Tieline®
The Codec Company

Gateway and Gateway-4
IP Codec User Manual

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1   Warnings & Safety Information

1. Both appliance power cables 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 Tieline codec 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 ISDN system, the LAN or the power.

ANY DAMAGE TO A TIELINE PRODUCT CAUSED BY LIGHTNING or an ELECTRICAL STORM WILL VOID THE WARRANTY. Use of this product is subject to Tieline's SOFTWARE LICENSE and WARRANTY conditions, which should be viewed at www.tieline.com/support before using this product.

SAFETY PRECAUTIONS:
- A readily accessible disconnect device shall be incorporated in the building installation wiring.
- Due to the risks of electrical shock, and energy, mechanical, and fire hazards, any procedures that involve opening panels or changing components must be performed by qualified service personnel only.
- To reduce the risk of fire and electrical shock, disconnect the device from the power line before removing the cover or panels.
- This unit has more than one power supply. Disconnect all power supplies before opening to avoid electric shock. Disconnecting one power supply disconnects only one power supply module. To isolate the unit completely, disconnect all power supplies.

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.
- This unit has more than one power supply. Disconnect all power cables before maintenance to avoid electric shock.

HIGH VOLTAGE WARNINGS:
- Any adjustment, maintenance, and repair of the opened instrument under voltage must be avoided as much as possible and, when inevitable, must be carried out only by a skilled person who is aware of the hazard involved.
- Capacitors inside the instrument may still be charged even if the instrument has been disconnected from its source of supply.

GROUNDING:

Before connecting this device to the power line, where required, the protective earth terminal screws of this device must be connected to the protective earth in the building installation.

LINE VOLTAGE:

Before connecting this device 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 device.

WARNING:

HIGH LEAKAGE CURRENT. EARTH CONNECTION ESSENTIAL BEFORE CONNECTING SUPPLY.

If the total leakage current exceeds 3.5 mA, or if the leakage current of the connected loads is unknown, connect the supplementary ground terminal to a reliable ground connection in your facility.

Supplementary ground connection

A supplementary ground terminal is provided on the codec to connect the unit to a ground connection. The ground terminal has an M4 stud with M4 retaining nuts and is compatible with all grounding wires. Remove only **NUT** 2 to connect your ground wire. The ground wire must have a suitable lug. When refitting **NUT** 2 ensure that both **NUT** 1 & **NUT** 2 are correctly tightened to establish and maintain a proper earth connection.
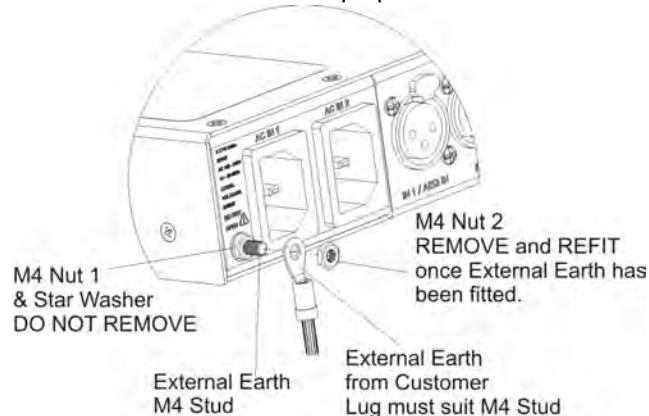
WARNING: To Reduce the Risk of Electrical Shock and Fire

1. This equipment is designed to permit connection between the earthed conductor of the DC supply circuit and the earthing conductor equipment.
2. Equipment connected to the protective earthing of the building installation through the mains connection or through other equipment with a connection to protective earthing - and to a cable distribution system using coaxial cable, may in some circumstances create a fire hazard. Connection to a cable distribution system must therefore be provided through a device providing electrical isolation below a certain frequency range (galvanic isolator, see EN 60728-11).
3. All servicing must be undertaken only by qualified service personnel. There are not user serviceable parts inside the unit.
4. DO NOT plug in, turn on or attempt to operate an obviously damaged unit.
5. Ensure that the chassis ventilation openings in the unit are NOT BLOCKED.
6. Do not operate the device in a location where the maximum ambient temperature exceeds 50°C (122°F), or is below 0°C (32°F).
7. Be sure to unplug both power supply cords from the wall socket BEFORE attempting to service the unit.

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 overcurrent protective device must be incorporated in the building wiring for the power input as follows:
1. Rated 20 A maximum in Canada and the United States.
2. Rated 13 A maximum in the United Kingdom and EU.
3. Rated 16 A maximum in Australia and other countries.

FUSES:

Make sure that only fuses with the required rated current and of the specified type are used for replacement. The use of repaired fuses and the short-circuiting of fuse holders must be avoided. Whenever it is likely that the protection offered by fuses has been impaired, the instrument must be made inoperative and be secured against any unintended operation.

AC units for Denmark, Finland, Norway, Sweden (marked on product):

• Denmark - "Unit is class I - unit to be used with an AC cord set suitable with Denmark deviations. The cord includes an earthing conductor. The Unit is to be plugged into a wall socket outlet which is connected to a protective earth. Socket outlets which are not connected to earth are not to be used!"
• Finland - "Laite on liitettävä suojamaadoituskoskettimilla varustettuun pistorasiaan"
• Norway - "Apparatet må tilkoples jordet stikkontakt"
• Unit is intended for connection to IT power systems for Norway only.
• Sweden - "Apparaten skall anslutas till jordat uttag."

To connect the power connection:

1. Connect the power cables to the power sockets, located on the rear panel of the device.
2. Connect the power cables to the grounded AC outlets.

WARNING: Risk of electric shock and energy hazard. Disconnecting one power supply disconnects only one power supply module. To isolate the unit completely, disconnect all power supplies.

CHINESE SAFETY WARNINGS:

本设备有两个电源供电，未避免电击危险，操作时需要加倍小心。只有当这两个电源完全断开时才可以安全操作

Translation of previous Chinese safety warning: This unit has two power supplies, please be very careful when using, disconnect two power supplies before maintenance to avoid electric shock.

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 metres.

JAPANESE SAFETY WARNINGS:

必ず接地接続を行って下さい

Translation of previous warning: Provide an earthing connection.

接地接続は必ず、電源プラグを電源につなぐ前に行って下さい。又、接地接続を外す場合は、必ず電源プラグを電源から切り離してから行って下さい。

Translation of previous warning: Provide an earthing connection before the mains plug is connected to the mains. And, when disconnecting the earthing connection, be sure to disconnect after pulling out the mains plug from the mains.

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.

Statement for Class B VCCI-certified Equipment:

この装置は，クラスB情報技術装置です。この装置は，家庭環境で使用することを目的としていますが，この装置がラジオやテレビジョン受信機に近接して使用されると，受信障害を引き起こすことがあります。取扱説明書に従って正しい取り扱いをして下さい。　　　VCCI－B

Translation of previous Class B VCCI Statement: This is a Class B product based on the standard of the Voluntary Control Council for Interference by Information Technology Equipment (VCCI). If this is used near a radio or television receiver in a domestic

environment, it may cause radio interference. Install and use the equipment according to the instruction manual.

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.

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. Tieline takes no responsibility for any damage to equipment attached to the codec.

2   French Warnings & Safety Information

Les deux câbles d'alimentation de l'appareil doivent en être retirés pour le débrancher de l'alimentation.

MISE EN GARDE RELATIVE À L'ORAGE ET AUX ÉCLAIRS:

N'UTILISEZ PAS les codecs Tieline dans des conditions d'orages et d'éclairs. Utiliser un codec Tieline pendant un orage pourrait vous faire subir une blessure. Cela peut entraîner des blessures corporelles qui, dans des cas extrêmes, pourraient s'avérer mortelles. Des dispositifs de protection peuvent être installés sur la ligne, cependant, en raison des tensions et des niveaux d'énergie extrêmement élevés associés à la foudre, ces dispositifs peuvent ne pas offrir de protection ni aux utilisateurs ni au codec Tieline, ni aux équipements qui y sont connectés.

Des coups de foudre secondaires peuvent se produire. Ces coups de foudre secondaires sont induits par la foudre et produisent également des courants et des niveaux d'énergie dangereusement élevés. Il vous suffit d'être à proximité d'un objet frappé par la foudre pour que cela vous causes des blessures corporelles ou endommage l'équipement. Par exemple, si vous êtes situé à proximité d'une tour d'éclairage dans une installation sportive, de points d'eau et de drains sur des terrains de golf, vous pourriez être touché par ces coups de foudre secondaires.

Des dommages aux personnes et aux codecs Tieline peuvent survenir pendant un orage, même si le codec est éteint alors qu'il reste branché à un réseau local ou à l'alimentation.

TOUT DOMMAGE CAUSÉ PAR UN PRODUIT TIELINE PAR UN ÉCLAIR ou une TEMPÊTE ÉLECTRIQUE ANNULERA LA GARANTIE. L'utilisation de ce produit est soumise à la LICENCE LOGICIELLE et aux conditions de GARANTIE de Tieline, qui doivent être consultées sur www.tieline.com/support avant l'utilisation de ce produit.

CONSIGNES DE SÉCURITÉ:

- Un dispositif de déconnexion facilement accessible doit être incorporé au câblage de l'installation du bâtiment.
- En raison des risques de décharge électrique et des dangers liés à l'énergie, à la mécanique et à l'incendie, toute procédure impliquant l'ouverture de panneaux ou le changement de composants doit uniquement être effectuée par du personnel d'entretien qualifié.
- Pour réduire le risque d'incendie et de décharge électrique, débranchez l'appareil du réseau électrique avant de retirer le capot ou les panneaux.
- Cet appareil dispose de plus d'une alimentation. Débranchez toutes les alimentations avant de les ouvrir pour éviter toute décharge électrique. Le débranchement d'une alimentation ne débranche qu'un seul module d'alimentation. Pour isoler complètement l'appareil, débranchez toutes les alimentations.

MISES EN GARDE RELATIVES À L'ENTRETIEN:

- N'effectuez aucun entretien autre que celui décrit dans les consignes d'utilisation, à moins que vous ne soyez qualifié pour le faire.
- Tous les travaux doivent être effectués par du personnel dûment qualifié.
- Cet appareil dispose de plus d'une alimentation. Débranchez tous les câbles d'alimentation avant l'entretien pour éviter les décharges électriques.

MISES EN GARDE RELATIVES À LA HAUTE TENSION:

- Tout réglage, entretien et réparation de l'instrument ouvert sous tension doit être évité autant que possible et, lorsque cela est inévitable, cela ne doit être effectué que par un homme du métier conscient du danger encouru.
- Les condensateurs à l'intérieur de l'instrument peuvent encore être chargés, même si l'instrument a été débranché de sa source d'alimentation.

MISE À LA TERRE:

Avant de brancher cet appareil au réseau électrique, le cas échéant, les vis de la borne de terre de protection de cet appareil doivent être connectées à la terre de protection de l'installation du bâtiment.

TENSION DU SECTEUR:

Avant de brancher cet appareil au réseau électrique, assurez-vous que la tension de la source d'alimentation correspond aux exigences de l'appareil. Reportez-vous aux spécifications de l'appareil pour plus d'informations sur la puissance nominale correcte de l'appareil.

MISE EN GARDE:

COURANT DE FUITE ÉLEVÉ. CONNEXION À LA TERRE ESSENTIELLE AVANT DE BRANCHER L'ALIMENTATION.
Si le courant de fuite total dépasse 3,5 mA, ou si le courant de fuite des charges connectées est inconnu, connectez la borne de terre supplémentaire à une connexion de terre fiable de votre installation.

Connexion à la terre supplémentaire

Une borne de terre supplémentaire se trouve sur le codec pour connecter l'appareil à la terre. La borne de terre comporte un goujon M4 avec des écrous de retenue M4 et est compatible avec tous les fils de terre. Retirez uniquement **l'ÉCROU** 2 pour connecter votre fil de terre. Le fil de terre doit comporter une cosse appropriée. Lors du repositionnement de **l'ÉCROU** 2, assurez-vous que **l'ÉCROU** 1 et **l'ÉCROU** 2 sont correctement serrés pour établir et maintenir une connexion correcte à la terre.

MISE EN GARDE : Réduire le risque de décharge électrique et d'incendie

1. Cet équipement est conçu pour permettre de connecter le conducteur mis à la terre du circuit d'alimentation c.c. au conducteur de mise à la terre de l'équipement.
2. L'équipement connecté à la mise à la terre de protection de l'installation du bâtiment par le biais du branchement au secteur ou d'un autre équipement disposant d'une connexion à la terre de protection et à un système de distribution par câble utilisant un câble coaxial, peuvent dans certaines circonstances créer un risque d'incendie. La connexion à un système de distribution par câble doit donc être assurée par un dispositif assurant une isolation électrique en dessous d'une certaine plage de fréquences (isolateur galvanique, voir EN 60728-11).
3. Tous les travaux d'entretien doivent être effectués uniquement par du personnel qualifié. Il n'y a aucune pièce réparable par l'utilisateur à l'intérieur de l'appareil.
4. Ne branchez pas, n'allumez pas ou n'essayez pas de faire fonctionner un appareil visiblement endommagé.
5. Assurez-vous que les ouvertures de ventilation du châssis de l'appareil NE SONT PAS BLOQUÉES.
6. N'utilisez pas l'appareil dans un endroit où la température ambiante maximale dépasse 50 °C (122 °F) ou est inférieure à 0 °C (32 °F).
7. Assurez-vous de débrancher les deux cordons d'alimentation de la prise murale AVANT d'essayer d'entretenir l'appareil.

Mentions spéciales pour les utilisateurs européens :

Pour le branchement européen, sélectionnez un cordon d'alimentation harmonisé au niveau international et marqué « <HAR> », 3 - conducteur, fil de 0,75 mm² minimum, calibré pour 300 V, avec une gaine isolée en PVC. Le cordon doit avoir un capuchon de fiche calibré pour 250 V, 3 A.

Câblage d'interconnexion:

Les câbles de connexion aux interfaces RS232 et Ethernet de l'appareil doivent être certifiés UL de type DP-1 ou DP-2. (Remarque : lorsque l'appareil réside dans un circuit non-LPS)

Protection contre les surintensités:

Un dispositif de protection contre les surintensités de circuit de dérivation répertorié facilement accessible doit être incorporé dans le câblage du bâtiment pour l'entrée d'alimentation comme suit :
1. Calibré pour 20 A maximum au Canada et aux États-Unis.
2. Calibré pour 13 A maximum au Royaume-Uni et dans l'UE.
3. Calibré pour 16 A maximum en Australie et dans d'autres pays.

FUSIBLES:

Assurez-vous que seuls les fusibles prévus pour le courant nominal requis et du type spécifié sont utilisés pour le remplacement. L'utilisation de fusibles réparés et le court-circuit des porte-fusibles doivent être évités. Chaque fois qu'il est probable que la protection offerte par les fusibles a été altérée, l'instrument doit être mis hors service et protégé contre toute utilisation involontaire.

MISE EN GARDE: Risque de décharge électrique et de danger énergétique. Le débranchement d'une alimentation ne débranche qu'un seul module d'alimentation. Pour isoler complètement l'appareil, débranchez toutes les alimentations.

CONSEILS EN MATIÈRE D'ÉCOUTE EN TOUTE SÉCURITÉ

MISE EN GARDE: ÉCOUTER L'AUDIO À DES VOLUMES EXCESSIFS PEUT CAUSER DES DOMMAGES AUDITIFS PERMANENTS. UTILISEZ UN VOLUME A AUSSI FAIBLE QUE POSSIBLE.

Une surexposition à des niveaux sonores excessifs peut endommager vos oreilles, entraînant une perte auditive permanente induite par le bruit (NIHL). Veuillez suivre les directives applicables de l'autorité en matière de santé et de sécurité concernant les limites maximales d'exposition. En règle générale, évitez les périodes prolongées d'écoute de niveaux de pression acoustique (NPA) de 85 dBA ou plus.

Déclaration de fin de durée de vie util

Tieline déclare par la présente que tous les matériaux, composants et produits fournis sont en pleine conformité avec les directives RoHS et DEEE. Ce produit doit être éliminé conformément aux lois et réglementations locales. Étant donné que le produit contient une batterie, il doit être jeté séparément des déchets ménagers. Ne l'incinérez pas, mais apportez-le à une installation de recyclage.

3 German Warnings & Safety Information

Zur Unterbrechung der Stromversorgung müssen beide Stromkabel vom Gerät entfernt werden.

ACHTUNG BEI GEWITTER UND BLITZSCHLAG:

Tieline-Codecs bei Gewitter und Blitzschlag NICHT VERWENDEN. Bei der Verwendung eines Tieline-Codecs während eines Gewitters können Sie verletzt werden. Dies kann dauerhafte Gesundheitsschäden oder im äußersten Fall den Tod zur Folge haben. Die Stromzuleitung kann mit Schutzgeräten versehen werden. Aufgrund der extrem hohen Spannungen und Energien, die bei Blitzschlägen auftreten, bieten diese Geräte jedoch möglicherweise keinen Schutz für die Anwender oder den Tieline-Codec und die an den Codec angeschlossenen Geräte.

Es können Sekundärblitze auftreten. Diese Sekundärblitze werden durch Blitzeinschläge hervorgerufen und erzeugen ebenfalls gefährlich hohe Strom- und Energieniveaus. Sie müssen sich lediglich in der Nähe eines von einem Blitz getroffenen Objekts aufhalten, um eine Verletzung oder einen Sachschaden zu erleiden. Wenn Sie sich z. B. in der Nähe eines Lichtmastes einer Sportanlage oder von Wasserflächen und Abflüssen auf Golfplätzen befinden, können Sie von diesen Sekundärblitzen getroffen werden.

Tieline-Codecs können während eines Gewitters auch dann beschädigt werden, wenn sie ausgeschaltet, jedoch weiter mit dem Netzwerk oder dem Stromnetz verbunden sind.

SCHÄDEN AN EINEM TIELINE-PRODUKT, DIE DURCH BLITZSCHLAG ODER GEWITTER VERURSACHT WERDEN, SIND VON DER GARANTIE NICHT GEDECKT. Die Verwendung dieses Produkts unterliegt den SOFTWARE-LIZENZ- und GARANTIE-Bedingungen von Tieline, die vor der Verwendung dieses Produkts unter www.tieline.com/support eingesehen werden sollten.

SICHERHEITSHINWEISE:

- Die Gebäudeinstallation muss über ein leicht zugängliches Trenngerät verfügen.
- Aufgrund der Gefahr eines Stromschlags sowie der energetischen, mechanischen und Brandgefahr dürfen alle Vorgänge, die das Öffnen des Gehäuses oder den Austausch von Komponenten beinhalten, nur von qualifiziertem Wartungspersonal durchgeführt werden.
- Um die Brand- Stromschlaggefahr zu verringern, trennen Sie das Gerät vom Stromnetz, bevor Sie die Abdeckung oder die Seitenteile entfernen.
- Dieses Gerät hat nicht nur eine Stromversorgung. Trennen Sie alle Stromversorgungen vor dem Öffnen, um einen Stromschlag zu vermeiden. Mit dem Trennen einer Stromversorgung wird nur ein Stromversorgungsmodul getrennt. Um das Gerät vollständig zu isolieren, trennen Sie alle Stromversorgungen.

WARTUNGSHINWEISE:

- Führen Sie keine anderen als die in der Bedienungsanleitung beschriebenen Wartungsarbeiten durch, es sei denn, Sie sind entsprechend qualifiziert.
- Alle Arbeiten sind von entsprechend qualifiziertem Personal auszuführen.
- Dieses Gerät hat nicht nur eine Stromversorgung. Trennen Sie vor Wartungsarbeiten alle Stromkabel, um einen Stromschlag.zu vermeiden.

HOCHSPANNUNGSHINWEISE:

- Jegliche Einstell-, Wartungs- und Reparaturarbeiten an einem unter Spannung stehenden geöffneten Gerät sind nach Möglichkeit zu vermeiden und dürfen, wenn es unvermeidlich

ist, nur von einem Fachmann durchgeführt werden, der sich der damit verbundenen Gefahr bewusst ist.

- Die Kondensatoren im Inneren des Geräts können noch aufgeladen sein, auch wenn das Gerät von der Stromversorgung getrennt wurde.

ERDUNG:

Vor dem Anschluss dieses Geräts an das Stromnetz müssen, falls erforderlich, die Schutzerdungsschrauben dieses Geräts mit der Schutzerde der Gebäudeinstallation verbunden werden.

NETZSPANNUNG:

Bevor Sie dieses Gerät an das Stromnetz anschließen, stellen Sie sicher, dass die Netzspannung den Spezifikationen des Geräts entspricht. Informationen zur Stromaufnahme des Geräts finden Sie in den technischen Daten.

ACHTUNG:

HOHER ABLEITSTROM. ERDUNGSANSCHLUSS VOR DEM ANSCHLUSS DER STROMVERSORGUNG UNBEDINGT ERFORDERLICH.

Wenn der Gesamtableitstrom 3,5 mA überschreitet oder wenn der Ableitstrom der angeschlossenen Lasten unbekannt ist, schließen Sie die zusätzliche Erdungsklemme an einen zuverlässigen Erdungsanschluss in Ihrem Gebäude an.

Zusätzlicher Erdungsanschluss

Der Codec verfügt über einen zusätzlichen Erdungsanschluss, um ihn mit einer Erdung zu verbinden. Der Erdungsanschluss verfügt über ein M4-Gewinde mit M4-Haltemuttern und ist mit allen Erdungskabeln kompatibel. Entfernen Sie zum Anschluss des Erdungskabels nur **MUTTER** 2. Das Erdungskabel muss über eine geeignete Öse verfügen. Achten Sie darauf, dass beim Aufschrauben von **MUTTER** 2 **MUTTER** 1 und **MUTTER** 2 richtig angezogen werden, um eine ordnungsgemäße Erdungsverbindung herzustellen.

ACHTUNG: Die Stromschlag- und Brandgefahr verringern

1. Dieses Gerät ist so konzipiert, dass der Schutzleiter des Gleichstromversorgungskreises und die Erdungseinrichtung verbunden werden können.
2. Geräte, die über den Netzanschluss oder über andere Einrichtungen mit Anschluss an die Schutzerdung an die Schutzerdung der Gebäudeinstallation – und per Koaxialkabel an ein Kabelverteilsystem – angeschlossen sind, können unter Umständen eine Brandgefahr darstellen. Der Anschluss an ein Kabelverteilsystem muss daher über eine Vorrichtung erfolgen, die unterhalb eines bestimmten Frequenzbereichs für eine elektrische Isolierung sorgt (Trennwandler, siehe EN 60728-11).
3. Wartungsarbeiten dürfen nur von qualifiziertem Wartungspersonal durchgeführt werden. Im Inneren des Geräts befinden sich keine Teile, die vom Anwender gewartet werden können.
4. Schließen Sie ein offensichtlich beschädigtes Gerät NICHT an, schalten Sie es nicht ein und versuchen Sie nicht, es zu betreiben.
5. Stellen Sie sicher, dass die Lüftungsöffnungen des Geräts NICHT VERSPERRT sind.
6. Betreiben Sie das Gerät nicht an einem Ort, an dem die Umgebungstemperatur über 50 ° C (122 °F) oder unter 0 °C (32 °F) liegt.
7. Ziehen Sie beide Netzkabel aus der Steckdose, BEVOR Sie Wartungsarbeiten am Gerät vornehmen.

Besondere Hinweise für europäische Anwender:

Wählen Sie für den Anschluss in Europa ein international vereinheitlichtes, mit „<HAR>" gekennzeichnetes 3-adriges Netzkabel mit einem Leiterquerschnitt von mindestens 0,75 mm², einer Nennspannung von 300 V und PVC-Isolierummantelung. Das Kabel muss einen angespritzten Stecker mit einer Nennbelastbarkeit von 250 V, 3 A haben.

Verbindungskabel:

Kabel für den Anschluss an die RS232- und Ethernet-Schnittstellen des Geräts müssen UL-zertifizierte Kabel des Typs DP-1 oder DP-2 sein. (Hinweis: bei einer Nicht-LPS-Schaltung)

Überstromschutz:

Für die Stromversorgung muss die Gebäudeinstallation über ein leicht zugängliches, handelsübliches Zweigstromkreis-Überstromgerät verfügen, das folgenden Spezifikationen entspricht:

1. Nennstrom von maximal 20 A in Kanada und den Vereinigten Staaten;
2. Nennstrom von maximal 13 A in Großbritannien und der EU;
3. Nennstrom von maximal 16 A in Australien und anderen Ländern.

SICHERUNGEN:

Stellen Sie sicher, dass nur Sicherungen mit dem erforderlichen Nennstrom und des angegebenen Typs zum Austausch verwendet werden. Es dürfen keine reparierten Sicherungen verwendet und keine Sicherungshalter kurzgeschlossen werden. Wenn der Sicherungsschutz mit einiger Wahrscheinlichkeit beeinträchtigt ist, muss das Gerät außer Betrieb genommen und gegen unbeabsichtigte Inbetriebnahme gesichert werden.

ACHTUNG: Stromschlaggefahr. Mit dem Trennen einer Stromversorgung wird nur ein Stromversorgungsmodul getrennt. Um das Gerät vollständig zu isolieren, trennen Sie alle Stromversorgungen.

HINWEIS ZUM SICHEREN MUSIKHÖREN

ACHTUNG: DAS HÖREN BEI ZU HOHER LAUTSTÄRKE KANN DAS GEHÖR DAUERHAFT SCHÄDIGEN. VERWENDEN SIE EINE MÖGLICHST GERINGE LAUTSTÄRKE.

Eine übermäßige Belastung durch zu hohe Schallpegel kann Ihr Gehör schädigen und eine dauerhafte Lärmschwerhörigkeit zur Folge haben. Halten Sie sich an die geltenden Richtlinien der Gesundheits- und Sicherheitsbehörden zu Expositionsgrenzwerten. Als Faustregel gilt: Vermeiden Sie es, über längere Zeit Schalldruckpegel ab 85 dBA oder höher zu hören.

Ende der Nutzungsdauer

Tieline erklärt hiermit, dass alle gelieferten Materialien, Komponenten und Produkte die RoHS- und die WEEE-Richtlinie vollständig einhalten. Dieses Produkt muss gemäß den örtlichen Gesetzen und Vorschriften entsorgt werden. Da das Produkt eine Batterie enthält, darf es nicht zusammen mit dem Hausmüll entsorgt werden. Verbrennen Sie es nicht, sondern führen Sie es dem Recycling zu.

4 How to Use this User Manual

Introduction to the Gateway and Gateway 4 User manual

This user manual documents how to use Tieline's Gateway and Gateway 4 codecs. This user manual outlines how to configure and monitor connections over IP WANs like the internet and ISDN. To configure AoIP streaming over AES67 networks, please download the Gateway and Gateway 4 AoIP User Manual from www.tieline.com.

The Gateway has a different front and rear panel to the Gateway 4. A table outlining the differences between the different codec models is placed at the head of sections affected by hardware and/or software differences between the two codec models.

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.

Codec Model Note: Informative notes highlighting differences between codec models, e.g. Gateway 4 versus Gateway.

Typographic Conventions

- Codec software elements are in Arial bold, e.g. **Contacts**
- Codec hardware elements are in bold Capitals, e.g. **KEYPAD**

Help Button

Press the [*i*] (information/help) button when navigating codec menus to display a dialog suggesting the actions which can be performed from within the current menu.

5   Glossary of Terms

AES3	Official term for the audio standard referred to often as AES/EBU
AES67	A technical standard for audio over IP and audio over Ethernet (AoE) interoperability developed by the AES. It is a layer 3 protocol suite facilitating interoperability between IP-based audio networking systems such as WheatNet-IP, RAVENNA, Livewire, and Dante.
AES/EBU	Digital audio standard used to carry digital audio signals between devices
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.
Destinations	An AoIP Destination in the codec is an AES67 stream received by the codec from an AES67 LAN.
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 or Diffserv Value is a field in an IP packet header for prioritizing data when traversing IP networks. This is often used in AES67 streaming.
Failover	Method of switching to an alternative backup Audio Stream if the primary connection is lost.
GUI	Graphical User Interface
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
IGMP	A communications protocol used by hosts and adjacent routers on IPv4 networks to establish multicast group memberships to IPv4 routers.
IGMP snooping	The process of listening to IGMP network traffic for delivery of IP multicasts. Network switches with IGMP snooping maintain a map of which links need which IP multicast transmission. Multicasts may be filtered to conserve bandwidth on links.
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
IPv4	Internet Protocol version 4 is the most widely deployed version of the protocol and is widely used on the Internet and on local area networks
IPv6	Internet Protocol version 6 is the most recent revision of the Internet Protocol and is intended to replace IPv4 eventually.
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.
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.
NMOS	Networked Media Open Specifications (NMOS) delivering Discovery and Registration to ensure that parts of a networked media system can find each other. NMOS also provides connection management and audio channel mapping to device I/O channels.
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
Primary Leader Clock	The primary source of synchronization for clock distribution via PTP.
PSTN	Public switched telephone network which is another term for POTS (see previous)
PSU	Power Supply Unit
PTP	The general class clock distribution protocol standardized in IEEE 1588-2002, IEEE 1588-2008 and IEEE 802.1AS-2011. PTP syncing requires a leader clock source (often an installed PTP primary leader clocking device) with clocking replicated on synced devices.
QoS (Quality of Service)	Priority given to different users or data flows across managed IP networks. This generally requires a Service Level Agreement (SLA) with a Telco or ISP to use over WANs.
RTP	A standardized packet format using UDP/IP networking for sending audio and video data streams and ensures consistency in the delivery order of voice data packets.
RTP stream	An RTP stream is a sequence of RTP packets with media data sent at regular interval. A stream may contain multiple channels. There may be multiple media streams per RTP session.
Runtime (edits)	Configuration changes which have not yet been saved, e.g. Matrix Editor edits.
SAP	SAP (Session Announcement Protocol) is used to distribute SDP descriptions to receivers, enabling simplified connection management for multicast streaming.
SDP	SDP 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
SIP	SIP is a common protocol which works with a myriad of other protocols to establish connections with other devices to provide interoperability
SIP URI	A SIP URI is a URI used by SIP to identify user agents. SIP URI take the form sip:<user>@<domain> or sips:<user>@<domain>.
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.

Sources	An AoIP source in the codec is an AES67 stream sent by the codec to an AES67 LAN.
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	TCP 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 muliticast servers to ensure data packets have a finite life and don't cause congestion over networks. Each time a packet passes through a router it reduces by 1 until it reaches zero, at which point a router will no longer pass the packet.
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 using Internet Protocol to enable audio to be intelligently distributed to devices across scalable networks

6   Overview of Gateway Codec Family

This user manual documents how to use Tieline's Gateway and Gateway 4 codecs. The Gateway has a different front and rear panel compared to the Gateway 4 and there are also some minor software differences between the two codec models.

Gateway

The Tieline Gateway is the most powerful DSP-based 1RU IP codec enabling transport of multiple channels of mono or stereo audio across any QoS-enabled IP network, including T1 and T3 connections and private WANs with MPLS. The Gateway streams up to 16 IP audio channels with support for AES67, ST2110-30, AES3 and analog I/O as standard.

The codec can be licensed for between 8 and 16 channels of audio and has two standard versions:

- Gateway supporting 8 Channels in/out (8 Mono or 4 Stereo)
- Gateway supporting 16 Channels in/out (16 Mono or 8 Stereo)

The codec also supports a flexible upgrade path that allows you to buy a Gateway with 8 channels and upgrade the codec over time as needs change or your network expands. See Purchase and Install Channel Licenses for more details about how to order licenses and upgrade a Gateway codec to support more channels.

Gateway 4

The Gateway 4 has a different front and rear panel and is designed for solutions requiring up to 4 audio streaming channels with advanced redundancy features and remote configuration and control. It supports 4 full-duplex audio channels* in 1RU and is backward compatible with all Tieline IP codecs. The Gateway 4 provides two stereo connections, or one stereo and two mono connections, or up to 4 mono connections.

The Gateway 4 codec supports 4 channels only and is not upgradable to support more channels.

Overview of this User Manual

Use this manual to learn how to:

- Configure codec 'programs' and connections.
- Adjust audio and connection settings within the codec.

To configure AoIP streaming over AES67 networks, please download the Gateway and Gateway 4 AoIP User Manual from www.tieline.com.

Codec Features

To view a list of the codec specifications for each codec see:
- Gateway Specifications
- Gateway 4 Specifications

7   Purchase and Install Channel Licenses

Order 8 Channels Now and Upgrade Later…

Customers can order a Gateway 8 and immediately have access to 8 input and output channels and stream up to 8 mono or 4 stereo streams of IP audio. If you need to expand channel and stream capacity, simply purchase an upgrade license and expand capacity in pairs up to 16 channels in total. Options include:

- Gateway: 8 channels in/out with the following license options or combination thereof:
- 2 Channel Upgrade: TLR6200-LIC1ST
- 4 Channel Upgrade: TLR6200-LIC2ST
- 8 Channel Upgrade: TLR6200-LIC4ST

Contact Tieline or your favorite dealer for pricing and to purchase an upgrade license. To install a license using the codec front panel use the following procedure, or to install using the Toolbox HTML5 Web-GUI see Software License Installation.

Updating Licensing Using TieServer

To update a codec after purchasing a channel license upgrade:

1. Connect the codec to the internet so it can communicate with Tieline's TieServer.
2. Press the **SETTINGS** 🔧 button.
3. Navigate to **System** and press the **OK** button.
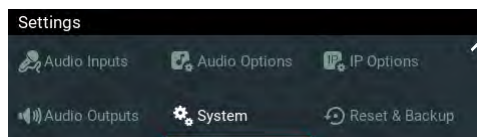
4. Navigate down to **Licenses** and press the **OK** button.
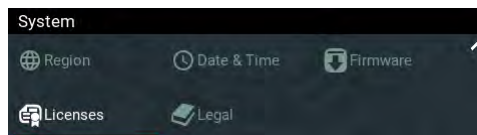
5. Select **TieServer Update** and press the **OK** button to update the codec license through TieServer. The number of channels licensed in a codec is also displayed in this menu.
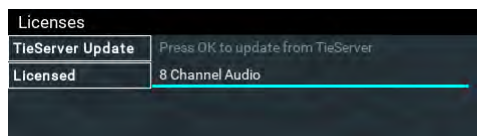
8   Gateway 8/16 Front Panel Controls

The hardware front panel interface features menu navigation buttons, an LCD display with PPM metering and a dialing keypad.

Navigation Buttons

The codec has four arrow shaped navigation buttons for navigating codec menus and adjusting levels, and an **OK** button for selecting menu items.

Dialing Keypad

The keypad has alpha-numeric buttons, plus star and hatch (pound) buttons, which can be used to enter contact and program information into the codec. Note: a virtual keyboard appears for entering alphanumeric characters in fields displayed on the codec **LCD SCREEN**.

Operation Button Description

	Features	Operation Button Descriptions
	Return Button	Press to move back through menus & delete characters
F1	Function Button 1	Press to activate codec user functions and relays
F1	Function Button 2	Press to activate codec user functions
	Connect Button	Press to create an IP connection
	Home Button	Press to return to home screen
i	Information Button	Press to view a help menu onscreen
	Settings Button	Press to adjust codec settings
	Disconnect Button	Press to end a connection
	Headphone Button	Press to adjust headphone audio levels

Front Panel LED Descriptions

LED	LED Description
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.
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. Note: the LED flashes until all accounts have been registered successfully when multiple SIP accounts have been added.
Connected	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.
Power	The **POWER LED** indicates power is connected to the codec.

Stereo RTS Headphone Output

The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for audio monitoring.

USB 2.0 Host Port

The USB 2.0 host port supports firmware upgrades, Wi-Fi USB modems and memory sticks.

9   Gateway 8/16 Rear Panel Connections
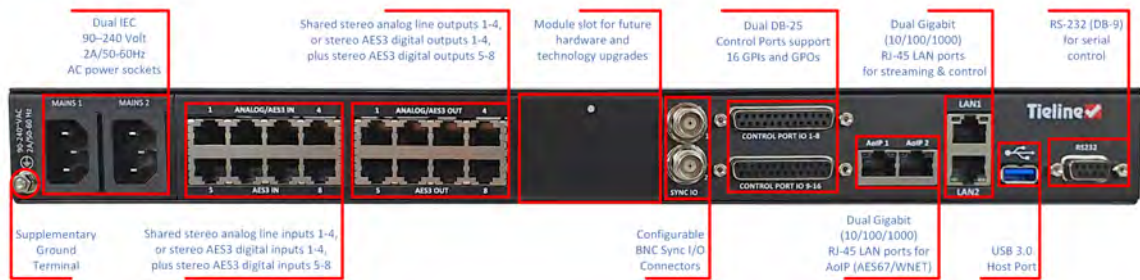
Supplementary Ground Terminal

Supplementary ground terminal for connecting the unit to a ground connection. See Warnings and Safety Information for more details.

Dual Redundant AC Power Inputs

The codec is powered by dual 90-240 volt, 2A/50-60Hz redundant AC power supplies, which use standard IEC connectors.

Shared Analog and AES3 Inputs and Outputs

Balanced **ANALOG/AES3 IN** RJ-45 connectors provide shared stereo analog line inputs 1-4 (analog inputs 1-8), or stereo AES3 digital inputs 1-4 (AES3 inputs 1-8). Plus, connectors 5-8 provide stereo AES3 digital inputs 5-8 (AES3 inputs 9-16). Balanced **ANALOG/AES3 OUT** RJ-45 connectors provide shared stereo analog line outputs 1-4, or stereo AES3 digital outputs 1-4, plus stereo AES3 digital outputs 5-8.

Module Slot

A module slot supports the optional Gateway Euro ISDN module, as well as future hardware and technology upgrades.

Configurable BNC Sync I/O Connectors

Two BNC type **SYNC I/O** connectors for Word Clock and other synchronization with external sources.

Dual DB-25 Control Port Connectors

Dual female DB-25 **CONTROL PORT I/O** connectors support 16 CMOS control inputs and 16 opto-isolated solid-state relay outputs. DB-25 pinouts are available in Appendix A.

Dual Gigabit AoIP Ports

Two Gigabit (10/100/1000) RJ-45 **AoIP** ports for AES67 or optional WheatNet-IP streaming (16 bidirectional mono or 8 bidirectional stereo). Note: Port 2 not operational in early firmware releases.

Dual Gigabit Ethernet Ports

Two Gigabit (10/100/1000) RJ-45 Ethernet **LAN** ports for IP connections. By default, the codec assumes **LAN1** is the primary LAN connection and **LAN2** is the backup LAN connection when in

use. If you are only using one Ethernet port, always use **LAN1**. 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

USB Host Port

Rear panel USB 3.0 host port supports record and playback of audio files.

DB-9 RS-232 Port

A nine pin female **RS-232** serial connection for local and remote control of equipment at either end of the link.

10 Gateway 4 Front Panel Controls

The hardware front panel interface features menu navigation buttons, an LCD display with PPM metering and a dialing keypad.

Navigation Buttons

The codec has four arrow shaped navigation buttons for navigating codec menus and adjusting levels, and an **OK** button for selecting menu items.

Dialing Keypad

The keypad has alpha-numeric buttons, plus star and hatch (pound) buttons, which can be used to enter contact and program information into the codec. Note: a virtual keyboard appears for entering alphanumeric characters in fields displayed on the codec **LCD SCREEN**.

