. If port 80 cannot be used across your network for some reason, type the IP address of your codec into your browser with a full colon and the port number 8080.

E.g. **192.168.0.176:8080**

It is also possible to specify a different port for connecting the Toolbox web-GUI to your codec.

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Web GUI**.
3. Tap **Alternate Port** to change the alternate port used to access the Web-GUI, then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

4. Type the IP address of your codec into your browser with a full colon and then the new port number.

Important Note: Any new port specified must be within the range 2000 to 65535 inclusive.

32.2 Security and Changing the Default Password

Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. In addition, search engines are widely available which can discover and expose unsecured 'internet connected devices'. Tieline recommends the following IP codec security precautions are followed as a bare minimum, to ensure your codec connections remain secure.

Maintaining Codec Network Security

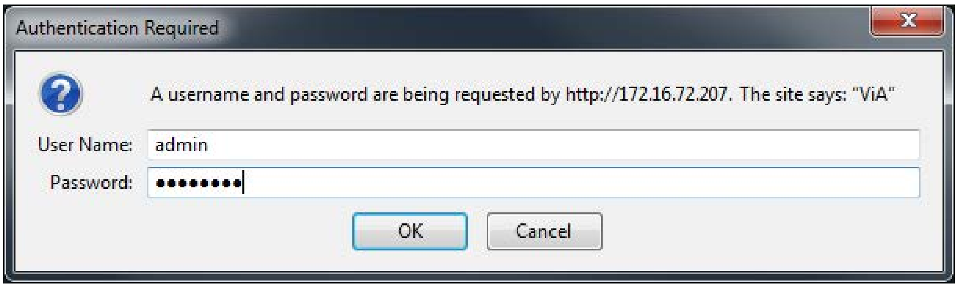
Adequate security is a major factor in ensuring your codecs and your broadcast network remain secure. There are several layers of security available in Tieline codecs to maintain secure connections. These include:

1. Immediately change the default password when you commission and install your codecs (see instructions which follow). Create a strong password which includes both capital and lower case letters, symbols and numbers (up to 15 characters can be entered). Password managers can be useful when managing multiple passwords within organizations.
2. Ensure your codec is behind a firewall and only open the TCP and UDP ports required to transmit session and audio data between your codecs. Using non-standard ports instead of Tieline default ports can also ensure the codec is more difficult to discover by external parties.
3. Ports 80 and 8080 are commonly used to access the Tieline codec web server. You can add an additional layer of security by translating these ports on the WAN side of your network into non-standard port numbers. Adjust ports using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Configuring TCP/UDP Ports.
4. By default SIP interfaces are disabled to avoid unwanted traffic. The **SIP Filter Lists panel** in the Toolbox HTML5 Web-GUI allows filtering of SIP URIs and User Agents to provide greater security when using SIP. See Configure SIP White and Blacklists for more information.
5. An SSL security certificate can be installed on each codec in your network to ensure it is a trusted device within your network. See Installing a Security Certificate for more information.
6. Firewall settings to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Adjust settings using the **Options panel** in the Toolbox HTML5 Web-GUI, or see Firewall Configuration.
7. Implementation of CSRF protection (Cross-Site Request Forgery). Enable and disable this setting using the **Options panel** in the Toolbox HTML5 Web-GU, or see Enabling CSRF Security for more info.

Be sure to document any port changes because this information will be required if you need to contact Tieline or other online support services.

Changing the Default Password

The default **User Name** for the HTML5 Toolbox Web-GUI is **admin** and the default **Password** is **password**. Tieline highly recommends changing the default password after initially connecting to increase network security and protect your codec from being tampered with during live broadcasts.



HTML5 Web-GUI Login Dialog

Caution: Codecs connected to the internet can be accessed by anyone with knowledge of the codec's public IP address. In addition, search engines are widely available which can discover and expose unsecured 'internet connected devices'. Tieline recommends the following IP codec security precautions as a bare minimum:

- Immediately change the default password when you commission and install your codecs. Create a strong password which includes both capital and lower case letters, symbols and numbers (up to 15 characters can be entered). Password managers can be useful when managing multiple passwords within organizations.
- Ensure your codec is behind a firewall and only open the TCP and UDP ports required to transmit data and audio between your codecs.
- Ports 80 and 8080 are commonly used to access the Tieline codec web server. You can add an additional layer of security by translating these ports on the WAN side of your network into non-standard port numbers.

Using these basic precautions will help to protect your equipment from being tampered with by nefarious characters.

Creating a New Password

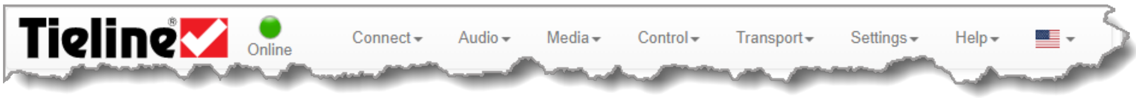
The authentication login password can be changed at any time using the **TOUCH SCREEN**. Note that passwords are case sensitive:

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to expand the **System** menu and then tap **Web GUI**.
3. Tap **Password** to change the Web-GUI login password.
4. Enter a new password and then tap **Done** in the top right-hand corner of the **TOUCH SCREEN**.

Important Note: The **Username** in the codec menu is permanently set to **admin** and cannot be changed; only the **Password** can be changed.

33 Using the HTML5 Toolbox Web-GUI

The following sections provide an overview of the different configuration panels available within the codec's HTML5 Toolbox Web-GUI. Navigate with the mouse pointer to the **Menu bar** at the top of the Web-GUI screen and click to select and open each panel in turn.



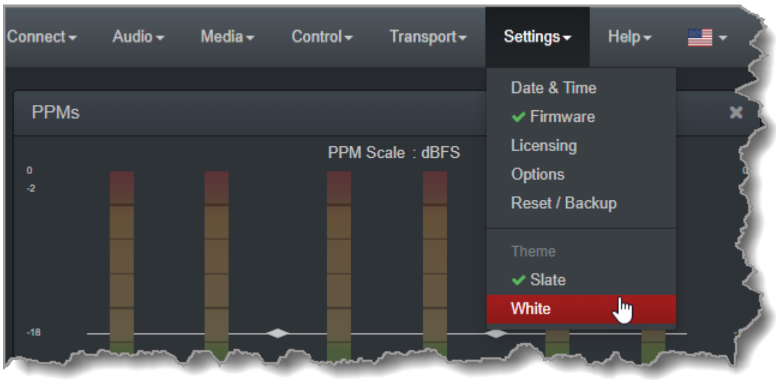
HTML GUI Menu Bar for Opening Panels

When you first open the HTML5 Toolbox Web-GUI the **Program Manager panel**, **Connections panel** and **PPMs panel** are loaded by default. If you retain cookies in your browser, any panels opened previously in the Web-GUI are automatically populated when you log in next. The default panel view is displayed on login if cookies have been cleared.

The green **Online** indication in the top left-hand corner of the Toolbox Web-GUI indicates it is online and can be used for codec control. A red **Offline** indication is displayed when the codec is unavailable. The **Upgrade** symbol is displayed when a new firmware version is available for the codec. Open the **Firmware panel** in the **Settings** menu to upgrade the codec with new firmware.

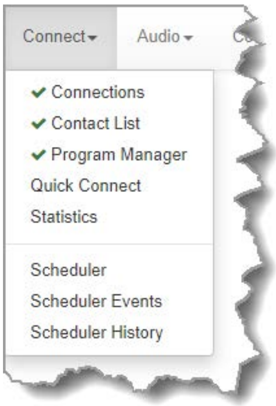
Adjusting the Theme

To adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI, navigate to the **Menu bar** at the top of the screen and click **Settings**, then click to select your preferred option. Note: this manual uses the **White** theme for most images.

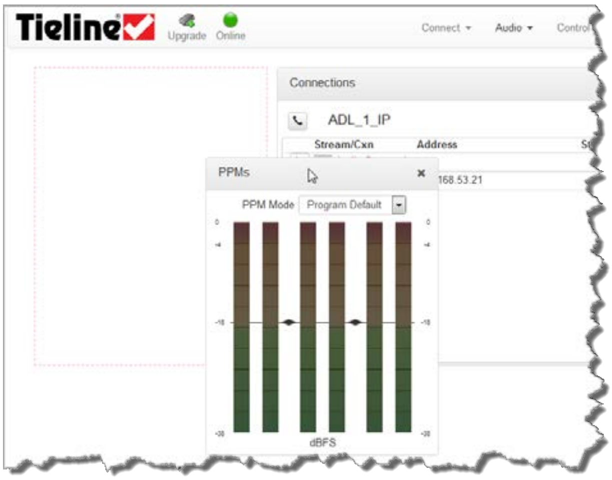


Opening a Panel & Adjusting Size or Screen Position

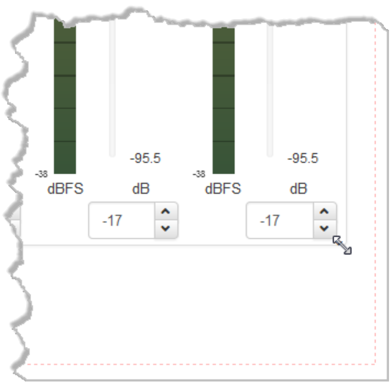
Click an item in the **Menu bar** to display available panel options, then click to select and open a panel. New panels automatically open in the top left of the screen. A green **Tick** adjacent to a panel name in the menu signifies it is already open in the web-GUI.



Position the mouse pointer over a panel's **Title bar** and click and drag to move a panel and reposition it in a preferred screen position.

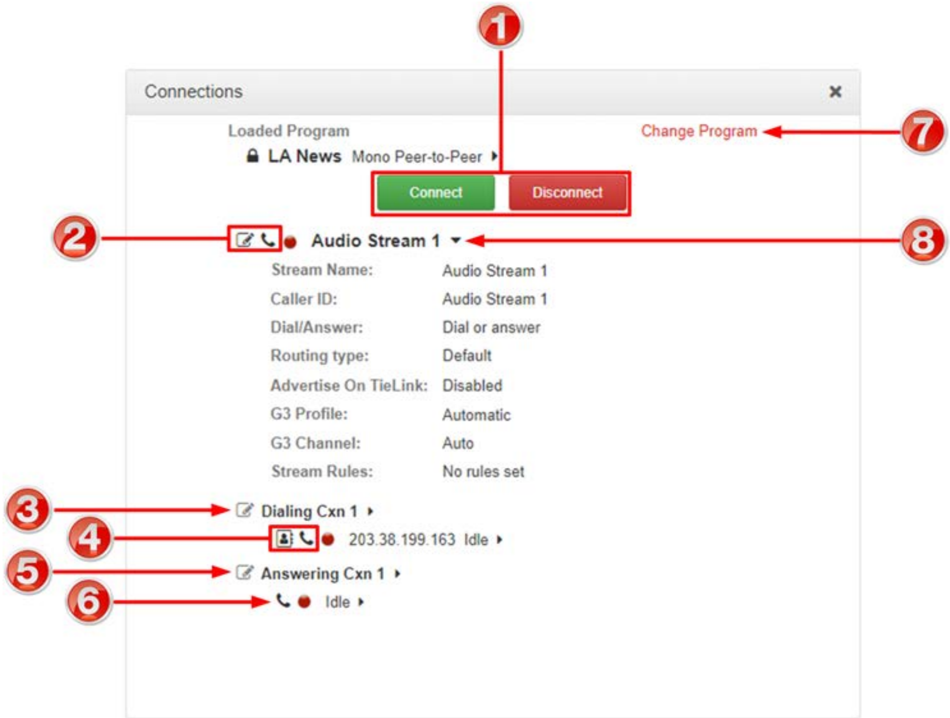


Some panels can be resized. Click the bottom right-hand corner of a panel supporting this feature and drag and resize as required.



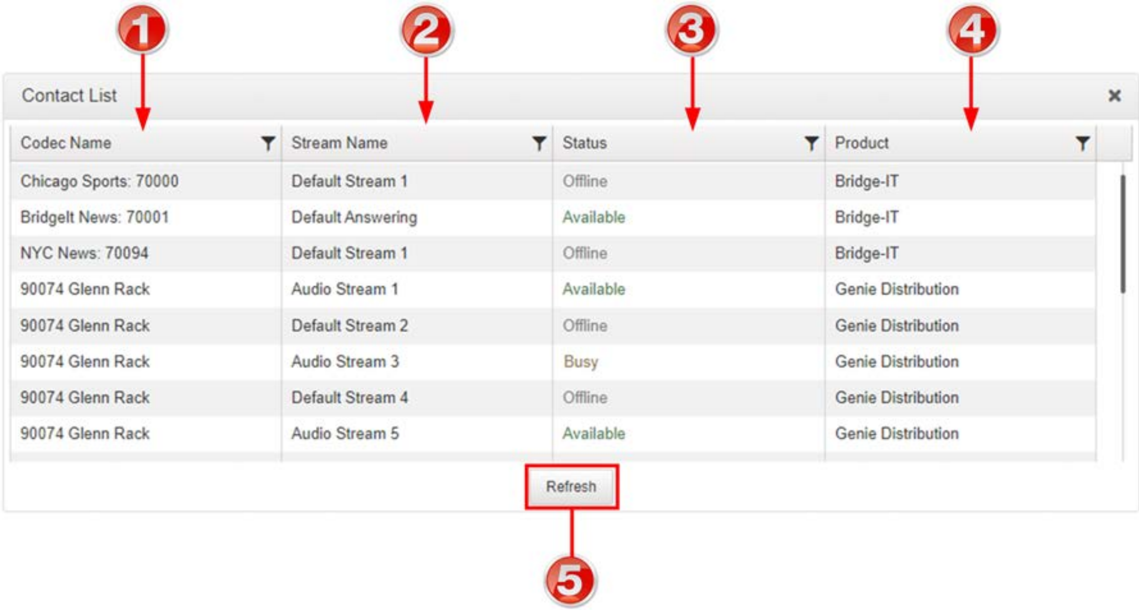
Connect Panels: Load & Connect Programs & Manage Audio Streams

Connections Panel



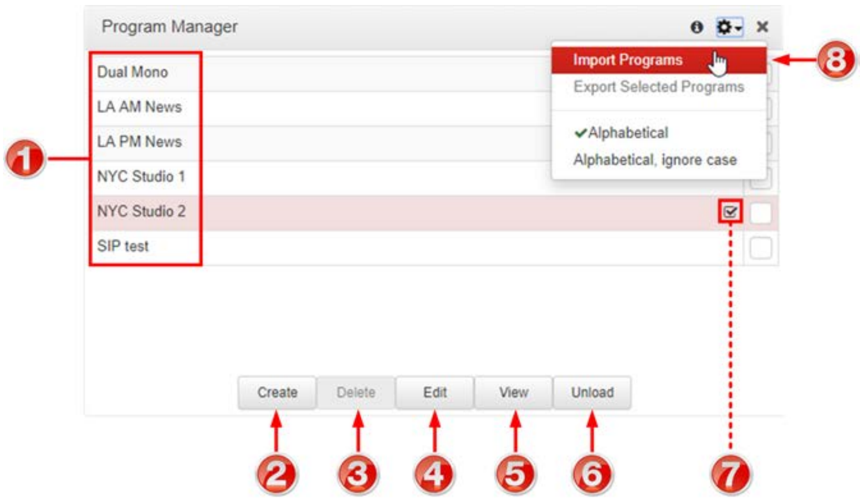
	Feature	Description
1	Program **Connect / Disconnect** button	Click to connect/disconnect all audio streams in a program.
2	Audio Stream **Edit** and **Connect / Disconnect** buttons	Click the **Edit** button to edit audio stream settings. Click the **Connect/Disconnect** button to connect/disconnect all connections in an audio stream.
3	Connection **Edit** button (dialing connection)	Click the **Edit** symbol to edit audio stream settings, including the IP address, or select a contact if a TieLink Traversal Server contact list is configured.
4	TieLink **Address Book** and Connection **Connect / Disconnect** button	Click the **Connect/Disconnect** button to connect/disconnect an individual connection. Also adjust the connection bit rate when a connection is active. Click the **Address Book** button to select a contact if a TieLink Traversal Server contact list is configured.
5	Answering connection **Edit** button	Click to edit answering connection settings
6	Answering Connection **Disconnect** button	Click to disconnect an answering connection
7	**Change Program**	Click to unload the current program and load a new program.
8	Show/Hide Arrow	Click to show/hide audio stream and connection details.

Contact List



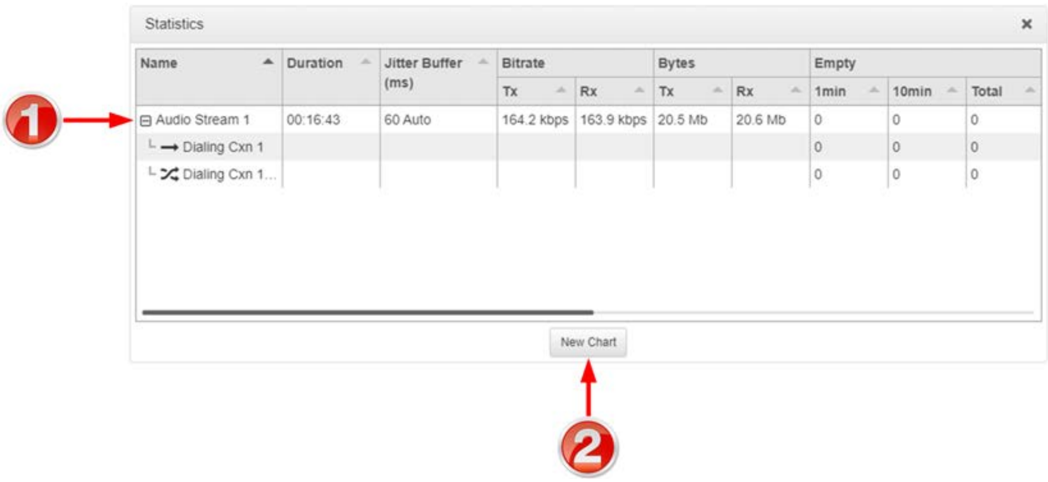
	Feature	Description
1	Codec Name	The name given to a codec to identify it when a program is created
2	Stream Name	The name given to a stream to identify it when a program is created
3	Status	Displays the status of an audio stream, which can be **Available**, **Busy** or **Offline**
4	Product	Displays the model of codec
5	Refresh	Click Refresh to refresh the codec and stream list

Program Manager Panel



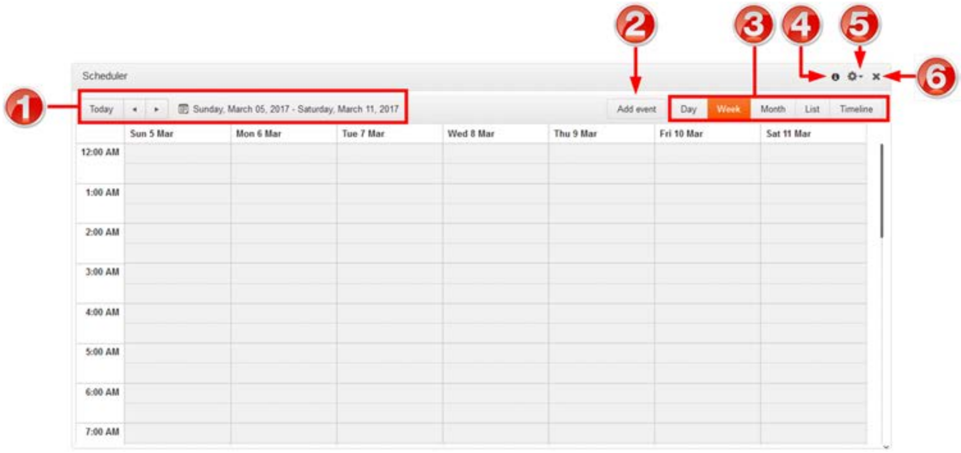
	Feature	Description
1	Program list	The list of saved programs in the codec
2	Create New Program button	Click to create a new program using the program wizard.
3	Delete Selected Programs button	Click to delete all selected programs
4	Edit Selected Program button	Click to edit the selected program
5	View Selected Program button	Click to view configuration settings for a selected program
6	Unload/Load program button	Click to load or unload a program
7	Loaded program symbol	Symbol identifies the currently loaded program
8	Import Programs and Export Selected Programs	Click the **Options** menu to select and import previously saved programs or export selected programs as a .zip file

Statistics Panel



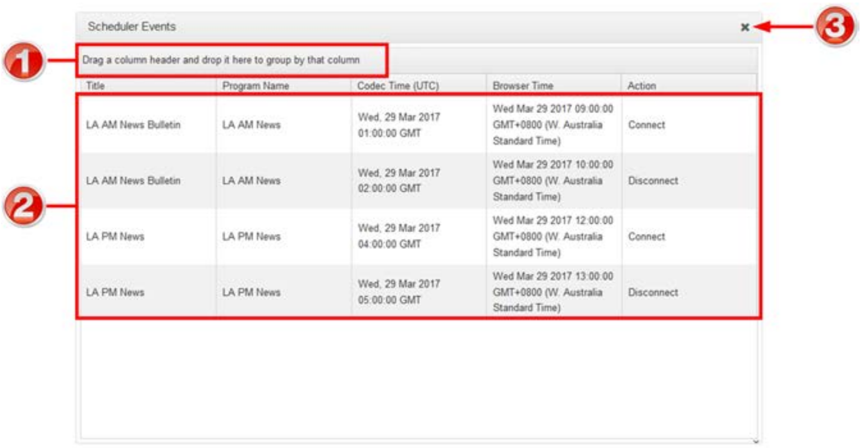
	Feature	Description
1	Expand/Collapse	Click to show/hide audio stream statistics, including packet arrival data info.
2	**New Chart** button	Click the **New Chart** button to select a **Data Series** and create a customized **Statistics panel**.

Scheduler Panel



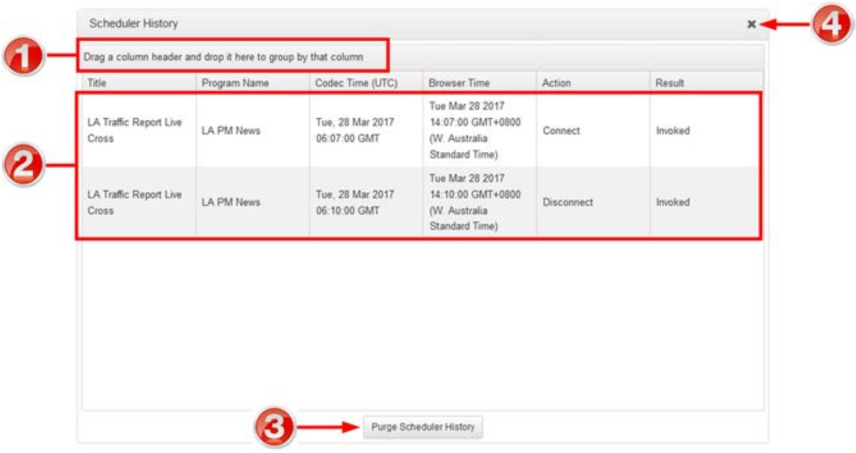
	Feature	Description
1	Date selection	Select the days you wish to view in the scheduler.
2	Add event button	Click to create a new scheduled event.
3	View type	Click to select the timeframe or timeline view of scheduled events.
4	**Information symbol**	Hover over the **Information symbol** to view information about the **Scheduler panel**.
5	**Options** menu	Click the **Options symbol** to view timezone options and generate a PDF view, or enable/disable the scheduler.
6	**Close** button	Click to close the panel.

Scheduler Events Panel



	Feature	Description
1	Grouping	Drag and drop a column header to group scheduled events by that column, e.g. **Codec Time**.
2	List of scheduled events	View scheduled events in a list view
3	**Close** button	Click to close the panel.

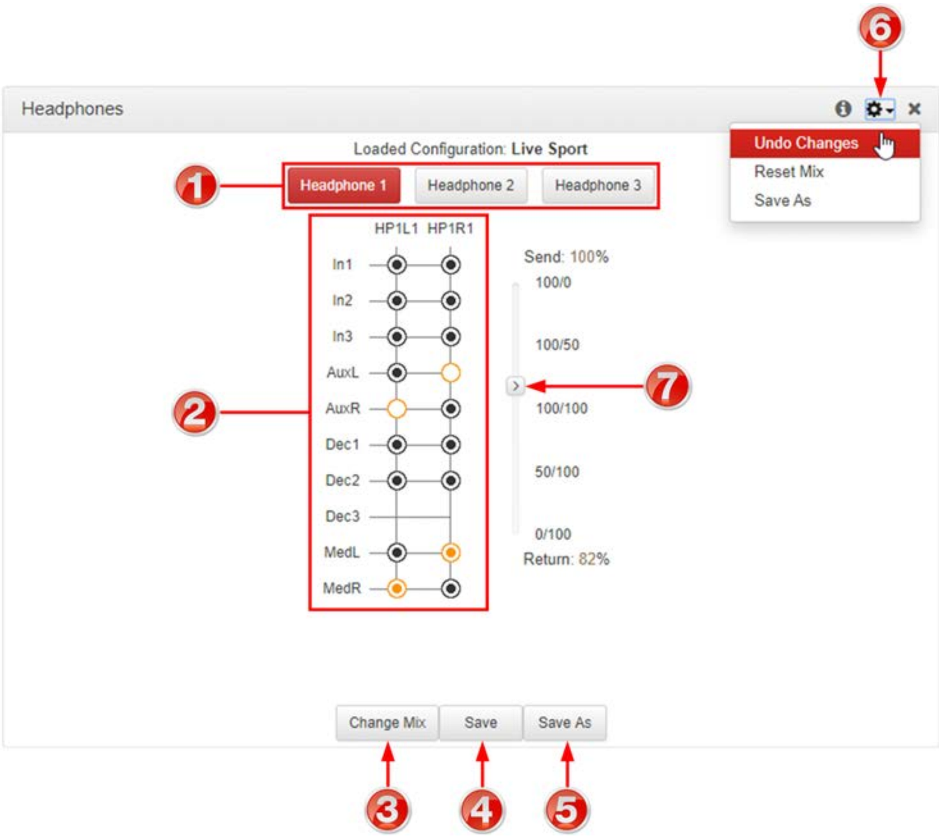
Scheduler History



	Feature	Description
1	Grouping	Drag and drop a column header to group event history by that column, e.g. **Codec Time**.
2	Event History List	View the codec's history of scheduled events.
3	**Purge Scheduler History** button	Click to clear all event history displayed.
4	**Close** button	Click to close the panel.

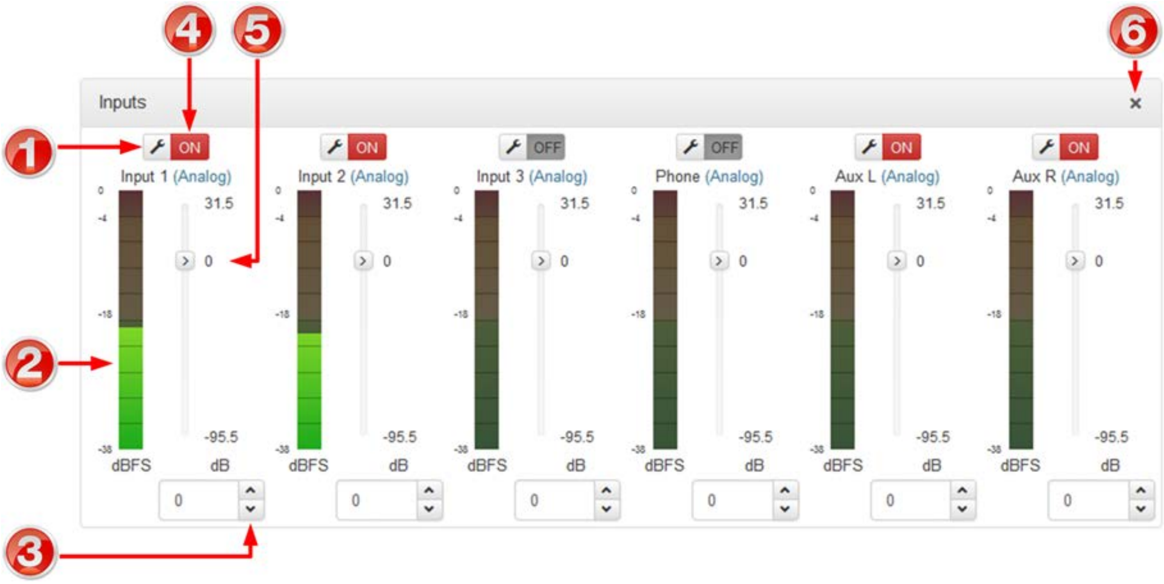
Audio Menu Panels

Headphones Panel



	Feature	Description
1	Headphone select buttons	Click to select and view a headphone mix and edit routing
2	Headphone crosspoint matrix	Click to select and deselect audio crosspoints to customize routing.
3	**Change Mix**	Click to select the default or Custom headphone mix to view and edit. Options include **Load**, **Delete** and **Rename** a mix
4	**Save**	Menu options available to undo any changes, reset the mix to the loaded matrix in the Matrix Editor, or save a new mix.
5	**Save as**	Click to save as a new custom headphone mix
6	**Options** menu	Click the **Options** menu to undo any changes, reset the mix to the loaded matrix in the Matrix Editor, or save as a new mix.
7	**Send/Return** balance slider	Click and drag to adjust the balance between outgoing and incoming audio.

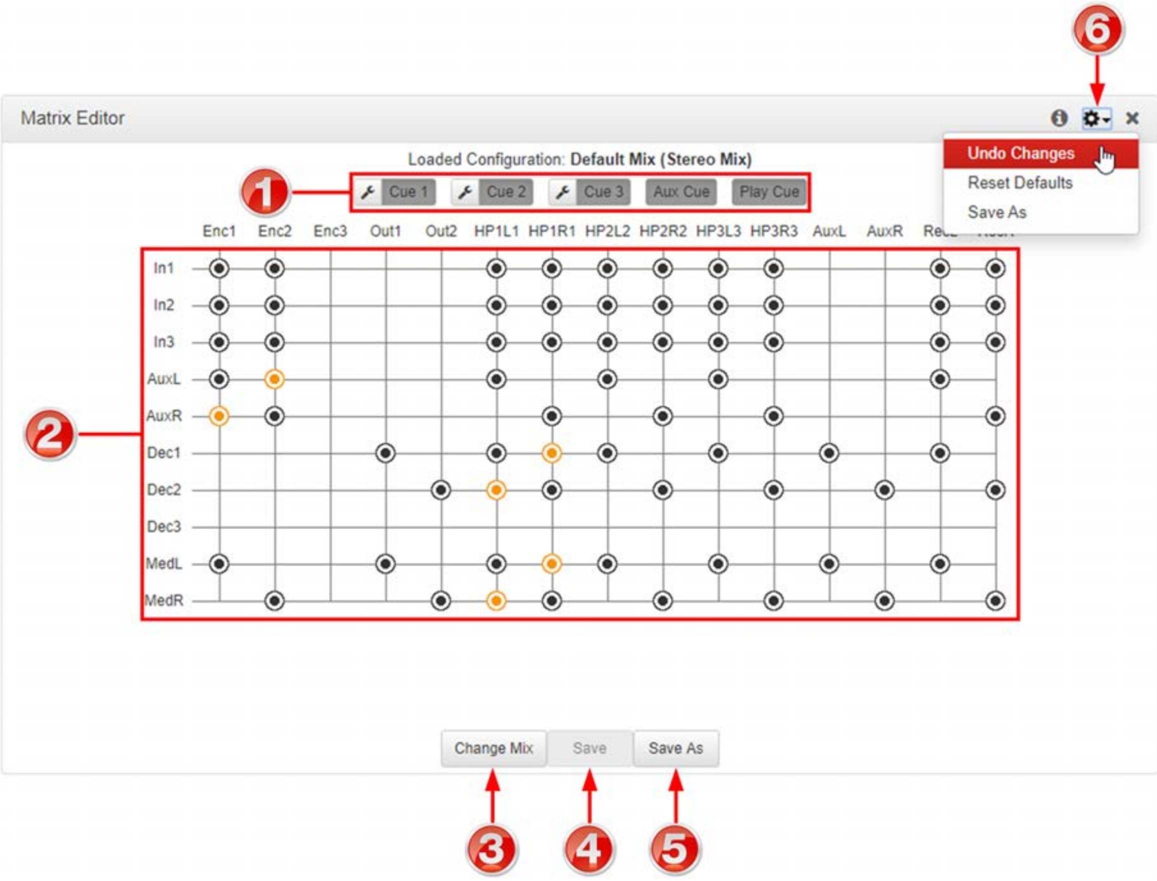
Inputs Panel



Important Note: Tieline codecs have different input configurations, therefore the image shown may not reflect the number of inputs displayed in your codec Web-GUI.

	Feature	Description
1	**Settings** button	Click to adjust input **Name**, **Type**, **Polarity Inverted** and **IGC** settings.
2	Input PPM meter	Input PPM meter.
3	Input gain adjustment	Click to adjust input gain in 0.5dB increments
4	**On/Off** button	Click to toggle an input on or off.
5	Input Sliders/Faders	Input gain control sliders/faders. Input gain adjustments in +/- 0.5dB increments
6	**Close** button	Click to close the panel.

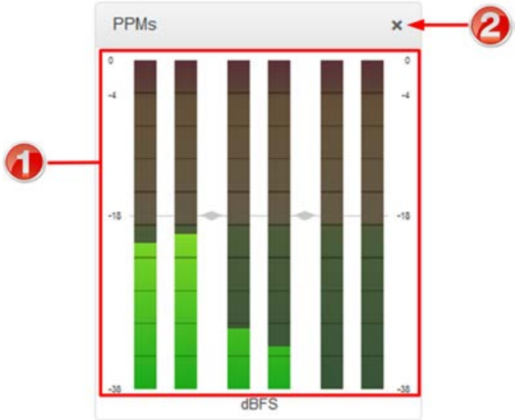
Matrix Editor Panel



	Feature	Description
1	Cue / Talkback select buttons	Click to select and view a Cue or Talkback mix and edit routing.
2	Matrix Editor crosspoint matrix	Click to select and deselect audio crosspoints to customize routing.
3	**Change Mix**	Click to select the default or Custom matrix mix to view and edit. Options include **Load**, **Delete** and **Rename** a mix
4	**Save**	Click to save changes to a custom mix
5	**Save as**	Click to save as a new custom matrix mix
6	**Options** menu	Menu options available to undo any changes, reset the mix to the factory default matrix mix for the program, or save as a new mix.

PPMs Panel

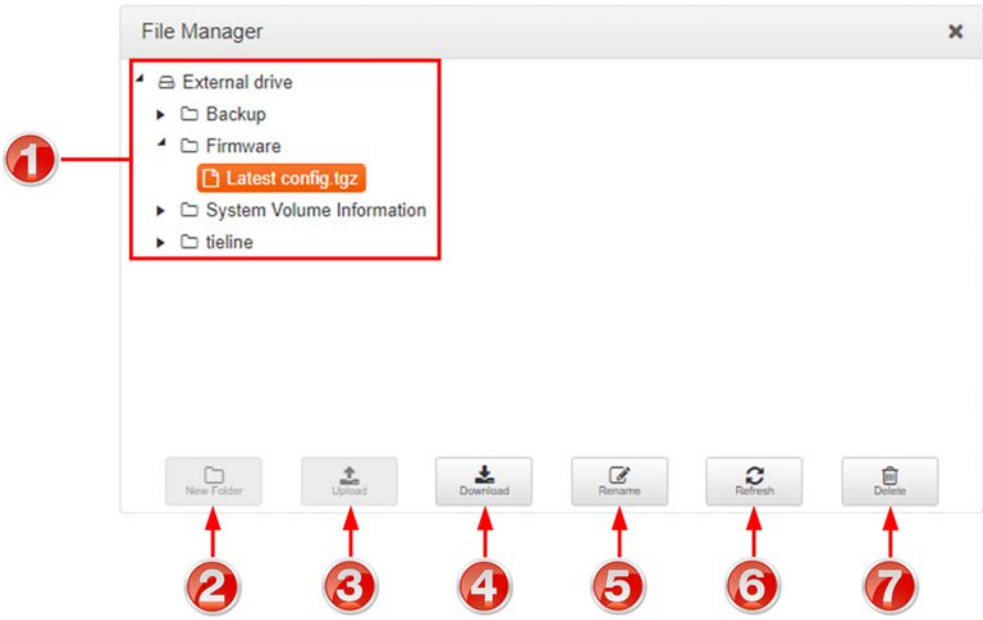
Note: Click and drag the bottom right-hand corner to expand the panel.



	Feature	Description
1	PPM Meters	6 PPM meters using dBFS metering
2	**Close** button	Click to close the panel.

Media Menu

File Manager

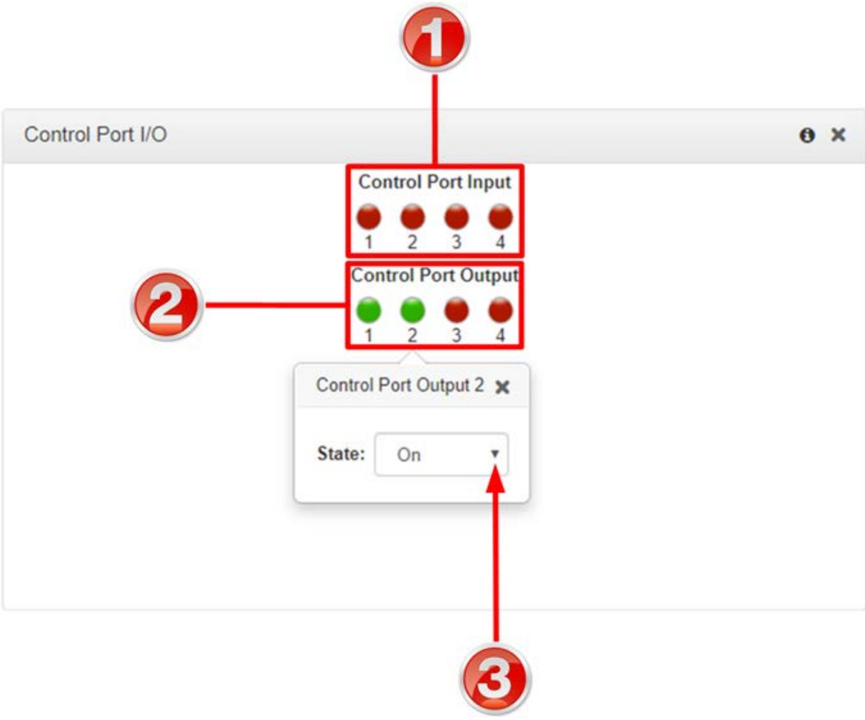


	Feature	Description
1	Folder and File View	View all folders and files on an external drive / removable media
2	**New Folder**	Select the drive or a folder and then click to create a new sub-folder
3	**Upload**	Click to select and upload a new file onto removable media
4	**Download**	Click to download a file onto removable media

5	**Rename**	Click to rename a selected file or folder
6	**Refresh**	Click to refresh the panel and view all files and folders
7	**Delete**	Click to delete a selected file or folder

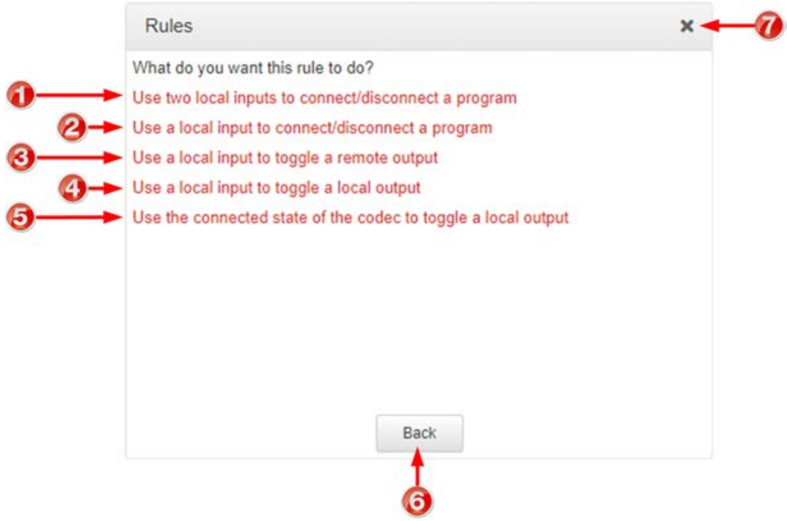
Control Menu

Control Port I/O



	Feature	Description
1	Control Port Input state	Displays the state of a control port input
2	Control Port Output	Displays the state of a control port output
3	**State**	Click a **Control Port Output** to change the On/Off state

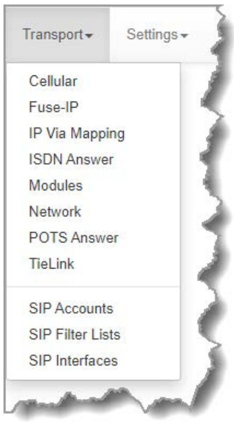
Rules Panel



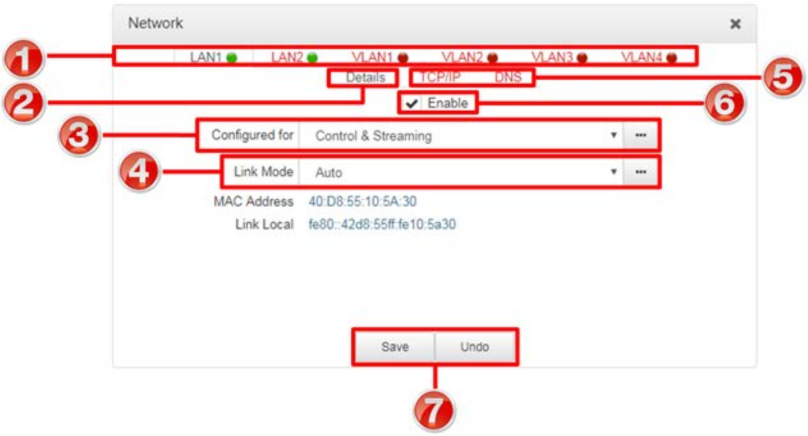
	Rule	Description
1	Use two local inputs to connect/disconnect a program	Click to configure connection and disconnection after different relay inputs are switched **ON**.
2	Use a local input to connect/disconnect a program	Click to configure connection and disconnection by toggling an input.
3	Use a local input to toggle a remote output	Click to configure a local relay input to synchronize with the state of a remote relay output.
4	Use a local input to toggle a local output	Click to configure a local relay input to synchronize with the state of a local relay output.
5	Use the connected state of the codec to toggle a local output	Click to configure a relay to toggle based on connection status.
6	**Back** / **Add New Rule** button	Click to add a new rule, or exit the rule creation function.
7	**Close** button	Click to close the panel.

Transport Panels

There are several **Transport** panels which can be opened in the Web-GUI. Each panel provides specific transport-related configuration settings and options. Click to select and open each panel.



As an example, the **Network panel** is displayed with network interface configuration options. A brief description of the other panels is also provided.



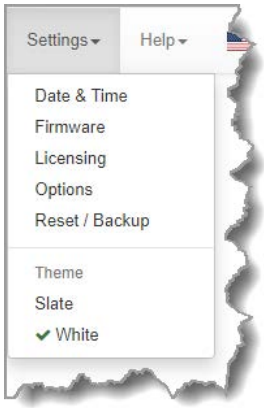
	Feature	Description
1	Network interfaces	Click to select and edit, or view network configuration settings for each Ethernet and VLAN interface.
2	**Details** tab	Display configuration options for a selected network interface, plus other device details.
3	Control/Streaming	Select **Control** and/or **Streaming** options for the selected interface.
4	**Link Mode**	Configure the Ethernet, VLAN or Wi-Fi link speed (10/100/1000/Auto) and whether an interface will operate in Full-Duplex or Half-Duplex mode.
5	**TCP/IP** and **DNS** tabs	Select the **TCP/IP** tab to configure **IPv4/IPv6** address details. Select the **DNS** tab to specify DNS addresses and domains to search.
6	**Enable check-box**	Select the check-box to enable an interface.
7	**Save/Undo** button	Click **Save** to store settings, or click **Undo** to revert to previously configured settings.
8	**Cellular**	Click to open the **Cellular panel** and configure USB air cards or an internal cellular module with a SIM card
9	**Fuse-IP**	Click to open the **Fuse-IP panel** and configure Fuse-IP bonding
10	**IP Via Mapping**	Configure default **Primary**, **Secondary** and **Tertiary** interfaces.
11	**ISDN Answer**	Click to open the panel and configure ISDN Answering settings.

12	**Modules**	Click to edit hardware module configuration.
13	**Network**	Click to open the **Network panel** and configure network settings.
14	**POTS Answer**	Click to open the panel and configure POTS Answering settings.
15	**TieLink**	Click to view the TieLink panel to enable, disable and prioritize interfaces, configure ports and Fuse-IP interfaces, and configure STUN server settings.
16	**SIP Accounts**	Click to open the panel and edit SIP account settings. Up to 6 SIP accounts are supported.
17	**SIP Filter Lists**	Add trusted network codecs to the **URI Whitelist**. Add SIP URIs to the **URI Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec.
18	**SIP Interfaces**	Click to open the panel and configure port, proxy and **Via** settings for the **SIP1** and **SIP2** interfaces. The codec supports dialing over these SIP interfaces simultaneously.

Settings Panels

There are several **Settings** panels which can be opened in the Web-GUI. Each panel provides different codec configuration settings and options. Click to select and open each panel. It is also possible to change the HTML5 Web-GUI theme.

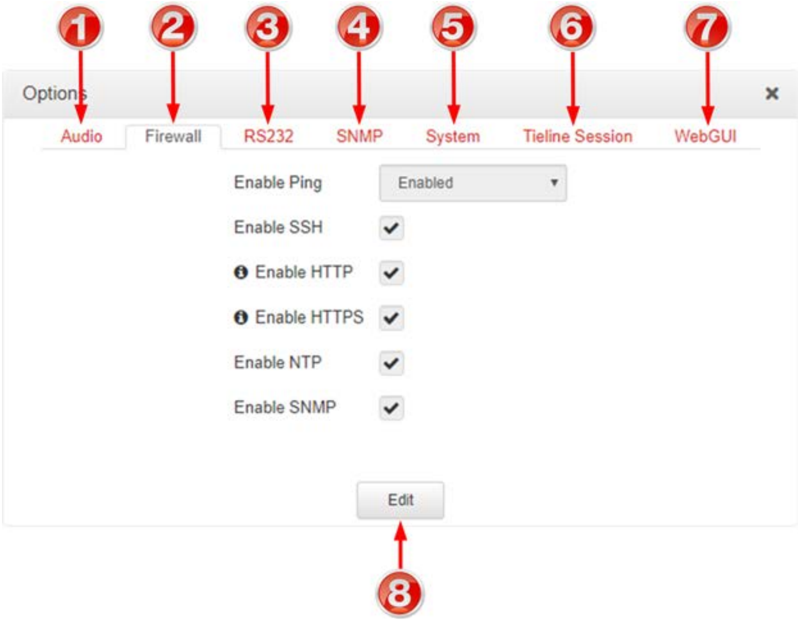
As an example, the **Options panel** is displayed with a brief description of the other panels available.



Settings panels

	Feature	Description
1	**Date and Time**	Click to open the panel view and sync the codec to NTP time.
2	**Firmware tab**	Click to open the panel; view software versions, download firmware and perform an upgrade.
3	**Licensing tab**	Click to open the panel; select a license file and install it in the codec.
4	**Options tab**	Click to open the panel and adjust a wide range of codec audio, firewall, RS232, SNMP, system, session data and Web-GUI settings.
5	**Reset / Backup**	Click to open the panel; reset codec default settings and perform backup/restore of codec programs and settings.
6	**Theme**	Adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI; options include **White** or **Slate**.

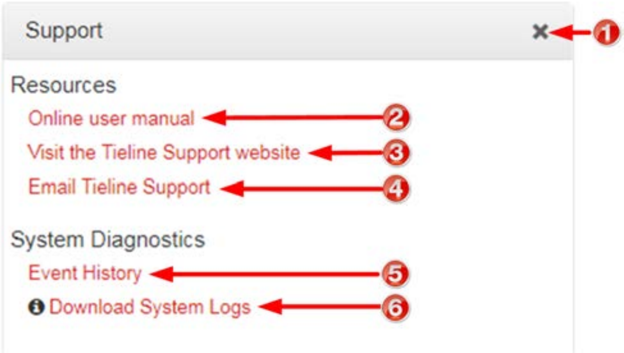
Options panel



	Feature	Description
1	**Audio** Settings	Configure various audio settings on the codec, including analog input PPM units, reference levels and phantom power voltage.
2	**Firewall** Settings	Enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.
3	**RS232** Settings	Click **Baud rate** to adjust the baud rate used by the RS-232 serial port on the codec. Select the check-box to **Enable Flow Control**.
4	**SNMP** Settings	Configure SNMP settings in the codec.
5	**System** Settings	Configure various system settings, including: Country setting; select the **Lock Loaded User Program** check-box to lock the currently loaded program in the codec; enable and assign a hostname to the codec to provide a flexible way of identifying the codec on a network; configure IP audio data packets for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks.
6	**Tieline Session**	Edit the Tieline session and alternative session port used by the codec.
7	**WebGUI**	Includes settings such as **Quick Connect Enabled**, **CCC Enabled** (select this option to enable Cloud Codec Controller use with the codec), enable **CSRF**, enter a **Browser Title**, adjust the **SSL Port**.
8	**Edit button**	Press to edit settings in the **Options panel**.

Help Panels

Support Panel



	Feature	Description
1	**Close** button	Click to close the panel.
2	User manual link	Click to open the codec user manual in a new browser, or view support information (Note: the codec name displayed will vary by product type)
3	Support website link	Click to visit the support page on the Tieline website.
4	Email Tieline Support	Click to email Tieline support.
5	Event History	Click to download user-viewable event logs
6	Download System Logs	Click to download diagnostic information that can be sent to Tieline support

About Panel

Details of the codec Toolbox and firmware version, as well as the codec serial number. Note: the codec name displayed will vary by product type.

	Feature	Description
1	**Close** button	Click to close the panel.

Language Selection

The HTML5 Toolbox Web-GUI offers language support for several languages.

1. Click on the **Language** drop-down menu arrow in the top right-hand corner of the Web-GUI page.
2. Select the preferred language to display.

33.1 Using the HTML5 Toolbox Quick Connect Web-GUI

The HTML5 Quick Connect Web-GUI is designed for simple peer-to-peer connections and non-technical users. It has a reduced feature-set and allows users to:

1. Load existing programs in a codec via the **Program Loader panel** and then dial via the **Quick Connect panel**.
2. Use the **Quick Connect panel** to create and dial a simple peer-to-peer connection using IP/SIP, ISDN or POTS.

See Opening the HTML5 Web-GUI and Login for details about launching the standalone HTML5 Quick Connect Web-GUI. The **Quick Connect panel** can also be launched from the **Connect** menu in the HTML5 Toolbox Web-GUI.

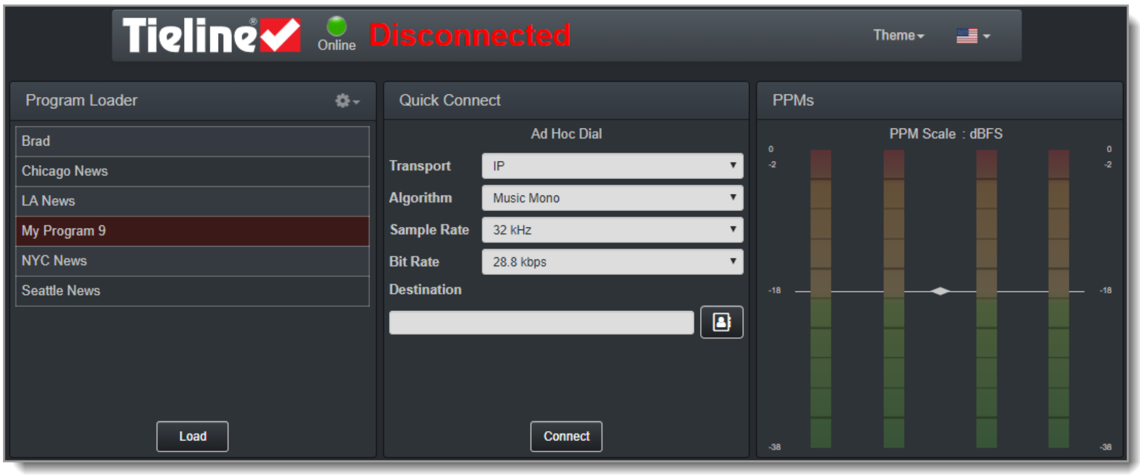
By default, the HTML5 Quick Connect panel is enabled in each codec. To disable this feature:

1. Navigate to **Settings** in the main HTML5 Toolbox web-GUI and click to open the **Options panel**.
2. Navigate to **Quick Connect Enabled** and deselect the check-box.

Important Note: Simple peer-to-peer connections are not saved as programs with unique names. Details of the last ad hoc dial are retained in the **Quick Connect panel**, even after a program is loaded and unloaded using the **Program Loader panel**.

Launching the HTML5 Quick Connect Web-GUI

1. Type the codec IP address in your web-browser.
2. Click to launch the HTML5 Toolbox Quick Connect Web-GUI.
3. Enter the authentication **Password** for the codec and click **OK**.
4. The panels in the Quick Connect Web-GUI will automatically be displayed.

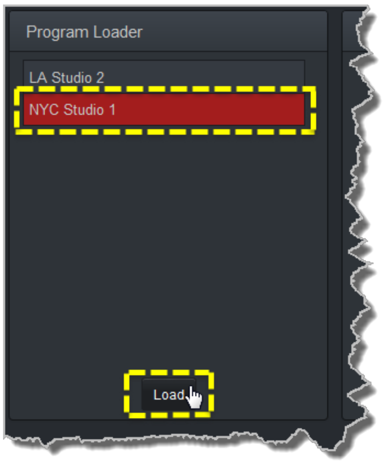


Important Note: To change the password using the codec **TOUCH SCREEN** navigate to **Settings > System > Web GUI > Password**. Use the **TOUCH SCREEN** keypad to enter a new password and tap **Done** in the top right-hand corner to save the new setting.

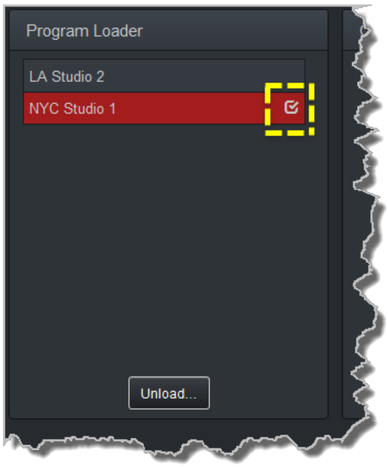
Loading and Unloading an Existing Program

If programs are saved in the codec they are displayed in the **Program Loader panel**.

1. Click to select a program in the **Program Loader panel** and click the **Load button** to load it in the codec.



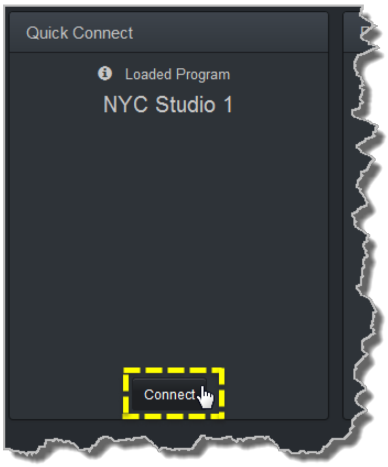
2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and the **Load button** changes to an **Unload button**.



3. To unload a program click the **Unload button**.

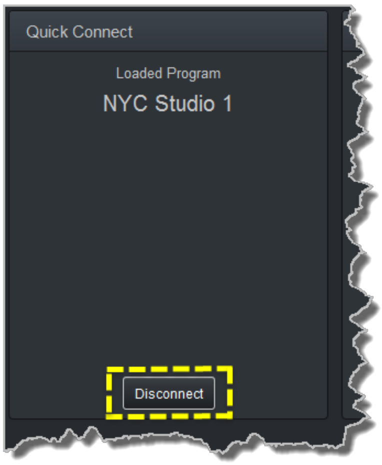
Dial a Loaded Program

1. Click the **Connect button** in the **Quick Connect panel** to dial a loaded program. Note: After connecting, the **Connect button** changes to a **Disconnect button**.

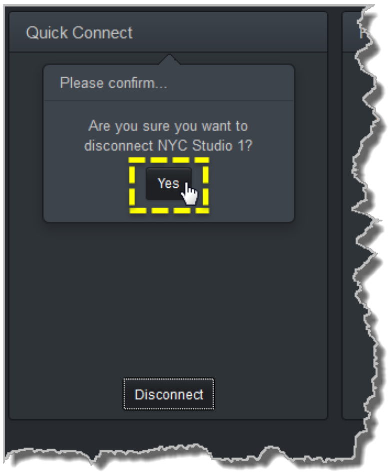


Disconnect a Loaded Program

1. Click the **Disconnect button** in the **Quick Connect** panel.



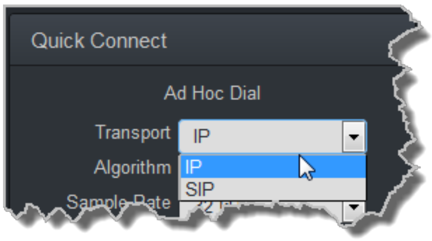
2. Click **Yes** in the confirmation dialog to disconnect the connection.



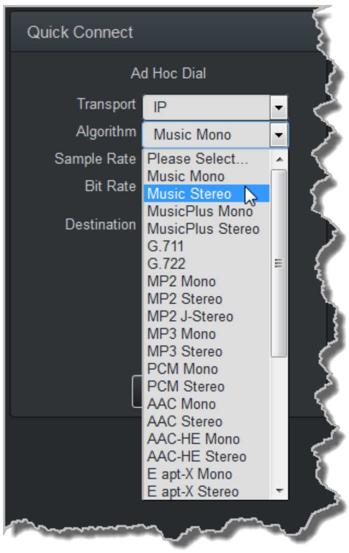
Dial Peer-to-Peer over IP with Quick Connect

> **Important Notes:**
> - Click the **Unload** button in the **Program Loader panel** if a program is currently loaded.
> - To dial over IP using **SIP**, click the drop-down arrow for **Transport** and select **SIP**.

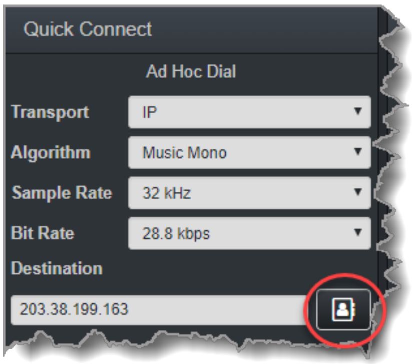
1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **IP**.



2. Click the drop-down **Algorithm** menu and select an algorithm.



3. Click the select the appropriate **Sample Rate** and **Bit Rate** for the connection. Note: If only one sample rate is available this will be automatically selected.
4. Click in the **Destination** text box and enter the IP address of the destination codec, or click the **Address Book button** to select a preconfigured TieLink contact.

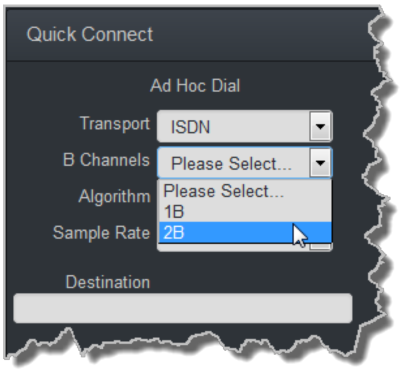


5. Click the **Connect button** to dial.

Dial Peer-to-Peer over ISDN with Quick Connect

Important Notes: Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.

1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **ISDN**.
2. Click the drop-down **Algorithm** menu and select an algorithm.
3. Click the drop-down **B Channels** menu and select the number of channels required for this connection.

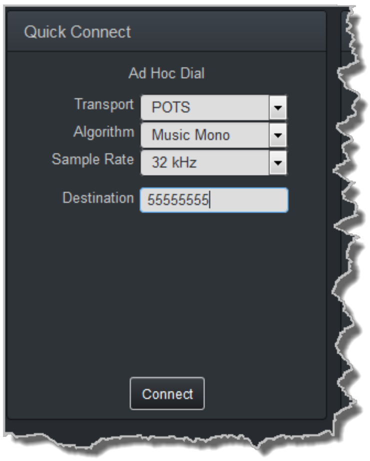


4. Click to select the appropriate **Sample Rate** for the connection.
5. Click in each **Destination** text box in turn to enter the ISDN number for each B Channel.
6. Click the **Connect button** to dial.