Operation Button Description

	Features	Operation Button Descriptions
↰	Return Button	Press to move back through menus & delete characters
F1	Function Button 1	Press to activate codec user functions and relays
F1	Function Button 2	Press to activate codec user functions
☎	Connect Button	Press to create an IP connection
⌂	Home Button	Press to return to home screen
i	Information Button	Press to view a help menu onscreen
⚒	Settings Button	Press to adjust codec settings
☎	Disconnect Button	Press to end a connection
⌒	Headphone Button	Press to adjust headphone audio levels

Front Panel LED Descriptions

LED	LED Description
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.
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. Note: the LED flashes until all accounts have been registered successfully when multiple SIP accounts have been added.
Connected	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.
Power	The **POWER LED** indicates power is connected to the codec.

Stereo RTS Headphone Output

The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output for audio monitoring.

USB 2.0 Host Port

The USB 2.0 host port supports firmware upgrades, Wi-Fi USB modems and memory sticks.

11 Gateway 4 Rear Panel Connections
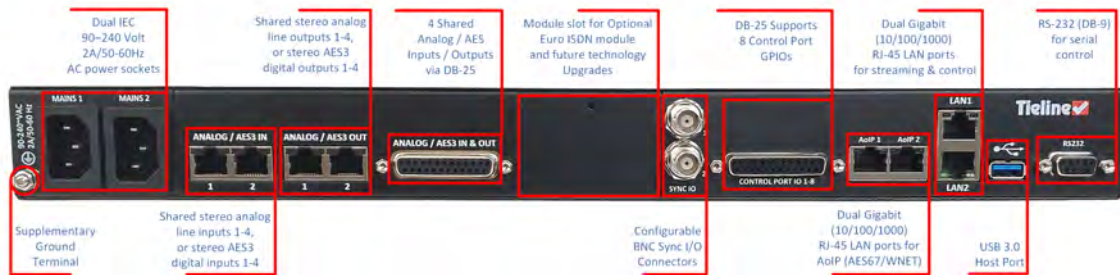
Supplementary Ground Terminal

Supplementary ground terminal for connecting the unit to a ground connection. See Warnings and Safety Information for more details.

Dual Redundant AC Power Inputs

The codec is powered by dual 90-240 volt, 2A/50-60Hz redundant AC power supplies, which use standard IEC connectors.

Shared Analog and AES3 Inputs and Outputs

Balanced **ANALOG/AES3 IN** RJ-45 connectors provide shared stereo analog line inputs 1-4, or stereo AES3 digital inputs 1-4. Balanced **ANALOG/AES3 OUT** RJ-45 connectors provide shared stereo analog line outputs 1-4, or stereo AES3 digital outputs 1-4.

Female DB-25 Shared Analog and AES3 Input/Output Connector

DB-25 connector provides shared stereo analog line inputs 1-4, or stereo AES3 digital inputs 1-4, plus shared stereo analog line outputs 1-4, or stereo AES3 digital outputs 1-4. DB-25 pinouts are available in Appendix C.

Module Slot

A module slot is available to support the optional Gateway Euro ISDN module, as well as future hardware and technology upgrades.

Configurable BNC Sync I/O Connectors

Two BNC type **SYNC I/O** connectors for Word Clock synchronization with external sources.

Female DB-25 Control Port Connector

The DB-25 **CONTROL PORT I/O** connector supports 8 CMOS control inputs and 8 opto-isolated solid-state relay outputs. DB-25 pinouts are available in Appendix A.

Dual Gigabit AoIP Ports

Two Gigabit (10/100/1000) RJ-45 **AoIP** ports for AES67 or optional WheatNet-IP streaming (4 bidirectional mono or 2 bidirectional stereo). Note: port 2 not operational in first firmware release.

Dual Gigabit Ethernet Ports

Two Gigabit (10/100/1000) RJ-45 Ethernet **LAN** ports for IP connections. By default, the codec assumes **LAN1** is the primary LAN connection and **LAN2** is the backup LAN connection when in use. If you are only using one Ethernet port, always use **LAN1**. 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

USB Host Port

Rear panel USB 3.0 host port supports record and playback of audio files.

DB-9 RS-232 Port

A nine pin female **RS-232** serial connection for local and remote control of equipment at either end of the link.

12   Menu Navigation

All front panel codec menus can be launched from the **Home** screen as follows:

	Features	Codec Home Screen Elements
1	Screen Name	The name of the current screen
2	Programs	View and edit Program configurations
3	Cxns	Adjust connection settings and show connections and connection details.
4	Settings	Menus for configuring a wide range of codec settings

Press the **RETURN** ⬅ button to navigate backwards through menus, or press the **HOME** 🏠 button to return to the **Home** screen from any menu. If a complete menu cannot be viewed on a single codec screen, arrows on the right hand side of the screen indicate that the current menu has options below and/or above the visible items. Use the navigation arrows to scroll up and down.
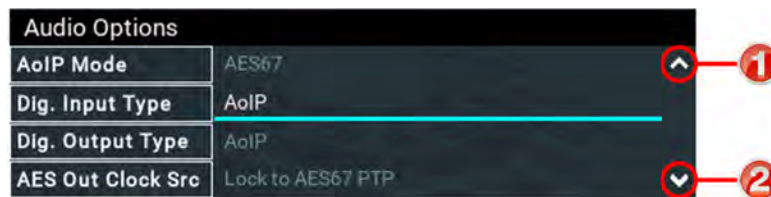
	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

Home Screen Menus
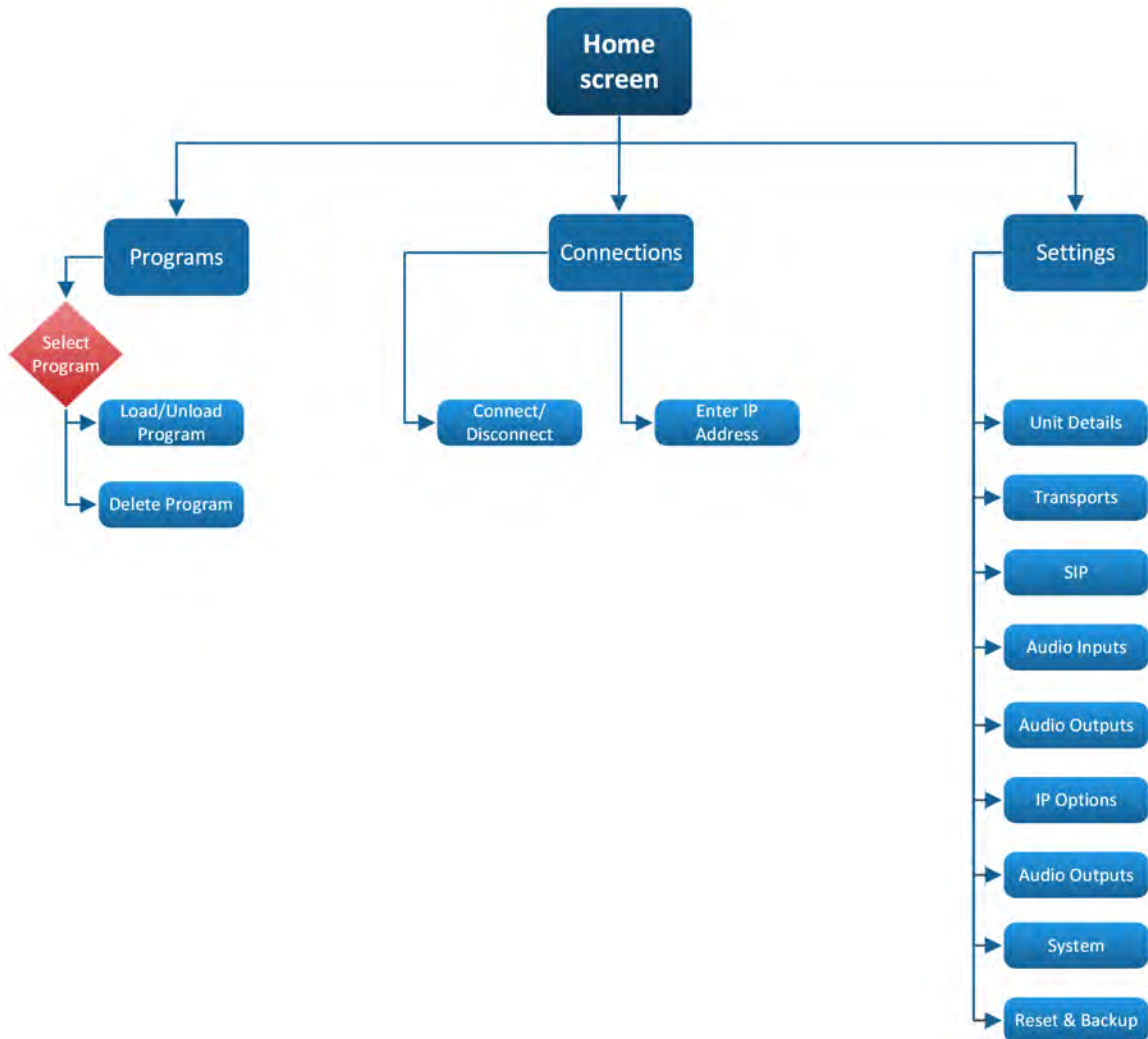
Settings Menu Navigation
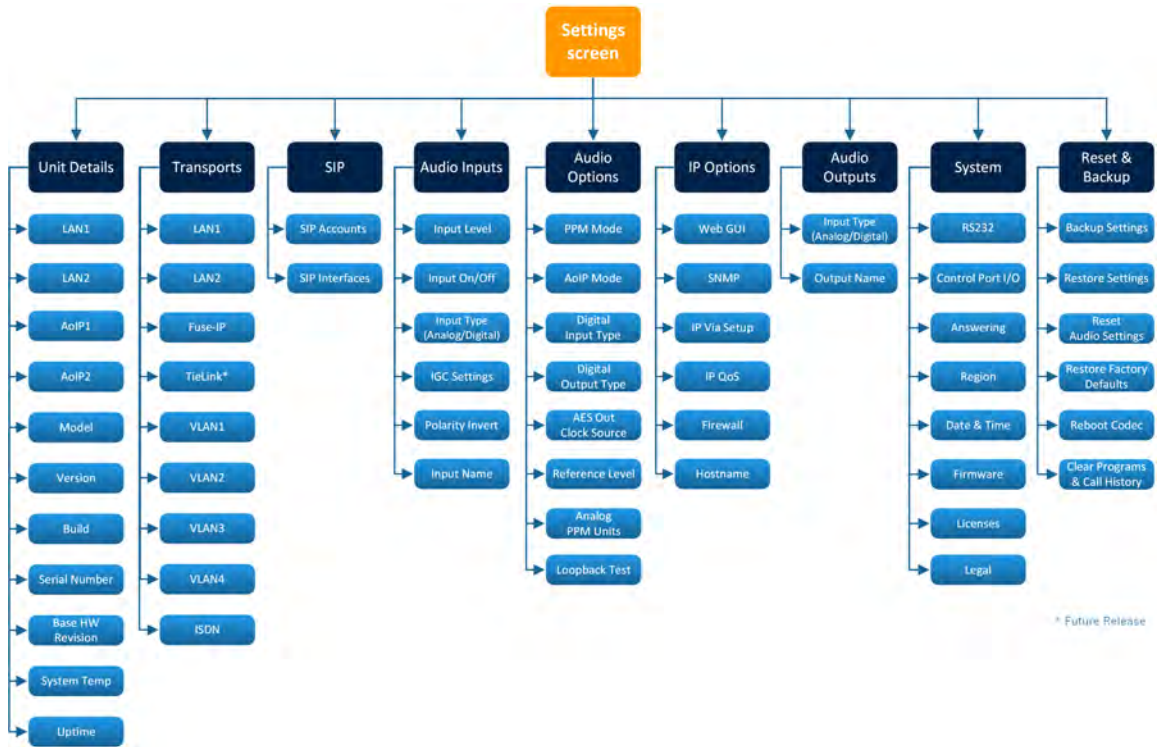
13 Input Configuration, Levels and PPMs

Product	Feature	Notification
Gateway 4	PPM Meters	8 PPMs on front panel display 4 channel inputs/outputs or encoders/decoders
Gateway 4	Inputs	4 Channels in/out
Gateway 4	Inputs	Inputs 1-4 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 8/16	PPM Meters	16 PPMs on front panel can display inputs, or outputs, or encoders, or decoders
Gateway 8/16	Inputs	8-16 Channels In/Out (depending on configuration purchased)
Gateway 8/16	Inputs	Inputs 1-8 can be configured as analog or digital inputs (AES3 or AES67) in pairs

ℹ️ **Important Note:** See Configuring AES3 audio for more information about digital in/outs. Input audio functions can also be configured using the Toolbox Web-GUI; see Configuring Input/Output Settings for more information.

Adjusting PPM Meter Reference Scale Settings

By default, the **PPM METERS** on the front of the codec, 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 G6** reference scale setting.

The default Tieline G6 audio reference scale displayed on the PPMs when connecting to a Tieline G6 codec is -40dBFS to 0dBFS. Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 20dB of headroom from the nominal 0vu point. The comparison table below outlines the reference scales for G6, G5, G3 codecs and Report-IT in dBFS, as well as the equivalent dBU scale.

Gateway-16 Codec PPMs Displayed

The audio reference level settings in the codec are:

	Reference Level	Description	dBu	dBFS
1	Tieline G6 (Gateway, Gateway 4)	PPM meter low point	-16dBU	-40dBFS
		Nominal 0vu reference level	+4dBU	-20dBFS
		Level at which audio will clip/distort	+24dBu	0dBFS
2	Tieline G5	PPM meter low point	-16dBu	-38dBFS

	(Genie, Merlin, Bridge-IT, ViA)	Nominal 0vu reference level	+4dBu	-18dBFS
		Level at which audio will clip/distort	+22dBu	0dBFS
3	Tieline G3 (Commander and i-Mix)	PPM meter low point	-11dBu	-29dBFS
		Nominal 0vu reference level	+4dBu	-14dBFS
		Level at which audio will clip/distort	+18dBu	0dBFS
4	Report-IT	PPM meter low point	-9dBu	-23dBFS
		Nominal 0vu reference level	+4dBu	-10dBFS
		Level at which audio will clip/distort	+14dBu	0dBFS

Important Note: If a codec supports multiple stream programs and the **Auto** (default) reference level is selected, the first codec to connect will configure the reference level used for all subsequent connections. I.e. If a G3 codec connects first then the G3 Audio Reference Level will be configured for all connections.

Configure Tieline G5 Audio Reference Scales

Tieline G6 codecs have slightly more headroom than Tieline G5 codecs, therefore the audio metering reference scale needs to be adjusted when G6 codecs connect to G5 Merlin, Genie, ViA and Bridge-IT codecs. The G5 metering scale is between -38dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -18dBFS. Audio peaks should not exceed 0dBFS.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the OK button.
3. Navigate to **Reference Level** and press the OK button.
4. Select Tieline G5 and press the OK button.
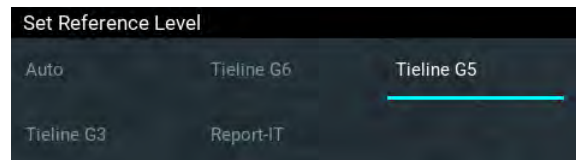
Configure Tieline G3 Audio Reference Scales

Tieline G6 and G5 codecs have more audio headroom than Tieline G3 audio codecs, therefore the audio metering reference scale needs to be adjusted when these codecs connect to a Commander or i-Mix G3 codec. The G3 metering scale is between -29dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -14dBFS. Audio peaks should not exceed 0dBFS.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the OK button.
3. Navigate to **Reference Level** and press the OK button.
4. Select **Tieline G3** and press OK.

Configure Report-IT Audio Reference Scales

The **Report-IT** setting is used for compatibility when streaming audio using Tieline's Report-IT smartphone application.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the OK button.

3. Navigate to **Reference Level** and press the button.
4. Select **Report-IT** and press .

The Report-IT metering scale is between -23dBFS and 0dBFS and audio levels should average around the nominal 0vu point at -10dBFS. Audio peaks should not exceed 0dBFS.

Adjusting PPM Metering

PPM metering options can be adjusted via **Settings > Audio Options > PPM Mode**. Note: Gateway 4 has 4 inputs, 4 outputs, 4 encoders, 4 decoders, and 8 PPMs only; Gateway 8/16 has 16 PPM meters and between 8 and 16 inputs depending on configuration purchased; encoders and decoders available will match input availability/configuration.

Gateway 8/16
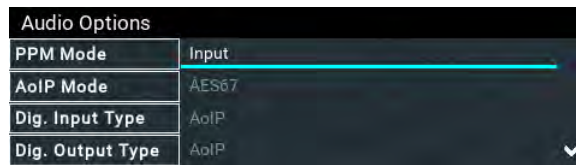
	PPM Mode	Gateway 16 Description	Gateway 8 Description
1	Input (default)	Maps inputs 1 to 16 with PPM meters 1 to 16	Maps inputs 1 to 8 with PPM meters 1 to 8
2	Output	Maps outputs 1 to 16 with PPM meters 1 to 16	Maps outputs 1 to 8 with PPM meters 1 to 8
3	Encoder	Maps input encoders 1 to16 with PPM meters 1 to 16	Maps input encoders 1 to 8 with PPM meters 1 to 8
4	Decoder	Maps decoders 1 to 16 with PPM meters 1 to 16	Maps decoders 1 to 16 with PPM meters 1 to 8
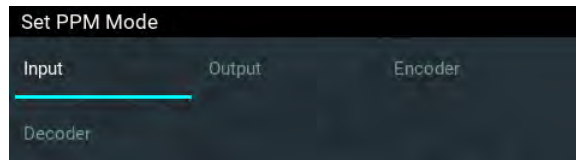
Gateway 4

	PPM Mode	Gateway 4 Description
1	Input/Output (default)	Maps inputs 1 to 4 with PPM meters 1 to 4; maps outputs 1 to 4 with PPM meters 5 to 8
2	Encoder/Decoder	Maps input encoders 1 to 4 with PPM meters 1 to 4; maps decoders 5 to 8 with PPM meters 1 to 4

Adjusting Input Settings

1. Press the **SETTINGS** button.
2. Navigate to **Audio Inputs** and press .

3. Adjustable input settings include:

- Input on/off.
- Input level: ganging of inputs is not currently supported.
- Input Type: Analog (line level input), Digital (AES3 or AES67).
- IGC.
- Polarity/phase inversion
- Input Name

Selecting Analog or Digital Inputs

1. Press the **SETTINGS** button.
2. Navigate to **Audio Inputs** and press .
3. Select an input to adjust and press .
4. Navigate to **Type** and press .
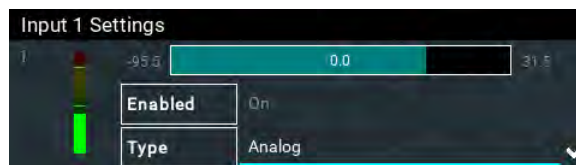
5. Select **Analog** or **Digital** as required. This will configure the associated input as well, e.g. in this example input 2 when input 1 is configured.
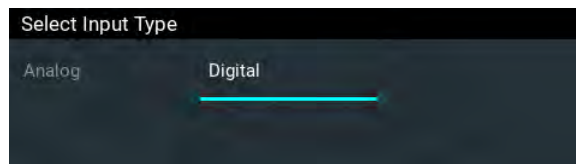
Adjusting Input Levels

1. Press the **SETTINGS** button.
2. Navigate to **Audio Inputs** and press .
3. Select an input to adjust and press .
4. Select the level adjustment bar and use the left ◁ and right ▷ navigation buttons to adjust levels up and down. Note: input ganging is not available in the first firmware release.
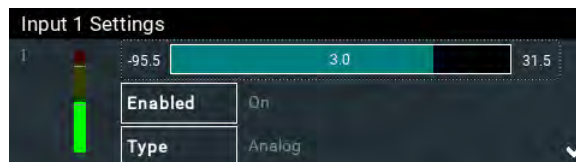
Important Note:
- Gain adjustments can be made in 0.5dB increments.
- There is a maximum of 6dB of additional gain available when adjusting a digital input.
- Audio signal processing occurs in the following order: IGC/Limiter (Intelligent Gain Control limiting on the input) > Input Compression > Input EQ > Mixer > Output AGC (Automatic Gain Control limiter).

Intelligent Gain Control (IGC)

The codec's inbuilt DSP limiter automatically takes care of any instantaneous audio peaks that occur in demanding broadcast situations. **Input IGC** (Intelligent Gain Control) is enabled by default and is automatically activated at +19 dBu (G6 audio scale), +17 dBu (G5 audio scale) and +13dBu (G3 audio scale) to prevent audio clipping.

There are three settings; **Auto**, **Fixed** and **Off**. If **Auto** is configured the codec will detect when incoming audio levels have reduced sufficiently and automatically return 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 Level** has been initiated) and will return the levels to the previous setting within half a second. This response is linear.

To adjust this setting in the codec:

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **Audio Inputs** and press (OK).
3. Select an input to adjust and press (OK).
4. Navigate to **IGC** and press (OK).

Input 1 Settings		
Type	Analog	
IGC	Auto	
Polarity Inv	Disabled, press to enable	
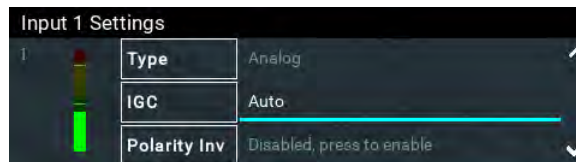
5. Select the preferred setting and press (OK).

Select IGC Options		
Auto	Fixed	Off
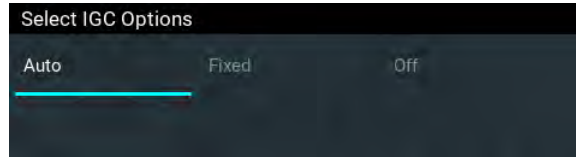
14   Configuring AES3 Audio

Product	Feature	Notification
Gateway 4	Analog / AES3 Inputs / Outputs (4 Analog / AES3 in/out)	**ANALOG/AES3 IN** RJ-45 connectors provide shared stereo analog line inputs 1-4, or stereo AES3 digital inputs 1-4. **ANALOG/AES3 OUT** RJ-45 connectors provide shared stereo analog line outputs 1-4, or stereo AES3 digital outputs 1-4.
Gateway 8/16	Analog / AES3 Inputs / Outputs (8 Analog in/out; 16 AES3 in/out)	**ANALOG/AES3 IN** RJ-45 connectors provide shared stereo analog line inputs 1-4, or stereo AES3 digital inputs 1-4. Connectors 5-8 provide stereo AES3 digital inputs 5-8. Balanced **ANALOG/AES3 OUT** RJ-45 connectors provide shared stereo analog line outputs 1-4, or stereo AES3 digital outputs 1-4, plus stereo AES3 digital outputs 5-8.

Codecs feature stereo AES inputs and outputs on the rear panel of the codec for AES3 (AES/EBU) format audio. These balanced 110 ohm inputs can operate effectively over distances of up to 100 meters and accept both mono and stereo AES3 signals.

Configuring AES3 Inputs and Outputs Globally

Either analog or digital audio is supported via RJ-45 inputs. To configure codec inputs and outputs for digital AES3 requires two settings to be configured for inputs and outputs 1-4 (Gateway 4) and inputs and outputs 1-8 (Gateway 8/16). First, the codec needs to be configured globally for AES3 input and output audio, instead of digital AES67 audio. Next, ensure inputs are configured for AES3 digital audio via **Settings > Audio Inputs > Type**.

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **Audio Options** and press the 🆗 button.
3. Select **Dig. Input Type** or **Dig. Output Type** and press the 🆗 button.
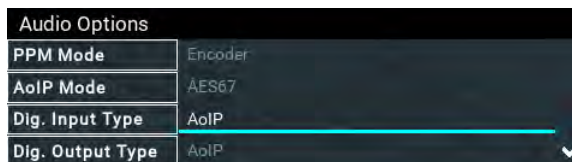
4. Select **AES3** and press the 🆗 button.
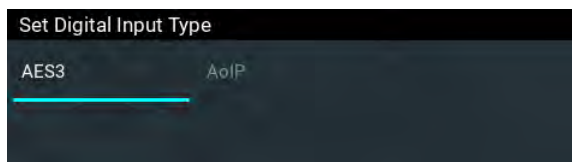
Note: Gateway 16 codec inputs and outputs 9-16 are configured for AES3 when the **Dig. Input Type** and **Dig. Output Type** are set for AES3 audio.

Configuring Inputs for Digital or Analog Audio

Inputs and outputs 1-4 (Gateway 4) and inputs and outputs 1-8 (Gateway 8/16) can be configured in pairs for either Analog or Digital audio. After selecting AES3 globally for inputs and outputs in the **Dig. Input Type** or **Dig. Output Type** menu, next ensure inputs and outputs are configured at the input/output level:

1. Press the **SETTINGS** ✐ button.
2. Navigate to **Audio Inputs** or **Audio Outputs** and press the OK button.
3. Navigate to **Type** and press the OK button.

Input 1 Settings			
1	-95.5	0.0	31.5
	Enabled	On	
	Type	Analog	⌄
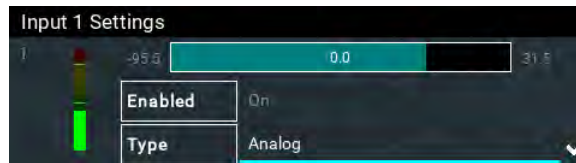
4. Select **Digital** and press the OK button.

Select Input Type	
Analog	**Digital**
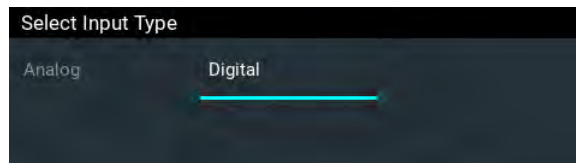
ℹ️ **Important Notes:**
- There is a maximum of 6dB of additional gain available when adjusting an AES3 digital input.
- If you switch back to the analog input setting after selecting AES3, the previous analog settings will be recovered.

AES3 Out Clock Source

The codec contains two sample rate converters.

Input Sample Rate Converter

The codec implements an Asynchronous Sample Rate Converter (ASRC) to convert the sample rate of an AES3 input to the sample rate set in the codec. The codec sample rate is determined by the selected algorithm. For example, if you select the Music algorithm, the sample rate will be set to 32kHz. By default the codec will up-sample all AES3 input sources to 96kHz sampling and then convert to match the AES output sample rate setting.

ℹ️ **Important Note:** All AES3 inputs must have the same sample-rates and must be synchronized to a common clock. See Appendix B for pin-outs of AES3 inputs.

Output Sample Rate Converter

The sample rate of the AES3 output is configured using the clock source setting via the **SETTINGS** ✐ button and then **Audio Options > AES3 Out Clock Src**. This configures the sample rate frequency of all AES3 output signals.

Wordclock Sync In
This setting configures the codec for a word clock source via the **SYNC I/O 2** BNC connector on the codec rear panel (this is the same as the **External Word Clock** setting in Tieline G3 codecs). Often this will be a studio reference signal. In television broadcasting facilities the audio reference signal should be locked to the video reference if there is one available. The sample rate being received is recognized by the codec and automatically adjusted. Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Lock to AES3 Input

With this setting the codec uses AES3 input sync information to set the codec output sample rate (Note: this is the same as the **AES Rx Clock** setting in Tieline G3 codecs). The codec initially tries to use the signal on AES inputs 1 and 2 as the clock to which the AES outputs are synchronized. If unavailable, it then attempt to use inputs 3 and 4, or inputs 5 and 6 and so on in that order. If you select this option, all AES inputs must always be synchronized to the same clock source, e.g. if AES3 inputs 1 and 2 use 48kHz sampling then all other inputs must also be synchronized to the same clock. Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Lock to AES11 Input

AES11 is a standard AES3 signal with an accurate clock reference (DARS, or Digital Audio Reference Signal), usually without the audio data. To use this input attach a female BNC connector to the **SYNC I/O 1** connector on the codec rear panel. Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Lock to AES67 PTP

AES67/ST2110-30 derives the sampling clock from a primary leader clock over the network. This clock can be selected as the **AES3 Output Clock Source**. The **AES3 Output Clock Source** is forced to **Lock to AES67 PTP** if either the *Digital Input Type* or *Digital Output Type* is set to AES67. If a WheatNet-IP card is installed, and the *Digital Input Type* or *Digital Output Type* is set to **WNIP**, then the **AES3 Output Clock Source** is forced to **Lock to WNIP Clock**.

Fixed Sample Clock

Select from a range of fixed output sample rates including:

1. 32 kHz
2. 44.1 kHz
3. 48 kHz
4. 88.2 kHz
5. 96 kHz

Note: The reference clock must be within +/- 50ppm of the listed sample rates.

15 Headphone Output

The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output on the front panel. Audio streams and analog inputs or digital inputs can be selected and monitored.

Adjust Headphone Output Settings

1. Press the **HEADPHONE** button to display the headphone monitoring screen.

2. Use the up △ or down ▽ navigation buttons to adjust the volume level up or down. Use the left ◁ and right ▷ navigation buttons to adjust the send and return audio balance. The send and return audio balance dictates whether the **HEADPHONE** output monitors send (input/encoder) audio only, return audio only (decoder audio from a connected device), or a mix of both send and return audio. Level and send/return audio balance adjustments occur in real-time.
3. Press 💿 to display the **Sources** screen and select an audio stream or audio inputs to monitor.
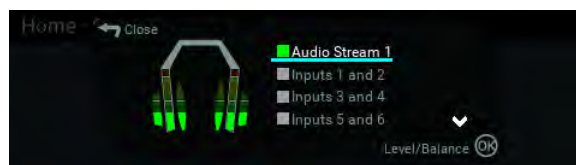
⚖ **Codec Model Note:** The number of inputs and audio streams displayed varies depending on the codec model purchased.

4. Use the up △ or down ▽ navigation buttons to select an audio stream or pair of inputs to monitor. Press 💿 again to return to the headphone monitoring screen.
5. Press the **Return** button to exit the menu.

16   Inserting Hardware Modules

A module slot on the codec rear panel supports the optional Gateway Euro ISDN module and future hardware and technology upgrades.

Gateway Euro ISDN module

Inserting or Removing a Module

Ensure the codec is not powered up 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. Remove power from the codec and then remove the 4 screws from the blanking panel or module installed in the codec.
2. 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 within the codec.
3. Reinsert the 4 screws to hold the module firmly in place.
4. Power up the codec.
5. Press the **SETTINGS** button.
6. Navigate to **Transports** and press the button.
7. Navigate to **ISDN** and press the button to verify it is installed correctly.

Important Note: If the module does not appear in the **Transports** menu in the codec, it is possible that the connector on the module has not lined up correctly with the connector inside the codec. Remove the module and reinsert it carefully to resolve this issue.

17   IP Streaming, Programs, and TieLink

Tieline codecs support high quality IP audio streaming connections using the following protocols:

1. Tieline Session Data: Proprietary session data sent when Tieline codecs connect to each other in order to establish, manage and terminate connections.
2. SIP (Session Initiation Protocol): EBU N/ACIP Tech 3326 compliant connections using SDP (Session Description Protocol) when connecting to other brands of IP codecs.
3. Sessionless: The codec does not send session data when attempting to connect. Requires the "send" audio port and "return" audio port to be configured.

Tieline Session Data

When a connection between two codecs is established:

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

For example, if you configure a stereo program on the dialing codec using Opus encoding at 128kbps with specific jitter settings, this will be configured on the dialing codec when it connects.

SIP

SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. The codec is fully EBU N/ACIP Tech 3326 compliant for connecting to any codec supporting SIP connections. For more information about SIP connections see About SIP.

TieLink Traversal Server Network

TieLink is a secure, independently hosted global server network, with multiple global backups. Linked to a TieServer domain, it centralizes Tieline codec contact list management and provides self-discovery of codecs within customized 'call-groups'. It also provides NAT traversal to simplify connections. This service is free to all Tieline customers but is not supported with Bridge-IT or G3 codecs.
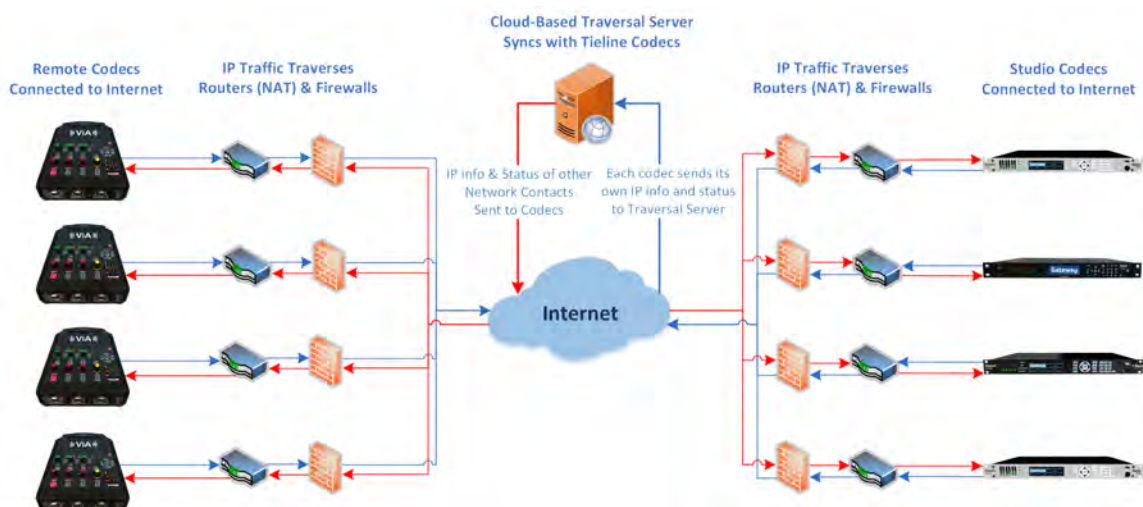
How does it work?

Using a simple HTML5 web-browser interface, engineers can log in and register their codecs with TieLink. Simply name your codec (e.g. BAYFM Studio A) and complete the registration details. Once registered, a codec will automatically connect to TieLink each time it connects to the internet:

1. TieLink automatically detects a codec's IP address and NAT routing information. Seconds later the codec address book populates with all codecs in the network, or a subset of codecs in an authorized call-group.
2. The registered codec will appear in the address books of all other codecs registered with TieLink and connected to the internet. Call-groups can be created to customize the number of codecs visible to users and engineers within large networks.
3. Users can easily view the online or offline status of all codecs in a group and whether it is connected or disconnected.

Manage Call Groups Easily

Whether you are running a large network with hundreds of IP codecs, or a small station with just a few, the TieLink Traversal Server and Toolbox software makes connecting over IP networks a simple, no-fuss task for non-technical broadcast personnel.

For networks with large numbers of devices, it is easy to create sub-groups and configure a codec to follow a specific group, e.g. news or sports. All of the codecs in your network are secure and cannot be seen by other users unless they have permission to join the network, or a specific codec call-group. See Configuring TieLink Settings for more information.

Programs

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. By default, Tieline codecs send 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 session data including information about how the codec receiving the call should be configured.
2. Once the answering codec receives session data it sends an acknowledgment to the dialing codec and streaming can commence.

For example, if you configure a stereo program on the dialing codec using Opus encoding at 128kbps with specific jitter settings, this will be configured on the dialing codec when it connects. It is also possible to lock a loaded program in a codec to ensure the currently loaded program cannot be unloaded by a codec dialing in with different program settings. For example, if your routing requirements 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 where possible. In some situations incompatible program types will be rejected.

Defining Audio Streams within Programs

Each audio stream within a program can be defined separately and contain a variety of settings relating to the number of connections (e.g. primary and backup) and the number of destinations to which each audio stream is distributed. Each audio stream is capable of being configured to include

dial and answer connections, dial connections only, or answer connections only. Each audio stream includes:

- Name.
- Connection, Transport, and Destination settings.
- Backup configuration options.

The following image displays a simple peer-to-peer program in the **Program Manager panel** within the HTML5 Toolbox Web-GUI, which can be used to configure and edit all program parameters. The following program is configured to send a single stereo audio stream and will allow the codec to both answer and dial if required. A backup **Failover** dialing connection is configured in case the primary connection fails.
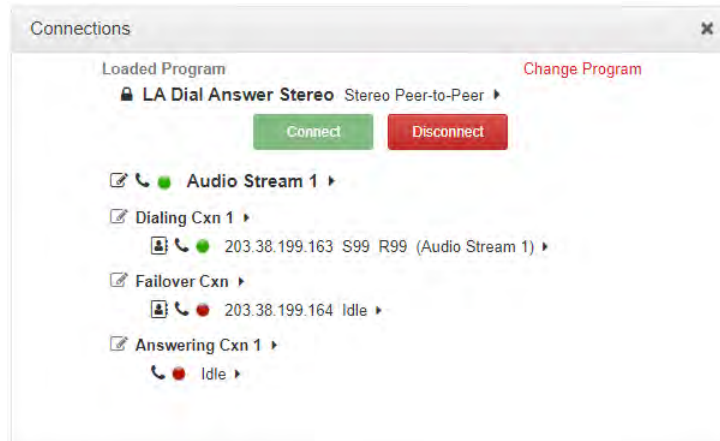
Creating Programs

The HTML5 Toolbox Web-GUI contains a **Program Manager panel** with a wizard for configuring program settings and backup connections. See Using the HTML5 Toolbox Web-GUI for more detailed information on configuring programs. Edit settings easily at the touch of a button and use existing programs as templates for creating other programs.

Multi-Unicast Programs

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. All connections that are part of a multi-unicast connection will automatically attempt to reconnect if they are terminated remotely.

Multi-unicasts are useful for distributing audio to several studios and can only be created using the **Program Manager panel** in the HTML5 Toolbox Web-GUI. Once multi-unicast connections have been created you can press **CONNECT** on the codec keypad to connect without using the HTML5 Toolbox Web-GUI. See Configure Multi-unicast Dialing Programs for more info.

Multicast Programs

A multicast audio stream can be used to broadcast to unlimited numbers of 'subscriber' (client codecs) wanting to receive a particular audio transmission. Multicast transmissions are broadcast by a multicast server codec using a dedicated IP multicast address, which looks similar to a regular IP address, and multicast subscribers request transmissions from this address. See Configure Multicast Server Programs or Configure Multicast Client Programs for more info.

18   About TieServer

A TieServer Domain is a unique domain associated with a particular broadcaster/customer, that is used to securely demarcate their Tieline assets from other broadcasters. 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.