Dial Peer-to-Peer Over POTS with Quick Connect

Important Note: Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.

1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **POTS**.
2. Click the drop-down **Algorithm** menu and select an algorithm. The connection bit rate is configured automatically. Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bit rate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.
3. Enter the phone number in the **Destination** text box.



4. Click the **Connect button** to dial.

Monitoring PPMs

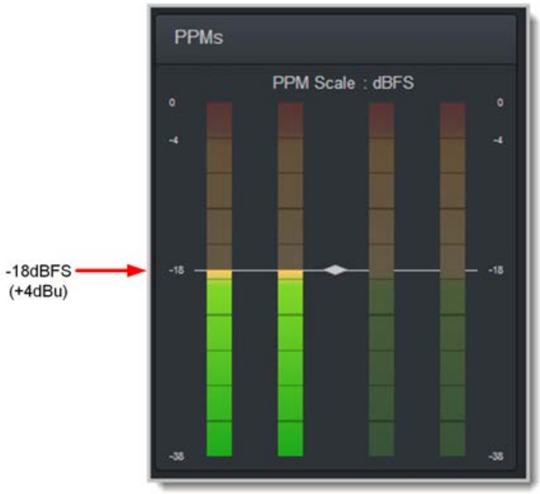
Set audio levels so that audio peaks average at the nominal 0vu point indicated below on the PPM meters. By default, the **PPM METERS** on the front of the codec, on the **TOUCH SCREEN**, or in the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels.

The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** reference scale setting.

The default Tieline G5 audio reference scale displayed on the PPMs is -38dBFS to 0dBFS (e.g. Merlin, Genie, Bridge-IT and ViA codec families). Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 18dB of headroom from the nominal 0vu point.

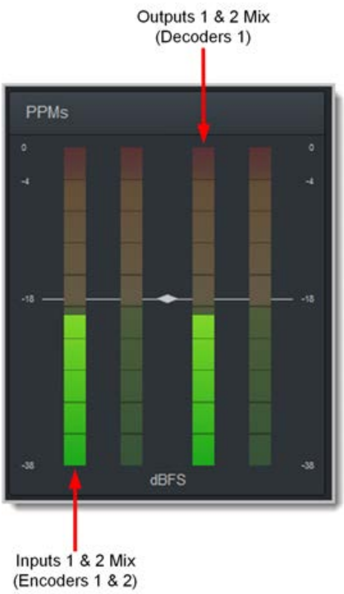
The audio reference level settings in the codec are:

	Reference Setting	Description	dBFS	dBu Equivalent
1	Tieline G5	PPM meter low point	-38dBFS	-16dBu
		Nominal 0vu reference level	-18dBFS	+4dBu
		Level at which audio will clip/distort	0dBFS	+22dBu
2	Tieline G3	PPM meter low point	-29dBFS	-11dBu
		Nominal 0vu reference level	-14dBFS	+4dBu
		Level at which audio will clip/distort	0dBFS	+18dBu



Mono and Stereo PPM Metering

When connected with a mono program the codec will display a mix of inputs 1 and 2 on **PPM1**. **PPM 3** displays the level of return audio.



When connecting with a stereo program, the codec displays audio on **PPM1** and **2** for inputs 1 and 2 and **PPM 3** and **4** for the return audio.

33.2 Configure Programs with the Connections Panel

The **Connections panel** allows:

- The configuration of new programs.
- Editing of existing programs.
- Loading and unloading of programs.

The **Connections panel** delivers the ability to edit destination settings for an audio stream when other audio streams are connected, without unloading a program. Please note: If you need to use an existing program as a template for a new program, or add a custom matrix mix to a program, please use the **Program Manager panel** to create a program.

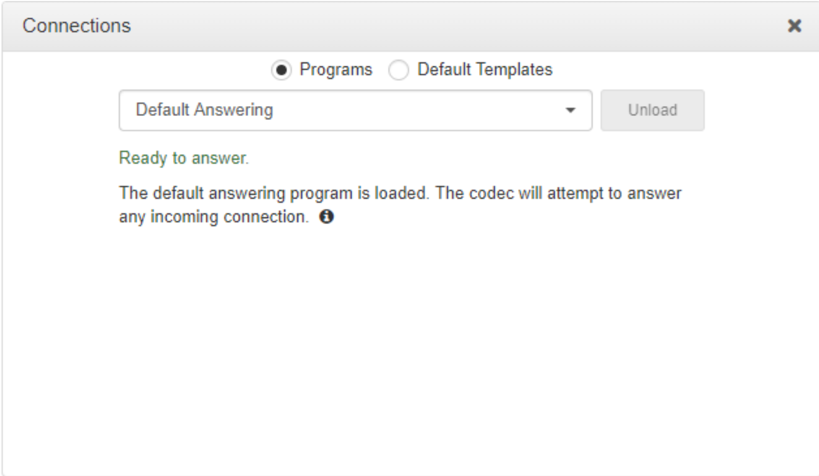
Configuring New Programs

Important Note: There are some limitations when creating and editing programs using the **Connections panel** in comparison to using the **Program Manager panel**. In the **Connections panel** is not possible to:

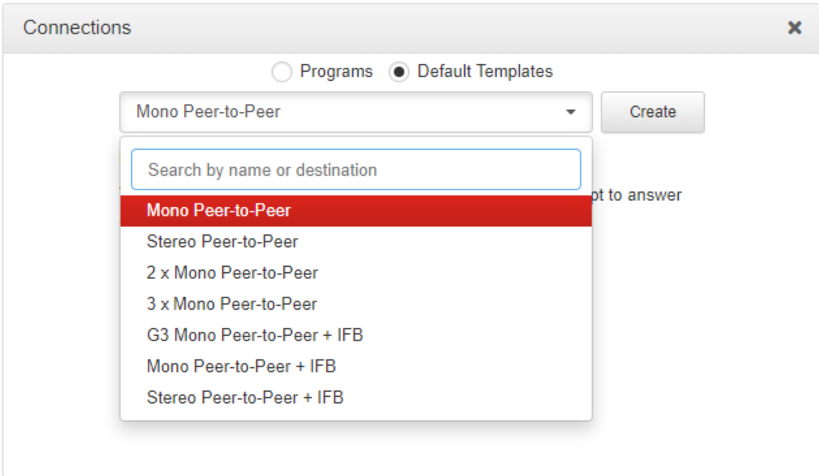
- Name or rename a program.
- Apply a custom matrix.
- Manage Program level Rules.
- Create a program from a copy of an existing program.

To configure new programs using the **Connections panel**:

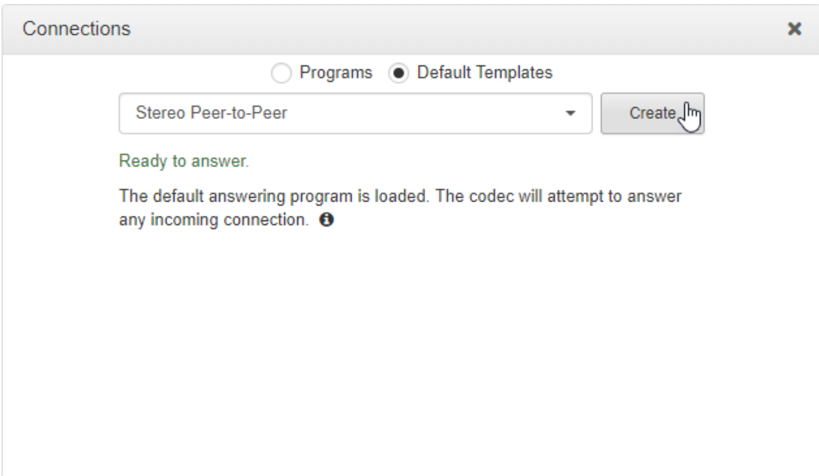
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



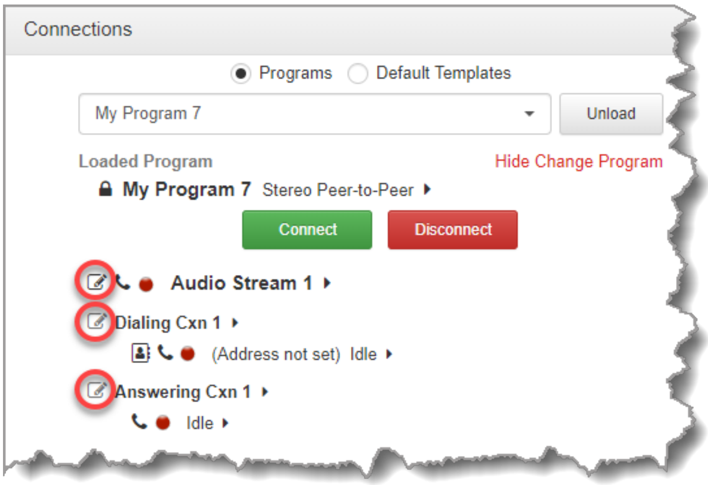
2. Select **Default Templates** and then click the drop-down arrow to select one of the supported default templates in the codec. Note: The default templates available will vary from codec-to-codec.



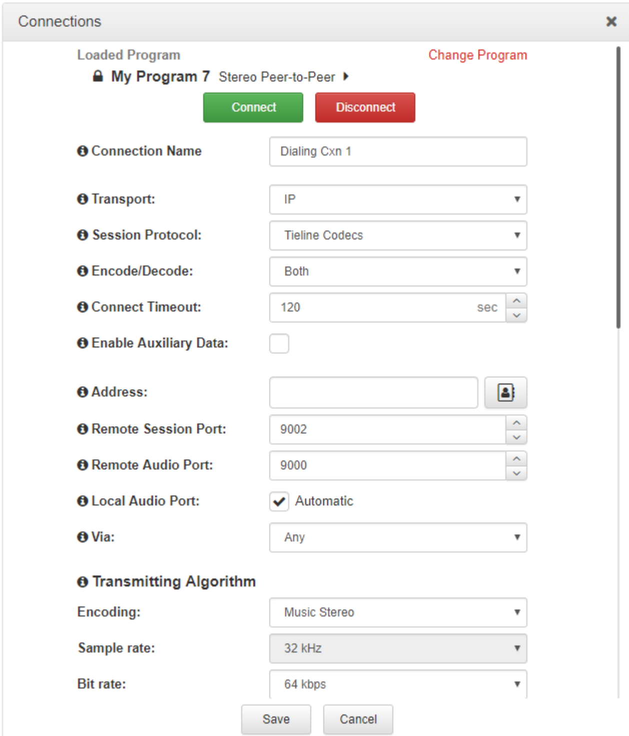
3. Click **Create** to add the new program.



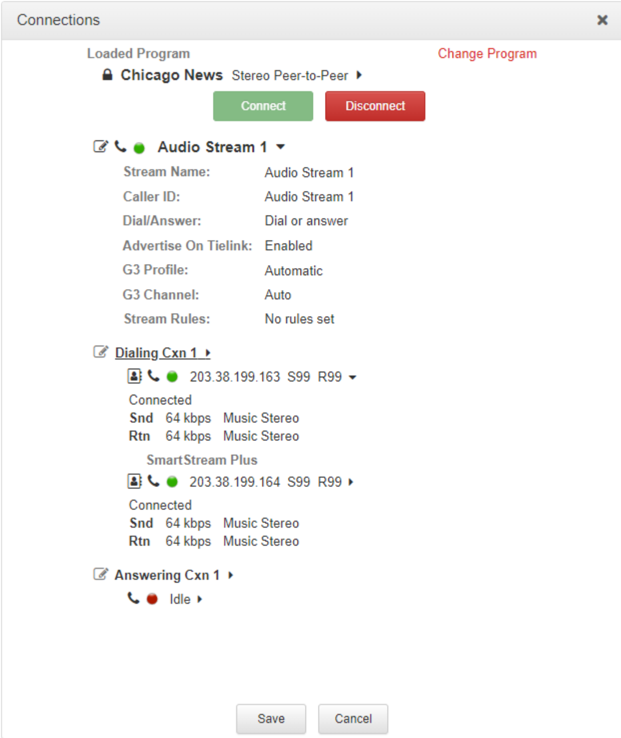
4. To configure new connection settings, or edit existing connection settings, click the **Edit symbol** .



5. The **Edit symbol** expands each audio stream, and each dialing and answering connection, to reveal configuration settings. Hover over the **Information symbol** to view details of each configuration setting.



6. Connections can also be connected, disconnected and managed from within the **Connections panel**.


Connections
Loaded Program
Change Program
Chicago News Stereo Peer-to-Peer
Connect
Disconnect
Audio Stream 1
Stream Name: Audio Stream 1
Caller ID: Audio Stream 1
Dial/Answer: Dial or answer
Advertise On Tielink: Enabled
G3 Profile: Automatic
G3 Channel: Auto
Stream Rules: No rules set
Dialing Cxn 1
203.38.199.163 S99 R99
Connected
Snd 64 kbps Music Stereo
Rtn 64 kbps Music Stereo
SmartStream Plus
203.38.199.164 S99 R99
Connected
Snd 64 kbps Music Stereo
Rtn 64 kbps Music Stereo
Answering Cxn 1
Idle
Save
Cancel

33.3 Configuring IP Settings

Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **Network** to view and configure Ethernet and VLAN interface settings in the Web-GUI.

Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

IPv4 versus IPv6

An IP address is a unique address to identify a device on a TCP/IP network. Your codec uses dual IP protocol stacks to allow your codec to work on both IPv4 and IPv6 networks. Tieline codecs support both DHCP (default) IP addressing and static IP addresses for dialing IPv4 connection endpoints.

If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN. Usually this will be set up at the studio end so you can dial into the studio from the remote codec.

Support for IPv6 connections allows you to use IPv6 infrastructure to connect to other codecs globally.

Configuring Ethernet Ports and VLANs

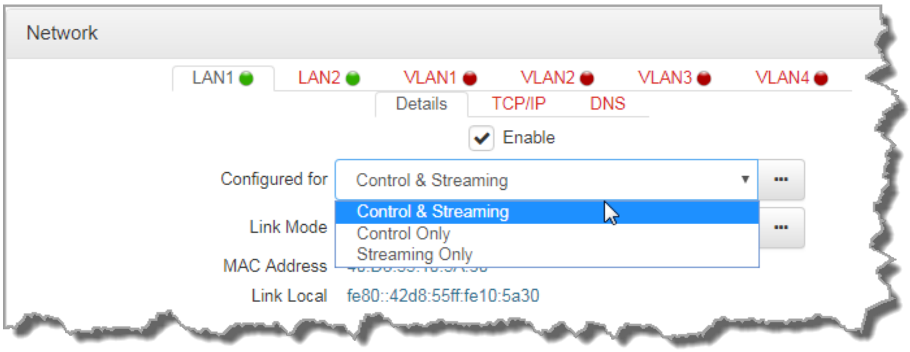
The codec features two physical Ethernet LAN ports, two USB ports and up to four additional VLAN interfaces.

VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec. These interfaces can be configured to stream or control the codec, or do both.

As an example, if only one physical Ethernet interface is available, VLANs or USB interfaces can be used to stream SmartStream PLUS audio or separate codec control and streaming functions as required. IP interfaces can be configured for:

- Controlling audio: codec control and command only from a selected IP interface.
- Controlling and Streaming: stream audio and control and command the codec from a selected IP interface.
- Streaming audio: stream audio only from a selected IP interface.
- Nothing: Disable the Ethernet port from streaming audio and codec command and control (**VLANs** only).

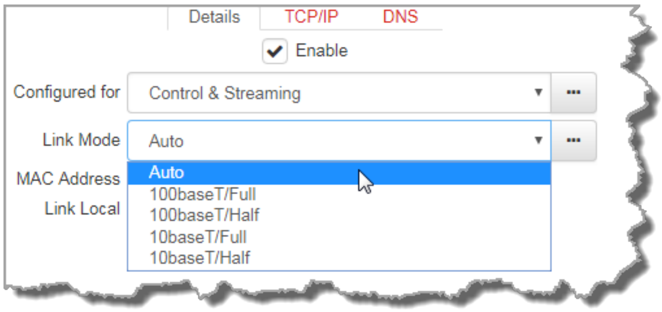
Select the **Details tab** in the **Network** panel to edit control and streaming settings. Select the **Enable** check-box to activate each interface. **LAN1** and **LAN2** are enabled by default. Note: An interface with a green status indication is enabled and has an active LAN cable attached. A LAN interface with red text and a yellow status indication is enabled, however a network cable is not attached, e.g. **LAN2** in the following image. An interface in red text with a red status indication is disabled, e.g. **VLANs** in the following image.



Configure Link Mode

It is possible to configure the Ethernet link speed (10/100/1000/Auto) and whether each available interface operates in full-duplex or half-duplex modes.

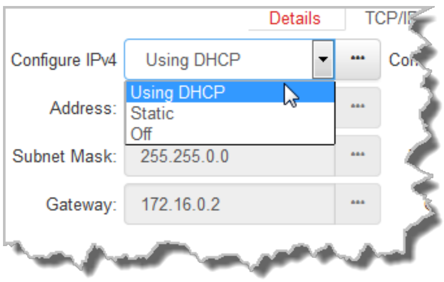
1. Click the drop-down **Link Local** arrow and select the preferred setting.



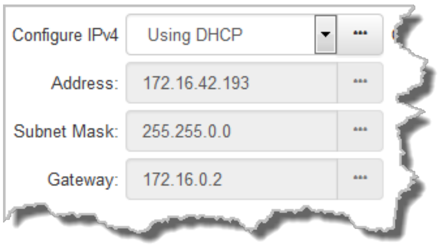
Click **Save** to store the new setting.

IPv4 Address Configuration

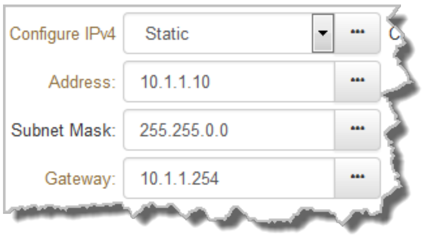
Click to select the **TCP/IP** tab in the **Network** panel to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the **Configure IPv4** drop-down menu. To ignore IPv4 settings select **Off**.



DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider, or to your own local area network (LAN). By default the codec is configured for DHCP-assigned IP addresses.



Static IP addresses are fixed addresses that are recommended for studio installations, so that IP address dialing remains the same over time for incoming codec connections.



Click **Save** to store all configuration settings.

Note: The **Subnet Mask** is used by the TCP/IP protocol to determine whether a host is on the local subnet or on a remote network. The default **Gateway** is the router linking the codec's subnet to other networks. See your IT administrator for more details.

IPv6 Address Configuration

An IPv6 address is represented by 8 groups of 16-bit hexadecimal values separated by colons (:). The drop-down **Configure IPv6** menu provides three address configuration options:

1. **Automatically**: An address is automatically assigned to the codec when you connect the codec to an IPv6 router. This process is similar to how an IPv4 DHCP address is assigned.
2. **Manually**: Select to enter static IPv6 address details.
3. **Off**: Select to ignore IPv6 address details.

Important Note: Select **Off** in the **Configure IPv6** drop-down menu if you are not using IPv6 to connect to another device. This ensures your codec will attempt to connect using IPv4 at all times.

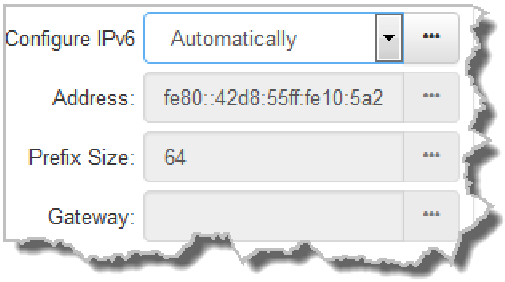
Types of IPv6 Addresses

There are two types of addresses displayed in the IPv6 section:

1. IPv6 address (normally global): A router-allocated IP address with 'global' visibility, details of which are displayed in the **Address, Prefix size** and **Gateway** text boxes.
2. Link Local: A local address which can only be used to connect to another device directly over a LAN. This address is allocated by the codec internally based on MAC address details.

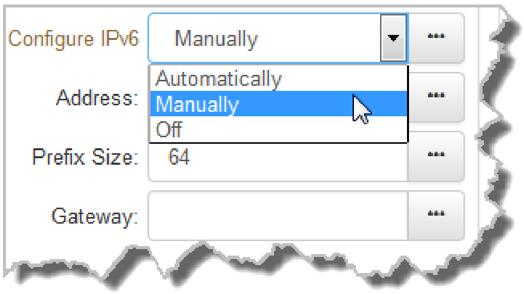
Auto Address Assignment

By default the codec is configured for connecting to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



Manual IPv6 Address Assignment

1. To configure IPv6 address details into the codec manually, select **Manually** and enter details into the **Address**, **Prefix** and **Gateway** text boxes.

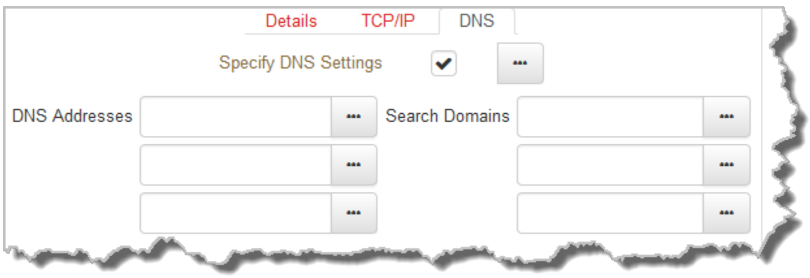


2. Click **Save** to store all configuration settings.

Specifying DNS Settings

It is possible to specify Domain Name Server (DNS) settings to allow easy look up of codecs within the specified **DNS Addresses** or **Domains**.

1. Select the **DNS** tab in the **Network** panel to configure settings.



2. Click **Save** to store all configuration settings.

IP Via Mapping

When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface will be used to dial a connection. The default **Via** interfaces in order of use when available are:

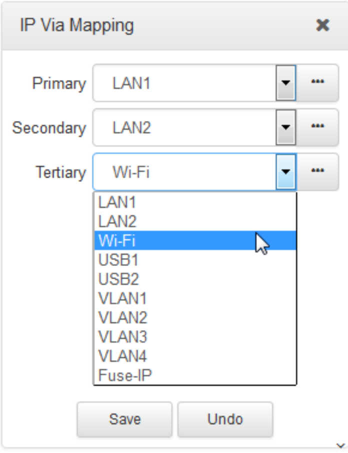
1. **LAN1** Ethernet port (default **Primary** Via interface)
2. **LAN2** Ethernet port (default **Secondary** Via interface)
3. Internal built-in Wi-Fi (default **Tertiary** Via interface)

Important Note: VLAN interfaces have features similar to physical Ethernet interfaces. However, your network administrator will need to configure VLAN support throughout your network for them to be supported in your codec.

Reconfigure Default Primary, Secondary and Tertiary Interfaces

It is possible to reconfigure the default **Primary** (LAN1), **Secondary** (LAN2) and **Tertiary** (Wi-Fi) interfaces in the codec. As an example, you may want to select **Primary** as the dialing interface in a program and then copy this program onto multiple codecs. However, the actual primary interface used at each location can vary for each codec. For one codec it may be a LAN interface and for another it may be a USB cellular interface. This allows you to configure site-specific settings to suit the available network interfaces at different remote locations.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** and then click **IP Via Mapping** to open this panel.
2. Click the drop-down arrow for each interface to select the preferred default setting.



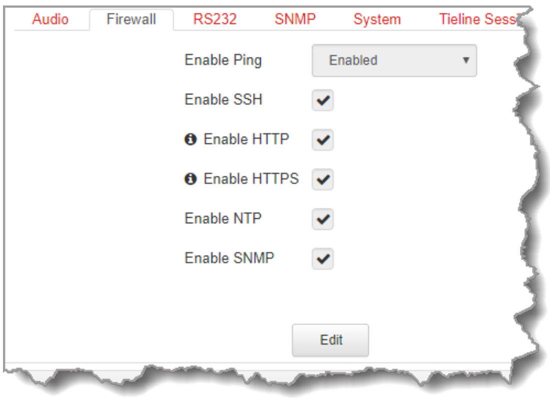
3. Click **Save** to store the configuration.

Important Note: Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary **Via**.

Configure Firewall Settings

The **Firewall** menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.

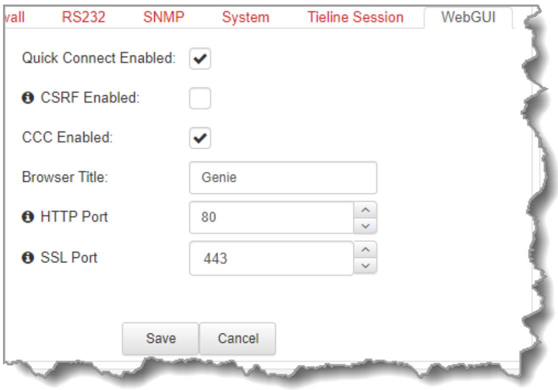
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **Firewall** and then click **Edit** to adjust settings. Click **Save** to store new configuration settings.



Configure Cross-Site Request Forgery

CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

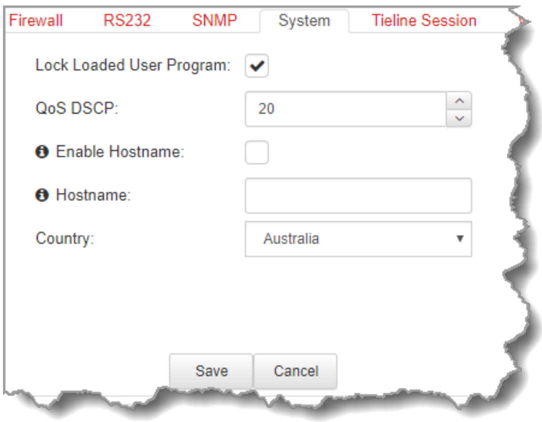
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **WebGUI**, then click **Edit** and select the **CSRF Enabled** check-box to enable this feature. Click **Save** to store new configuration settings.



Configuring QoS

The codec can be configured to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (**DSCP**) field within the header of data packets transmitted over the network.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** and then click **Options** to open the **Options panel**.
2. Select **System** and then click **Edit**. Click in the **QoS DSCP** text box and enter the preferred value.
3. Click **Save** to store configuration settings.



Important Note: Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

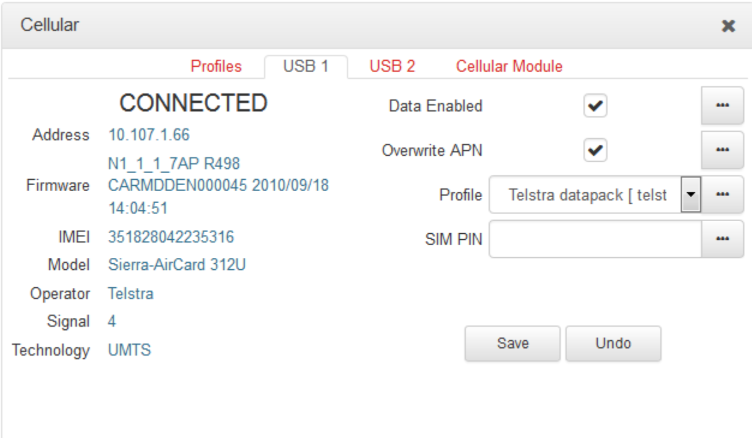
33.4 Configure Cellular Modems and Module

It may be necessary to add a custom access point number (APN) profile if a USB modem or cellular module does not connect to a cellular network. Often this is necessary when you are using different SIM cards in an unlocked USB modem. E.g. When using SIM cards from various carriers in different countries. As a rule of thumb, Tieline recommends always using the **Overwrite APN** menu setting and entering the correct APN for your Telco. This ensures use of the correct data APN. It may also be necessary to enter a SIM PIN unlock code.

Up to 10 custom access point profiles can be added to the codec. Before configuring the custom access point you need to obtain the access point details from the cellular network provider; this is normally found on a Telco's website. Usually a Telco will list internet and MMS APN information and you need to enter the internet APN details in the codec, as well as the correct authentication type. See Adding Cellular Access Points and a SIM PIN for more details on configuration via the codec **TOUCH SCREEN**.

Cellular Configuration

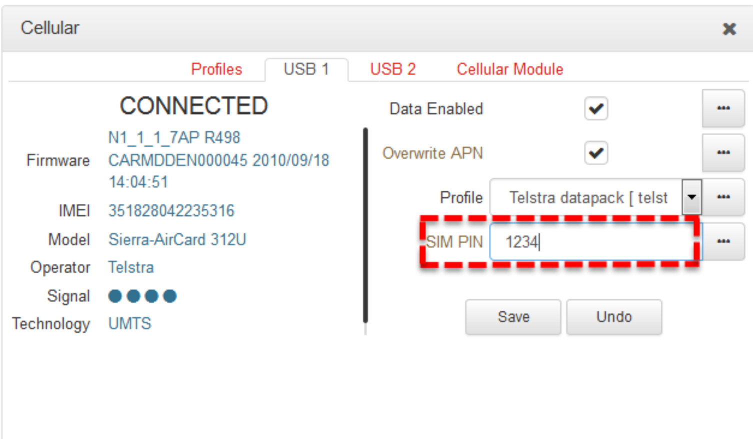
Open the HTML5 Toolbox Web-GUI and click **Settings** and then **Cellular** to view and configure a cellular modem attached to the codec. Data is enabled by default for USB modems.



Cellular panel with modem attached to USB1

Entering a SIM PIN Unlock Code

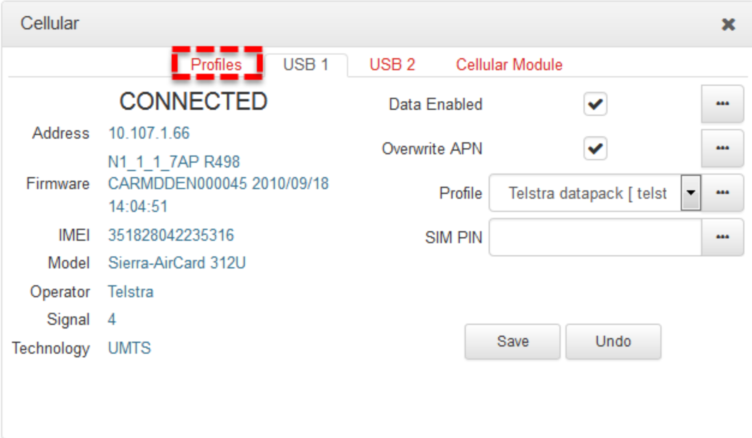
1. Enter the **SIM PIN** in the text box to unlock a SIM card using this feature in a USB modem or cellular module if it is locked.



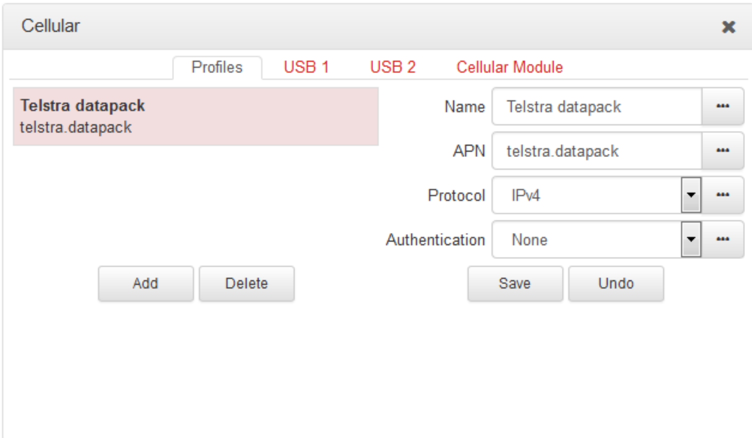
2. Click the **Save** button.

Adding a Custom APN

1. Click to select the **Profiles** tab to add a custom APN profile.

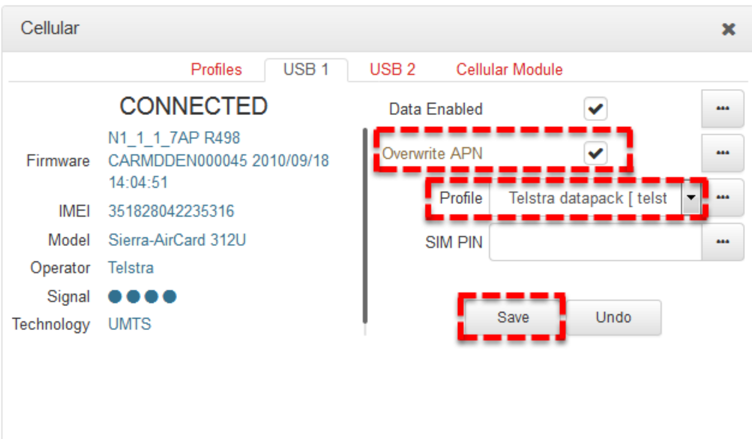


2. Enter the APN details and click the **Save** button.



Important Note: Up to 10 custom access point profiles can be added to the codec.

3. Click the tab for the modem or module you are configuring and select the **Overwrite APN** checkbox, then choose the correct profile in the **Profile** drop-down menu and click **Save**.



33.5 Configuring TieLink Settings

The TieLink Traversal server centralizes Tieline codec contact list management and provides self-discovery of codecs within customized ‘call-groups’. It also provides NAT traversal to simplify connections. The following needs to be configured before TieLink connections can be created.

Getting Started with TieLink

1	Register a TieServer Domain	Visit www.tieline.com/register to register a TieServer Domain if you don’t have Report-IT Enterprise.
2	Configure TieLink Traversal Server	1. Log in to TieLink at www.tieserver.com/tielink and add codecs to TieServer. 2. Create a contact list. 3. Set the answering codec as a member of the contact list. 4. Set the dialing codec as a follower of the contact list.
3	Enable TieLink and Connection Interfaces	Enable TieLink and enable interfaces for use with TieLink (‘TieLink panel' in Toolbox, or TieLink screen in the codec).
4	Adjust TieLink Interface Priority	Adjust TieLink interface settings in the Toolbox web-GUI ‘TieLink panel’, e.g. priority of use.
5	Configure Answering Codecs	1. A default program will automatically be ‘available’ in TieLink and answer calls, or 2. Create a program with “Advertise on TieLink” selected, and 3. Lock this program in the codec.
6	Configure Dialing Codecs	On the dialing codec: 1. Refresh the ‘Contact List panel’ in the Toolbox web-GUI. 2. Create a program and select a TieLink contact to dial.

To learn how to obtain a domain or add a codec to a TieServer Domain, please visit www.tieline.com and download the TieLink user manual. There are two ways of connecting using the TieLink Traversal Server:

1. Use the front panel of a codec to select a contact and dial without creating and saving a program.
2. Create a Program with TieLink configuration parameters specified.

TieLink Settings in Toolbox

TieLink needs to be enabled in the codec before it can communicate with the TieLink Traversal Server. To facilitate connections the **TieLink panel** is used to configure interfaces and options like STUN. To view the panel open the HTML5 Toolbox Web-GUI and click **Transport** and then **TieLink** to open the **TieLink panel**.

1. Enable TieLink

Click **Enable TieLink** to enable TieLink connectivity.

2. Enable Interfaces

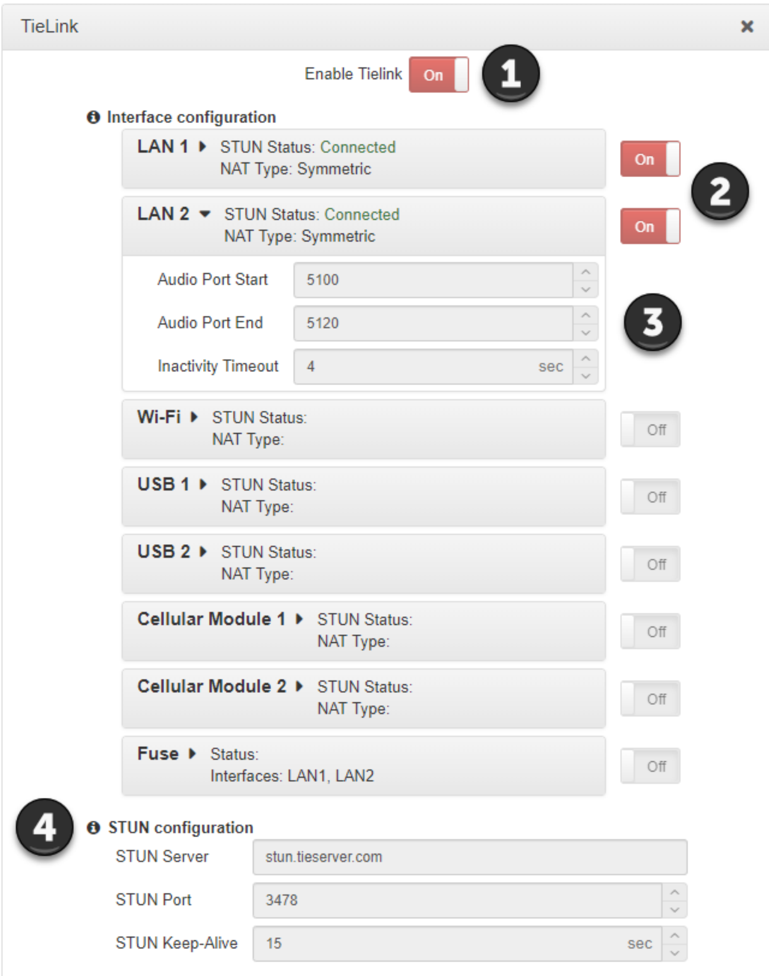
Click the **ON/OFF** button to enable each individual interface and/or select the interfaces used for **Fuse-IP** connections (Note: This is a simpler way to configure Fuse-IP. TieLink supports one connection to a single codec using Fuse-IP).

3. Adjust Tieline Interface Settings

Adjust TieLink interface ports, inactivity timeouts and Fuse-IP interface selections.

4. STUN Configuration

A STUN server is required to use TieLink. Adjust TieLink **STUN configuration** and use a different STUN server if required, e.g. to use a different port.

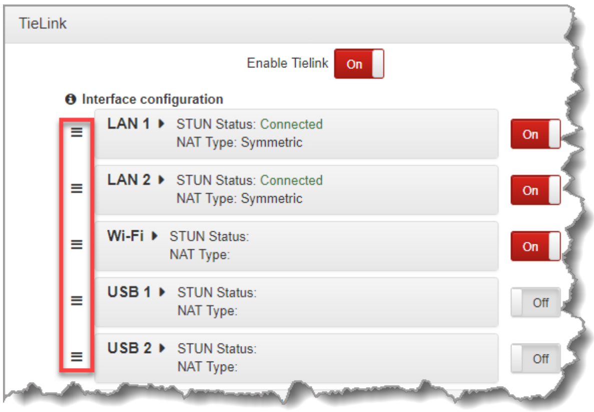


A green **TieLink Symbol** is displayed at the top of the Toolbox HTML5 Web-GUI when TieLink has been enabled successfully. Note: the symbol is orange if STUN is not successfully enabled on any interface.

Reorder TieLink Interface Priority

When the codec attempts to connect to a TieLink contact, local interfaces are used in the order in which they are listed. For example, if both **LAN1** and **Wi-Fi** are configured for use when dialing a contact, TieLink will attempt to use these interfaces in the order in which they are listed in the panel. It is possible to reorder local TieLink interface priority in the **TieLink panel** in Toolbox.

1. Click **Edit** in the TieLink panel.
2. Click the **Drag Handle** to the left of an interface to drag and reorder it.



3. Click **Save** to save the new settings.

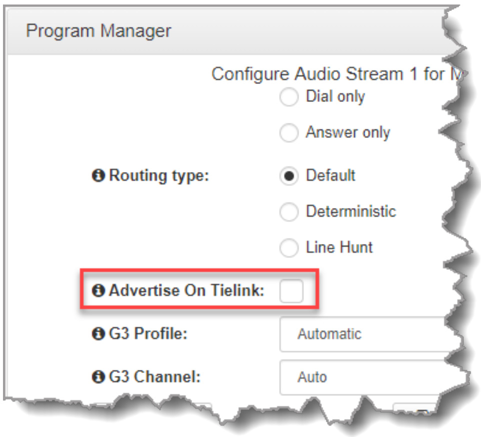
Configuring TieLink Connections in the Program Manager Panel

Important Note: The following settings can also be edited in the **Connections panel**.

Include a Codec's Answering Audio Streams in TieLink Contact Lists

A default program will automatically display audio streams as ‘available’ in TieLink and allow answering of calls from codec 'followers' in a contact list. When creating custom programs, from an answering perspective the **Program Manager panel** is used to select which answering streams are "Stream Members" visible within a TieLink contact list. I.e if an answering stream is 'advertised' on TieLink then is becomes an available endpoint for other codecs to dial.

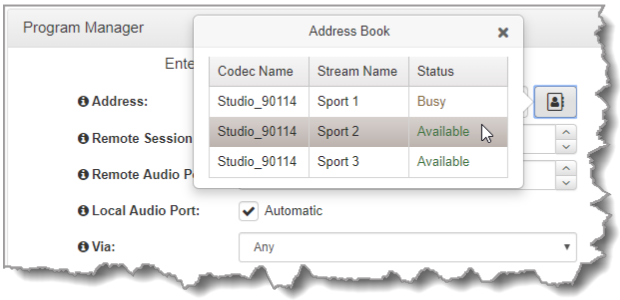
When creating a program select the **Advertise On TieLink** check-box to register answering audio streams on a loaded program with the TieLink Traversal server. These audio streams will be visible in TieLink contact lists which include this codec's audio streams. In other words, the answering audio streams will appear as available to dial (or "Busy" when connected) by other codec 'followers' in the same contact list.



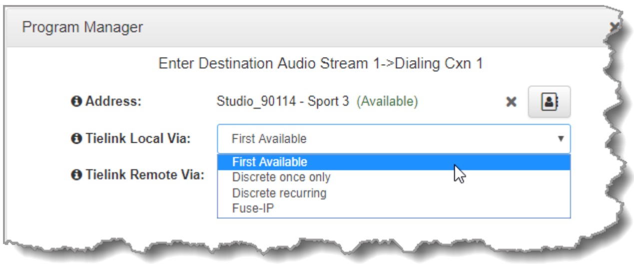
Select a Contact to Dial using TieLink

From a dialing perspective, the **Program Manager panel** is used to select a contact to dial from within a TieLink contact list that has been shared with a codec. When creating a program:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.

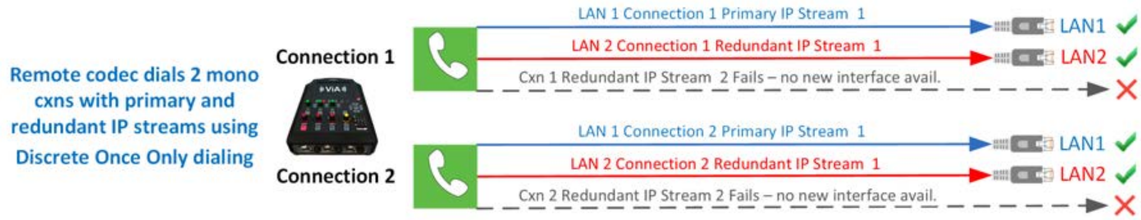


2. Select the interface options to use on local and remote codecs when dialing TieLink connections.

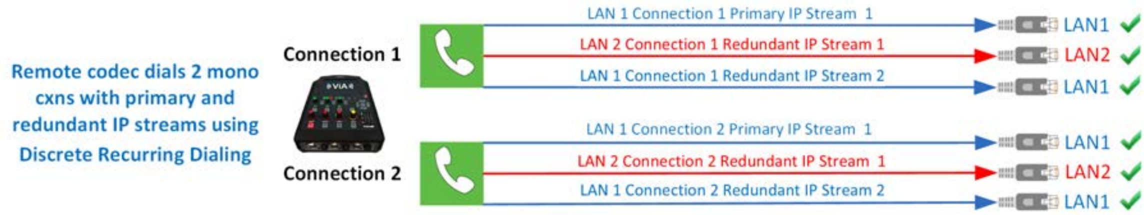


i. **First Available**: The first available interface that can successfully connect will be used to dial. If you dial two connections the same interface will be used for both connections, e.g. LAN1, even if another interface is available.

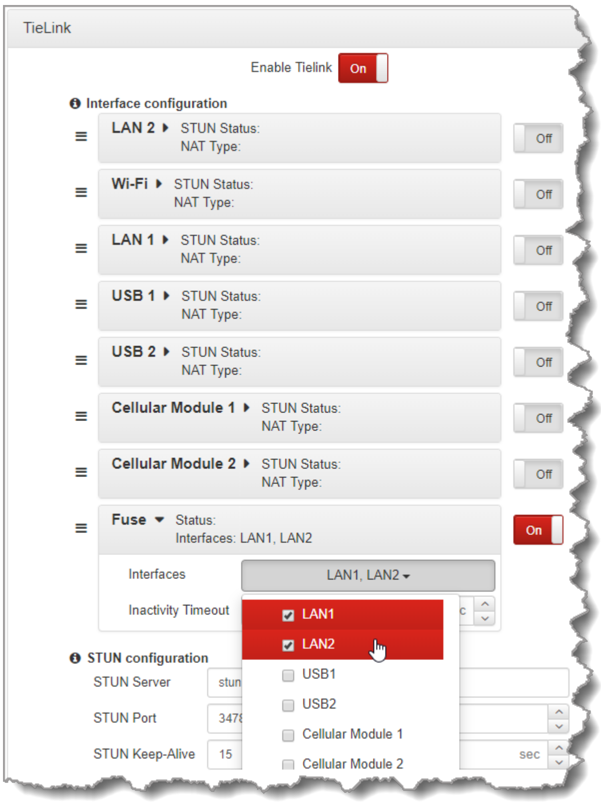
ii. **Discrete once only**: The first interface not being used by another connection will be used to dial. In the following example using two LAN interfaces, the codec connects a primary and redundant IP stream using different interfaces for each connection. A second redundant IP stream configured on each connection fails because both available interfaces are already being used for that connection.



iii. **Discrete recurring:** Uses the first interface that is not being used by another connection (within the same audio stream). After all interfaces have been used, each audio stream will scan each interface from the start again, in the order in which they are listed, and connect using the first available interface that will successfully dial and connect. The dialing order of the interfaces in the TieLink panel can be adjusted by clicking a dragging interfaces on the left side of the panel when it is in **Edit** mode.



iv. **Fuse-IP**: Connect using Fuse-IP data bonding. Multiple interfaces can be selected and bonded to aggregate data and create a higher bandwidth connection. This is configured in the **TieLink panel** as displayed in the following example. Please note that this is the simplest way to configure a Fuse-IP connection.



Important Note: Ensure an answering program is configured for Fuse-IP on the destination codec and locked.

Helpful Hints for Troubleshooting

If TieLink Does Not Connect:

1. Ensure TieLink and at least one relevant interface is enabled and 'stunned' on both codecs; i.e. the **STUN Status** for the interface is **Connected**.
2. Check STUN server settings.

If a TieLink Fuse-IP Connection Does Not Connect:

1. Ensure Fuse-IP is enabled in the **TieLink panel** in Toolbox, or in the codec's **TieLink Settings screen**, for both the dialing and answering codecs
2. Check at least one interface has been added to the Fuse-IP configuration in the **TieLink panel** in Toolbox, or in the codec's **TieLink Settings screen**, for both the dialing and answering codecs
3. Verify that the **STUN Status** for at least one interface in the Fuse-IP configuration is **Connected** on both the answering and dialing codecs.
4. Check that the answering codec is not already connected using TieLink Fuse-IP. Only one Fuse-IP connection is currently supported.

If SmartStream PLUS Connections do not connect:

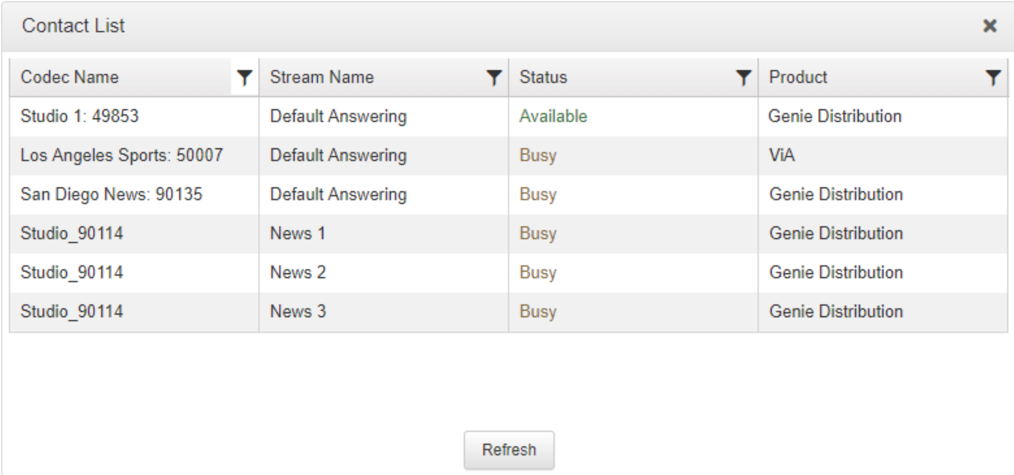
1. Based on the priority order of interfaces in the TieLink panel, a SmartStream PLUS connection may automatically use TieLink Fuse-IP as one of the connections if Fuse-IP is enabled at the dialing or answering codec. If TieLink Fuse-IP is chosen, then that connection will only succeed if the Fuse-IP stream is available at both ends. Disable Fuse-IP as an interface in TieLink to avoid Fuse-IP being selected in this scenario.
2. When "Discrete once only" or "Discrete recurring" is selected, with TieLink Fuse-IP enabled, TieLink Fuse-IP may be selected to transmit one of the streams. In this situation, if the answering codec is already connected to another codec using TieLink Fuse-IP, then the connection will fail for this stream because only one Fuse-IP connection is supported using TieLink.
3. When "First Available" is selected, TieLink Fuse-IP may be used if it is the first interface "ready" in the **TieLink panel** interface list. In this situation, the connection will fail if the answering codec is already connected to another codec using TieLink Fuse-IP.

Configuration Restrictions for TieLink

TieLink only supports one SmartStream PLUS redundant streaming connection on each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.

Updating TieLink Contacts

A TieLink administrator updates codecs included in, and excluded from, Contact Lists. Codec members of a Contact List shared with a codec are displayed from the **Menu Bar** by selecting **Connect > Contact List** to open the **Contact List** panel.



Contact List

Codec Name	Stream Name	Status	Product
Studio 1: 49853	Default Answering	Available	Genie Distribution
Los Angeles Sports: 50007	Default Answering	Busy	ViA
San Diego News: 90135	Default Answering	Busy	Genie Distribution
Studio_90114	News 1	Busy	Genie Distribution
Studio_90114	News 2	Busy	Genie Distribution
Studio_90114	News 3	Busy	Genie Distribution

Refresh

If a contact list is adjusted in TieLink it is necessary to click the **Refresh** button in the **Contact List panel** to display the latest contact info. This updates the **Contact List panel**, selectable contacts in Toolbox, and the codec **TOUCH SCREEN** when dialing and creating programs..

Stream Status in TieLink

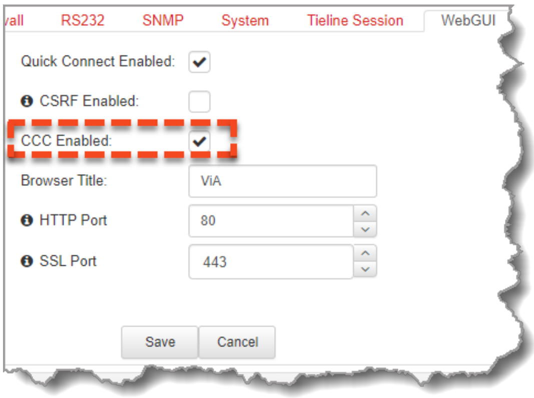
The **Contact List panel** displays the availability of a TieLink audio stream. The status may display:

1. **Available**: The TieLink Traversal Server Network has provided reachable streaming interfaces and indicates the audio stream is not busy.
2. **Busy**: The codec is connected to the TieLink Traversal Server network and TieLink indicates the stream is busy.
3. **Offline**: The audio stream is unavailable within the TieLink Traversal Server Network. Possible reasons an audio stream may be offline include:
   a. The codec is offline.
   b. An audio stream is not advertised on TieLink.
   c. TieLink and/or TieLink interfaces are disabled in the codec.
   d. Ports are not open.
   e. The **Contact List panel** in the Toolbox Web-GUI may need 'refreshing'.
   f. DNS - Not resolving the STUN server set in the **TieLink panel**. Note: If a codec has a statically configured IP address, ensure that DNS Server settings are also configured.
   g. DNS - Not resolving https://tieserver.com
   h. The codec firmware version is not compatible.

33.6 Enabling the Cloud Codec Controller

To allow the codec to be configured and managed by Tieline's Cloud Codec Controller over the public internet it needs to be enabled for CCC management:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **WebGUI**, then click **Edit** and select the **CCC Enabled** check-box to enable this feature.



3. Click **Save** to store the new configuration.

Important Notes:
- Ensure **CSRF** is disabled in the codec or it will not be able to connect to the CCC. This setting is **[OFF]** by default and is also available in the codec menu via **Settings > WebGUI**, and in the **Options** panel in Toolbox.
- Locally Defined Codecs over a private network do not need to be enabled for CCC operation. Only codecs that require internet access need to be enabled.
- The CCC needs to continually send and receive data between codecs to update information displayed. If the CCC is left open on a computer and is not used for more than 4 hours, the Codec Viewer is placed in 'sleep' mode to save on data use.

33.7 Configure Fuse-IP Bonding

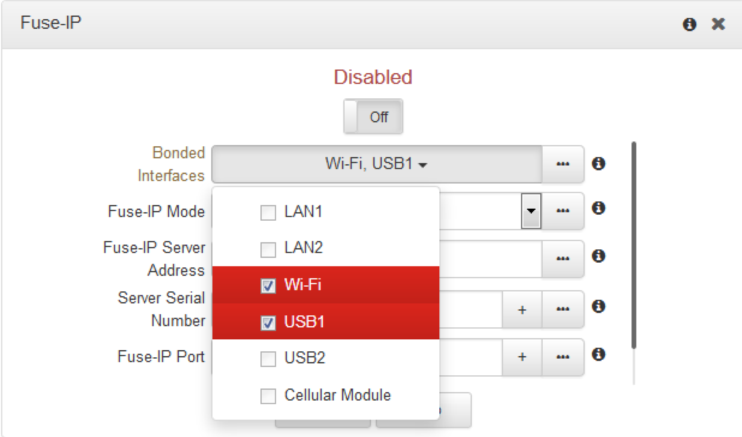
Tieline's proprietary Fuse-IP data aggregation technology uses a point-to-point tunnel between two codecs to bond multiple IP interfaces (peers). Fuse-IP automatically distributes data over any two bonded interfaces. IP interfaces from which you can choose to bond include:

- 2 USB modems.
- Built-in Wi-Fi.
- An internal cellular module.
- Dual Ethernet LAN ports.

See Connecting with Fuse-IP for more details on configuration using the codec **TOUCH SCREEN**.

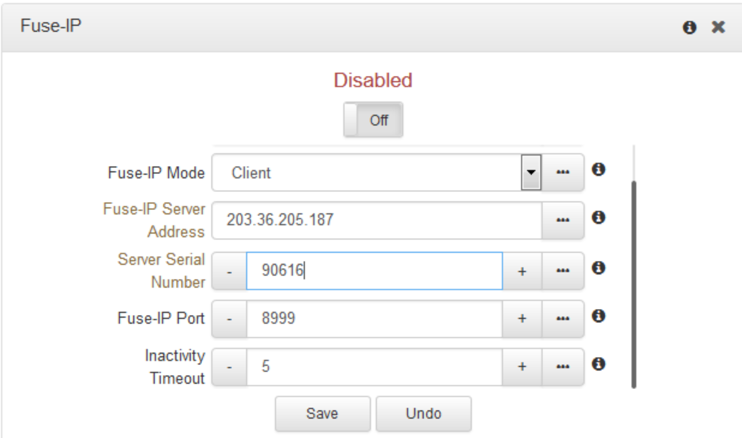
Configuring a Fuse-IP Remote Client

1. Open the HTML5 Toolbox Web-GUI and click **Settings** and then **Fuse-IP**.
2. Click the **Bonded Interfaces** drop-down menu to select the interfaces to be bonded.

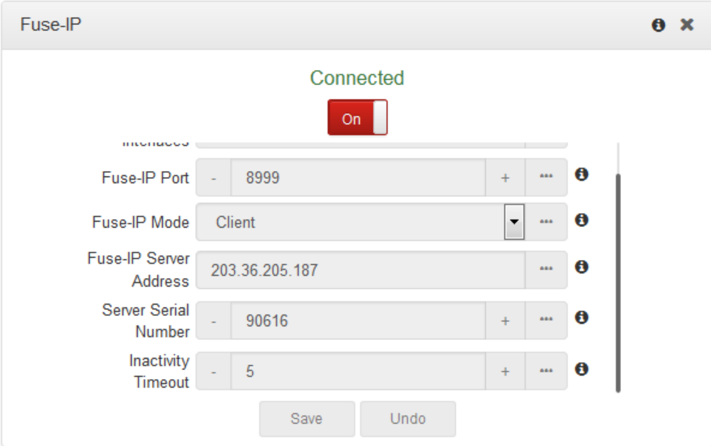


3. Click the **Fuse-IP Mode** drop-down menu to select **Client** as the Fuse-IP mode if you are dialing the studio codec. Note: the studio codec should be configured in server mode.

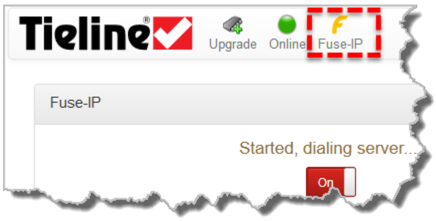
4. Enter the **Fuse-IP Server Address**, which is the public static IP address of the server codec at the studio. Then enter the server codec's serial number in the **Server Serial Number** text box. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. you have multiple codecs behind a firewall using Fuse-IP, therefore you need to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same. Configure the **Inactivity Timeout** if you want to turn the Fuse-IP tunnel off after a predetermined time period to save data, then click **Save**. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter **0** to disable the timeout.



5. Click the **On/Off** button for Fuse-IP to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds.



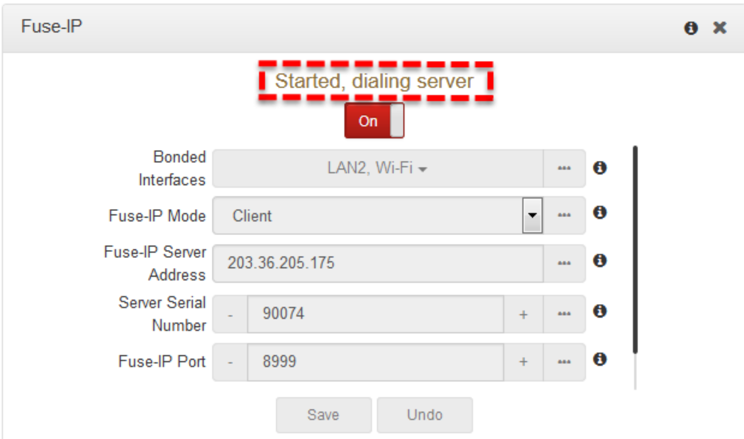
6. The status indicator is orange when Fuse-IP is enabled but no tunnel is created. It turns green when a tunnel is active.



Enabled: no tunnel created

Enabled & tunnel created

7. Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.



Important Notes:

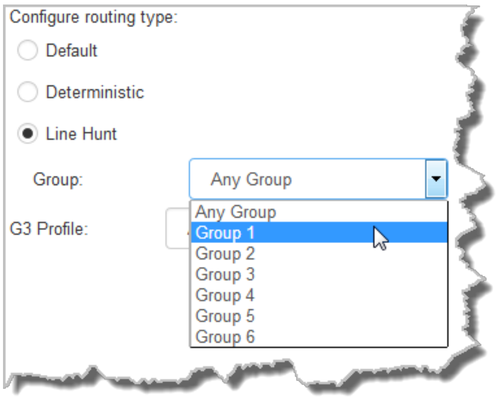
- Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary **Via**.
- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, the codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- The codec remembers the Fuse-IP enabled/disabled state on power up.
- For additional stability it is recommended that a fixed jitter buffer is configured when streaming using Fuse-IP. The actual jitter buffer depth should account for the difference in delay between the interfaces and the maximum jitter experienced. To determine the jitter over each link you can connect and stream audio over each interface separately and look at the jitter reading displayed on the **Connection Statistics** screen.

- Use a dotted quad IPv4 address when configuring the Fuse-IP Server Address.

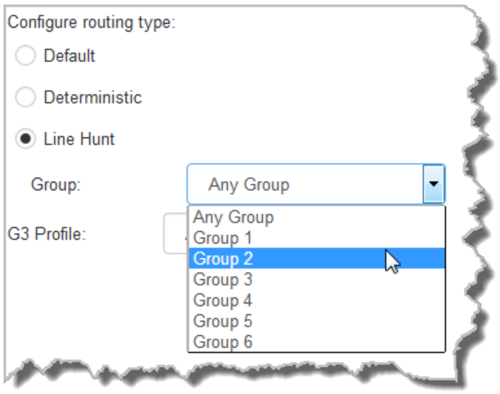
33.8 Line Hunt Call Answering

The codec supports line hunt call answering, whereby you can create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations.

As an example, when creating a program which supports connecting six mono audio streams, select **Line Hunt** as the routing type. Then select **Group 1** for the first three audio streams, to route outgoing and incoming calls via inputs and outputs 1 to 3 of a codec at the studio. These physical inputs and outputs on the codec can be routed to a particular studio or station.

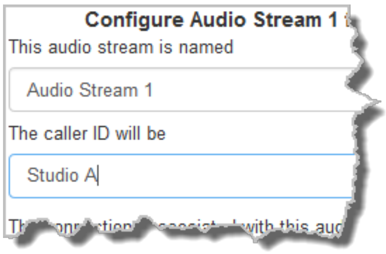


For the next three audio streams, select **Group 2** to route outgoing and incoming calls via inputs and outputs 4 to 6. These physical inputs and outputs on the codec can be routed to a different studio or station.

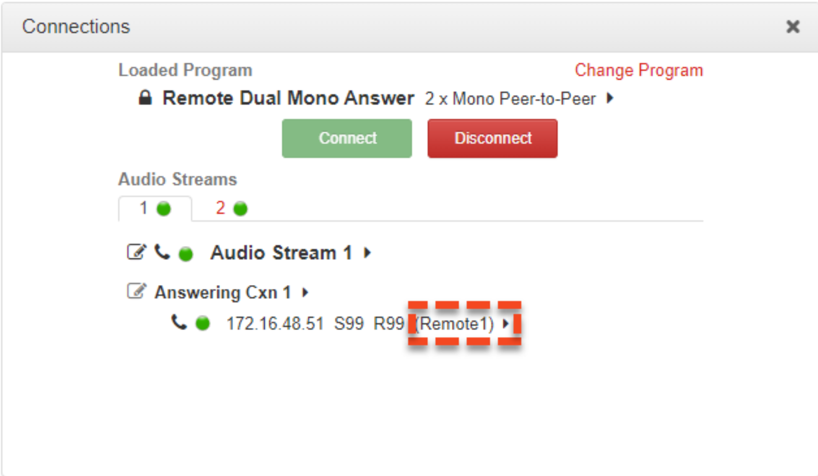


Incoming Caller ID

Tieline codecs also support incoming caller IDs, so you can uniquely identify codecs or Report-IT users when they call in. This is particularly useful for identifying inbound callers when using line hunt answering mode.



Any Tieline G5 codec dialing can display a designated **Caller ID** in the **Connection panel**. In the following example, **Remote1** has called into the codec using a specific caller ID, which is displayed next to the **Send** and **Return** link quality connection.



33.9 Configuring ISDN

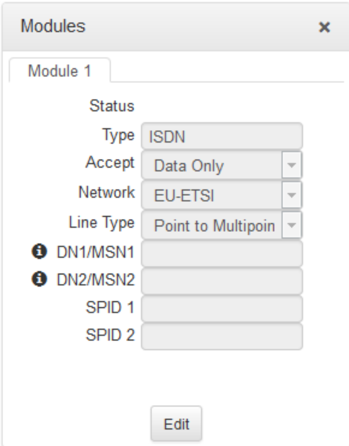
A ViA ISDN module can be inserted into the codec's rear panel module slot and a dial and/or answer program can be configured using the HTML5 Toolbox Web-GUI. See About ISDN Modules for additional information on ISDN. It may also be necessary to:

1. Configure ISDN module settings.
2. Configure ISDN Answering settings.

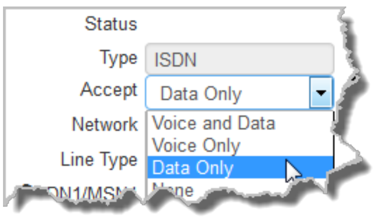
33.9.1 Configuring ISDN Modules

ISDN settings in the **Modules panel** determine how each codec module operates at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each ISDN module should connect. ISDN module settings may need to be adjusted depending on your country and network requirements.

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Modules** to view and configure ISDN site settings.

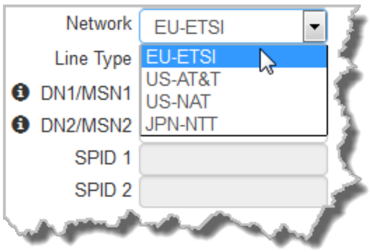


2. Click the **Edit** button to configure settings.
3. Click the drop-down arrow for **Accept** to select whether to allow or disallow circuit switched voice and data calls. The default setting allows **Data only**.

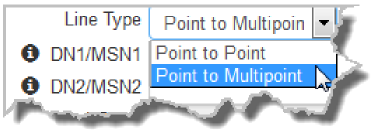


Important Note: G.711 is the algorithm used when **Voice Only** is selected.

4. Click the drop-down **Network** arrow and select the **Network Type** corresponding to the region in which you are using the codec (see ISDN Module Settings for more details).



5. Click the drop-down **Line Type** arrow and select your preferred option. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is an AT&T 5ESS custom point-to-point.



6. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required.
7. Click **Save** when configuration is complete.

Important Notes:

Directory Numbers and Multiple Subscriber Numbers
Directory Numbers (DN) in North America and Multiple Subscriber Numbers (MSN) in the rest of the world are simply phone numbers associated with an ISDN B channel, like lines listed in a typical phone directory. Your Telco will normally supply 2 DN/MSN numbers for each pair of B channels. However, these numbers may or may not be associated with a specific B channel.

Often broadcasters prefer to predict which B channel will answer an incoming call to ensure audio routing is consistent. However, if a DN or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number. Programming DN/MSN numbers for each B channel allows the codec to ignore calls without matching DN/MSN numbers. This is the best way to answer calls from codecs in a predictable manner.

SPID Numbers in North America
ISDN relies on an initialization procedure for associating Service Profiles with specific terminating equipment (e.g. your audio codec) rather than lines. In the US Telcos assign a Service Profile ID (SPID) number which assists in identifying different ISDN services across the network. Your Telco must provide a SPID for each B channel you order when connecting over US-Nat or US-AT&T networks in the US. A SPID is not required when using the AT&T PTP protocol.

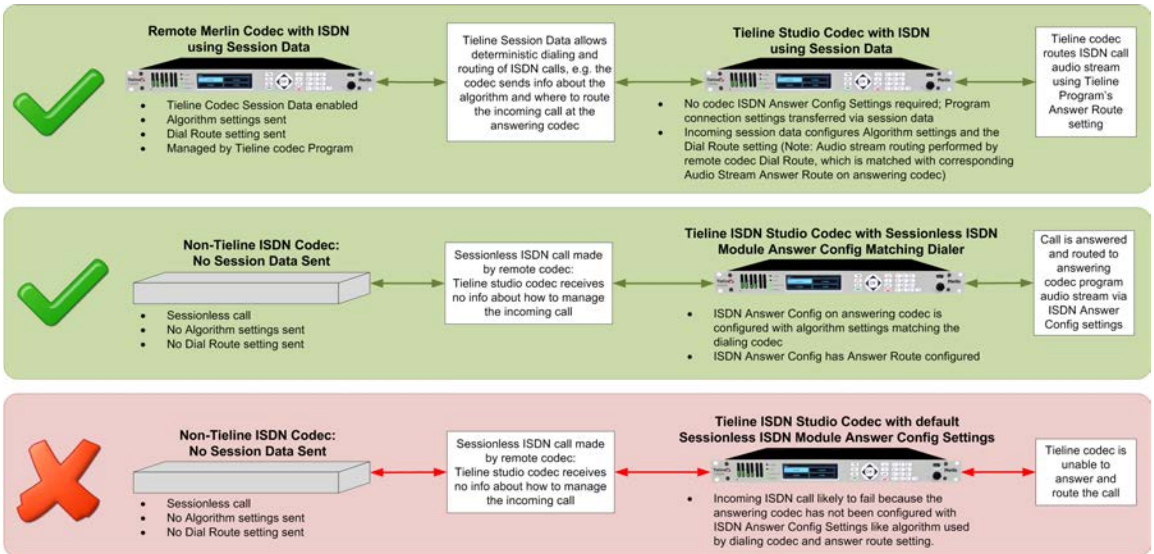
Typically, each ISDN BRI service in the US will have two SPIDs and these must be entered correctly. When you enter a SPID into your codec and connect it to an ISDN line, an initialization and identification process takes place, whereby the terminating equipment (your codec) sends the SPID to the switch. The switch then associates the SPID with a specific Service Profile and directory number.

Note: SPID numbers normally include the phone number and additional prefix or suffix digits up to 20 digits long.

33.9.2 Configuring ISDN Answering

ISDN Answer Configs are used to determine how codec ISDN modules will behave when answering ISDN calls.

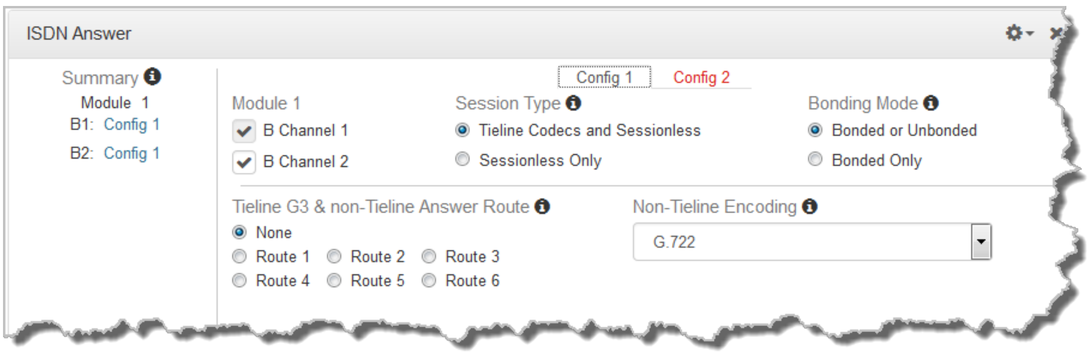
The following image explains the difference between answering calls from Tieline codecs sending session data, and non-Tieline codecs making sessionless ISDN calls. Codecs sending Tieline Session Data contain all the information required to connect, e.g. algorithm and audio stream routing settings. When answering sessionless calls it is necessary to configure the answering codec with an **ISDN Answer Config**, which tells the answering codec how a sessionless call will try and connect.



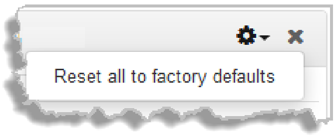
After installing a ViA ISDN module it is possible to save up to two different **ISDN Answer Configs**, which allow each ISDN B channel to be individually configured for unique answering behaviors. ISDN answering can be configured to suit:

- Hardware available in the codec, i.e. the number of B channels available.
- Expected dialing behaviors, e.g. if B channels should bond or not, and whether audio streams need to use **Dial** and **Answer Route** tags.
- The type of call being received by the codec, e.g. Tieline (with Tieline Session Data) versus non-Tieline sessionless calls.
- The algorithm expected when receiving sessionless calls.

The two available **Configs** allow you to select which B channel or channels are used to answer a call or calls from incoming ISDN codecs.



To reset ISDN answering to default settings click the **Options symbol** in the top right-hand corner of the panel and select **Reset all to factory defaults**.

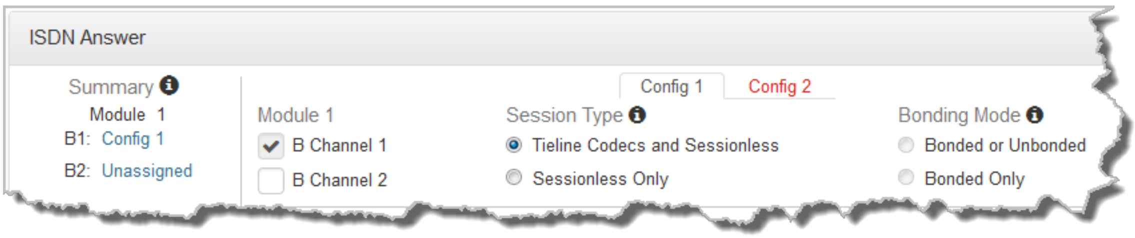


Important Note: B channels can only be selected once and are greyed out once they have been selected in one of the two ISDN **Configs**.

Single B Channel Config

To use a single 64kbps B channel for a connection (e.g. a 1 x Mono Peer-to-Peer audio stream):

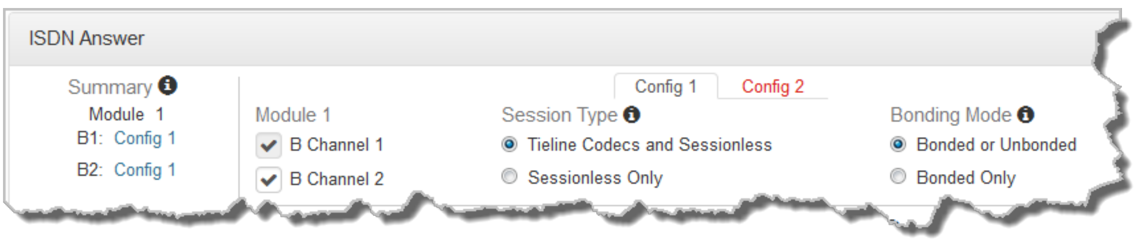
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **ISDN Answer** to view and configure ISDN answering site settings.
1. Click to select a **Config**.
2. Select a B channel from those available and then click **Save**. The connection is not bonded if only one B channel is selected.



Multiple B Channel Bonding Config

A point-to-point audio stream can also bond multiple B channels to create higher bandwidth connections.

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **ISDN Answer** to view and configure ISDN answering site settings.
2. Select multiple B channels in the **Config**. In the following example, two B channels from **Module 1** have been selected within **Config 1**.



2. Configure the bonding setting that best suits the audio stream with which this **Config** is associated. **Bonded or Unbonded** is the best setting in most situations.

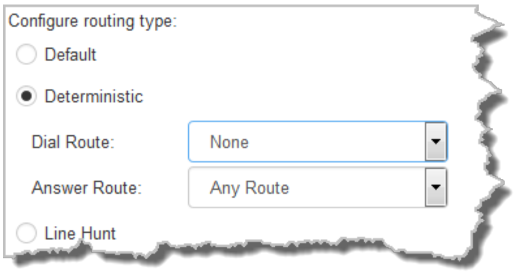
Bonding Setting	Behavior
Unbonded	When an unbonded single B Channel is selected
Bonded or Unbonded (May Bond)	Calls using the same algorithm from the same Tieline codec, or sessionless calls, will attempt to bond when received. Calls using incompatible algorithms will not be bonded

Bonded Only	Will only bond compatible algorithms. This mode will reject incompatible calls which cannot be bonded, e.g. G.711 and G.722

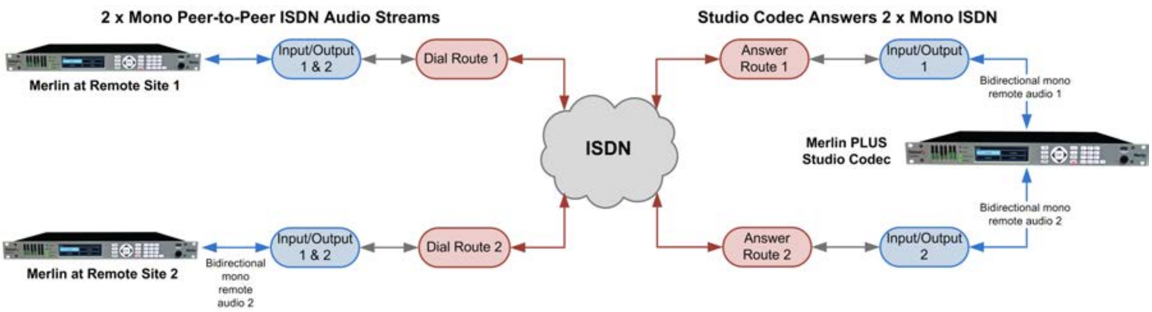
3. Click **Save** to apply changes to the **Config**.

Dial and Answer Route Settings in Programs

Dial Route and **Answer Route** tags allow you to associate a B channel (or channels) in a **Config** with a particular incoming audio stream from either Tieline G3 or non-Tieline codecs. This is not necessary in simple point-to-point ISDN audio stream configurations, however it is very useful in multiple audio stream codecs using multiple B channels. When dialing Tieline to Tieline over ISDN using the Merlin, Genie and ViA families family of codecs, you can configure a **Dial Route** in the dialing codec's program and a corresponding **Answer Route** in the answering codec's program. This will ensure a particular audio stream is routed between two codecs consistently. This feature is not available in Tieline G3 codecs, so an **Answer Route** should be used for deterministic routing when answering calls from these codecs.



In principle, the concept of 'routes' operates similarly to how audio ports are used to route multiple audio streams over IP. Selecting different IP audio port numbers allows users to define which incoming IP audio stream is routed to a specific answering audio stream configuration on the codec. This ensures inbound calls from multiple codecs can be consistently routed to the same answering codec audio streams, and therefore the same inputs and outputs. Following is an example of how to consistently route incoming ISDN audio streams using dial and answer routes.



Answer Routes for Non-Tieline (Sessionless) or Tieline G3 ISDN Calls

In some situations you may receive a call from a non-Tieline codec which doesn't support session data and **Dial Route** tags. In this situation you can still specify the audio stream **Route** on the answering codec using **Config 1-2** in **ISDN Answer**. You can also select the default algorithm.

For example, if a call from a non-Tieline codec is received via **B Channel 1** on **Module 1** (i.e. no **Dial Route** has been specified in the dialing codec):

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **ISDN Answer** to view and configure ISDN answering site settings.
2. Click to select a **Config**.

3. Select a **Route** for this B channel in one of the two **Configs** within **ISDN Answer**, e.g. **Route1**, then select the default **Non-Tieline encoding** algorithm to use when connecting (default setting is **G.722**).

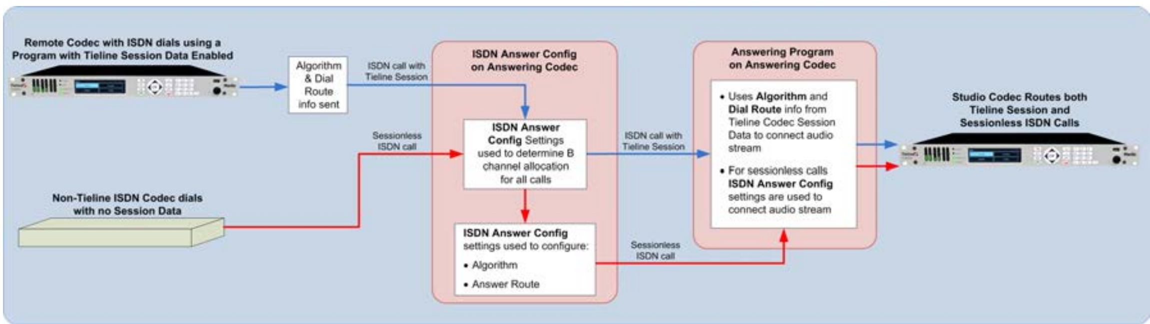
3. Click **Save** when configuration is complete to store the new **Config** settings.
4. This will associate the incoming call with a corresponding **Answer Route** configured in the answering codec program, e.g. **Answer Route 1** in the following image.

More detailed information about how to configure the codec to answer and route multiple sessionless ISDN calls is available in Using ISDN Answer Routes for Sessionless ISDN Calls. This uses examples to explain how to set up consistent deterministic routing of multiple incoming sessionless calls.

Answering both Tieline Session and Sessionless ISDN Calls

Leave the **Sessionless Only** check-box in the **ISDN Answering Config** unchecked if the codec is expected to receive ISDN calls from Tieline codecs, or both Tieline and non-Tieline codecs (i.e. you are not sure which type of codec may call). In this mode, when the codec answers a call it initially expects to receive Tieline session data from the dialing codec and configure its own algorithm settings according to that. If it fails to receive Tieline session data within 5 seconds (i.e. a non-Tieline codec is calling, or a Tieline codec with session data disabled), it will use the settings in the **ISDN Answering Config** instead.

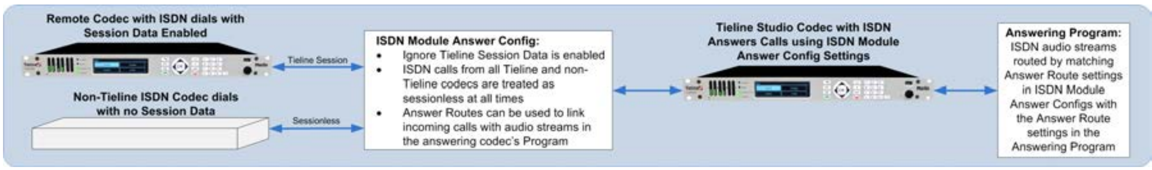
The following image displays how the answering codec will behave in this mode when receiving calls from both Tieline and non-Tieline codecs.



Allow Answering of Sessionless ISDN Calls Only

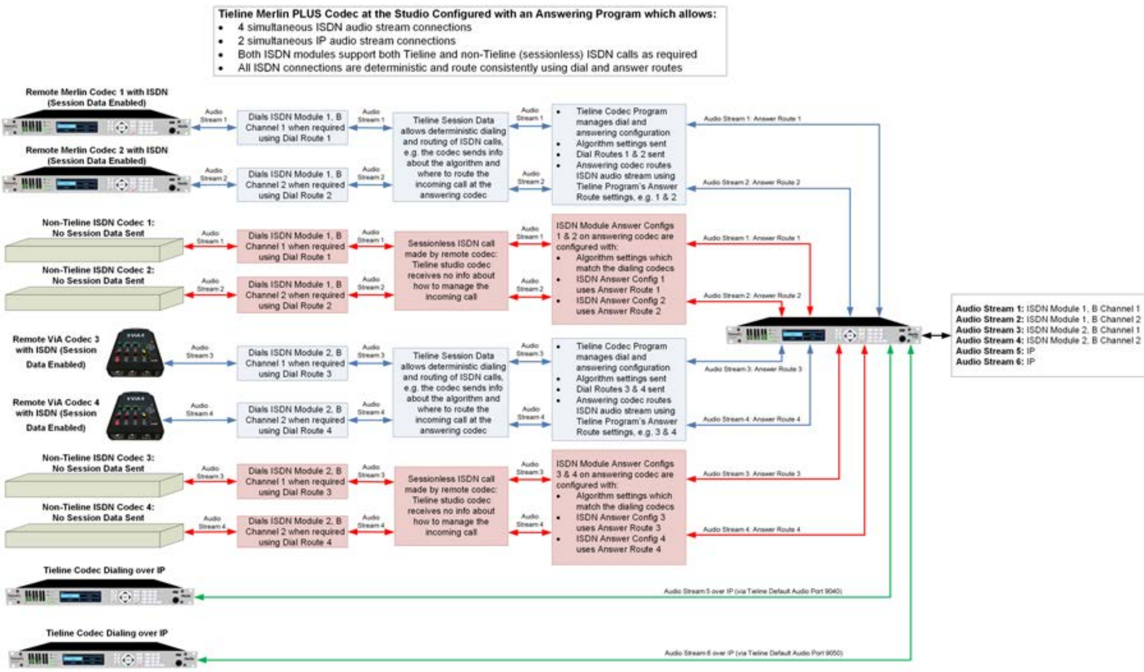
Select **Sessionless Only** when answering ISDN calls from non-Tieline codecs only. When **Sessionless Only** is selected, the codec will not wait to receive the Tieline session data. This reduces the time taken to answer an inbound sessionless call.

The following image displays how the answering codec will respond with **Sessionless Only** selected, i.e. calls from both Tieline and non-Tieline codecs are always regarded as sessionless.



Answering Multiple ISDN Calls from Tieline and non-Tieline Codecs

Tieline codecs capable of answering multiple incoming audio streams can be configured to answer both Tieline session data and sessionless ISDN calls at different times. They can also support connections using other transports such as IP or POTS. The following example shows how a Tieline codec can be configured to answer up to 4 separate mono ISDN calls at different times from both Tieline and non-Tieline codecs, as well as two mono IP audio streams.



Default Answering Settings

When a B channel is not associated with a **Config** it inherits the following default settings:

- Tieline Session
- Unbonded
- G.722 algorithm
- Audio route: None

33.10 Configuring POTS

A ViA POTS module can be inserted into the codec's rear panel module slot and a dial and/or answer program can be configured using the HTML5 Toolbox Web-GUI. See About POTS Modules for additional information on POTS. It may also be necessary to:

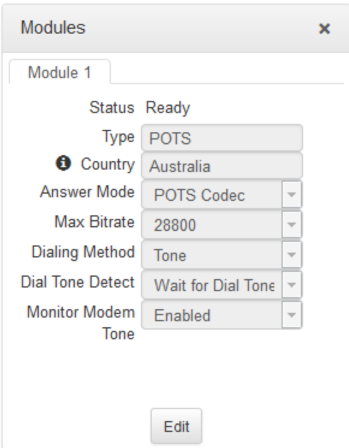
1. Configure POTS module settings.
2. Configure POTS Answering settings

33.10.1 Configuring POTS Modules

POTS settings in the **Modules panel** menu determine how your codec will connect at a particular site. You can copy similar programs between codecs installed at different locations and also configure site-specific settings for how each module should connect. The default **Config** settings for POTS modules are designed to suit Tieline codecs. These settings will need to be adjusted to connect to non-Tieline POTS codecs or connect in **Analog Phone** mode.

Configuring POTS G5 Modules

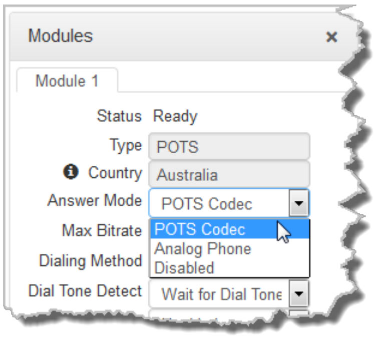
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Modules** to view and configure POTS site settings.



Important Notes: The POTS module **Status** is displayed in the **Modules panel**. **No Phone Line** is displayed when a cable is detached; **Ready** is displayed when a cable is attached and the line voltage is good.

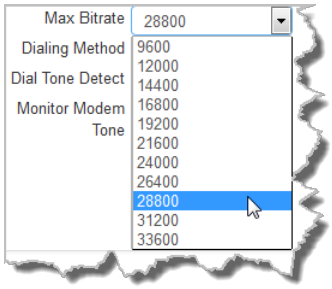
2. Click the **Edit** button to configure settings.
3. **Country** displays the current country setting in the codec. The country setting makes adjustments relating to POTS line ring tones and POTS line impedance for different countries. To adjust this setting see Configure Country Setting.
4. Click the drop down arrow to adjust the **Answer Mode** and select how the module in the codec will be able to answer incoming POTS calls. Options include:

   - **POTS Codec:** allows the POTS G5 module to receive incoming audio data over a POTS line.
   - **Analog Phone:** configures the POTS G5 module to receive a standard analog phone call.
   - **Disabled:** disables the POTS G5 module from receiving a **POTS Codec** or **Analog Phone** call.



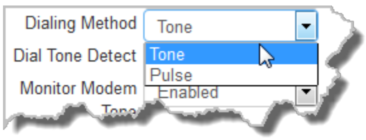
Calls are answered based on the **POTS Answer settings** in **Config 1** & **2**. Adjustments to these **Config** settings are not normally necessary when connecting between Tieline codecs. They are usually adjusted when connecting to non-Tieline codecs over POTS (see Configuring POTS Answering for more info).

5. Click the **Max Bitrate** drop-down arrow to adjust the maximum bit rate (dialing and answering). The default setting is **28800** (28.8kbps) and this only affects **POTS Codec** calls. The range of the setting is 9.6kbps to 33.6kbps. Even if the line is capable of establishing a connection at a higher bit rate, the **Max Bitrate** setting is the highest bit rate that will be attempted. Reducing this value can improve connection reliability on poor quality lines. If two codecs are not configured with the same setting, they will attempt to connect at the lowest of the two **Max Bit rate** settings.



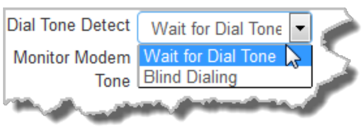
Important Note: G5 POTS modems initially attempt to establish a link at the lowest **Max Bitrate** setting configured in the two modules being connected. If the POTS line doesn't support this bit rate, the modems will attempt to connect at the highest possible bit rate to suit the prevailing line quality at each end of the link.

6. Click the drop-down arrow for **Dialing Method** to select **Tone** (DTMF) or **Pulse** dialing over POTS Codec connections. Tone dialing is used always when the **Answer Mode** is **Analog Phone**.

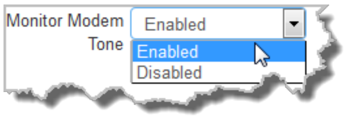


7. Click the drop-down arrow for **Dial Tone Detect** to select either:

   - **Wait for Dial Tone**: The module will only be allowed to dial when a dial tone is present on the line.
   - **Blind Dialing**: Allows the module to dial when no dial tone is present.



8. Click the drop-down arrow for **Monitor Modem Tone** to select either **Enabled** or **Disabled**. Note: When enabled, the module will allow audio monitoring of modem tones via the phone input while connecting in **POTS Codec** mode.



Important Notes:
- Modem tone monitoring will work even if **Phone Input Enable** is **Off** via **Settings > Audio > Phone Input > Phone Input Enable [Off]**.
- Modem tone monitoring is only enabled during the initial connection training and negotiation period in **POTS Codec** mode.
- The monitoring volume can be adjusted using the codec front panel via **Settings > Audio > Phone Input > Level**, or by opening the **Inputs panel** in the Web-GUI and adjusting the **Phone** input volume slider.

9. Click **Save** when configuration is complete.

33.10.2 Configuring POTS Answering

POTS answering can be configured to suit:

- The type of call being made, e.g. Tieline (with Tieline Session Data) versus non-Tieline (Sessionless).
- Expected dialing behaviors and encoding, e.g. whether audio streams use **Route** tags and which algorithm is used.

The POTS answering "Config" determines the settings used when answering a call from a non-Tieline codec .

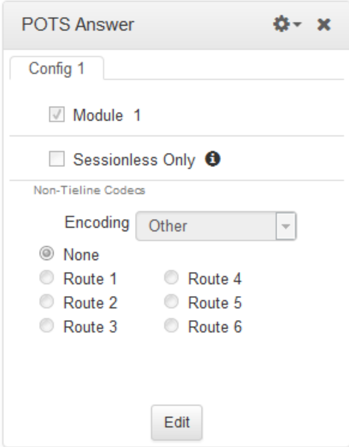
Important Notes:
- **POTS Answer Config** settings are applied to **POTS Codec** connections and not **Analog Phone** connections.
- When receiving a call from a Tieline codec with session data enabled (i.e. not **Sessionless**), the algorithm setting from the dialing codec overrides the **Encoding** setting in the **POTS Answer Config** menu.

POTS Config Settings

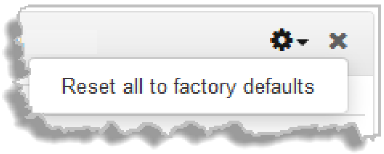
The default **POTS Answer** module **Config** settings, which can be viewed in the **POTS Answer panel** are:

- **Tieline Codecs** Session Data,
- The **Other** algorithm.

This configuration will accept the settings from an incoming Tieline codec when it dials with session data enabled. It will also allow the codec to answer a call from a Comrex POTS codec supporting the **Other** algorithm setting.

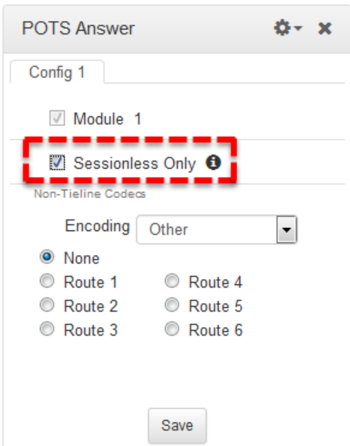


To reset POTS answering to default settings click the **Options symbol** in the top right-hand corner of the panel and select **Reset all to factory defaults**.



Answering Calls from Non-Tieline POTS Codecs

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **POTS Answer** to view and configure POTS answering site settings.
2. Click the **Edit** button to configure settings.
3. Select the **Sessionless Only** check-box when only non-Tieline codecs are dialing a Tieline codec over POTS. This allows you to choose the default encoding setting and **Route** the incoming call to a nominated audio stream via a corresponding **Answer Route** in the answering codec program if required.

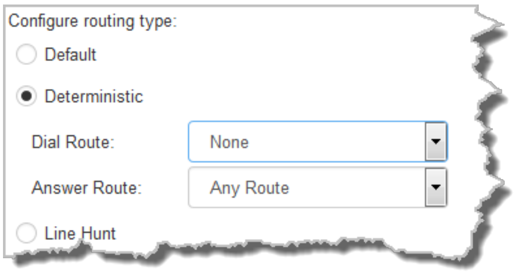


4. Click **Save** to apply changes to the **Config**.

Important Note: Select **Other** in the **Encoding** drop-down menu when connecting to Comrex® Vector, Matrix® and BlueBox® codecs. On the Comrex codec select its "Music" algorithm. Please note that 9.6kbps connections are not supported by the Comrex codecs.

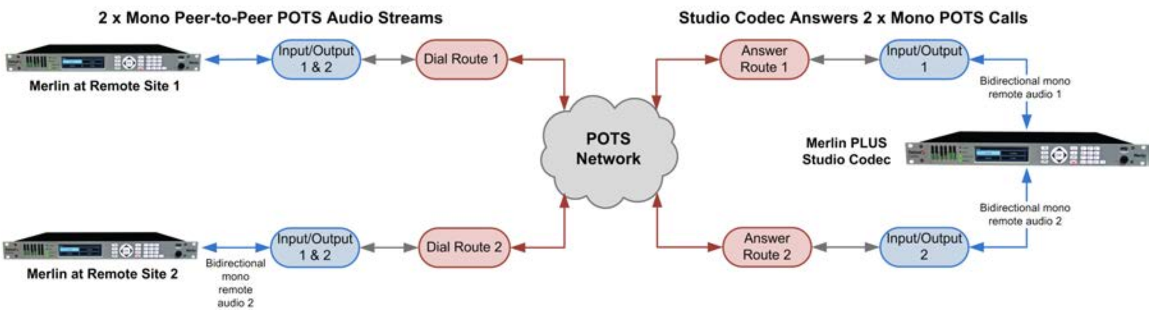
Dial and Answer Route Settings in Programs

Dial Route and **Answer Route** tags allow you to associate a POTS **Config** with a particular incoming audio stream from either Tieline or non-Tieline codecs.



In principle, this operates similarly to how audio ports are used to route multiple audio streams over IP. Selecting different IP audio port numbers allows users to define which incoming IP audio stream is routed to a specific answering audio stream configuration on the codec. This ensures inbound calls from multiple codecs can be consistently routed to the same answering audio streams, and therefore the same inputs and outputs.

This is not necessary in simple point-to-point POTS audio stream configurations, however it is very useful in multiple audio stream codecs which support POTS connections. When dialing Tieline to Tieline over POTS using ViA, or Merlin and Genie codecs, you can configure a **Dial Route** in the dialing codec's program and a corresponding **Answer Route** in the answering codec's program. This will ensure a particular audio stream is routed between two codecs consistently.

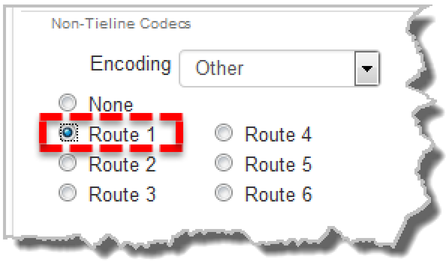


Answer Routes for Non-Tieline POTS Codecs

In some situations you may receive a call from a non-Tieline POTS codec which doesn't support **Dial Route** tags. In this situation you can still specify the audio stream **Route** on the answering codec using **Config 1 or 2** in **POTS Answer**. You can also select the default algorithm.

For example, if a call from a non-Tieline codec is received via POTS **Module 1** (i.e. no **Dial Route** has been specified in the dialing codec):

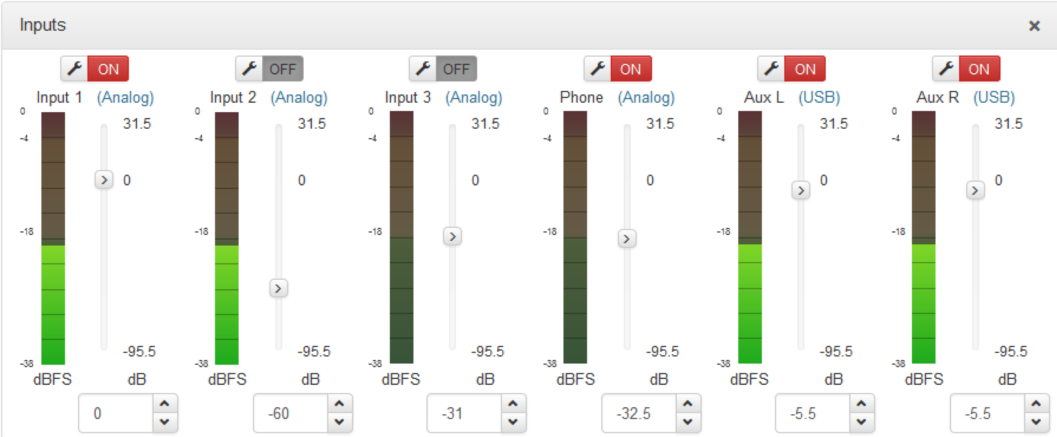
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **POTS Answer** to view and configure POTS answering site settings.
2. Click the **Edit** button to configure settings.
3. Select an answering **Route** for this POTS module in one of the two **Configs** available in the **POTS Answer panel**, e.g. **Route1**, then select the default **Encoding** algorithm **Other** (Note: **Other** is used for connecting to Comrex POTS codecs).



4. Click **Save Settings** to store the new **Config** settings.
5. This will associate the incoming call with a corresponding **Answer Route** configured in the answering codec program, e.g. **Answer Route 1** as displayed in the following example.

33.11 Configuring Input/Output Settings

Open the HTML5 Toolbox Web-GUI and click **Inputs** to display the **Inputs panel**.



Adjusting Audio Levels

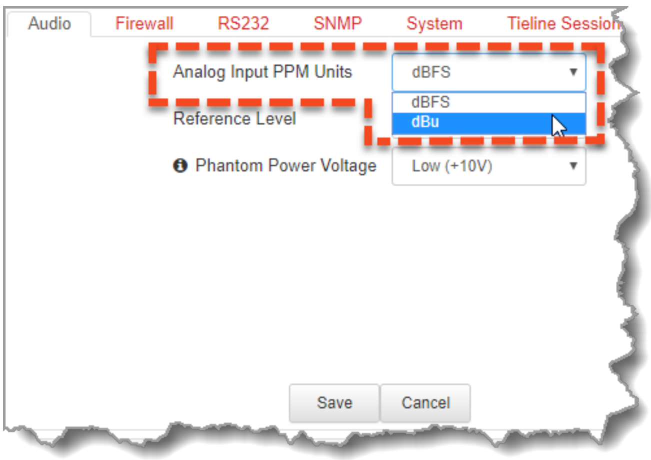
To adjust input audio levels, click on the input slider and drag it to the desired input gain level. Alternatively, click the arrows below a PPM meter to incrementally increase or decrease the input level in 0.5dB steps. Input levels on the **Input panel** should be set to ensure audio peaks average at the first yellow indications on the PPM meters, which represents nominal 0 VU at -18dBFS. Audio levels should also be verified using the meters in the **PPMs panel**.

Important Notes:
- There is a maximum of 6dB of additional gain available when adjusting a digital input.
- Turn off all unused inputs to avoid additional noise in program audio.

Changing the Input PPM Meter Units from dBFS to dBU

It is also possible to switch the analog input PPM meter unit of measurement from dBFS (default) to dBU:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **Audio** and then click **Edit**.
3. Click the **Analog Inputs PPM Units** drop-down menu and select **dBU**.



3. Click **Save** to change the setting.

For more information on reference scales see General Audio Settings. For more information on input levels and input settings see ViA Input Levels and Input Settings. For more info on PPM meters see PPM Meters and Analog Audio Outputs.

Configuring Input Settings

Select Analog and Digital Input Type

VOLTAGE WARNING: DO NOT attach non-digital microphones or an AES3 source to input 1 when **AES42** input mode is selected, or equipment may be damaged by high voltages supplied in this mode. See Configuring AES3 and AES42 Input Audio for more info.

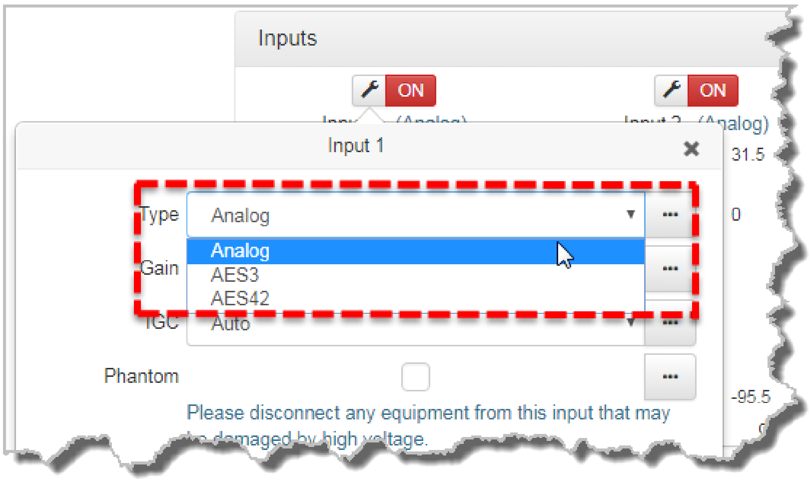
The following input type settings are available for each input on the codec:

		Input Type Options				
	Input	**Analog**	**AES3**	**AES42**	**S/PDIF**	**USB**
1	Input 1	✓	✓	✓		
2	Input 2	✓	✓			
3	Input 3	✓				
4	Phone	✓				
5	Auxiliary input 1	✓			✓	✓
6	Auxiliary input 2	✓			✓	✓

Important Note: When the auxiliary input (**AUX IN**) is **On** the default mixer configuration routes audio to the encoders and analog **OUTPUT 1** and **2**. If you are not using the auxiliary input ensure it is **Off** to avoid additional noise in program audio.

Codec inputs 1-3 and the stereo auxiliary input are configured for analog audio by default. To reconfigure the input type:

1. Click the **Input Settings** symbol.
2. Select **Type** and choose your preferred option from **Analog**, **AES3, AES42**, **S/PDIF** and **USB**. Note: refer to the previous table for supported input types on each input.

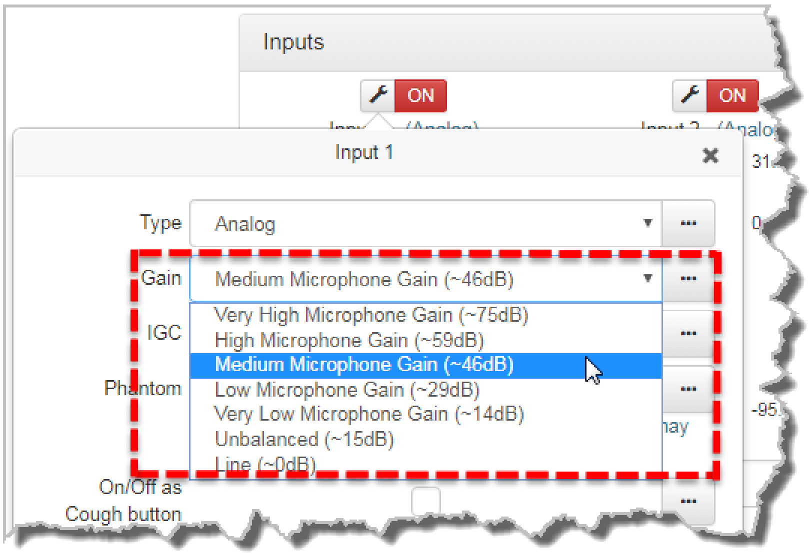


3. Click **Save** to confirm changes.

Select Input Gain

The default analog input gain setting for inputs 1-3 is **Medium Microphone Gain**. The stereo auxiliary input is a line level input when operating in analog mode. To adjust input gain settings:

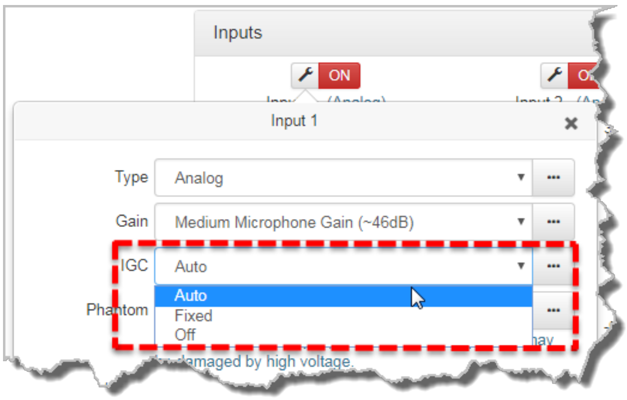
1. Click the **Input Settings** symbol on the input you want to adjust.
2. Click the **Gain** drop-down arrow to view menu items and click to select the preferred option.



3. Click **Save** to confirm the change.

Intelligent Gain Control (IGC)

1. Click the **Input Settings** symbol on the input you want to adjust.
2. Click the **IGC** (Intelligent Gain Control) drop-down arrow and select **Auto**, **Fixed** or **Off** as required.



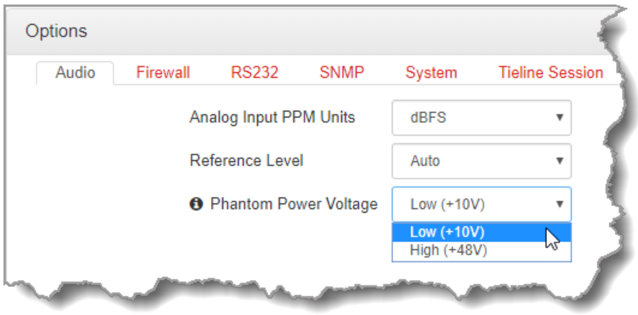
3. Click **Save** to confirm the change.

Phantom Power

Phantom power can be enabled or disabled when inputs 1-3 are in analog mic level input mode (default setting disabled). Phantom power of 10V or 48V is supplied to all inputs when enabled. The default setting is 10V and the currently configured voltage is displayed in brackets in the

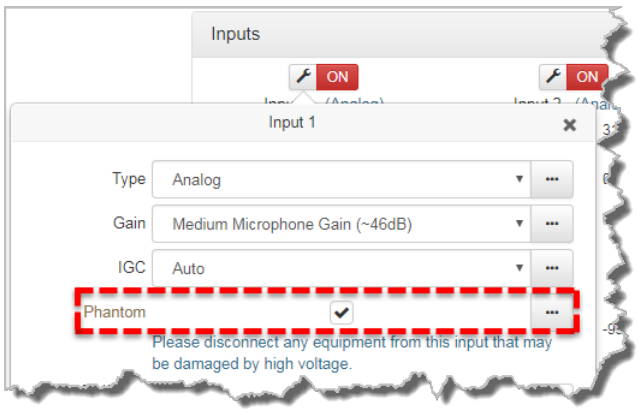
Phantom power menu via **Home** screen, then tap **Audio** > **Inputs** > **[tap to select an input] > Phantom Power**.

See General Audio Settings to change the voltage setting and default on-screen warnings, or open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**. Select **Audio**, then click **Edit** and click the **Phantom Power Voltage** drop-down menu to select 10V or 48V phantom power.



VOLTAGE WARNING: Check the specifications of all microphones attached to the codec to ensure they will not be damaged by either 10V or 48V phantom power supplied by the codec inputs.

1. Click the **Input Settings** symbol on the input you want to adjust.
2. Click to select the **Phantom** check-box to enable phantom power on the selected input.

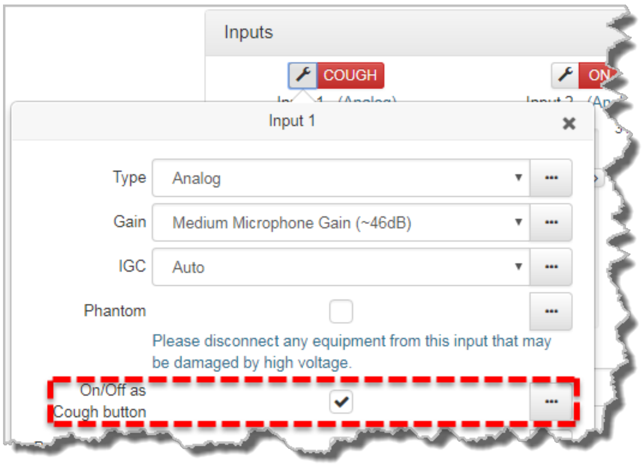


3. Click **Save** to confirm the change.

On/Off as Cough Button

This feature on **Inputs 1-3** allows the **ON/OFF** button on the codec to be turned into a cough button, whereby an input is turned off while the **ON/OFF** button is pressed. Each input can be configured independently.

1. Click the **Input Settings** symbol on the input you want to adjust.
2. Click to select the **On/Off as Cough button** check-box and enable cough button mode on the selected input.

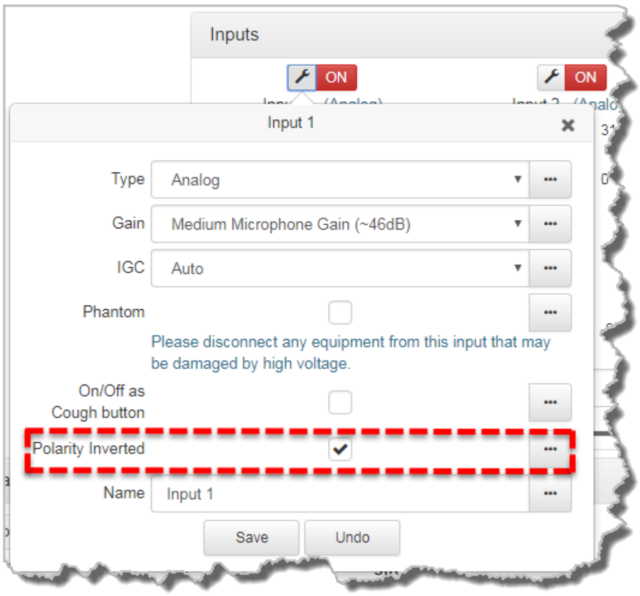


3. Click **Save** to confirm the change. Note: The label on the input changes from **ON** to **COUGH** when enabled.

Polarity Inverted

Select the **Polarity Inverted** check-box to invert the polarity of an input.

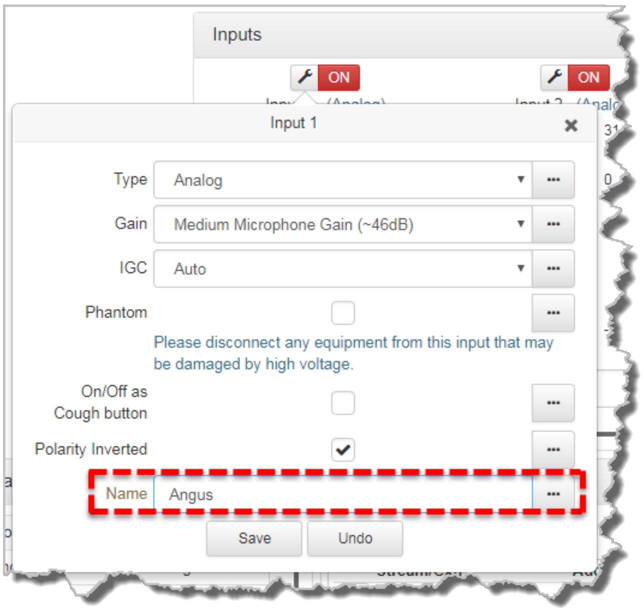
1. Click the **Input Settings** symbol on the input you want to adjust.
2. Click to select the **Polarity Inverted** check-box and reverse the phase of the selected input.



3. Click **Save** to confirm the change.

Renaming Inputs

1. Click the **Input Settings** symbol on the input you want to rename.
2. Click in the **Name** text box to enter a new name, or edit an existing name.



3. Click **Save** to confirm the change.

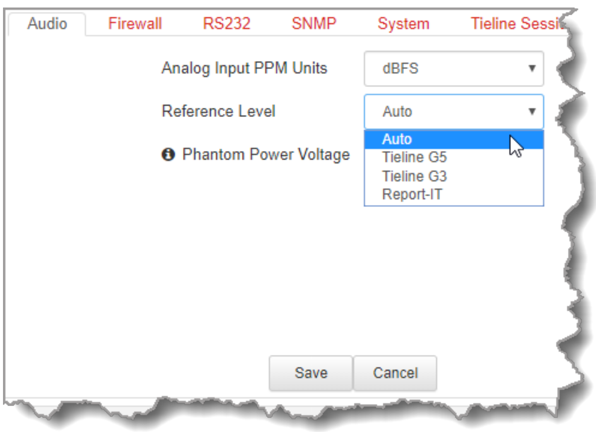
Audio Reference Levels

By default, codec the **PPM METERS** on the front of the codec, on the **TOUCH SCREEN**, or the HTML5 Toolbox Web-GUI, use dBFS to express nominal operating, headroom and noise floor levels. The codec can also automatically adapt to different Tieline reference scales. A Tieline codec with proprietary Tieline session data enabled will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. When connecting to a non-Tieline codec, or a Tieline codec without session data enabled, the codec will use the **Tieline G5** reference scale setting. See General Audio Settings for more details.

Important Note: When dialing a multistream program to both G3 and G5 codecs, by default the Audio Reference Level will be configured for the compatibility of the codec that connects first. I.e. if you dial a G3 codec first then the G3 Audio Reference Level will be configured for all connections.

To adjust this setting:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **Audio** and then click **Edit**.
3. Click the **Reference Level** drop-down menu and select the correct option.



33.12 Configure Mono or Stereo Peer-to-Peer Programs in ViA

The **Programs panel** incorporates a wizard to configure a new program and all audio stream settings. Before you configure a new codec program consider if:

- You want your codec to be capable of dialing and answering, dialing only or answering only.
- A backup connection is required.

This section contains instructions for:

1. Configuring Peer-to-Peer Programs: Dialing
2. Configuring a Backup Connection or Auto Reconnect
3. Configuring Answering Connections

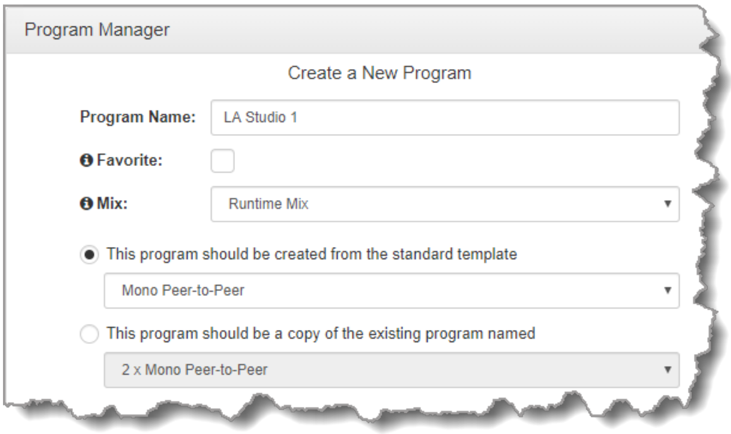
For more information about programs and audio streams within programs see the section titled Load, Connect and Manage Programs. Note: The following connection setup instructions will show how to configure a dial and answer program, with a backup connection. If you want the codec to either dial or answer only, select the option and the wizard will automatically display relevant screens to allow you to configure the codec correctly.

Configuring Peer-to-Peer Programs: Dialing

Important Notes: Before you start program configuration please note:

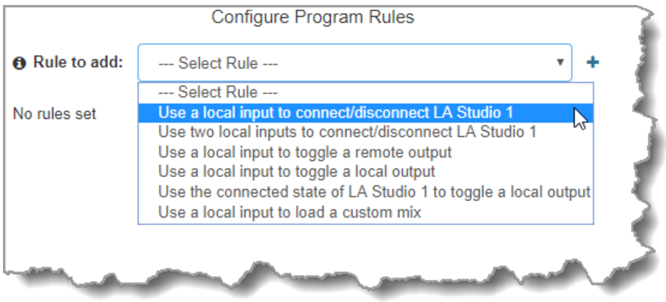
- It is not possible to edit a program when loaded in the codec.
- Lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see Load, Connect and Manage Programs.

1. Open the HTMLL5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option.
   - Click to select the **Favorite** check-box if you want to add the new program to the list of favorites in the codec.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required, then click **Next**.



Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

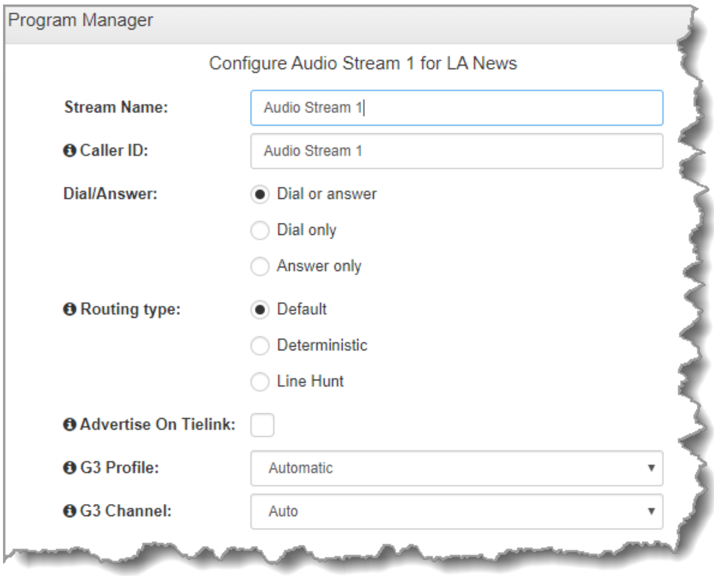
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT** GPIOs; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **SOURCE** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

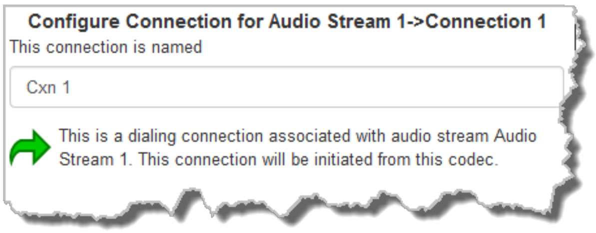
1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. Note: This is overridden in a ViA codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

Important Notes on G3 Channel Settings:

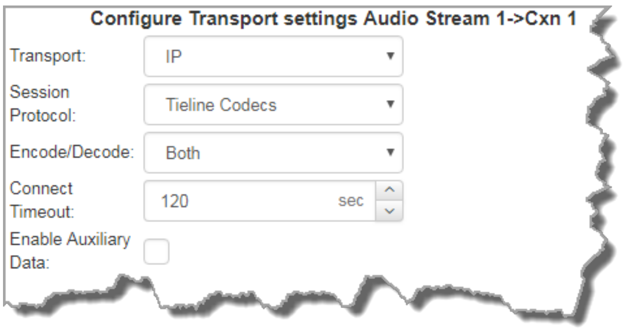
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



6. Configure the transport settings for the connection, then click **Next**. For SIP connections see Configuring SIP. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm.

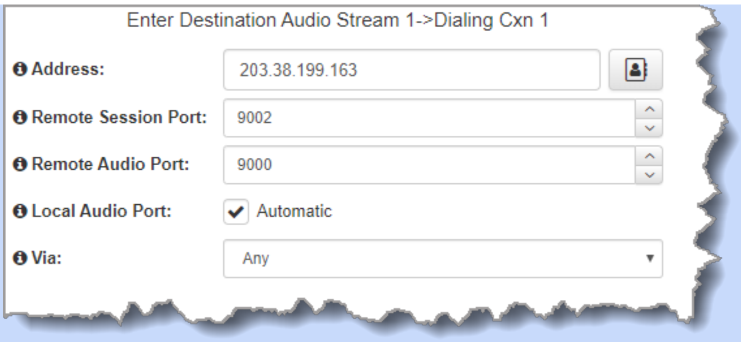


Important Note:
- If you select **Sessionless** or **SIP** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
- See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings. Enter an IP address, or click the **Address Book** button to select a TieLink contact.