Each Tieline domain has unique administrator login credentials and codecs can be added to a TieServer Domain with TieLink or the TieServer Console. If you already have Report-IT Enterprise or the Cloud Codec Controller, the same domain can be used for TieLink. Following is a diagram showing the relationship between a TieServer Domain and Tieline's software products; TieLink, the TieServer Console, and the Cloud Codec Controller.
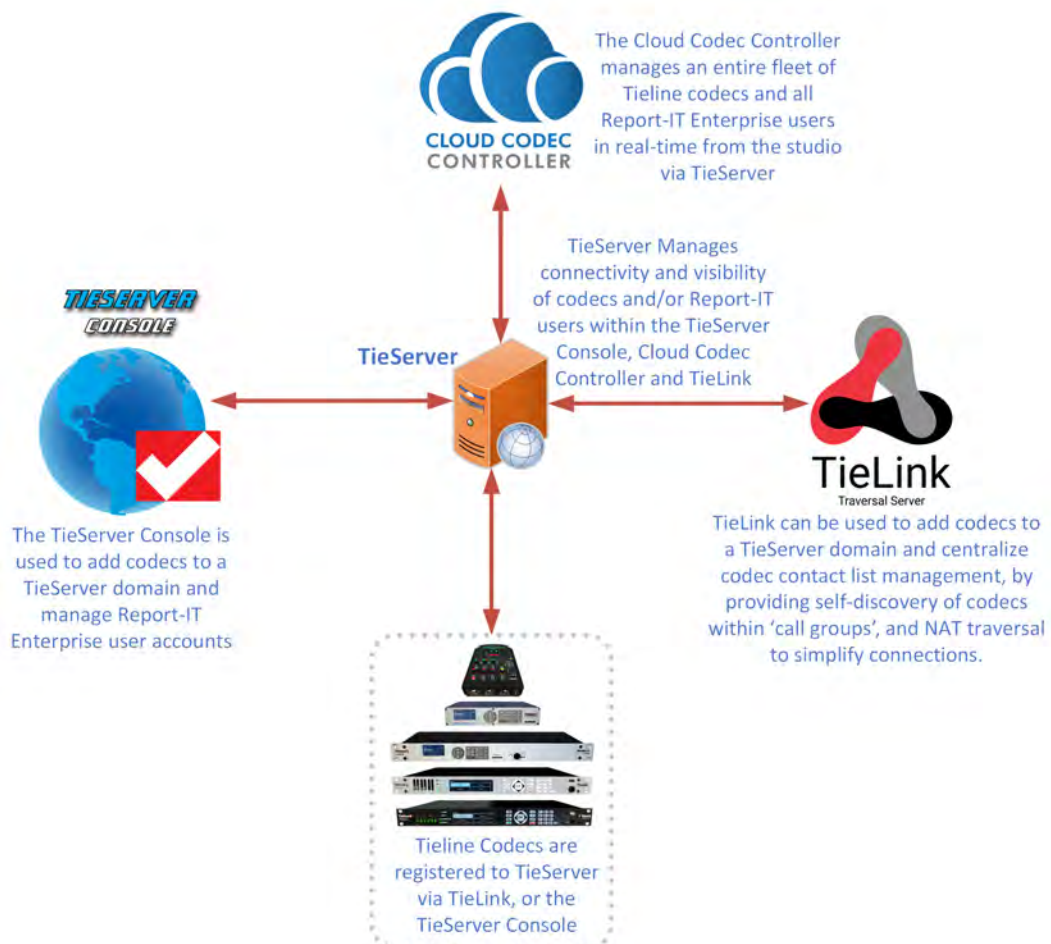
It is critically important to remember that removing a codec from a domain removes it from:

- The TieServer Console: used to manage an organization's codecs and Report-IT users.
- Report-IT user accounts: users authorized to connect to a particular codec will no longer be able to 'see' or connect to a codec after it is removed.
- The Cloud Codec Controller: used to manage an entire fleet of Tieline codecs and Report-IT Enterprise users remotely.
- TieLink Traversal Server: used to configure codec dialing groups and traverse firewalls and NAT more simply.

19   Configure Language and Country

Configure Country Setting

To configure the country setting in the codec:

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **System** and press the OK button.
3. Navigate to **Region** and press the OK button.

Region Settings	
Language	English
Country	United States

4. Use the navigation buttons to select your country of operation and press the OK button.
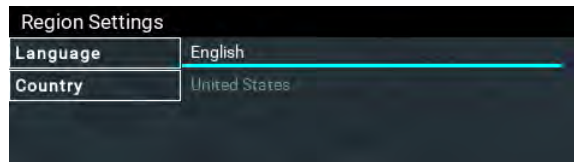
Select Country		
Tuvalu	Uganda	Ukraine
United Arab Emirates	United Kingdom	**United States**

ℹ️ **Important Note:** The country setting affects whether G.711 µ-Law (North America/Japan) or A-Law (Europe/Australasia) coding is used over IP and SIP connections.

Configure Language Setting

The codec menus are available in English, French, Spanish and Portuguese. To configure the language setting in the codec:

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **System** and press the OK button.
3. Navigate to **Region** and press the OK button.
4. Use the navigation buttons to select a preferred language and press the OK button.

20   About SIP

SIP provides interoperability between different brands of codecs due to its standardized protocols for connecting different devices. The codec is fully 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. 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).

All the mandatory EBU N/ACIP 3326 algorithms are supported in the codec, including G.711, G.722, MPEG-1 Layer 2 and 16 bit PCM, as well as optional algorithms including Opus, LC-AAC, AAC-LD, HE-AACv2 and aptX Enhanced.

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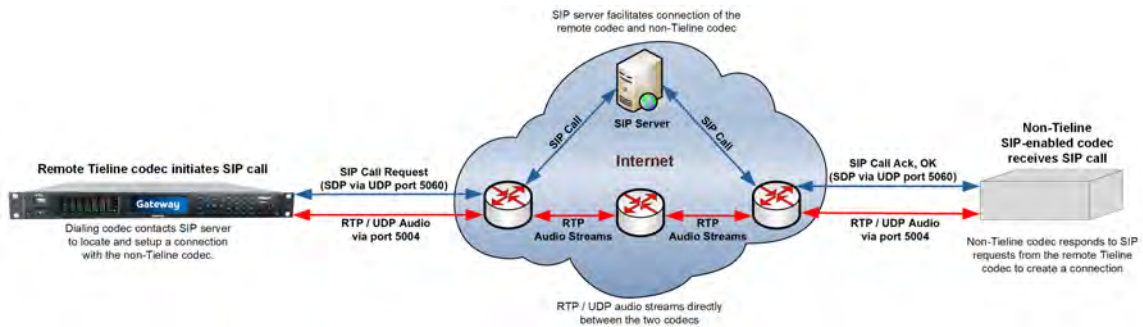
Some SIP servers route RTP audio through the SIP server as well and Tieline recommends avoiding this type of server whenever possible. Otherwise you will be reliant on the SIP server for streaming broadcast audio packets and most servers are not designed for mission critical packet streaming.

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 ports 5004 to 5054.

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

- Username
- Authorized User
- SIP address
- Domain
- Realm
- Registrar
- Registar port
- Outbound Proxy
- Proxy port

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**.

> **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.
> - 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 **SETTINGS** 🔧 **> Audio Options > Reference Level > Tieline G3**. In addition, configure 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]**

20.1   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 **LAN2** by default, which is mapped to the **Secondary** 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 Address** text box 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 **SETTINGS** 🔧 button, then navigate to **SIP** and press the 🆗 button.
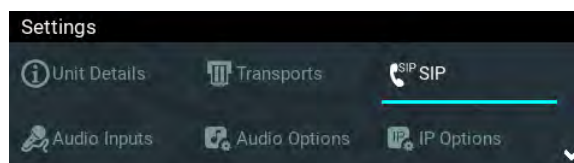
2. Navigate to **SIP Interfaces** and press the 🆗 button.
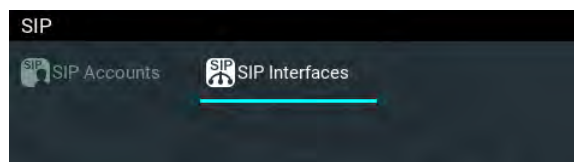
3. Select either **SIP1** or **SIP2** and press 🆗 to view and edit settings. Note: A SIP interface must be disabled before settings can be edited.

SIP Interfaces

| SIP1 | LAN1 (Disabled) |
| SIP2 | LAN2 (Disabled) |
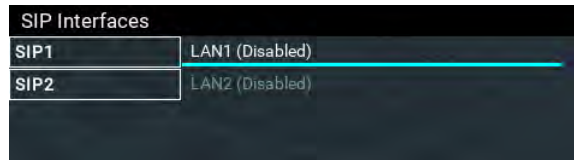
4. Select **Edit** in the dialog to edit the SIP interface settings.

Select action to proceed

Cancel   Enable   Edit
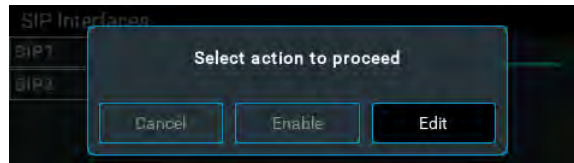
5. Navigate to each setting in turn and press the ⊙ button to edit SIP interface settings as required.

Edit Interface: SIP1

State	Disabled
Session Port	5060
Audio Port Start	5004
Audio Port End	5054

6. Select **STUN Server** and press the ⊙ button to enter server details if using STUN (Session Traversal of User Datagram Protocol). Note: UDP Port 3478 is the default port assigned.

Edit Interface: SIP1

Proxy Server	Press OK to configure
Proxy Server Port	Proxy Server not configured
Stun Server	stun.ekiga.net
Stun Server Port	3478
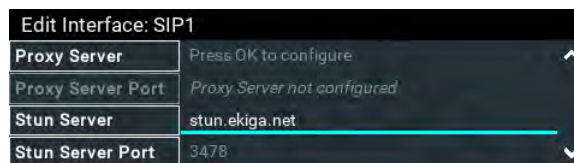
7. Select **Public IP** and press the ⊙ button if you want to dial over SIP from behind a firewall. Enter the public IP address and then configure port forwarding to route traffic to the codec's local IP address behind your firewall. Note: Do not enter a **Public IP** address if STUN is configured. They 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.

Edit Interface: SIP1

Stun Keep-Alive	15
Public IP Address	Press OK to configure
NAT Type	STUN Not Configured
Answer Route	Any
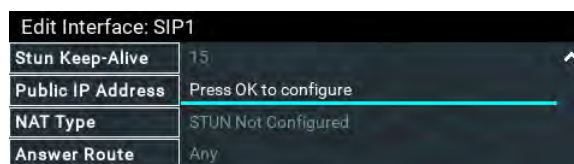
8. The **NAT Type** used by the firewall is updated if STUN has been enabled and the **NAT Type** can be determined. Note: the SIP interface must be enabled to view this.

Edit Interface: SIP1

Stun Keep-Alive	15
Public IP Address	Press OK to configure
NAT Type	Port Restricted
Answer Route	Any
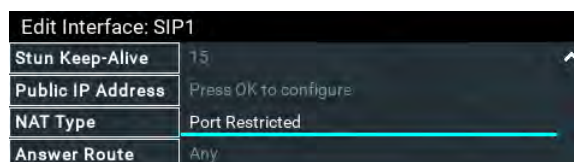
9. Select **Answer Route** and use the **Change Interface Answer Route** screen to route calls using this SIP interface to a specific audio stream. The route setting in this menu must correspond with the answering route configured in an audio stream within the loaded

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.
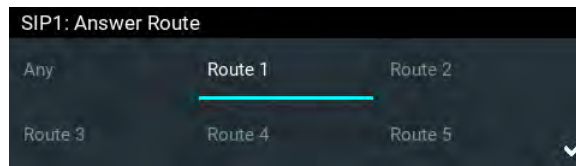
10. Press the **Return** ⤺ button to exit the menu and select **Save** in the dialog to store new settings. Note: to enable an interface after configuration press the ⊙ button and select **Enable** in the onscreen dialog.
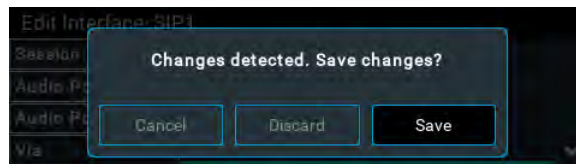
20.2 Configuring SIP Accounts

Getting Started

Up to 16 SIP accounts can be configured and registering codecs for SIP connectivity is simple. First, select the SIP server to which the codec will be registered. On a LAN this may be your own server, or it could be one of the many internet servers available. When registering an account with a SIP server you will be provided with:

- The SIP registrar and Proxy Server details.
- An authorized user or username (often the same as a SIP number).
- A password.
- Domain details.
- Realm details (in most cases leave blank).
- 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** (**LAN1**) or **SIP 2** (**LAN2**).
- Up to 16 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 **SETTINGS** �‾ button, then navigate to **SIP** and press the ⊙ button.
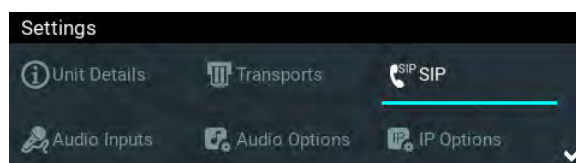
2. Navigate to **SIP Accounts** and press the ⊙ button.

SIP

SIP Accounts   SIP Interfaces
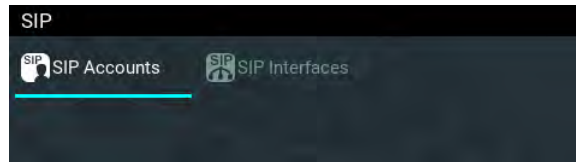
3. Select an account which is **Not Registered** and press ⊙.

SIP Accounts

Not registered	1. Press OK to configure
Not registered	2. Press OK to configure
Not registered	3. Press OK to configure
Not registered	4. Press OK to configure
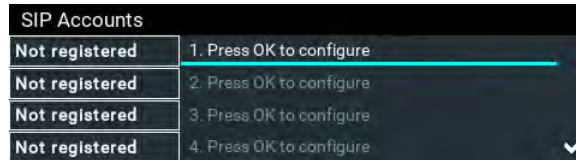
4. Select **Edit** and then press ⊙ to individually select and enter SIP server account details.

SIP Accounts

Select action to proceed

Cancel   Enable   Edit
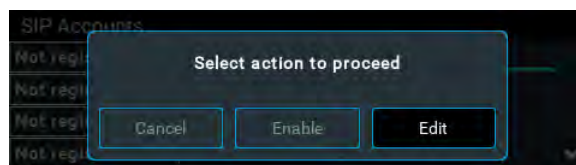
5. Navigate to each field in turn and press the ⊙ button to enter SIP account credentials.

Edit SIP Account

Registrar Server	sip2sip.info
Registrar Port	5060
Proxy Server	proxy.sipthor.net
Proxy Port	5060
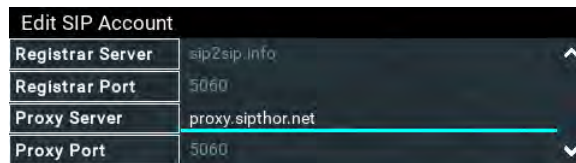
6. Configure an **Answer Route** to route incoming calls (audio streams) to this SIP account.

Edit SIP Account

Reg. Timeout	3600
Answer Route	Any
Interface	SIP2 (ETH2)
Session Port	Use Interface Port
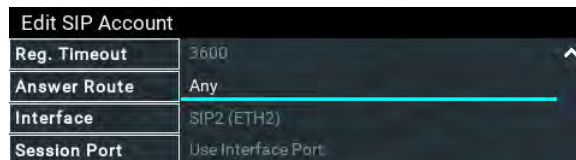
7. The route setting in this menu must correspond with the answering route configured in an audio stream within a loaded program. If the default value **Any** is used then a call will be routed to an audio stream on a first-come-first-served basis in a multi-stream program.

SIP1: Answer Route

| Any | Route 1 | Route 2 |
| Route 3 | Route 4 | Route 5 |
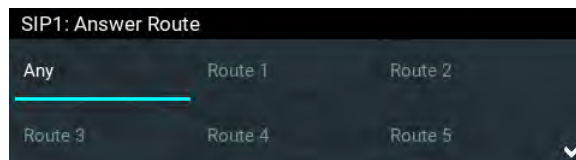
8. By default, the SIP account will use the interface's session port, e.g. port 5060. However, some SIP software applications like CallManager (Unified CM) require each interface to support its own Session Port. Therefore, a different **Session Port** can be specified for each SIP Account.
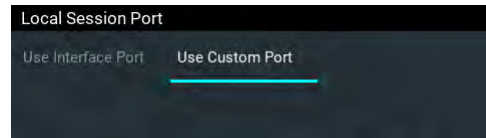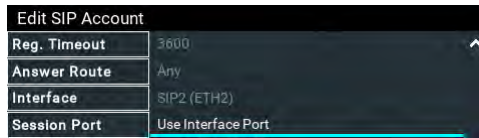
After completing account configuration, press the **Return** ⟵ button to exit the account configuration menu. Then select the account and press the ⊙ button to enable the SIP account and register it with the SIP server.
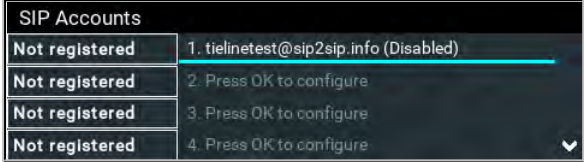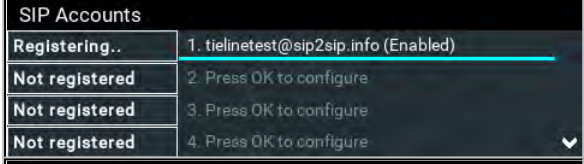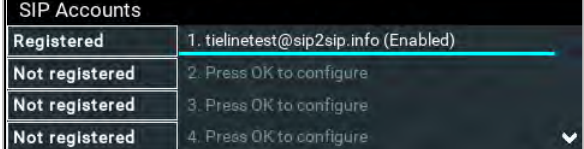
9. To confirm the account has been registered successfully, verify the **SIP LED** on the front panel of the codec is illuminated solid orange. Alternatively, verify the account has a **Registered** next to it in the **SIP Accounts** menu.

ⓘ **Important Note:** Once enabled, the SIP account can be used when creating a new SIP connection. Configure this menu setting via **Connect > IP > SIP > Setup > Account**.

Confirming Account Registration

Three SIP account registration status indications may be displayed.

Symbol	Description
	Not registered is displayed before an account is registered.
	Registering is displayed when account registration is being attempted.
	Registered is displayed when an account is registered to a SIP server.

The **SIP LED** on the front panel of the codec also displays the following SIP account registration states:

USER (SIP) LED State	Description
Not Illuminated	Indicates an 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**.

21   Connecting Quickly

Product	Feature	Notification
Gateway 4	Programs	Supports program connections using up to 4 channels of audio
Gateway 8/16	Programs	Supports program connections using up to 16 channels of audio, depending on codec configuration purchased (between 8 and 16 channels available for purchase)

Before creating a new IP connection adjust the following:

1. Attach power to the codec.
2. Attach an RJ45 Ethernet cable to one of the **LAN** ports on the codec's rear panel.
3. Attach headphones to the 6.35mm (1/4") headphone jack on the codec's front panel.
4. Verify the correct country is selected in the codec:
i. Press the **SETTINGS** button.
ii. Navigate to **System** and press the OK button.
iii. Navigate to **Region** and press the OK button.
iv. Use the navigation buttons to select your country of operation and press the OK button.
5. Make sure you know the IP address for dialing the destination codec.

> **Important Note:** The country setting affects whether G.711 μ-Law (North America/Japan) or A-Law (Europe/Australasia) coding is used over IP and SIP connections.

Creating a New Connection Program

The codec supports a large number of different connection programs. To load a program it is necessary to launch the Toolbox HTML5 Web-GUI and perform the following steps:

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**. (See Opening the HTML5 Web-GUI & Login for how to launch the Web-GUI).
2. Click the **New Program** button to open the wizard and:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select a program from the list of program types available.
3. Configure and save the new program using the procedure outlined in the section Configuring Mono or Stereo Programs.

Load and Connect a Program

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the OK button.
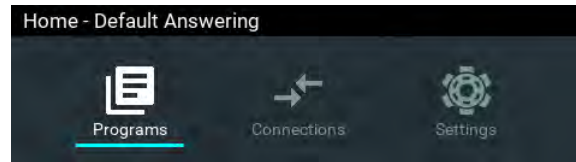
3. Navigate to a program and press the ⊙ button to select it.
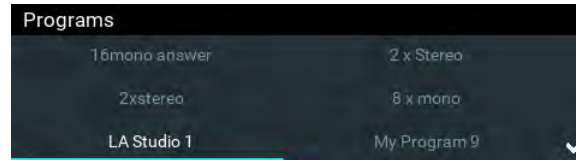
4. Select **Load** and press the ⊙ button to load the program.
5. Navigate to the **Connections** screen and then press the **CONNECT** 🔘 button to make a connection.

Edit the IP Address in a Loaded Program

It is not possible to create a program from the codec front panel, however it is possible to edit an IP address for each audio stream connection.

> **Important Note:** The IP address for an audio stream connection can only be edited when it is disconnected.

1. Press the **HOME** 🔘 button to return to the **Home** screen.
2. Use the navigation buttons to select **Connections** and press the ⊙ button.
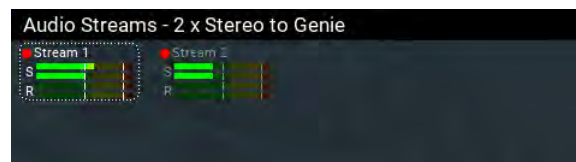

3. Select a disconnected stream and press the ⊙ button.
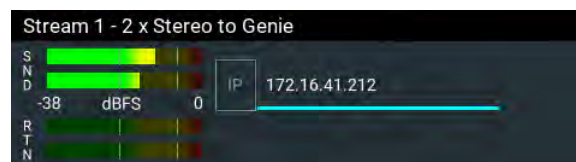
4. Press the ⊙ button again to edit the IP address using the keypad or onscreen keyboard, then press the **Enter** button on the onscreen keyboard.

21.1 Monitoring Connections

Connections Screen Explained
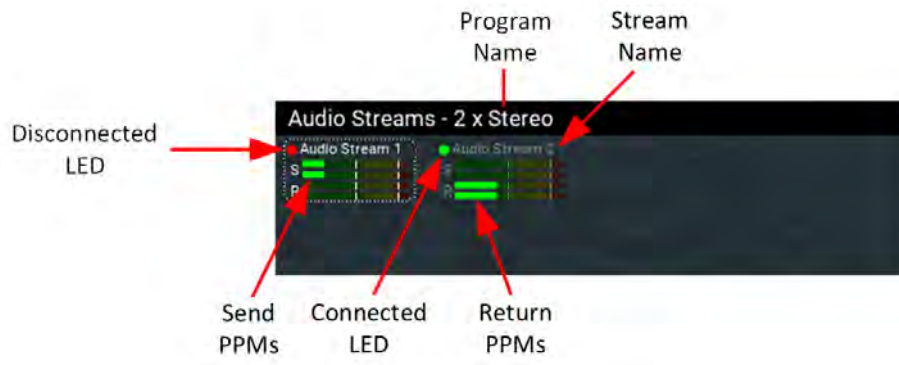
Connection status of up to 16 audio streams can be monitored using the **Connections** screen.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Connections** and press the button.
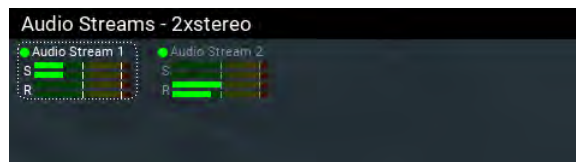
21.2 Load and Dial Custom Programs

Custom programs stored on the codec are simple to load and dial from the codec front panel.

1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.
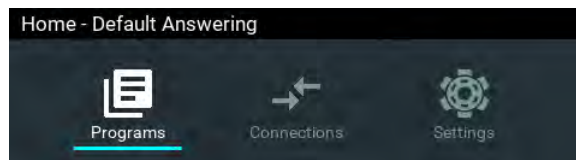
3. Use the navigation buttons to select a program and press the button to load it.
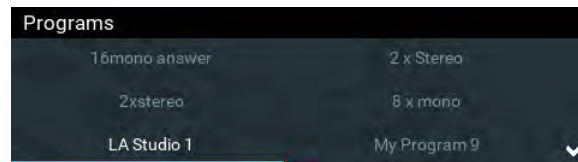
4. Navigate to the **Connections** screen and press the **CONNECT** [icon] button to make a connection.

5. A dialog appears during the connection process with connection details displayed.

Editing the IP Address or ISDN Number

It is possible to edit the IP address of a destination or the ISDN number for ISDN calls.

1. Press the **HOME** [icon] button to return to the **Home** screen.
2. Use the navigation buttons to select **Connections** and press the [OK] button.
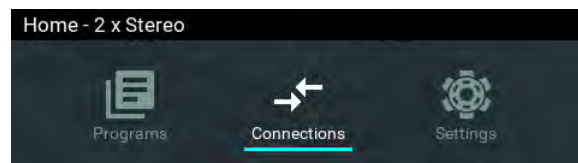
3. Select a connection and press the [OK] button.
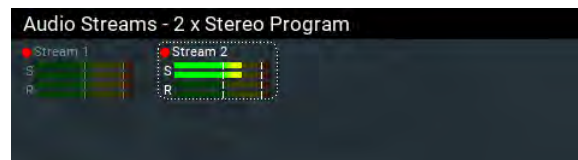
4. Press the [OK] button to edit the destination address/number.
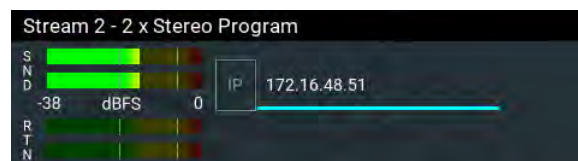
5. Use the onscreen keypad to edit the dial string, then press the [OK] button to save it.

6. Press the **CONNECT** [icon] button to make a connection.

21.3 Disconnecting a Connection

1. Navigate to the **Connections** screen and select a connection, then press the red **DISCONNECT** button on the numeric **KEYPAD** at any time to hangup an individual connection.

2. Use the right ▷ navigation button to select **Disconnect** and press the 🆗 button to confirm the disconnection.

Note: Press the red **DISCONNECT** button on the numeric **KEYPAD** from other menus at any time to disconnect all active connections.
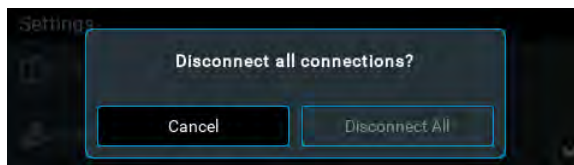
21.4 Deleting Programs

1. Press the **HOME** button to return to the **Home** screen.

2. Use the navigation buttons to select **Programs** and press the 🆗 button.

3. Navigate to the program you want to delete and press the 🆗 button.
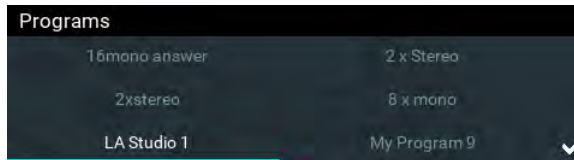
4. Navigate to **Delete** and press the 🆗 button.
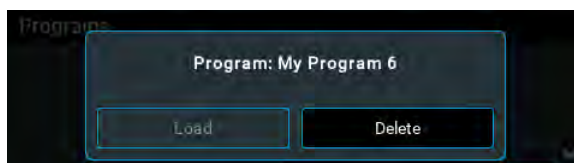
5. Confirm the deletion and press the 🆗 button.

21.5   Lock or Unlock a Program in the Codec

It is possible to lock a loaded custom program in a codec to ensure the currently loaded program type, e.g. dual stereo, cannot be unloaded by a codec dialing in with a different program type, e.g. stereo. It is essential to lock multi-stream programs in a codec or they will be unloaded by the first call received. Only mono and stereo connections can be answered when the **Lock Program** feature is disabled.

For example, if routing requirements require the codec at the studio to always connect using dual stereo connections, 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 algorithm preferences and bit rates via session data.

1. Press the **SETTINGS** 🔧 button.
2. Use the navigation buttons to select **System** and press the 🆗 button.
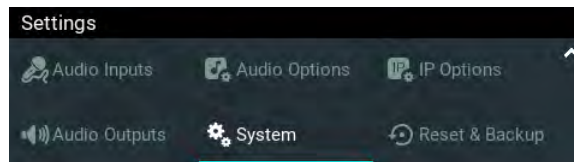
3. Navigate to **Answering** and press 🆗.
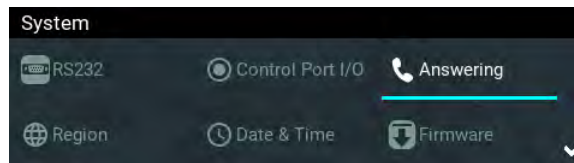
4. Navigate to **Lock Program** and press 🆗 to toggle between **Enabled** and **Disabled**.
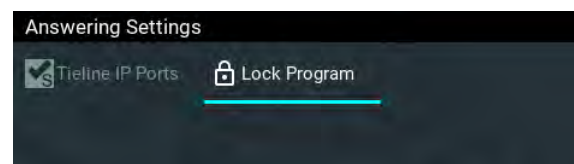
> **ⓘ** **Important Note:** It is only possible to lock custom programs in a codec. If **Lock Program** is enabled and a new custom program is loaded, **Lock Program** remains enabled and locks the most recently loaded custom program.

22   Connecting to the ToolBox Web-GUI

There are two graphical user interface (GUI) options for configuring and connecting Tieline G5 and G6 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.
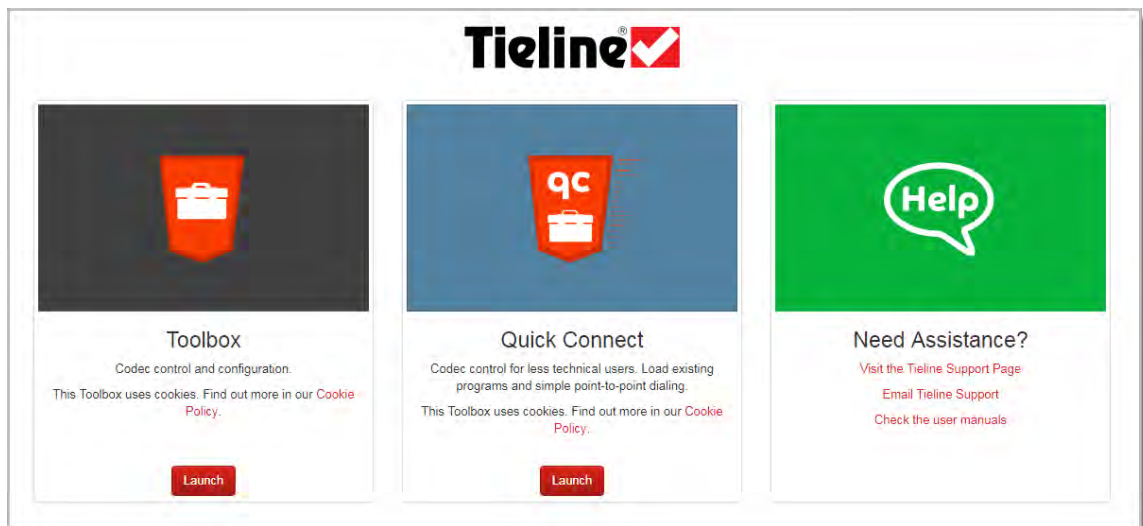
About the HTML5 Toolbox Web-GUI

The HTML5 Toolbox Web-GUI improves the user experience with 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 simple peer-to-peer connections with the **Quick Connect panel**. To enable the Quick Connect panel press the **SETTINGS** button and navigate to **IP Options > WebGUI > Quick Connect > [Enabled]**.

22.1   Opening the HTML5 Web-GUI & Login

1. Attach an Ethernet cable to the **LAN1** port on the codec.
2. Press the **SETTINGS** button and select **Unit Details** to view IP address details for the codec.
3. Ensure the PC used to configure the codec is connected to the same LAN.
4. Launch a web browser and type the IP address of the codec into the address bar of the 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 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.
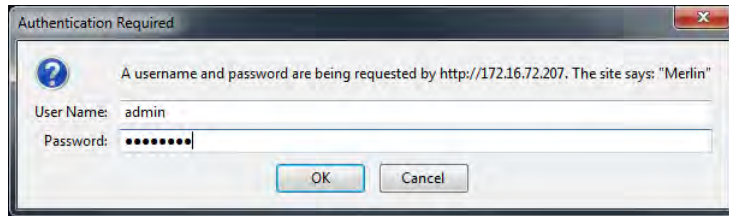

Launching the AES67 Web-GUI

To access the AES67 Web-GUI type the IP address of the AES67 AoIP interface into a browser. IP address details can be found in the codec via the **AES67 Host Network panel** accessed via the HTML5 Toolbox Web-GUI **AoIP** menu. Note: This can only be accessed when a computer is attached to the same AES67 LAN as the Tieline codec.
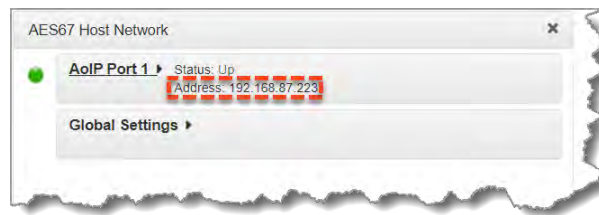
Alternatively, press the **SETTINGS** button on the codec front panel and use the navigation buttons to select **Unit Details** and press the button to discover the AoIP port address details.

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.
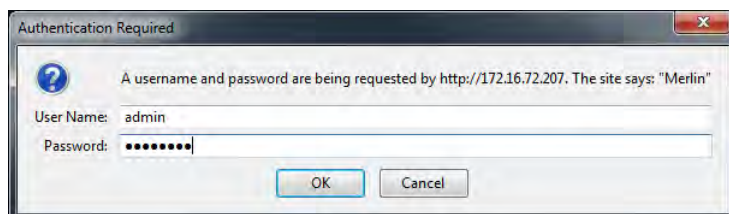
Using the Web-GUI over the Internet

If the codec is connected to the internet and is assigned a public static IP address it is possible to connect and configure it from any PC connected to the internet. If multiple browsers are open on a PC for different codecs it is possible to customize the browser title for simple identification. 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**.

22.2   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 Allow and Block lists 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 facilitate enabling or disabling a range of firewall-related network services, or limit ping to only work in a local subnet. Tieline also recommends SNMP is disabled if a codec is connected to a public network like the internet. Adjust settings using the Toolbox HTML5 Web-GUI **Options panel** in the **Firewall** tab, 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 password for the Toolbox Web-GUI is **password**. Enter this in the authentication dialog to use the Web-GUI initially and then Tieline highly recommends changing the default password to protect your codec from being tampered with during live broadcasts. Note: In the HTML5 Web-GUI authentication dialog it is necessary to enter **admin** as the **User Name**.
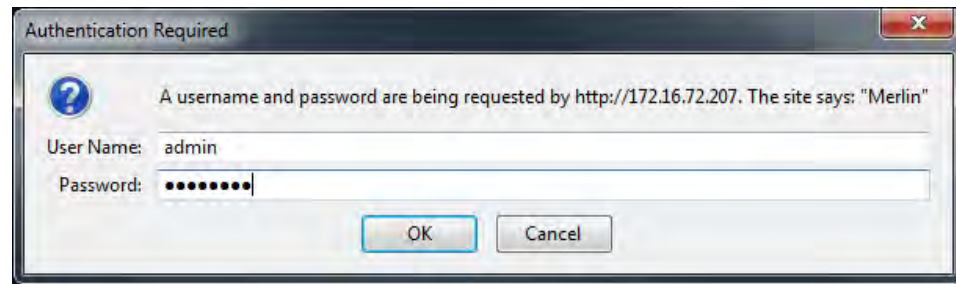
Toolbox HTML5 Web-GUI Login Dialog on a Merlin Codec

Creating a New Password

The authentication login password can be changed at any time using the codec keypad and **LCD SCREEN**. Note that passwords are case sensitive:

1. Press the **SETTINGS** button.
2. Navigate to **IP Options** and press the button.
3. Select **WebGUI** and press the button.
4. Select **Password** and press .
5. Use the **KEYPAD** or onscreen keyboard to enter a new password and press the button to save the new setting (Note: there is no character limit for passwords).

If you forget the password for the Toolbox web-GUI press the **SETTINGS** button on the codec and navigate to **IP Options > WebGUI** to view the current password.

Important Note: The **Username** in the codec menu is permanently set to **admin** and cannot be changed; only the **Password** can be changed.

23   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 logging in subsequently. 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.

> **Important Note:** The AoIP menu and associated panels are for managing AES67/ST2110-30 AoIP streaming. Please see the AoIP Streaming Manual for configuration and streaming information.