IP

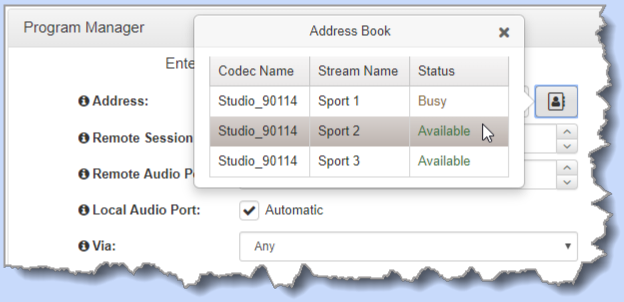
For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.



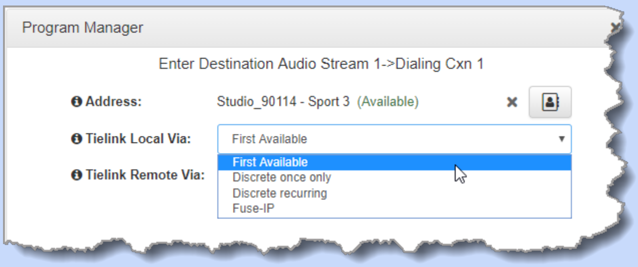
Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting. When you select **Sessionless** as the **Session Protocol**, the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.

If TieLink Contact List dialing is configured:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.

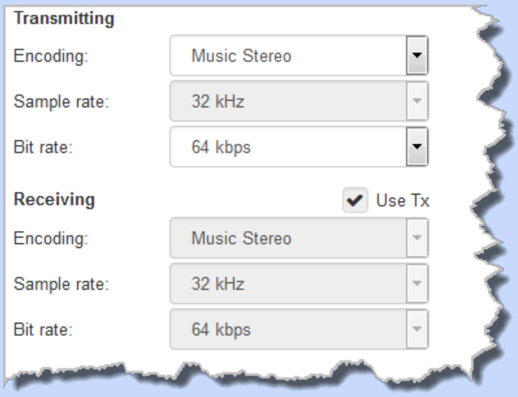


2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see [Configuring TieLink Settings](#) for more info.



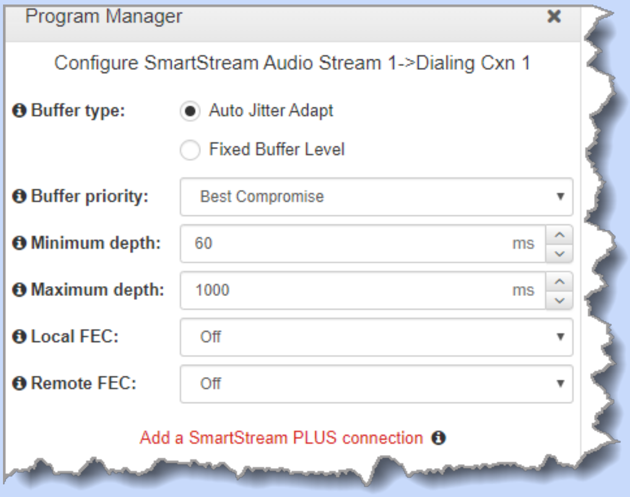
Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a failover connection or SmartStream PLUS for this audio stream (recommended).

Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.



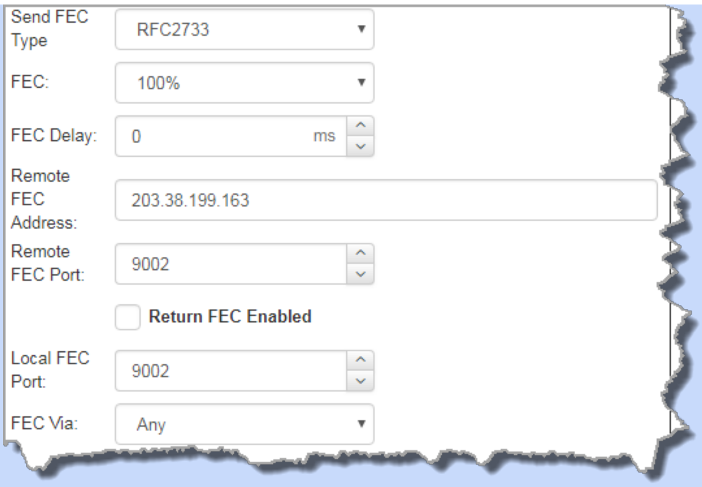
For IP connections click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.



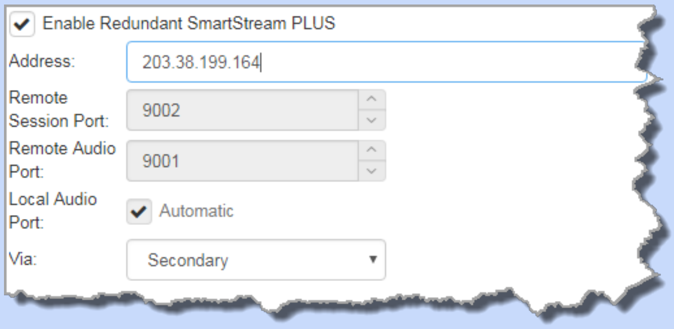
Important Notes:

- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent at the selected setting.



Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.

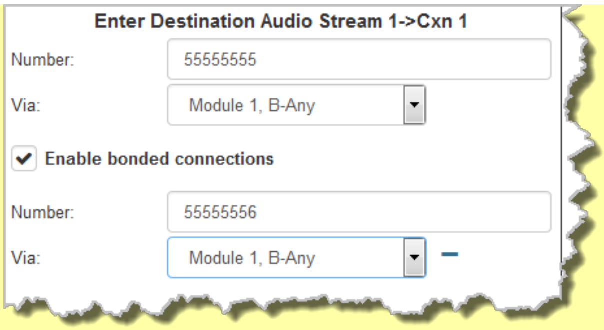
By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.



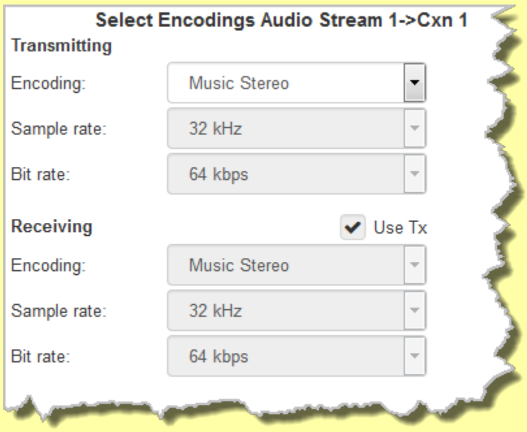
Important Notes:
- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

ISDN

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



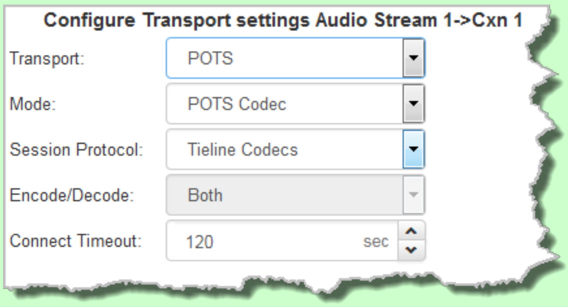
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



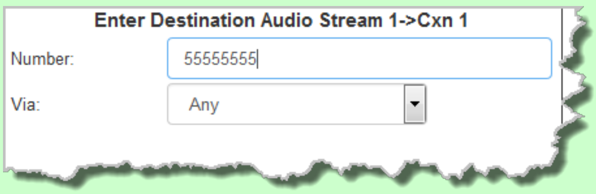
Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



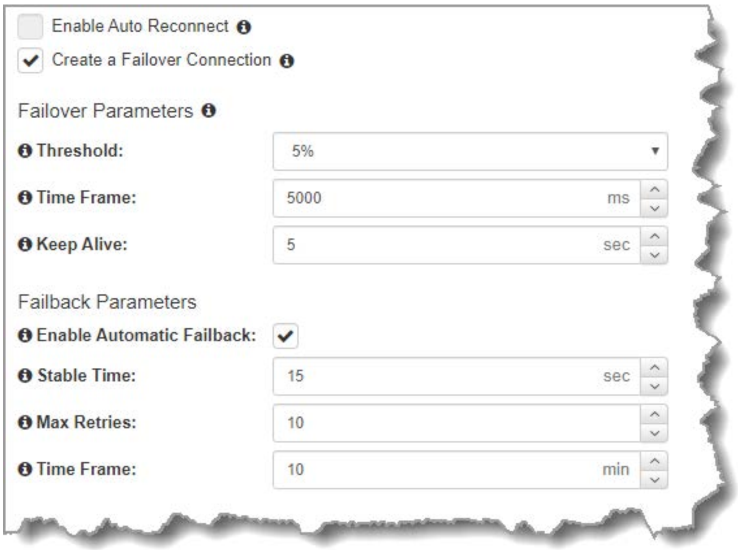
Dialing settings for this POTS audio stream are now complete.

Configuring a Failover Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a failover connection for the audio stream you are configuring.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the Web-GUI.

1. Click to select the check-box for **Create a Failover Connection**. Adjust the parameters and click **Next**.



Important Note: When Failover is enabled, the codec will fail over to a backup connection under the following circumstances:

- There is sudden loss of connectivity to the primary destination.
- The remote codec disconnects the primary connection.
- The user manually attempts to connect to the failover connection.
- When the conditions for failover are triggered by the **Failover Parameters**.

The codec will always attempt Failback if any of the following conditions are met:

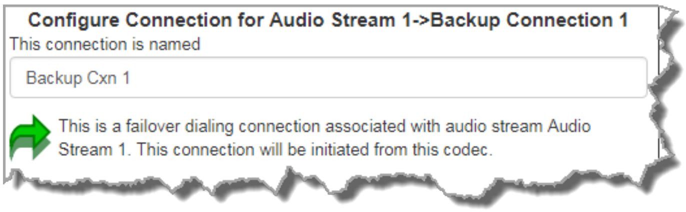
- There is sudden loss of connectivity to the backup destination.
- The remote codec disconnects the backup connection.
- The user manually attempts to connect to the primary connection.

The explanations within the following table can be used to assist with failover connection configuration.

	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity
4	Enable Automatic Failback	Select the check-box to fail back to a higher priority connection when failback parameters are met

5	Stable Time	The amount of time a primary connection must remain stable before attempting to fail back from the backup connection
6	Maximum Retries	The maximum number of fail back retries a codec can try before ending fail back attempts
7	Time Frame	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the backup connection and click **Next**.

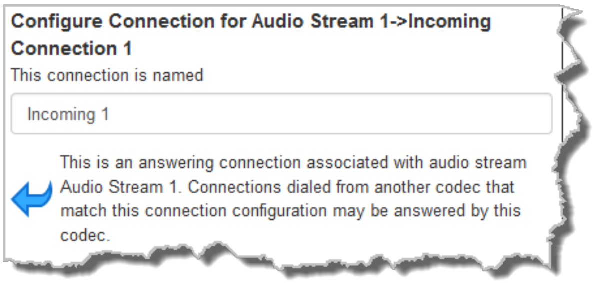


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

Configuring ViA to Answer Connections

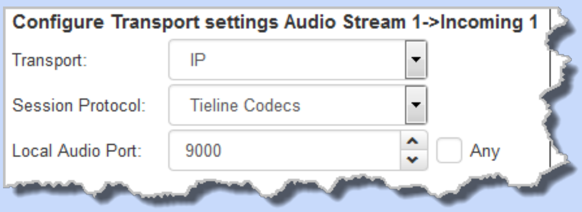
The codec is capable of being configured to accept calls via different transports (e.g. IP, ISDN and POTS), or to accept calls using different audio ports. To answer one or more incoming audio stream connections:

1. Enter a name for the answering connection and click **Next**.



2. Configure the transport settings:

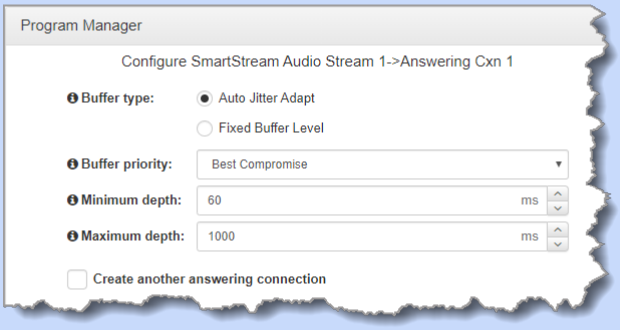
IP For IP select the **Session Protocol** and **Audio Port**.



Important Note: The **Local Audio Port** is the port used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the **Local Audio Port** is automatically configured as UDP audio port 9000 by default for the first audio stream connection. Click to deselect the **Any** check-box to adjust this setting.

Click **Next** to specify jitter buffer settings, or create another answering connection. Click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details, or
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.



ISDN For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN Answering.

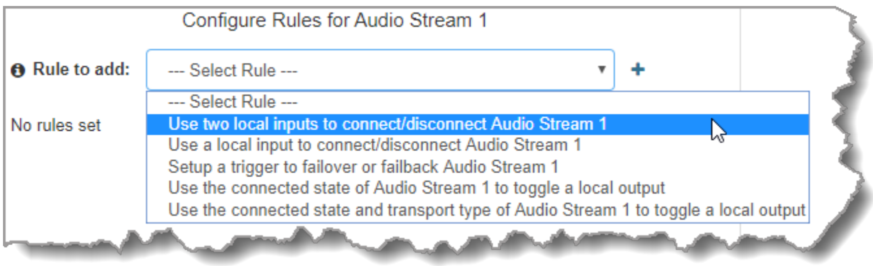
POTS For POTS, settings are determined by POTS module answering settings. For more details see Configuring POTS Answering.

3. After configuring all settings there are 3 options:

   i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
   ii. Click **Save Program** to save the program at this point.
   iii. Click **Next** to configure rules options.

Configuring Rules

1. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Note: Connection-related rules are not displayed in **Answer only** audio streams.

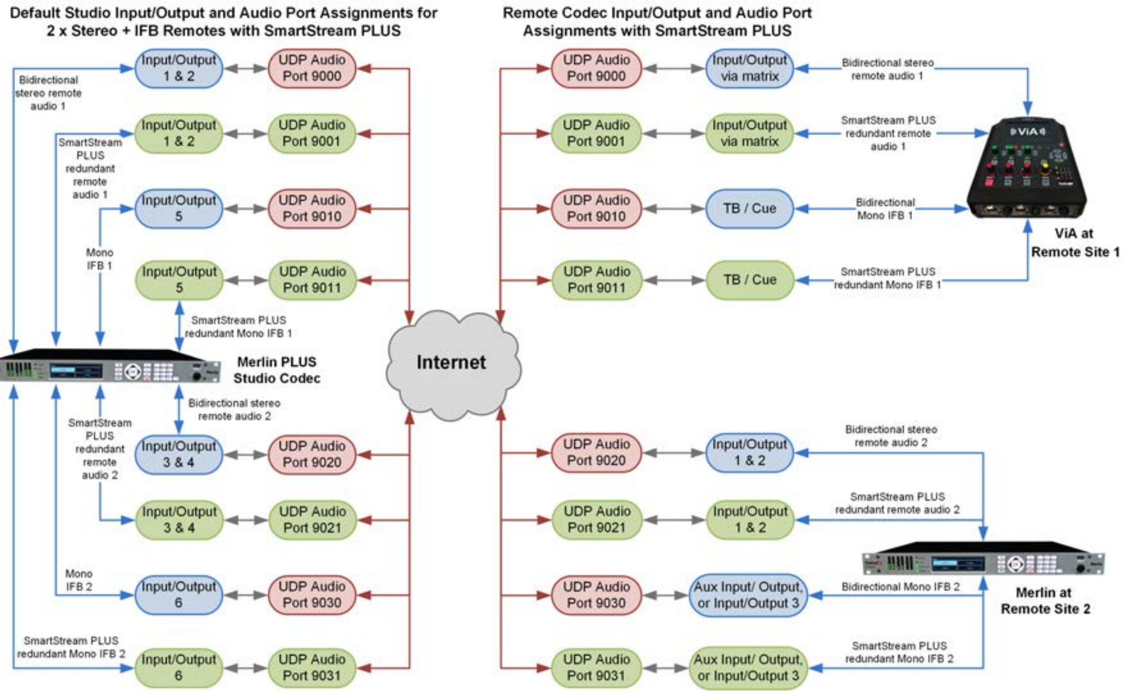
2. Click **Save Program** to save the program.
3. Click **Finish** to exit the wizard or **Load** to load the program.
4. The newly created program can be loaded from within the **Program Manager panel**, **Connections panel** and the **Program Loader panel** (in the Quick Connect web-GUI). Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec **TOUCH SCREEN**.

33.13 Configure Mono or Stereo + IFB Dialing Programs

This program is designed to allow ViA to dial a Merlin or Merlin PLUS codec at the studio and transmit:

1. A bidirectional mono or stereo audio stream connection.
2. A separate bidirectional mono IFB audio stream for communications.

This program can also incorporate SmartStream PLUS redundant IP streaming. The following diagram indicates the default input, output and port assignments for 2 x **Mono or Stereo Peer-to-Peer + IFB** Programs using SmartStream PLUS to dial a Merlin PLUS codec at the studio.



2 x Mono/Stereo Peer-to-Peer + IFB Remotes dialing Merlin PLUS at the studio

The following setup instructions describe how to configure a dialing stereo audio stream and IFB audio stream, with a failover (backup) connection.

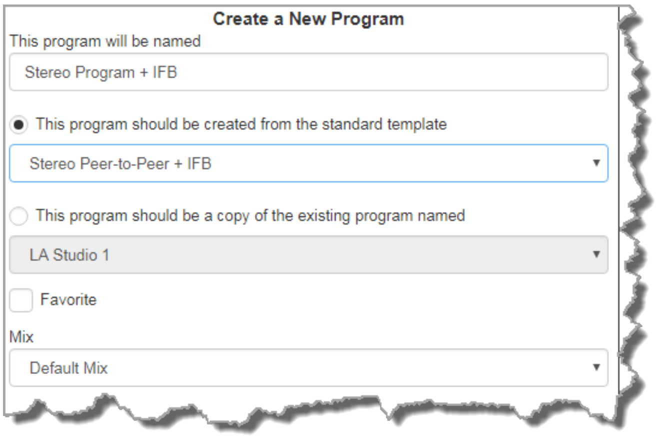
Configuring a Mono or Stereo Audio Stream: Dialing

Important Notes: Before you commence program configuration please note:

- When an announcer presses an input's **TB CUE** button, audio is routed to the IFB audio stream encoder and outgoing audio is simultaneously monitored in the right side of the selected input's headphone output. Return IFB audio is also audible in the right side of the headphones. For more info see Cue and Talkback Operation.
- The auxiliary input is used by default for the IFB communications channel in Merlin codecs. In Merlin PLUS codecs the auxiliary input and XLR input 3 are mixed together by default for Mono/Stereo Peer-to-Peer + IFB programs.
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.

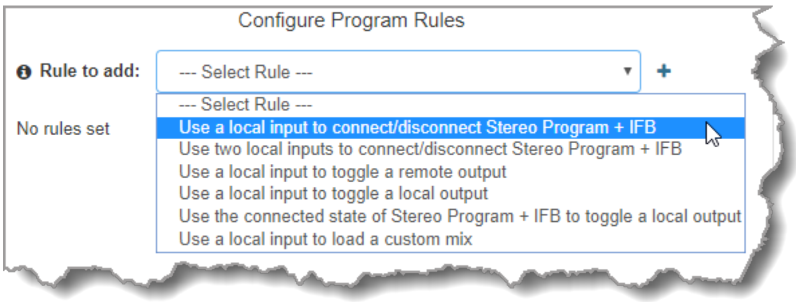
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
- POTS is not supported for stereo audio stream connections.
- To learn more about programs see the section titled Load, Connect and Manage Programs.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer + IFB,** or if you want to use an existing program as a template, select this option.
   - Click to select the **Favorite** check-box if you want to add the new program to the list of favorites in the codec. Note: The following example is configured to connect a stereo audio stream and mono IFB stream.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required, then click **Next**.



Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.

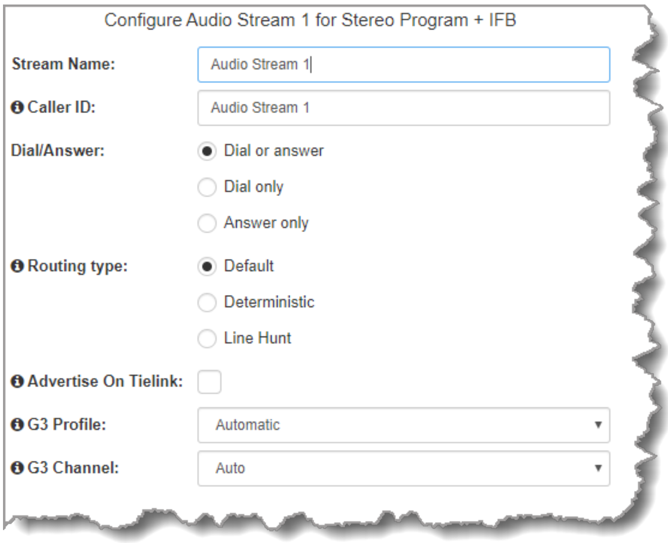


Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT** GPIOs; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **SOURCE** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.

- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

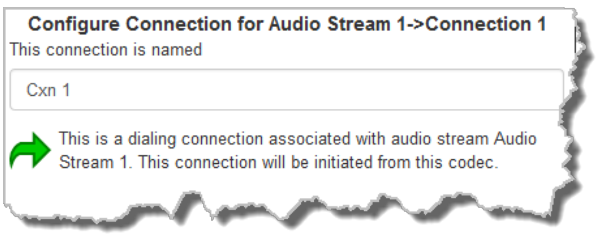
Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. This is overridden in a Merlin or Merlin PLUS codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

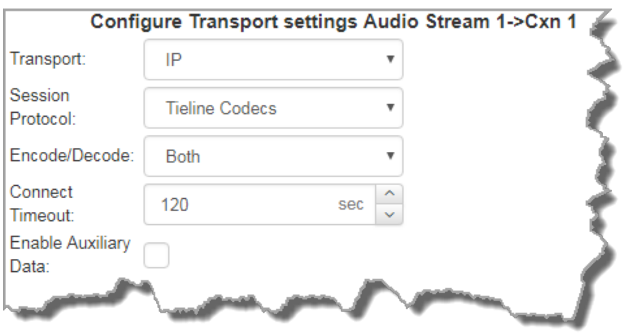
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

5. This audio stream connection in the wizard will allow the codec to dial. Enter the connection name in the text box, then click **Next**.



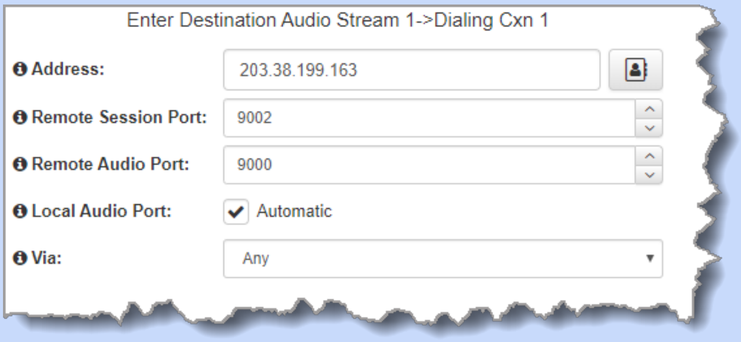
6. Configure the transport settings for the connection, then click **Next**. For SIP connections see Configuring SIP. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm.



Important Note: See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings. Enter an IP address, or click the **Address Book** button to select a TieLink contact.

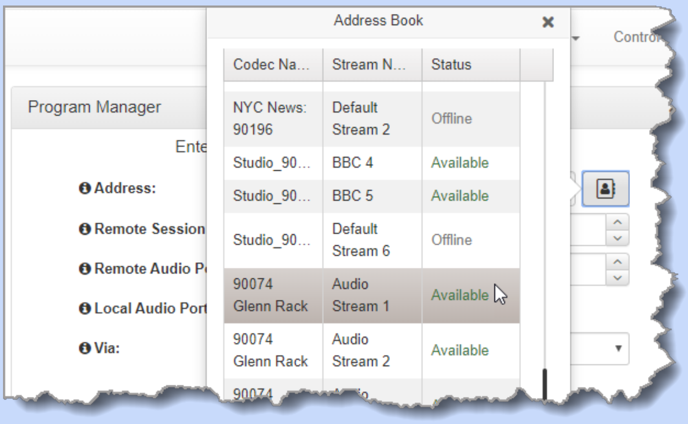
IP For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.



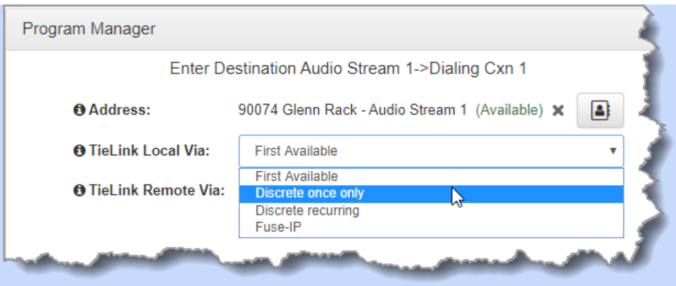
Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting. When you select **Sessionless** as the **Session Protocol**, the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.

If TieLink Contact List dialing is configured:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.



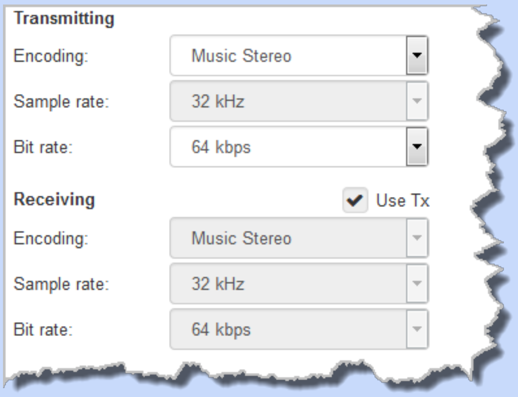
2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see Configuring TieLink Settings for more info.



Click **Next Stream** to configure the first audio stream with default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a failover connection or SmartStream PLUS for this audio stream (recommended).

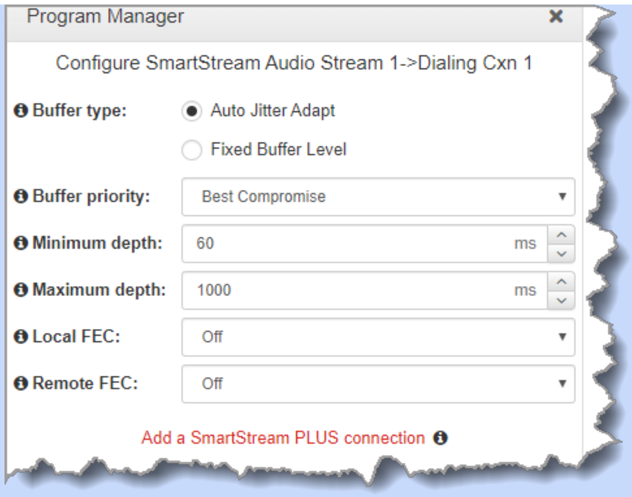
Note: If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio (when creating 2 x Mono or Stereo Peer-to-Peer + IFB connections), use **Remote Audio Port 9020** to configure the second mono/stereo dialing connection at the studio. Mono or stereo program audio over this audio stream connection will be routed via audio inputs/outputs 3 and 4 on the studio Merlin PLUS codec.

When adjusting encoding settings, click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.



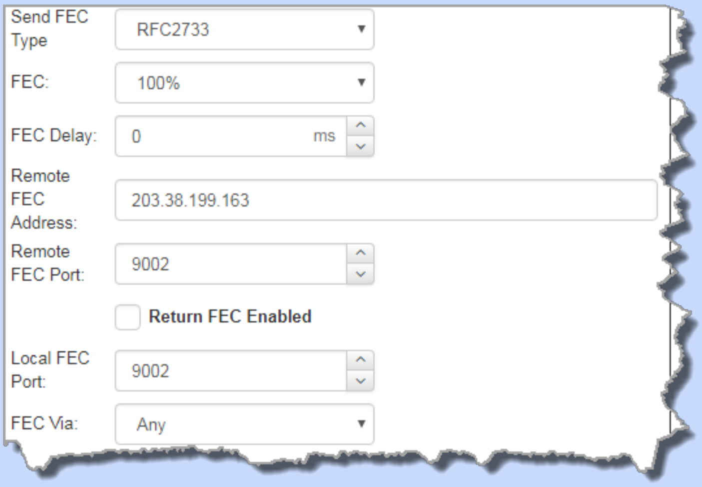
For IP connections click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.



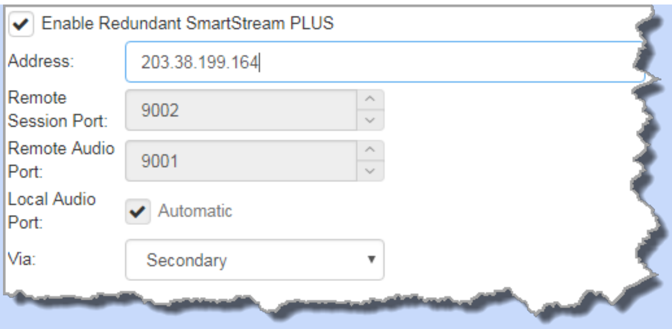
Important Notes:

- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent at the selected setting.



Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.

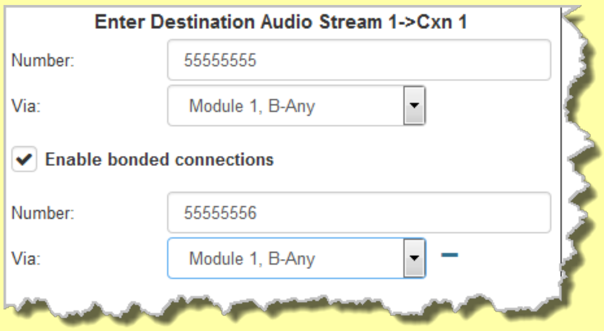
By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.



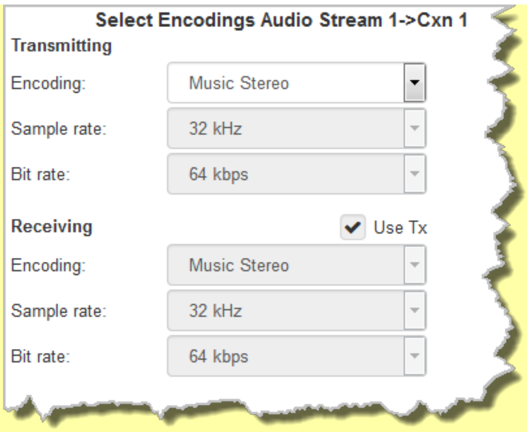
Important Notes:

- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



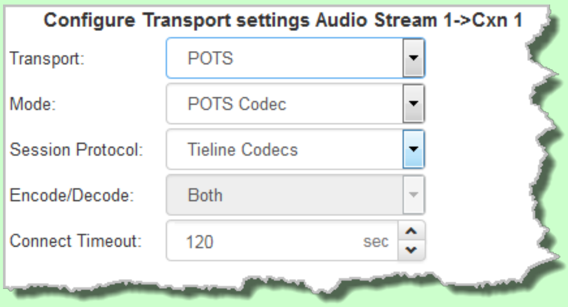
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



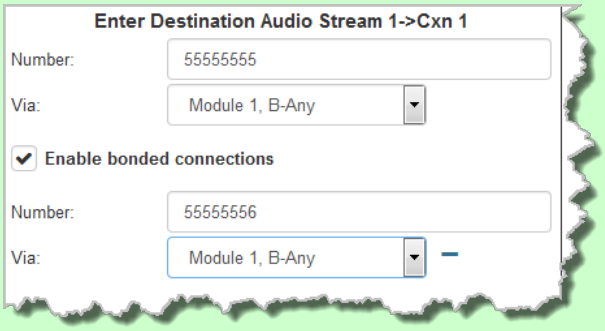
Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



Dialing configuration settings for this POTS audio stream are now complete.

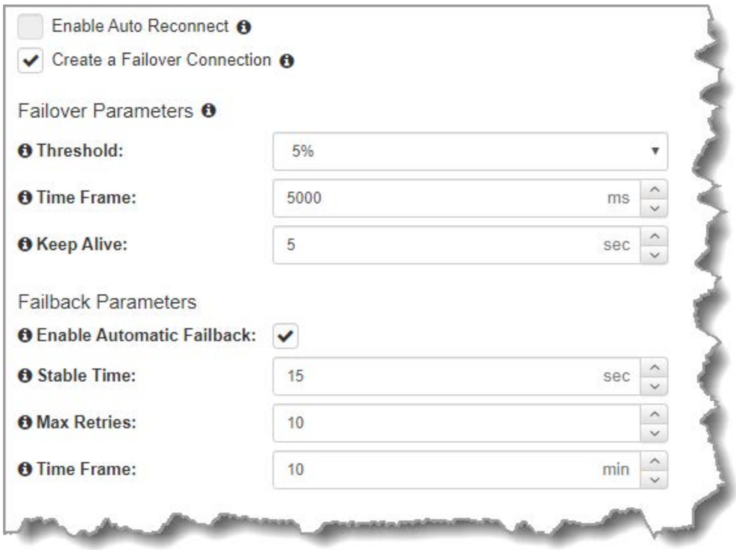
Configuring a Failover Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a failover connection for the audio stream you are configuring.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the Web-GUI.

To configure a backup connection:

1. Click to select the check-box for **Create a Failover Connection**. Adjust the parameters and click **Next**.



Important Note: When Failover is enabled, the codec will fail over to a backup connection under the following circumstances:

- There is sudden loss of connectivity to the primary destination.
- The remote codec disconnects the primary connection.
- The user manually attempts to connect to the failover connection.
- When the conditions for failover are triggered by the **Failover Parameters**.

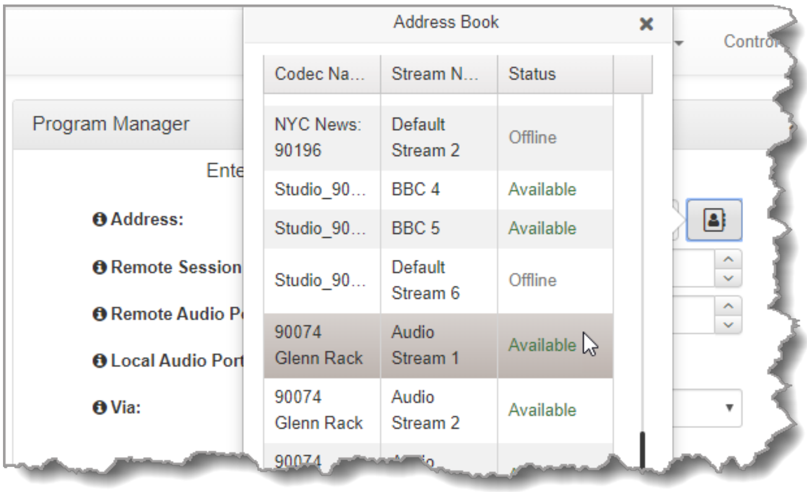
The codec will always attempt Failback if any of the following conditions are met:

- There is sudden loss of connectivity to the backup destination.
- The remote codec disconnects the backup connection.
- The user manually attempts to connect to the primary connection.

The explanations within the following table can be used to assist with failover connection configuration.

	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity
4	Enable Automatic Failback	Select the check-box to fail back to a higher priority connection when failback parameters are met
5	Stable Time	The amount of time a primary connection must remain stable before attempting to fail back from the backup connection
6	Maximum Retries	The maximum number of fail back retries a codec can try before ending fail back attempts
7	Time Frame	The time frame used to measure the number of fail back retries attempted

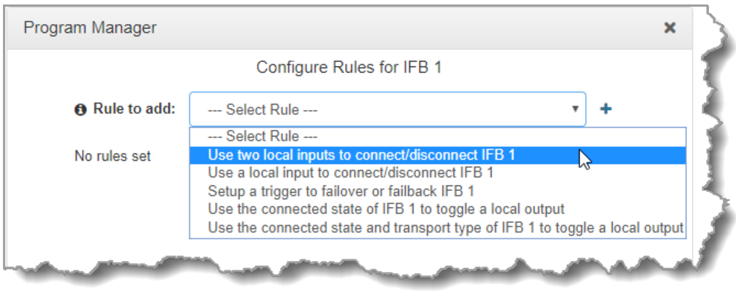
2. Enter a name for the backup connection.



3. After configuring all settings there are 2 options:

   i. Click **Next** to configure rules options.
   ii. Click **Next Stream** to configure the second audio stream.

4. To configure new rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



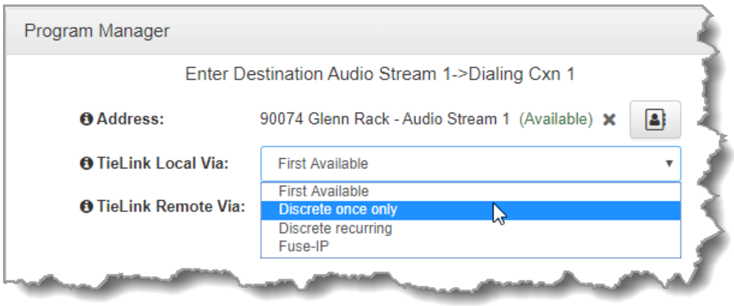
Important Note: Connection-related rules are not displayed in **Answer only** audio streams.

5. Click **Next** to configure the IFB audio stream.

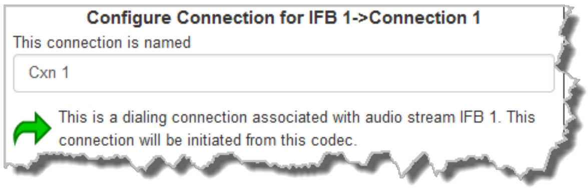
Configure the Bidirectional IFB Audio Stream

When you have finished configuring SmartStream PLUS, Auto Reconnect or a backup connection, proceed with configuration of the IFB audio stream in the wizard.

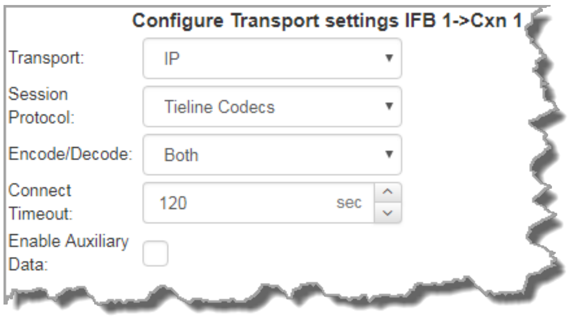
1. Enter the IFB **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



2. This audio stream connection in the wizard will allow the codec to dial. Enter the connection name in the text box, then click **Next**



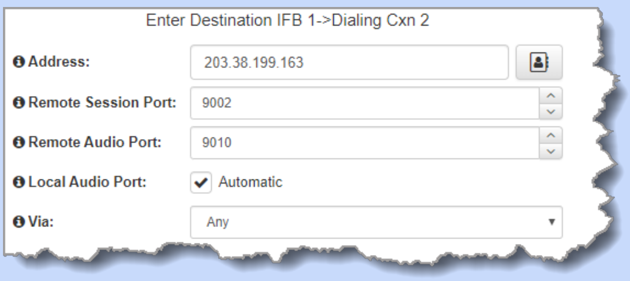
3. Configure the transport settings for the connection, then click **Next**.



Important Note: See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

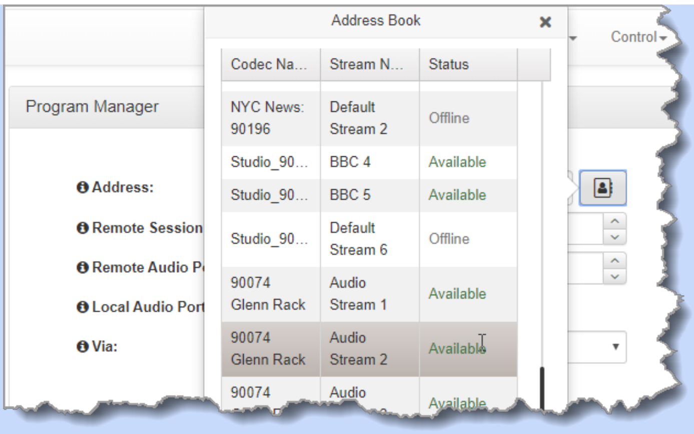
4. Configure destination codec dialing and encoding settings:

For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

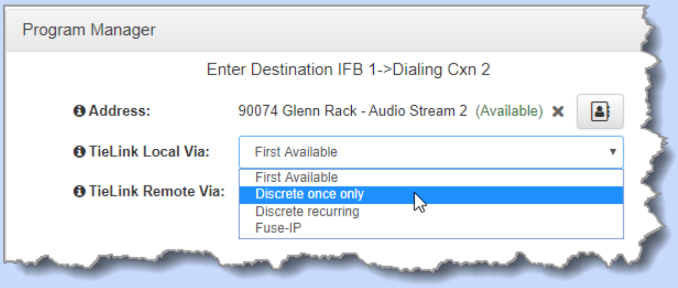


If TieLink Contact List dialing is configured:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.



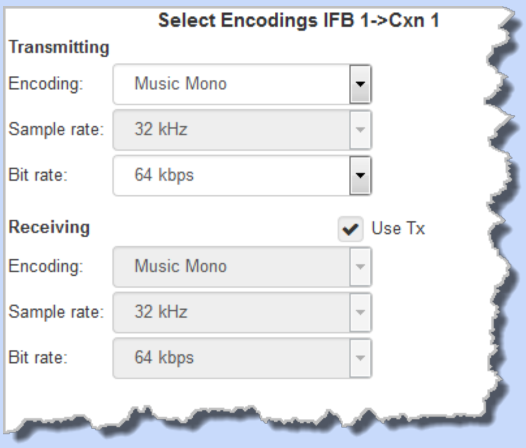
2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see [Configuring TieLink Settings] for more info.



Click **Save Program** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a backup connection or SmartStream PLUS for this audio stream (recommended).

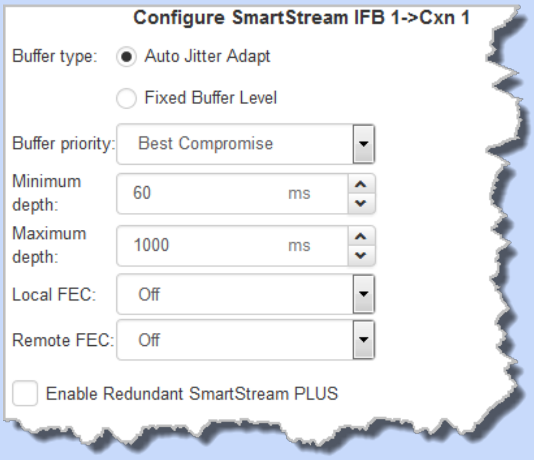
Note: The default **Remote Audio Port** is **9010** for this IP audio stream. If you connect multiple remote codecs simultaneously to a Merlin PLUS codec at the studio (when creating 2 x Mono or Stereo Peer-to-Peer + IFB connections), use **Remote Audio Port 9030** to configure the second IFB connection at the studio. IFB audio over this audio stream connection will be routed via audio input/output 6 on the studio Merlin PLUS codec.

Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** and **Bit rate** options.

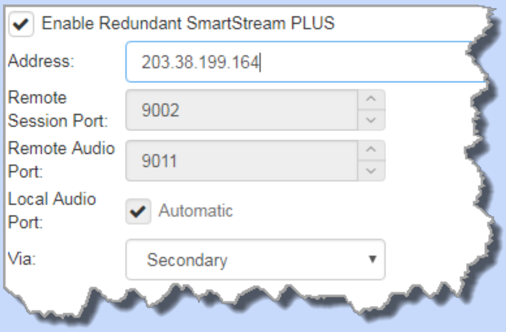


For IP connections click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.

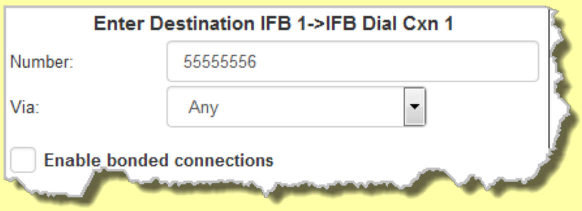


Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.



By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9011** by default, and provides automatic IP streaming backup in case one IP connection fails.

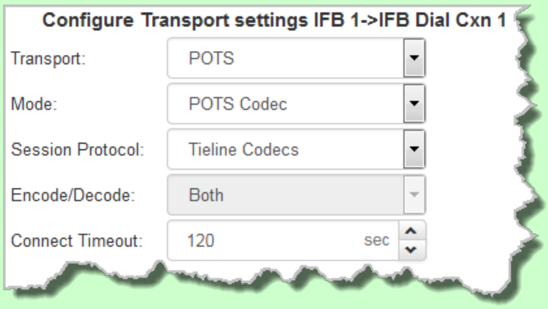
For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



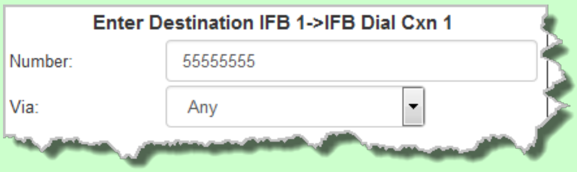
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended). Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).

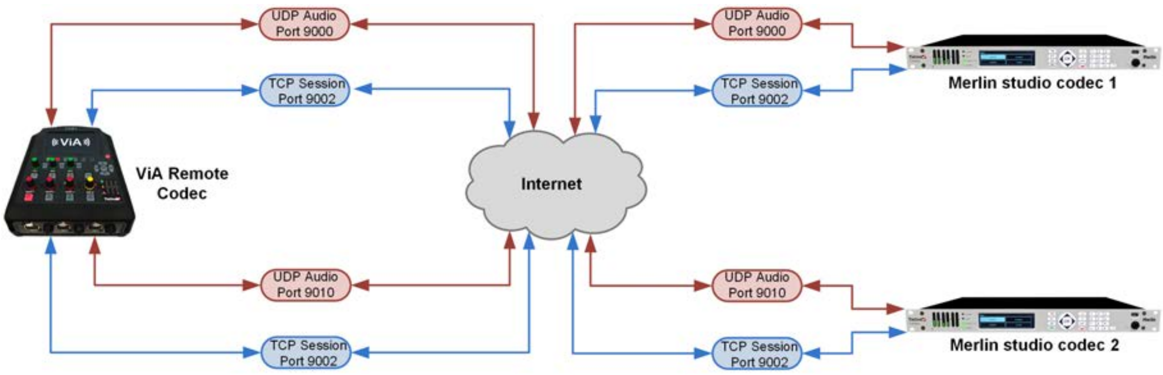


Dialing configuration settings for this POTS audio stream are now complete.

5. Continue through the steps in the wizard to complete configuration in the same way as the first connection was configured. Then click **Save Program** to complete configuration, then click **Finish** to exit the wizard or **Load** to load the program.
6. The newly created program can be loaded from within the **Program Manager panel** or **Connections panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

33.14 Configure 2 Mono Peer-to-Peer Dialing Connections

ViA is capable of sending a mono program mix to two separate codecs over different connections.



ViA 2 x Mono Peer-to-Peer Dialing Configuration

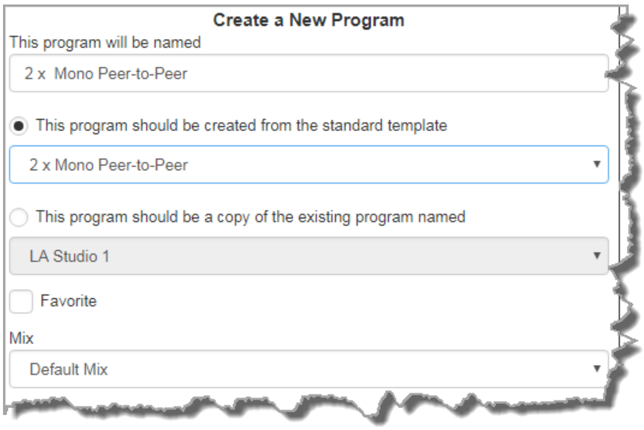
This requires the creation of a dialing program in ViA with two separate mono audio streams and associated dialing connections.

Configure a 2 x Mono Peer-to-Peer Dialing Program

Important Notes:

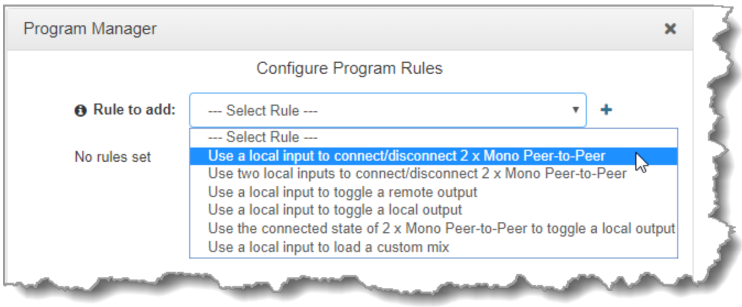
- A program cannot be edited when it is loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save audio stream or program settings at several points throughout the program wizard and use default settings to save configuration time.
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
- When dialing a multistream program to both G3 and G5 codecs, by default the Audio Reference Level will be configured for the compatibility of the codec that connects first. I.e. if you dial a G3 codec first then the G3 Audio Reference Level will be configured for all connections.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **2 x Mono Peer-to-Peer,** or if you want to use an existing program as a template, select this option.
   - Click to select the **Favorite** check-box if you want to add the new program to the list of favorites in the codec. See Load, Connect and Manage Programs for more details.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required, then click **Next**.



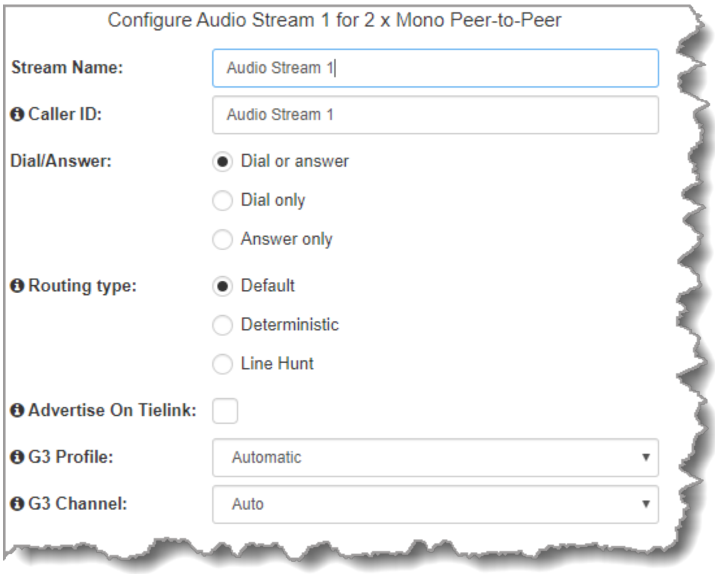
Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:
- The codec has 4 physical **CONTROL PORT** GPIOs; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **SOURCE** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

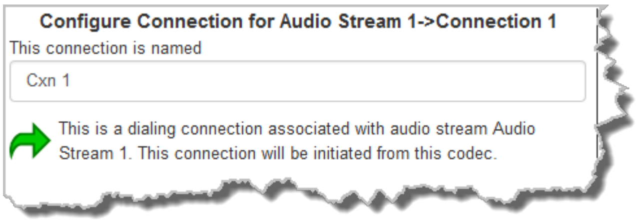
1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. This is overridden in a Merlin or Merlin PLUS codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

Important Notes on G3 Channel Settings:

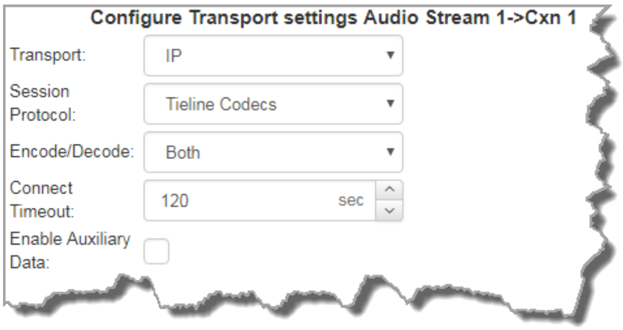
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

5. This audio stream connection in the wizard will allow the codec to dial and connect the first audio stream of the two audio streams being configured. Enter the name of the connection in the text box, then click **Next**.



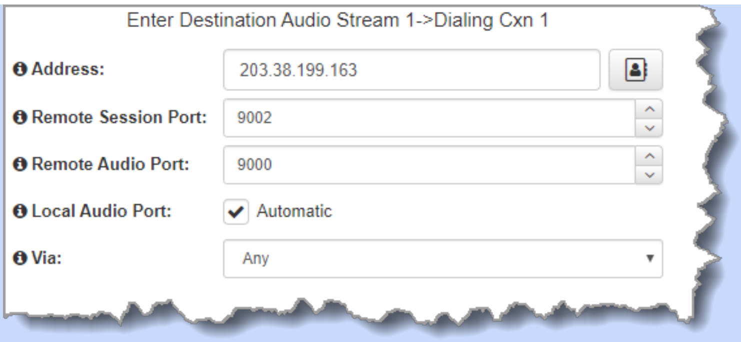
6. Configure the transport settings for the connection, then click **Next**. For SIP connections see Configuring SIP. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm.



Important Note:
- If you select **Sessionless** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
- See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings. Enter an IP address, or click the **Address Book** button to select a TieLink contact.

For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

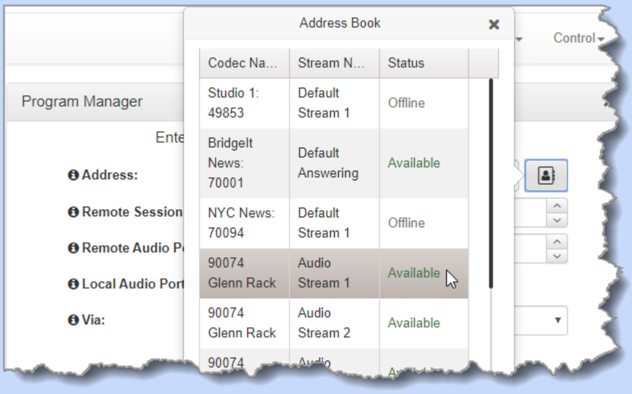


Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting. When you select **Sessionless** as the **Session Protocol**, the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.

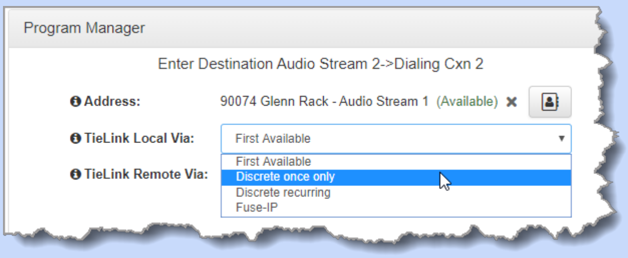
Session Port 9002 and **Remote Audio Port 9000** are used by default for the first IP audio stream. The audio for this audio steam will use analog or digital input and output 1 for both the dialing and answering codecs. If you need to change default port settings on the codec from which the call originates, this will also need to be adjusted in the answering codec using a custom program.

If TieLink Contact List dialing is configured:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.

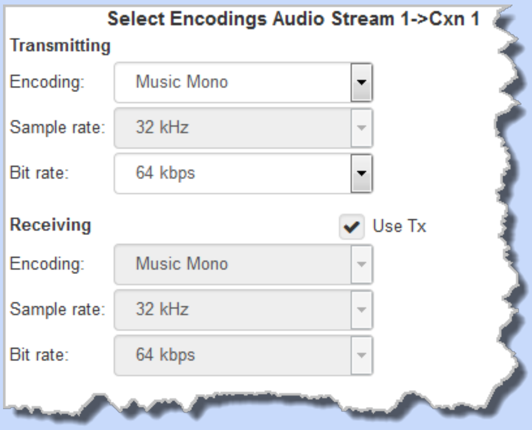


2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see [Configuring TieLink Settings] for more info.



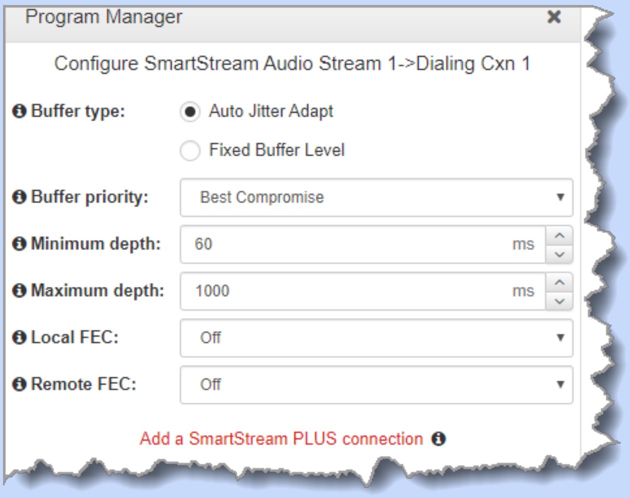
Click **Next Stream** to configure the first audio stream with default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a failover connection or SmartStream PLUS for this audio stream (recommended).

Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** or **Bit rate** options.



For IP connections click to configure:

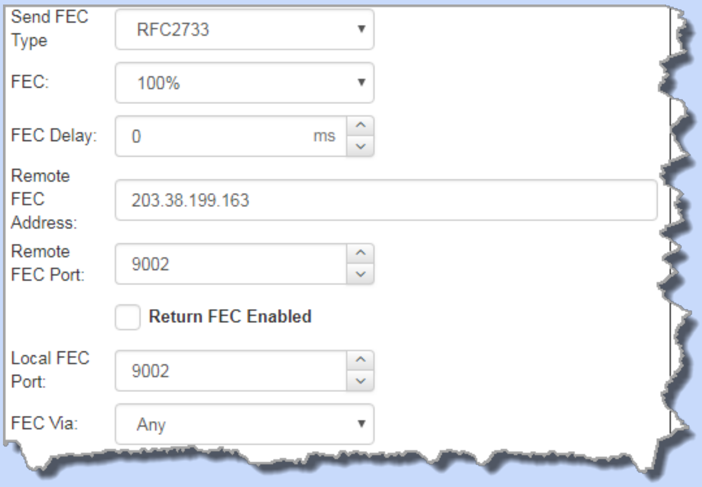
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- **Local** and **Remote FEC** settings if required.



Important Notes:

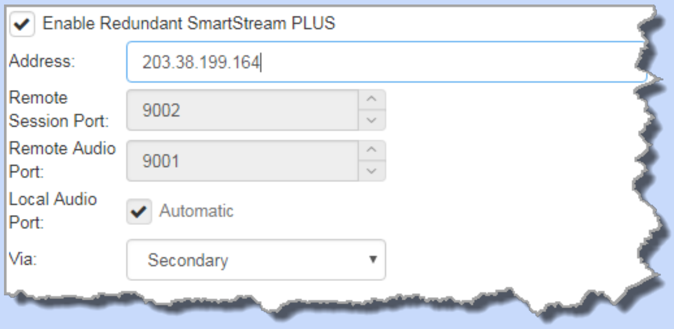
- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically

delayed based on the ratio of primary packets to FEC packets sent at the selected setting.



Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.

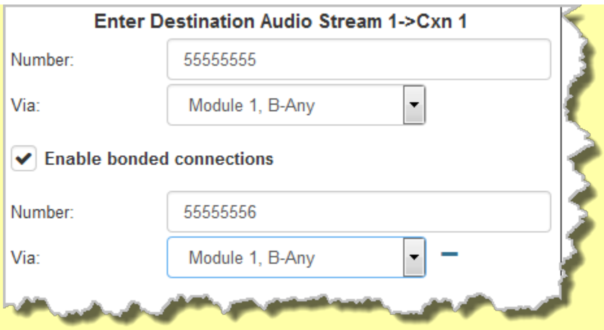
By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.