Features in Gateway and Gateway 4

Codec capabilities such as inputs, outputs and control port capabilities vary between different models. The following table outlines major feature differences between the most popular codec configurations available.

| Product	Feature	Notification
Gateway 4	PPM Meters	8 PPMs on front panel display 4 inputs/outputs or 4 encoders/decoders
Gateway 4	Inputs	4 Channels in/out
Gateway 4	Inputs	Inputs 1-4 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 4	Web-GUI Inputs panel	4 inputs and PPMs displayed
Gateway 4	Control Port GPIOs	Supports 8 control port inputs and 8 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connectors
Gateway 4	Software LIOs	3 virtual inputs, 56 logical outputs, and 64 WheatNet Logic Inputs/Outputs
Gateway 8/16	PPM Meters	16 PPMs on front panel can display inputs/outputs/encoders/decoders
Gateway 8/16	Inputs	8-16 Channels In/Out (depending on codec configuration purchased)
Gateway 8/16	Inputs	Inputs 1-8 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 8/16	Web-GUI Inputs panel	8 - 16 inputs and PPMs displayed (depending on codec configuration purchased)
Gateway 8/16	Control Port GPIOs	Supports 16 control port inputs and 16 opto-isolated outputs via dual DB25 **CONTROL PORT I/O** connectors
Gateway 8/16	Software LIOs	3 virtual inputs, 48 logical outputs, and 64 WheatNet Logic Inputs/Outputs

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 the preferred option. Note: this manual uses the **White** theme for most images.
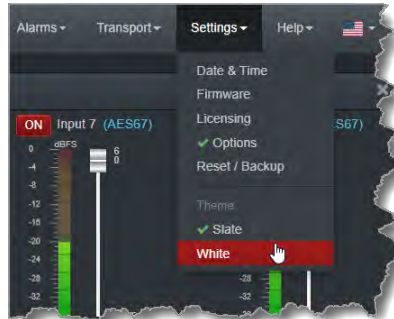

Opening a Panel & Adjusting Size or Screen Position

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.
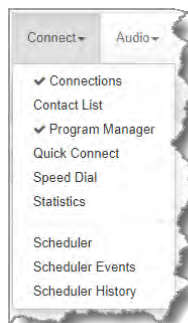
Position the mouse pointer over a panel's **Title bar** and click and drag to move a panel and reposition it in a preferred screen position.
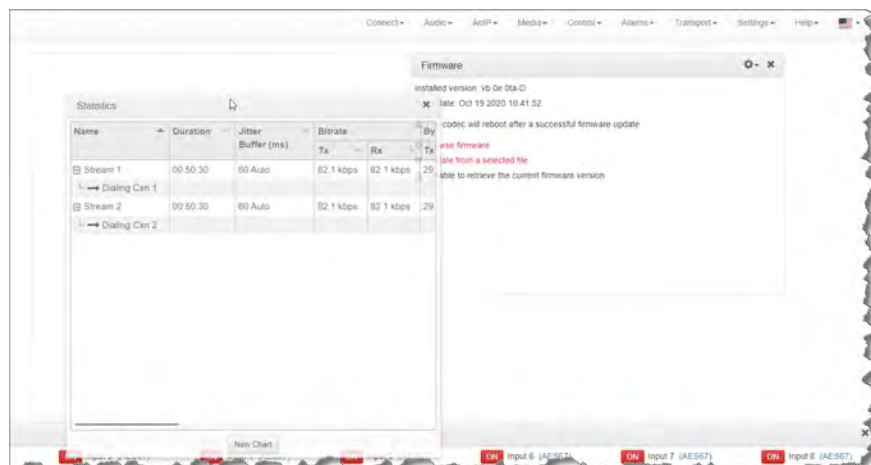
Connect Panels: Load & Connect Programs & Manage Audio Streams

Connections Panel

	Feature	Description
1	Program **Connect / Disconnect** buttons	Click to connect or disconnect all audio streams in a program.
2	**Connect / Disconnect** button and **Connection State LED**	Click the **Connect/Disconnect** 📞 button to connect/disconnect all connections in an audio stream. The **Connection State LED** displays status of an audio stream connection.
3	Audio Stream **Edit**	Click the **Edit** 📝 button to edit audio stream settings.
4	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.
5	TieLink **Contact List** 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.
6	Answering connection **Edit** button	Click to edit answering connection settings
7	Answering Connection **Disconnect** button	Click to disconnect an answering connection
8	Additional Connections	Other connections listed
9	**Change Program**	Click to unload the current program and load a new program.
10	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
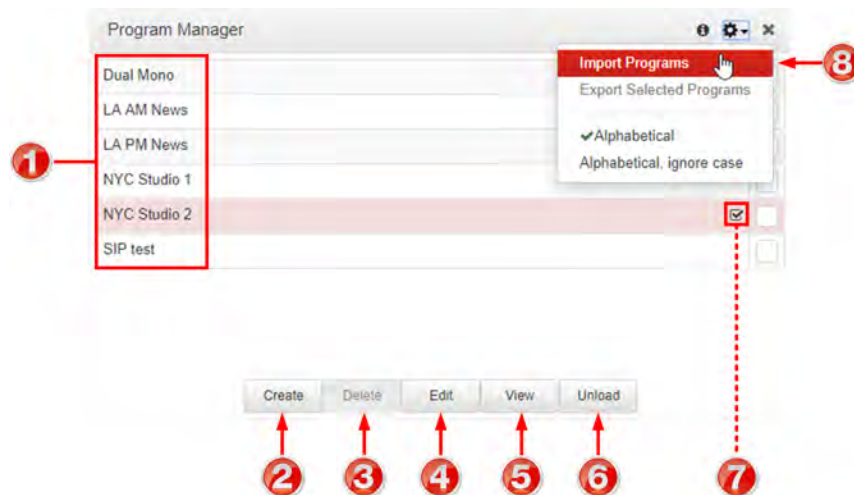
	Feature	Description
1	Program list	The list of saved programs in the codec
2	Create New Program button	Click to create a new program using the program wizard.
3	Delete Selected Programs button	Click to delete all selected programs
4	Edit Selected Program button	Click to edit the selected program
5	View 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

Speed Dial
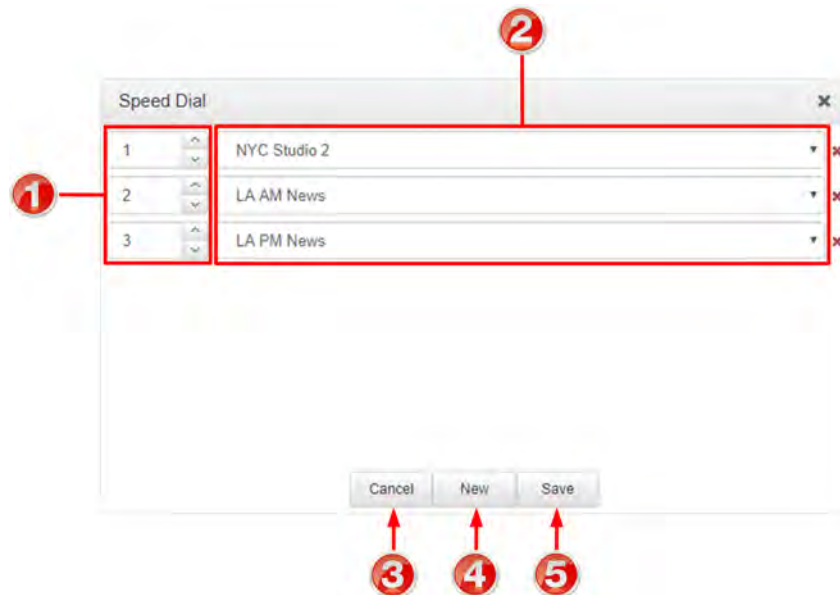
	Feature	Description
1	Speed dial numbers	The number allocated for each speed dial configured
2	Speed dial program	Select the program to associate with a speed dial number
3	**Cancel** button	Click to cancel pending changes
4	**New** button	Click to create a new speed dial
5	**Save** button	Click to save configuration changes

Statistics Panel
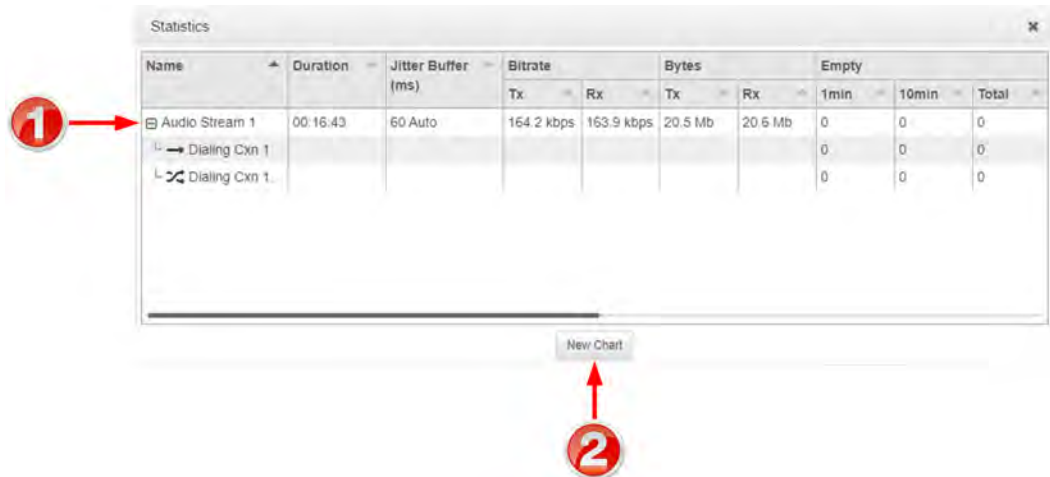
	Feature	Description
1	Expand/Collapse	Click to show/hide audio stream statistics, including packet arrival data info.
2	**New Chart** button	Click the **New Chart** button to select a **Data Series** and create a customized **Statistics panel**.

Scheduler Panel
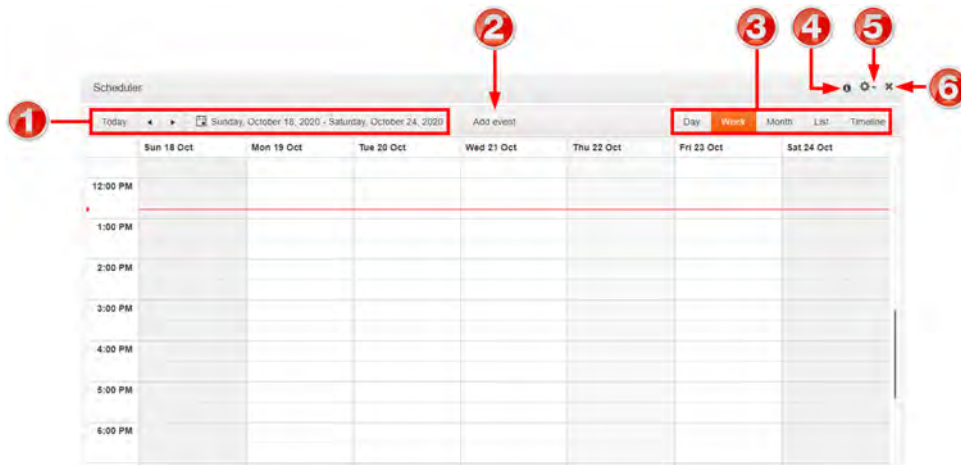
	Feature	Description
1	Date selection	Select the days you wish to view in the scheduler.
2	Add event button	Click to create a new scheduled event.
3	View type	Click to select the timeframe or timeline view of scheduled events.
4	**Information symbol**	Hover over the **Information symbol** to view **Scheduler panel** information.
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
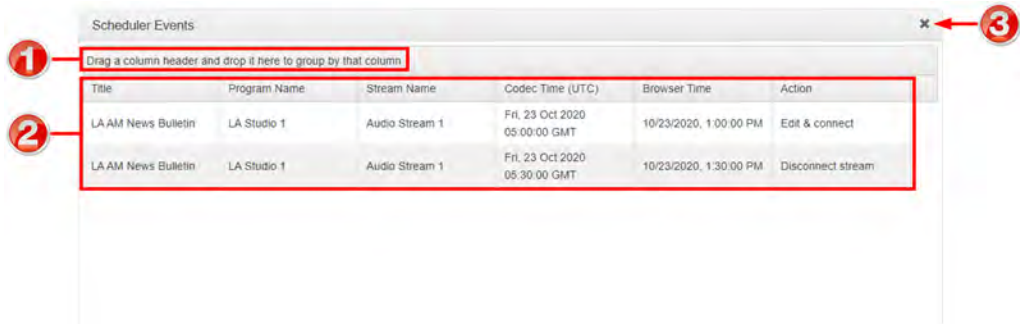
	Feature	Description
1	Grouping	Drag and drop a column header to group scheduled events by that column, e.g. **Codec Time**.
2	List of scheduled events	View scheduled events in a list view
3	**Close** button	Click to close the panel.

Scheduler History
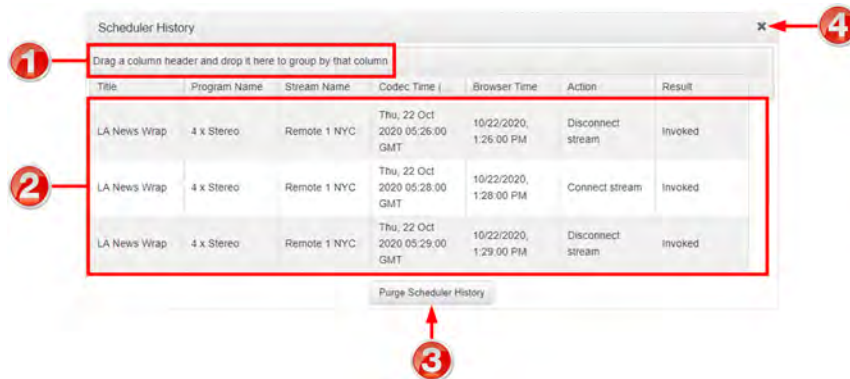
	Feature	Description
1	Grouping	Drag and drop a column header to group event history by that column, e.g. **Codec Time**.
2	Event History List	View the codec's history of scheduled events.
3	**Purge Scheduler History** button	Click to clear all event history displayed.
4	**Close** button	Click to close the panel.

Audio Menu Panels

ℹ️ **Important Note:** Tieline codecs have different input/output configurations, therefore the images shown in this section may not reflect the number of inputs displayed in your codec Web-GUI.

Audio Options

	Feature	Description
1	Audio Over IP Mode	If an optional WheatNet-IP card is installed in the codec select WheatNet-IP as the AoIP mode. Select AES67 as the AoIP mode when connecting to any other AES67 network protocol.
2	Digital Input Type	Select either AES3 or AES67 as the **Digital Input Type** for the codec. This setting is global for all digital inputs.
3	Digital Output Type	Select either AES3 or AES67 as the **Digital Output Type** for the codec. This setting is global for all digital outputs.
4	AES Output Clock	The sample rate of the AES3 output is configured using the clock source which sets the sample rate frequency of all AES3 output signals. Adjust using this setting.
5	PPM Mode	PPM metering can be adjusted to display inputs, outputs, encoders and decoders with this setting.
6	Analog Input PPM Units	Use this setting to change the analog input PPM meter unit of measurement from dBFS (default) to dBU
7	Reference Level	The codec will automatically adjust the reference level to suit G5 and G3 codecs, or Report-IT. The reference level can also be preconfigured using this setting.

Inputs Panel

	Feature	Description
1	**Settings** button	Click to adjust input **Name**, **Type**, **Polarity Inverted** and **IGC** settings
2	**EQ button**	Click to enable and adjust 10 band EQ settings for individual inputs
3	Compressor button	Click to enable the input compressor on each input
4	**On/Off** button	Click to toggle an input on or off
5	Input Sliders/Faders	Input gain control sliders/faders. Adjust input gain in +/- 0.5dB increments
6	Input PPM meter	Input PPM meter
7	Input gain adjustment	Enter a dB value to adjust input gain in 0.5dB increments
8	**Close** button	Click to close the panel

Matrix Editor Panel
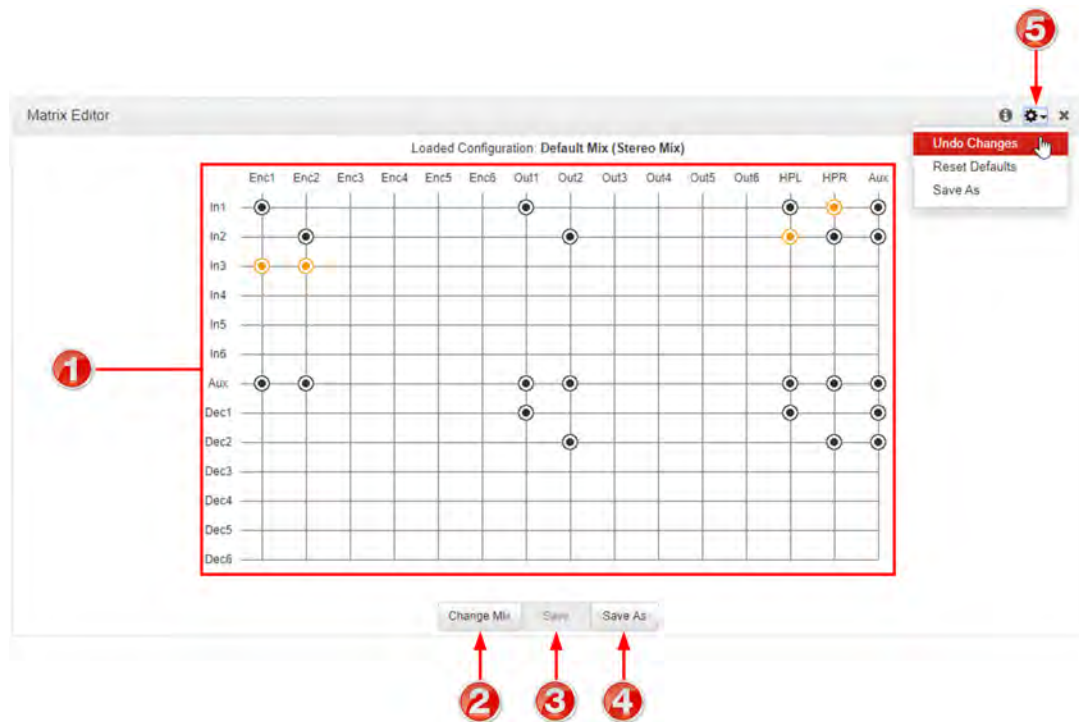
	Feature	Description
1	Matrix Editor crosspoint matrix	Click to select and deselect audio crosspoints to customize routing.
2	**Change Mix**	Click to select the default or Custom matrix mix to view and edit. Options include **Load**, **Delete** and **Rename** a mix
3	**Save**	Click to save changes to a custom matrix mix
4	**Save as**	Click to save as a new custom matrix mix
5	**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.

Outputs Panel

	Feature	Description
1	Settings	Select analog or digital as the output type. Note: configure the global digital output type via **Setting > Audio Options > Dig.Output Type > [select AoIP or AES3]**.
2	**Send/Return slider**	Use the **Send/Return** slider to adjust the balance between the mix of incoming (return/decoder) audio and outgoing (send/input) audio fed to each codec output
3	Output PPM meter	Output PPM meter
4	**Send/Return** indications	Balance of send and return audio included in the mix for each output (expressed as a percentage)
5	**Close** button	Click to close the panel

PPMs Panel

Note: Click and drag the bottom right-hand corner to expand the panel.
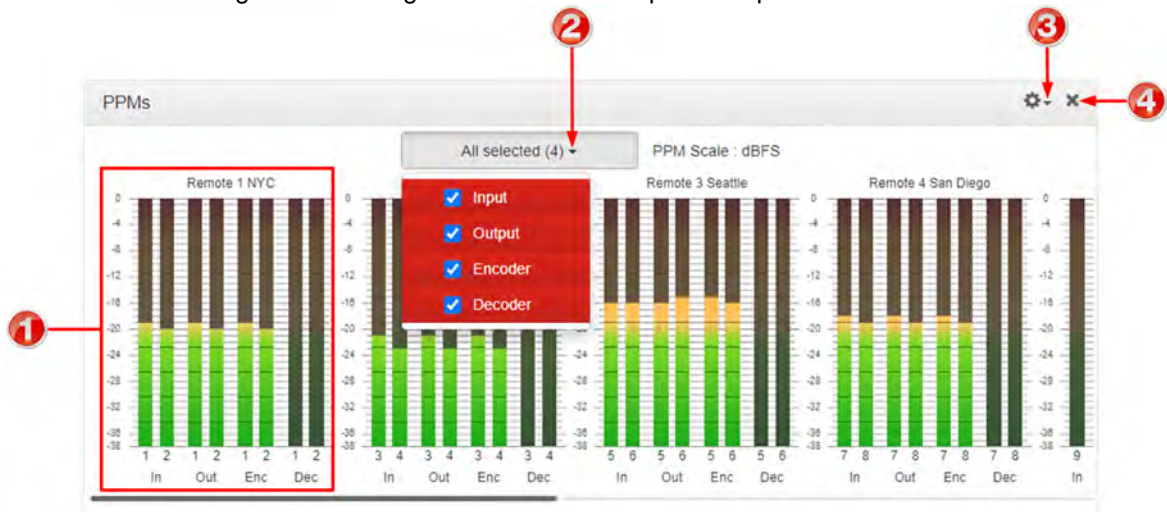
	Feature	Description
1	**PPMs**	Audio stream PPMs grouped
2	**PPM Select** menu	Selectable audio stream PPM metering options using dBFS audio scale
3	**Options menu**	Options menu allows users to group PPMs by stream, or by type, e.g. input, output, encoder and decoder.
4	**Close** button	Click to close the panel.

Important Note: Codec front panel **PPMs 1-6** and PPMs in the **PPMs panel** always reflect the default program settings and do not display customized mix adjustments using the **Matrix Editor**.

Media Menu

File Manager
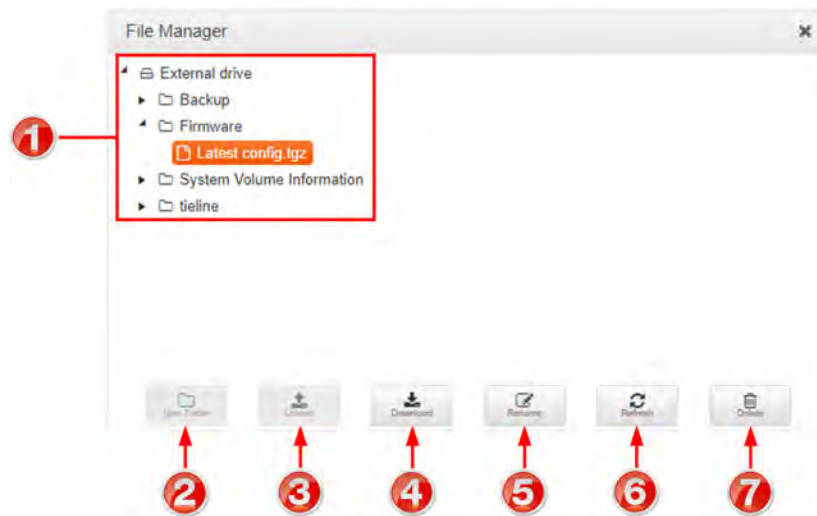
	Feature	Description
1	Folder and File View	View all folders and files on an external drive / removable media
2	**New Folder**	Select the drive or a folder and then click to create a new sub-folder
3	**Upload**	Click to select and upload a new file onto removable media
4	**Download**	Click to download 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
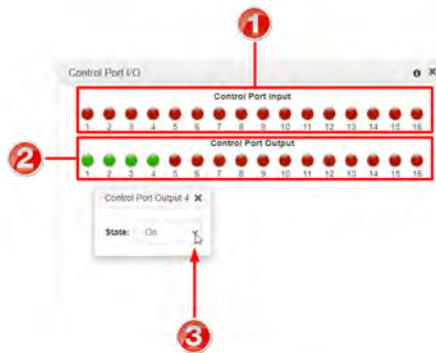
	Feature	Description
1	Control Port Input state	Displays the state of a control port input
2	Control Port Output	Displays the state of a control port output
3	**State**	Click a **Control Port Output** to change the On/Off state

Rules Panel
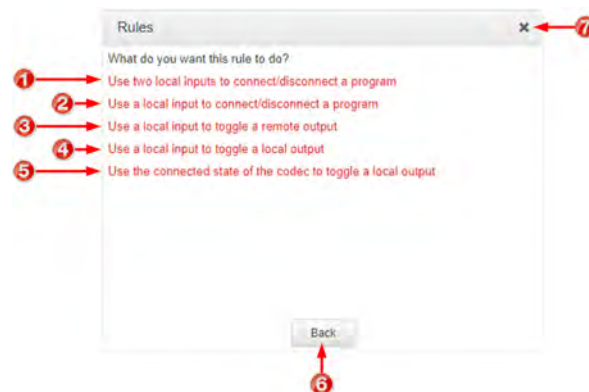
	Rule	Description
1	Use two local inputs to connect/disconnect a program	Click to configure connection and disconnection after different relay inputs are switched **ON**.
2	Use a local input to connect/disconnect a program	Click to configure connection and disconnection by toggling an input.
3	Use a local 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.

Alarms Panels: Configure & Monitor Alarms

Configure Alarms Panel

	Feature	Description
1	List of alarm types	Click to select an alarm type to configure.
2	Enable Alarm check-box	Click the **Enabled** check-box to enable the currently selected alarm.
3	**Edit** / **Save** button	Click to edit an alarm, or save configured alarm settings when in edit mode.
4	Alarm Severity Setting	Click the drop-down arrow to select an alarm severity setting.
5	**Close** button	Click to close the panel

Alarm Dissemination Panel

Alerts for each alarm severity level are configured using the **Alarm Dissemination panel**.

	Feature	Description
1	List of alarm severity levels	Click to select an alarm severity level to configure it.
2	**Front panel alarm & Toolbox** check-box	Select the check-box (default enabled) to deliver front panel **ALARM LED** notifications and HTML5 Toolbox Web-GUI alarm notifications.
3	**Close** button	Click to close the panel.
4	**Relay** drop-down selection	Click the drop-down arrow to select a relay to open when an alarm using the current severity level is activated.
5	SNMP Trap Target text-box	Click in the text box in edit mode to enter the SNMP trap target for alarms using the currently selected severity level.
6	**Edit** / **Save** button	Click to edit alarm dissemination settings, or save configured settings when in edit mode.
7	**Send SNMP trap** check-box	Select the check-box to enable SNMP traps to be sent (for alarms using the selected severity level).

Current Alarms
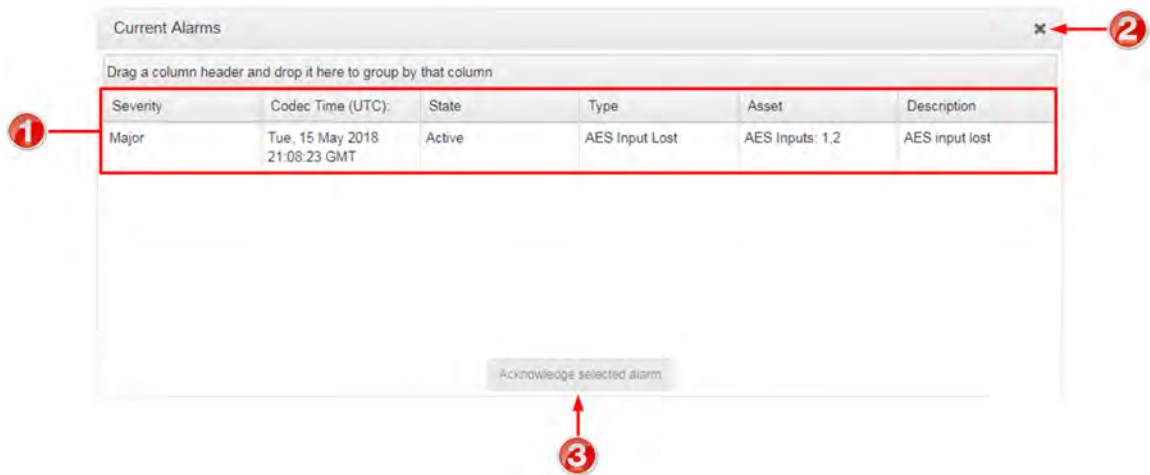
	Feature	Description
1	Current alarm description	View a list of active alarms in the codec
2	**Close** button	Click to close the panel
3	**Acknowledge selected alarm** button	Click to acknowledge a selected alarm.

Alarm History
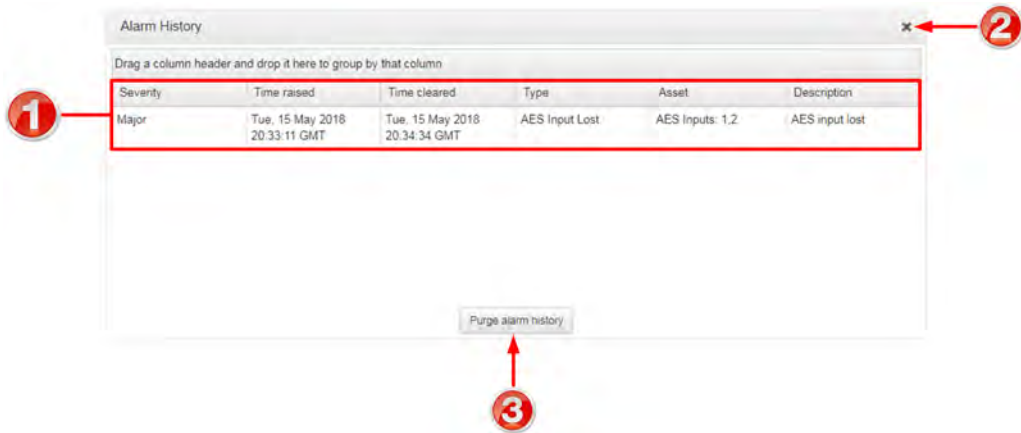
	Feature	Description
1	Alarm history description	View the history of previous alarms in the codec.
2	**Close** button	Click to close the **Alarms** panel.
3	**Purge Alarm History** button	Click to clear the alarm history.

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.
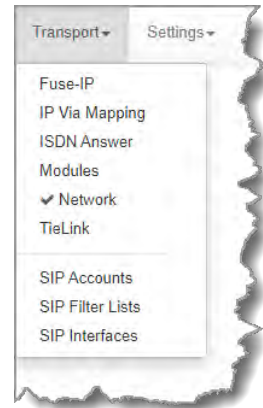
As an example, the **Network panel** is displayed with network interface configuration options. A brief description of the other panels is also provided.
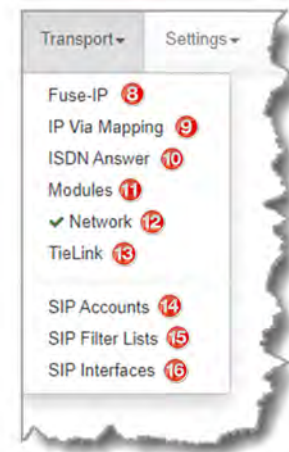
	Feature	Description
1	Network interface	Click to select, expand and view configuration, and edit network configuration settings for Ethernet, Wi-Fi and VLAN interfaces.
2	**Details** tab	Displays configuration options for the selected network interface, plus other device details.
3	Control/Streaming	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	Selectable check-box to enable/disable an interface.
7	**Save/Undo** button	Click **Save** to store settings, or click **Undo** to revert to previously configured settings.
8	**Fuse-IP**	Click to open the **Fuse-IP panel** and configure Fuse-IP bonding.
9	**IP Via Mapping**	Configure default **Primary**, **Secondary** and **Tertiary** interfaces.
10	**ISDN Answer**	Click to open the panel and configure ISDN Answering settings if an ISDN module is installed.
11	**Modules**	Click to edit hardware module configuration settings.
12	**Network**	Click to open the **Network panel** and configure network settings.
13	**TieLink**	Click to view the TieLink panel to enable, disable and prioritize interfaces, configure ports and Fuse-IP interfaces, and configure STUN server settings.
14	**SIP Accounts**	Click to open the panel and edit SIP account settings. Up to 6 SIP accounts are supported.
15	**SIP Filter Lists**	Add trusted network codecs to the **URI Allow List**. Add SIP URIs to the **URI Block List** and add user agents to the **User Agent Block List** to deny them access to the codec.
16	**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.
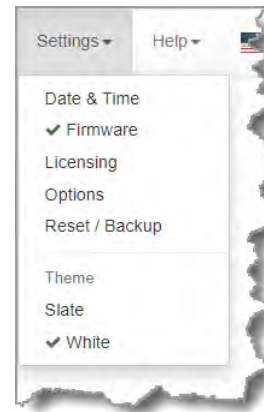
Settings panels

	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 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	**Firewall** Settings	Enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet.
2	**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**.
3	**SNMP** Settings	Configure SNMP settings in the codec.
4	**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.
5	**Tieline Session**	Edit the Tieline session and alternative session port used by the codec.
6	**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**.
7	**Edit button**	Press to edit settings in the **Options panel**.

Help Panels

Support Panel
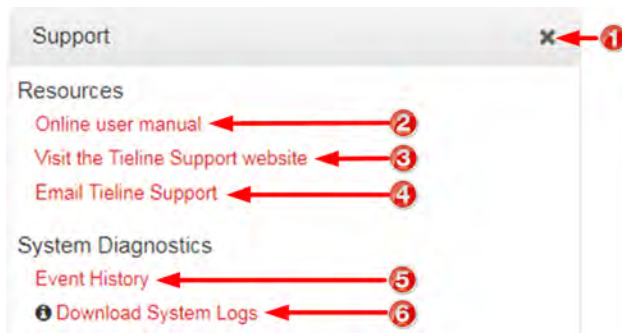
	Feature	Description
1	**Close** button	Click to close the panel.
2	User manual link	Click to open the codec user manual in a new browser, or view support information (Note: the codec name displayed will vary by product type)
3	Support website link	Click to visit the support page on the Tieline website.
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 in this panel displayed include the Toolbox and firmware version, as well as the codec serial number. Note: The codec name displayed varies as per how many channels are enabled in the codec. E.g. An 8 channel codec will display **Gateway 8**. The following image confirms this is a 16 channel I/O unit named **Gateway 16**.
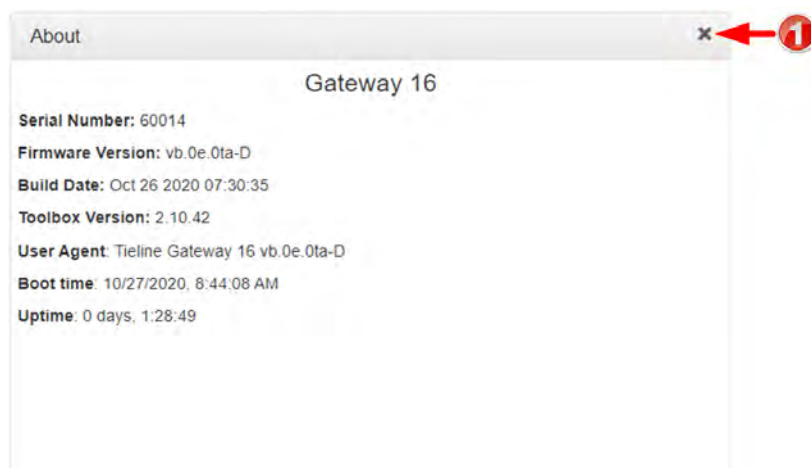
	Feature	Description
1	**Close** button	Click to close the panel.

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.

23.1   Configure Programs with the Connections Panel

Product	Feature	Notification
Gateway 4	Programs	Supports program connections using up to 4 channels of audio
Gateway 8/16	Programs	Supports program connections using up to 16 channels of audio, depending on codec configuration purchased (between 8 and 16 channels available for purchase)

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**.
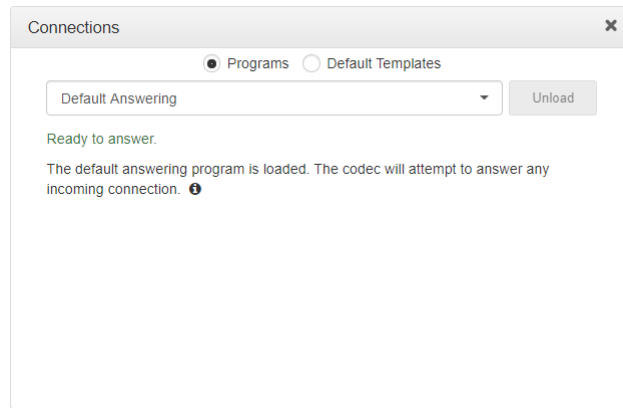
2. Select **Default Templates** and then click the drop-down arrow to select one of the supported default templates in the codec. Note: Default templates available vary depending on the codec purchased, e.g. Gateway 4, Gateway 8, and Gateway 16.
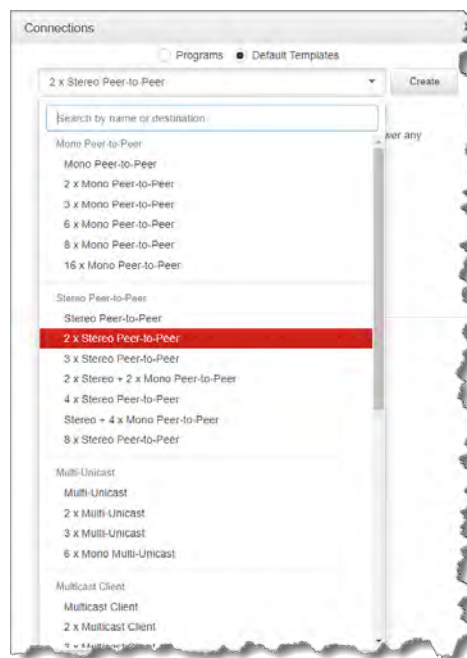
3. Click **Create** to add the new program.
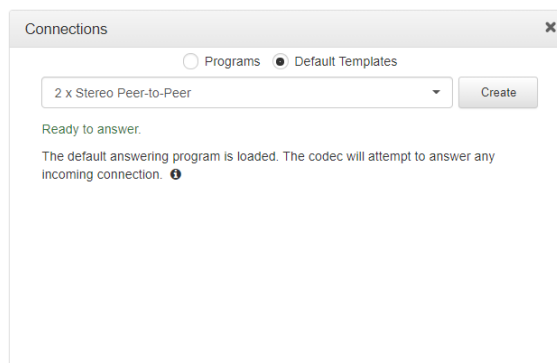
4. Click the right-facing arrow ▶ to expand an audio stream and view default settings.
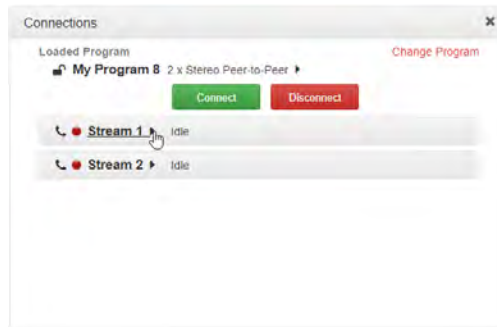
5. To configure new connection settings, or edit existing connection settings, click the **Edit symbol** 🖉.
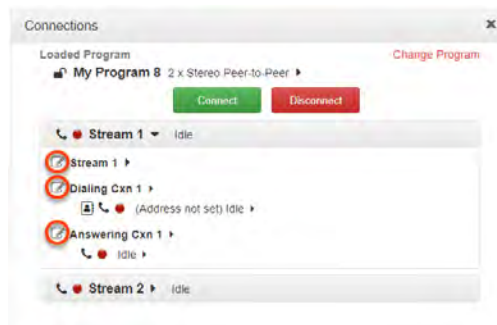
6. The **Edit symbol** 🖉 expands each audio stream and dialing and answering connection to reveal configuration settings. Hover over the **Information symbol** ❶ to view details of each configuration setting. Click **Save** to store changed settings.
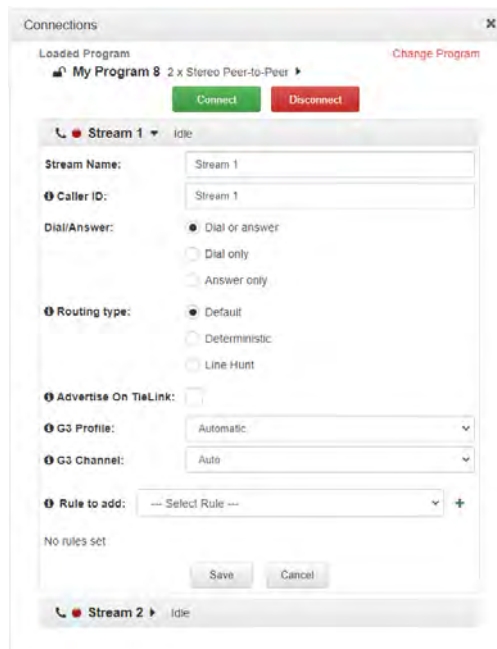
7. To connect an individual connection, click the **Connect/Disconnect** 📞 symbol for an inactive connection.
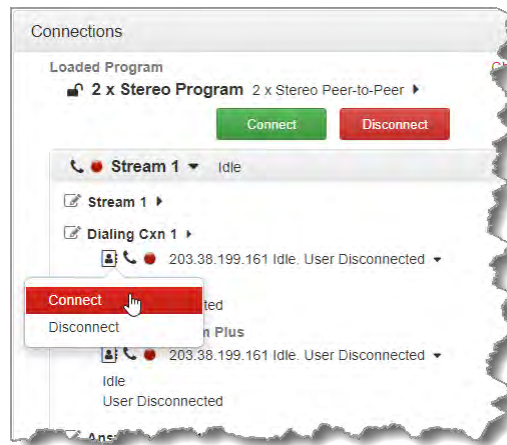
8. Click the right-facing arrows ▶ to expand stream settings. Click the down-facing arrows ▼ to collapse expanded stream settings. Audio stream connections can be connected, disconnected and managed from within the **Connections panel**.
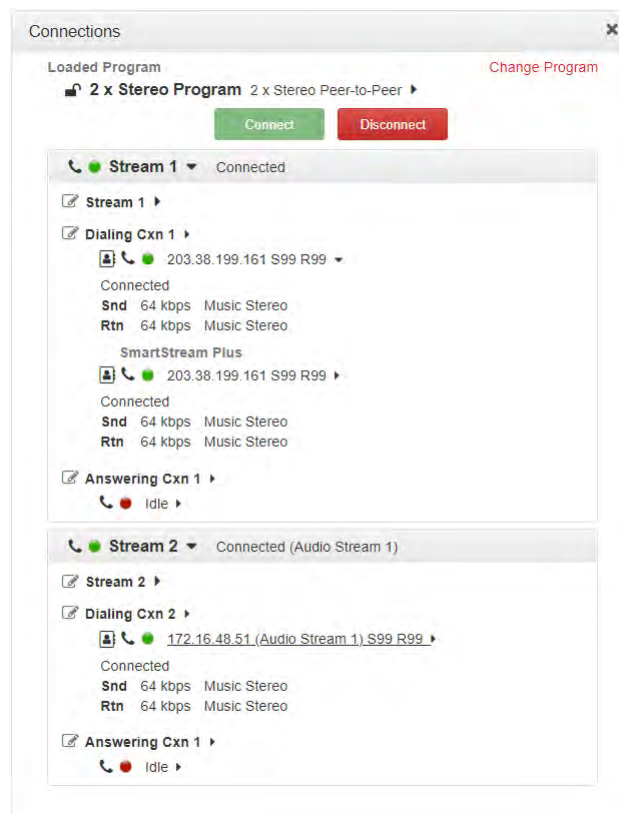
9. Click the **Connect/Disconnect** ☎ symbol for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate.
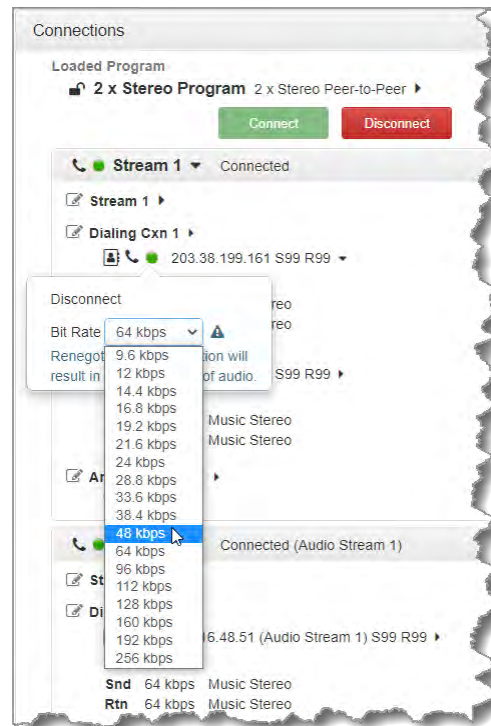
Important Notes:
- Renegotiation will result in a temporary loss of audio.
- It is not possible to renegotiate the connection bit rate of a SIP connection.

10. To disconnect an individual connection, click the **Connect/Disconnect** 📞 symbol for an active connection.
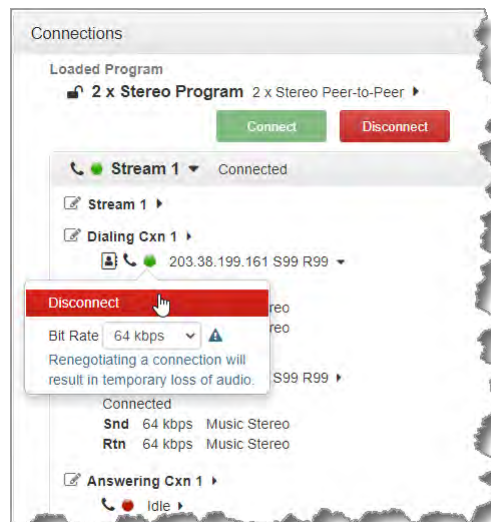
23.2   Using the HTML5 Toolbox Quick Connect Web-GUI

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.

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.

> **ⓘ** **Important Notes:**
> - Simple peer-to-peer connections configured in the **Quick Connect panel** 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**.

To enable the **Quick Connect panel** press the **SETTINGS** 🔧 button, then navigate to **WebGUI > IP Options > Quick Connect > Enabled**.

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.

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.
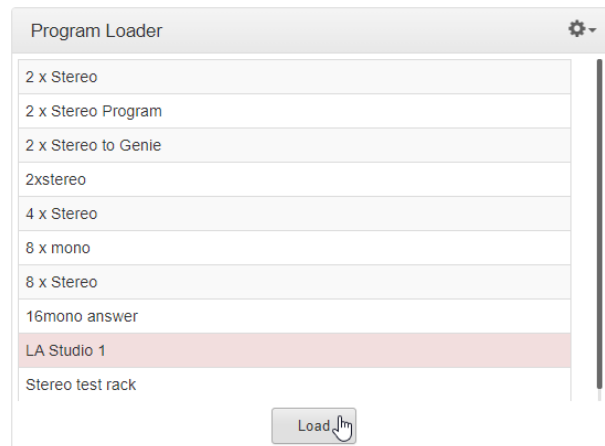
2. The **Check-box symbol** appears next to the program name to confirm it has been loaded and the **Load button** changes to an **Unload button**.
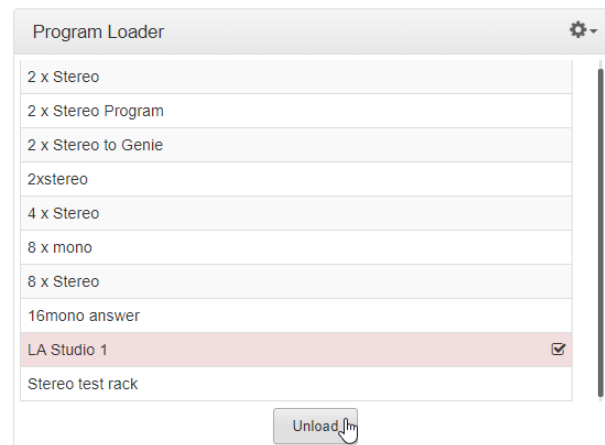
To unload a program click the **Unload button**.

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**.
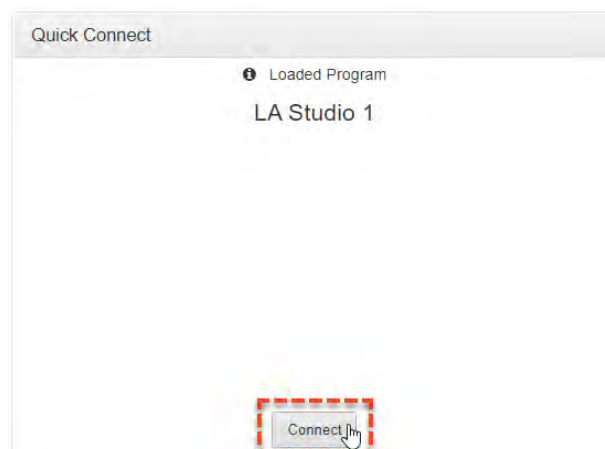
Disconnect a Loaded Program

1. Click the **Disconnect button** in the **Quick Connect** panel.
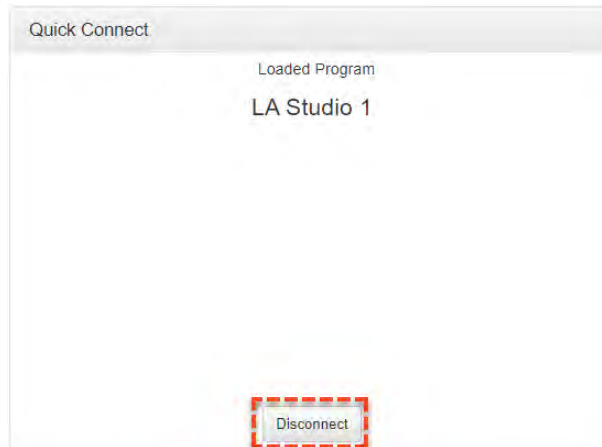
2. Click **Yes** in the confirmation dialog to disconnect the connection.
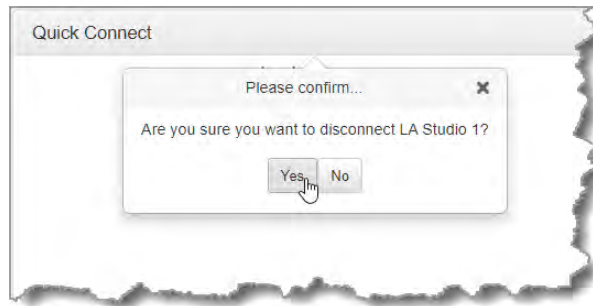
Dial Peer-to-Peer over IP with Quick Connect

Important Notes:
- Click the **Unload button** in the **Program Loader panel** if a program is currently loaded.
- The transcriptor algorithm is for closed captioning and not normal broadcast configurations.

1. Click the drop-down **Transport** menu arrow in the **Quick Connect panel** and select **IP**.
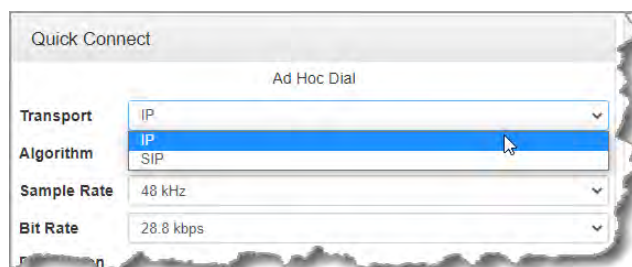
2. Click the drop-down **Algorithm** menu and select an algorithm.
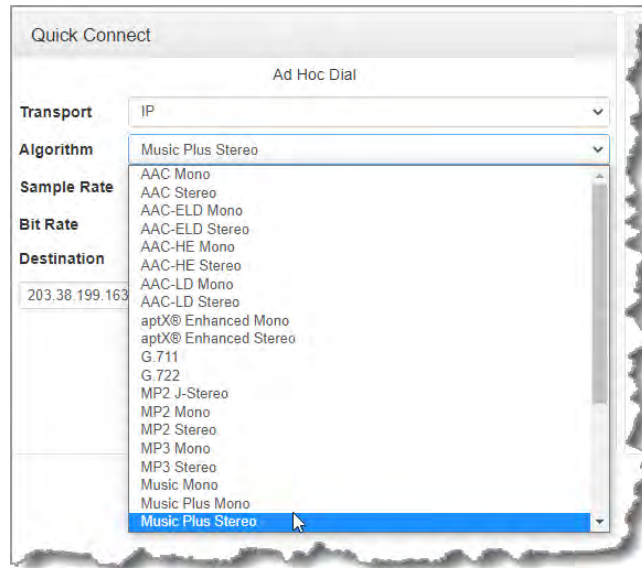
3. Click the select the appropriate **Sample Rate** and **Bit Rate** for the connection. Note: If only one sample rate is available this will be automatically selected.
4. Click in the **Destination** text box and enter the IP address of the destination codec, or click the **Address Book button** 📇 to select a preconfigured TieLink contact.
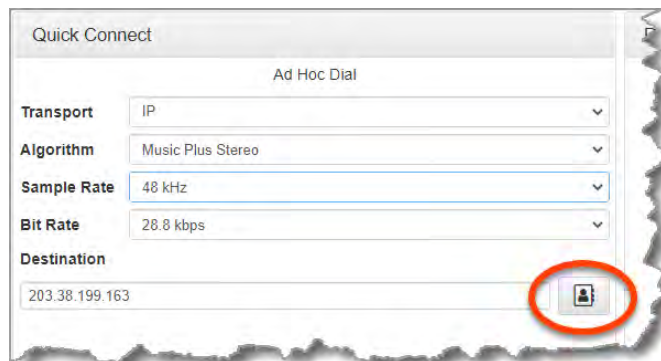
5. Click the **Connect button** to dial.

Monitoring PPMs

See Monitoring PPMs for details of how to monitor audio streams using the **PPMs panel**.

23.3   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.

> ℹ️ **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 port interfaces and up to four additional VLAN interfaces.

VLAN interfaces have features similar to physical Ethernet interfaces. However, a network administrator needs to configure VLAN support across a LAN for them to be supported in the codec.

As an example, if only one physical Ethernet interface is available, VLANs can be used to operate SmartStream PLUS or to separate codec **Control and Streaming** functions if required. Ethernet and VLAN interfaces can be configured for:

- Controlling audio: codec control and command only from the Ethernet port.
- Controlling and Streaming: stream audio and control and command the codec via the Ethernet port.
- Streaming audio: stream audio only from an Ethernet port.
- Nothing: Disable the Ethernet port from streaming audio and codec command and control.

To edit control and streaming settings:

1. Click an interface to expand the details for the selected interface.
2. From the **Details** tab click the **Edit** button.
3. Select the **Details tab** and configure control and streaming as required.
4. Click **Save**.
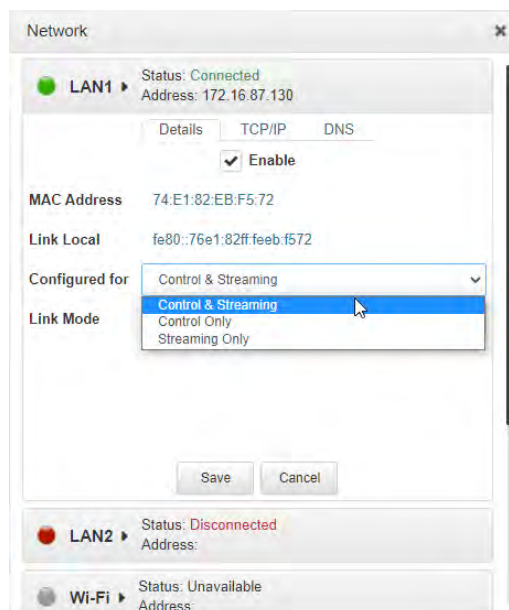
To enable or disable an interface select the **Enable** check-box when in **Edit** mode, then click **Save**. **LAN1** and **LAN2** are enabled by default. Please note that the **Status LED** color indicates the status of each network interface.

1. Any interface with a grey **Status LED** indication is disabled.
2. Any interface with a green **Status LED** indication is enabled and active.
3. Any interface with an amber **Status LED** indication is enabled and attempting to connect.
4. Any interface with a red **Status LED** indication is enabled and unavailable for some reason, e.g:
   - Ethernet cable is not connected.
   - Incorrect configuration, e.g. VLAN Identifier not correct.
   - Wi-Fi access point is not selected.

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 Mode** arrow and select the preferred setting.
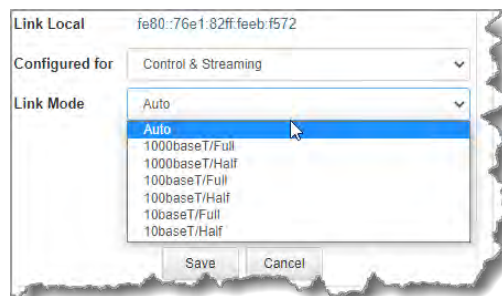
2. Click **Save** to store the new setting.

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**.
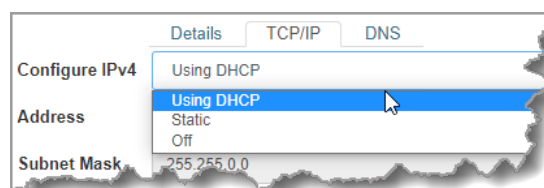
DHCP IP addresses are automatically assigned and can change each time you connect to your Internet Service Provider, or to your own local area network (LAN). By default the codec is configured for DHCP-assigned IP addresses.

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.

> **Important 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 drop-down **Configure IPv6** 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.
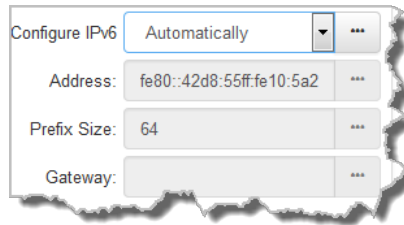
Types of IPv6 Addresses

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

1. By default the codec is configured for connecting to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.
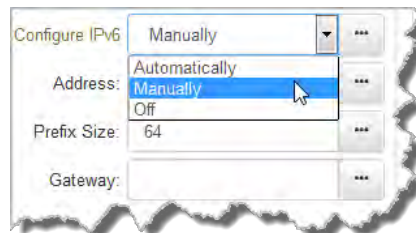
2. Click **Save** to store all configuration settings.

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.
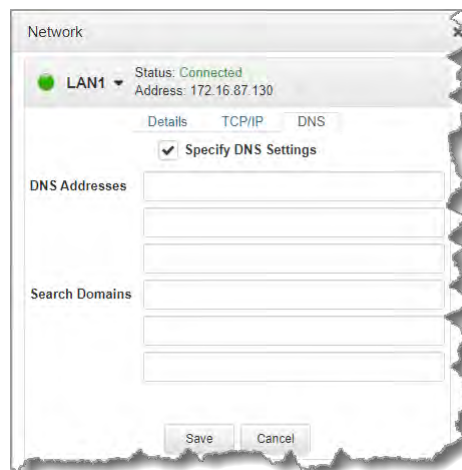
2. Click **Save** to store all configuration settings.

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 for the interface in the **Network** panel.

2. Click **Edit** to configure settings.
3. 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 is 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. **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 an Ethernet interface and for another it may be a Wi-Fi connection. 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 **Transport** 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.
3. Click **Save** to store the new configuration settings.
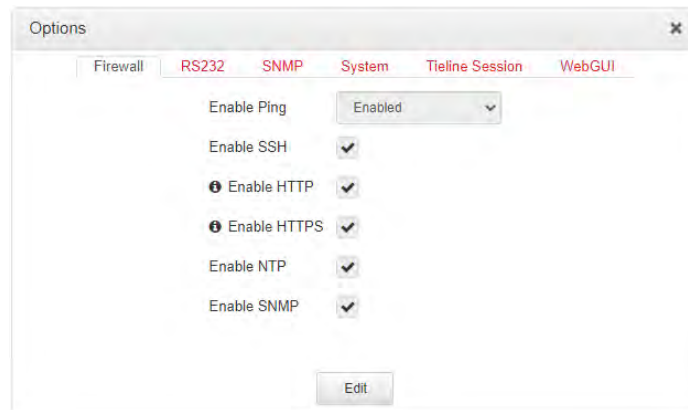
Configure Cross-Site Request Forgery

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.
3. Click **Save** to store new configuration settings.
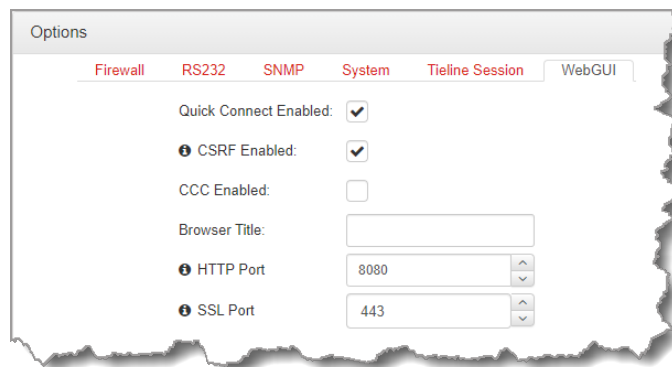
Configuring QoS

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.
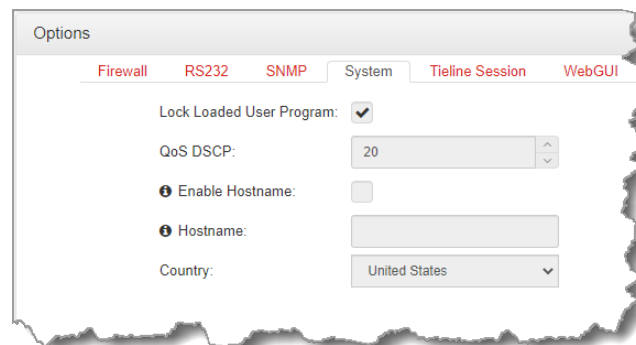
> **Important Note:** Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

23.4   Enabling the Cloud Codec Controller

For 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.
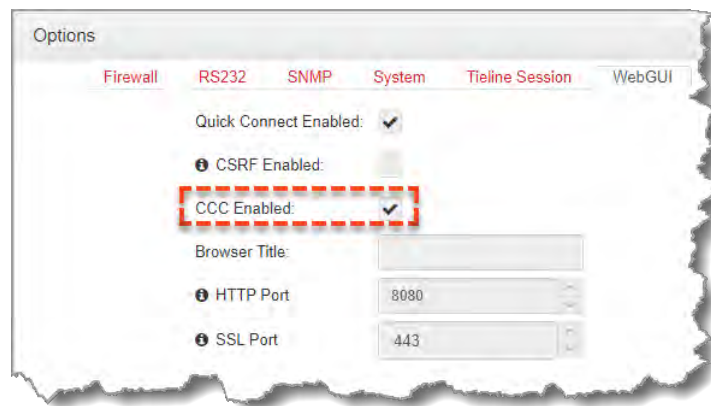
3. Click **Save** to store the new configuration.

> **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.

23.5   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, which may include:

- Dual Ethernet LAN ports.
- An Ethernet LAN port with Wi-Fi.

See Connecting with Fuse-IP for more details on configuration using the codec front panel controls.

Configuring a Fuse-IP Server at the Studio

1. Open the HTML5 Toolbox Web-GUI and click **Transport** 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 and select **Server** if the codec is not initiating the connection.

4. 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.

5. Click the **Save** button to store all settings.
6. Click the **On/Off** button to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds if both codecs are configured correctly and Fuse-IP is enabled on both codecs.
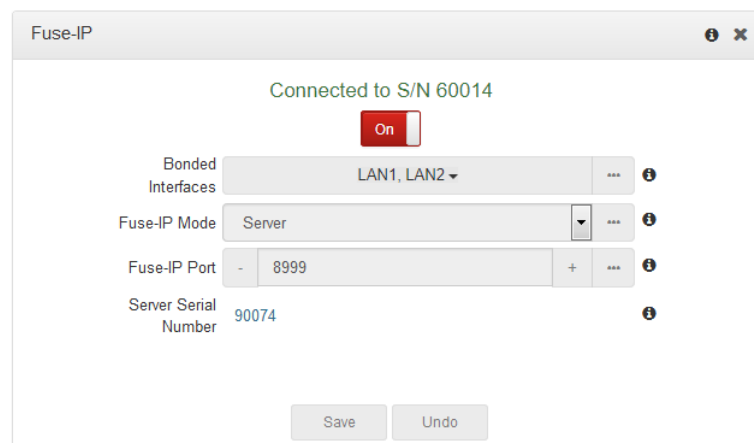
7. Double-check all settings on both the server and client codecs if the message **Started, dialing server...** persists after turning Fuse-IP **On**.
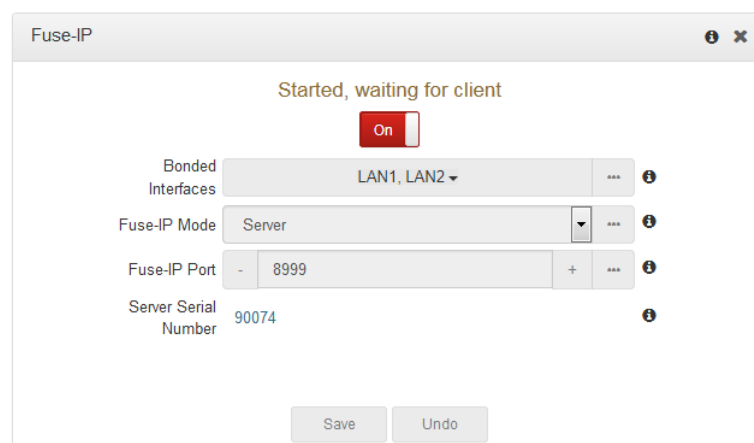
Configuring a Fuse-IP Remote Client

1. Open the HTML5 Toolbox Web-GUI and click **Transport** and then **Fuse-IP**.
2. Click the **Bonded Interfaces** drop-down menu to select the interfaces to be bonded.
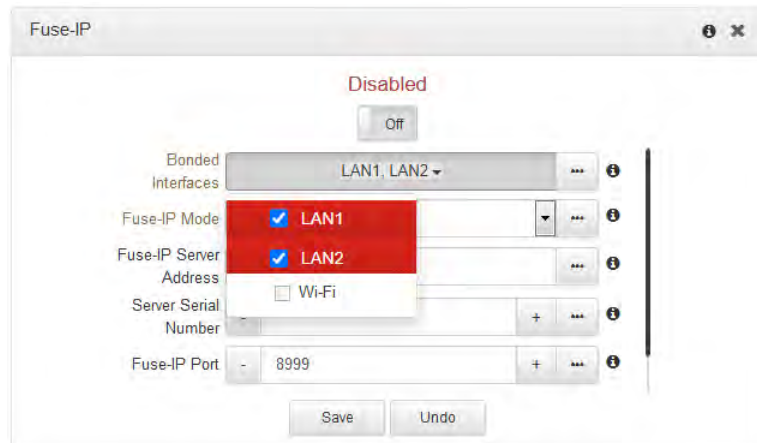
3. Click the **Fuse-IP Mode** drop-down menu to select **Client** as the Fuse-IP mode if dialing the studio codec. Note: the studio codec should be configured in server mode.

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.
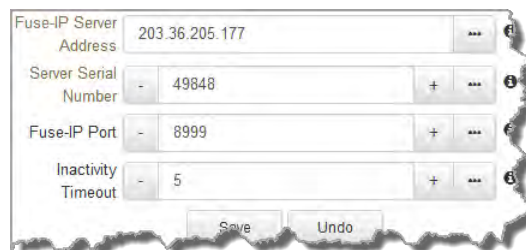
5. Click the **On/Off** button for Fuse-IP to create a Fuse-IP tunnel between the server and client codecs. Note: **Connected** should be displayed after a few seconds if both codecs are configured correctly and Fuse-IP is enabled on both codecs.
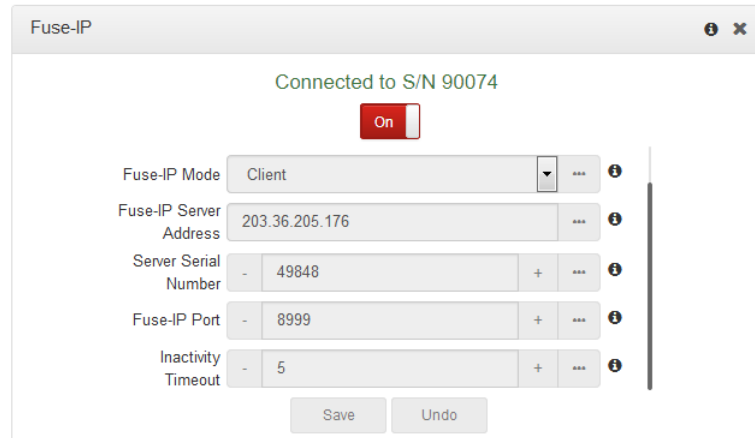
6. The status indicator is orange when Fuse-IP is enabled but no tunnel is created. It turns green when a tunnel is active.

Fuse-IP Enabled & no tunnel created

Fuse-IP 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**.
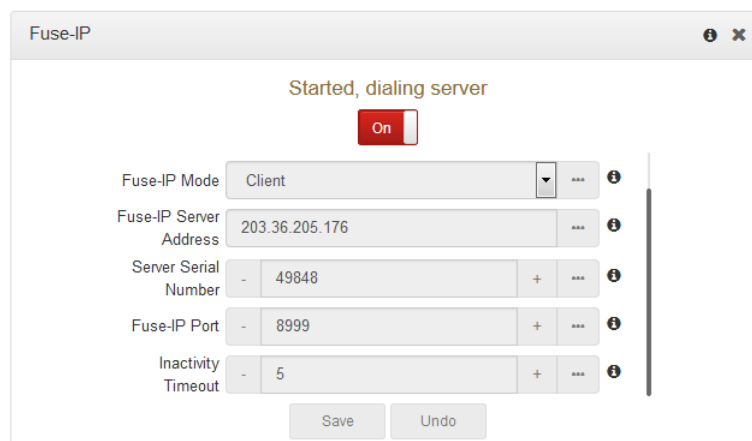
Important Notes:
- Data is sent by the codec over the newly created 'tunnel' as soon as Fuse-IP is enabled, even if a connection has not been configured and dialed. Depending on the number of interfaces being used, the codecs may transmit and receive up to 24MB of data per hour at each end of the link.
- 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.

23.6   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 sixteen mono audio streams, select **Line Hunt** as the routing type. Then select **Group 1** for the first eight audio streams, to route outgoing and incoming calls via inputs and outputs 1 to 8. These inputs and outputs on the codec can be routed to a particular studio or station.
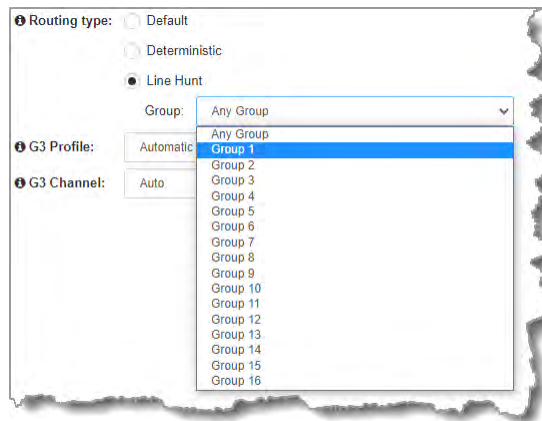
For the next eight audio streams, select **Group 2** to route outgoing and incoming calls via inputs and outputs 9 to 16. These inputs and outputs on the codec can be routed to a different studio or station.
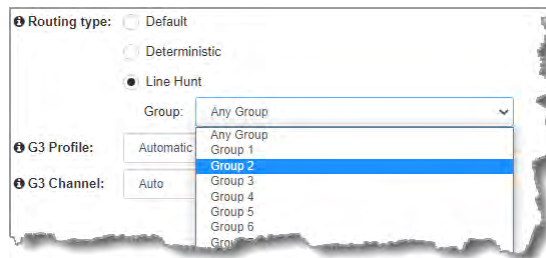
In a 16 stream answering program example, when **Any Group** is selected for all streams in **Line Hunt** mode, the codec will route all incoming calls on a first come, first served basis. The dialer doesn't have to specify a line hunt group, however they do need to select **Line Hunt** mode in the dialing program for this to function correctly.

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.
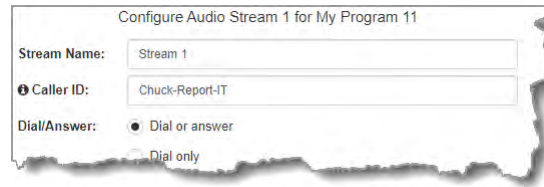
Any Tieline G5 codec dialing can display a designated **Caller ID** in the **Connection panel**. In the following example, **Remote1** has called into the codec using a specific caller ID, which is displayed next to the **Send** and **Return** link quality connection.
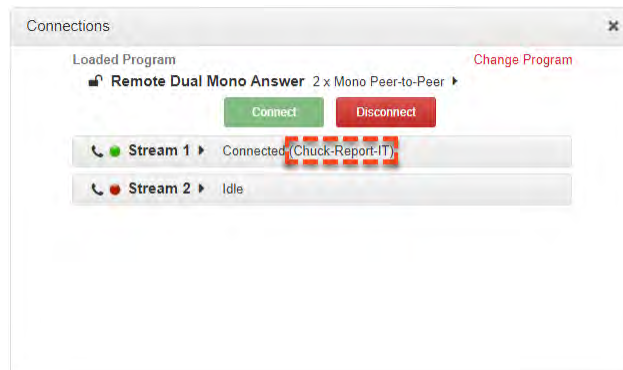
23.7   Configuring Input Settings

Product	Feature	Notification
Gateway 4	PPM Meters	8 PPMs on front panel display 4 channel inputs/outputs or encoders/decoders
Gateway 4	Inputs	4 Channels in/out
Gateway 4	Inputs	Inputs 1-4 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 4	Web-GUI Inputs panel	4 inputs and PPMs displayed
Gateway 8/16	PPM Meters	16 PPMs on front panel can display inputs/outputs/encoders/decoders
Gateway 8/16	Inputs	8-16 Channels In/Out (depending on codec configuration purchased)
Gateway 8/16	Inputs	Inputs 1-8 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 8/16	Web-GUI Inputs panel	8 - 16 inputs and PPMs displayed (depending on codec configuration purchased)

Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Inputs** to display the Inputs panel.
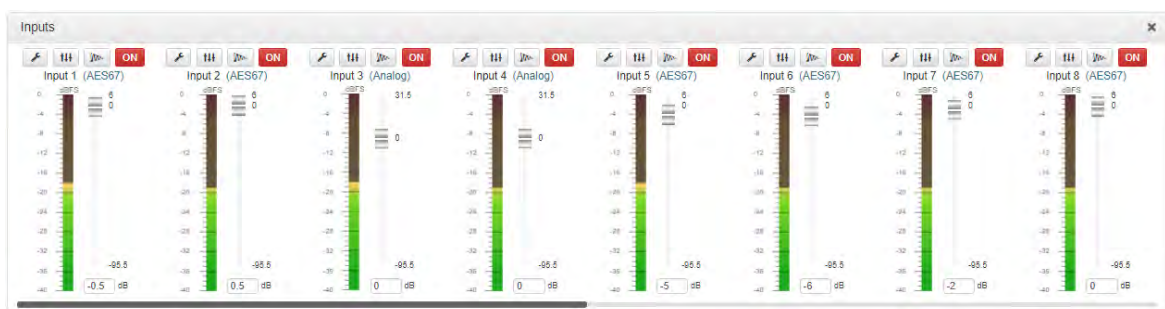
Gateway (8/16) input panel
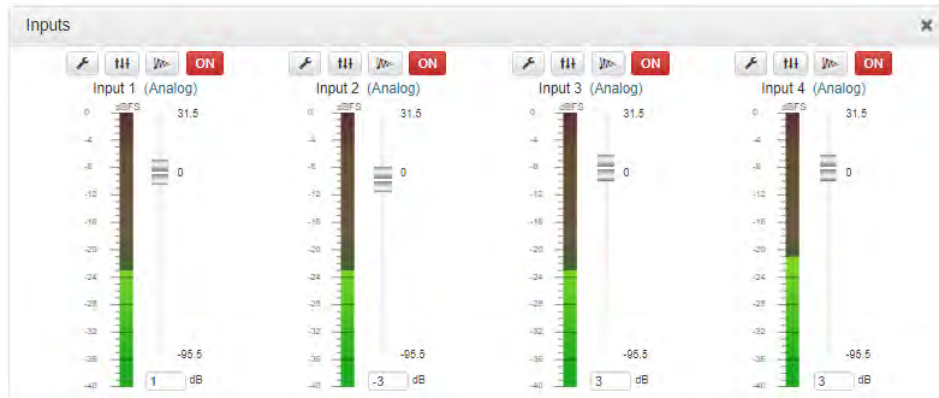
Gateway 4 inputs panel

Adjusting Audio Input Levels

To adjust input audio levels, click on the input slider and drag it to the desired input gain level. Alternatively, type a value below a PPM meter to increase or decrease the input level value. Note: 0.5dB increments are accepted for increasing and decreasing audio levels. 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 -20dBFS. 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.

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 **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**, next click **Edit**.
2. Click the **Analog Inputs PPM Units** drop-down menu and select **dBU**, then click **Save**.
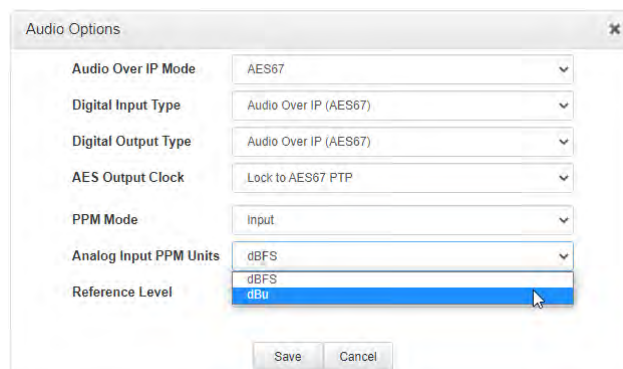
Configuring Input Settings

Selecting Analog and Digital Audio Sources

Codec inputs can be configured for AES3, AES67 or analog line level audio sources. Use the **Audio Options panel** in the HTML5 Toolbox web-GUI to configure digital inputs globally for

either AES3 or AES67 AoIP input sources. See Input Configuration, Levels and PPMs for more details. After setting the **Digital Input Type** in the **Options panel**, configure the codec for either analog or digital input types using the **Inputs panel** as follows.

1. Click the **Input Settings** 🔧 symbol.
2. Select **Type** and then **Analog** or **AES3**.
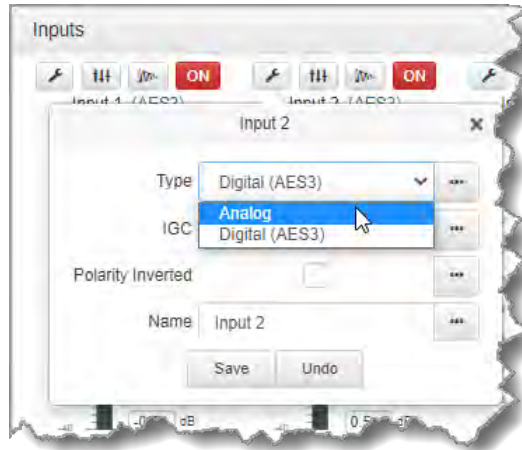
3. Click **Save** to confirm the new setting.

Adjusting IGC

Input IGC (Intelligent Gain Control) is enabled by default and is automatically activated at +19 dBu (G6 audio scale), +17 dBu (G5 audio scale) and +13dBu (G3 audio scale) to prevent audio clipping.

1. Click the **Input Settings** 🔧 symbol.
2. Select **IGC** (Intelligent Gain Control) and then **Auto**, **Fixed** or **Off** as required.
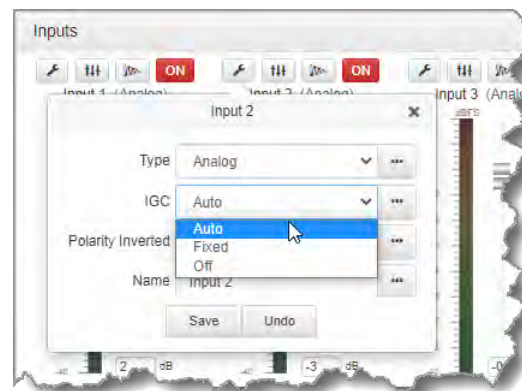
3. Click **Save** to confirm the new setting.

Invert Polarity

Select the **Polarity Inverted** check-box to reverse the polarity of an analog or digital input.

1. Click the **Input Settings** 🔧 symbol on the input you want to adjust.
2. Click to select the **Polarity inverted** check-box.
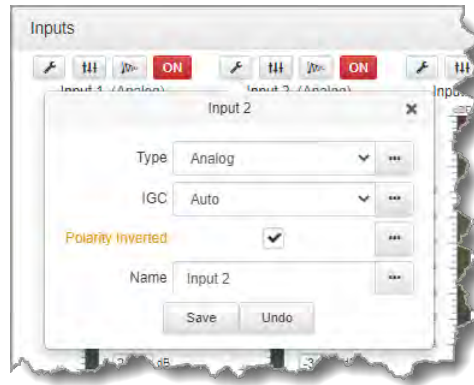
3. Click **Save** to store the new setting.