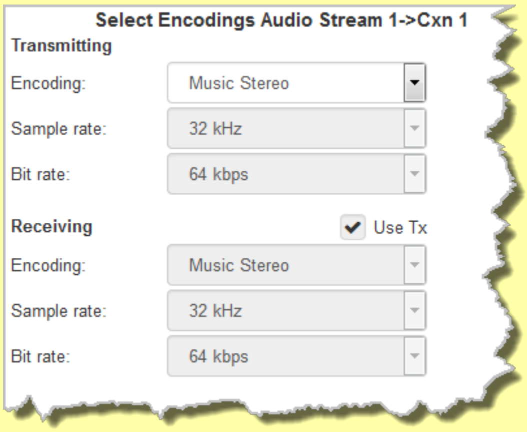
Important Notes:
- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

ISDN

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



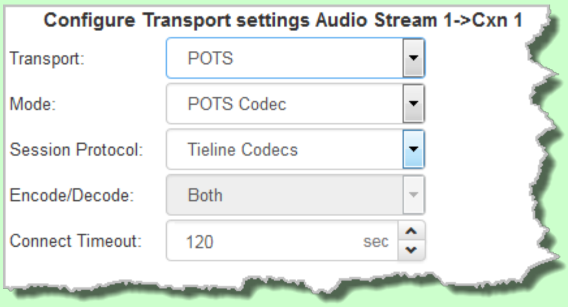
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



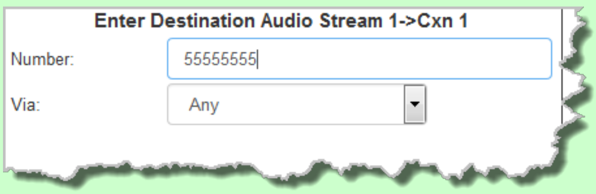
Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2** to specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



Dialing configuration settings for this POTS audio stream are now complete.

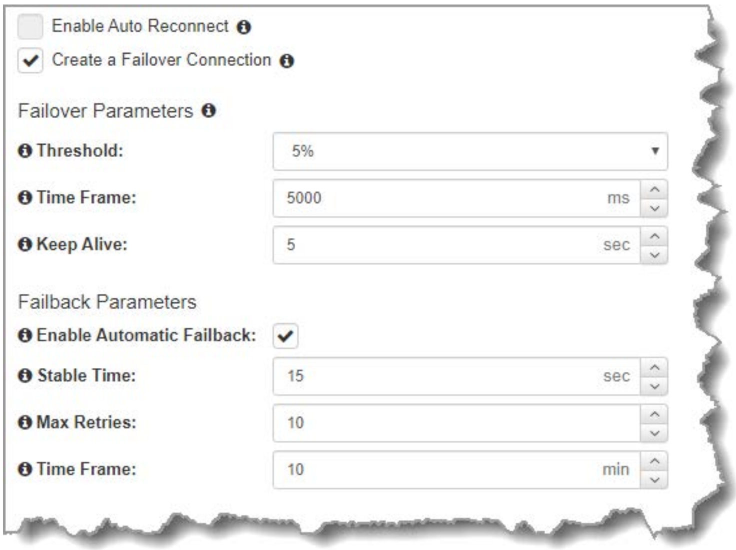
Configuring a Failover Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a failover connection for the audio stream you are configuring.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the Web-GUI.

To configure a backup connection:

1. Click to select the check-box for **Create a Failover Connection**. Adjust the parameters and click **Next**.



Important Note: When Failover is enabled, the codec will fail over to a backup connection under the following circumstances:

- There is sudden loss of connectivity to the primary destination.
- The remote codec disconnects the primary connection.
- The user manually attempts to connect to the failover connection.
- When the conditions for failover are triggered by the **Failover Parameters**.

The codec will always attempt Failback if any of the following conditions are met:

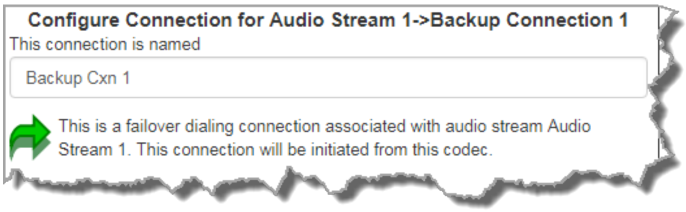
- There is sudden loss of connectivity to the backup destination.
- The remote codec disconnects the backup connection.
- The user manually attempts to connect to the primary connection.

The explanations within the following table can be used to assist with failover connection configuration.

	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to

		confirm connectivity
4	Enable Automatic Failback	Select the check-box to fail back to a higher priority connection when failback parameters are met
5	Stable Time	The amount of time a primary connection must remain stable before attempting to fail back from the backup connection
6	Maximum Retries	The maximum number of fail back retries a codec can try before ending fail back attempts
7	Time Frame	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the backup connection and click **Next**.

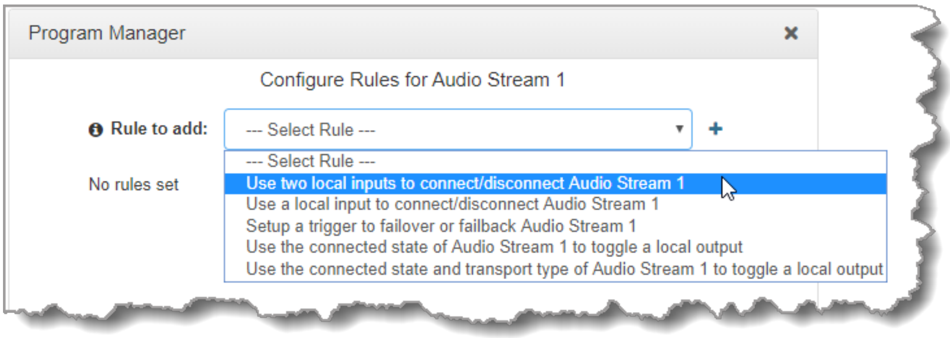


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

After configuring these settings there are 2 options:

i. Click **Next** to configure rules options.
ii. Click **Next Stream** to configure the second audio stream.

4. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.

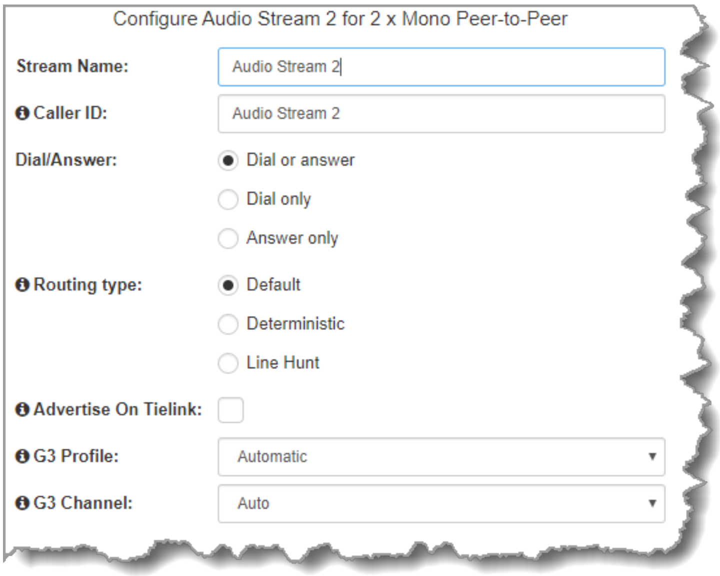


Important Note: Connection-related rules are not displayed in **Answer only** audio streams.

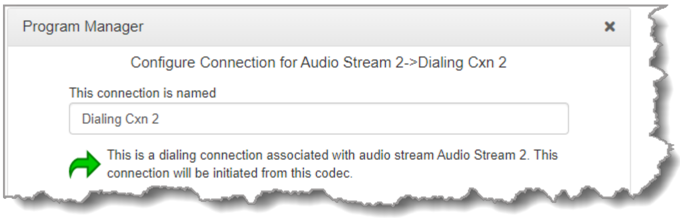
5. Click **Next** to configure the second audio stream.

Configure the Second Peer-to-Peer Audio Stream

1. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



2. This audio stream connection in the wizard will allow the codec to dial and connect the second audio stream. Enter the name of the connection in the text box, then click **Next**.



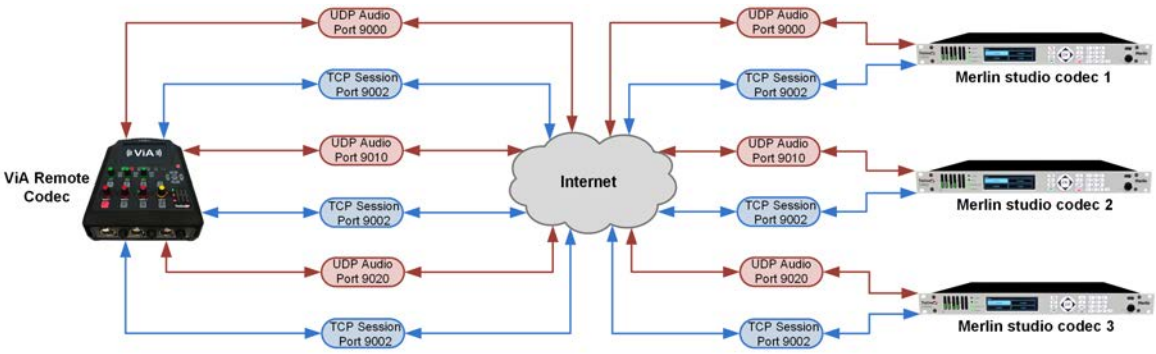
3. Continue through the steps in the wizard to complete configuration in the same way as the first peer-to-peer connection was configured. Click **Save Program** to save all program settings, then click **Finish** to exit the wizard, or **Load** to load the program.

Important Notes: Session Port 9002 and **Remote Audio Port 9010** are used by default for the second IP connection. If you connect to a G3 codec using IP2 you will need to change the session port on the answering codec to 9012 for the second audio stream connection, as this is used by default on G3 codecs for IP2.

4. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

33.15 Configure 3 Mono Peer-to-Peer Dialing Connections

ViA is capable of sending a mono program mix to three separate codecs over different connections.



ViA 3 x Mono Peer-to-Peer Dialing Configuration

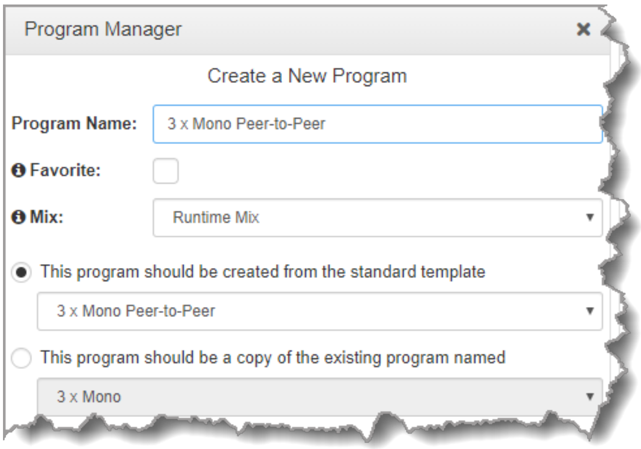
This requires the creation of a dialing program in ViA with three separate mono audio streams and associated dialing connections.

Configure a 3 x Mono Peer-to-Peer Dialing Program

Important Notes:

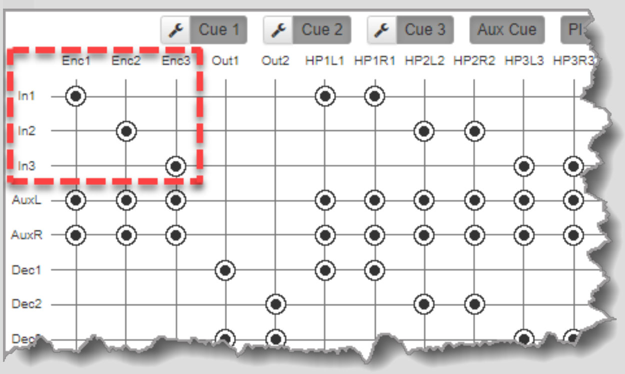
- You cannot edit a program when it is currently loaded in the codec.
- You can lock a loaded custom program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with a different type of program.
- Some drop-down menus and settings may be greyed out intentionally depending on features available and the transport selected (e.g. IP or ISDN).
- It is possible to save audio stream or program settings at several points throughout the program wizard and use default settings to save configuration time.
- Failover and SmartStream PLUS redundant streaming are not available with SIP or sessionless IP connections.
- When dialing a multistream program to both G3 and G5 codecs, by default the Audio Reference Level will be configured for the compatibility of the codec that connects first. I.e. if you dial a G3 codec first then the G3 Audio Reference Level will be configured for all connections.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **3 x Mono Peer-to-Peer,** or if you want to use an existing program as a template, select this option.
   - Click to select the **Favorite** check-box if you want to add the new program to the list of favorites in the codec. See Load, Connect and Manage Programs for more details.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required, then click **Next**.

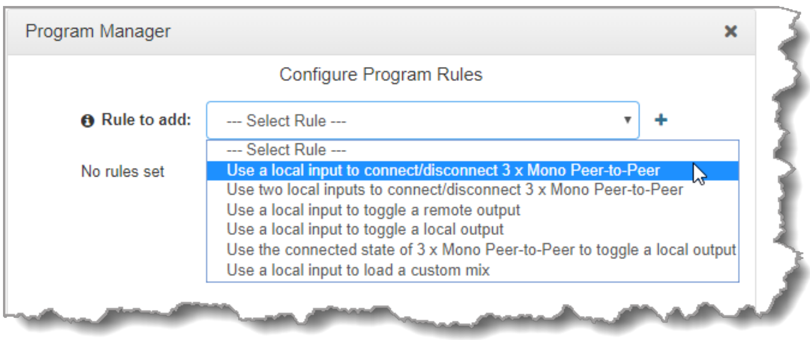


Important Notes:

- When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.
- The **3 x Mono Peer-to-Peer** program matrix by default has Input 1 routed to Encoder 1, Input 2 routed to Encoder 2 and Input 3 routed to Encoder 3. Edit the matrix as required to include or exclude inputs in each mono audio stream.



3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.

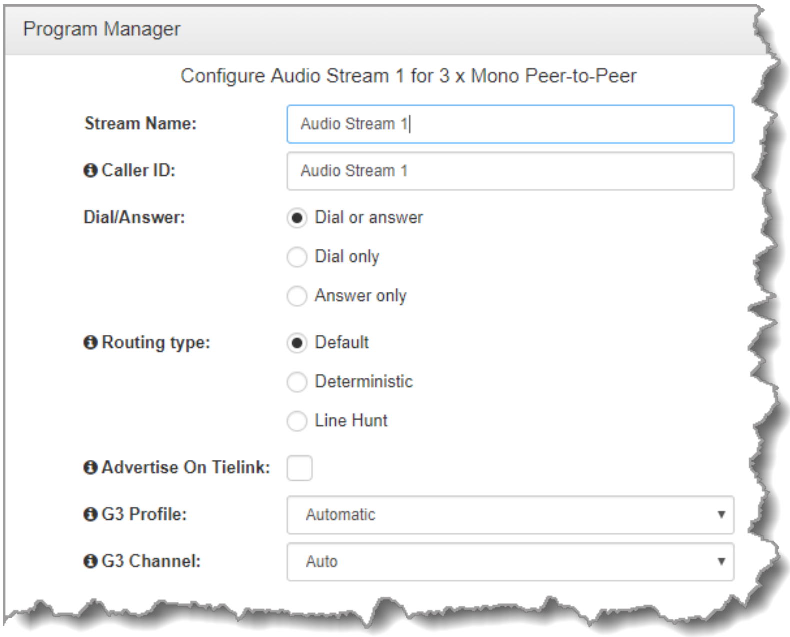


Important Notes for Rules:

- The codec has 4 physical **CONTROL PORT** GPIOs; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **SOURCE** buttons 1-3.

- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **caller ID** and configure the codec to **Dial only**. Then click **Next**. Note: The caller ID is used to identify calls. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



Routing Type Options:	
Default ISDN POTS	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.
Deterministic IP ISDN POTS	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using transports like ISDN or POTS. Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is employed. However dial routes can be used over IP when a single stream on an answering codec answers using POTS and/or ISDN connections, as well as IP. This effectively creates an answering group using different transports. See Configuring ISDN Answering or Configuring POTS Answering for more information.
Line Hunt IP ISDN POTS	Create line hunt groups for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. See Line Hunt Call Answering for more information.

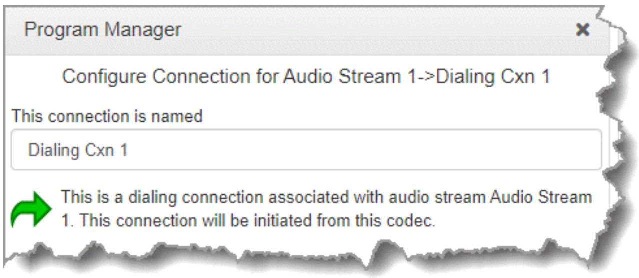
Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

1. **Auto**: The codec will dial the G3 codec and connect in mono or stereo. This is overridden in a Merlin or Merlin PLUS codec when a G3 Main + IFB use-case is configured.
2. **Dual Program**: This allows the codec to dial a G3 codec with a Dual Program profile loaded and support two simultaneous mono connections. Note: When connecting in dual mono (2 x Mono Peer-to-Peer) mode to a G3 codec over IP both audio streams must encode using the same algorithm and sample rate or the G3 codec will not connect.
3. **Runtime**: The G3 codec will retain runtime settings when answering a call from a G5 codec.
4. **Custom**: The G3 codec will load a specified profile, e.g. profile 6, which is the first custom profile number.

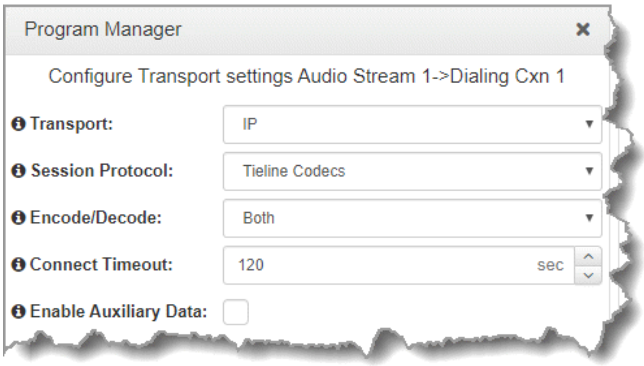
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

1. **Auto** (default): The answering codec will route incoming calls on a first come first served basis.
2. **Channel 1**: The answering codec will always route incoming calls to codec **Channel 1** (left output).
3. **Channel 2**: The answering codec will always route incoming calls to codec **Channel 2** (right output).

5. This audio stream connection in the wizard will allow the codec to dial and connect the first audio stream of the two audio streams being configured. Enter the name of the connection in the text box, then click **Next**.



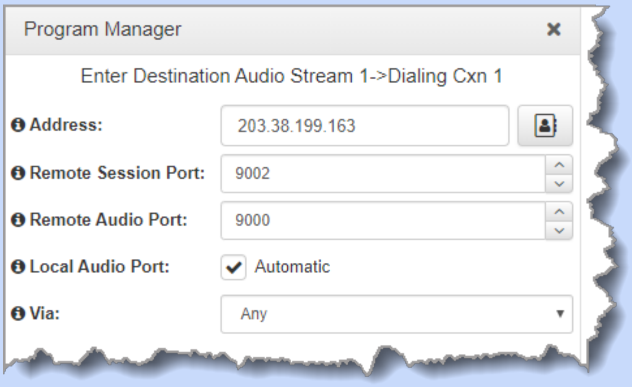
6. Configure the transport settings for the connection, then click **Next**. For SIP connections see Configuring SIP. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm.



Important Note:
- If you select **Sessionless** or **SIP** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
- See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings. Enter an IP address, or click the **Address Book** button to select a TieLink contact.

For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

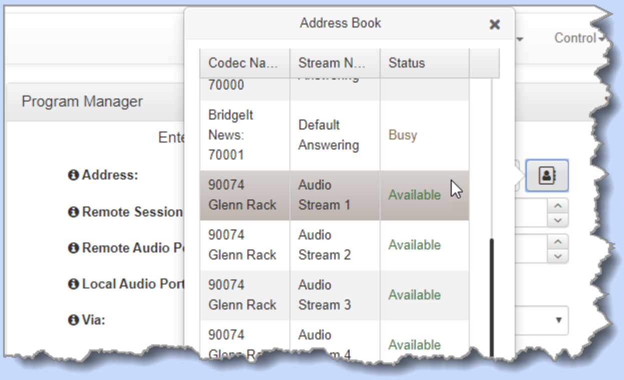


Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), the default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing. Click to deselect the **Automatic** check-box and change this setting. When you select **Sessionless** as the **Session Protocol**, the **Session Port** is not configurable and you can manually configure the **Remote Audio Port** and **Local Audio Port**.

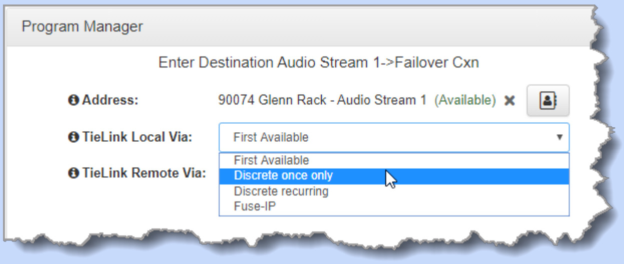
Session Port 9002 and **Remote Audio Port 9000** are used by default for the first IP audio stream. The audio for this audio steam will use analog or digital input and output 1 for both the dialing and answering codecs. If you need to change default port settings on the codec from which the call originates, this will also need to be adjusted in the answering codec using a custom program.

If TieLink Contact List dialing is configured:

1. Click the **Address Book** button to select a contact to dial from the TieLink **Address Book**.

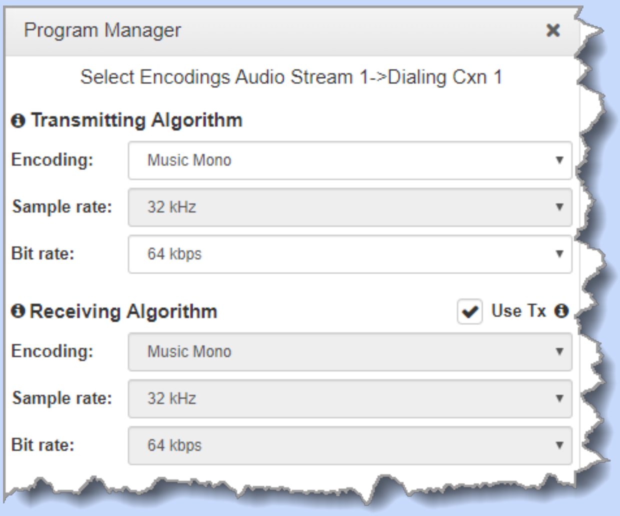


2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see [Configuring TieLink Settings](#) for more info.



Click **Next Stream** to configure the first audio stream with default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a failover connection or SmartStream PLUS for this audio stream (recommended).

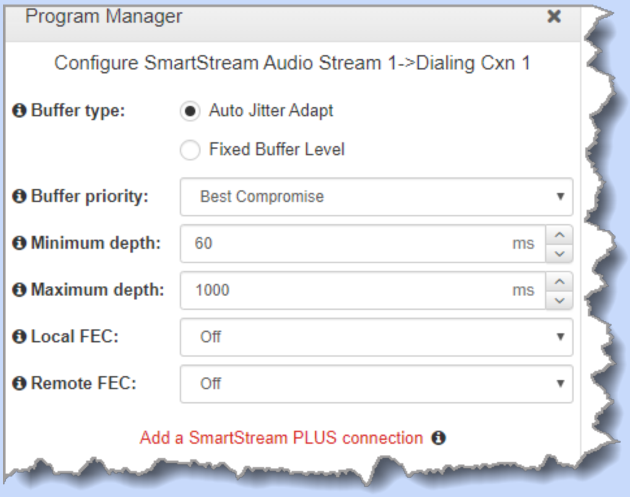
Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** or **Bit rate** options.



For IP connections click to configure:

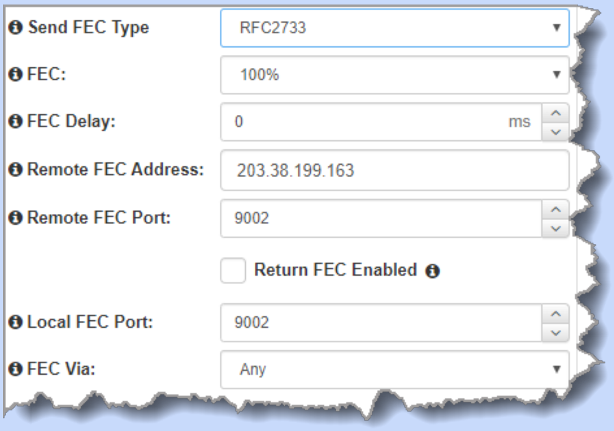
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See [Configuring the Jitter Buffer](#) for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.

- **Local** and **Remote FEC** settings if required.



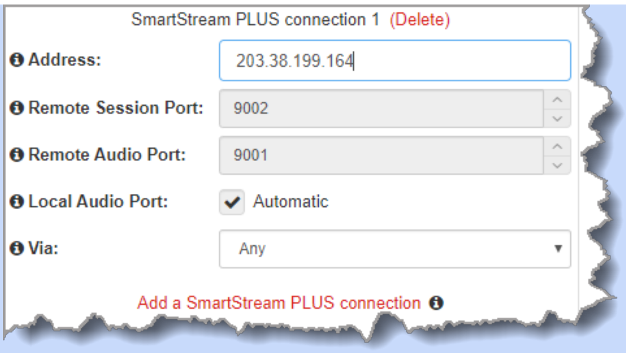
Important Notes:

- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio stream's packets are sent. This will increase the likelihood that the FEC packets will take an alternate route to the primary stream's packets. This means that if primary audio stream packets are not received at the remote codec, there is a good chance that FEC packets taking an alternate route will be received and replace them. When a **FEC** percentage lower than **100%** is configured, FEC packets are automatically delayed based on the ratio of primary packets to FEC packets sent at the selected setting.



Select **Add a SmartStream PLUS Connection** to configure redundant IP streaming. Alternatively, click **Next** to configure **Auto Reconnect** or a failover connection, whereby the alternative connection is dialed if the primary connection fails.

By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio Port** allocated is **Automatic**. Note: **Automatic** indicates that the codec will arbitrarily allocate the local port value and send this information to the codec to which you are dialing.

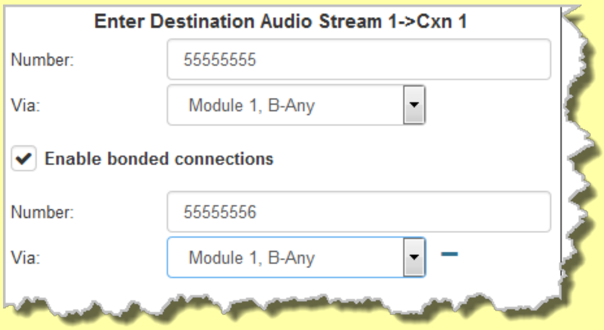


Important Notes:

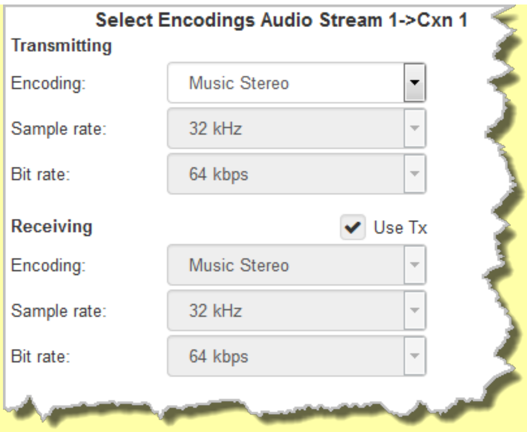
- SmartStream PLUS redundant streaming over multiple IP interfaces mitigates lost packets and provides IP network backup if an IP link is lost.
- Only one SmartStream PLUS connection per audio stream is supported with uncompressed PCM, or when encoding with aptX Enhanced, G.711 or G.722 algorithms.
- Two SmartStream PLUS connections are supported per audio stream with Music PLUS encoding.
- Up to three SmartStream PLUS connections are supported per audio stream when encoding using Tieline Music, AAC algorithms, MP2, MP3 and Opus.
- TieLink only supports one SmartStream PLUS redundant connection for each audio stream. TieLink also does not support uncompressed PCM connections, or encoding using aptX Enhanced, G.711 or G.722.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

ISDN

For ISDN connections enter a number and select which B channel to use. Select the **Enable bonded connections** check-box to configure and bond multiple B channels.



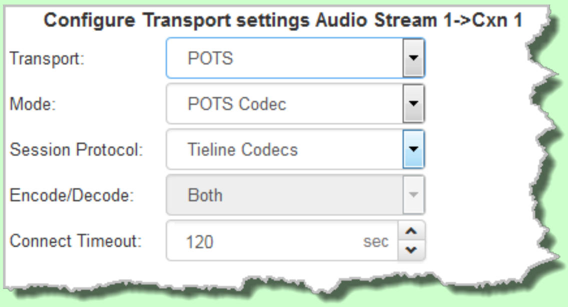
Next, click **Save Program** to save the program with default algorithm settings, or click **Next** to specify a different algorithm and configure a backup connection if required. (recommended).



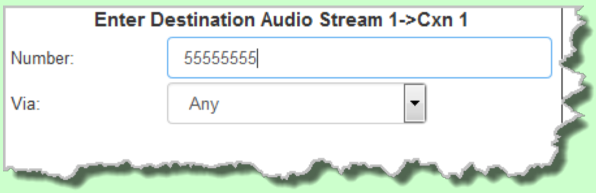
Dialing settings for this ISDN audio stream are now complete.

POTS

Select **POTS Codec** in the **Mode** drop-down menu to encode/decode using POTS, or select **Analog Phone** to configure a standard analog phone call, then click **Next**.



Next, enter the phone number of the codec or device you want to dial. When multiple POTS modules are installed, click the **Via** drop-down menu and select **Module 1** or **Module 2** to specify which POTS module will dial. Next, click **Save Program** to save the program with default settings, or click **Next** to specify algorithm settings and configure a backup connection if required (recommended).



Dialing configuration settings for this POTS audio stream are now complete.

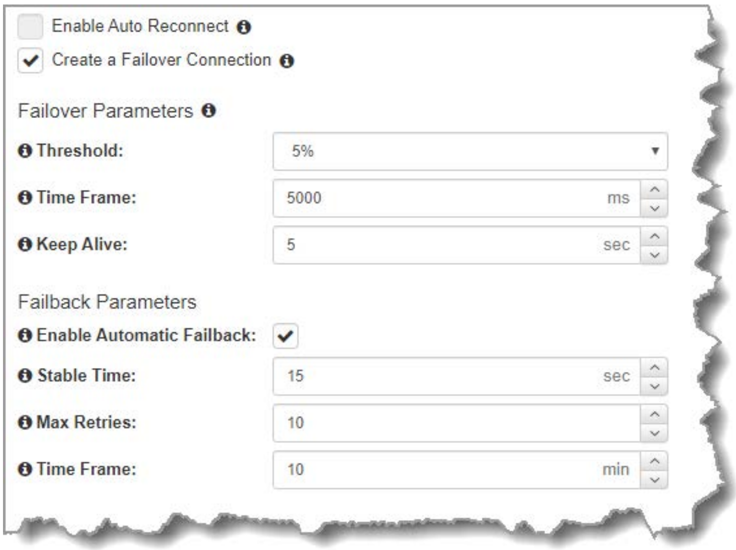
Configuring a Failover Connection or Auto Reconnect

At this point in the wizard you can choose to configure **Auto Reconnect** or create a failover connection for the audio stream you are configuring.

Important Note: When **Auto Reconnect** is enabled, the dialing codec will continue to attempt a connection with the remote codec until **Disconnect** is pressed either on the dialing codec's keypad, or in the Web-GUI.

To configure a backup connection:

1. Click to select the check-box for **Create a Failover Connection**. Adjust the parameters and click **Next**.



Important Note: When Failover is enabled, the codec will fail over to a backup connection under the following circumstances:

- There is sudden loss of connectivity to the primary destination.
- The remote codec disconnects the primary connection.
- The user manually attempts to connect to the failover connection.
- When the conditions for failover are triggered by the **Failover Parameters**.

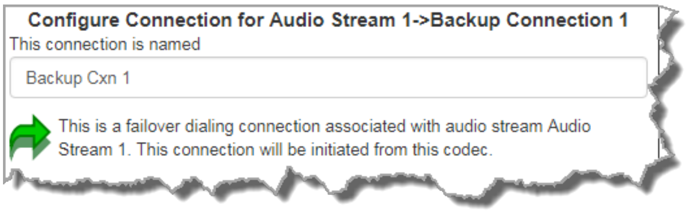
The codec will always attempt Failback if any of the following conditions are met:

- There is sudden loss of connectivity to the backup destination.
- The remote codec disconnects the backup connection.
- The user manually attempts to connect to the primary connection.

The explanations within the following table can be used to assist with failover connection configuration.

	Screen Display	Description
1	Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame against which lost data is measured
3	Keep Alive	The keep connection alive time before failing over to a backup connection; Tieline RTP pings every second to confirm connectivity
4	Enable Automatic Failback	Select the check-box to fail back to a higher priority connection when failback parameters are met
5	Stable Time	The amount of time a primary connection must remain stable before attempting to fail back from the backup connection
6	Maximum Retries	The maximum number of fail back retries a codec can try before ending fail back attempts
7	Time Frame	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the backup connection and click **Next**.

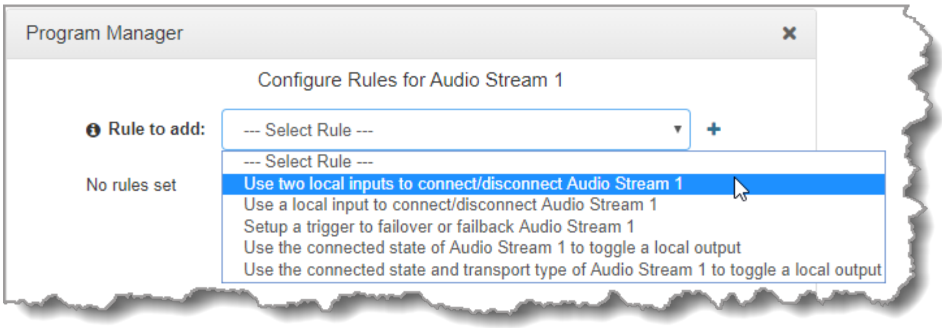


3. Click **Next** to continue through the wizard and configure the backup connection in a similar manner to how you configured the primary connection.

After configuring these settings there are 2 options:

i. Click **Next** to configure rules options.
ii. Click **Next Stream** to configure the second audio stream.

4. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.

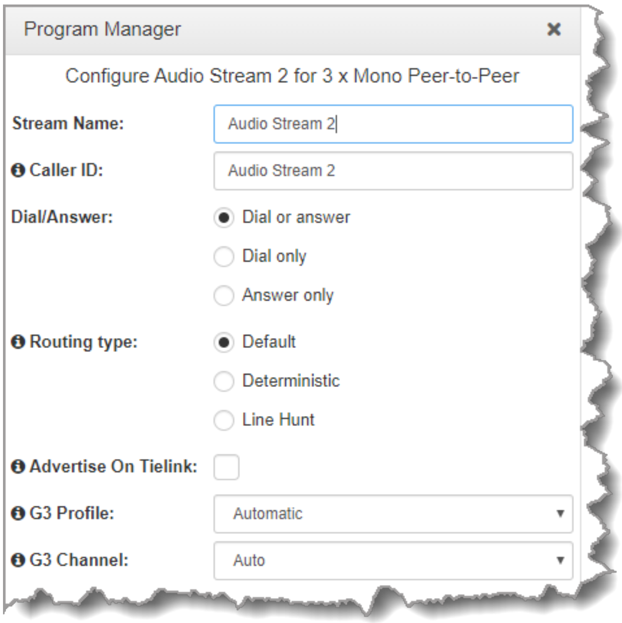


Important Note: Connection-related rules are not displayed in **Answer only** audio streams.

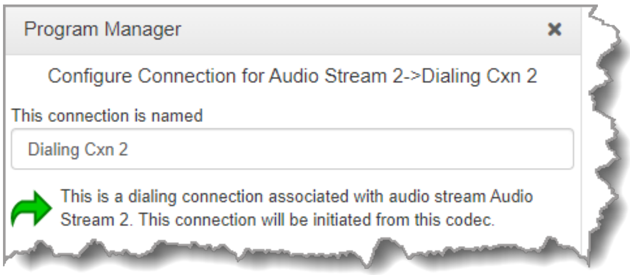
5. Click **Next** to configure the second audio stream.

Configure the Second and Third Peer-to-Peer Audio Streams

1. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**. Select **Advertise On TieLink** if configuring answering audio streams that will be used as dialing destinations in TieLink Traversal Server Contact Lists. For more details see Configuring TieLink Settings.



2. This audio stream connection in the wizard will allow the codec to dial and connect the second audio stream. Enter the name of the connection in the text box, then click **Next**.



3. Continue through the steps in the wizard to complete configuration in the same way as the first peer-to-peer connection was configured. Then configure the third audio stream and click **Save** to save all program settings, then click **Finish** to exit the wizard, or **Load** to load the program.

Important Notes:
- **Session Port 9002** and **Remote Audio Port 9010** are used by default for the second IP connection. **Session Port 9002** and **Remote Audio Port 9020** are used by default for the third IP connection.
- If you connect to a G3 codec using IP2 it is necessary to change the session port on the answering codec to 9012 for the second audio stream connection, as this is used by default on G3 codecs for IP2.

4. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**. [Select and connect audio streams] in a program using the **Connections panel**, or [dial the program manually] using the codec front panel.

33.16 Configuring SIP

The codec is fully EBU N/ACIP Tech 3326 compliant when connecting using SIP (Session Initiation Protocol) to other brands of IP codecs. For more background on SIP connections and the differences between registered and unregistered peer-to-peer SIP connections see About SIP.

To configure the codec to dial over SIP using a SIP Server you will need to:

1. Register the codec to a SIP server using SIP account credentials.
2. Configure a SIP interface in the codec. Note: This **SIP1** or **SIP2** interface will include the proxy and port settings, as well as the selected IP interface used to make the connection, e.g. **LAN1** or **LAN2**.
3. Create a peer-to-peer SIP program using the Dialer wizard using the codec **TOUCH SCREEN**, or create a SIP program using the HTML5 Toolbox Web-GUI.

Important Notes:
- The codec supports dialing over SIP using a registered SIP server account, or peer-to-peer using one of the two SIP interfaces **SIP1** and **SIP 2**.
- SIP dialing is only supported over point-to-point connections.
- Some ISPs and/or cellular networks may block SIP traffic over UDP port 5060.
- Tieline G3 codecs do not support connections using algorithms like AAC, aptX Enhanced and Opus and will default to MPEG Layer 2 if an incoming call is configured to use these algorithms.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- For detailed information about connecting with other brands of codec using SIP visit www.tieline.com.
- When connecting to a Tieline G3 codec using SIP you need to manually select the G3 audio reference level. To do this select **Home screen > Audio** **> General** **> Reference Level > Tieline G3**. In addition, select the following on the G3 codec prior to dialing.
  - Select either a mono or stereo profile
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Session Type] > [SIP]**
  - Select **[Menu] > [Configuration] > [IP1 Setup] > [Algorithm] > [G711/G722 or MP2]**

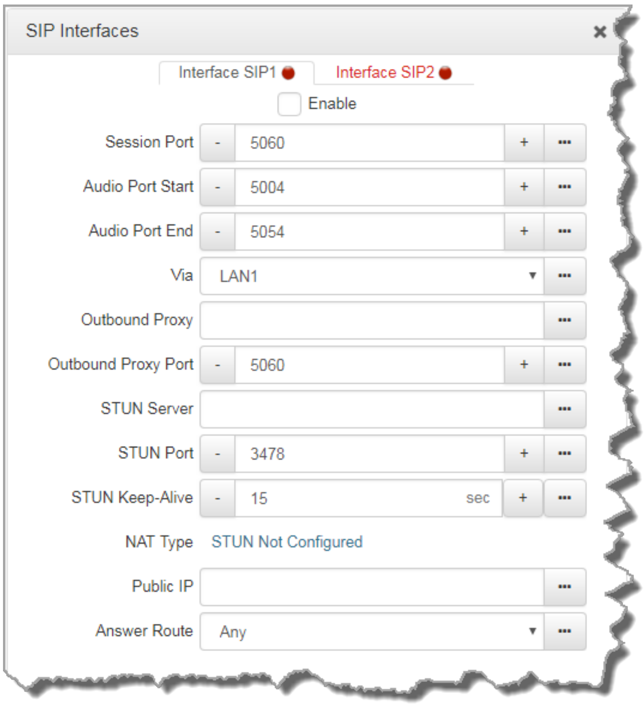
33.16.1 Configuring SIP Interfaces

The codec supports dialing over two SIP interfaces simultaneously.

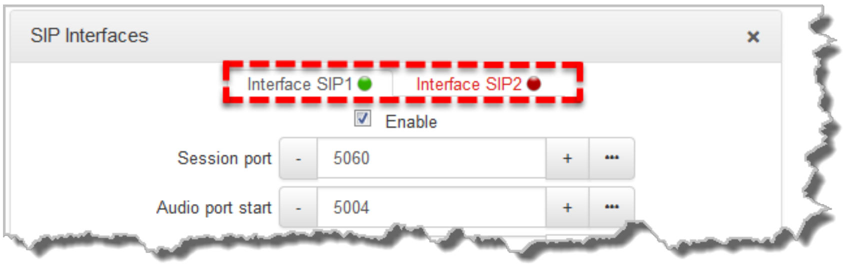
Important Notes:
1. SIP interfaces are disabled by default.
2. **SIP1** is configured to use **LAN1** by default, which is mapped to the **Primary** Via interface by default.
3. **SIP2** is configured to use **Wi-Fi** by default, which is mapped to the **Tertiary** Via interface by default.
4. **SIP1** and **SIP2** each need to use a separate IP interface when connecting, e.g. **LAN1** or **LAN2**.
5. **SIP1** and **SIP2** can however each make multiple SIP calls, e.g. two calls can be made over **SIP1**, or two calls can be made over **SIP2**.
6. The settings for **SIP1** and **SIP2** cannot be edited if the interface has been enabled.
7. Enter a public IP address in the **Public IP** menu if you want to dial over SIP from behind a firewall. Then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They cannot be used together because both will attempt to use a public IP address over SIP. STUN settings are prioritized and used if both are configured.

To configure **SIP1** or **SIP2**:

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Interfaces** to view and configure SIP interface settings.
2. Default SIP settings are configured and select **Interface SIP1** or **Interface SIP2** to adjust each interface. Note: Ensure each interface uses a unique "Via" IP interface because they can't share one, e.g. **LAN1**.



3. Select the **Enable** check-box and then click the **Save** button to confirm settings.
4. The SIP interface indicator is green when an interface is enabled and red when it is disabled.



33.16.2 Configuring SIP Accounts

Up to 6 SIP accounts can be configured in the codec and registering codecs for SIP connectivity is simple. First, choose the SIP server to which you will register your codec. On a LAN this may be your own server, or it could be one of the many internet servers available. We recommend that you use your own SIP server and configure it to use G.711, G.722, MP2 and AAC algorithms. This is because most internet SIP servers are for VoIP phones and are only configured for G.711 and GSM algorithms.

When you register an account with a SIP server you will be provided with:

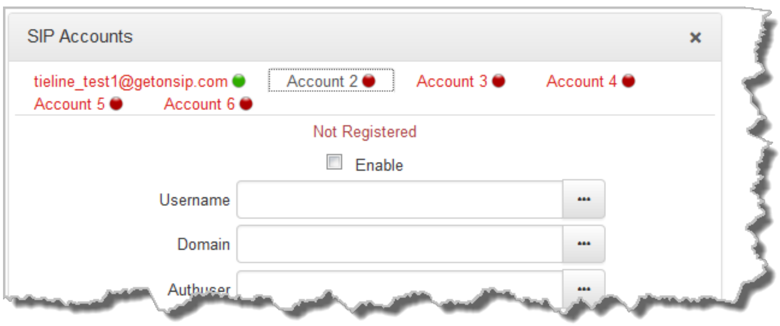
- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port
- Password
- Registration Timeout (this shouldn't need to be adjusted from the default setting).

Important Notes:

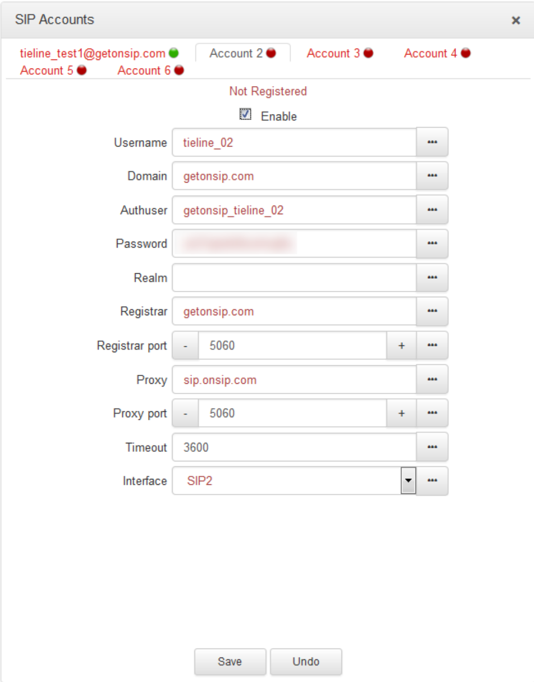
- In most situations it is best to configure a SIP account when the codec is configured with a public IP address.
- Each SIP account can only be mapped to a single SIP interface, i.e. **SIP1** or **SIP 2**.
- Up to 6 SIP accounts can be added to the codec.
- To configure a SIP Account using the codec **TOUCH SCREEN** see Configuring SIP Accounts.

Adding a SIP Account

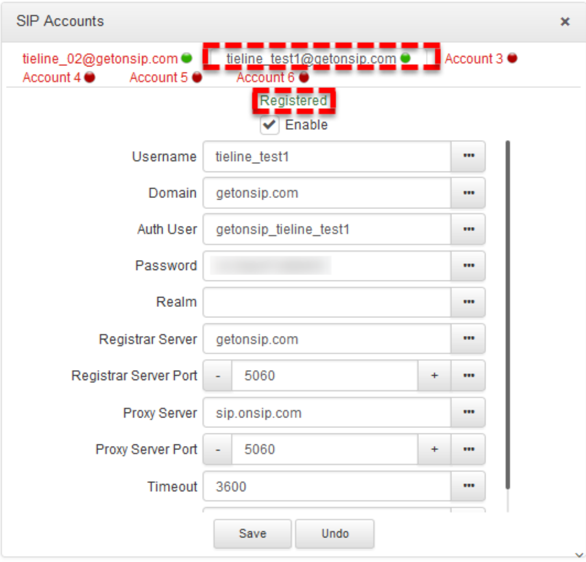
1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **SIP Accounts** to view and configure SIP account settings.
2. Click to select one of the unused **Accounts** at the top of the **SIP Accounts panel**.



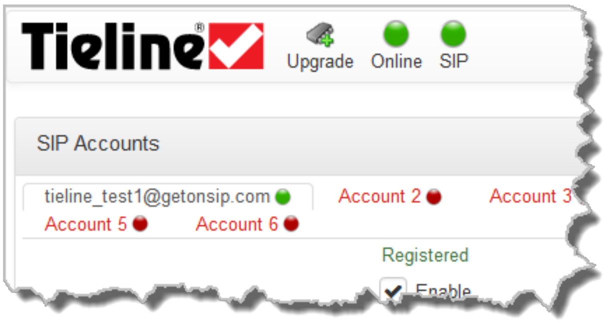
3. Enter the SIP account details into the relevant text boxes, including the registration **Timeout** (which shouldn't need to be adjusted from the default setting). Also ensure a SIP **Interface** is selected (e.g. **SIP1** or **SIP2**.) The SIP interface contains settings related to ports and the selected **Via** interface, e.g. **LAN1** or **LAN2**. See Configuring SIP Interfaces for more details.



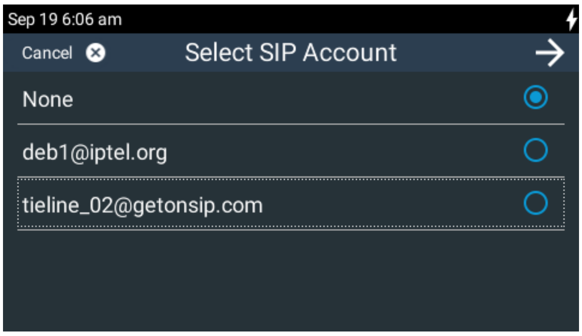
4. Click the **Enable** check-box at the top of the panel and then click the **Save** button to register the codec to the server.
5. If an account is registered successfully, the account registration indicator changes from red to green, and **Not Registered** (above the **Enable** check-box) becomes **Registered**.



6. In the Toolbox Web-GUI the red **SIP** indicator adjacent to the codec **Online** indicator also changes to green when an account is currently registered in the codec and ready to be used when dialing over SIP.



7. Once enabled, the SIP account can be selected when creating a new SIP connection via the **Home screen > Dialer > Destination > Via.**



Important Notes: Some ISPs may block SIP traffic over UDP port 5060.

Troubleshooting SIP Registration

If a SIP account is not being registered please check the following:

1. Confirm all account registration information has been entered correctly.
2. Confirm the SIP interface (**SIP1** or **SIP2**) configured as the **Via** in the account is enabled.
3. Verify that the **Via** selection in the **SIP1** or **SIP2** interface settings corresponds with the network interface being used by the codec to register the account. E.g. **LAN1**, **LAN2**.

33.16.3 Configure SIP White and Blacklists

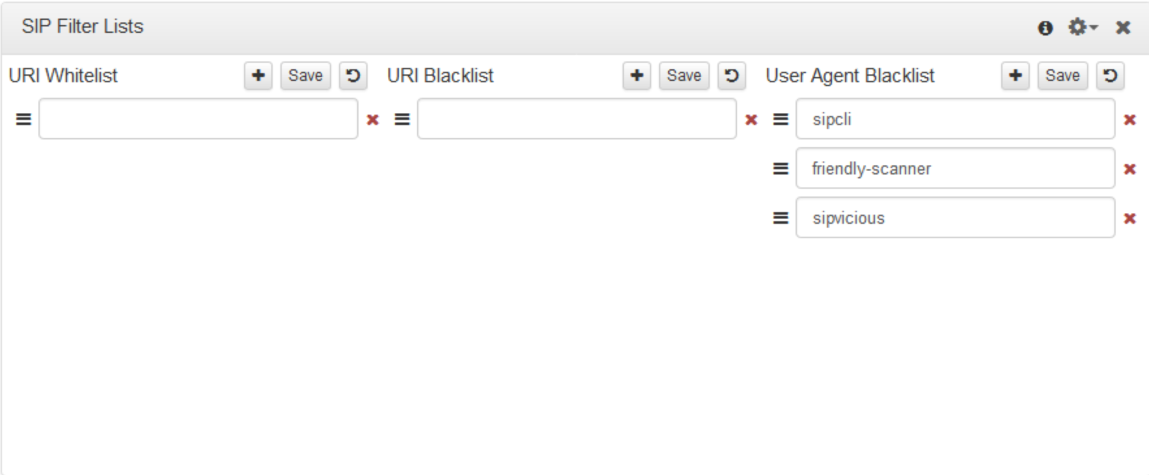
The **SIP Filter Lists panel** allows filtering of SIP URIs and User Agents to provide greater security for your codec connections. For example, add trusted network codecs to the **URI Whitelist** in this panel and only codecs using these SIP URIs will be able to connect. This is like saying, "if you have the key you can open the door" and is perhaps the easiest way to filter outside access to your codec's "front door".

It is also possible to add SIP URIs to the **URI Blacklist** and add user agents to the **User Agent Blacklist** to deny them access to the codec. These blacklists also filter unwanted traffic and increase the likelihood of rejecting unwanted traffic. Note: If an incoming SIP caller is not on the URI Whitelist it will be scanned using the **URI Blacklist**. If there is no match it will be scanned using the **User Agent Blacklist**. A connection will be established if there is no match on either Blacklist.

Important Note: To only allow a predefined list of codecs to connect, add them to the **URI Whitelist** and add a wildcard (asterisk) * to the **URI Blacklist**: all incoming calls will be blocked except for codecs in the Whitelist.

Filter URIs and User Agents

1. Open the HTML5 Toolbox Web-GUI and click **Transport** in the **Menu Bar**, then click **SIP Filter Lists** to launch the **SIP Filter Lists panel**.

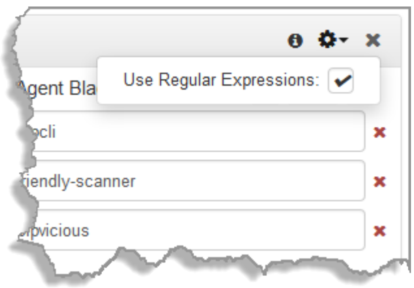


2. Click the **Plus symbol** for **URI Whitelist**, **URI Blacklist** or **User Agent Blacklist** to add a new item to the list.
3. Enter the new item in the text box, click to select the check-box and then click **Save** to store the new setting.
4. Click the **Undo symbol** to undo editing and click and drag the **List symbol** ≡ to shift the position of whitelist and blacklist items.

Important Note: Some codec manufacturers allow whitelisting of calls by 'User Agent.' Therefore, it may be necessary to enter a Tieline codec user agent into a non-Tieline codec to allow it to connect to a Tieline SIP-enabled codec. From firmware v2.16.xx the user agent in the codec is configured as "Tieline <Product Name> <Firmware Version>". E.g. Tieline Bridge-IT v2.18.68

Using Regular Expressions

To filter using regular expressions in the **SIP Filter Lists panel**, click the **Options symbol** in the top right-hand corner of the panel and then click to select the **Use Regular Expressions** check-box.



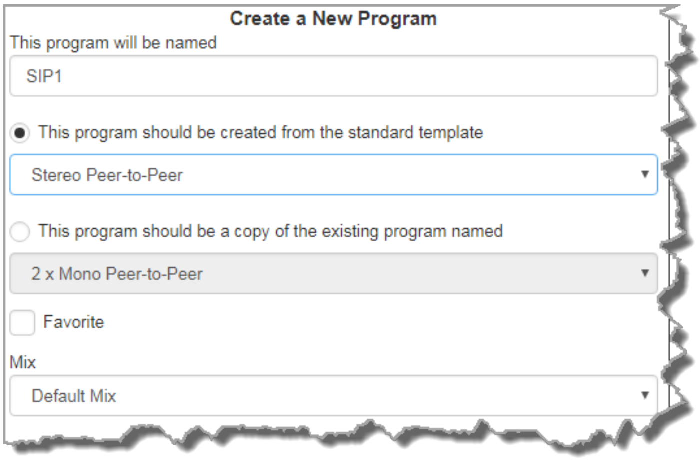
Important Note: Regular expressions should not use ^ and $ anchors because searches implicitly try to match anywhere in the line.

33.17 Configure Peer-to-Peer SIP Programs

SIP programs are like a normal IP program to configure, with two small differences; entering a SIP address and selecting SIP as the **Session Protocol**.

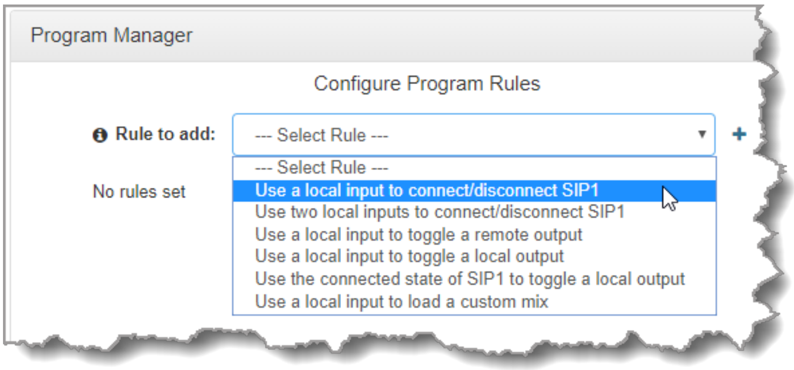
Important Notes: Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- To learn more about programs see the section titled Load, Connect and Manage Programs. Remember to lock an answering program in a codec when answering multiple SIP calls.
- Ensure the appropriate TCP and UDP audio ports are open in your firewall to allow SIP audio streams to connect. See Installing the Codec at the Studio for more information.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the **Program Name** text box to name the new program.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option.
   - Click to select the **Favorite** check-box if you want to add the new program to the list of favorites in the codec. See Load, Connect and Manage Programs for more details.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required, then click **Next**.



Important Notes: When you use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

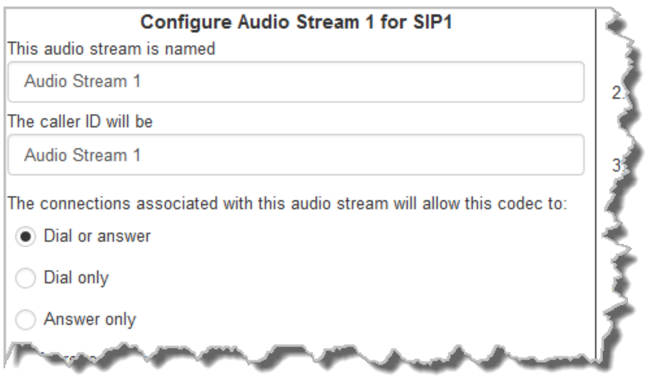
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



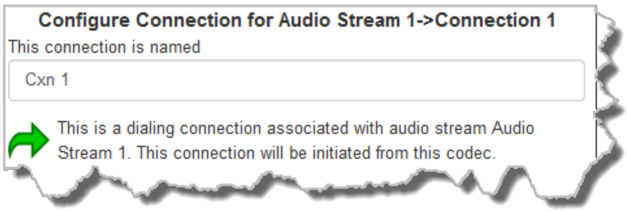
Important Notes for Rules:
- The codec has 4 physical **CONTROL PORT** GPIOs; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Note: The following example will display how to configure a dial and answer program. If you want the codec to either dial or answer only, select the option and the wizard will automatically display screens to allow you to configure the codec correctly. Please note that caller ID, dial routes and G3 profile or G3 channel information can not be used for SIP connections because Tieline session data is replaced by SDP for SIP connections.



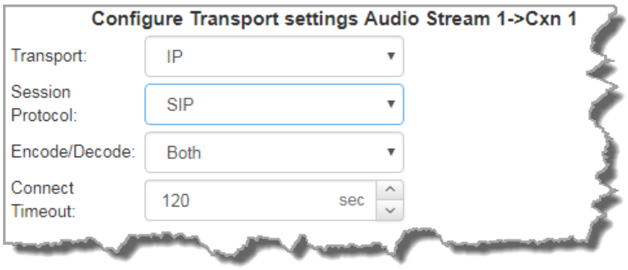
5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



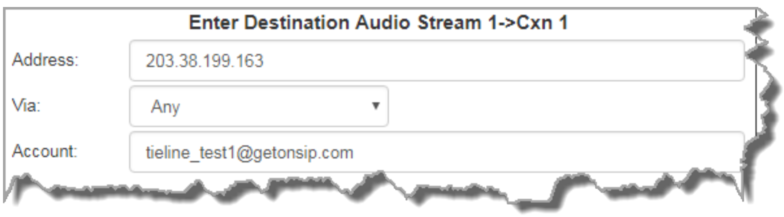
6. Configure the transport settings for the connection: Ensure that you select:

- **IP** as the **Transport**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.



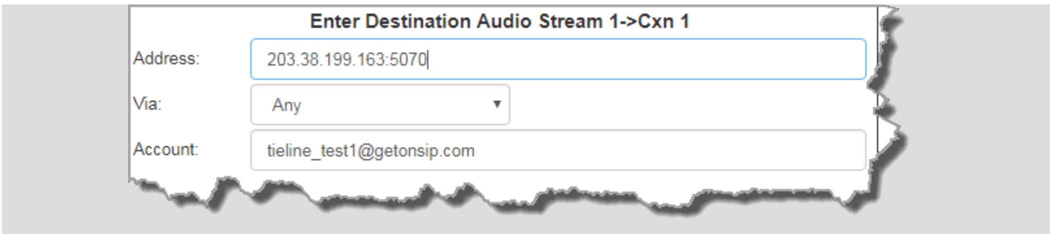
7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** (LAN port 1). Enter the name of a registered SIP account if you are using a SIP server to establish a connection. If you wish to dial from one of the codec's registered accounts, enter the account name in the **Account** field using the format accountname@sipserverdomain, e.g. tieline_test1@getonsip.com. In this configuration the account interface will be used rather than the specified **Via**, e.g. if the account is using **SIP2** and this is configured to use **LAN 2** then the call will proceed using **LAN 2**. If you do not wish to use an account for dialing then leave the **Account** field blank and select the required interface. Note: the interface must be associated with either **SIP1** or **SIP2** for the call to proceed.