Renaming Inputs

1. Select the **Input Settings** 🔧 symbol on the input being renamed.
2. Click in the **Name** text box to enter a new name, or edit an existing name.

3. Click **Save** to confirm the name change.

Adjust EQ Settings

EQ and compression can be adjusted for each input if required. To enable and adjust the built-in 10 band EQ:

1. Click the **EQ button** 🎚️ for an input in the **Inputs panel**. The **EQ button** is blue 🎚️ when input EQ is enabled.
2. Click the **Graphic button** to enable EQ for the input if it is in **Bypass**.
3. Adjust the EQ frequency sliders as required.
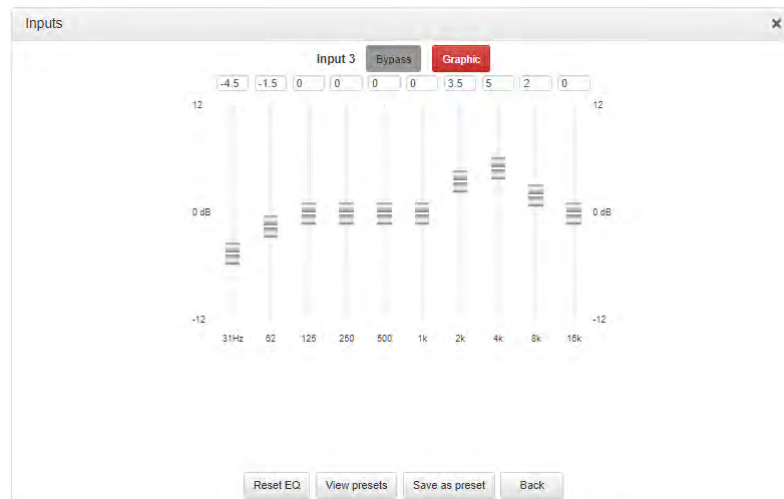
4. Click **Save as preset** to save an EQ preset, e.g. for a particular announcer. Click **View presets** to recall a previously stored EQ preset.
5. Click **Back** to view all inputs.

Adjust Compression Settings

To to enable and adjust input compression settings.

1. Click the **Compressor button** [icon] for an input in the **Inputs panel**. The **Compressor button** is blue [icon] when input compression is enabled.
2. Click the **Disabled button** to enable compression for the input if it is disabled.
3. Adjust the compression settings as required.
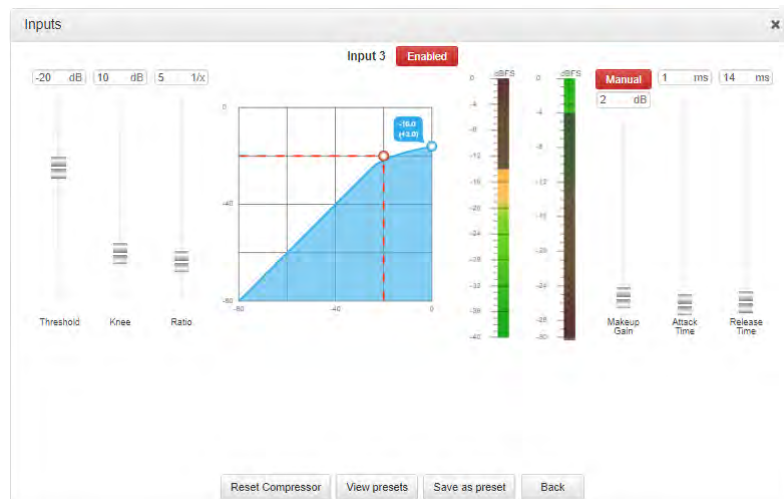
4. Click **Save as preset** to save a compressor preset, e.g. for a particular announcer. Click **View presets** to recall a previously stored compressor preset.
5. Click **Back** to view all inputs.

> **Important Note:** 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).

AES3 Output Sample Rate Configuration

The AES3 output sample rate can be configured using the HTML5 Toolbox Web-GUI. See Configuring AES3 Audio for more information about AES3 input and output sample rate clocking.
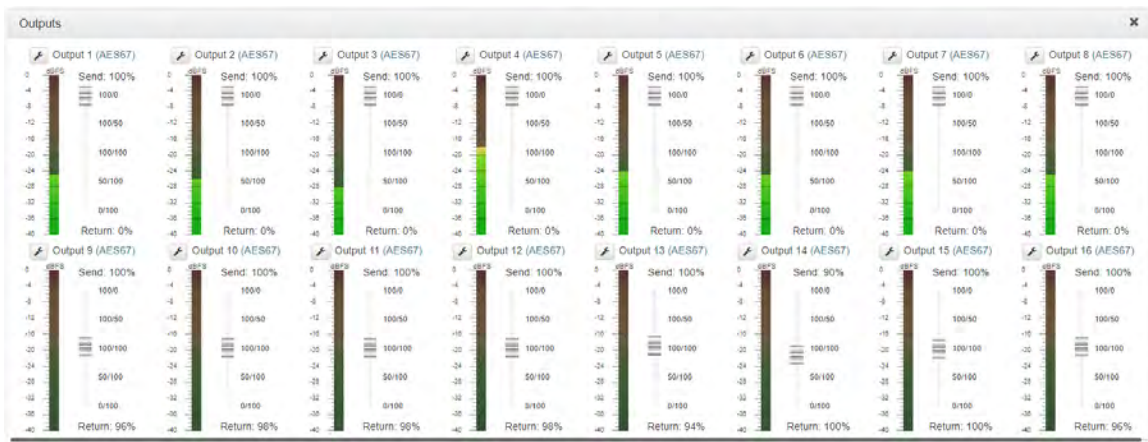
1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **AES Output Clock** drop-down menu to select your preferred **AES Output Clock** setting, then click **Save**.
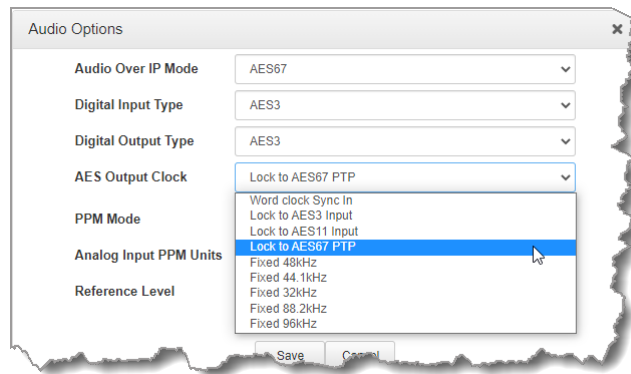
Audio Reference Levels

By default, the **PPM METERS** on the front of the codec, 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 G6** reference scale setting.

The default Tieline G6 audio reference scale displayed on the PPMs when connecting to a Tieline G6 codec is -40dBFS to 0dBFS. Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 20dB of headroom from the nominal 0vu point.

> **Important Note:** If a codec supports multiple stream programs and the **Auto** (default) reference level is selected, the first codec to connect will configure the reference level used for all subsequent connections. I.e. If a G5 codec connects first then the G5 Audio Reference Level will be configured for all connections.

To adjust this setting:

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **Reference Level** drop-down menu and select the correct option, then click **Save**.
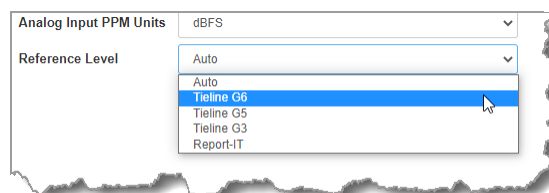
23.8   Configuring Output Settings

Product	Feature	Notification
Gateway 4	Outputs	Outputs 1-4 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 4	Web-GUI Outputs panel	4 Outputs and PPMs displayed
Gateway 8/16	Outputs	Inputs 1-8 can be configured as analog or digital inputs (AES3 or AES67) in pairs
Gateway 8/16	Web-GUI Outputs panel	8 - 16 Outputs and PPMs displayed (depending on codec configuration purchased)

The **Outputs panel** can be configured to support either analog or digital output types for outputs 1 to 4 (Gateway 4) or outputs 1 to 8 (Gateway 8/16). Configuration of the digital output type, either AES3 or AES67, is global and is configured for all 16 Gateway outputs using the **Audio Options panel**.

Gateway-16 Outputs panel
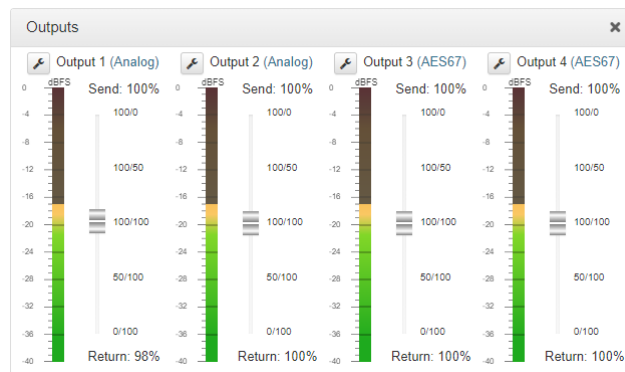
Gateway-4 Outputs panel

Selecting the Global Digital Output Type

To select the digital output type for all outputs:

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **Digital Output Type** drop-down menu and select either **Audio Over IP (AES67)** or **AES3**, then click **Save**.
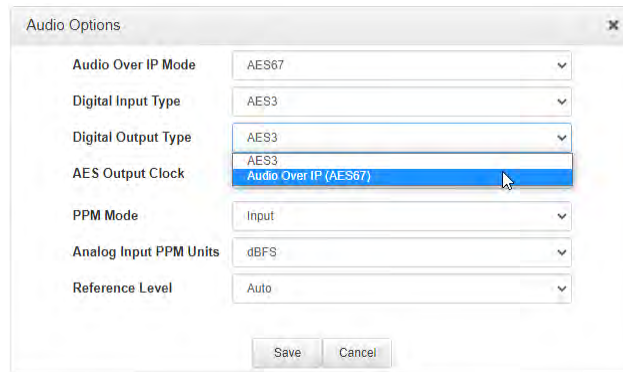
Select Output Type as Analog or Digital

To select either analog or digital output types for outputs 1 to 8 (in pairs):

1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Outputs** to display the **Outputs panel**.
2. Click the **Settings** button.
3. Select from **Analog** or **Digital**.
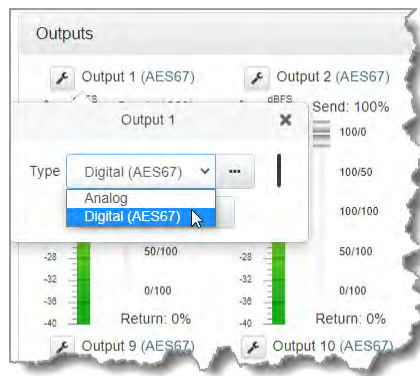
Output Send / Return Balance

The balance between send and return audio signals fed to each output can be individually adjusted using each output slider. This provides flexibility and the ability to feed input sources and decoded incoming signals to outputs as required.
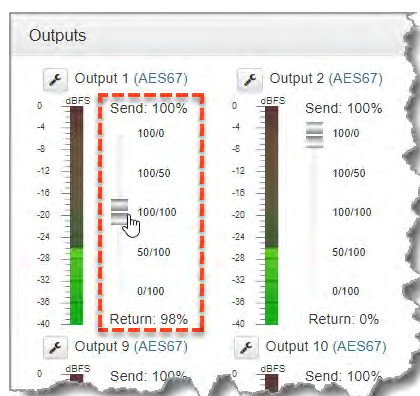
In many situations incoming **Return** decoder audio will be the only audio fed from an output, however there may also be a requirement to mix input audio with return audio. Or input (**Send**) audio may be fed directly to an output for AES67 streaming.

Output Assignments for AES67 Source Audio Streams

When streaming in AES67 mode, the hard coded **Matrix Editor** output channel assignments depend on the stream configuration mode, as displayed in the following table.

AES67 Audio Stream	Mode	Mix Outputs Used
Source 1	Mono	1
	Stereo	1 and 2
	8 Channel	1 to 8
Source 2	Mono	2
Source 3	Mono	3
	Stereo	3 and 4
Source 4	Mono	4
Source 5	Mono	5
	Stereo	5 and 6
Source 6	Mono	6
Source 7	Mono	7
	Stereo	7 and 8
Source 8	Mono	8
Source 9	Mono	9
	Stereo	9 and 10
	8 Channel	9-16
Source 10	Mono	10
Source 11	Mono	11
	Stereo	11 and 12
Source 12	Mono	12
Source 13	Mono	13
	Stereo	13 and 14
Source 14	Mono	14
Source 15	Mono	15
	Stereo	15 and 16
Source 16	Mono	16

Verify audio is displayed on the PPMs in the **Sources** panel before commencing AES67 streaming.

23.9   Monitoring PPMs

Product	Feature	Notification
Gateway 4	PPM Meters	8 PPMs on front panel display 4 channel inputs/outputs or encoders/decoders
Gateway 4	Web-GUI PPMs Panel	Capable of monitoring 4 inputs, 4 outputs, 4 encoders and 4 decoders simultaneously
Gateway 8/16	PPM Meters	Gateway 8 has 8 PPMs on codec front panel capable of monitoring 8 inputs, 8 outputs, 8 encoders or 8 decoders; Gateway 16 capable of monitoring 16 inputs, 16 outputs, 16 encoders or 16 decoders
Gateway 8/16	Web-GUI PPMs Panel	Gateway 8 capable of monitoring 8 inputs, 8 outputs, 8 encoders or 8 decoders simultaneously; Gateway 16 capable of monitoring 16 inputs, 16 outputs, 16 encoders and 16 decoders simultaneously.

The **PPMs panel** is able to monitor incoming and outgoing audio streams in the codec.
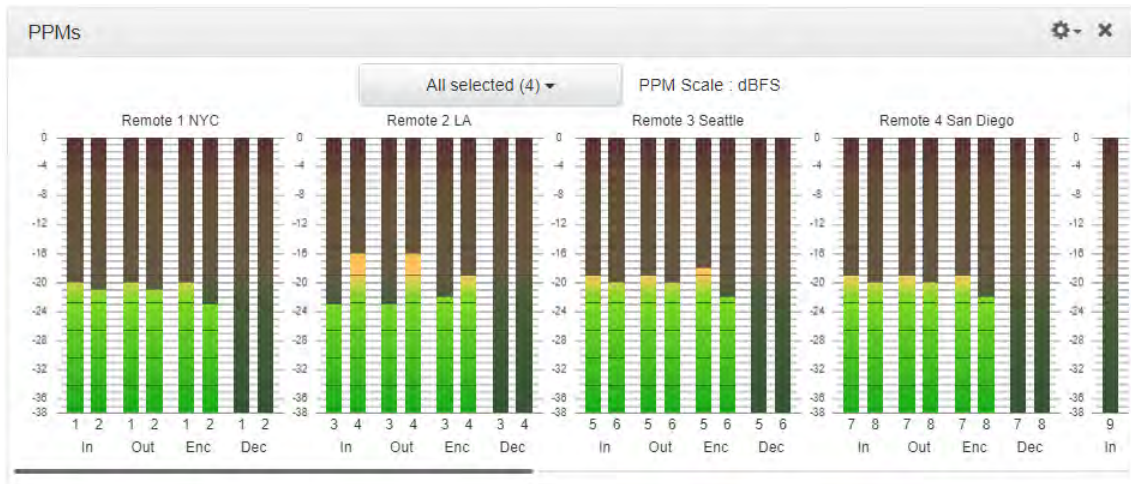
PPMs grouped by stream and identified by Stream Name

Click the drop-down PPM selector arrow to select whether the panel will display PPMs for:

- Inputs
- Outputs
- Encoders
- Decoders.

Select the check-boxes to select or deselect the PPMs displayed for each audio stream.
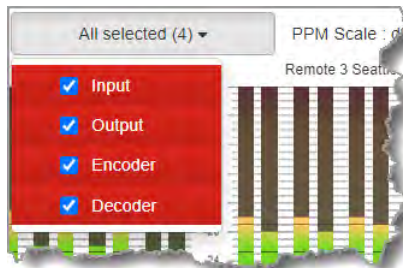
Click the **Options** menu in the panel to choose whether PPMs are grouped by stream or by type.
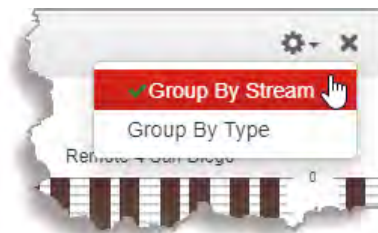
PPMs grouped by type appear as follows:
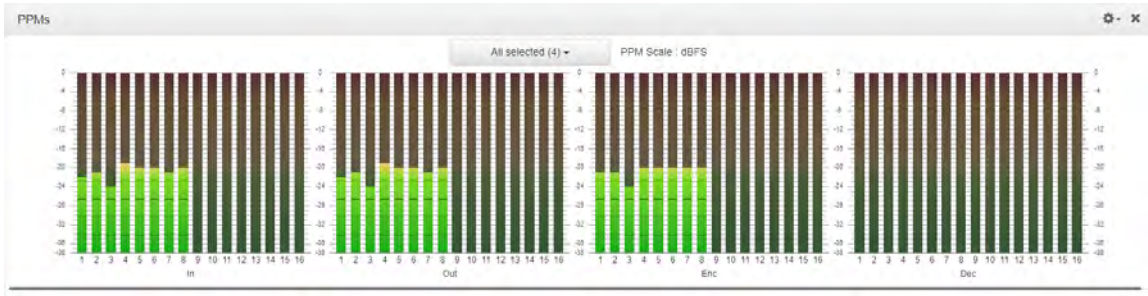
AES67 Stream PPMs

PPMs for AES67 streams are displayed in the **AES67 Destinations panel** and **AES67 Sources panel** in the Toolbox HTML5 Web-GUI and the AES67 Web-GUI. For more details please download the AoIP User Manual.

Toolbox AES67 Destinations panel displaying incoming codec audio

Toolbox AES67 Sources panel displaying outgoing codec audio

Important Note: PPMs on the AoIP **Destinations panel** are inactive if the **Digital Input Type** is not set to **AES67** and the corresponding input **Type** is not set as **Digital**.

23.10   Configure Dial and Answer Programs

Product	Feature	Notification
Gateway 4	Programs	Supports program connections using up to 4 channels of audio
	Program Manager panel	Displays program types that support 4 channels in/out
Gateway 8/16	Programs	Supports program connections using up to 16 channels of audio, depending on codec configuration purchased (between 8 and 16 channels available for purchase)
	Program Manager panel	The Gateway 8 displays program types that support 8 channels in/out; the Gateway 16 displays program types that support 16 channels in/out.

The **Program Manager 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.

The following section contains instructions for:

1. Configuring Point-to-Point Programs: Dialing
2. Configuring a Failover Connection or Auto Reconnect
3. Configuring the Codec to Answer Connections
4. Configuring Output Source Failover options, e.g. File Playback on Silence Detection.

For more information about programs and audio streams within programs see the section titled IP Streaming, Programs and TieLink. Note: The following instructions will display how to configure a dial and answer program, with a backup connection and file playback. If the codec will either dial or answer only, select the option and the wizard will automatically display relevant screens to allow you to configure the codec correctly.

Creating Multistream Programs in the Gateway

When configuring multiple stream programs use the same process to configure each audio stream in turn. Port numbers are automatically assigned and can be adjusted as required. The default Tieline ports are outlined in Default Program Ports. A wide range of different program types are available for different contribution and distribution streaming applications.
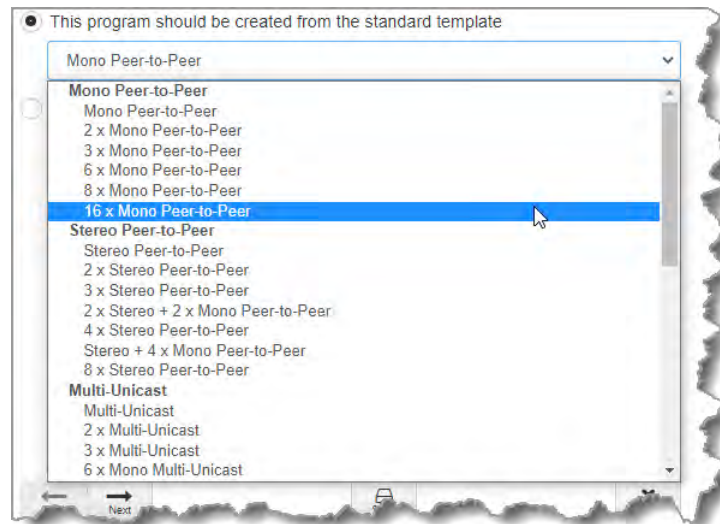
Some Programs displayed in the Gateway-16

Configuring a Mono or Stereo Peer-to-Peer Program: Dialing

> **Important Notes:** Before configuring a program please note:
> - A program cannot be edited in the **Program Manager** when loaded in the codec.
> - Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different program type. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
> - Some drop-down menus and settings may be greyed out intentionally depending on features available.
> - A program can be saved at several points throughout the program wizard and use default settings to save configuration time.
> - Failover is not available with SIP.
> - SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

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.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. 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 Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. (WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network. WheatNet logical inputs are only available if a codec has a WheatNet-IP card installed). 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 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.
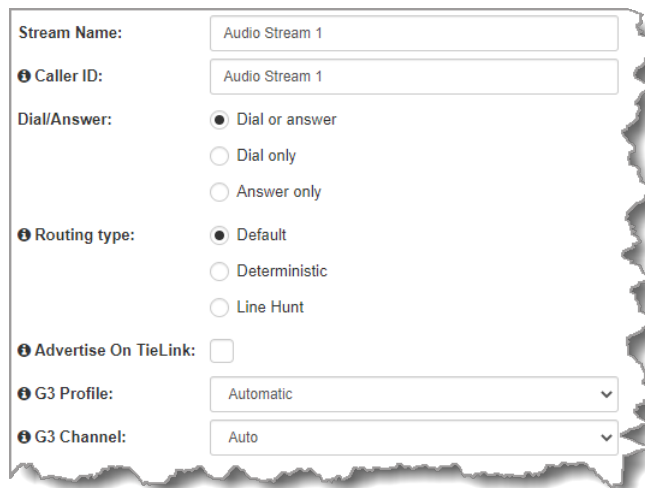
Routing Type Options:	
Default ISDN	No **Dial Route** or **Answer Route** is configured. An incoming call will be routed to an audio stream on a first-come, first-served basis in a multi-stream program. Note: By default IP streams are routed using audio ports.

Deterministic IP ISDN	Select a **Dial Route** or **Answer Route** to configure deterministic routing of multiple audio streams using ISDN. 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 both ISDN connections and IP at different times. This effectively creates an answering group using different transports. See Configuring ISDN Answering for more information.
Line Hunt IP ISDN	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 or G6 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 or G6 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**. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm.

Configure Transport settings Audio Stream 1->Dialing Cxn 1

ℹ Transport:	IP
ℹ Session Protocol:	Tieline Codecs
ℹ Encode/Decode:	Both
ℹ Connect Timeout:	120 sec
ℹ Enable Auxiliary Data:	☐
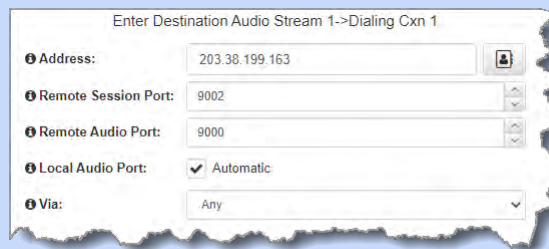
> **Important Notes:**
> - If you select **Sessionless** as the **Session Protocol** select **UDP/IP +RTP** for RFC-compliant IP streaming.
> - See RS232 Data Configuration for detailed information on RS232 data and see Enabling Relays and RS232 Data for more information on relay operations.

7. Configure destination codec dialing and encoding settings:

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.

Enter Destination Audio Stream 1->Dialing Cxn 1

ℹ Address:	203.38.199.163
ℹ Remote Session Port:	9002
ℹ Remote Audio Port:	9000
ℹ Local Audio Port:	✔ Automatic
ℹ Via:	Any

> **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**.

Program Manager			
	Address Book		✕
Ente	**Codec Name**	**Stream Name**	**Status**
ℹ Address:	Studio_90114	Sport 1	Busy
ℹ Remote Session	Studio_90114	Sport 2	Available
ℹ Remote Audio P	Studio_90114	Sport 3	Available
ℹ Local Audio Port:	✔ Automatic		
ℹ Via:	Any		

2. Select the interface options to use on local and remote codecs when dialing TieLink connections. Note: see Configuring TieLink Settings for more info.
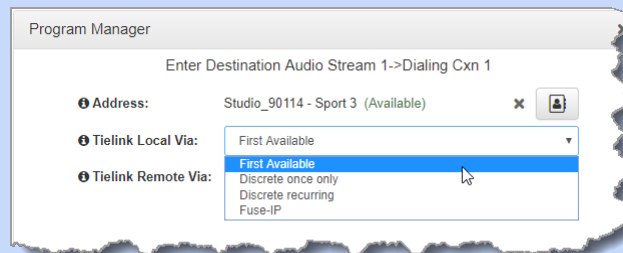
Click **Save** to save the program with the default algorithm, jitter and FEC settings which are physically entered in the codec. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings and configure a failover connection or SmartStream PLUS for this audio stream (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.
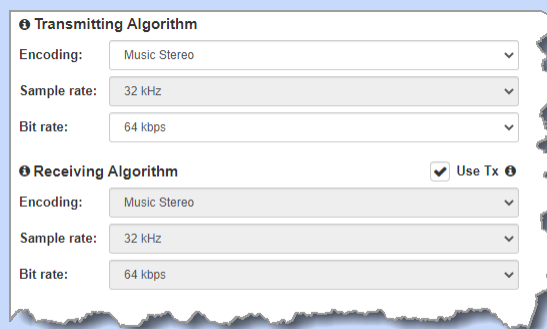
For IP connections click to configure:
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more 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.
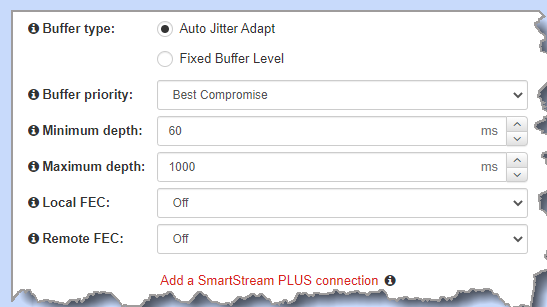
Important Notes:
- If you select **Sessionless** or **SIP** as the **Session Protocol** then RFC-compliant FEC is displayed.
- **FEC Delay** is only available when the **FEC** percentage is **100%**. This is designed to delay the sending of FEC packets for a predetermined period after the primary audio 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 The codec supports an optional Euro ISDN module for primary or backup connections. 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.

Enter Destination Stream 1->Dialing Cxn 1

ⓘ Number: 55555555

ⓘ Via: Any

☑ Enable bonded connections ⓘ

ⓘ Number: 55555556

Via: Any

Next, click **Save** to save the program with default encoding settings, or click **Next** to specify a different algorithm and configure a backup connection if required.

Select Encodings Stream 1->Dialing Cxn 1

ⓘ Algorithm

Encoding: Music Plus Stereo

Sample rate: 48 kHz

Bit rate: 128 kbps

Dialing settings for the ISDN 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.

ⓘ **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 the drop-down **Failure Mode** arrow and select **Failover** to create a failover connection. Adjust **Failover Parameters** and click **Next**.
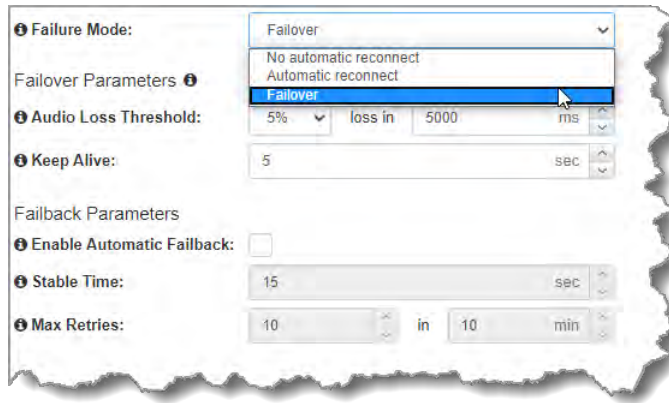
Important Note: When Failover is enabled, the codec will fail over to a backup connection under the following circumstances:
- There is sudden loss of connectivity to the primary destination.
- The remote codec disconnects the primary connection.
- The user manually attempts to connect to the failover 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 in the following table define configurable failover parameters.

	Failover Parameter	Description
1	Audio Loss Threshold	The percentage of lost data measured during a given time frame
2	Time Frame	The time frame in milliseconds against which lost data is measured
3	Keep Alive (Time Frame)	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 (Maximum Retries)	The time frame used to measure the number of fail back retries attempted

2. Enter a name for the failover connection and click **Next**.
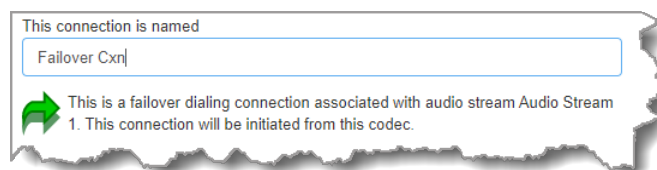
3. Click **Next** to continue through the wizard and configure the failover connection in a similar manner to how you configured the primary connection.

Configuring the Codec to Answer Connections

The codec is capable of being configured to accept calls via different transports (e.g. IP and ISDN), or to accept calls using different audio ports. If you are configuring the codec to allow it to answer one or more incoming audio stream connections:

1. Enter a name for the answering connection and click **Next**.

> Configure Connection for Audio Stream 1->Answering Cxn 1
>
> This connection is named
>
> Answering Cxn 1
>
> This is an answering connection associated with audio stream Audio Stream 1. Connections dialed from another codec that match this connection configuration may be answered by this codec.
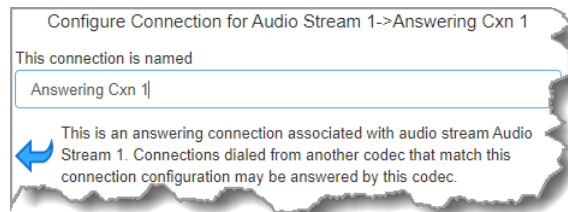
2. Configure the transport settings:

IP For IP select the **Session Protocol** and **Local Audio Port**.

❶ Transport:	IP	⌄
❶ Session Protocol:	Tieline Codecs	⌄
❶ Local Audio Port:	9000	◆ Any
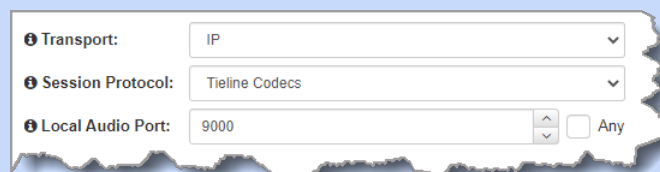
Click **Next** to specify jitter buffer settings, or create another answering connection. Click to configure:
- **Auto Jitter Adapt** and the preferred auto jitter setting using the drop-down arrow for **Buffer priority**. It is also possible to configure the **Minimum depth** and **Maximum depth** of jitter over the connection. See Configuring the Jitter Buffer for more details.
- Alternatively, select a **Fixed Buffer Level** and enter the **Jitter Depth**, which must be between 12ms and 5000ms depending on the algorithm you select.

❶ Buffer type:	◉ Auto Jitter Adapt
	◯ Fixed Buffer Level
❶ Buffer priority:	Best Compromise	⌄
❶ Minimum depth:	60	ms
❶ Maximum depth:	1000	ms

Click **Next** to configure **Failure Parameters** for the answering connection if required. **Please note**: In most situations the default answering **Failure Parameters** do not need adjustment. These settings may be useful to troubleshoot certain connections, e.g. satellite IP links.

ISDN For ISDN, settings are determined by ISDN module answering settings. For more details see Configuring ISDN 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** to save the program at this point.
    iii. Click **Next** to configure **Output Audio Source** options.

Configuring Output Audio Source Options

1. Configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program audio at transmitter sites. **Output Audio Source** options include:

- **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
- **Input**: Input audio looped to the physical codec outputs.
- **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
- **USB File Playback**: Audio file playback from a USB stick.

2. Click the blue **Plus symbol** ✚ to add a backup **Output Audio Source**, or click the **Minus symbol** ➖ to remove an **Output Audio Source**.

3. Click the drop-down arrow to select an **Output Audio Source** option.

4. Configure silence threshold parameters to enable a preferred backup option, as well as resume thresholds for reactivating a previous source. Then click **Save Program** to save program settings.

5. After configuring **Output Audio Source** options you can:

    i. Click **Save** to save the program at this point.
    ii. Click **Next** to configure rules options.

Important Notes for File Playback:
* Ensure MP3 recordings used are not variable bit rate files.
* File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
* USB backup audio is only sent to the outputs of the local codec to which a USB stick is attached. USB file audio is not sent to encoders and cannot be transmitted via an audio stream to another codec.
* The USB stick can be inserted or removed at any time as long as the codec is not already playing audio in failover mode. Removing the USB stick while audio is playing from it will result in poor audio quality and should be avoided. If it is removed accidentally you must reboot the codec to ensure USB failover will work reliably in future.
* If you enter a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
* If you enter a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
* File playback will occur automatically if the silence threshold parameters are breached; if the codec is not connected for any reason file playback will commence. To stop file playback open the **Connections panel** in the HTML5 Web-GUI, click to select the file playback connection, then click **Disconnect**.

Configuring Rules

1. 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.

Configure Rules for Audio Stream 1

ℹ **Rule to add:** Use the connected state of Audio Stream 1 to toggle a ⌄ +

Use two local inputs to connect/disconnect Audio Stream 1 —

Connect on: Local Control Port Input ⌄ | 1

Disconnect on: Local Control Port Input ⌄ | 2

Connect Audio Stream 1 when Local Control Port Input 1 is activated and disconnect it when Local Control Port Input 2 is activated.

> ℹ **Important Note:** Connection-related rules are not displayed in **Answer only** audio streams..

2. Click **Save** to save the program.
3. Click **Finish** to exit the wizard.
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 connect the program manually using the codec front panel.

23.11   Configure Phase Locked Surround Programs

Product	Feature	Notification
Gateway 4	Programs	Supports programs with up to 4 channels of audio.
	Program Manager panel	Displays program types that support 4 channels in/out
Gateway 8/16	Programs	Supports program connections using up to 16 channels of audio, depending on codec configuration purchased (between 8 and 16 channels available for purchase)
	Program Manager panel	The Gateway 8 displays program types that support 8 channels in/out; the Gateway 16 displays program types that support 16 channels in/out.

The codec is capable of streaming multichannel phase-locked surround sound audio streams according to SMPTE ST 2110-30 standards. Support is available for streaming 4, 6 and 8 digital channels in phase-locked mode.

It is also possible to stream 6 channels of phase locked audio, plus a phase locked stereo IP stream. This allows networks to deliver live content in both stereo and surround sound formats for a variety of live broadcast applications and audiences, or STL and studio-to-studio applications. The Gateway 16 also supports sending 6 + 6 + 2 + 2 channels of phase-locked streams, or 8 + 8 channels of phase-locked streams, or 4 + 4 + 4 + 4 channels of phase-locked streams.

Configuring Phase Locked Streaming Programs

Configuration is similar to setting up a regular connection program. In the **Program Manager panel** select the program with the preferred connection profile suiting a phase-locked 6 channel (5.1 or 6.0) or 8 channel (7.1 or 8.0) surround sound stream, or a 4 channel audio stream.
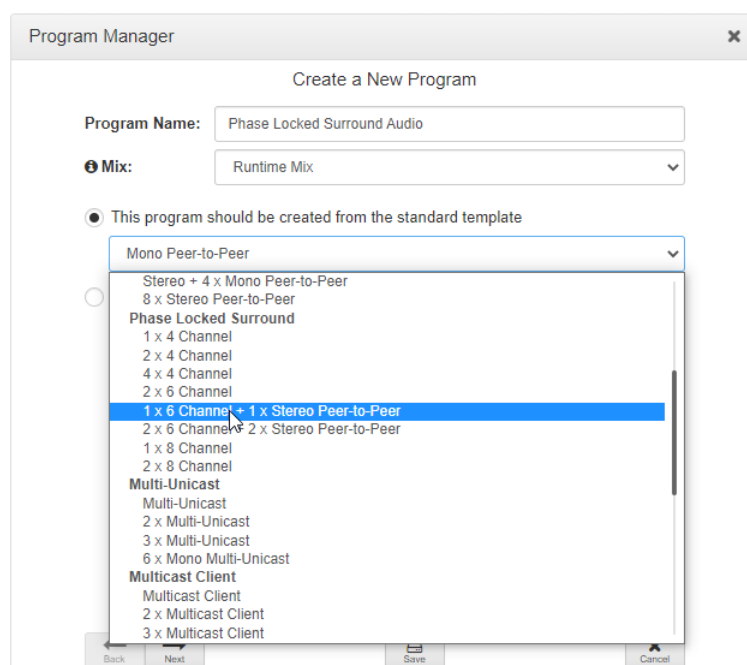
Gateway-16 phase locked program options

Select a preferred encoding algorithm when configuring a program using the **Program Manager panel**.
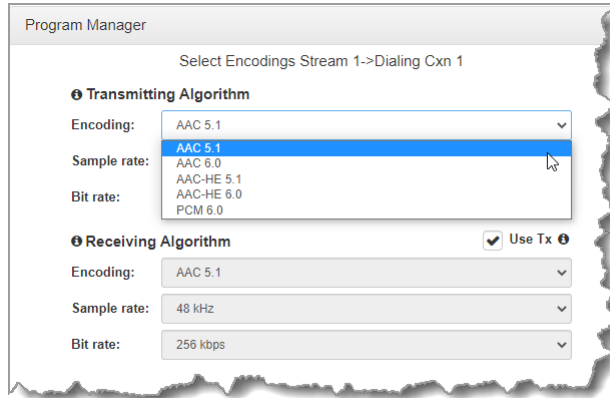
The following table displays the mapping of channels to encoders and decoders for 5.1 channel and 7.1 channel surround sound streams. This is based on the audio channel group order defined in the SMPTE ST 2110-30 standard. Please note: 6.0 and 8.0 channel streams feature undefined channel grouping. All channels are the same quality and are not bandwidth limited. The selection of 7.1 or 8.0, and 5.1 or 6.0, for streaming is determined when selecting the encoding algorithm when creating a program.

⚖ **Codec Model Note:** Only 4.0 channel phase-locked audio is supported in the Gateway 4.