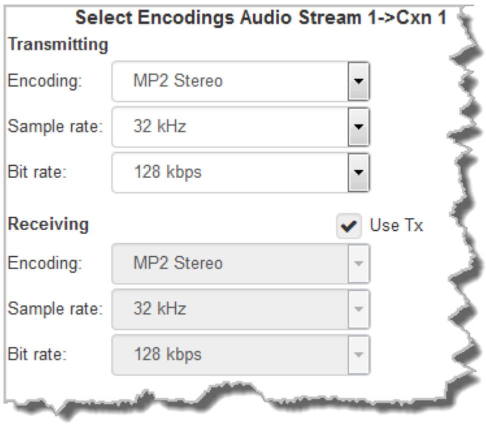
At this point you can click **Save Program** and save the program with default algorithm and jitter settings. Alternatively, click **Next** to specify and confirm algorithm and jitter settings for this connection and configure backup audio settings (recommended).

Important Notes:

- The default UDP audio port when using SIP for a peer-to-peer connection is 5004 in Tieline codecs. To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.
- Tieline codecs automatically add "**sip:**" to the address you enter in the **Address** field when dialing, so it's not necessary to add this.
- Enter the IP address or SIP URI, then a full colon and the session port number to change the session port from the default setting 5060.

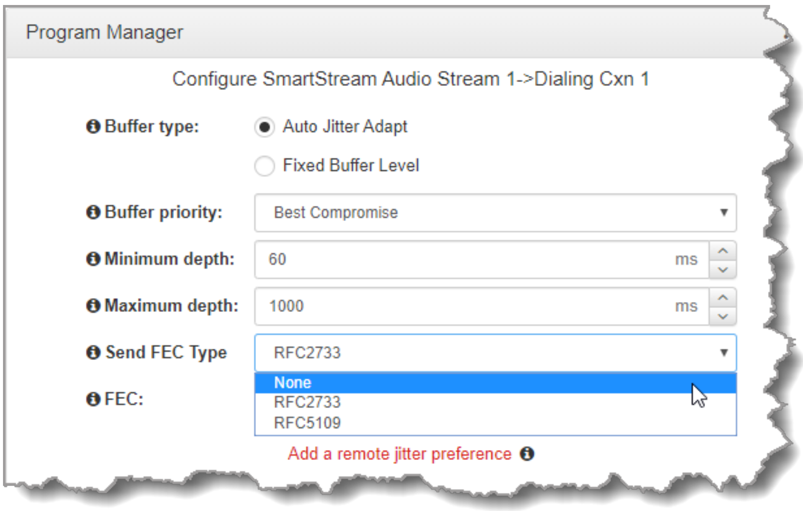


8. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click **Next** to continue.



9. Click to configure:

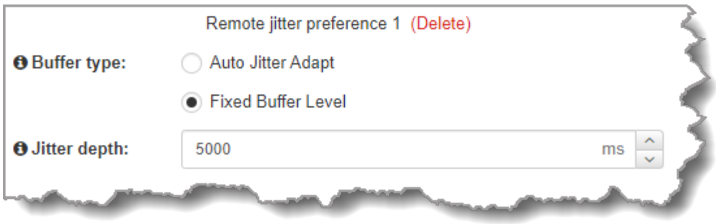
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See [Configuring the Jitter Buffer](#) for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- RFC-compliant FEC can also be configured if required and the percentage is configurable.



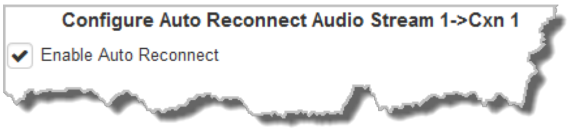
10. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote codec to ensure settings are correct. Recommended jitter buffer limits are as follows:

- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.

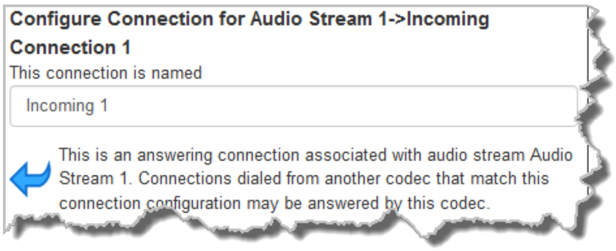
- 5,000 for all other algorithms including Opus, MP2, AAC, AAC-HE, Tieline Music and Music PLUS.



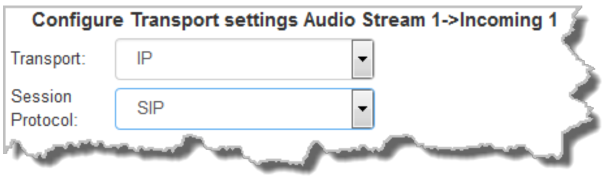
11. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.



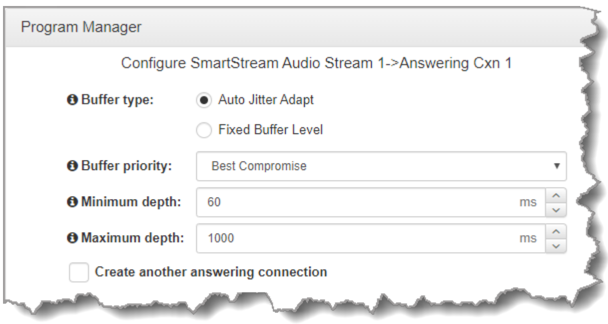
12. Click **Next** to name the answering connection for when calls are received by the codec.



13. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.



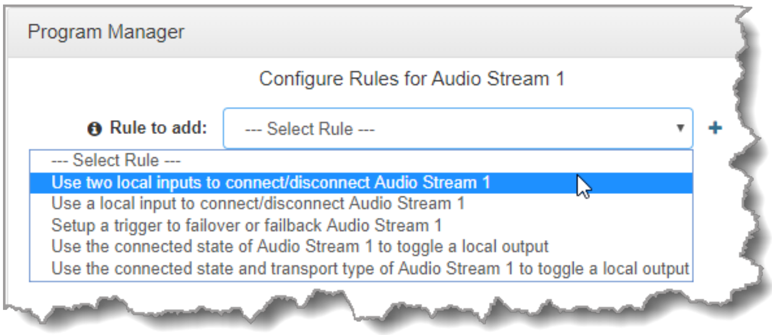
14. Click **Next** to configure the jitter settings for the answering connection.



15. After configuring all settings there are 3 options:

    i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
    ii. Click **Save Program** to save the program at this point.
    iii. Click **Next** to configure rules options.

16. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



Important Notes for Rules:
- Connection-related rules are not displayed in **Answer only** programs.
- Relay reflection is not available for SIP and Multicast Client programs.

17. Click **Save Program** to save the program.
18. Click **Finish** to exit the wizard.
19. The newly created program can be loaded from within the **Program Manager panel**, **Connections panel** and the **Program Loader panel** (in the Quick Connect web-GUI). Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

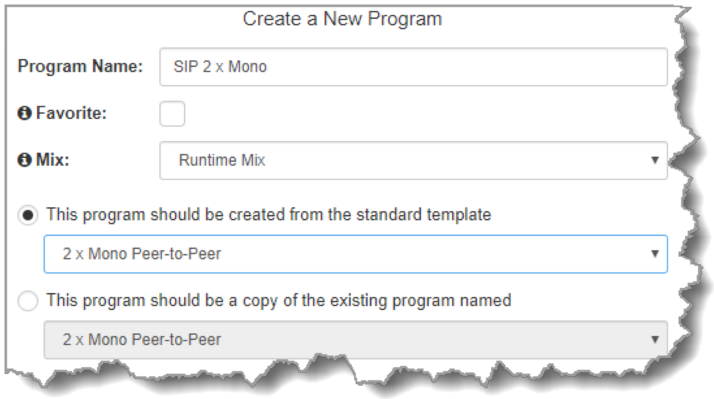
33.18 Configure Multiple Stream SIP Programs

To configure a multiple stream SIP program simply create a new program and configure each SIP audio stream as you would for a single SIP Peer-to-Peer program. In the following example a **2 x Mono Peer-to-Peer** dial and answer program is being configured.

Configuring a 2 or 3 x Mono Peer-to-Peer SIP Program

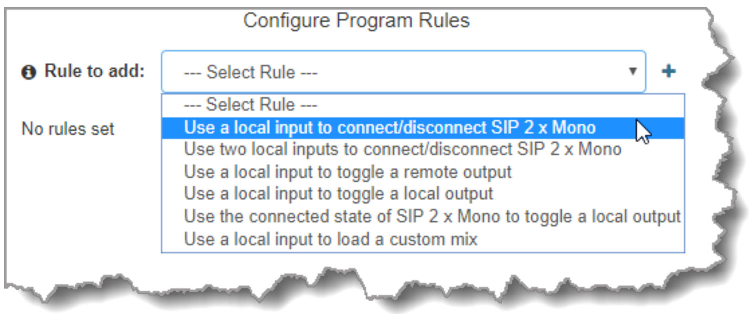
Important Notes: Before you start program configuration please note:
- You cannot edit a program when it is currently loaded in the codec.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections.
- To learn more about programs see the section titled Load, Connect and Manage Program Dialing.Remember to lock an answering program in a codec when answering multiple SIP calls.
- Ensure the appropriate TCP and UDP audio ports are open in your firewall to allow SIP audio streams to connect. See Installing the Codec at the Studio for more information.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Select **2 x Mono Peer-to-Peer**, or 3 **x Mono Peer-to-Peer**, or if you want to use an existing program as a template, select this option.
   - Click to select the **Favorite** check-box if you want to add the new program to the list of favorites in the codec. See Load, Connect and Manage Programs for more details.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required, then click **Next**.



Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

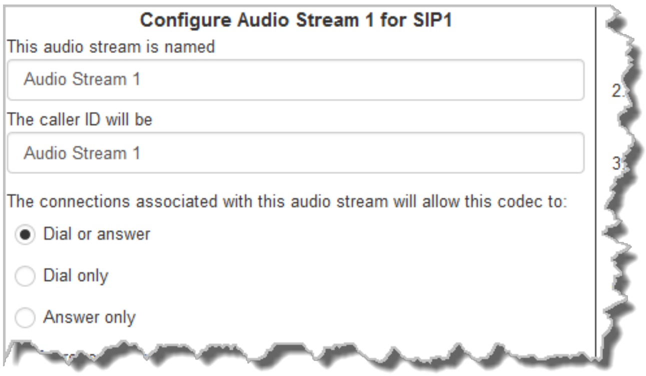
3. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



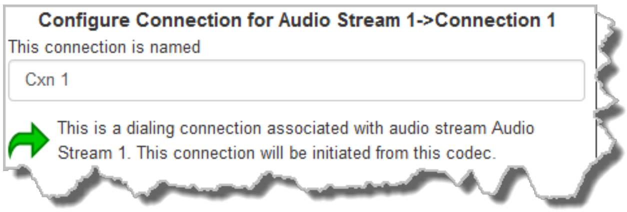
Important Notes for Rules:
- The codec has 4 physical **CONTROL PORT** GPIOs; 7 Tieline and WheatNet virtual inputs (1-7); 64 Tieline virtual logic outputs; and, 64 virtual WheatNet logic outputs (these allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network). See Enabling Relays & RS232 Data for more info.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4. Enter a **Stream Name**, then add a **Caller ID** and configure the codec to dial, answer or dial and answer. Then click **Next**.

Note: The following example will display how to configure a dial and answer program. If you want the codec to either dial or answer only, select the option and the wizard will automatically display screens to allow you to configure the codec correctly. When answering multiple SIP connections, answer routes can be used to create deterministic dialing, which routes audio from incoming calls consistently to the same inputs and outputs. For more information please see Answering Multiple SIP Peer-to-Peer Programs. Please note that caller ID, dial routes and G3 profile or G3 channel information can not be used for SIP connections because Tieline session data is replaced by SDP for SIP connections.



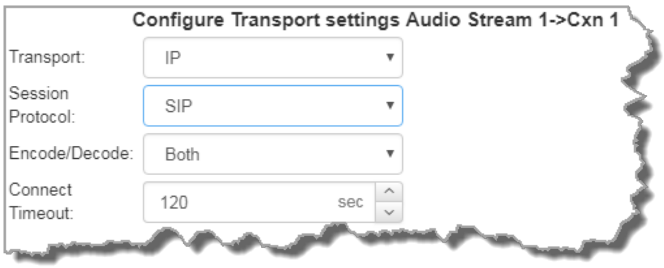
5. This audio stream connection in the wizard will allow the codec to dial. Enter the name of the connection in the text box, then click **Next**.



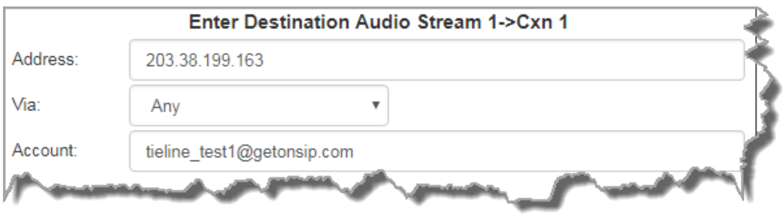
6. Configure the transport settings for the connection: Ensure that you select:

- **IP** as the **Transport**.
- **SIP** from the **Session Protocol** menu option.

Then click **Next**.



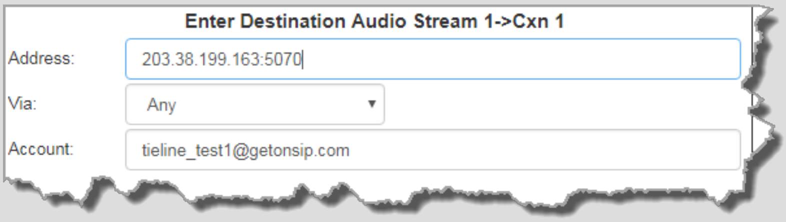
7. Configure the destination codec **Address** if you are dialing peer-to-peer, then specify the network interface used to dial the connection, e.g. **Primary** (LAN port 1). Enter the name of a registered SIP account if you are using a SIP server to establish a connection. Use the format accountname@sipserverdomain, e.g. tieline_test1@getonsip.com



At this point you can click **Save Program** and save the program with default algorithm and jitter settings. Alternatively, click **Next** to confirm and specify algorithm and jitter settings for this connection and configure backup audio settings (recommended).

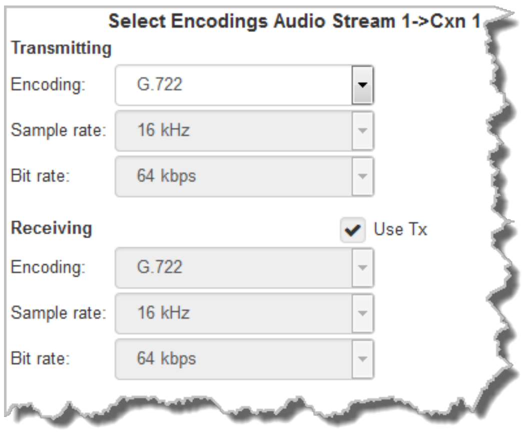
Important Notes:

- Enter the IP address or SIP URI, then a full colon and the session port number to change the session port from the default setting 5060.



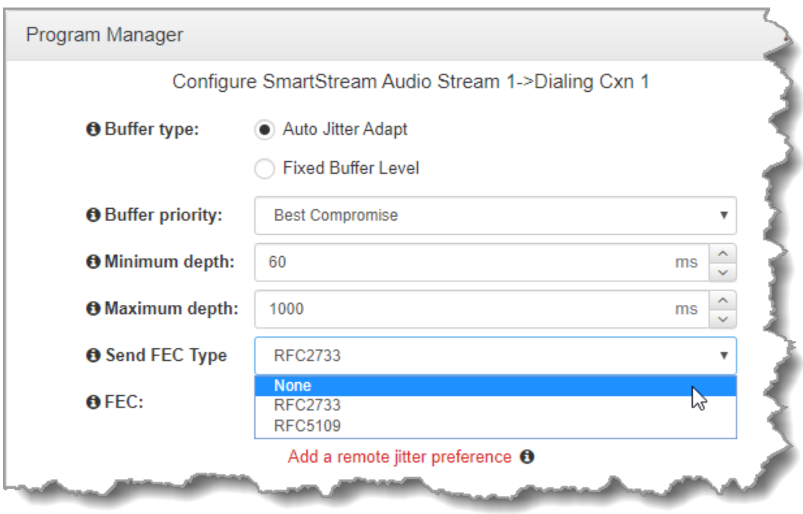
- The audio port used by the codecs is allocated automatically from the 'pool' of ports specified in the **SIP Interfaces** panel. The default setting is 5004 to 5054.
- To contact a codec that is behind a firewall or NAT-enabled router, it is essential that this and all other relevant ports are open and forwarded to the other device.
- Tieline codecs automatically add "**sip:**" to the address you enter in the **Address** field when dialing, so it's not necessary to add this.

8. Click the drop-down arrows on the right-hand side of each active drop-down menu to adjust the **Encoding, Sample rate** or **Bit rate** parameters. Click **Next** to continue.



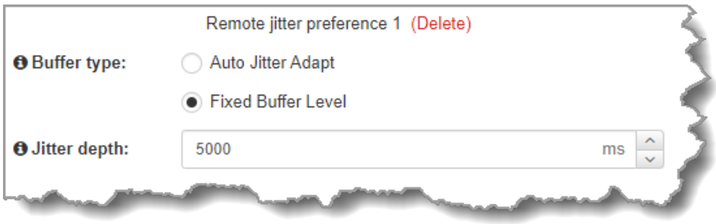
9. Click to configure:

- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.
- RFC-compliant FEC can also be configured if required and the percentage is configurable.

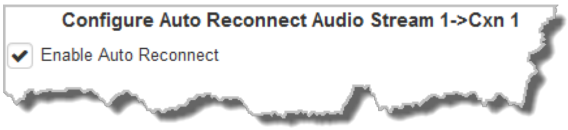


10. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote codec to ensure settings are correct. Fixed jitter limits are as follows:

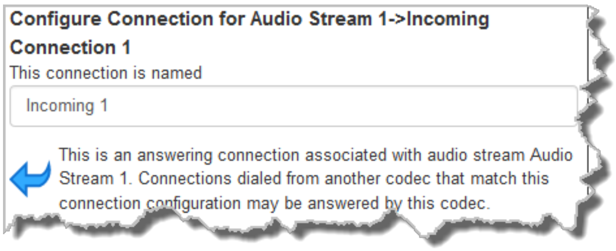
- 1,000ms for PCM and G.711, G.722 and aptX Enhanced encoding.
- 2,500ms for AAC ELD, AAC LD.
- 5,000 for all other algorithms including Opus, MP2, Tieline Music and Music PLUS.



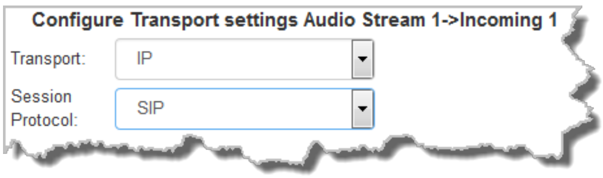
11. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.



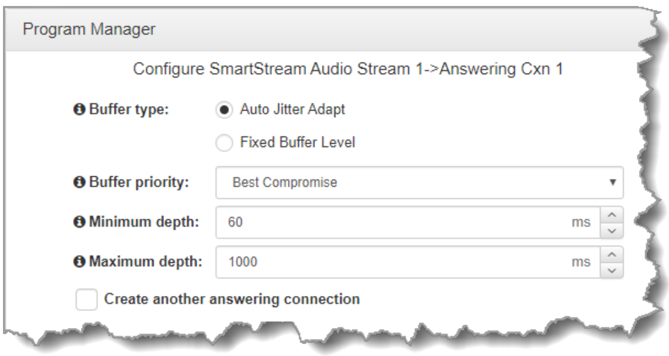
12. Click **Next** to name the answering connection for when calls are received by the codec.



13. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.



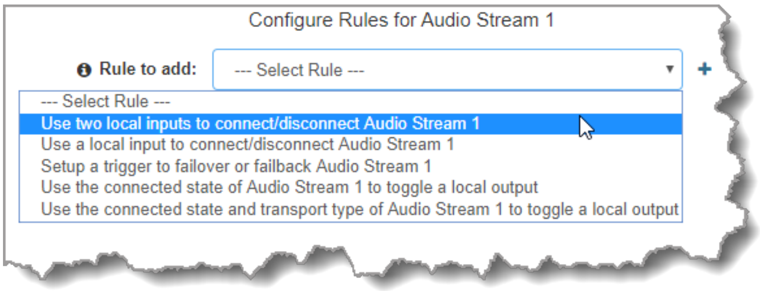
14. Click **Next** to configure the jitter settings for the answering connection.



15. After configuring all settings there are 3 options:

i. If you want to create another answering connection, select the check-box for **Create another answering connection** and continue through the wizard.
ii. Click **Next Stream** to configure the next audio stream.
iii. Click **Next** to configure rules options.

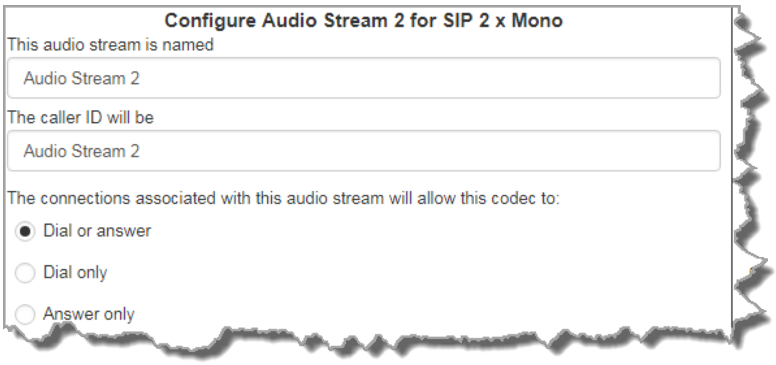
16. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



Important Notes for Rules:
- Connection-related rules are not displayed in **Answer only** programs.
- Relay reflection is not available for SIP and Multicast Client programs.

Configure the Second Peer-to-Peer SIP Audio Stream

1. Click **Next Stream** to name the second **Audio Stream** and configure the codec to dial, answer or dial and answer. Then click **Next**.



Note: When answering multiple SIP connections, answer routes can be used to create deterministic dialing, which routes audio from incoming calls consistently to the same inputs and outputs. For more information please see Answering Multiple SIP Peer-to-Peer Programs. Please note that caller ID, dial routes and G3 Profile and G3 Channel information can not be used for SIP connections because Tieline session data is replaced by SDP for SIP connections.

2. Continue through the wizard and configure additional dial and answer stream connections in a similar way to the first audio stream. Then click **Save Program** to save all settings and click **Finish** to exit the Program Manager wizard. The newly created program can be loaded from within the **Program Manager panel** and **Connections panel**. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

Answering Multiple SIP Calls

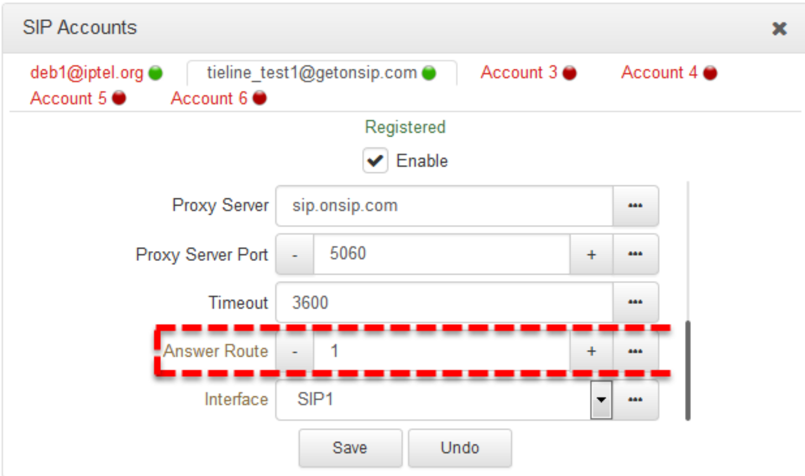
To answer multiple SIP calls you need to create and lock a suitable multiple stream answering program in the codec, or it will be unloaded by the first SIP call and a default peer-to-peer program will be loaded.

33.19 Answering Multiple SIP Peer-to-Peer Calls

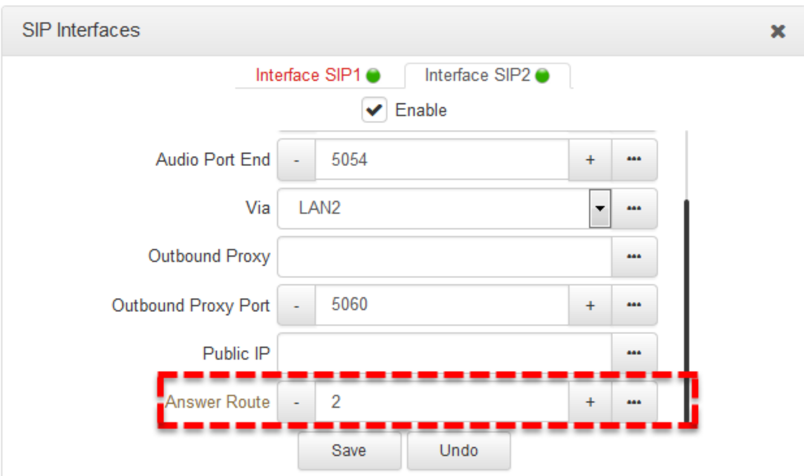
The codec is capable of accepting multiple SIP Peer-to-Peer connections and routing them in a deterministic manner to specific audio input and outputs.

Tieline session data, in conjunction with unique audio ports, normally facilitates routing of multiple incoming streams to specific inputs and outputs on a single answering codec. Session Description Protocol (SDP) is used over SIP instead of Tieline session data, therefore deterministic routing of incoming calls needs to be done a bit differently. There are two options:

1. **SIP Accounts**: Using the **SIP Accounts panel** in the HTML5 Toolbox web-GUI, register up to 6 SIP accounts and configure a unique answer **Route** in each SIP account. Then create a multiple stream answering program (e.g. **2 x Mono Peer-to-Peer**) and configure an answer route in each audio stream matching the routes used for each SIP account you have registered to the codec. In this way, when a call is received via a particular account, it is routed to the audio stream with the matching audio answer route. This process also reliably predetermines which inputs and outputs are used on the answering codec (e.g. **Input 1** and **Output 1** for the first audio stream in a multiple stream program).



2. **SIP Interfaces**: When answering multiple incoming peer-to-peer SIP calls, configure an answer **Route** for incoming peer-to-peer calls via the **SIP Interface panel** and route them to specific audio streams. Configure answer routes in **Interface SIP1** and **Interface SIP2** to match answer routes configured in an answering program's audio streams. This process also reliably predetermines which inputs and outputs are used on the answering codec (e.g. **Input 2** and **Output 2** for the second audio stream in a multiple stream program).



Important Notes:

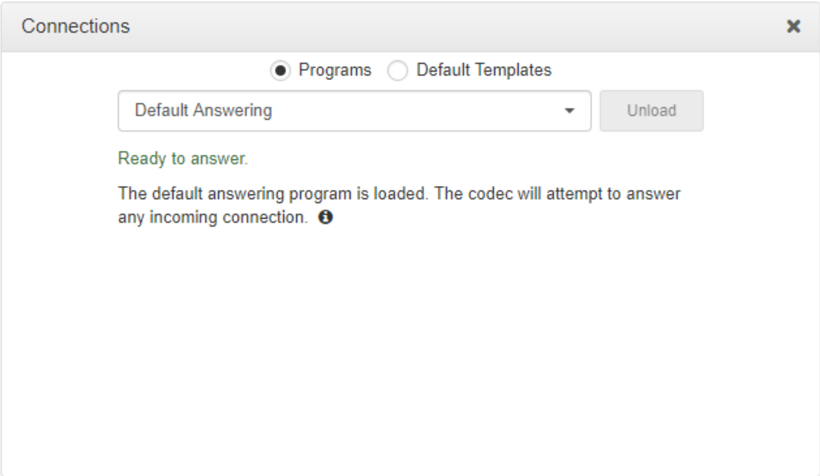
- Ports are arbitrarily assigned when answering calls using SIP accounts. By default the UDP audio port range used is from 5004 to 5054. This setting can be adjusted in the **SIP Interfaces panel** in the HTML5 Toolbox web-GUI. Ensure your firewall has the required TCP and UDP ports open if you are receiving multiple SIP calls.
- Remember to lock an answering program in a codec when answering multiple SIP calls.
- Ensure the appropriate TCP and UDP audio ports are open in your firewall to allow multiple SIP audio streams to connect. See Installing the Codec at the Studio for more information.
- Failover and SmartStream PLUS redundant streaming are not available with SIP connections

33.20 Load, Unload and Dial a Program

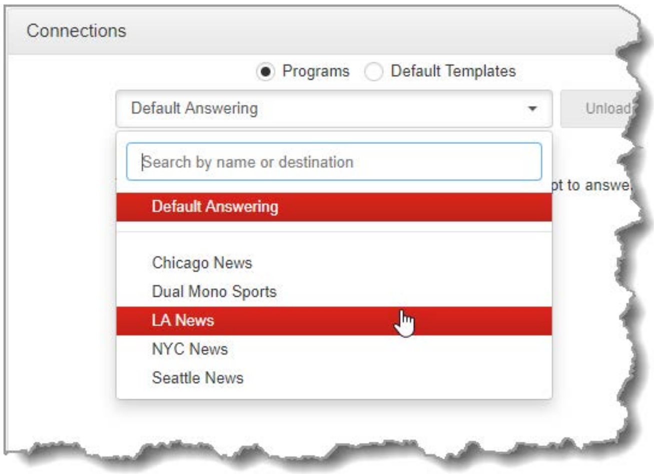
A program can be loaded, unloaded and edited using the **Program Manager panel** or the **Connections panel**. Audio stream dialing settings can be edited without unloading a program, even if other audio streams are concurrently connected. Please note: If you need to use an existing program as a template for a new program, or add a custom matrix mix to a program, the **Program Manager panel** must be used to create a program.

Load and Unload a Program

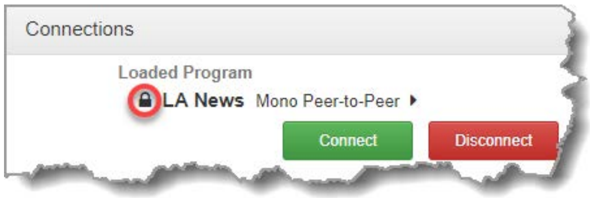
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



2. Select **Programs** and then click the drop-down arrow to select one of the programs in the codec, then click **Load**.



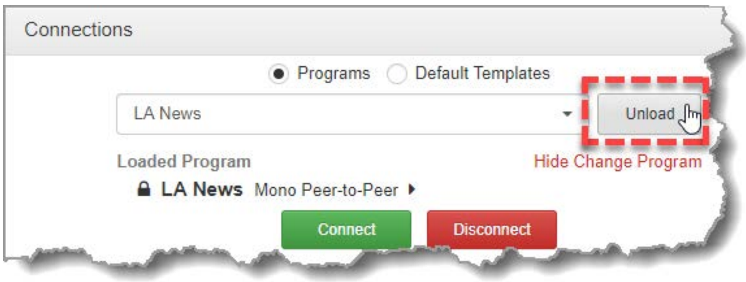
3. If **Lock Loaded User Program** has been configured in the **Options panel**, a black **Padlock** symbol appears next to the program name in the **Connections panel**, to indicate a program is locked in the codec.



Unload a Program

1. To unload a loaded program click **Change Program**.

2. Click **Unload**.

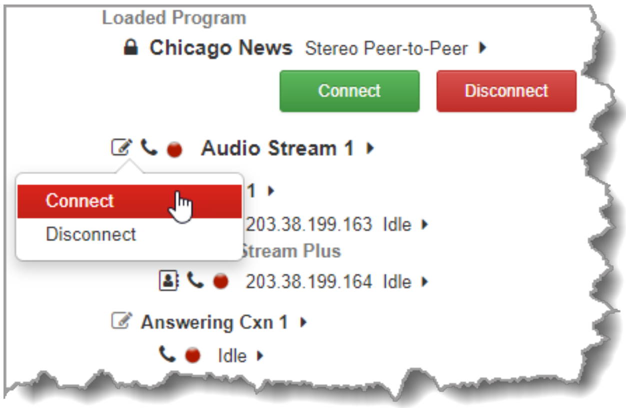


Connecting a Program

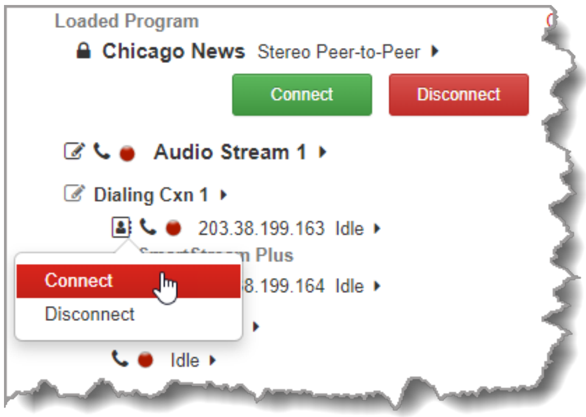
To connect audio streams and connections within an existing program there are three options:

1. Click the **Connect** 

 button to connect all audio streams and connections configured in a program.
2. Click the audio stream **Connect/Disconnect** symbol and then click **Connect**; this connects all connections associated with this audio stream.



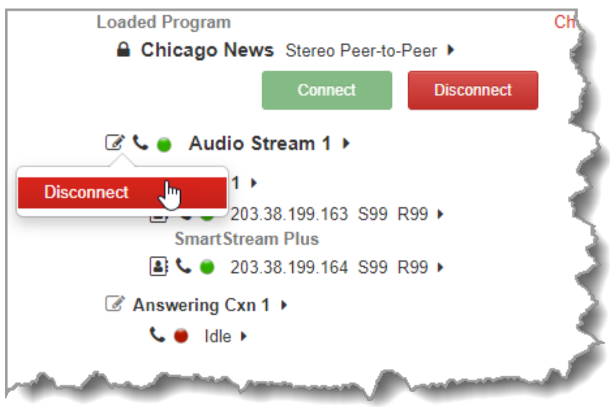
3. Click the connection **Connect/Disconnect** symbol and then click **Connect**; this connects an individual audio stream connection.



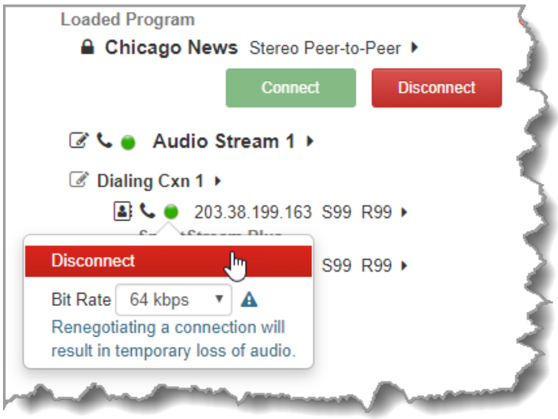
Disconnecting a Program

To disconnect audio streams and connections within an existing program there are three options:

1. Click the **Disconnect** Disconnect button to disconnect all audio streams and connections configured in a program.
2. Click the audio stream **Connect/Disconnect** symbol and then click **Disconnect** to disconnect an individual audio stream and all associated connections.



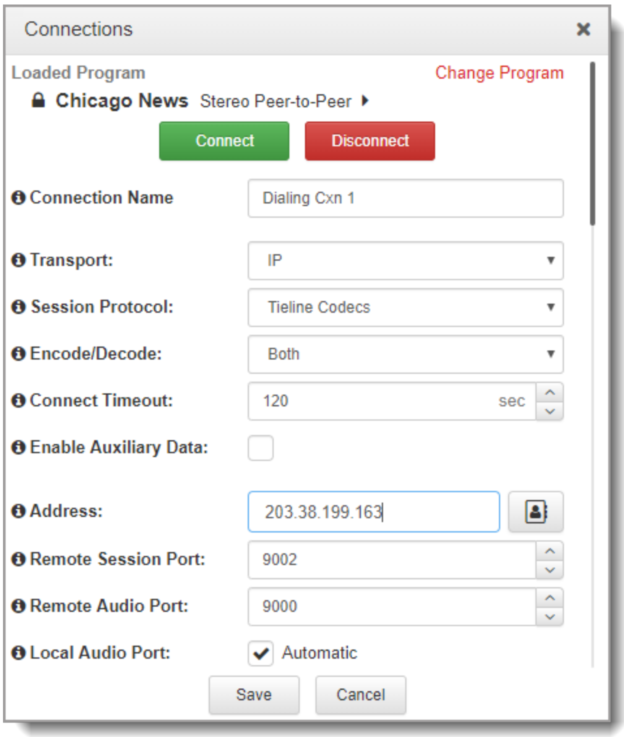
3. Click the connection **Connect/Disconnect** symbol to disconnect an individual audio stream connection.



Change Dialing Settings

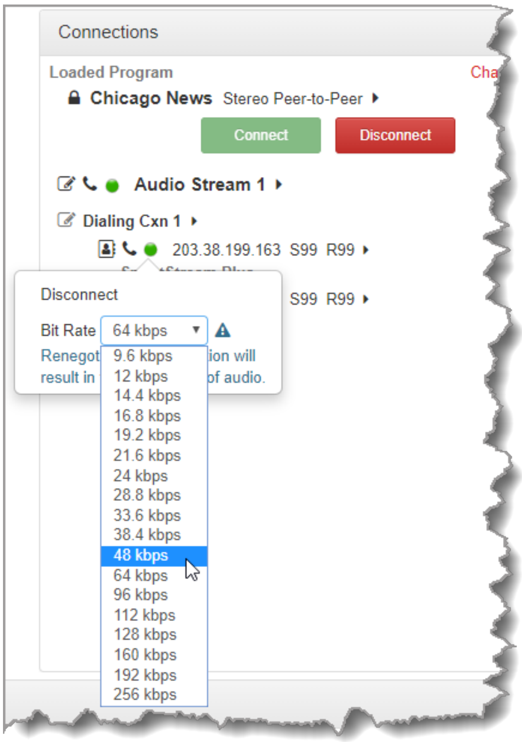
To edit destination dialing settings:

1. Click the **Edit symbol** adjacent to a connection.
2. Adjust dialing and connection settings and then click **Save** to change edited settings in the program. Note: The IP address can be changed, or a **TieLink contact** can be selected if Traversal Server Contact Lists have been configured. See Configuring TieLink Settings for more info.



33.21 Adjusting the Connection Bit Rate

1. Open the HTML5 Toolbox Web-GUI and click **Connect**, then select **Connections** to open the **Connections panel**.
2. Click **Connect/Disconnect** symbol for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit rate.



Important Notes:
- Renegotiation will result in a temporary loss of audio.
- It is not possible to renegotiate the connection bit rate of a SIP connection.

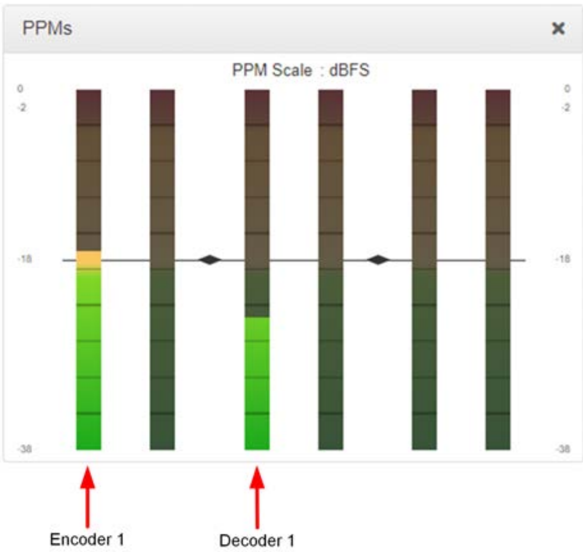
33.22 Monitoring Program PPMs

Important Note: Codec **LED PPMs** always reflect the **following:**

- Front panel LED **PPM L** displays the Encoder 1 mix.
- Front panel LED **PPM R** displays the Encoder 2 mix.
- Front panel LED **PPM RTN** displays a mix of Decoders 1 & 2.

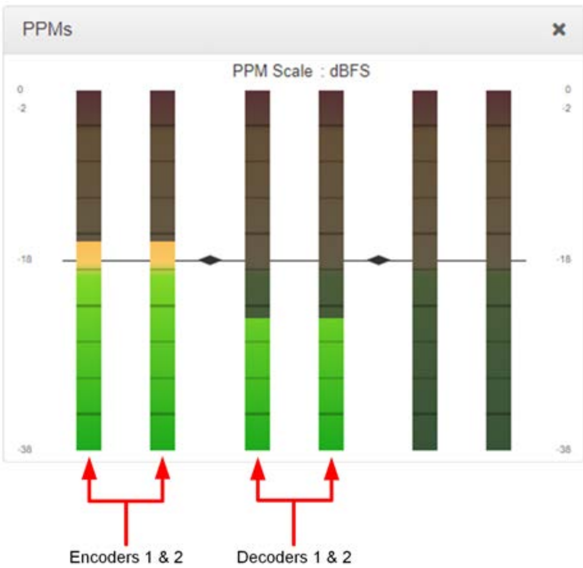
Mono Connections

When connected with a mono program the codec displays outgoing program audio via encoder 1 on **PPM1**. **PPM3** displays return audio via decoder 1.



Stereo Connections

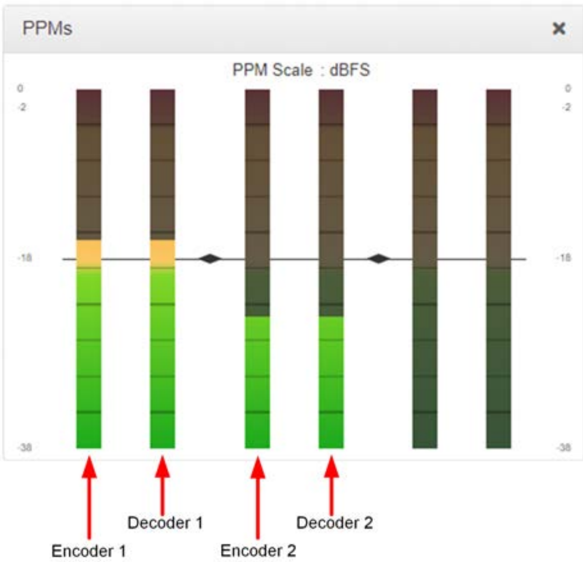
When connecting with a stereo program the codec displays outgoing program audio via encoders 1 and 2 on **PPM1** and **PPM2**. **PPM3** and **PPM4** display return audio via decoders 1 and 2.



2 x Mono Peer-to-Peer Connection Metering

The codec is capable of creating two independent mono audio stream connections simultaneously. In this situation:

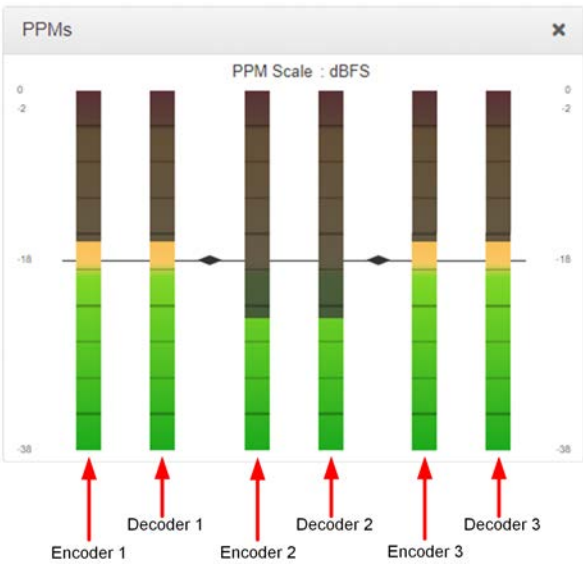
1. The codec displays outgoing connection 1 audio via encoder 1 on **PPM1** and **PPM2** displays return audio via decoder 1.
2. Outgoing connection 2 audio via encoder 2 is displayed on **PPM3** and **PPM4** displays return audio via decoder 2.



3 x Mono Peer-to-Peer Connection Metering

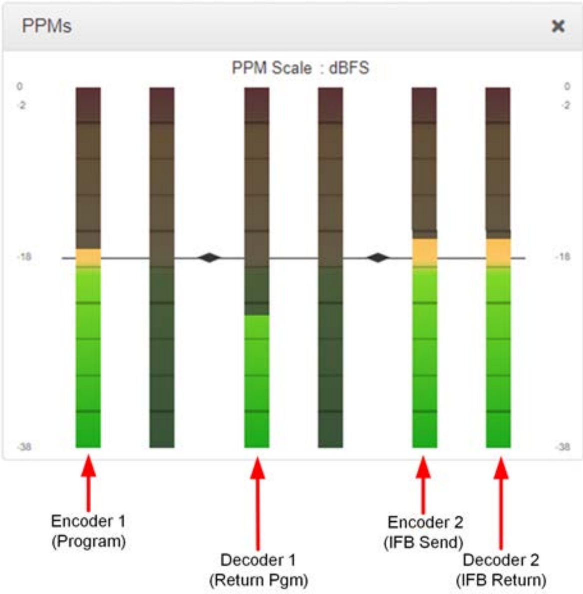
The codec is capable of creating three independent mono audio stream connections simultaneously. In this situation:

1. The codec displays outgoing connection 1 audio via encoder 1 on **PPM1** and incoming connection 1 return audio via decoder 1 on **PPM 2**.
2. Outgoing connection 2 audio via encoder 2 is displayed on **PPM3** and **PPM 4** displays incoming connection 2 return audio via decoder 2.
3. Outgoing connection 3 audio via encoder 3 is displayed on **PPM5** and **PPM 6** displays incoming connection 3 return audio via decoder 3.



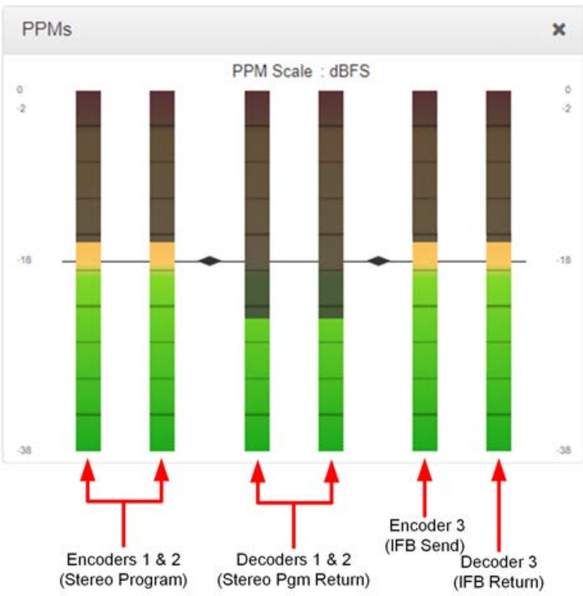
1 x Mono Peer-to-Peer + IFB Metering

This program transmits a bidirectional mono audio stream and a separate bidirectional mono IFB communications audio stream. Outgoing program via encoder 1 is displayed on **PPM1**. **PPM 3** displays incoming return program audio via decoder 1. **PPM5** displays outgoing IFB audio via encoder 2 and **PPM6** displays incoming IFB audio via decoder 2.



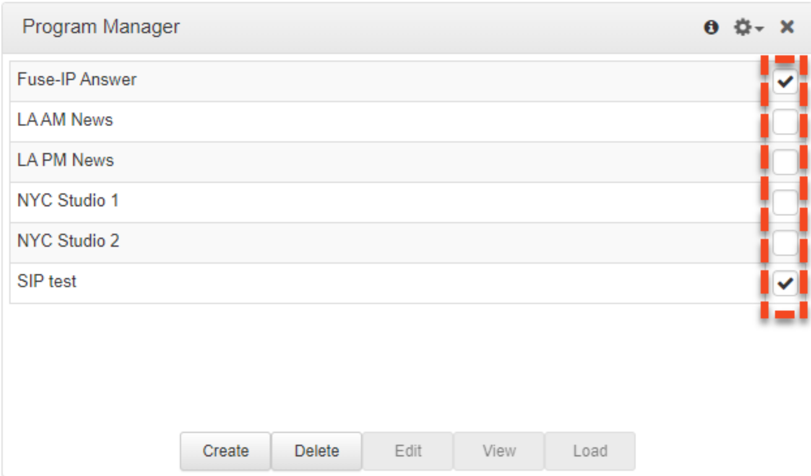
1 x Stereo Peer-to-Peer + IFB Metering

This program transmits a bidirectional stereo audio stream and a separate bidirectional mono IFB communications audio stream. Outgoing program via encoders 1 and 2 is displayed on **PPM1** & **PPM2**. **PPM 3** & **PPM4** display incoming return program audio via decoders 1 and 2. **PPM5** displays outgoing IFB audio via encoder 3 and **PPM6** displays incoming IFB audio via decoder 3.

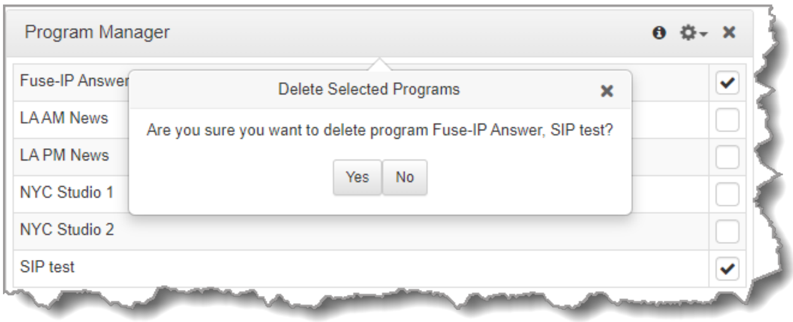


33.23 Delete a Program

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Program Manager** to open the **Program Manager panel**.
2. Click to select the check-box for each program to be deleted. Note: multiple programs can be selected and deleted simultaneously.



3. Click the **Delete** button.
4. Click **Yes** in the **Delete Selected Programs** confirmation dialog to delete all selected programs.



Important Notes: Any program that is currently loaded or listed in the **Scheduler Events panel** cannot be deleted.

33.24 Matrix, Cue and Talkback Editing

The matrix editor in the codec allows any input to be routed to any output. Default routing settings are configured for each program type and these default matrices can be edited, saved and recalled as required. All saved custom matrices are available if a compatible program is loaded. If a matrix is not compatible with a program type it will not be visible in the menu, e.g. a saved stereo matrix is not visible when a mono program is loaded.

Custom matrices can be created, saved and then backed up with program and scheduler data. This allows them to be copied between codecs by using the Backup and Restore feature.

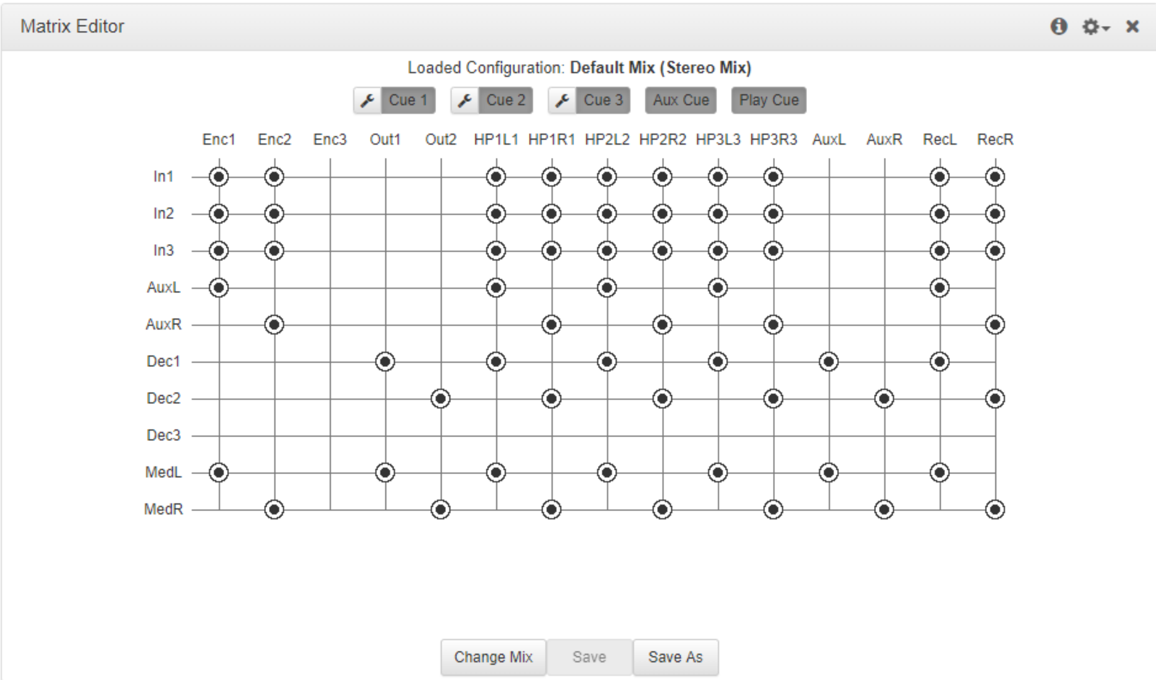
Important Notes:

- If you create custom headphone mixes and load them, and then save all routing in the **Matrix Editor** as a custom matrix, all the custom headphone mixes and their **Send/Return** balance settings are also saved. In essence, custom headphone mixes are "attached" to a **Matrix Editor** matrix when it is saved.

- Only crosspoint routing settings for XLR and Digital Outputs are stored in a custom matrix. The send/return balance, ganging, output mute and output level settings are not saved.

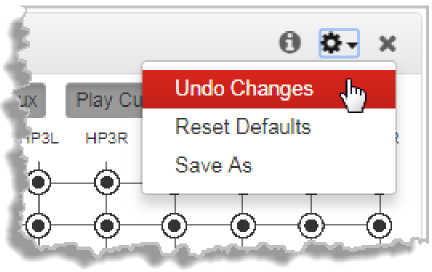
Viewing the Matrix Editor

Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Matrix Editor** to open the **Matrix Editor panel**. In the following image the default stereo matrix is loaded, as indicated by the **Loaded Configuration** at the top of the panel.



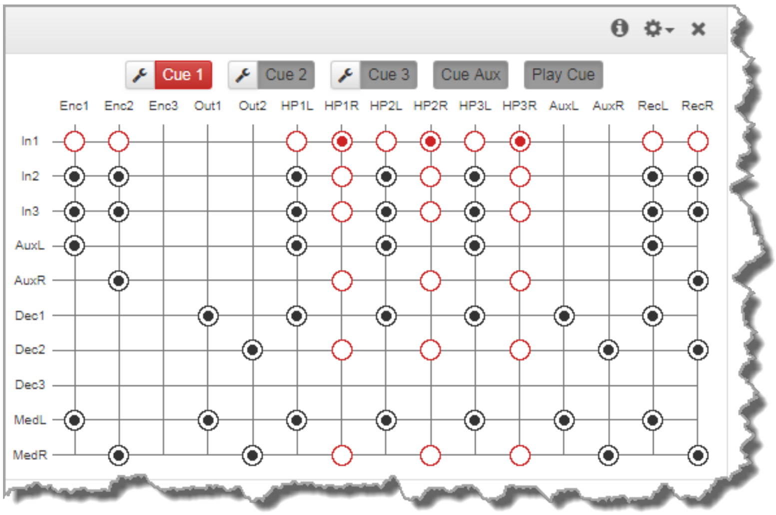
Click the **Options symbol** to view other matrix editor options, including:

1. **Undo Changes**: Clears any changes that have not been saved.
2. **Reset Defaults**: Resets the matrix to defaults for the currently loaded program, e.g. mono, stereo, mono/stereo plus IFB.
3. **Save as**: Save the current matrix settings as a new **Custom Mix** with a unique name (includes headphone and Cue/TB matrices).



View Cue and Talkback Matrices

Click the **Cue/Talkback 1-3** buttons at the top of the **Matrix Editor** to view Cue/Talkback routing in red.

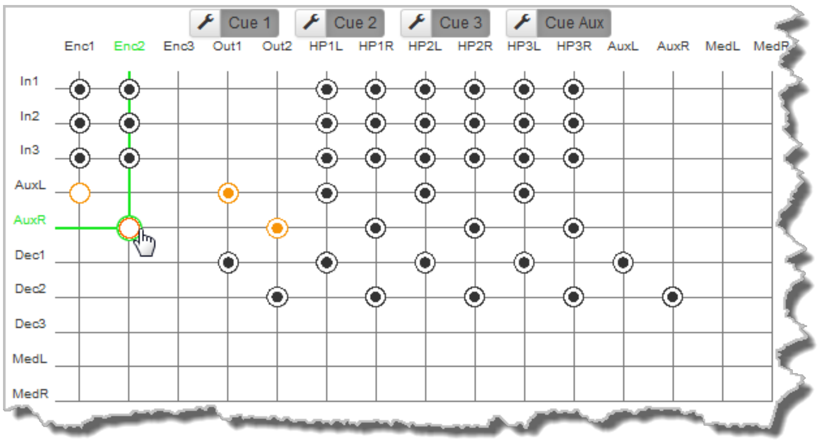


Important Notes: Click the **Information symbol** at the top of the panel to view details about the current matrix.

Using the Matrix Editor

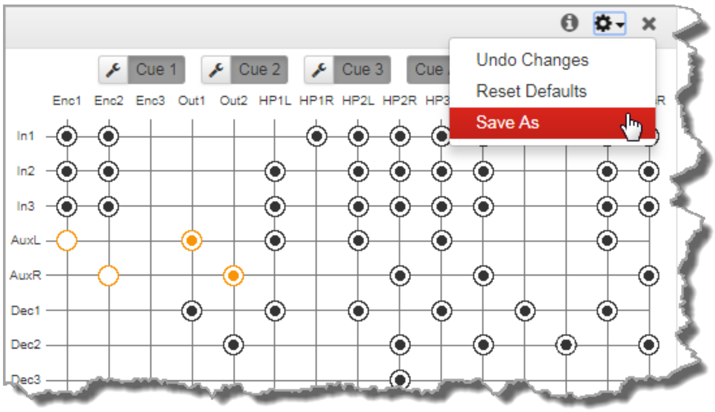
Routing can be adjusted very simply in the matrix editor. To edit default matrix settings click a crosspoint to select or deselect it. Routing edits are also reflected automatically in:

1. The Headphone mix visible in the **Headphones panel**.
2. The **XLR Outputs** and **Digital Outputs** accessed via **HOME** > **Audio** > **XLR Outputs** / **Digital Outputs** on the codec **TOUCH SCREEN**. In the following example, the auxiliary input (**AuxL** and **AuxR**) have been deselected from the encoder outputs (**Enc1** and **Enc2**) and have instead been routed to the analog XLR outputs (**Out1** and **Out2**). Note: Changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.

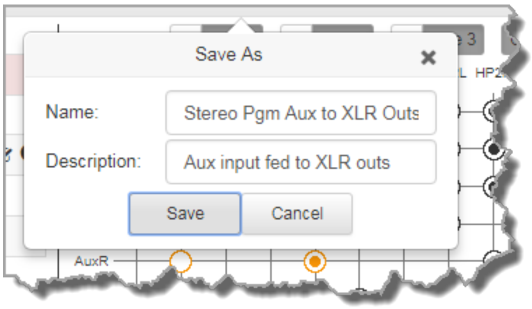


Edits can be saved as a custom mix, which also includes any edits to **Headphone**, **XLR Output**, **Digital Output** or **Cue/TB Mix** matrices. To save a custom mix:

1. Click the **Options symbol** and select **Save As**.



2. Enter a new **Name** and **Description** (if required) and click **Save**.



3. The new custom mix stays loaded in the codec until a new mix is loaded, an incompatible program is loaded, or program defaults are restored via **Reset Defaults**.

Important Notes:
- If a new program is loaded and it is compatible with the current custom mix then it will remain loaded. If an incompatible program type is loaded, the last compatible mix to suit the new program type is loaded, including any runtime changes made previously. E.g. If a custom stereo mix is loaded and then a mono program is loaded, the last mono mix used will be loaded.
- If you make runtime matrix edits to a loaded mono program, then load a stereo program, and subsequently reload the original mono program, the runtime matrix edits are recalled.

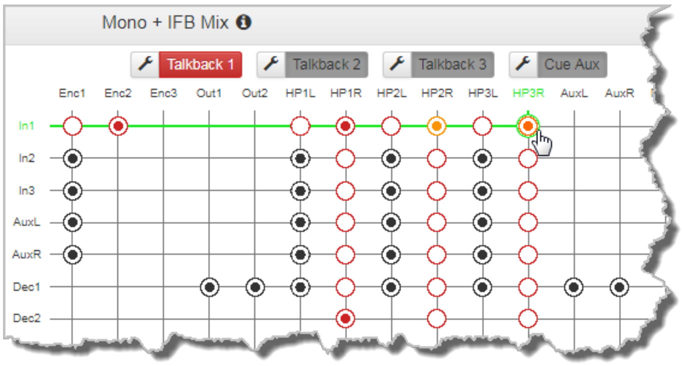
Editing Cue and Talkback Matrices for Cue 1-3

Typically, local cue intercom is configured for commentators to talk to each other offline. In cue mode, offline communications audio from inputs 1-3 is routed to the right side of all local headphone outputs (**HP 1-3**) when the **TB CUE** button is pressed. By default this is the same for:

- Mono Peer-to-Peer programs.
- Stereo Peer-to-Peer programs.
- 2 x Mono Peer-to-Peer programs.

When you load a **Mono Peer-to-Peer + IFB** program, or a **Stereo Peer-to-Peer + IFB** program, two independent audio streams are configured to stream program and communications audio separately. When an announcer presses an input's **TB CUE** button, audio is routed to the IFB audio stream encoder and outgoing audio is simultaneously monitored in the right side of the selected input's headphone output. Return IFB audio is also audible in the right side of the headphones.

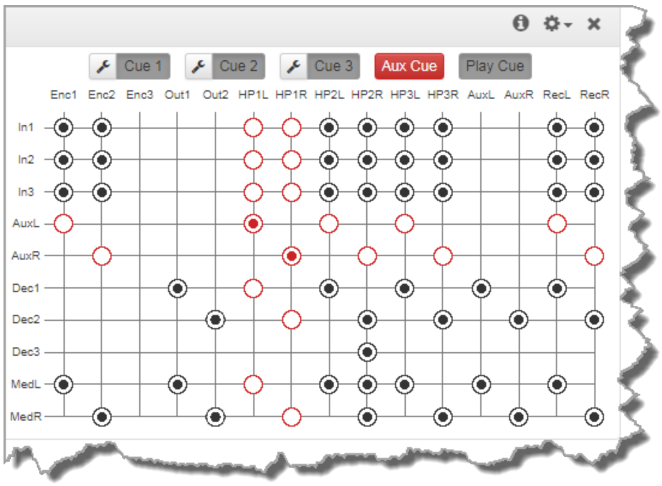
1. Click the **Cue/Talkback 1-3** buttons at the top of the **Matrix Editor** to view current routing.
2. Click a crosspoint to select or deselect an audio crosspoint. Note: Changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.



To learn more about the differences between cue and talkback functions, and editing cue and talkback matrices with the **TOUCH SCREEN**, see Cue and Talkback Operation.

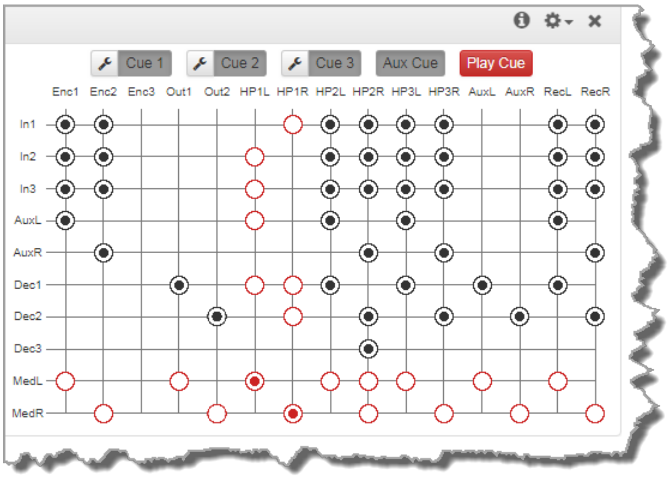
Auxiliary Input Cue

The codec supports offline cueing of external analog and digital sources attached to the stereo auxiliary input. By default, offline cue monitoring of the auxiliary input is only available via headphone output 1 (**HP1**). This routing can be adjusted as required.



Editing Play Cue

When using the file playback feature it is possible to configure file playback audio routing in playback Cue mode. By default, audio is routed to the left and right headphone output for headphone output 1 (**HP1L** and **HP1R**). This routing can be adjusted as required.

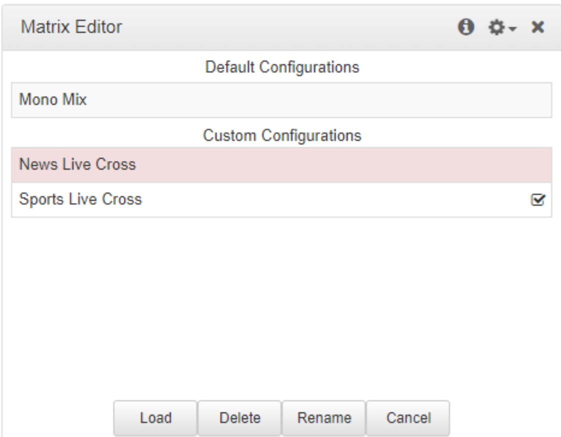


Important Notes: To enter offline cue mode for file playback select **HOME** > **Media** > **Playback** and then tap the **Cue** button on the **Playback** screen. An orange **Cue symbol** is displayed in the **Status Bar** at the top of the **TOUCH SCREEN** to notify the user that offline cue monitoring is enabled.

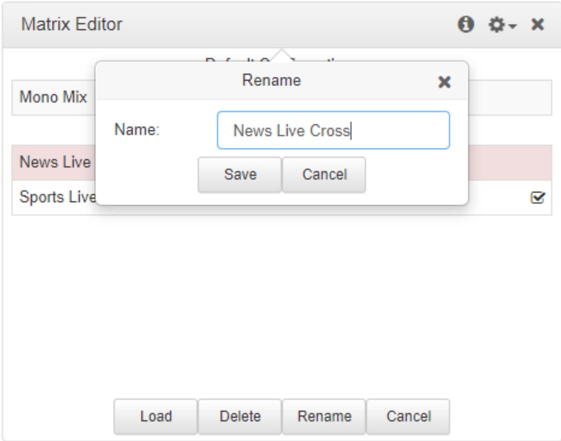
Load, Rename and Delete a Custom Matrix Editor Mix

When a new program is loaded it may be necessary to load a new custom mix. It is also possible to rename or delete mixes that are no longer required.

1. To load a custom mix click **Change Mix** and then select a mix listed under **Custom Configurations** in the left-hand pane, then click **Load**.



2. To rename a custom mix, click to select a saved mix and then click **Rename**. Edit the name and click **Save**.

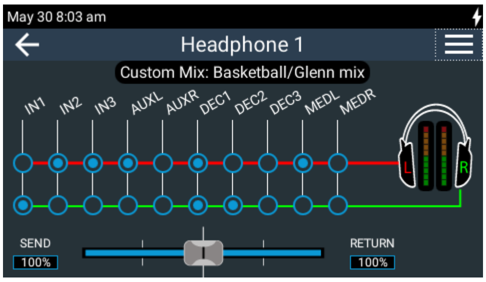


3. To delete a custom mix, click to select a custom mix and then click **Delete**. Note: A loaded mix cannot be deleted.

33.25 Headphone Matrix Editing

The **Headphones panel** in the HTML5 Toolbox allows the customization of audio source monitoring for each headphone output. The send/return balance between incoming and outgoing audio sources can also be adjusted.

The mix displayed for **Headphone 1-3** in the panel is reflected in the loaded mix in the **Matrix Editor**. If you create custom headphone mixes and load them, and then save all routing in the **Matrix Editor** as a custom matrix, all the custom headphone mixes and their **Send/Return** balance settings are also saved. In essence, custom headphone mixes are "attached" to a **Matrix Editor** matrix when it is saved. This is displayed in the codec as per the following image. In this example, "Basketball" is the name of the custom **Matrix Editor** matrix and "Glenn mix" is the name of the custom headphone mix attached to this matrix.

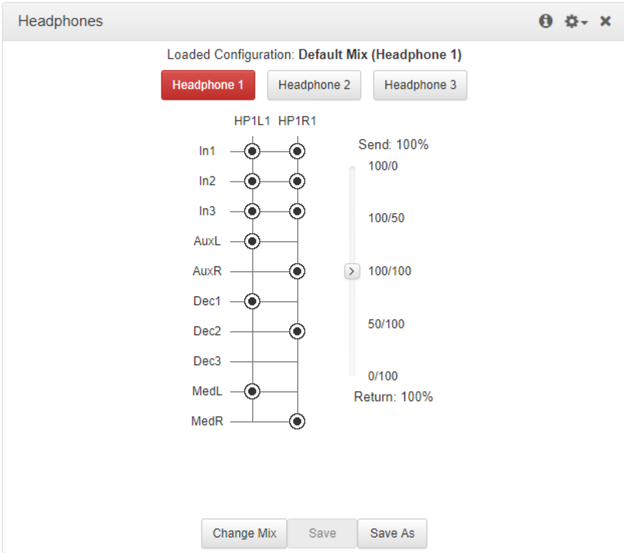


Important Notes:

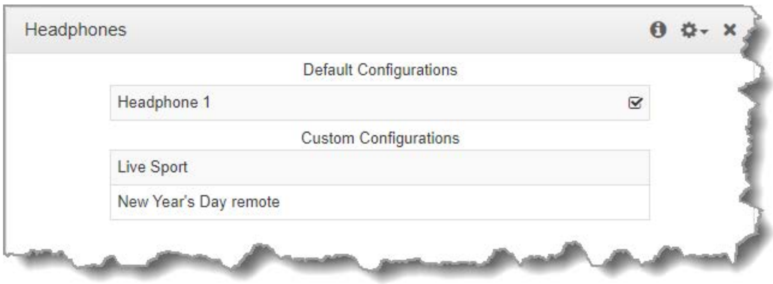
- If a custom headphone mix is saved with a custom matrix and the headphone mix is subsequently deleted, the custom headphone mix settings are retained in the saved custom matrix.
- Individual headphone source level adjustments for each headphone output are not saved with a **Matrix Editor** custom matrix. These individual source levels are accessed by swiping left in the **Headphone Mix** view on the **TOUCH SCREEN**.

Adjusting the Headphone Mix

To adjust settings open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Headphones** to open the **Headphones panel**. In the following image the default stereo matrix is loaded in the codec and the headphone matrix reflects this routing configuration. This is indicated by the **Loaded Configuration** at the top of the panel.

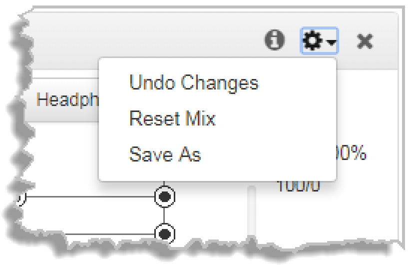


Click **Change Mix** to view and load previously saved custom headphone mixes listed under **Custom Configurations**.

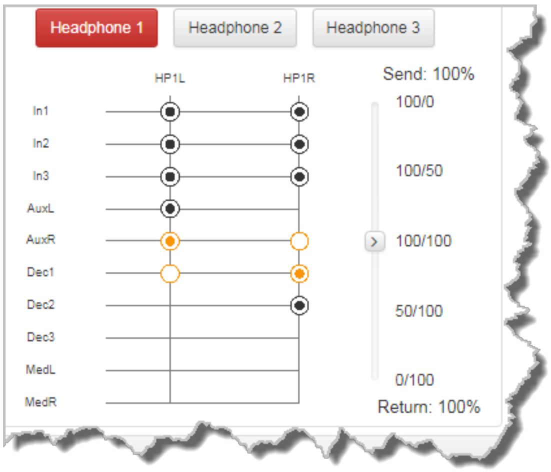


Click the **Options symbol**  to view other options, including:

1. **Undo Changes**: Clears edits in a headphone mix that have not been saved.
2. **Reset Mix**: Unloads a custom headphone mix and resets the mix to match the currently loaded matrix in the **Matrix Editor**, e.g. default matrix for a program type, or a custom loaded matrix.
3. **Save as**: Saves the current headphone mix as a **Custom Configuration** with a unique name.

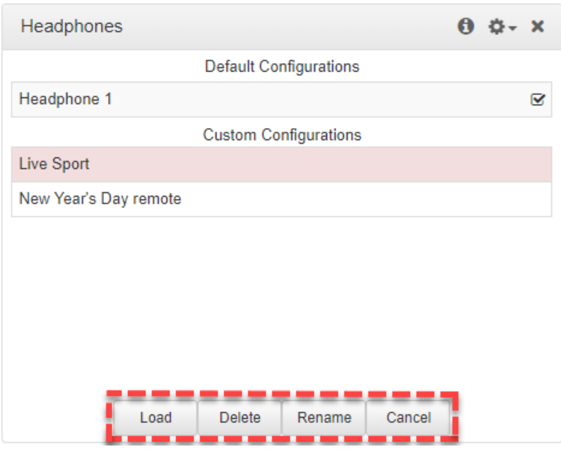


To edit headphone routing click to select **Headphone 1**, **2** or **3**, then click a crosspoint to select or deselect it. Routing edits are also reflected automatically in the **Matrix Editor**. Click and drag the **Send/Return** slider to adjust the balance between incoming and outgoing audio sources. Note: Routing changes not yet saved as a custom mix are displayed in orange. Runtime changes are audible in real-time and persist if the unit is powered down and rebooted.



Load, Rename and Delete Custom Mixes

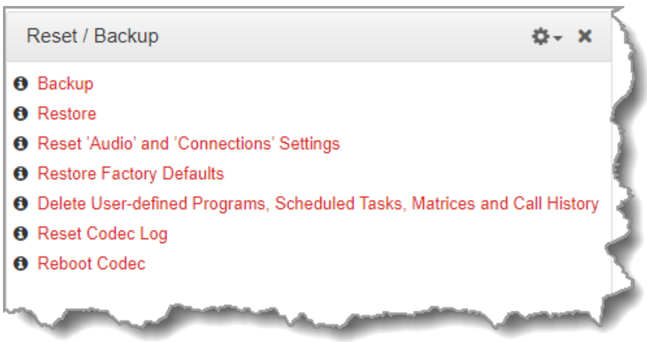
Select **Headphone 1**, **2** or **3**, then click **Change Mix** to load, rename or delete custom headphone mixes listed under **Custom Configurations**. Select a mix and then click the **Load**, **Rename** or **Delete** buttons to perform these functions. See ViA Headphone Controls for more details on adjusting, loading, saving and deleting headphone mixes.



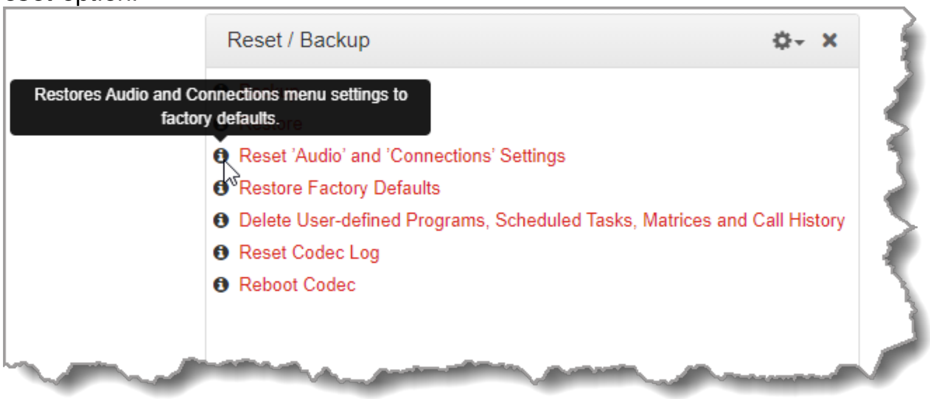
33.26 Reset Factory Default Settings

There are several options which allow you to backup, restore or reset specific factory default settings within the codec. See Reset and Restore Factory Defaults for more details on each option.

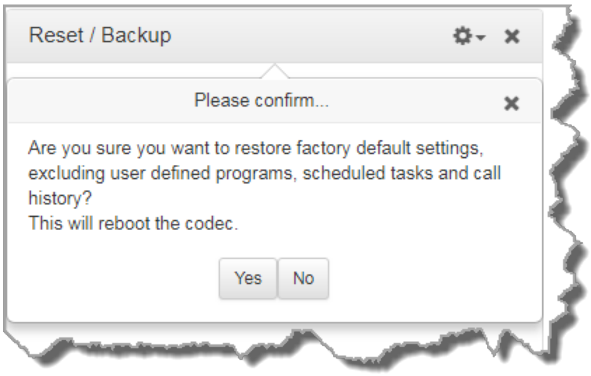
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset/Backup** to display the **Reset/Backup panel**.



2. Click one of the available reset options to adjust codec settings, or reboot the codec. Note: Hover with the mouse pointer over the **Information** ℹ symbol to view a tool-tip for each reset option.



3. A confirmation dialog appears for each option; click **Yes** to proceed.



33.27 Backup and Restore Functions

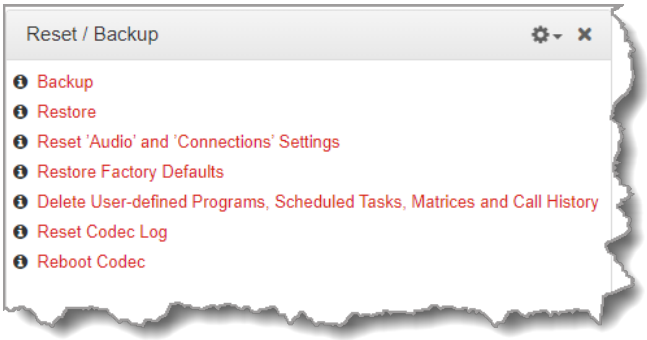
The HTML5 Toolbox Web-GUI can be used to backup and restore codec settings, including:

- Programs (containing a variety of connection settings) and scheduler data.
- All system settings (saved and current run time settings).

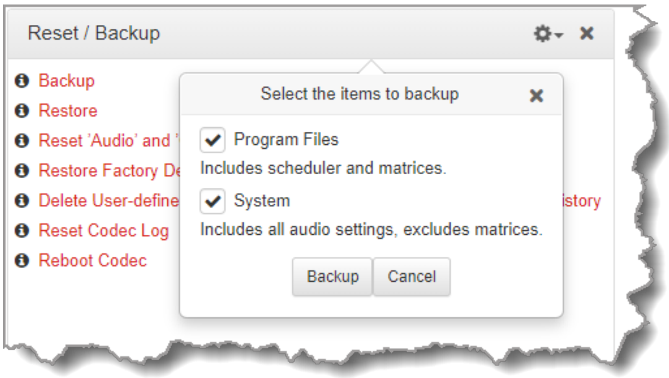
Backup files can also be used to copy configurations onto other codecs.

Creating Backup Files

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset/Backup** to display the **Reset panel**.



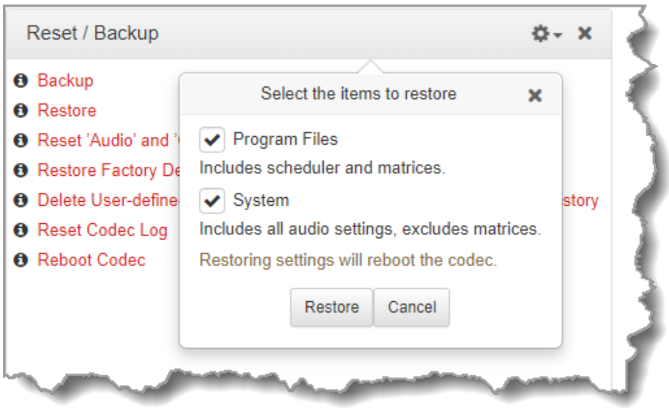
2. Click **Backup**.
3. Click to select the check-boxes to confirm your backup requirements, then click **Backup**.



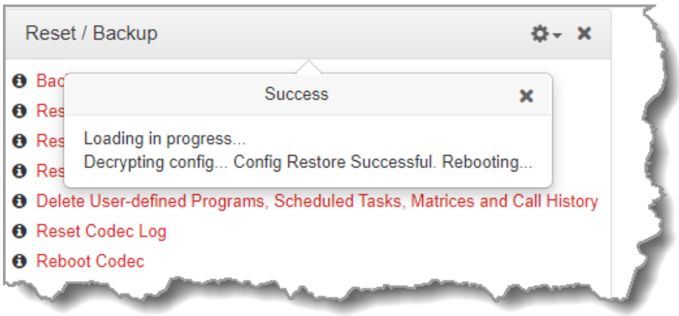
4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

Restoring Configuration File Settings

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset/Backup** to display the **Reset/Backup panel**.
2. Click **Restore**.
3. Click to select the check-boxes and confirm your restore settings. For example, you could select the **Program Files** check-box and deselect the **System** check-box if you are only copying programs onto codecs.



4. Click **Restore** and select a saved .tgz file to load onto the codec. A **Success** dialog confirms the files have been restored. Note: The codec will automatically reboot after restoring settings.



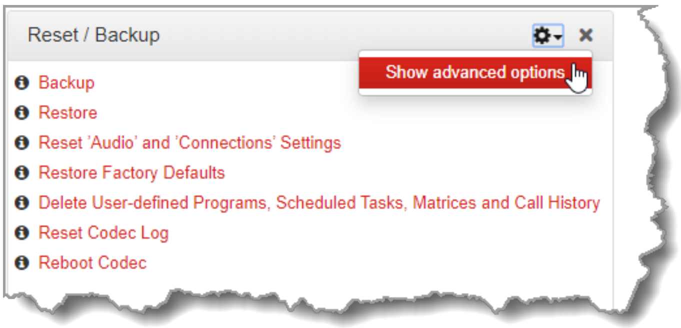
Note: The codec will automatically reboot when restoring system settings.

Advanced Settings: XML Config

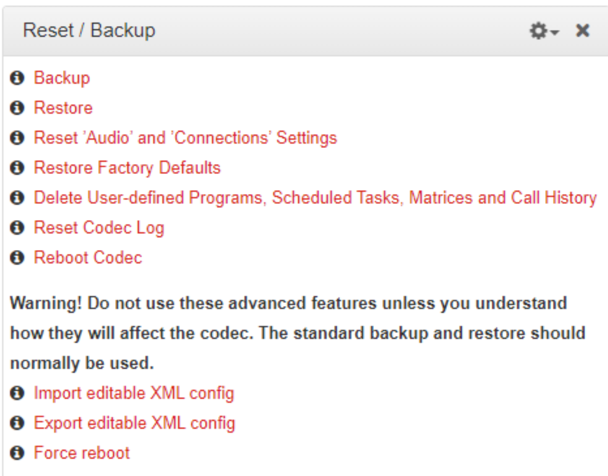
Caution: DO NOT use advanced XML configuration features unless you fully understand how they will affect the codec. The standard backup and restore function should normally be used. Damage to the codec may occur if this feature is used without fully understanding how it will affect the codec.

XML Config is a highly advanced feature which should only be performed by suitably qualified personnel. To import or export XML config files:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset/Backup** to display the **Reset/Backup panel**.
2. Click the **Options symbol** to view **Show Advanced Options**.



3. Click to select **Import/Export editable XML config** as required, or force the codec to reboot.

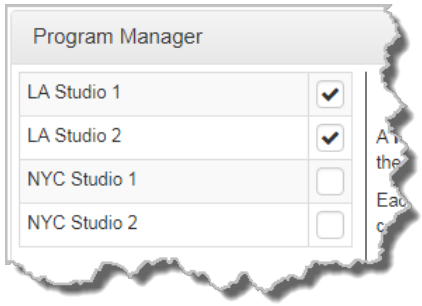


33.28 Import and Export Programs

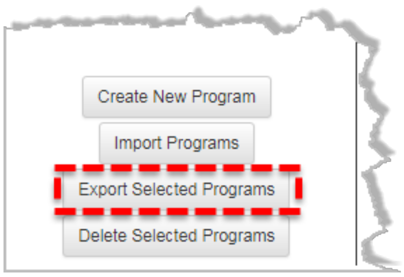
It is possible to import and export individual programs using the **Program Manager panel**.

Exporting Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click to select the check-box for the program or programs you wish to export.



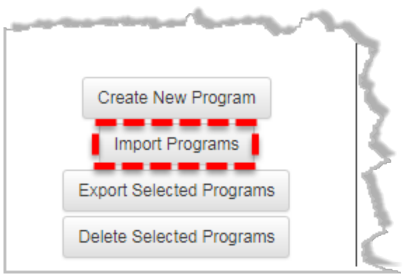
3. Select **Export Selected Programs** in the bottom-left corner of the **Program Manager panel**.



4. Navigate to a file folder and save the program .zip file.
5. Click **Save** to save the program file.

Importing Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Select **Import Programs** in the bottom-left corner of the **Program Manager panel**.

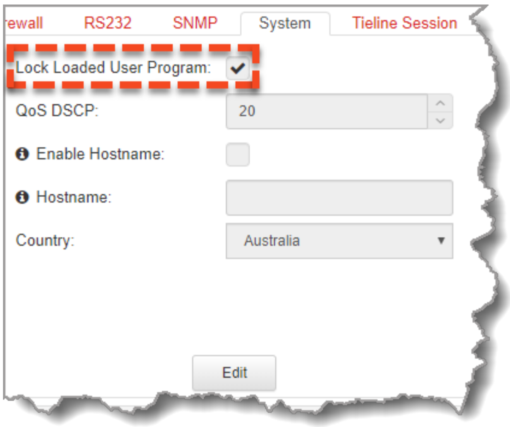


3. Navigate to the file folder containing the program .zip file to be imported.
4. Click to select the .zip file and click **Open** to import it.

33.29 Lock or Unlock Programs

It is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. mono, cannot be unloaded by a codec dialing in with a different program type, e.g. stereo. For example, if you require the codec at the studio to always connect in mono, simply load and lock a mono program in the codec. Generally programs will be up or down-mixed by the answering codec to match the loaded program type. In some situations incompatible program types will be rejected.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **System** and then click **Edit**.
3. Click the **Lock Loaded User Program** check-box to lock or unlock a user program in the codec.



4. Click **Save** to store the new configuration.

Important Note:
- A black **Padlock** symbol appears next to the program name in the **Connections panel** and in the **Program Loader panel** (in the Quick Connect web-GUI), to indicate a program is locked in the codec.
- It is only possible to lock custom programs in a codec.
- If **Lock Program** is enabled and you load a new custom program in the codec, **Lock Program** remains enabled and locks the most recently loaded custom program.

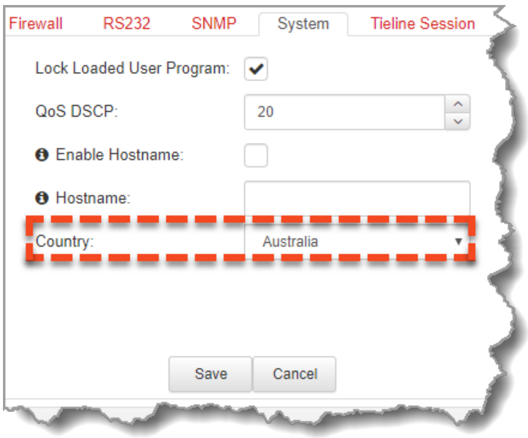
33.30 Configure Country Setting

The **Country** setting in the codec configures country-specific settings like:

- POTS line ring tones.
- POTS line impedance.
- Use of G.711 µ-law for North America and Japan, and G.711 a-law in most other regions of the world (e.g. Europe/Australasia), when the G.711 algorithm is used for IP/SIP or ISDN connections.

To configure the **Country** setting:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **System** and then click **Edit**.
3. Click the **Country** drop-down menu arrow to select the correct country of operation.



4. Click the **Save** button to save the new configuration.

33.31 Configuring SNMP in the Codec

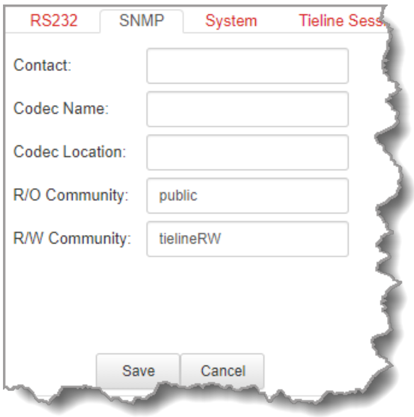
The codec supports Simple Network Management Protocol (SNMP) for managing devices on IP networks. To use SNMP you need to configure SNMP Device settings in your codec.

Description of SNMP Settings in the Codec

Features	Operation Button Descriptions
Codec Name	A user-specified alphanumeric identifier which may be used by third-party SNMP software to identify a device. The device name corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysName" SNMP attribute and is completely independent of DNS, NIS, WINS or other device naming and identification schemes, though convention is to use the device's fully-qualified domain name.
Codec Location	A user-specified alphanumeric string which may be used by third-party SNMP software to identify a device. Device location corresponds to the ".iso.org.dod.internet.mgmt.mib-2.system.sysLocation" SNMP attribute.
Contact	A text identifier for the contact person for this managed node, together with information on how to contact this person.
R/O Community	SNMP provides two types of access, namely Read-Only access and Read-Write access. The R/O Community identifier allows Read Only level access.
R/W Community	The R/W Community identifier allows Read/Write level access.

Configuring SNMP Settings in the Codec

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **SNMP** and click **Edit**.
3. Click in the text boxes to enter SNMP configuration settings.



4. Click **Save** to store the new configuration.

MIB Files for SNMP Configuration

Management Information Base (MIB) files are required for SNMP applications to interact with your Tieline codec and interpret SNMP data. The codec supports SNMPv1 and SNMPv2 MIB protocols. The required MIB files can be downloaded from the codec using the following link in a PC web browser connected to the same network as your codec:

- http://<YOUR_CODEC_ADDRESS>/mibs/tieline-mibs.zip

Save the .zip file to your PC and import the contents into the MIB browser you use to manage SNMP-enabled network devices.

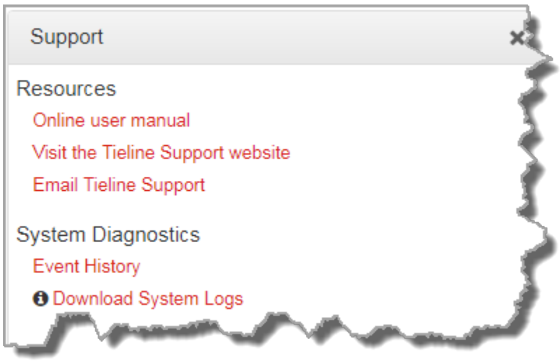
Important Note: The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213. An example of a free MIB browser is available at http://www.ireasoning.com/.

33.32 Download Logs

The codec is capable of providing diagnostic information via user logs, which can either be sent to Tieline support, or downloaded for user diagnostics.

Procedure for Sending Logs to Tieline

1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Download System Logs**.



3. Save the file to your computer and then send it as a .zip file to Tieline support at support@tieline.com

Download Event Logs

Event logs can be downloaded from the codec and viewed in your browser.

1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Event History** to view the event log in a new web-browser window.

Clearing Logs

This option should only be used if instructed to by Tieline support staff. To clear all event and other logs in the codec via the front panel, see the Reset and Restore Factory Default Settings section of this manual, or see Reset Factory Default Settings to clear recent log history using the Web-GUI.

33.33 Using the Program Scheduler

The program **Scheduler** is a powerful tool which facilitates automatically connecting and disconnecting programs using a simple calendar-based user interface. Key features include:

- Drag and drop to add programs into the scheduler.
- Automatically load and unload programs.
- Drag the top or bottom of a scheduled program to adjust the scheduled time.
- Customization of time-zones displayed in the scheduler.
- Day, week, month, list and timeline views available.
- View a list of scheduled upcoming 'events' in a separate panel.
- Enable and disable scheduled events in a snap.

Scheduler Overview

There are several panels associated with the Program Scheduler:

1. **Scheduler**: Create and manage scheduled "events."
2. **Scheduler Events**: View a list of all scheduled events.
3. **Scheduler History**: View previously scheduled events.

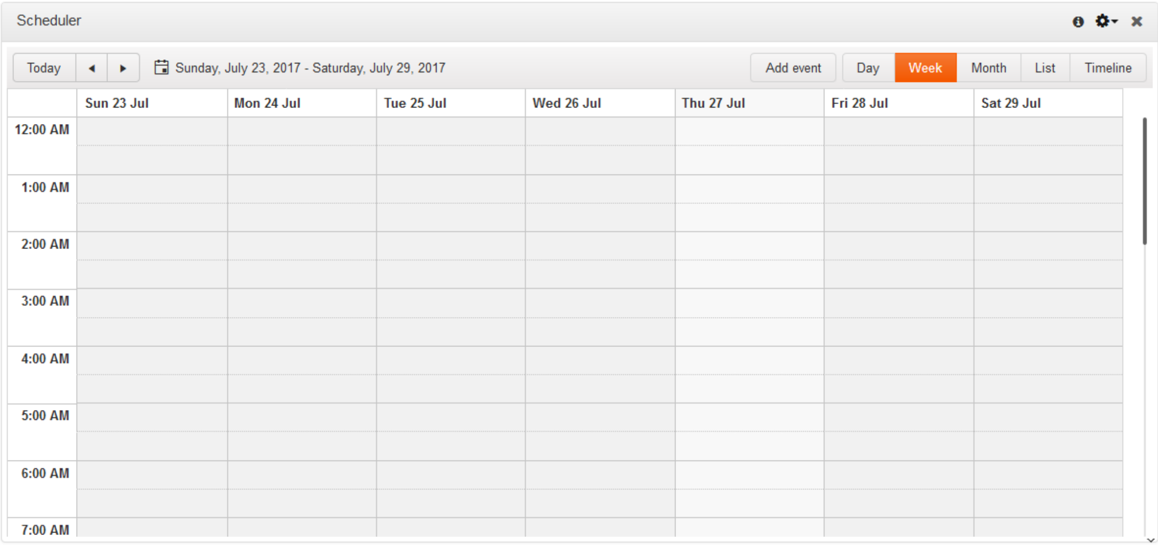
Important Notes:

- A program attempts to dial and connect at the scheduled event start time. Allow enough dial time when scheduling events.
- Allow sufficient time to disconnect and dial between consecutive events.
- A scheduled event will fail if a program is already connected.
- Consider configuring events in UTC time during daylight saving transitions to simplify set up.
- Events shorter than 15 minutes display a 15 minute time slot in the scheduler to enhance event visibility.

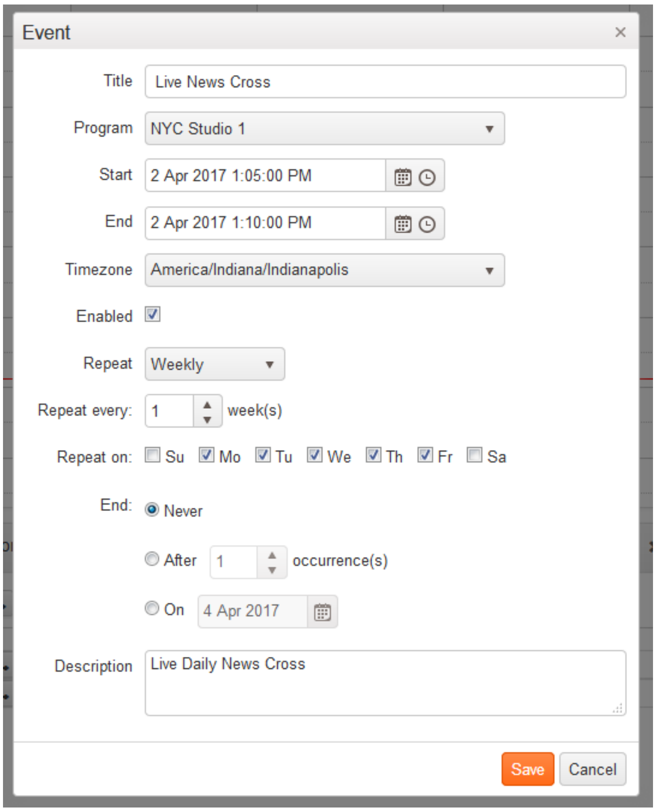
Scheduling New Events

To launch the **Event dialog** and schedule a new program event:

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Scheduler** to launch the **Scheduler panel**.
2. To add a new event click the **Add event** button in the **Scheduler panel**, or navigate to a day and time in the **Scheduler panel** and double-click when you want the programmed event to commence.



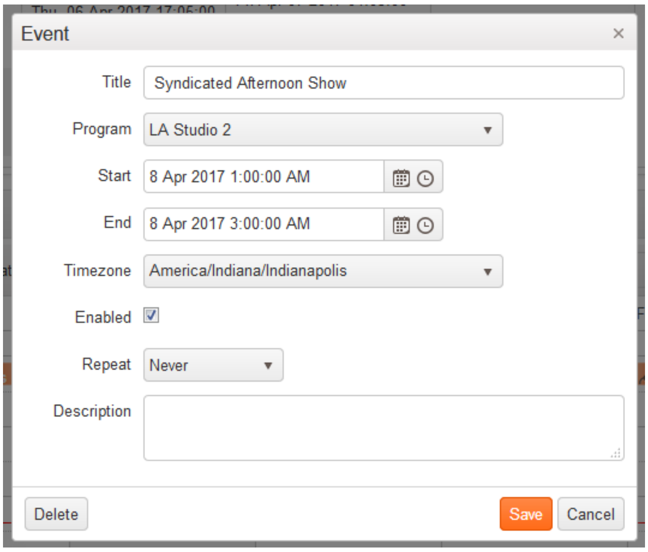
3. Add a **Title**, and adjust the **Start** and **End** time for each event. It is also possible to adjust the frequency of an event, e.g. daily, weekly or monthly.



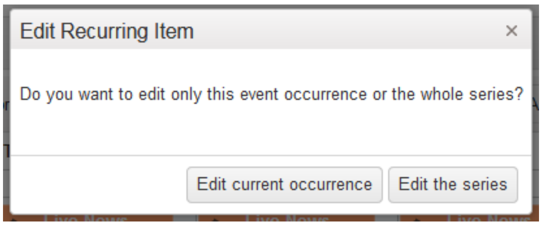
4. Click **Save** to add the new event to the **Scheduler panel**.

Edit or Delete an Event

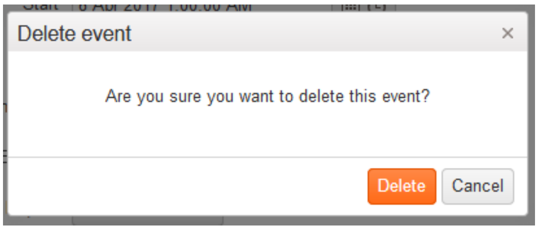
1. Double-click an event displayed in the **Scheduler panel** to open the **Event** dialog.



2. Two options are presented if the event is a recurring event:
   - Edit the current event you have selected, or
   - Edit all events in the series.

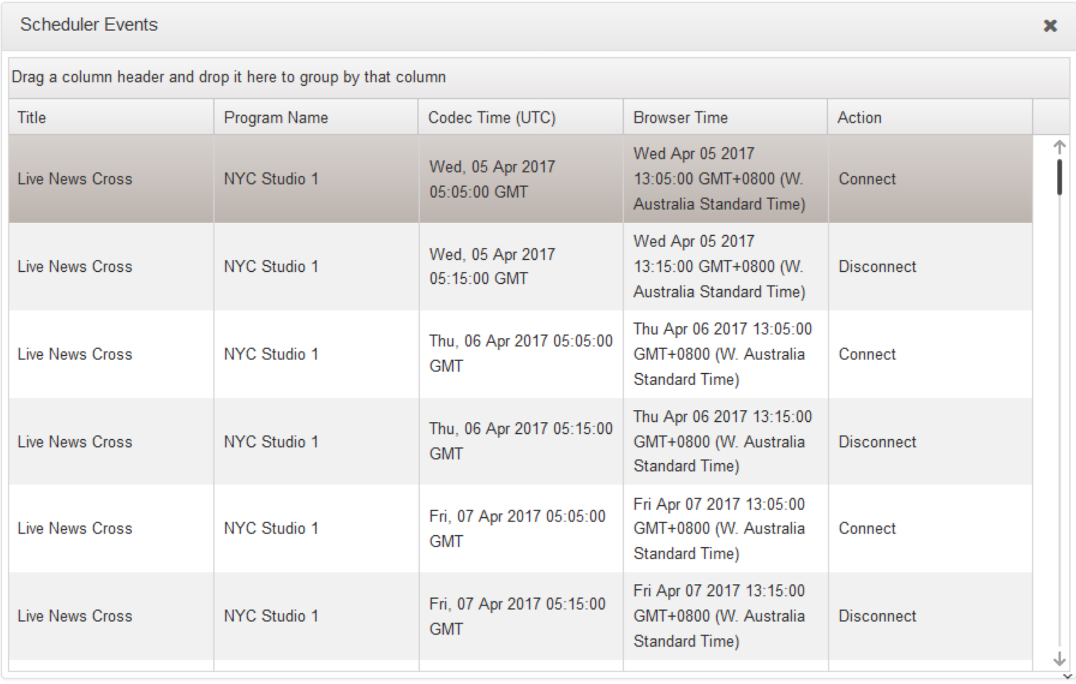


3. Select the preferred option and edit settings in the Event, then click **Save** to store all changes. Click **Delete** to delete an Event.
4. A confirmation dialog is presented to confirm the deletion of an event.



View Scheduler Events

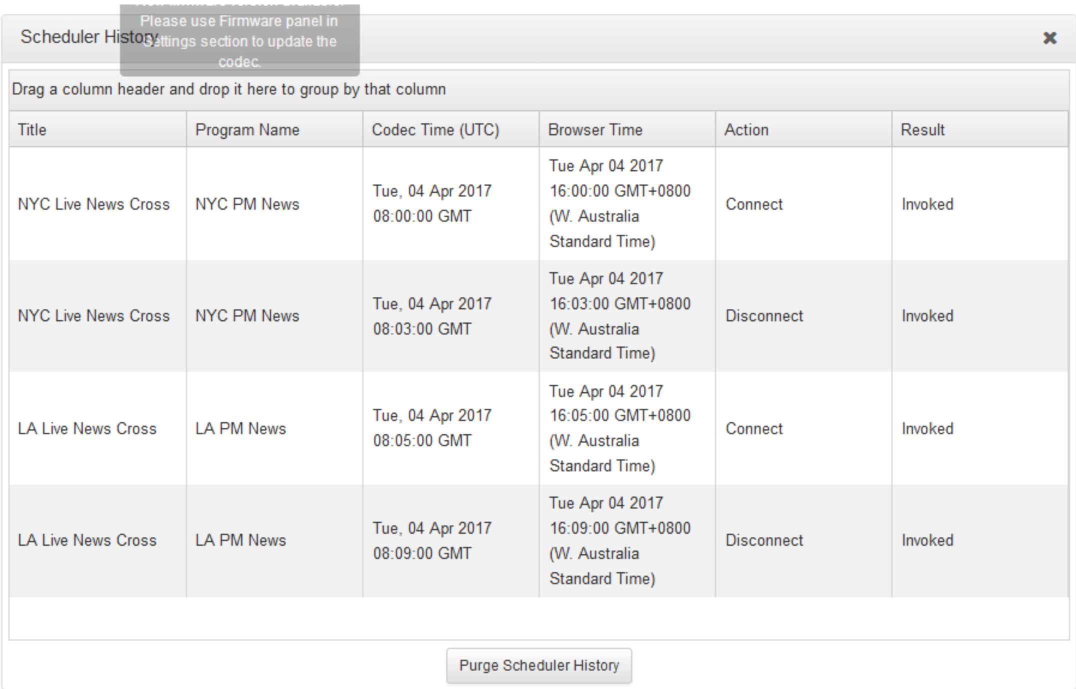
To view all scheduled events click **Connect** in the **Menu Bar**, then select **Scheduler Events** to launch the **Scheduler Events panel**. Use the scroll bar to view future events.



Title	Program Name	Codec Time (UTC)	Browser Time	Action
Live News Cross	NYC Studio 1	Wed, 05 Apr 2017 05:05:00 GMT	Wed Apr 05 2017 13:05:00 GMT+0800 (W. Australia Standard Time)	Connect
Live News Cross	NYC Studio 1	Wed, 05 Apr 2017 05:15:00 GMT	Wed Apr 05 2017 13:15:00 GMT+0800 (W. Australia Standard Time)	Disconnect
Live News Cross	NYC Studio 1	Thu, 06 Apr 2017 05:05:00 GMT	Thu Apr 06 2017 13:05:00 GMT+0800 (W. Australia Standard Time)	Connect
Live News Cross	NYC Studio 1	Thu, 06 Apr 2017 05:15:00 GMT	Thu Apr 06 2017 13:15:00 GMT+0800 (W. Australia Standard Time)	Disconnect
Live News Cross	NYC Studio 1	Fri, 07 Apr 2017 05:05:00 GMT	Fri Apr 07 2017 13:05:00 GMT+0800 (W. Australia Standard Time)	Connect
Live News Cross	NYC Studio 1	Fri, 07 Apr 2017 05:15:00 GMT	Fri Apr 07 2017 13:15:00 GMT+0800 (W. Australia Standard Time)	Disconnect

View Event History

To view the history of scheduled events in a codec click **Connect** in the **Menu Bar**, then select **Scheduler History** to launch the **Scheduler History panel**. Press the **Purge Scheduler History** button to delete all events listed.



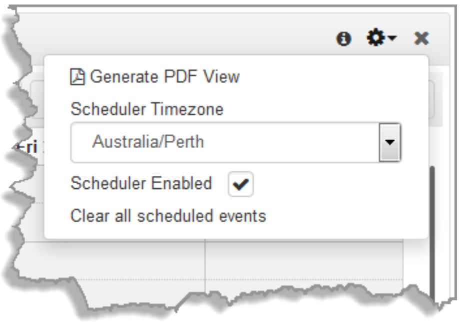
Title	Program Name	Codec Time (UTC)	Browser Time	Action	Result
NYC Live News Cross	NYC PM News	Tue, 04 Apr 2017 08:00:00 GMT	Tue Apr 04 2017 16:00:00 GMT+0800 (W. Australia Standard Time)	Connect	Invoked
NYC Live News Cross	NYC PM News	Tue, 04 Apr 2017 08:03:00 GMT	Tue Apr 04 2017 16:03:00 GMT+0800 (W. Australia Standard Time)	Disconnect	Invoked
LA Live News Cross	LA PM News	Tue, 04 Apr 2017 08:05:00 GMT	Tue Apr 04 2017 16:05:00 GMT+0800 (W. Australia Standard Time)	Connect	Invoked
LA Live News Cross	LA PM News	Tue, 04 Apr 2017 08:09:00 GMT	Tue Apr 04 2017 16:09:00 GMT+0800 (W. Australia Standard Time)	Disconnect	Invoked

Other Scheduler Options

Click the **Options symbol** in the top right-hand corner of the panel to reveal a drop-down menu displaying other available options in the scheduler, including:

1. Generate a PDF view of scheduled events.

2. Adjust the Scheduler Timezone.
3. Enable / Disable the Scheduler.
4. Clear all Scheduled Events.



33.34 RS232 Data Configuration

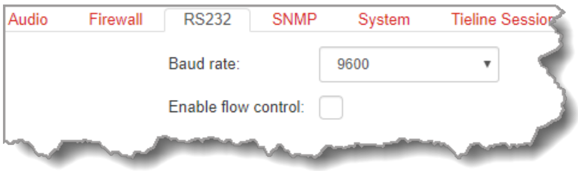
The codec supports both in-band and out-of-band data depending on the connection transport and algorithm you are using. In-band RPTP data is automatically included within an audio stream when using Tieline Music and Music PLUS algorithms over any transport – IP, ISDN or POTS. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP	ISDN and POTS
Tieline Music and MusicPLUS	• In-band RPTP data is enabled automatically • Synchronized out-of-band data can be enabled and disabled as required • Using out-of-band data with rules between G5 codecs employing relay reflection minimizes latency • These algorithms must be used when connected to G3 codecs as they don't support out-of-band data	• In-band RPTP data is enabled automatically and used for all rules including relay reflection
All other algorithms	• No in-band data available; synchronized out-of-band data can be enabled and disabled	• No in-band or out-of-band data available

Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** or **Connections panel** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. This will allow the codec to connect to external devices and send RS232-compatible data via the serial port on the rear panel. Alternatively, select **Home screen > Dialer > Auxiliary Data > On** to enable auxiliary data using the codec's **TOUCH SCREEN**.

Setting RS232 Data Rates and Flow Control

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **RS232**, then click **Edit** and click the **Baud rate** drop-down menu arrow to select the serial port baud rate. Ensure this matches the baud rate of the external device connected to the RS232 port on the codec.
3. Click to select the **Enable flow control** check box and enable flow control, then click **Save** to store the new settings.



To adjust data rates and flow control settings using the codec **TOUCH SCREEN** select **Home screen > Settings** **> System > RS232** **> Flow Control > Enable**.(see Enabling RS232 Data).

Important Notes:

- When connecting to G3 codecs over IP, ISDN or POTS only in-band data is available via the Music and MusicPLUS algorithms.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data over multicast connections.
- It is important to enable serial port flow control as it regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you configure the serial port baud rate to match the setting of the external device to which you are connecting. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Tieline session data sent from the dialing codec will configure all other compatible Tieline codecs (non-G3) when you connect.
- Bidirectional RS232 data is supported over the first audio stream in a program with multiple audio streams, e.g. 2 x mono peer-to-peer, or mono/stereo program with a separate IFB connection.

33.35 Creating Rules

Codec 'rules' configure events based on specific codec actions. A range of default rules are preconfigured in the codec to facilitate activation of the most common events required by broadcast engineers. Typically rules are based on a change in the state of a physical **CONTROL PORT GPIO**, or a WheatNet-IP logic IO, or a codec program being connected or disconnected. There are three categories of rules:

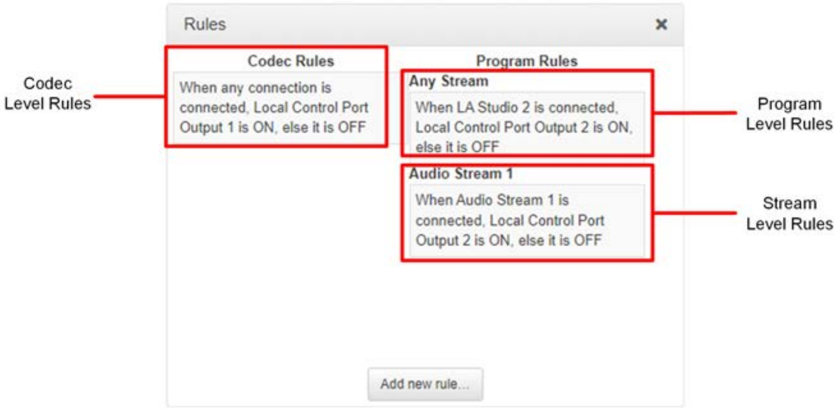
1. Codec level rules: Rules based on programs or codec hardware and software I/O states, e.g. Connect or disconnect a program when an input is toggled, or synchronize a local input to a remote relay.
2. Program level rules: Rules based on codec behaviors at the program level, e.g. Connect and disconnect program A when an input is toggled, set a custom mix when a relay is activated, or synchronize a local input to a remote relay.
3. Stream level rules: Rules based on codec behaviors at the stream level, e.g. Connect and disconnect stream A when an input is toggled, or synchronize a local input to a remote relay.

There are three ways to create rules in the HTML5 Toolbox Web-GUI:

1. **Rules panel**: Configure codec level rules related to programs and/or hardware and software I/O states.
2. **Program Manager panel**: Configure program level rules early in the **Program Manager panel** wizard.
3. **Program Manager panel**: Configure stream level rules for each audio stream as you proceed through the **Program Manager panel** wizard.

Important Notes:
- Rules can only be created with the Web-GUI while the codec is disconnected.
- Program and stream level rules configured in the **Program Manager panel** are only active when the program is loaded.

Following is a summary of how codec, program and stream level rules are displayed in the **Rules panel** when configured.



Enabling Data

Data is disabled by default and must be enabled to allow contact closure operation and transmission of RS232 data. Select **Enable Auxiliary Data** when creating a program in the **Program Manager panel** or **Connections panel** to enable out-of-band RS232 data and activate rules employing relay reflection over a connection. Data can also be enabled using the codec's **TOUCH SCREEN** by selecting **Home screen > Dialer > Auxiliary Data > On** (see Enabling RS232 Data for more info).

Important Notes:

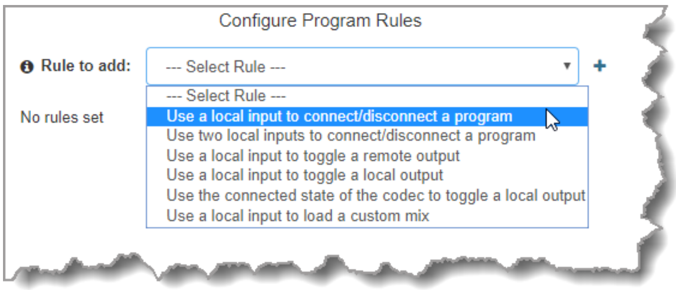
- The codec has 4 physical **CONTROL PORT** GPIOs and 64 Tieline logic IOs. These Tieline logic IOs are only supported between Genie, Merlin, ViA and Bridge-IT IP codecs.
- Up to 64 WheatNet logic IOs are available in Genie Distribution and Merlin PLUS WheatNet-IP codecs, which allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **SOURCE** buttons 1-3.
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet logic IO (LIO) in a Tieline WheatNet-IP codec.
- Tieline WheatNet-IP codecs require Wheatstone Razor firmware version 1.4.22 or later to support WheatNet LIOs. In addition, the WheatNet-IP codec must have the **WNet Enable LIO** checkbox selected in the **Options panel** of the HTML5 Toolbox Web-GUI.
- Relay reflection is not available for SIP and Multicast Client programs.

Configure Rules with the Program Manager Panel

To configure program or stream level rules follow the instructions in this user manual for setting up connections.

Program Level Rules

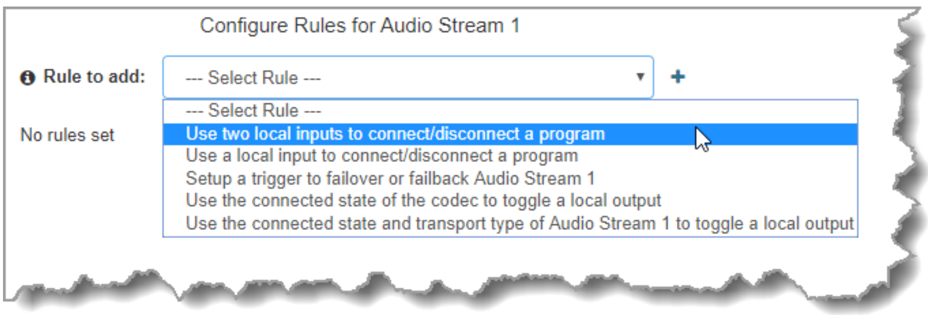
In the **Program Manager panel** wizard use the **Configure Program Rules** screen to configure program level rules. The rules available are displayed in the following image.



Note: Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.

Stream Level Rules

In the **Program Manager panel** wizard use the **Configure Rules for Audio Stream** screen later in the wizard to configure stream level rules. The rules available are displayed in the following image.

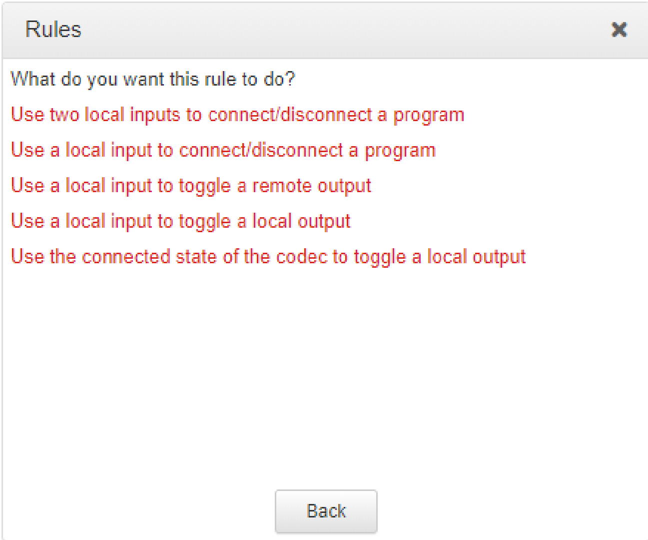


Note: A subset of filtered rules will be displayed for an **Answer only** audio stream connections.

Configuring Rules with the Rules Panel

To view rules options in the **Rules panel**:

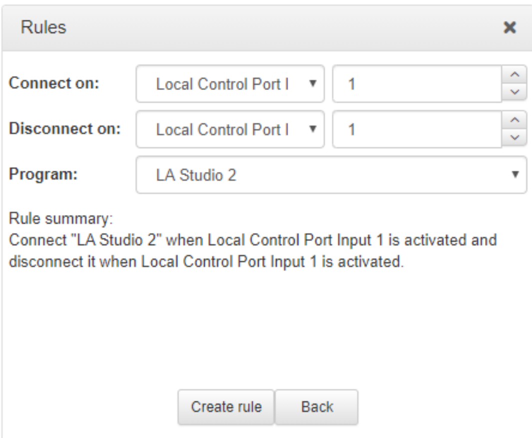
1. Open the Toolbox HTML5 Web-GUI and click **Control** in the **Menu Bar**, then click **Rules** to display the **Rules panel**.
2. Click **Add New Rule**
3. Click to select the appropriate rule for your requirements.



Rule 1: Use Two Local Inputs to Connect/Disconnect a Program

This rule is used to connect and disconnect a selected program when different codec control port inputs or virtual inputs are activated.

1. Click the first rule in the **Rules panel** titled **Use two local inputs to connect/disconnect a program**.
2. Click the drop-down arrows to select the control port input used to connect the selected program, and then select the alternative input used to disconnect the program.
3. Click the drop-down **Program** arrow to select the program to be connected.

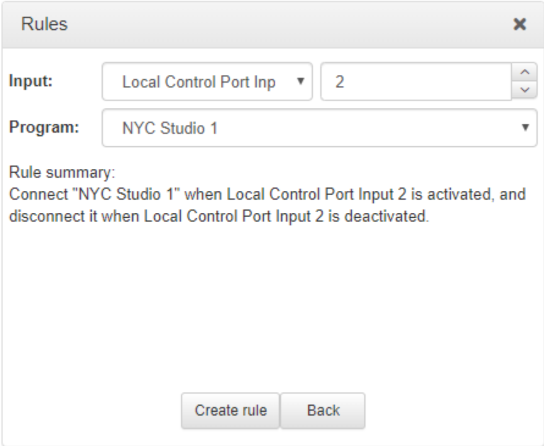


4. Check the **Rule Summary** and click **Create Rule** to save the settings.

Rule 2: Use a Local Input to Connect/Disconnect a Program

This rule is used to connect and disconnect a selected program when a control port or virtual input is toggled.

1. Click the second rule in the **Rules panel** titled **Use a local input to connect/disconnect a program**.
2. Click the drop-down arrows to select the control port input or virtual input used to toggle connecting and disconnecting a program.
3. Click the drop-down **Program** arrow to select an individual program which will connect and disconnect when the input is toggled.

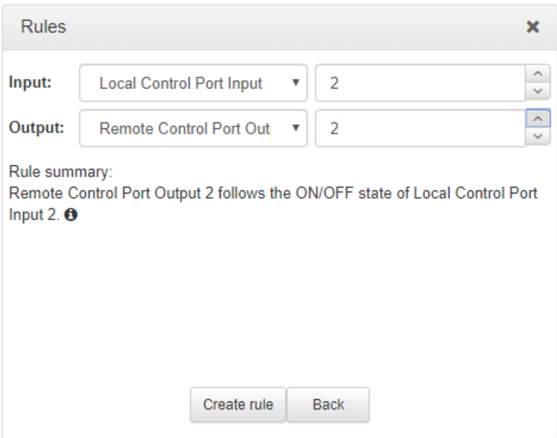


4. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 3: Use a Local Input to Toggle a Remote Output

Use this rule to allow a local codec's control port input or virtual input to change the state of a remote output.