5.1 Channel Mapping for 5.1 and 7.1 Phase-Locked Surround Streams

Encoder/Decoder	1 x 5.1 Channel Program	1 x 7.1 Channel Program
Encoder 1 / Decoder 1	Left	Left
Encoder 2 / Decoder 2	Right	Right
Encoder 3 / Decoder 3	Center	Center
Encoder 4 / Decoder 4	LFE (bandwidth limited channel)	LFE (bandwidth limited channel)
Encoder 5 / Decoder 5	Surround Left	Side Surround Left
Encoder 6 / Decoder 6	Surround Right	Side Surround Right
Encoder 7 / Decoder 7		Rear Surround Left
Encoder 8 / Decoder 8		Rear Surround Right

Encoder/Decoder	2 x 5.1 Channel Program	2 x 7.1 Channel Program
Encoder 1 / Decoder 1	Left	Left
Encoder 2 / Decoder 2	Right	Right
Encoder 3 / Decoder 3	Center	Center
Encoder 4 / Decoder 4	LFE (bandwidth limited channel)	LFE (bandwidth limited channel)
Encoder 5 / Decoder 5	Surround Left	Side Surround Left
Encoder 6 / Decoder 6	Surround Right	Side Surround Right
Encoder 7 / Decoder 7		Rear Surround Left
Encoder 8 / Decoder 8		Rear Surround Right
Encoder 9 / Decoder 9	Left	Left
Encoder 10 / Decoder 10	Right	Right
Encoder 11 / Decoder 11	Center	Center
Encoder 12 / Decoder 12	LFE (bandwidth limited channel)	LFE (bandwidth limited channel)
Encoder 13 / Decoder13	Surround Left	Side Surround Left

Encoder 14 / Decoder 14	Surround Right	Side Surround Right
Encoder 15 / Decoder 15		Rear Surround Left
Encoder 16 / Decoder 16		Rear Surround Right

23.12 Default Program Ports

Product	Feature	Notification
Gateway 4	Program Manager panel	• Supports port configuration for up to 4 peer-to-peer audio streams. • Supports port configuration for one multi-unicast and multicast audio stream
Gateway 8/16	Program Manager panel	• Supports port configuration for up to 8 peer-to-peer audio streams in Gateway 8 and up to 16 peer-to-peer audio streams in Gateway 16 • Supports port configuration for up to 6 different multi-unicast audio streams, or up to 8 different multicast streams in Gateway 8, and 16 different multicast streams in Gateway 16

The following table displays the default TCP and UDP ports used when configuring audio stream connections over IP using different program types. It is a good idea to keep a record when changing these default port numbers, to ensure a record of these settings is maintained in case it is necessary to reconfigure the codec in the future. To view other important ports that need opening for SIP, Fuse-IP, Web-GUI use and G3 Toolbox software, please see Installing the Codec at the Studio in this user manual.

Audio Stream	TCP Session Port	UDP Audio Port	SmartStream + 1 (UDP)	SmartStream + 2 (UDP)	SmartStream + 3 (UDP)
1	9002	9000	9001	9002	9003
2	9002	9010	9011	9012	9013
3	9002	9020	9021	9022	9023
4	9002	9030	9031	9032	9033
5	9002	9040	9041	9042	9043
6	9002	9050	9051	9052	9053
7	9002	9060	9061	9062	9063
8	9002	9070	9071	9072	9073
9	9002	9080	9081	9082	9083
10	9002	9090	9091	9092	9093
11	9002	9100	9101	9102	9103
12	9002	9110	9111	9112	9113
13	9002	9120	9121	9122	9123
14	9002	9130	9131	9132	9133
15	9002	9140	9141	9142	9143
16	9002	9150	9151	9152	9153

Default Multi-Unicast Audio Stream Ports

Audio Stream	TCP Session Port	UDP Audio Port
1	9002	9000
2	9002	9010
3	9002	9020

4	9002	9030
5	9002	9040
6	9002	9050

Default Multicast Audio Stream Ports

Multicast Audio Stream	TCP Session Port	UDP Audio Port
1	9002	9000
2	9002	9010
3	9002	9020
4	9002	9030
5	9002	9040
6	9002	9050
7	9002	9060
8	9002	9070
9	9002	9080
10	9002	9090
11	9002	9100
12	9002	9110
13	9002	9120
14	9002	9130
15	9002	9140
16	9002	9150

23.13   Configure Multi-Unicast Dialing Programs

Product	Feature	Notification
Gateway 4	Multi-unicasting	Supports 1 or 2 multi-unicast streams being sent to up to 20 endpoints
Gateway 8/16	Multi-unicasting	Supports up to 8 multi-unicast audio streams in Gateway 8 and up to 16 multi-unicast streams in Gateway 16 being sent to up to 80 endpoints in total. For example, 4 separate audio streams can each be streamed to 20 endpoints.

Important Notes: Before you start program configuration please note:

- You cannot edit a program when it is currently loaded in the codec.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- If a codec supports multiple stream programs and the **Auto** (default) reference level is selected, the first codec to connect will configure the reference level used for all subsequent connections. I.e. If a G3 codec connects first then the G3 Audio Reference Level will be configured for all connections.
- All streams must be dialed initially for a multi-unicast program. Streams can be disconnected and dialed individually after this.
- Connections are "dial only" for multi-unicast programs.
- Up to 80 streams of 24 bit, 48kHz PCM audio are supported. 24 bit, 96kHz PCM is not supported for multi-unicast audio streams.

- SmartStream PLUS is supported for multi-unicast connections. Each additional redundant stream is included and added in the 80 stream limit for the Gateway and 20 stream limit for the Gateway 4.
- Some drop-down menus and settings may be greyed out intentionally depending on features available.
- It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time. The first connection in each multi-unicast stream determines default settings (e.g. algorithm, sample rate, bit rate), for all subsequent connections in that stream, except for dialing settings.
- All algorithms are supported for multi-unicast connections, however only one can be used for each audio stream.
- Bidirectional audio is only available on the first connection dialed for each audio stream.
- Renegotiation of connection bit rates is not possible when connected.
- Ensure you have sufficient connection bandwidth at the local codec to support all the connections to which you are connecting.
- FEC is not available for multi-unicast connections.
- If a codec is answering more than one mono or stereo multi-unicast connection it is necessary to create an answering program to suit the answering configuration and lock this program in the codec.

Creating a Multi-Unicast Program

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** symbol to open the wizard and then:
   - Click in the text box to name the new program.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Multi-unicast** or **2x Multi-unicast**, **3 x Multi-unicast or 8 x Multi-unicast**, (or **16 x Multi-unicast** in Gateway 16) or if you want to use an existing program as a template, select this option. Then click **Next**. Note: In this example we create a **2 x Multi-unicast** program.

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 Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. (WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network. WheatNet logical inputs are only available if a codec has a WheatNet-IP card installed). 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 the **Stream Name** and add a **caller ID**. Then click **Next**. Note: The caller ID is used to identify calls.

Routing Type Options:	
Default	By default IP streams are routed using audio ports.
Deterministic **IP**	Use **Answer Routes** to configure deterministic routing of multiple audio streams using SIP. Answer routes can be configured for up to 6 SIP accounts or SIP interfaces. Then create a multiple stream answering program and configure an **Answer Route** for each audio stream matching the SIP Account or SIP Interface **Answer Routes**. A call received via a particular account or interface is routed to the audio stream with a matching audio answer route. See Answering Multiple SIP Peer-to-Peer calls for more information.
Line Hunt	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

| IP | 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**.

Configure Connection for Stream 1->Endpoint 1
This connection is named
Endpoint 1
This is a dialing connection associated with audio stream Stream 1. This connection will be initiated from this codec.

6. Configure the transport settings for the connection, then click **Next**. Note: only the first connection dialed in a multi-unicast audio stream can encode and decode audio. All other connections are unidirectional and encode only.
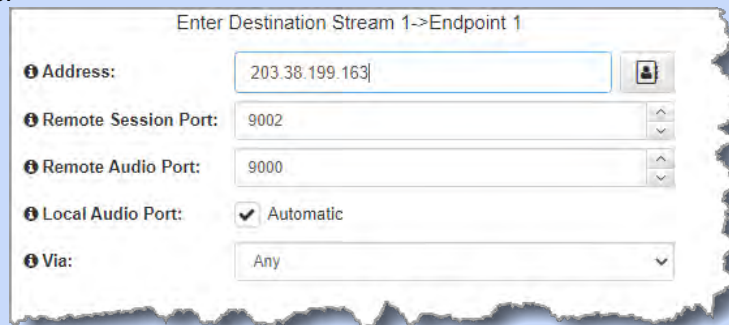
Configure Transport settings Stream 1->Endpoint 1

❶ Transport: IP

❶ Session Protocol: Tieline Codecs

❶ Encode/Decode: Both

❶ Connect Timeout: 120 sec

❶ Enable Auxiliary Data: ☐

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. Select **Enable Auxiliary Data** to enable synchronized out-of-band data in separate packets using any algorithm. Only in-band auxiliary data using the Music or MusicPLUS algorithms is possible when connecting to G3 Commander and i-Mix codecs.

7. Configure destination codec dialing and encoding settings:

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.

Enter Destination Stream 1->Endpoint 1

❶ Address: 203.38.199.163

❶ Remote Session Port: 9002

❶ Remote Audio Port: 9000

❶ Local Audio Port: ✔ Automatic

❶ Via: Any

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.

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 (recommended) and more connection endpoints.

Click the drop-down arrows on the right-hand side of each text box to adjust the **Encoding**, **Sample rate** or **Bit rate** options.

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.

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.

8. Click the drop-down **Failure Mode** arrow and select **Automatic reconnect** to enable this feature and configure **Failure Parameters**. To add another stream endpoint select the check-box for **Create another dialing connection**, then click **Next**.

Configure Failure Parameters Stream 1->Endpoint 1

ℹ Failure Mode: Automatic reconnect

Failure Parameters ℹ

ℹ Audio Loss Threshold: 50%   loss in   30000   ms

ℹ Keep Alive: 5   sec

✔ Create another dialing connection

9. Name the connection and then click **Next**.

Configure Connection for Stream 1->Endpoint 2

This connection is named

Endpoint 2

This is a dialing connection associated with audio stream Stream 1. This connection will be initiated from this codec.

Delete this connection

10. Enter destination settings for the dialing connection, then click **Next**.

Enter Destination Stream 1->Endpoint 2

ℹ Address: 203.38.199.161

ℹ Remote Session Port: 9002

ℹ Remote Audio Port: 9000

ℹ Local Audio Port: ✔ Automatic

ℹ Via: Any

11. There are four options at this point:
- Click **Add a SmartStream PLUS connection** to add a redundant stream for the current connection.
- Select **Create another dialing connection** to create another stream endpoint.

- Click **Next** to configure the **Output Audio Source** for the audio stream.
- Click **Next Stream** to configure the second multi-unicast audio stream.
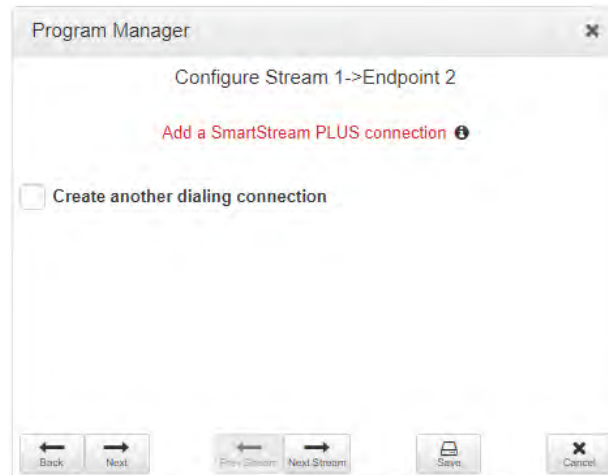
Configuring Output Audio Source Options

1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program
   audio at transmitter sites. **Output Audio Source** options include:

   - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **Input**: Input audio looped to the physical codec outputs.
   - **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
   - **File Playback**: Audio file playback from a supported storage medium.

2. Click the blue **Plus symbol** ✚ to add a backup **Output Audio Source**, or click the **Minus symbol** ➖ to remove an **Output Audio Source**.
3. Click the drop-down arrow to select an **Output Audio Source** option.

4. Configure silence threshold parameters for enabling a preferred backup option, as well as resume thresholds for reactivating a previous source.

> ℹ️ **Important Notes for File Playback:**
> - Ensure MP3 recordings used are not variable bit rate files.
> - File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
> - Backup file audio is only sent to the outputs of the local codec. File playback audio is not sent to encoders and cannot be transmitted via an audio stream to another codec.
> - Removing storage media while audio is playing from it will result in poor audio quality and should be avoided. If it is removed accidentally reboot the codec to ensure failover file playback will work in future.
> - If you enter a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
> - If you enter a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
> - File playback occurs automatically if the silence threshold parameters are breached; if the codec is not connected for any reason file playback will commence. To stop file playback open the **Connections panel** in the HTML5 Web-GUI, click to select the file playback connection, then click **Disconnect**.

5. After configuring **Output Audio Source** options you can:

    i.   Click **Next** to configure rules options.
    ii.   Click **Next stream** to configure the next audio stream.

6. 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.

Configure Rules for Stream 1

❶ Rule to add: --- Select Rule --- ➕

Use a local input to toggle a remote output ➖

Input: Local Control Port Input 1

Output: Remote Output 1

Remote Output 1 follows the ON/OFF state of Local Control Port Input 1. ❶

7. Click **Next stream** to configure the next audio stream.

Configure the Second Multi-unicast Audio Stream

1. Click **Next Stream** to configure the second multi-unicast audio stream.

Configure Audio Stream 2 for 2 x Multi-unicast

Stream Name: Stream 2

❶ Caller ID: Stream 2

Dial/Answer: ● Dial only

❶ G3 Profile: Automatic

❶ G3 Channel: Auto

2. Enter the name of the first connection for this audio stream in the text box, then click **Next**.

Configure Connection for Stream 2->Endpoint 1

This connection is named

Endpoint 1

➡ This is a dialing connection associated with audio stream Stream 2. This connection will be initiated from this codec.

3. Continue through the program wizard and configure all multi-unicast connections for this audio stream. After all connections are configured there are three options:
   - Click **Next** to configure the **Output Audio Source** for the audio stream.
   - Click **Next** to configure rules for this audio stream.
   - Click **Save** to save the currently configured settings.

4. Click **Save** to save the program settings, then click **Finish** or **Load** to load the new program.

> ℹ **Important Notes:** Configure new audio streams and connections as required for **3 x Multi-unicast** or **6 x Multi-unicast** programs. Following are the default ports used when configuring up to 6 multi-unicast audio streams (up to 80 connections supported in total):
> - Audio Stream 1: **Audio Send Port 9000**
> - Audio Stream 2: **Audio Send Port 9010**
> - Audio Stream 3: **Audio Send Port 9020**
> - Audio Stream 4: **Audio Send Port 9030**
> - Audio Stream 5: **Audio Send Port 9040**
> - Audio Stream 6: **Audio Send Port 9050**

5. 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 connect the program manually using the codec front panel.

## 23.14   Configure Multicast Server Programs

Product	Feature	Notification
Gateway 4	Multicasting	Supports 1 or 2 multicast server and/or client streams being sent to unlimited endpoints
Gateway 8/16	Multicasting	Supports up to 8 different multicast server and/or client streams in Gateway 8, and up to 16 different multicast server and/or client streams in Gateway 16

In multicast server mode a 16 channel Gateway codec supports sending 1, 2, 3, 4, 6, 8 or 16 multicast audio streams to unlimited endpoints over compatible IP networks. It is also possible to use multicast client mode to simultaneously receive up to 16 separate multicast streams.
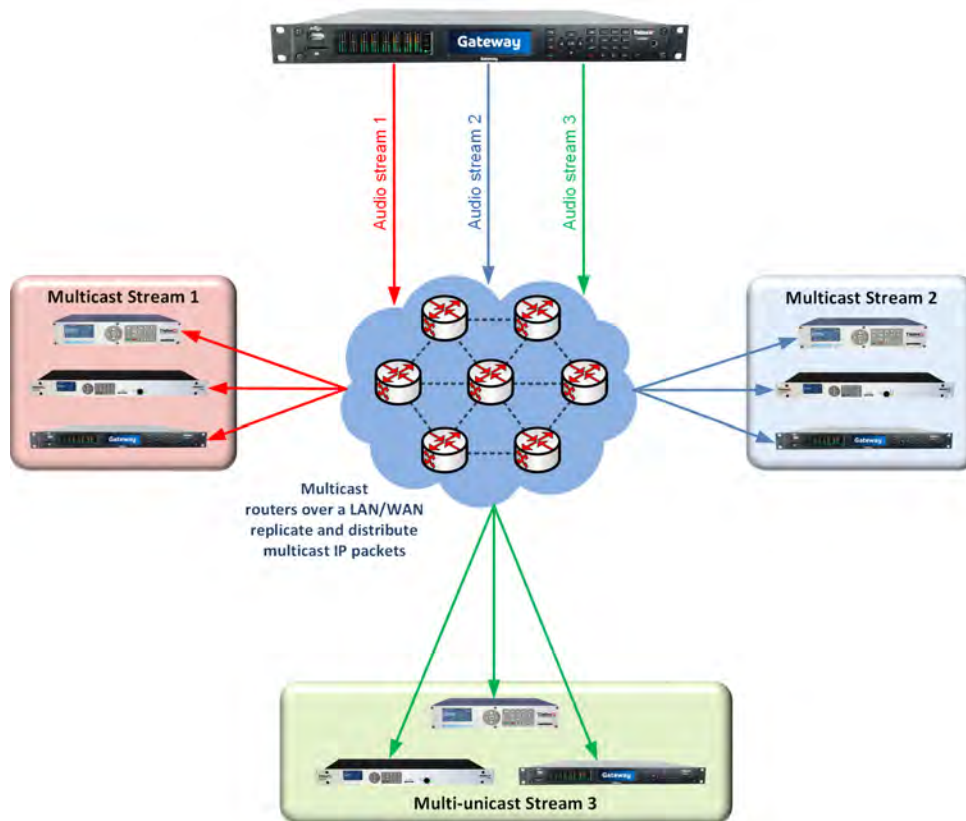
Gateway codec in 3 x multicast server mode distributing 3 multicast IP streams

How Multicasting Works

Multicast transmissions are sent using a dedicated IP multicast address that looks similar to a regular IP address and multicast subscribers request transmissions from this address. This unique address allows multicast routers to identify multicast requests from a group of codecs interested in a particular transmission and packets are replicated depending on demand. This can create large demands on network bandwidth if the multicast group is significant in size.

As a result, only small sections of the internet are multicast enabled and many internet service providers (ISPs) block multicast traffic over wide area networks. This restricts most multicast broadcasts to private local area networks. Some ISPs provide quality of service (QoS) priority to multicast streams for an increased service charge. You need to check with your ISP to find out what multicast services, if any, are available over WANs.

> **Important Notes:**
> - When a connection is dialed Tieline codecs normally use session data to configure settings like the algorithm, connection bit rate and sample rate etc. Multicast connections do not use session data and it is imperative that all codecs are configured with the same connection settings prior to connecting, or they will not be able to join multicast streaming sessions.
> - Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

Multicast Server versus Multicast Client Programs

Two different types of multicast programs need to be created when multicasting:

- A multicast server program is used by the broadcasting codec to send multicast IP packets to multicast routers on a network.
- A multicast client program is used by codecs to receive multicast IP audio packets.

A multicast server codec sends audio packets only and a multicast client codec receives audio packets only. Codecs using the client program request multicast packets (sent from the server codec), which are distributed by multicast routers.

Creating Multicast Server Programs

> **Important Notes:** Before you start program configuration please note:
> - Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
> - You cannot edit a program when it is currently loaded in the codec.
> - Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
> - Some drop-down menus and settings may be greyed out intentionally depending on features available.
> - It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
> - Always dial the multicast server codec connection first before connecting multicast client codecs.
> - Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
> - The default UDP audio port setting is 9000 for the first multicast, 9010 for the second multicast and 9020 for the third multicast, and so on. E.g. if a client codec wishes to receive multicast audio stream 2 then it must use audio port 9010. Default Tieline Audio Stream ports are outlined in Default Program Ports.
> - Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to transmit auxiliary data.
> - Relay functionality is available for multicast connections between Tieline codecs.
>   - It is not possible to connect to a G3 codec and receive multicast IP audio streams.

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.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select a preferred Multicast Server program for the options available.

- If you want to use an existing program as a template, select this option. 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.

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 Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. (WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network. WheatNet logical inputs are only available if a codec has a WheatNet-IP card installed). 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 the **Stream Name**, then click **Next**.

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**. Note: select **UDP/IP +RTP** for RFC compliant streaming.

> **Important Note:**
> - The encode and decode direction is configured automatically for **Encode Only** (server program) or **Decode Only** (client program). This setting is configured when either **Multicast Server** or **Multicast Client** is first created in the wizard.

- The **Time to Live** (TTL) value you need to use is dependent upon your network infrastructure. Please consult your network administrator if you are unsure about how to configure this setting.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.

7. Configure the multicast IP address and **Remote Audio Port** (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**), then click **Next**. Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

Enter Destination Stream 1->Multicast Server Cxn 1

ⓘ Address:	224.0.255.225
ⓘ Audio Port:	9000
ⓘ Via:	Any

8. Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. Click **Next** to continue.

Select Encodings Stream 1->Multicast Server Cxn 1

ⓘ Transmitting Algorithm

Encoding:	Music Stereo
Sample rate:	32 kHz
Bit rate:	64 kbps

9. Click the drop-down arrow for **Send FEC Type** to enable RFC-compliant FEC. Configuration instructions are displayed in the right-hand **Program Manager panel** pane. **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.

Configure Stream 1->Multicast Server Cxn 1

ⓘ FEC Type:	RFC2733
ⓘ FEC:	100%
ⓘ FEC Delay:	0 ms
ⓘ FEC Address:	224.0.255.225
ⓘ FEC Port:	9002
ⓘ FEC Via:	Any

10. Click **Next** to select configure **Enable Auto Reconnect** if required.
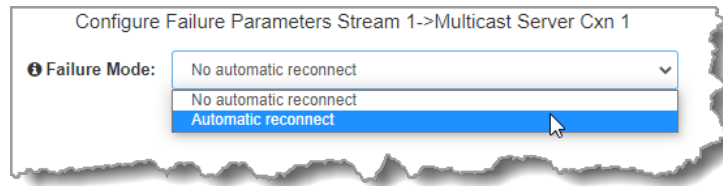
11. Next you can either:

    i. Click **Next** to configure rules options.
    ii. Click **Next Stream** if configuring multiple audio stream programs.

12. 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.
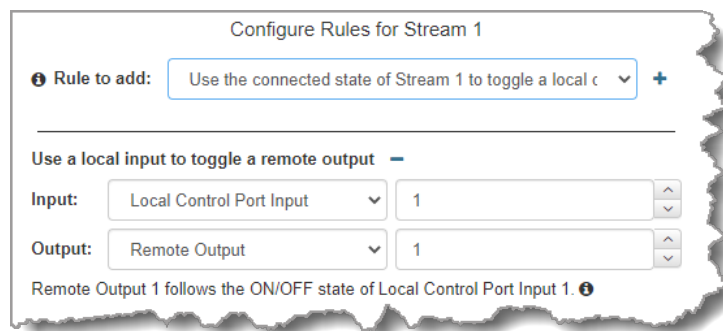

13. After configuring all streams in the multicast server program click **Save** to store the program settings, then click **Finish**.

> **Important Notes:** There is no jitter buffer setting on the server codec because it never receives audio packets.

14. Configure multicast server and multicast client programs and load all codecs with the appropriate program. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast audio packets. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel.

23.15   Configure Multicast Client Programs

Product	Feature	Notification
Gateway 4	Multicasting	Supports 1 or 2 multicast client streams being received
Gateway 8/16	Multicasting	Supports up to 8 different multicast client streams in Gateway 8, and up to 16 different multicast client streams in Gateway 16

> **Important Notes:** Before you commence program configuration please note:
> - Ensure all connection related settings like the port, algorithm, bit rate (etc) match on both multicast server and client programs or they will not connect successfully.
> - You cannot edit a program when it is currently loaded in the codec.
> - Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
> - Some drop-down menus and settings may be greyed out intentionally depending on features available.
> - It is possible to save a program at several points throughout the program wizard and use default settings to save configuration time.
> - Always dial the multicast server codec connection first before connecting multicast client codecs.
> - Multicast client codecs will display return link quality (LQ) only. The **Return** reading represents the audio being downloaded from the network locally. Multicast server codecs do not display LQ readings.
> - The default UDP audio port setting is 9000 for the first multicast, 9010 for the second multicast and 9020 for the third multicast, and so on. E.g. if a client codec wishes to receive multicast audio stream 2 then it must use audio port 9010. Default Tieline Audio Stream ports are outlined in Default Program Ports.
> - Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data.
>   - It is not possible to connect to a G3 codec and receive multicast IP audio streams.
>   - To copy multicast client programs onto multiple codecs see Backup and Restore Functions.

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.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Multicast Client** to configure a multicast program, or if you want to use an existing program as a template, select this option. Then click **Next**.
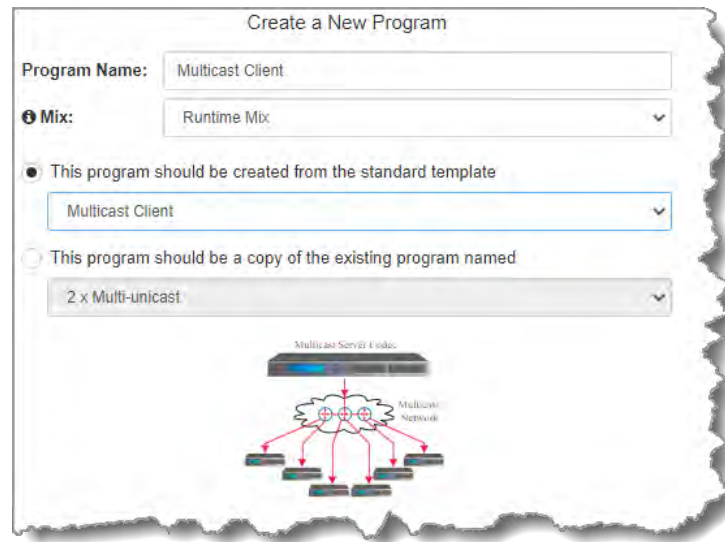
Important Notes:

- 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.
- Gateway 16 channel codecs support simultaneously receiving up to 16 separate multicast streams in multicast client mode.

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 Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. (WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network. WheatNet logical inputs are only available if a codec has a WheatNet-IP card installed). See Enabling Relays & RS232 Data for more info.
- Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** 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.
- Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
- For more details about rules see Creating Rules.

4.   Enter the **Stream Name**, then click **Next**.

Configure Audio Stream 1 for Multicast Client

Stream Name: Stream 1

Dial/Answer: ⦿ Dial only

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**.

Configure Connection for Stream 1->Multicast Client Cxn 1

This connection is named

Multicast Client Cxn 1

➡ This is a dialing connection associated with audio stream Stream 1. This connection will be initiated from this codec.

6.   Configure the transport settings for the connection, then click **Next**. Note: select **UDP/IP +RTP** for RFC compliant streaming.

Configure Transport settings Stream 1->Multicast Client Cxn 1

🛈 **Transport:**   IP

🛈 **Session Protocol:**   Sessionless

🛈 **Audio Protocol:**   UDP/IP + RTP

🛈 **Encode/Decode:**   Decode Only

🛈 **Enable Auxiliary Data:**   ☐

🛈   **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 the multicast IP address and **Local Audio Port** (the same multicast address and port must be used for both the server and client programs), then specify which IP streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**), then click **Next**. Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

Enter Destination Stream 1->Multicast Client Cxn 1

🛈 **Address:**   224.0.255.255

🛈 **Audio Port:**   9000

🛈 **Via:**   Any

8.   Click the drop-down arrows on the right-hand side of each text box to select the **Encoding**, **Sample rate**, **Bit rate** or **Sample size** options. 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.

Next, select **Return FEC Enabled** to enable RFC2733 compliant Forward Error Correction, which is available because multicast connections are sessionless. Follow the instructions in the right-hand pane. When configuration is complete click **Next**.

Important Notes: Automatic or fixed jitter buffer settings can be adjusted on individual client codecs as required. There is no jitter buffer setting on the server codec because it never receives audio packets.

10. Click the **Failure Mode** drop-down arrow to select **Automatic reconnect** if required, then click **Next**.

11. After configuring these settings there are 2 options:

     i.   Click **Next** to configure **Rules** and/or **Output Audio Source** options.
     ii.   Click **Save** to save the program.

Configuring Output Audio Source Options

1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program audio. **Output Audio Source** options include:

   - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **File Playback**: Audio file playback from a supported storage medium.
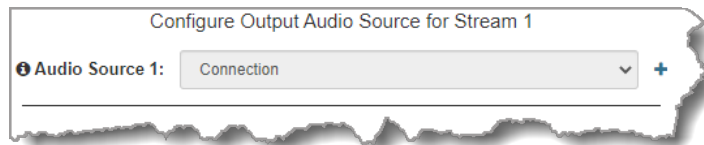
2. Click the blue **Plus symbol** + to add a backup **Output Audio Source**, or click the **Minus symbol** − to remove an **Output Audio Source**.
3. Click the drop-down arrow to select an **Output Audio Source** option.
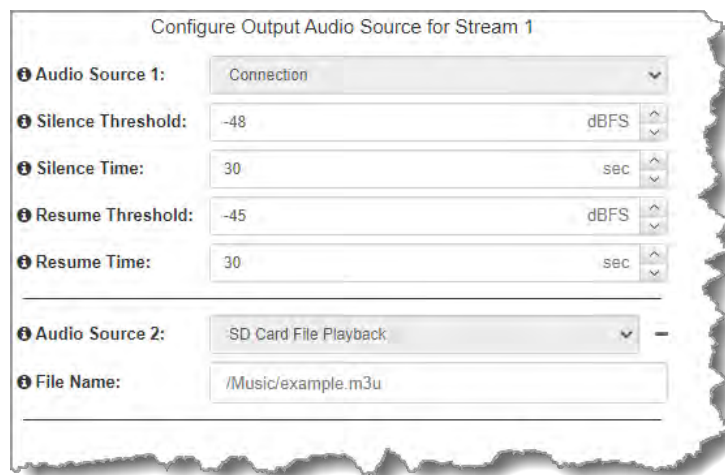
4. Configure silence threshold parameters for enabling a configured backup option, as well as resume thresholds for reactivating a previous source.
5. After configuring **Output Audio Source** options you can:

   i. Click **Save** to save the program at this point.
   ii. Click **Next** to configure rules options.

Important Notes for File Playback:
- Ensure MP3 recordings used are not variable bit rate files.
- File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
- Backup file audio is only sent to the outputs of the local codec. File playback audio is not sent to encoders and cannot be transmitted via an audio stream to another codec.
- Removing storage media while audio is playing from it will result in poor audio quality and should be avoided. If it is removed accidentally reboot the codec to ensure failover file playback will work in future.
- If you enter a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you enter a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to

play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.

- File playback will occur automatically if the silence threshold parameters are breached; if the codec is not connected for any reason file playback will commence. To stop file playback open the **Connections panel** in the HTML5 Web-GUI, click to select the file playback connection, then click **Disconnect**.

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.

2. Click **Save** to complete configuration of the program. Alternatively, continue through the wizard to configure the codec to receive multiple audio streams as required, then click **Save**.
3. Click **Finish** to exit the wizard or **Load** to load it.
4. Configure multicast server and multicast client programs and load all codecs with the appropriate program. Select and connect audio streams in a program using the **Connections panel**, or dial the program manually using the codec front panel. Dial the multicast server program connection first and then connect multicast client codec programs to begin receiving multicast streams.

23.16   Multicast Server and Client Programs

Product	Feature	Notification
Gateway 4	Multicasting	Supports 1 or 2 multicast server and client streams being received and also sent to unlimited endpoints
Gateway 8/16	Multicasting	Supports up to 8 different multicast server and client streams in Gateway 8, and up to 16 different multicast server and client streams in Gateway 16

Gateway codecs support programs configured with both multicast client and server capability simultaneously. Simply select the program to support both server and client capability and configure the program as per the previous instructions in this user manual. Note: the number of streams supported will vary depending on the model of codec.
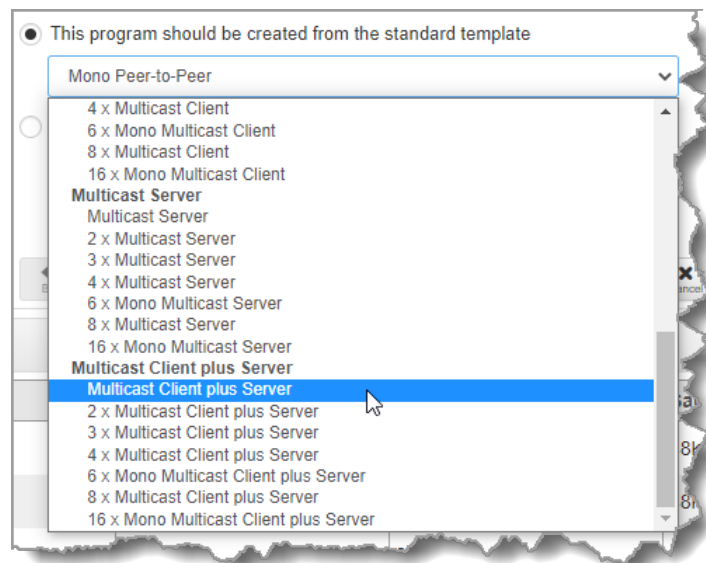
23.17 Configuring SIP

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 SIP program using the front panel via the **Home 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 is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless 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 > Settings > Audio > 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]**

23.17.1   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 **LAN2** by default, which is mapped to the **Secondary** 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** text box 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.
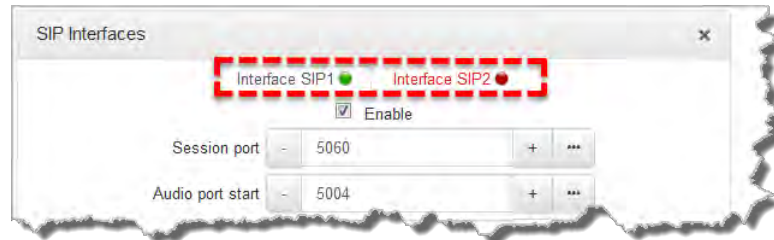
23.17.2 Configuring SIP Accounts

Up to 16 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. 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 16 SIP accounts can be added to the codec.
> - To configure a SIP Account using the codec **LCD SCREEN** see Configuring SIP Accounts.

Adding a SIP Account

Enter SIP account details and register the account in your codec. Once configured, the codec will contact the SIP server automatically to acknowledge its presence over a wide area network when connected to a public IP address.

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**.
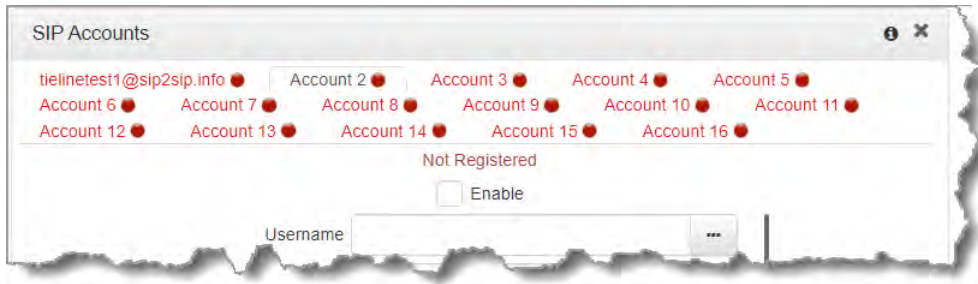
3. Enter the SIP account details into the relevant text boxes, including the registration **Timeout** (which shouldn't need to be adjusted from the default setting). Also ensure a SIP **Interface** is selected (e.g. **SIP1** or **SIP2**.) The SIP interface contains settings related to ports and the selected **Via** interface, e.g. **LAN1** or **LAN2**. See Configuring SIP Interfaces for more details.
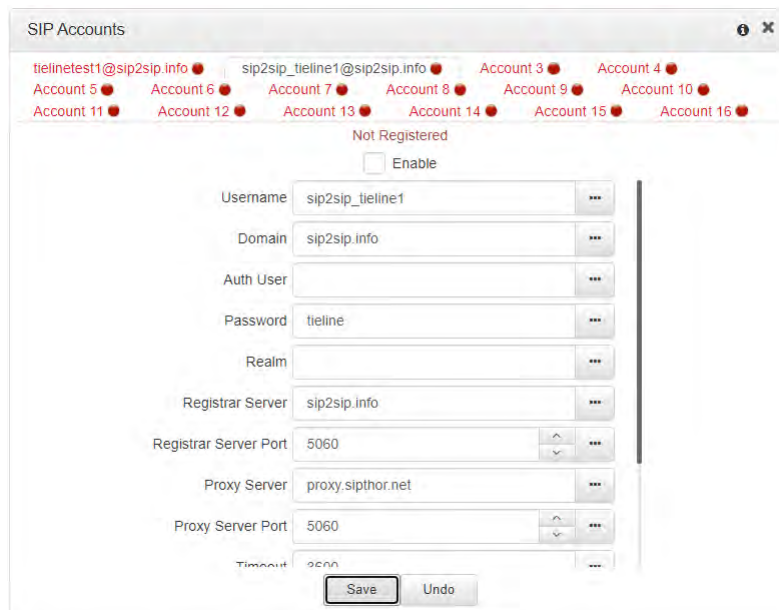
4. Click the **Enable** check-box at the top of the panel and then click the **Save** button to register the codec to the server.
5. If an account is registered successfully, the account registration indicator changes from red to green, and **Not Registered** (above the **Enable** check-box) becomes **Registered**.
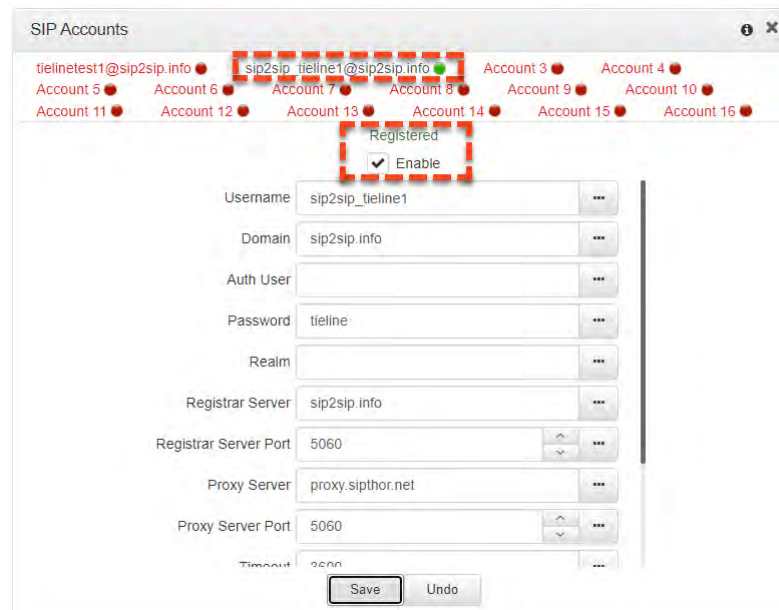
6. In the Toolbox Web-GUI the red **SIP** indicator adjacent to the codec **Online** indicator also changes to green when an account is currently registered in the codec and ready to be used when dialing over SIP.

7. Once enabled, the SIP account can be selected when creating a new SIP connection.

Important Notes:
- Some ISPs may block SIP traffic over UDP port 5060.
- By default, the session port used for each SIP account is the associated SIP interface session port. The default session port is the registered UDP port number 5060. It is also possible to configure a custom local session port for each SIP account for compatibility with Cisco Unified Communications Manager (CUCM). Ensure your firewall has the required TCP and UDP ports open when receiving multiple SIP calls.
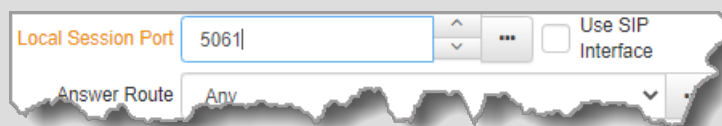
Troubleshooting SIP Registration

If a SIP account is not registering successfully 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**.