1. Click the rule in the **Rules panel** titled **Use a local input to toggle a remote output**.
2. Click the drop-down arrow to select the local input used to control a remote output.

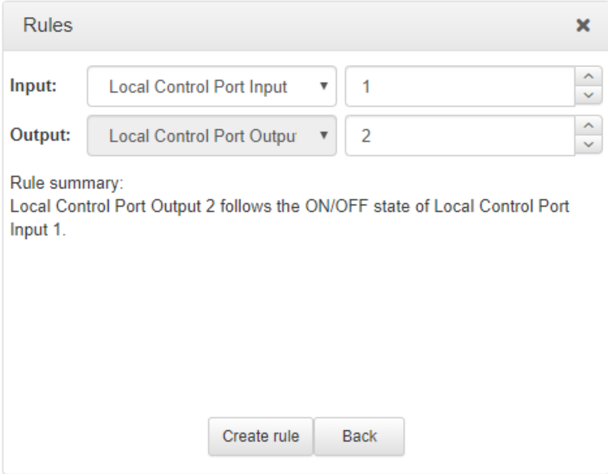


3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 4: Use a Local Input to Toggle a Local Output

Use this rule allow a local control port input or virtual input to change the state of a local relay output.

1. Click the rule in the **Rules panel** titled **Use a local input to toggle a local output**.
2. Click the drop-down arrow to select the local control port input used to control a local control port output.

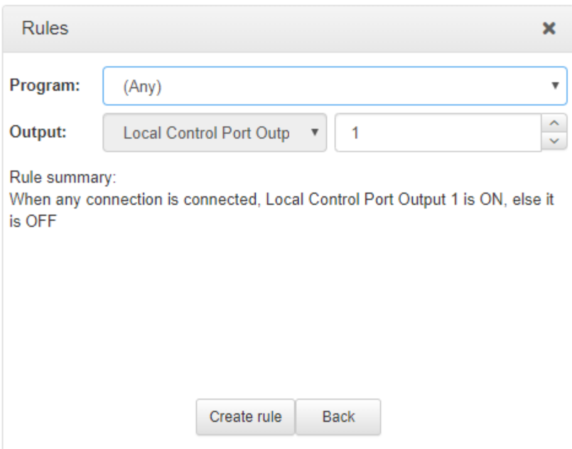


3. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 5: Use the Connected State of the Codec to Toggle a Local Output

This rule is used to toggle a codec's control port relay output each time a program connects and disconnects.

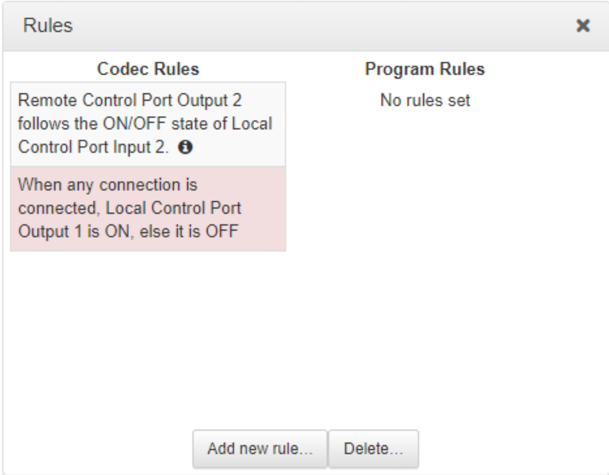
1. Click the rule in the **Rules panel** titled **Use the connected state of the codec to toggle a local output**.
2. Click the drop-down **Program** arrow to select the program which will affect the relay toggle function, or use the default setting whereby any program connecting will toggle the relay output.
3. Click the drop-down arrow and select the relay output you want to toggle.



4. Check the **Rule summary** and click **Create Rule** to save the settings.

Deleting Rules

1. Open the HTML5 Toolbox Web-GUI and click **Rules** in the **Menu Bar** to display the **Rules panel**.
2. Click to select the rule you want to delete.
3. Click the **Delete** button.

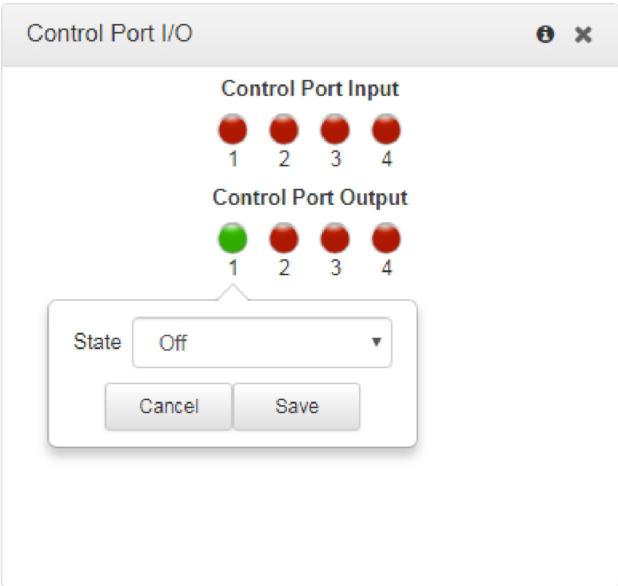


4. Click **Yes** in the confirmation dialog.

33.36 Monitoring Control Port I/O Status

It is possible to monitor the status of the four control port inputs and four opto-isolated outputs available using the DB15 **CONTROL PORT I/O** connector. To monitor status:

1. Open the Toolbox HTML5 Web-GUI and click **Control** in the **Menu Bar**, then click **Control Port I/O** to display this panel.
2. Click on an output to change the state from **Off** to **On**, then click **Save**. Note: Input states cannot be changed.



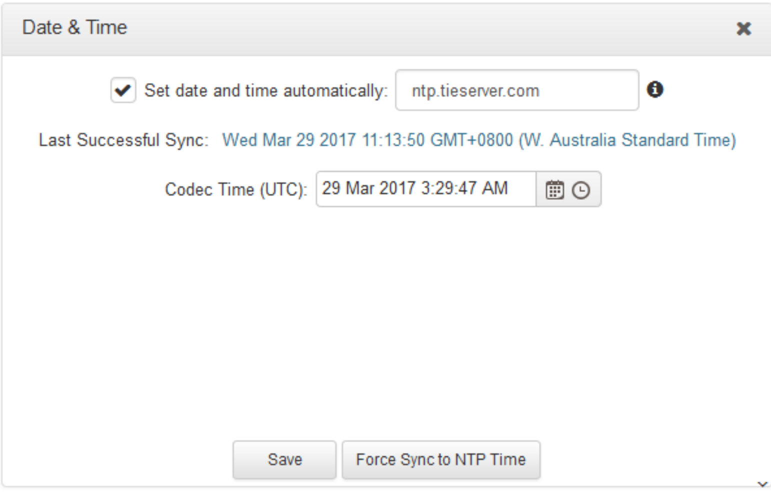
Tieline and WheatNet virtual logic inputs 5-7 can be activated by pressing the F1 button and the **SOURCE** buttons on inputs 1-3. E.g. F1 + **SOURCE 1**. Note: The relay state is activated for the duration of the key press.

33.37 Adjusting Codec Time and Date

Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks. By default **Use NTP** time is enabled in the codec and it will synchronize with **ntp.tieserver.com**. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected.

To manually synchronize time settings in the codec:

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Date & Time** to display the **Date & Time panel**.
2. Click **Force Sync to NTP Time** to manually synchronize the codec to NTP time.



Important Notes:
- It may take more than one attempt to **Force Sync to NTP Time**.
- When NTP address settings are configured and enabled, the codec will immediately jump to the new time when it synchronizes with the server. This may cause scheduled events to be missed.

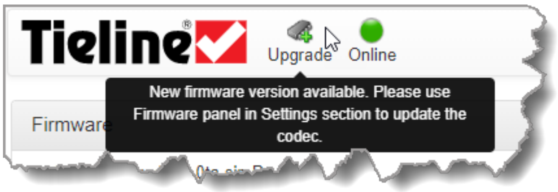
33.38 Upgrading Codec Firmware

To download the latest codec firmware visit www.tieline.com. See Upgrading Firmware via SD to upgrade codec firmware using a USB stick with new firmware copied onto it.

New Firmware Notifications

By default the HTML5 Web-GUI integrates with TieServer to automatically update users when a firmware upgrade is available.

1. Connect the codec to a PC using a LAN connection and open the HTML5 Toolbox Web-GUI.
2. If new software is available the **Upgrade** symbol appears in the top-left of the screen.



3. Click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel** to perform the firmware upgrade.

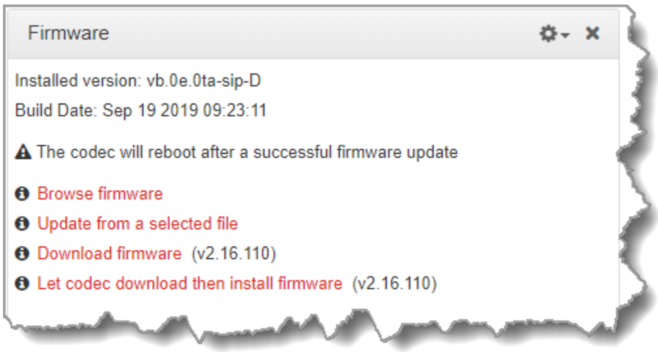
Performing a Firmware Upgrade

There are several firmware upgrade options available:

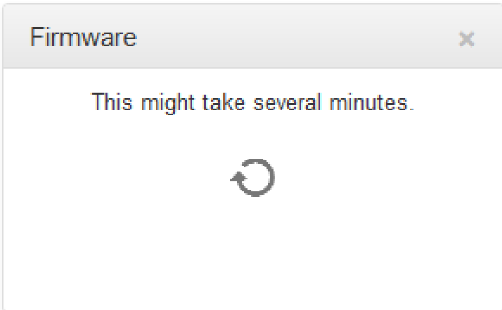
1. **Browse Firmware**: Click to navigate to the Tieline website and download the latest firmware for the codec.
2. **Update from a Selected file**: Click to navigate to a firmware file saved on a computer or network drive.
3. **Download firmware**: Click to download a previous reliable firmware version. Note: Only visible when a new release is available.
4. **Let codec download and install firmware**: Click to download a previous reliable firmware version directly into the codec and then complete the update. Note: Only visible when a new release is available.
5. Install from HTTP sources from within private networks: Click to select the **Options symbol** and select **Show Advanced Options** to install official firmware versions when internet access is unavailable.

The following procedure explains how to perform codec firmware upgrades with a downloaded firmware file saved to your PC.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel**.



2. Click **Browse firmware** to search for the latest firmware and download it to your computer. Note: the **Download firmware** and **Let codec download then install firmware** links are only visible if firmware more recent than the currently installed version is available.
3. Once the firmware has been saved locally, click **Update from a selected file** in the **Firmware panel**.
4. Select the **.bin** file you are using to perform the upgrade and click **Open** to start the upgrade. **IMPORTANT**: The codec will reboot automatically after the firmware upgrade. DO NOT remove power or reboot the codec before the update has completed and the codec has rebooted itself.



5. We recommend clearing your browser cache after the upgrade is complete. The short cuts for this are:
   - Google Chrome: shift+Ctrl+delete
   - Mozilla Firefox: Ctrl+shift+delete
   - Internet Explorer: Ctrl+shift+delete
   - Safari: Ctrl+alt+e

34 Reference

The following sections contain reference and troubleshooting information.

34.1 Installing the Codec at the Studio

Studio IP Streaming Setup for Tieline Audio Codecs

The following instructions are intended to help you configure your internet connection and Tieline codecs at the studio to enable incoming calls over the internet from a remote Tieline codec. It is assumed that you have a basic understanding of your IP network and how to configure IP devices. If you have limited IT network knowledge, we recommend you engage the services of an IT professional to install the public IP address and perform the Network Address Translation (NAT) and port forwarding between the public internet and your private Local Area Network (LAN) at the studio.

Prerequisites

The following procedures are valid for:

- All firmware versions in the Genie and Merlin codec families and ViA.
- All Bridge-IT Basic and Pro and Bridge-IT XTRA codecs with firmware release v.2.x or higher.
- All Commander G3 and i-Mix G3 codecs.

Getting Started at the Studio

To perform a typical codec installation at the studio you will need to:

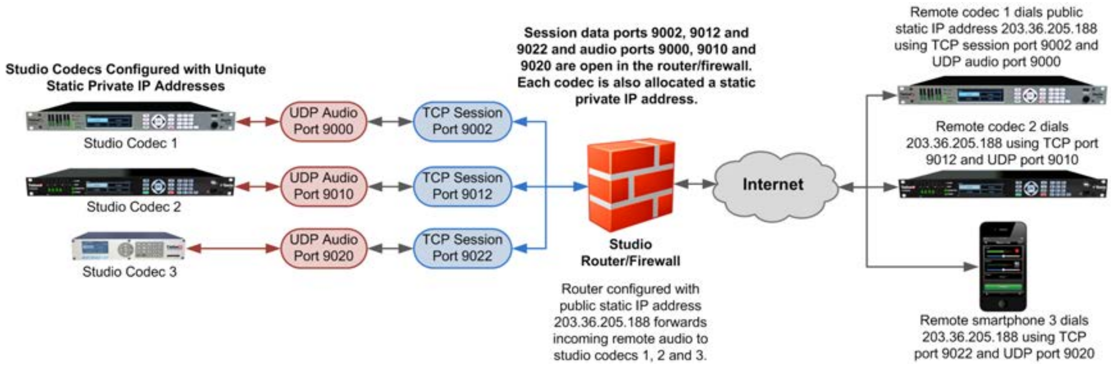
1. Contact your Internet Service Provider and organize a dedicated high speed broadband connection at the studio for your codec with a public static IP address. Do not share this connection with other devices.
2. Install your codec at the studio and attach an active RJ-45 LAN cable to the “LAN” or “Ethernet” port on the rear of the codec. Please note:
   - The green LED underneath the “LAN” or "Ethernet" port will illuminate and the orange LED will flash steadily if you are connected to an active LAN connection.
   - The Genie and Merlin families of IP codecs support two simultaneous Ethernet connections.
3. If you are connecting a single codec to a router without a firewall you can enter the public IP address, Subnet Mask and Gateway directly into the codec and your work is done. Note: your Telco should be able to provide this information.
4. Alternatively, if you are connected to a router with a firewall, configure Network Address Translation (NAT) in your router. NAT is performed between the public internet and your private Local Area Network (LAN) by your router. Your remote codec sends IP data packets to the studio router's public static IP address and the router performs NAT, which forwards these data packets to the private IP address allocated by the router to your codec. As part of this process we recommended you:
   - Connect to your router using a web-browser.
   - Configure it to allocate a static private IP address for each codec.

Important Note: The IP address may change if the codec is allocated a DHCP IP address by the router and it loses power or is temporarily disconnected from the LAN. This will cause problems for remote codecs attempting to dial and connect.

5. Ensure your router's firewall is configured with the relevant TCP and UDP IP ports open to allow data traffic between your codec and the remote codec. The process is fairly simple if you use the following procedure:

a. Connect to your router using a web-browser.
b. Navigate to https://portforward.com/router.htm
c. Select your router manufacturer from the list.
d. Next, select your router model from the list.
e. Follow the instructions to complete port forwarding. Note: It is necessary to select the device you are configuring in the drop-down "Application" menu (usually step 4). Select "Tieline Bridge-IT", "Tieline-G3" or "Tieline -G5" to suit the codec you are configuring.

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open.
7. Configure the static IP address in your codec using the instructions in the next section. To allow multiple codecs to share a single public static IP address behind a firewall and route the calls correctly, your codecs and the firewall need to be configured similarly to the example diagram which follows. Ensure the port, IP address, Subnet Mask and Gateway settings in your codecs match those configured in your router.



Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address

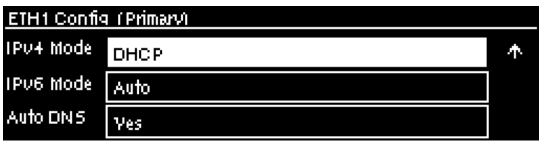
Important Note:
- The most common studio configuration issue is a firewall which blocks the incoming and/or outgoing TCP and UDP ports, or not configuring NAT and port forwarding correctly. The following table lists the firewall ports you need to open for each model of Tieline codec if they are dialing your router at the studio. If the remote codec is also connected to a LAN with a firewall you may also need to open the ports at the remote end of the link to connect successfully.
- Some firewalls require symmetric port configuration. The codec supports configuration of the "send" audio port (codec port at the remote end of the link to which you are sending audio) and "return" audio port (port used by the local codec to receive audio from the remote codec).

Firewall Ports							
Commander G3 / i-Mix G3		Bridge-IT / Bridge-IT XTRA		Merlin and Genie Codec Families		ViA Codec	
TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP
IP1 Session Port: 9002	IP1 Audio Port: 9000	Session Port (Sess): 9002	Audio (Proto): 9000	Session Port: 9002	Audio Port Stream 1: 9000	Session Port: 9002	Audio Port Stream 1: 9000
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alternative Session: 9012	Audio Port Stream 2: 9010	Alternative Session: 9012	Audio Port Stream 2: 9010
Toolbox Software: 5550	Toolbox Software: 5550	Alternative Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060
	SIP Session: 5060	Alternative Web-GUI: 8080	Fuse-IP 8999	Alternative Web-GUI: 8080	Audio Port Stream 4: 9030	Alternative Web-GUI: 8080	SIP Audio: 5004-5054
	SIP Audio: 5004	TLS/SSL 443		TLS/SSL 443	Audio Port Stream 5: 9040	TLS/SSL 443	Fuse-IP 8999
					Audio Port Stream 6: 9050		
					SIP Session: 5060		
					SIP Audio: 5004-5054		
					Fuse-IP 8999		

Configuring a Static Public or Private IP Address in Genie, Merlin and Bridge-IT (v.2.x firmware) Codecs

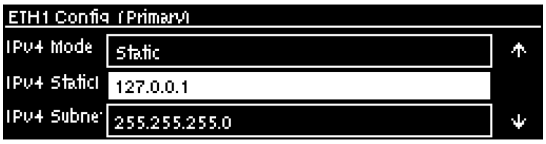
To enter a static IP Address into the codec for NAT:

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Settings** and press .
3. Use the down navigation button to select **LAN** and press .
4. Select **Eth1** and navigate to IPv4 mode and press .



5. DHCP is enabled by default. Select **Static** and press .

6. The Static IP address menu is revealed after DHCP is disabled. Use the navigation buttons to select **IPv4 Static IP** and press .



7. Use the numeric **KEYPAD** to enter the IP address and press to store the setting. Note: use the * or # buttons to enter the periods in the IP address and use the **RETURN** button to delete any numbers already entered.

8. Enter changes to the **IPv4 Subnet** (Subnet Mask) or **IPv4 Gateway** (Default Gateway) in the same way if they are required (check with your network administrator for these settings).
9. After all changes have been made use the navigation buttons to scroll to the top of the menu and select **Apply Setting**, then press the button to save all changes.
10. From the **Home** screen select **Settings > Unit > Eth** in the codec menus to ensure the new static IP address has been entered correctly.

Configuring a Static IP Address in Commander G3 and i-Mix G3 Codecs

To set up a static IP address in Commander G3 and i-Mix G3 codecs select **Menu > Configuration > Advanced > LAN settings > IP Setup > Setup > Static > IP Address > [enter IP address] > press OK > Subnet Mask [enter Subnet Mask] > press OK > Gateway [enter Gateway] > press OK > reboot the codec**.

Record IP Address Details

IPv4 Static IP Address	
IP Address	. . .
Subnet Mask	. . .
Default Gateway	. . .
IPv6 Mode: Manual	**(Bridge-IT, Genie and Merlin codecs only)**
IP Address	: : : : : : :
IPv6 Prefix Size	
IPv6 Gateway	: : : : : : :

Configuring a Static IP Address ViA

1. Press the **HOME** button to return to the **Home** screen, then tap **Settings**.
2. Tap to select **Transport Interfaces** and then **LAN1** or **LAN2**.
3. Tap **Network Settings**.
4. Tap **IPv4 configuration**.
5. Tap to select **Static**.
6. Use the on-screen keyboard to enter the static IP address into the codec, and then tap the **Right Arrow** in the top-right hand corner of the **TOUCH SCREEN**.

7. Enter the **IPv4 Subnet** mask into the codec, and then tap the **Right Arrow**.
8. Enter the **IPv4 Gateway** details as required, then tap **Done** in the top-right hand corner of the **TOUCH SCREEN**.
9. Tap **Save** in the top right-hand corner to reconfigure the codec for static address configuration.

Getting Connected

Once the studio codec is configured you are now ready to receive an incoming call from the remote codec over the internet. Always dial from the field codec to the studio codec over the internet unless the remote codec is assigned a public static IP address and you know this address.

If you dial the studio using a cell-phone data network at the remote site you will not normally experience any firewall or port blocking issues at the remote end of the link using default Tieline ports.

Troubleshooting: How to Determine Where Firewall Port Blocking is Occurring

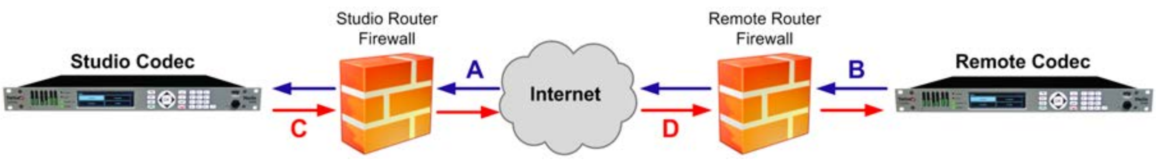
If you find you are unable to either send or receive audio between the studio and remote codecs you can use Tieline's Link Quality reading to diagnose where ports are being blocked. LQ can be displayed on the front LCD screen of Tieline's Bridge-IT, Merlin and Genie codecs by selecting **Cxns**, then select the connection you want to view and press the **OK** button. LQ readings are also displayed on the home screen of all Commander and i-Mix G3 codecs.

Link Quality (LQ) Readings

Send and Return LQ numbers help you to determine if a problem is occurring at either end of a connection. For example, on an IP connection the Return LQ reading represents the audio being downloaded from the network locally (i.e. audio data is being sent by the remote codec). Conversely, the Send LQ reading represents the audio data being sent by the local codec (i.e. being downloaded by the remote codec). To ensure a stable connection, try to maintain a reliable reading of 80 or higher for both the **Send** and **Return** LQ reading.

Important Note:
- The **Return** link quality reading is the same as the Local (**L**) setting displayed in early software versions on a G3 codec.
- The **Send** link quality reading is the same as the Remote (**R**) setting displayed in early software versions on a G3 codec.

Diagnosing Port Blocking via the Studio Codec LQ
If the studio codec **Return** LQ reading is **01** then incoming audio from the remote codec is being blocked by a firewall at either point A or B in the following diagram. If the studio codec **Send** LQ reading is **01** then outgoing audio from the studio is being blocked by a firewall at either point C or D in the following diagram.



Diagnosing Port Blocking via the Remote Codec LQ
If you attach your Tieline codec at the remote site to a LAN with access to the internet you can often dial and connect to the studio without any problem. It is less likely that a firewall will block outgoing TCP and UDP ports. However, if there is a firewall at the remote site it may block incoming data packets from the studio.

The principle is the same at the remote codec for diagnosing blocked ports. If the remote codec **Return** LQ reading is 01 then incoming audio from the studio codec is being blocked by a firewall at either point C or D in the preceding diagram. If the remote codec **Send** LQ reading is 01 then the outgoing audio from the remote codec is being blocked by a firewall at either point A or B in the preceding diagram.

Troubleshooting TCP Port Blocking

Error messages on the codec screen can help to diagnose TCP port blocking.

1. **"Connection Refused"** usually means that the firewall is configured correctly but the codec is not using the expected port. For example, the firewall is set up to forward via port 9002 but codec is 'listening' to port 10,000. "Connection Refused" is not normally shown if the firewall is not configured correctly because a firewall will by design silently drop any forwarding requests to ports that it doesn't have open (see next point). Note: "Connection Refused" will also be displayed if the Commander G3 or i-Mix G3 codec you are calling is already connected.
2. **"Connection Timeout"** can mean one of two things:
   - The firewall is not configured correctly and the attempted codec connection is being silently dropped, e.g. a remote codec is dialing to port 9002 but the studio firewall port forwarding is not configured.
   - The UDP port is not port forwarded correctly. Tieline codecs send test data during connection establishment to make sure that the audio path is configured correctly; if this process fails then it will also result in a "Connection Timeout".

How do I determine which end is blocking data flow?

Tieline test codec firewalls have the default Tieline TCP and UDP ports open. You can dial into these test codecs (or other codecs you know are configured correctly) from your recently configured studio and remote codecs and use the LQ readings to diagnose whether your studio or remote codec firewall is blocking your data packets. If one codec connects OK and the other one doesn't, then you will know which end is likely to be causing the problem. As an example:

1. Dial from site 1 to a Tieline test codec.
2. Dial from site 2 to Tieline test codec.

If both of these connect successfully then the "outbound" TCP path for session data is OK, and the inbound UDP audio path is OK.

3. Dial to site 1 from a codec you know is configured correctly.
4. Dial to site 2 from a codec you know is configured correctly.

If either of these calls fail then TCP and/or UDP inbound data is being blocked on the failed connection (see "Troubleshooting TCP Port Blocking" previously).

Testing your Codec

- Visit the Tieline website at www.tieline.com and select "Support" and then "Test Lines" for a list of test IP codec addresses you can use to verify your codec is configured correctly.
- See Testing IP Network Connections for more IP test information.

Learning More About IP Networks

For more IP network information please see the section titled Understanding IP Networks which discusses:

- Private versus public IP addresses.

- Static versus DHCP assigned IP addresses.
- Network Address Translation (NAT), port forwarding and firewalls.

34.2 Understanding IP Networks

Types of IP Addresses Available

	Type of IP Address	How the IP Address is Allocated	Description
Public	Static Public IP Address	Internet Service Providers (ISPs)	ISP's allocate a static public IP address to allow network devices to communicate with each other over the internet. It works like a public telephone number and will allow your remote codec to call your studio codec over the Internet.
	Dynamically Assigned Public IP Address	Internet Service Providers (ISPs)	ISP's usually allocate dynamically (automatically) assigned public IP addresses to allow network devices to communicate with each other over the Internet. (Not recommended for studio installations because each time you connect to your ISP the IP address can change).
Private	Dynamically Assigned Private IP Address	DHCP Server/Router on your own private LAN network.	A DHCP server-allocated IP address that is automatically assigned to a device on a LAN to allow it to communicate with other devices and the internet. This address can change each time a device connects.
	Static Private IP Address	LAN Administrator	A network administrator-allocated static address which is programmed into a device to allow it to connect to a LAN. Often a security measure to only allow access to devices approved by a network administrator.

Obtaining Public IP Addresses

To send audio streams over the public internet you need to use a public IP address assigned to you by your ISP (Internet Service Provider).

A public IP address is like your public telephone number and allows you to be contacted over the internet in much the same way people dial your public telephone number. They come in two forms; dynamic (DHCP) and static. Most ISPs assign a dynamic public IP address by default, which can often change without you knowing. This is suitable for a quick demo of your Tieline codec, but for a permanent installation you will need to request a permanent static public IP address.

Once the Static Public IP address is assigned to your internet connection (router) at the studio you need to create a link between the public IP address and your codec's private IP address on the LAN. This is called Network Address Translation.

Depending upon how your network is configured, it may also be possible to simply connect your Tieline codec directly into your ADSL modem/router and receive a public address from the router.

Private LAN IP Addresses

By default your Tieline codec will normally be automatically assigned a private IP address when you connect it to a typical router over a LAN.

Private IP Addresses are associated with LANs and normally reside behind a firewall and are not visible to the internet. They are generally in the ranges: 10.0.0.1 – 10.255.255.255, 169.254.0.0 – 169.254.255.255, 172.16.0.0 – 172.31.255.255 and 192.168.0.0 – 192.168.255.255 and are assigned by network DHCP servers and routers.

These IP Addresses are generally assigned for a predefined period (known as a lease) by your network's DHCP server or router. This IP address will generally expire after the lease period. DHCP assigned IP Addresses may also change if the device is disconnected for lengthy periods or if power to the device is turned off and back on. As a result, it is advised that you make this IP address permanent by assigning it as a Static DHCP IP Address. This will ensure you are able to always forward incoming audio packets to your codec using the same private IP address at the studio using port forwarding (see the section on port forwarding for more details). Consult your Network Administrator if you are unsure how to do this.

Network Address Translation (NAT)

Network Address Translation (NAT) is a method of connecting multiple devices to the internet using one public IP address.

The best way to explain NAT is to use the example of a phone system at an office that has one public telephone number and multiple extensions. This type of telephone system allows people to call you on a single public telephone number and performs the translation and routing of the public number to a particular private extension. Similarly, in order to receive an IP call from a remote codec over the public internet, the same network address translation principle applies. NAT and port forwarding allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN.

The relationship between public and private IP addresses and NAT is displayed in the following diagram and the following section explains port forwarding configuration in more detail.

Port Forwarding: Tieline TCP and UDP Port Settings

For your Tieline Codec to communicate over the public internet an IP Address alone is not sufficient. In TCP/IP and UDP networks the codec port is the endpoint of your connection. Ports are doorways for IP devices to communicate with each other. Picture a house and imagine the front door is the entry point represented by a public or private IP address. Then you want to get to several codecs in different rooms of the same house and ports represent the doors to each of those rooms. In principle this is how port addressing works.

For example, several codecs may dial into your studio using the same public static IP address. In this situation it is necessary to configure codec 'programs' with audio streams using different audio ports for discretely routing each incoming and outgoing audio stream. By doing this your studio's network routers know where IP packets for each audio stream should be routed, i.e. to which codec and respective audio outputs.

When data packets are received from remote codecs at a particular public IP address, port information is translated from data packets to ensure the correct packets are sent to the correct studio codecs. This process is performed by PAT (Port Address Translation), which is a feature of NAT (Network Address Translation) devices.

Tieline codecs use TCP ports for setting up the communication session and UDP ports for streaming audio. While TCP ports are generally open, UDP ports are generally blocked by network devices which contain firewalls and will stop you delivering your audio. Depending on the codecs you are using, you need to configure your firewall to allow TCP and UDP protocols to pass through the ports listed in the table below.

34.3 Tips for Creating Reliable IP Connections

The following 10 tips are provided to help obtain the best possible IP connection between two codecs, without paying for Quality of Service (QoS).

1. Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the internet 'backbone'. Wikipedia lists the major service providers that make up the internet backbone at: http://en.wikipedia.org/wiki/internet_backbone. In Australia Telstra is equivalent to one of these service providers.
2. You will get the best quality connection if both the local (studio) and remote codecs use the same ISP. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.
3. Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. Domestic plans are often speed-limited or "shaped" when usage exceeds a predefined limit. These plans are cheap but they are dangerous for streaming broadcast audio.
4. Ensure that the speed of the connection for both codecs is adequate for the job. The minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.
5. Use good quality equipment to connect your codecs to the internet. (Tieline successfully uses Cisco® switching and routing equipment.):

- If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.
- If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. Tieline has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!
- If using a wireless connection ensure that the antenna signal strength received is strong. The type of antenna used and the amount of output gain also affects connection quality.

Important Note: You should be able to stream audio between two codecs on your LAN and get high send/return 'link quality' readings of around 99. If you see less than this then get a network engineer to investigate the issue.

6. Once your internet connection is installed at the studio check that the connection performance is approximately what you ordered and are paying for. A connection can perform below advertised bit rates if:

- There is an error in ISP configuration;
- There is an error in modem configuration;
- There is a poor quality line between the studio and the exchange;
- There are too many phones or faxes connected to the phone line; or
- Line filters have been connected incorrectly.

7. Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organization has network equipment and engineers that can implement and manage quality of service (QoS) on its network.
8. Use UDP as the preferred audio transport protocol.
9. When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate. IP headers require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower.
10. Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection.

Important Note: Be careful when using cell-phone connections at special events where thousands of people have mobile phones. This can result in poor quality connections and audio drop-outs if cell-phone base stations are overloaded.

Wi-Fi Connectivity Tips

- Don't share a router for other purposes when you are streaming to ensure you have the most bandwidth possible.
- The 5GHz band handles interference better and tends to be less congested than 2.4GHz.
- Try to minimize your distance from the Wi-Fi Access Point if possible. This can make a huge difference in reliability.
- You may be able to increase your speed by switching to a less busy channel. Download a wireless channel analyzer app such as Wifi Analyzer for Android or a desktop program such as NirSoft's Wi-FiInfoView for Windows. Macs have the tool built in. These programs show each channel on each Wi-Fi frequency, and the ones nearby networks are using. If you discover you're on a crowded channel, you can manually change it. Type your router's IP address into your web browser. (The IP address is usually on the back of the router, or you can Google your router's model. You can also use http://setuprouter.com/). Enter your username and password, then click to view Wi-Fi settings and select the channel recommended by the Wi-Fi analyzer program.

IP Connection Checklist

Complete the following check list and aim for a score of at least 8 out of 10 before going live.

Number	Check	Result
1	Using a reputable Tier1 ISP that's part of internet backbone.	
2	The same ISP is being used for both codec connections.	
3	The ISP Plan is a Business Plan or equivalent.	
4	The ISP connection speed is adequate.	
5	Equipment is high quality and suitable for media streaming.	
6	The ISP connection speed has been tested and is suitable.	
7	The ISP connection is not shared with other PCs or devices.	
8	UDP is being used as the audio transport protocol.	
9	No more than 80% of ISP connection bandwidth is being used.	
10	There are no wireless connections being used.	

34.4 Testing IP Network Connections

There are a few very simple tools that you can use to test whether a codec can be reached over an IP network.

- Visit http://www.speedtest.net/ to test the upload and download speed of your IP connections and identify your public IP address.
- Visit www.portforward.com and download the port checking application to verify your router's ports are open. Note: Using a port scanner to test a codec will be unsuccessful if you try to scan and the port is already in use, i.e. the codec is connected.
- Visit www.subnetonline.com and use an online port scanner to check for open and closed TCP ports. This site also has numerous other software tools, including an online ping webtool for IPv4, plus TraceRoute and TracePath software tools.

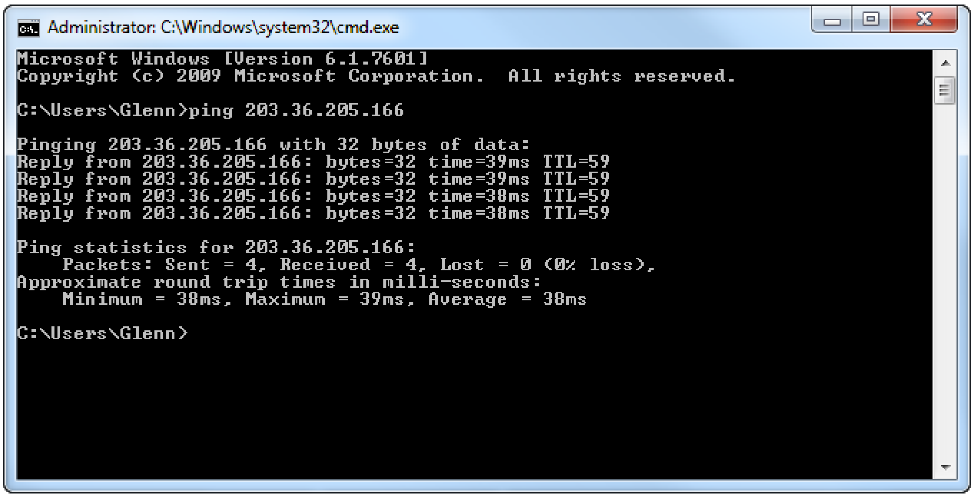
Ping the Codec

A ping test can be used to test whether it is possible to reach a codec or any device over an IP network. A ping test measures:

- The round-trip time of packets.
- Any packet loss.

There are two types of ping tests:

1. **Short test:** sends 4 packets and delivers statistics.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping** and the IP address of the codec you are pinging (i.e. **ping 192.168.0.159**) and press the **Enter** key on your keyboard.
   v. The round trip time of the packets is displayed, as well as any packet loss.



2. **Long test:** sends packets continuously until stopped.
   i. Point to the **start** menu on your PC and click once.
   ii. In the search text box type **Run** and press **Enter.**
   iii. Type **CMD** in the **Run dialog** text box and click **OK**.
   iv. Type **ping**, the IP address of the codec you are pinging, and then **-t** (i.e. **ping 203.36.205.163 -t**) and press the **Enter** key on your keyboard.
   v. Let the test run for several minutes and then press **CTRL C**.
   vi. The round trip time of the packets is displayed, as well as any packet loss for the period of time that the test occurred.

Trace the Route of IP Packets

Another utility available on your PC is traceroute. This tool can be used to determine the route and number of hops that data packets are taking to their destination (codec). This is useful because the more routers that packets traverse, the more latency your connection will have, and the less reliable it will be.

i. Point to the **start** menu on your PC and click once.
ii. In the search text box type **Run** and press **Enter.**
iii. Type **CMD** in the **Run dialog** text box and click **OK**.
iv. Type **tracert**, the IP address of the codec you are contacting (i.e. **tracert 203.36.205.163**) and press the **Enter** key on your keyboard.

34.5 Testing ISDN Connections

To test your ISDN line is working you can dial a standard phone line or your cell-phone number. If the call is successful this verifies the line is active. To verify ISDN data is being sent you can:

- Dial a codec you know is connected to an active ISDN line, e.g. another codec in your network or a Tieline test codec.
- Dial the test ISDN data number provided by your Telco (when available).
- Create a program and perform a loopback test by dialing out on the main ISDN number and receive the call on the auxiliary ISDN number. (**Note:** To create a loopback program create a 2 x Mono or Stereo Peer-to-Peer program and configure a dial only audio stream using your main ISDN number. For the second audio stream create an answer only audio stream connection configured for ISDN. If you dial the connection and can hear the audio you are sending on the return B channel, you have confirmed ISDN data is being sent successfully.
- If you dial using a loopback program and a "disconnect" message appears, you may have the incorrect **Line Type** configured. Change the **Line Type** setting and this should hopefully resolve the issue.

On-Demand ISDN Services

If **Sync** appears for approximately 60 seconds when you connect an ISDN line to the codec and then disappears, or if **Sync** does not appear and you know you are connected to an active ISDN interface, then the line may have ‘On-demand’ enabled by your Telco. To test this you can dial a codec on an ISDN line known to be operational. Dial over ISDN and if **Sync** appears after connecting it indicates the service has now been activated. Disconnect and then dial again. If this dial is successful ‘On Demand’ is enabled. We recommend you contact your network service provider and get them to disable ‘On Demand’ to circumvent any possible connection issues.

34.6 Using Answer Routes for Sessionless ISDN Calls

Tieline Genie Distribution, Merlin, Merlin PLUS and ViA audio codecs support multiple connections using a variety of connection transports such as IP, ISDN and POTS. Tieline codecs support using Tieline session data, which assists with configuration and routing of multiple incoming calls to these codecs. In addition, audio ports can be used to successfully route IP calls to your preferred codec inputs/outputs.

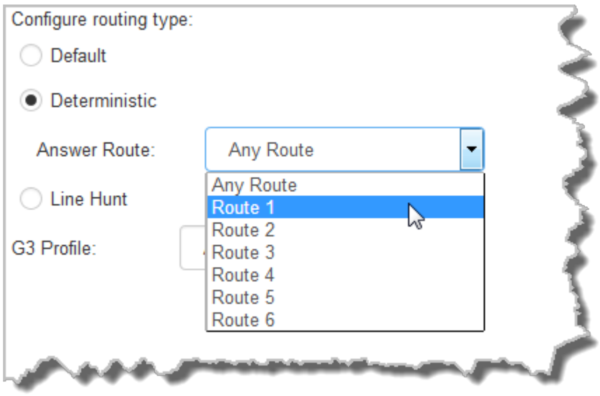
If you are accepting calls from multiple non-Tieline ISDN codecs then you will be making "sessionless" connections which require the codecs at both ends to be configured with the same connection settings. In addition you can use “Answer Routes” and 'site-specific' module settings in Genie Distribution and Merlin PLUS to route incoming calls to specific codec outputs. (Note: Merlin codecs can also be configured to accept 2 ISDN calls from non-Tieline codecs and would use similar settings).

In the following example we will configure two incoming sessionless ISDN audio stream connections (Note: Genie Distribution and Merlin PLUS support up to 4 sessionless ISDN audio streams/connections/configs using 2 ISDN modules and 4 B channels). If you want 2 incoming mono ISDN calls to use inputs/outputs 1 and 2, then use answering audio stream connections 1 and 2 in a 2 x mono peer-to-peer program.

So let's get started. There are 2 or 3 steps to ensure this is configured correctly, depending on whether you want specific incoming calls to always use the same B channels and codec outputs or not.

Step 1: Configure the Answer Route for the two ISDN Audio Stream answering connections in the codec program.

Setup two ISDN audio stream answering connections in your program and use the **Answer Route** setting in the program wizard (as displayed in the following image):

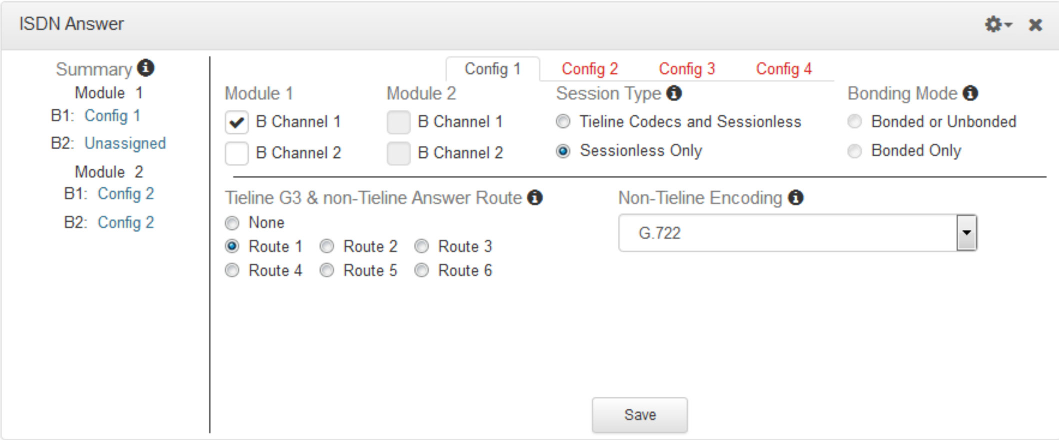


You can use any **Answer Route**, for example **Route 1** for ISDN Audio Stream 1 and **Route 2** for ISDN Audio Stream 2. The **Answer Route** number doesn't have to match the audio stream number because the route you select will be used by the incoming ISDN call. This is similar to how an "extension number" is used to route a phone call.

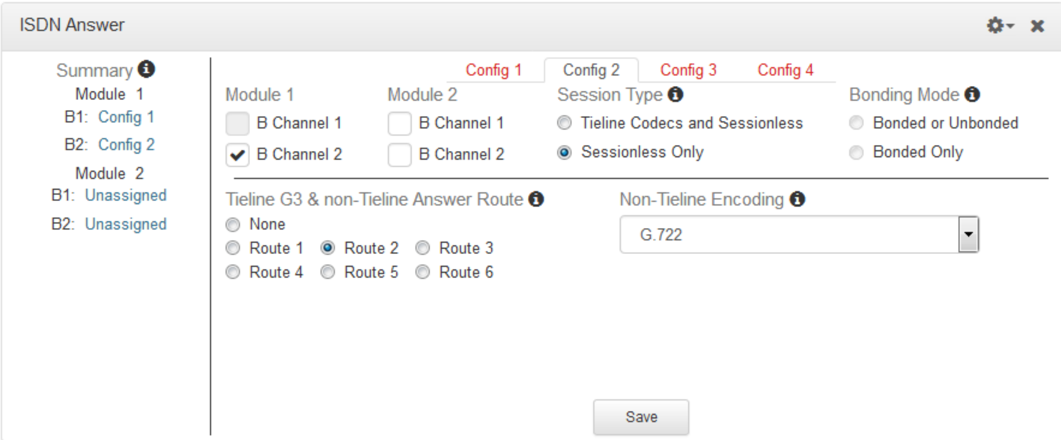
Step 2: Configure the ISDN Module to accept two sessionless ISDN calls.

This can be configured via **Settings > Modules** or use the Toolbox Web-GUI via **Transport > ISDN Answer Modules**.

1. Select **Config 1** and **Sessionless Only** and **Route 1**. Select your preferred algorithm, then click **Save**. This means that Module 1 B channel 1 will answer a sessionless ISDN call using these settings.



2. Next select **Config 2** and **Sessionless Only** and **Route 2**. Select your preferred algorithm, then click **Save**. This means that Module 1 B channel 2 will answer a sessionless ISDN call using these settings.

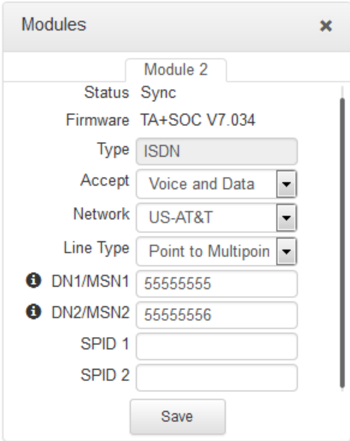


Both ISDN B channels can now answer incoming sessionless ISDN calls. If it doesn't matter which incoming codec call is answered by which B channel then that's all you need to do. If, however, you want each non-Tieline codec to use the same B channel and be routed to the same codec output consistently, you must configure this in the site config for the ISDN module via **Settings > Answering > ISDN Answer Configs**, or via the Web-GUI using the **Settings panel > Modules**.

Step 3: Configuring the module to answer calls from a specific non-Tieline codec consistently.

If a Directory Number (DN) or MSN number is not entered in the codec and multiple B channels are available, the codec may use any channel to answer an incoming call. To ensure calls are routed consistently, enter a DN/MSN number (without the country or area code) as the DN/MSN for a B channel, then only that corresponding B channel will answer an incoming call to that number.

Enter the number for the first B channel into the field for **Directory Number/MSN1**. (This has been allocated **Route 1** previously.) Enter the number for the second B channel into the field for **Directory Number/MSN2**. (This has been allocated **Route 2** previously.) Next, click **Save**.



If codec 1 always uses the first directory number to call then it will always be routed via **Route 1** to the Answering Audio Stream Connection using **Answer Route 1** (configured in step 1). Codec 2 should always use the second directory number and then it will always be routed via **Route 2** to the Answering Audio Stream Connection configured with **Answer Route 2**.

34.7 POTS Connection Tips & Precautions

POTS Operation Precautions

POTS performance is greatly affected by the quality of the line being used. Precautions must be taken to ensure the Tieline codec is not sharing the line with other devices. Please remove these possible sources of interference:

- DSL or ADSL Modems
- Other telephone handsets
- Portable phone base stations
- Unused parallel phone sockets
- Fax machines
- Computer modems
- Burglar alarm systems
- Extension bells
- Call waiting

Call Waiting

Call waiting tones may cause the codec to malfunction. Most phone companies supply call waiting as a feature and you will need to turn it off. Your Telco should be able to provide a number you can dial to disabling the call waiting feature on the line.

Private Branch Exchanges

Avoid connecting the codec to a digital PBX or PABX system, key station, business system or any other local switchboard. It may sometimes be tricky to detect if you are connected to one of these systems, however, as a general guide, these devices require you to dial an additional digit to access the POTS network.

WARNING: Many of these systems are digital and have non standard telephone line operating voltages. If you attach your POTS module to a digital PABX or PBX system permanent damage may result from the high voltage pulses these systems generate. Even if the PBX is not digital, the performance of the codec is unlikely to be as good as a normal POTS line.

If you have no option other than to use a PBX or PABX System, search for a fax machine. The overwhelming majority of fax machines are designed for analog POTS line operation and are normally on an extension optimized for fax machines and data transmission. Substitute a normal phone for the fax machine to verify correct operation. Use a normal phone, not a venue-supplied phone, because this may have characteristics to match the existing PBX/PABX and not a POTS line. After confirming correct phone operation, you can unplug the phone and attach the phone line to the codec.

Check the Length of the Line

It is desirable to have a local loop which is as short as possible, i.e. the line from your location to the local Central Office or Local Exchange. Optimum performance can be expected for lines up to about 2 miles (3 kilometers) in length. Line quality will be reduced over longer distances and the codec can be expected to perform at lower bit rates. Line quality will also be affected by the age, condition and type of cabling used, e.g. plastic insulation or paper insulation, water or moisture entering the cable, age and state of repair of joins.

POTS Party Lines or Stubs

In some countries, it was the practice to have more than one phone service attached to one line - sometimes called a 'Party Line'. As more lines are installed, services are separated but the redundant cabling may remain connected across the line, causing problems with the operation of your codec.

Leakage Problems on the Line

A good line should have an earth isolation of better than ten mega-ohms. If your line is located in an area where water is a problem, ask your Telco to check out the earth leakage.

Equipment Problems at the CO or Local Exchange

Although there are many factors at the Telco end that can cause problems, a problem that does occasionally occur is that the clock on the interface codec to your line is not synchronized to the network. A drifting clock will cause instability and unreliable codec performance. If you suspect that this could be the problem, contact your local Telco.

POTS Exchange Problems

On most good POTS lines, Tieline codecs can achieve a 28.8kbps connection at a line quality of approximately 50% or greater. If you are not able to achieve this level of operation, you may have a problem with your line, or the line at the other end of the connection.

Tips for Successful POTS Operation

1. Take a phone when you are doing a remote broadcast. Connect it to the line you want to use and dial the number to check for any unusual noises. If present, these may be caused by other devices connected to the line.
2. Take an ADSL/DSL filter to all remote locations. ADSL/DSL modems can generate noise on a line which will degrade the performance of your codec. Simply place the ADSL/DSL filter between the POTS line and your codec to remove the interference.
3. Tieline USA has a POTS test codec you can dial on +1-317-913 6911 to facilitate line tests at each end of your connection to diagnose line problems.
4. Tieline recommends that you confirm your broadcast POTS line works well before you try to go live.

34.8 ViA Compliances and Certifications

FCC Compliance Notice

This equipment has been tested and found to comply with the limits for a class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area may cause harmful interference, in which case the user will be required to correct the interference at his/her own expense. There is no guarantee, however, that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment to an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio or TV technician for help.

CAUTION:
Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate this equipment.

FCC Compliance Information

This ViA codec displays all FCC compliance information on the LCD **TOUCH SCREEN**. To view FCC compliance information power up the codec and press the **HOME** button to return to the **Home** screen and then tap **Settings > About & Firmware > Legal** .

Declaration of Conformity

This ViA codec meets the requirements of directives for CE and RCM certifications. Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021.

EN 55 022 Statement

Warning: This is a Class A product. In a domestic environment this product may cause radio interference in which case the user may be required to take adequate measures.

Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline ViA) does not exceed the Class A limits for radio-noise emissions from digital apparatus as set out in the Radio Interference Regulations of the Canadian Department of Communications.

Règlement sur le brouillage radioélectrique du ministère des Communications

Cet appareil numérique (Tieline ViA) respecte les limites de bruits radioélectriques visant les appareils numériques de classe a prescrites dans le Règlement sur le brouillage radioelectrique du ministère des Communications du Canada.

34.9 ViA Declaration of Conformity

Tieline TECHNOLOGY

Declaration of conformity

Declaration of Conformity
according to ISO/IEC 17050-1 and EN 17050-1

Manufacturer's Name	Tieline Pty Ltd ACN 008-962-472 Trading as Tieline Technology
Manufacturer's Address	1/25 Irvine Drive Malaga WA 6090 Australia

declares, that the product

Product Name	ViA
Regulatory Model Number	TLF5300
Product Options	ALL

conforms to the following Product Specifications

EMC	CISPR22- 2006 / EN55022:2006 – Class A CISPR24-2010 / EN55024 2010 EN 61000-4-2 2008 EN 61000-4-3 2008 EN 61000-4-4 2006 EN 61000-4-6 1995 +A1 FCC Title 47 CFR, Part 15 Class A) / ICES-003, Issue 4

Supplementary Information:
The product herewith complies with the requirements of the EMC Directive 2014/30/EU, the R&TTE Directive 1999/5/EC (Annex II), and carries the CE Marking CE accordingly.

This Device complies with Part 15 of the FCC Rules. Operation is subject to the following two Conditions: (1) this device may not cause harmful interference, and (2) this device must accept any interference received, including interference that may cause undesired operation.

Signed:

Name: Rod Henderson

Director, Tieline Pty. Ltd.

Place: Perth, Western Australia

October 12th, 2016

For regulatory topics only:

Tieline Pty Ltd
1/25 Irvine Drive, Malaga WA 6090 Australia

Ph: 61 8 92496688
Fax: 61 8 92496858

Web: www.tieline.com

Please note: Technical documentation required by the conformity assessment procedure is kept at the head office of Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021.

Ph +61 8 9413 2000 or email info@tieline.com (web page www.tieline.com) for repair and warranty information.

34.10 Software Licences

This product uses a combination of proprietary and open source software packages which are licensed as follows.

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o This product includes software developed by Yen Yen Lim and North Dakota State University o This product uses the FreeType library.

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35 ViA Specifications

Input/Output Specifications	
Analog Audio Inputs	3 x Female XLR mic/line inputs
Analog Audio Outputs	2 x Male XLR
AES3 In (via input 1)	1 x female XLR (Channel 1 in; shared with Ch1 analog input & Ch 1 AES42 input)
AES 42 (via input 1)	1 x female XLR (Channel 1 in; shared with Ch1 analog input and Ch1 AES3 input)
Auxiliary Inputs/Outputs	S/PDIF in/out (RCA), or micro-USB in/out, or 3.5mm (1/8") Jack line input
Headphones Out	3 x 6.35mm (1/4") Jacks
Control Port In/Out	Four relay inputs and four opto-isolated outputs for machine control via a DB15 connector.
Mic Phantom Power	Selectable 10V or 48V phantom power on all analog XLR inputs
AES42 Digital Phantom Power	10V nominal
Input Impedance	High Impedance > 5K ohm (line input); approximately 2K ohm (mic input)
Output Impedance	<50 ohm Balanced
Clipping Level	+22dBu (XLR input and outputs)
A/D & D/A Converters	24 bit
Frequency Response at 48kHz	20Hz to 20kHz
THD and Noise (Analog)	< 0.005% (-86dB) at +16dBu unweighted
THD and Noise (Digital)	<0.0001% (-120dB) at -1 dBFS
Analog Signal to Noise	>90dB at +22dBu, unweighted
Sample Frequencies	
IP Sample Frequencies	8kHz, 16kHz, 24kHz, 32kHz, 44.1kHz, 48kHz
Algorithms	
IP	Tieline Music, Tieline MusicPLUS, G.711, G.722, MPEG-1 Layer 2, MP3, LC-AAC, HE-AAC and HE-AACv2, AAC-LD, AAC-ELD, Opus, 16/24 bit aptX Enhanced
IP (PCM uncompressed)	Linear PCM16/24 bit 48 kHz sampling
Data and Control Interfaces	
USB	2 x USB 2.0 Host ports
LAN	2 x 10/100/1000 RJ45 Ethernet ports
Advanced Networking	VLAN tagging (IEEE 802.1q,802.1p)
Serial	RS232 up to 115kbps with or without CTS/RTS flow control via female DB9 connector, can be used as a proprietary data channel
Protocols / RFC support	Tieline, DHCP, SNMP, DNS, HTTP, IGMP, ICMP, VLAN, IPv4/v6, FEC, SIP/SDP (EBU N/ACIP Tech 3326 & 3368 compliant), RTP, RTCP, STUN, SSL, CSRF, RFC5109, RFC5956, RFC5588, RFC4756, RFC3388, RFC5956, RFC 5588, RFC2733, RFC3190
Cellular via module	Optional via module slot
ISDN via module	Optional via module slot
POTS via module	Optional via module slot
Wireless	IEEE 802.11 a/b/g/n with dual band support (2.4 and 5 GHz); support for WPA2-PSK encryption.

Front Panel Interfaces	
Display	4.3 inch TFT Color LCD touch screen
Navigation	Touch screen or 5 button navigation keypad
General	
Dimensions	7 11/64" x 3 5/16" x 8 15/32" 182mm (W) x 84mm (H) x 215mm (D) includes protruding front connectors, rear battery compartment lugs and rubber feet
Weight	1.62 kg (with battery)
Power	Li-ion internal battery or external 12VDC 3A Power Supply (4 pin XLR power inlet accepts 9VDC - 14VDC)
Operating Temp.	0°C to 40°C (32°F to 104°F)
Humidity Operating Range	15% ≤RH ≤80% (0 to 35°C/32°F to 95°F), non-condensing
SD/SDHC Card Slot	Supports SDHC Flash Cards up to 32GB capacity
Battery	
Battery	Rechargeable Li-ion battery pack RRC2057; 7.5 V, 6.4 AH, 8A; Fast charging; 240g
Battery Operation	Up to 6.5 hours (depending on the connection type and power saving modes configured)
Battery Storage Temperature	Max: -20°C to +50°C (-4°F to 122°F) Recommended: -20°C to 25°C (-4°F to 77°F)
External Battery Charger	Contact RRC Power Solutions at http://www.rrc-ps.com
Battery Charge Time (inside codec with power supply attached)	Charge time with maximum charge current < 4hrs at 25°C
Battery Life Expectancy	Minimum 300 charge/discharge cycles as per manufacturer's charge/discharge recommendations

36 Appendix A: RS232 and Control Port Wiring

Relays

The codec uses a DB15 connector to facilitate use of four CMOS solid state relays for the control of equipment, consisting of four relay closures and four opto-isolated outputs.

Inputs

The input signal is referenced to chassis ground, i.e. the ground reference terminal on the connector is connected the chassis. The input device is a high impedance CMOS device with a 330 ohm pull-up resistor to +5 volts.

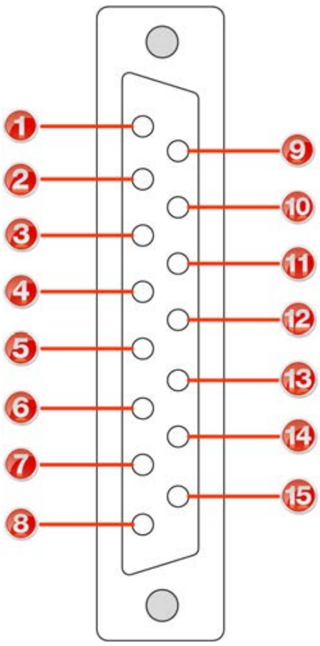
Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

Outputs

CMOS field effect transistors switch a low impedance path between the two pins when activated. These are opto-isolated and floating above ground. It is important to current-limit the source as damage will result where the current exceeds 100mA peak-to-peak. No more than 48 volts peak-to-peak should be used as a safety precaution. The resistance of the CMOS element is approximately 25 ohms in the ON state.

Control Port Pin-outs

A closing contact across Inputs 1-4 to Ground will provide a closing contact on the remote codec Outputs 1 to 4. If your codec supports multi-unicast connections to multiple codecs, a contact closure will appear on each of the compatible (non-G3) remote codecs' corresponding contacts. I.e. Input 1 shorted, Output 1 contacts on all connected codecs closed.



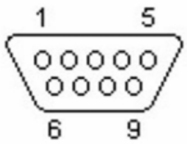
Female DB-15
Codec Connector

Pins	Pin Function
1	Ground
2	Output 4
3	Output 3
4	Output 2
5	Output 1
6	Ground
7	Input 3
8	Input 1
9	Output 4
10	Output 3
11	Output 2
12	Output 1
13	Ground
14	Input 4
15	Input 2

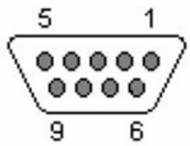
Important Note: For more information about how to program relay operations with a PC using the Toolbox Web-GUI, please see [Creating Rules](#).

RS232 Pin-outs and Data Connections

Pin	INTERFACE Female DB9 (RS232) DCE	DATA Male DB9 (RS232) DTE
1	No Connection	No connection
2	TX Data	RX Data
3	RX Data	TX Data
4	No connection	No connection
5	Signal Ground	Signal Ground
6	No Connection	No connection
7	CTS	RTS
8	RTS	CTS
9	No connection	No connection



DB9 Male Connector Pins



DB9 Female Connector Pins

Important Notes:
- When connecting to G3 codecs over IP, ISDN or POTS, only in-band data is available via the Music and MusicPLUS algorithms.
- It is important to enable serial port flow control as it regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you configure the serial port baud rate to match the setting of the external device to which you are connecting. Ideally the settings on both codecs should match, or you could have data overflow issues.
- Only the dialing codec needs to be configured to send RS232 data. Session data sent from the dialing codec will configure all other compatible codecs (non-G3) when you connect.

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