23.17.3 Configure SIP Allow and Block Lists

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 Allow List** 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 Block List** and add user agents to the **User Agent Block List** to deny them access to the codec. These block lists also filter unwanted traffic and increase the likelihood of rejecting unwanted traffic. Note: If an incoming SIP caller is not on the **URI Allow List** it will be scanned using the **URI Block List**. If there is no match it will be scanned using the **User Agent Block List**. A connection will be established if there is no match on either Block List.

> **Important Note:** To only allow a predefined list of codecs to connect, add them to the **URI Allow List** and add a wildcard (asterisk) * to the **URI Block List**: all incoming calls will be blocked except for codecs in the Allow List.

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 Allow List**, **URI Block List** or **User Agent Block List** 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 allow list and block list items.

> **Important Note:** Some codec manufacturers allow calls based on 'User Agent' identification. It may be necessary to enter a Tieline codec user agent into a non-Tieline codec 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
> - User agent for Report-IT SIP connections: Tieline Report-IT EE (3.5.6_2894). Note: In this example "3.5.6" is the Report-IT version number and "2894" is the build number.
> - 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:

    o Commander G3 Rack TLR300 = Model Number TLR300
    o Commander G3 Rack TLR300B = Model Number TLR350
    o Commander G3 Field TLF300 = Model Number TLF300
    o i-Mix G3 TLM600 = Model Number TLM600

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.
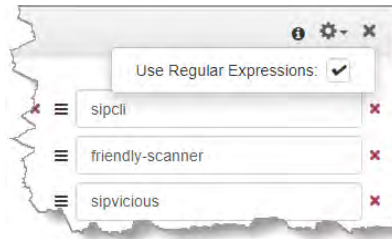
Important Note: Regular expressions should not use ^ and $ anchors because searches implicitly try to match anywhere in the line.

23.18   Configure Peer-to-Peer SIP Programs

SIP programs are very similar to how a normal Tieline IP program is configured. 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**. It is also necessary to select SIP as the **Session Protocol**.

To configure a SIP multiple stream program simply create a new program and configure each SIP audio stream like a single SIP Peer-to-Peer program. The codec is capable of registering up to 16 SIP accounts, each of which has an associated **Answer Route** field, which can be matched to a loaded answering program's audio stream Answer Route. Without using SIP accounts, each SIP interface also has an **Answer Route** field. However, only 2 SIP interfaces are supported, limiting this method of routing configuration to a maximum of 2 audio streams. Note: An account's Answer Route setting is applied first.

Important Notes: Before commencing 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 is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- 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.
   - Click the **Mix** drop-down arrow to associate a custom matrix mix with the program if required.
   - Select **Mono/Stereo Peer-to-Peer,** or if you want to use an existing program as a template, select this option. 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 Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. (WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network. WheatNet logical inputs are only available if a codec has a WheatNet-IP card installed). See Enabling Relays & RS232 Data for more info.
* Virtual inputs 5-7 can be activated by pressing the **F1** button and **KEYPAD** 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.
* Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.
* For more details about rules see Creating Rules.

4. Enter the **Stream Name** 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,

TieLink, 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.

Stream Name:	Stream 1
ⓘ Caller ID:	Stream 1
Dial/Answer:	● Dial or answer
	○ Dial only
	○ Answer only

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**.

Configure Connection for Stream 1->Dialing Cxn 1

This connection is named

Dialing Cxn 1

→ This is a dialing connection associated with audio stream Stream 1. This connection will be initiated from this codec.

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**.

Configure Transport settings Stream 1->Dialing Cxn 1

ⓘ Transport:	IP
ⓘ Session Protocol:	SIP
ⓘ Encode/Decode:	Both
ⓘ Connect Timeout:	120	sec

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** (Ethernet 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, then 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 LAN2, then the call will proceed using LAN2. If you do not wish to use an account for the dial 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 be able to proceed.

Enter Destination Stream 1->Dialing Cxn 1

ⓘ Address:	203.38.199.163
ⓘ Via:	Any
ⓘ Account:	tieline_test@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:**
> * 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.

<div align="center">

Enter Destination Stream 1->Dialing Cxn 1

ⓘ Address:	203.38.199.163:5070
ⓘ Via:	Any ⌄
ⓘ Account:	tieline_test@getonsip.com

</div>

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.

<div align="center">

Select Encodings Stream 1->Dialing Cxn 1

ⓘ Transmitting Algorithm

Encoding:	G.722 ⌄
Sample rate:	16 kHz ⌄
Bit rate:	64 kbps ⌄

ⓘ Receiving Algorithm   ☑ Use Tx ⓘ

Encoding:	G.722 ⌄
Sample rate:	16 kHz ⌄
Bit rate:	64 kbps ⌄

</div>

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.
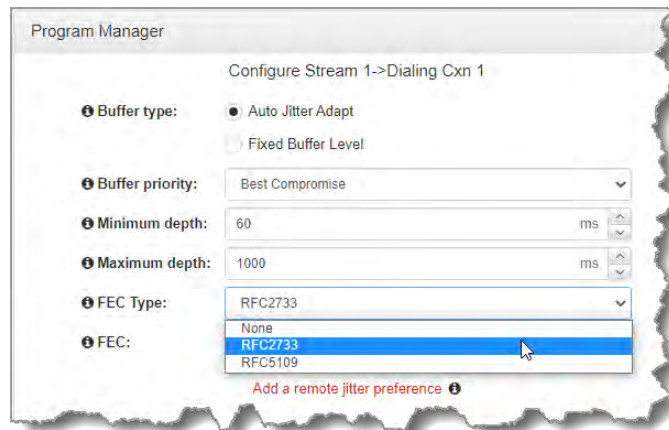
10. Click **Add a remote jitter preference** to send preferred jitter settings to a remote codec. Note: this is just a preference as per EBU Tech 3368 and there is no guarantee that the remote codec will accept or support these jitter configuration settings. Verify configuration settings on the remote 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.
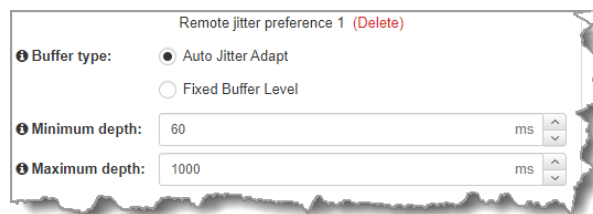
11. Click **Next** to select the check-box if you want to **Enable Auto Reconnect**.
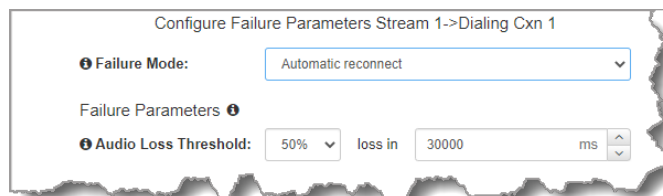
12. Click **Next** to name the answering connection for when calls are received by the codec.
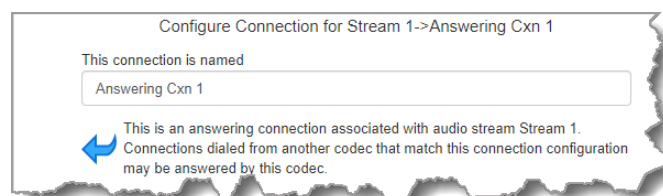
13. Click **Next** to configure the **Session Protocol** as **SIP** for the answering connection to receive a SIP call.

Configure Transport settings Audio Stream 1->Answering Cxn 1

❶ Transport: IP ▼

❶ Session Protocol: SIP ▼

14. Click **Next** to configure the jitter buffer settings for the answering connection. Note: it is also possible to configure remote jitter preferences if the remote codec supports RFC5109.

Configure Stream 1->Answering Cxn 1

❶ Buffer type: ⦿ Auto Jitter Adapt
◯ Fixed Buffer Level

❶ Buffer priority: Best Compromise ⌄

❶ Minimum depth: 60 ms

❶ Maximum depth: 1000 ms

❶ Jitter depth: 500 ms

15. Click **Next** to configure **Failure Parameters** for the answering connection if required. **Please note**: In most situations the default answering **Failure Parameters** do not need adjustment. These settings may be useful to troubleshoot certain connections, e.g. satellite IP links.

Configure Failure Parameters Stream 1->Answering Cxn 1

Failure Parameters ❶

❶ Audio Loss Threshold: 50% ⌄ loss in 30000 ms

☐ **Create another answering connection**

16. 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** to save the program at this point.
   iii. Click **Next** to configure **Output Audio Source** options.

Configuring Output Audio Source Options

1. Click **Next** to configure **Output Audio Source** options and automatically switch between up to 4 backup audio sources to maintain program audio at transmitter sites. **Output Audio Source** options include:

- **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
- **Input**: Input audio looped to the physical codec outputs.
- **HTTP**: Icecast client mode to allow media server streaming from a specified URL.
- **USB File Playback**: Audio file playback from a USB stick.

2. Click the blue **Plus symbol** ✚ to add a backup **Output Audio Source**, or click the **Minus symbol** ➖ to remove an **Output Audio Source**. Click the drop-down arrow to select an **Output Audio Source** option.
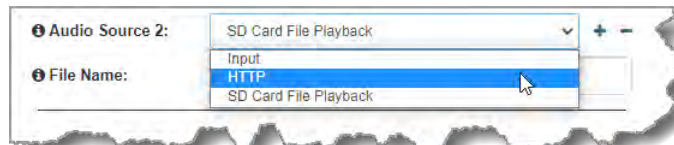
3. Configure silence threshold parameters for enabling a preferred backup option, as well as resume thresholds for reactivating a previous source. Then click **Save Program** to save program settings.
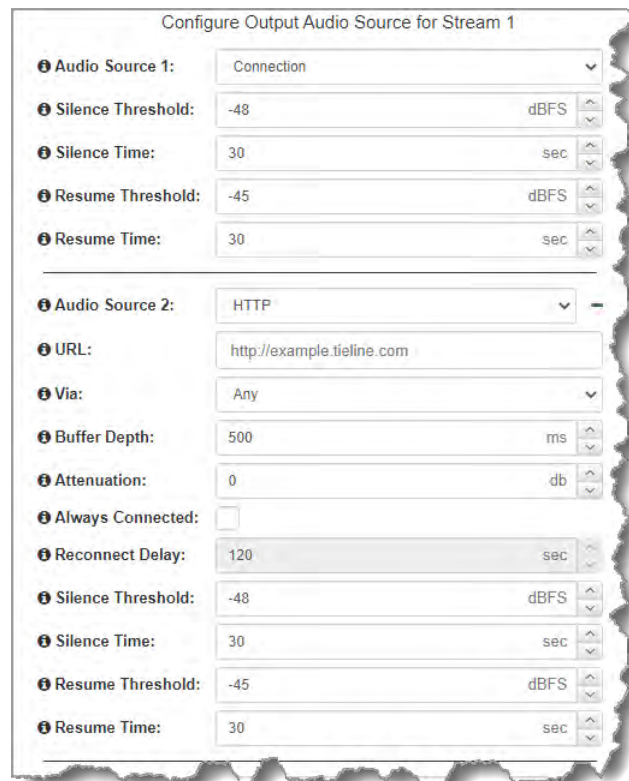
4. After configuring **Output Audio Source** options you can:

   i. Click **Save Program** to save the program at this point.
   ii. Click **Next** to configure rules options.

> **Important Notes for File Playback:**
> - Ensure MP3 recordings used are not variable bit rate files.
> - File playback audio is sent directly to the codec outputs and therefore IGC is not available. When you create your MP2 or MP3 files ensure the audio levels match the audio reference level of your codec and that peaks average at the correct levels.
> - USB backup audio is only sent to the outputs of the local codec to which a USB stick is attached. USB file audio is not sent to encoders and cannot be transmitted via an audio stream to another codec.

- The USB stick can be inserted or removed at any time as long as the codec is not already playing audio in failover mode. Removing the USB stick while audio is playing from it will result in poor audio quality and should be avoided. If it is removed accidentally you must reboot the codec to ensure USB failover will work reliably in future.
- If you enter a single file name ensure you add the file extension, e.g. "test.mp3", or the file will not play back.
- If you enter a directory name, all the files within the directory will be played back. We recommend you save all audio files as a playlist and link to this if you want them to play out sequentially. Please note that "M3U" is the playlist file format supported by the codec.
- File playback will occur automatically if the silence threshold parameters are breached; if the codec is not connected for any reason file playback will commence. To stop file playback open the **Connections panel** in the HTML5 Web-GUI, click to select the file playback connection, then click **Disconnect**.

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: Program level rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

2. Click **Save** to save the program.
3. Click **Finish** to exit the wizard.
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 connect the program manually using the front panel.

23.19 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 16 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, the call 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** by default for the first audio stream in a multiple mono stream program).
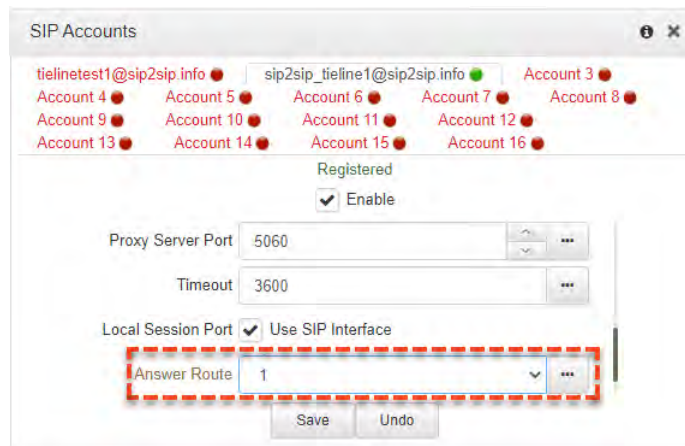
2. **SIP Interfaces**: When answering multiple incoming peer-to-peer SIP calls, configure an answer **Route** for incoming peer-to-peer calls via the **SIP Interface panel** and route them to specific audio streams. Configure answer routes in **Interface SIP1** and **Interface SIP2** to match answer 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** by default for the second audio stream in a multiple mono stream program).

Important Notes:
- By default local audio ports are arbitrarily assigned when answering calls using SIP accounts. The default 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.
- By default, the session port used for each SIP account is the associated SIP interface session port. The default session port is the registered UDP port number 5060. It is also possible to configure a custom local session port for each SIP account for compatibility with Cisco Unified Communications Manager (CUCM). Ensure your firewall has the required TCP and UDP ports open when receiving multiple SIP calls. See Installing the Codec at the Studio for more information.

Local Session Port 5061 ··· ☐ Use SIP Interface
Answer Route Any

- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono or stereo call.
- Failover is not available with SIP and SmartStream PLUS redundant streaming is not available with SIP or sessionless IP.

23.20   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, you must use the **Program Manager panel** 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**.

Connections   ✕

⦿ Programs   ◯ Default Templates

Default Answering   ▾   Unload

Ready to answer.

The default answering program is loaded. The codec will attempt to answer any incoming connection. ❶
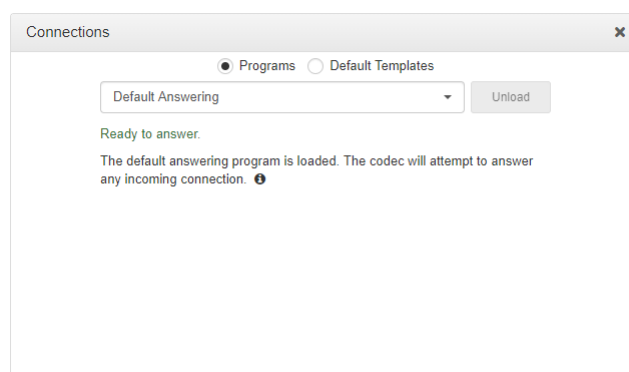
2. Select **Programs** and then click the drop-down arrow to select one of the programs in the codec, then click **Load**.
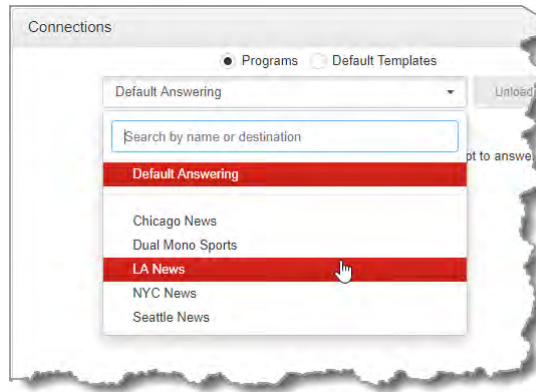
3. If **Lock Loaded User Program** has been configured in the **Options panel**, a black **Padlock** symbol appears next to the program name in the **Connections panel**, to indicate a program is locked in the codec.
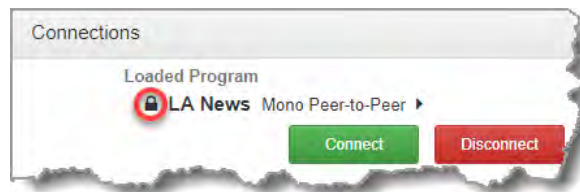
Unload a Program

1. To unload a loaded program click **Change Program**.
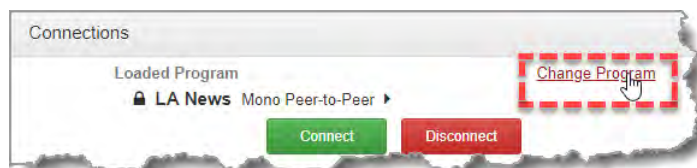
2. Click **Unload**.
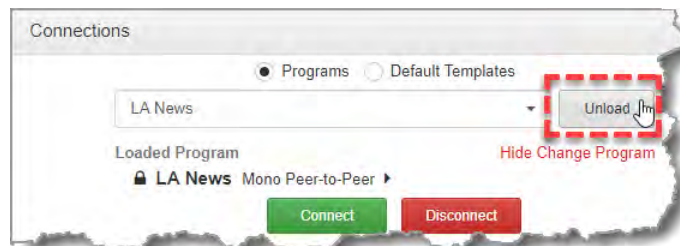
Connecting a Program

To connect audio streams and connections within an existing program there are three options:

1. Click the **Connect** [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.
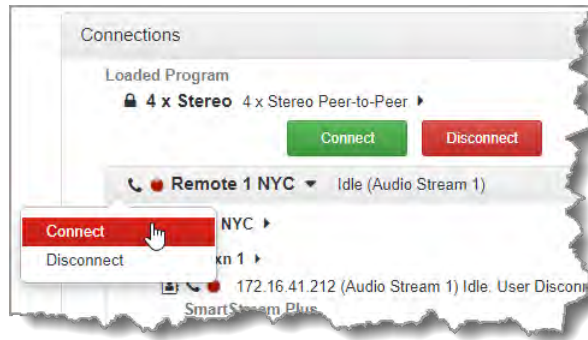
3. Click the connection **Connect/Disconnect** 📞 symbol and then click **Connect**; this connects an individual audio stream connection.
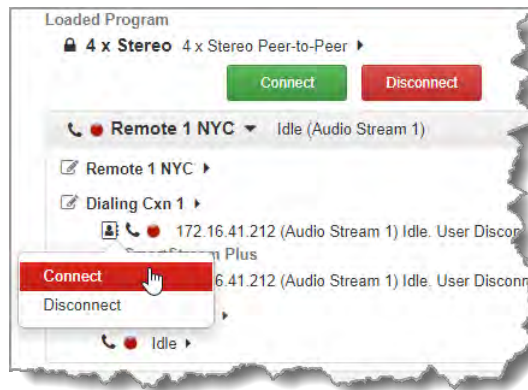
Disconnecting a Program

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.
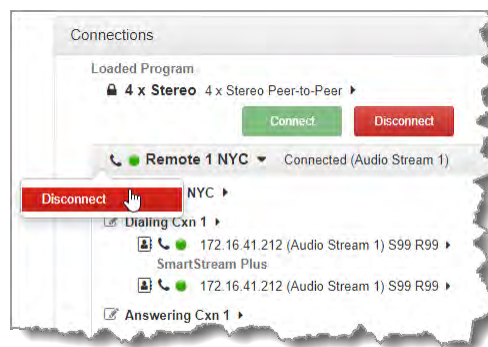
3. Click the connection **Connect/Disconnect** 📞 symbol to disconnect an individual audio stream connection.
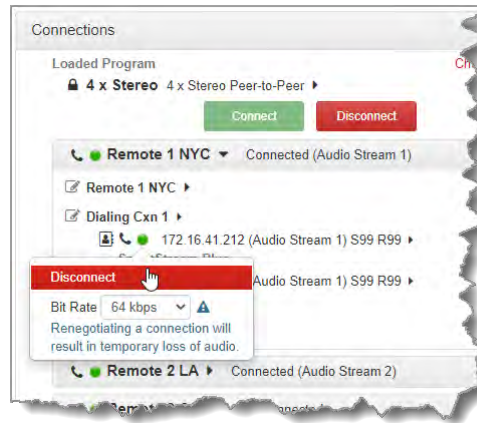
Change Dialing Settings

To edit destination dialing settings:

1. Click the **Edit symbol** ✎ adjacent to a connection. Note: The connection must be disconnected before editing.
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.
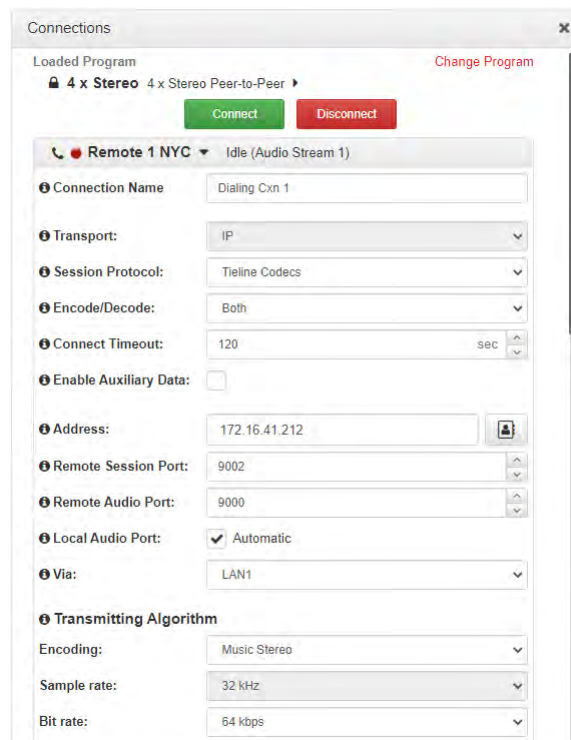
23.21   Configure Speed Dialing

The Toolbox HTML5 web-GUI supports dialing programs using preconfigured speed dials. This is configured and managed using the **Speed Dial panel**.

Create a New Speed Dial

To configure speed dialing:

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then click **Speed Dial** to open the **Speed Dial panel**.
2. Click **Edit**.
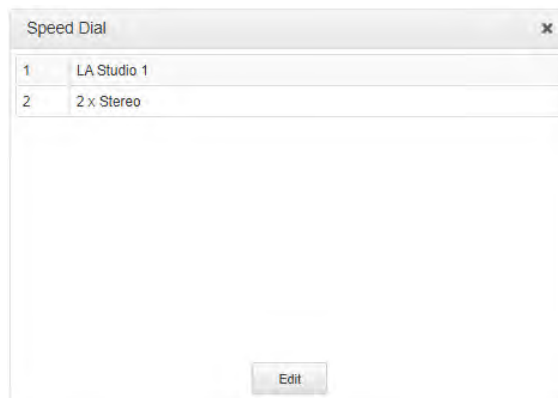
3. Click **New** to create a new speed dial and then click the arrows on the left side of the panel to select the speed dial number.
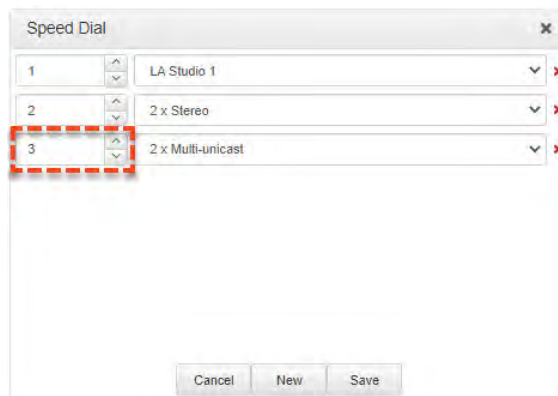
4. Click the drop-down arrow on the right side of the panel to select the program to associate with the speed dial number.
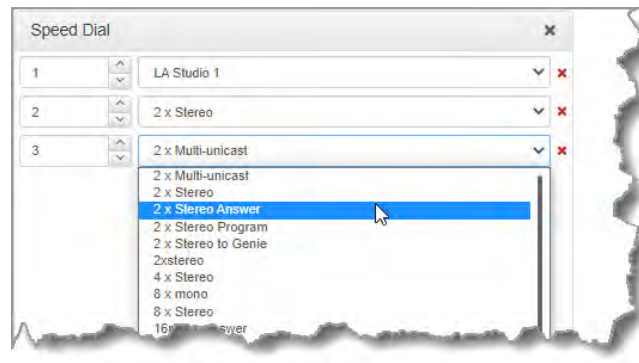
5. Click **Save** to store all edited speed dials.

> **Important Note:** To view speed dial configurations using the front panel of the codec select **SETTINGS** 🔧 > Speed **Dial** and press the 🆗 button.

23.22   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.
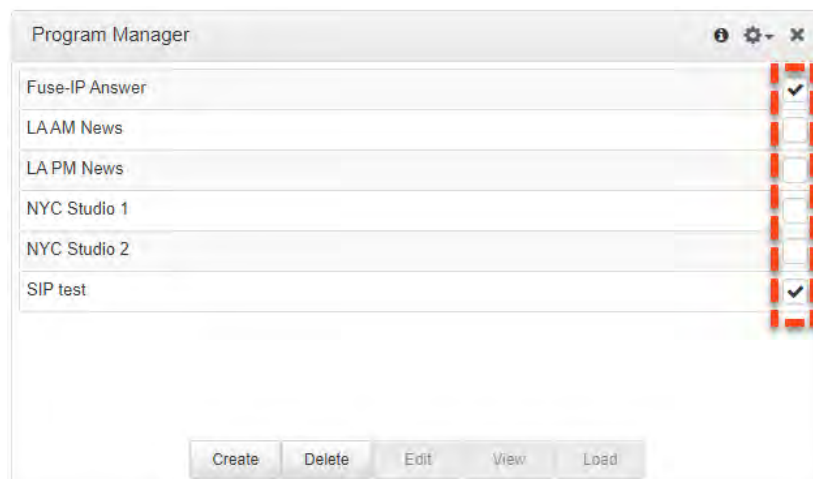
3. Click the **Delete** button.
4. Click **Yes** in the **Delete Selected Programs** confirmation dialog to delete all selected programs.
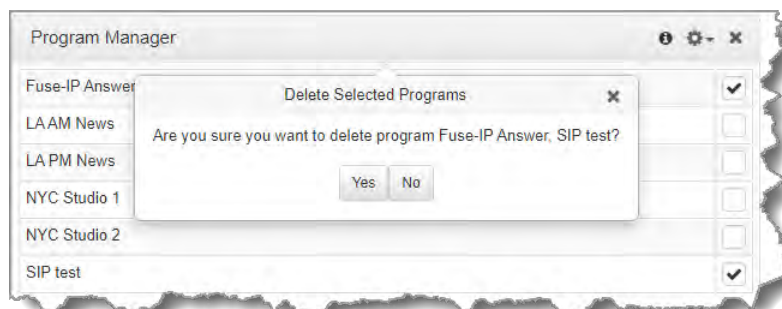
> **Important Notes:** Any program that is currently loaded or listed in the **Scheduler Events panel** cannot be deleted.

23.23   Matrix Editing

Product	Feature	Notification
Gateway 4	Matrix Editor panel	The Gateway 4 codec supports 4 audio channels only and this cannot be upgraded or expanded. 4 inputs and outputs are displayed in the Gateway 4 Matrix Editor panel.
Gateway 8/16	Matrix Editor panel	The Gateway displays between 8 and 16 channels in the Matrix Editor panel based on the number of channels purchased. If a codec supports fewer than 16 channels, it can be upgraded to support more channels by purchasing a channel license upgrade from Tieline and installing it in the codec.

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.

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 for the Gateway 8 is loaded, as indicated at the top of the panel. The Gateway 4 codec will only display 4 inputs and outputs only.
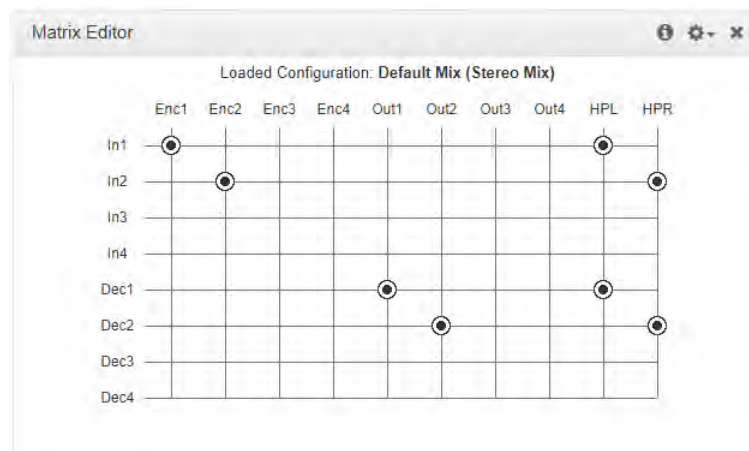
Default Stereo Matrix for Gateway-4 codec

The Gateway codec supports more channels and these are displayed in an expanded matrix in the panel. When a codec is upgraded to support more channels the matrix displays more inputs, outputs, encoders and decoders as per the license upgrade.
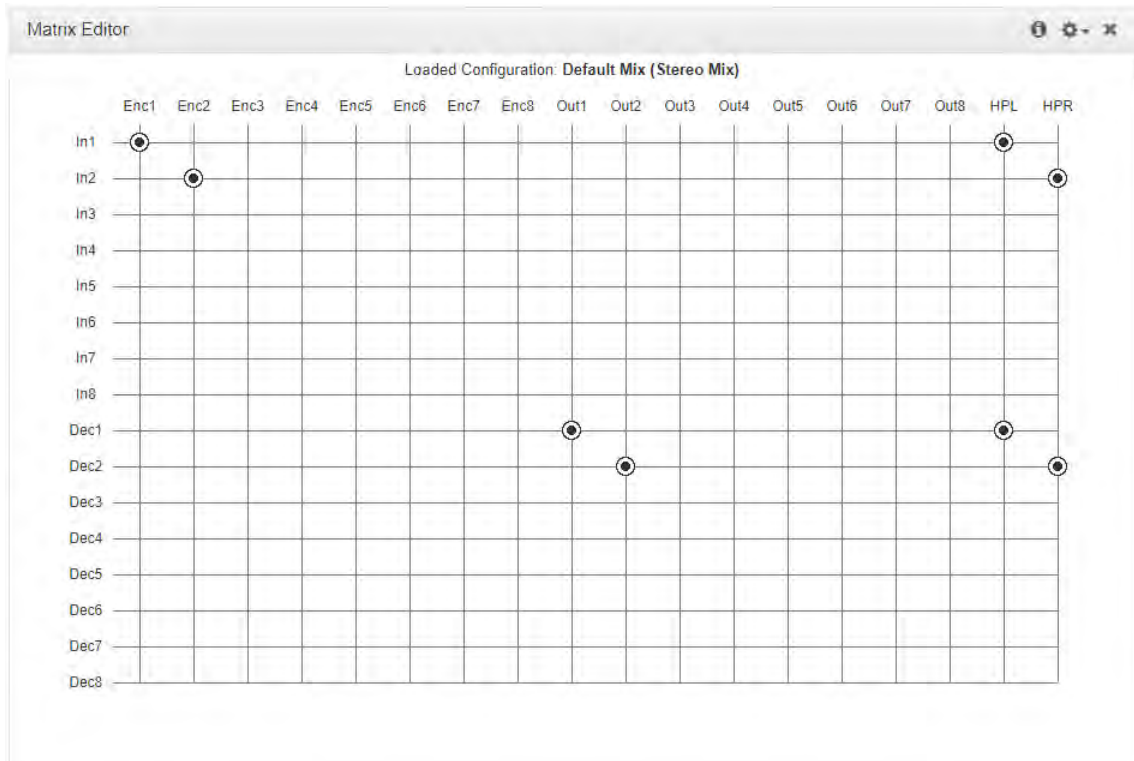
Default Stereo Matrix for Gateway Licensed for 8 Channel Streaming

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).
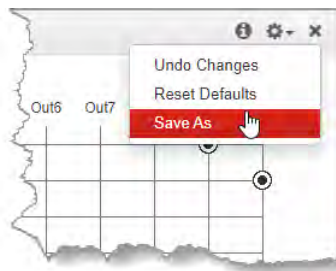
Important Notes:
- Click the **Information symbol** ⓘ at the top of the panel to view details about the current matrix.
- The number of channels available in the **Matrix Editor** depends on how many channels are licensed. This can vary between 8 and 16 channels for a Gateway codec.

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. 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.
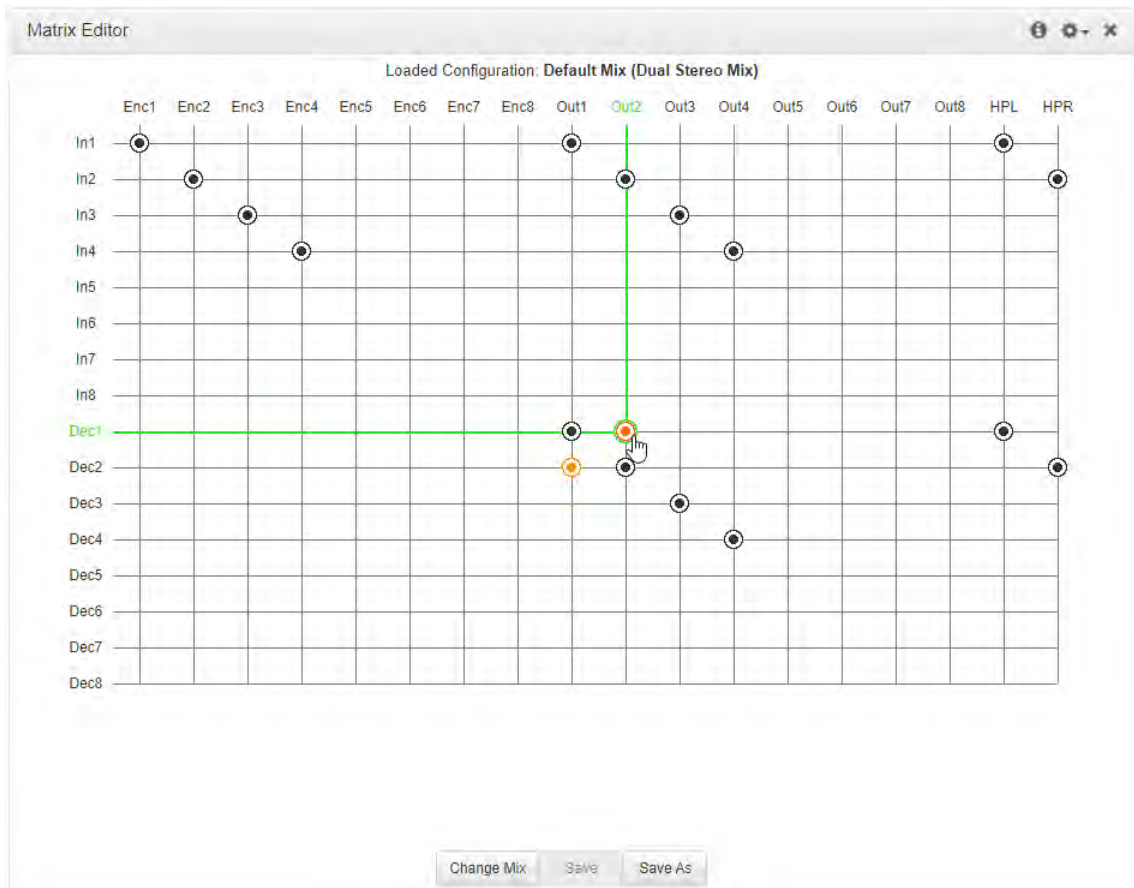
Edits can be saved as a custom mix as follows:

1. Click the **Options symbol** ⚙ and select **Save As**.
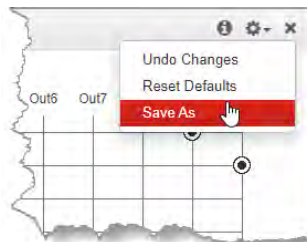
2. Enter a new **Name** and **Description** (if required) and click **Save**.

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 previous runtime changes. 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.

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**.
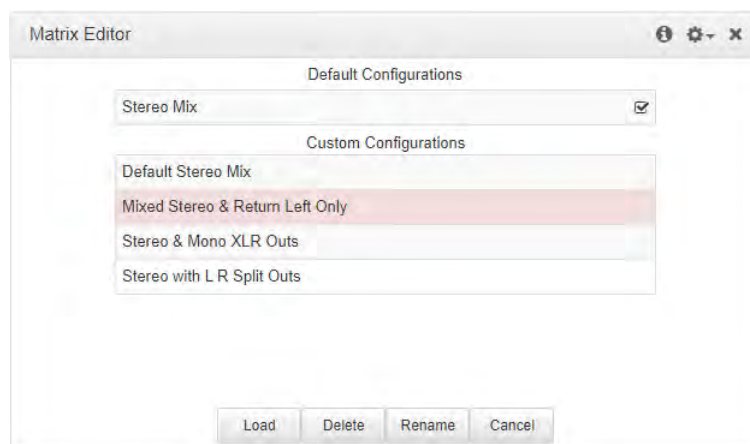
2. To rename a custom mix, click to select a saved mix and then click **Rename**. Edit the name and click **Save**.
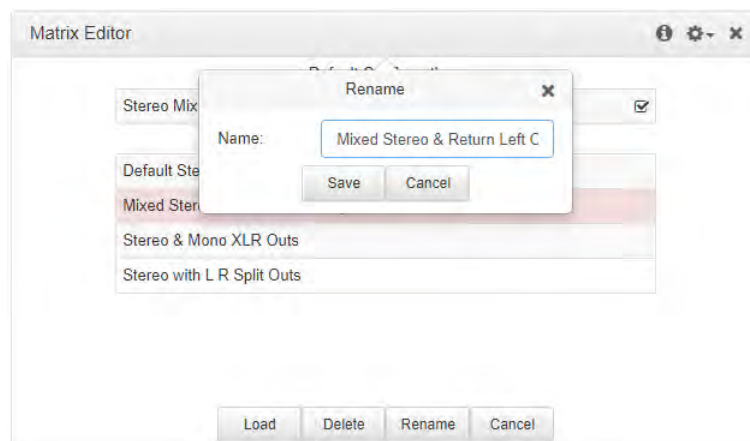
3. To delete a custom mix, click to select a custom mix and then click **Delete**. Note: A loaded mix cannot be deleted.
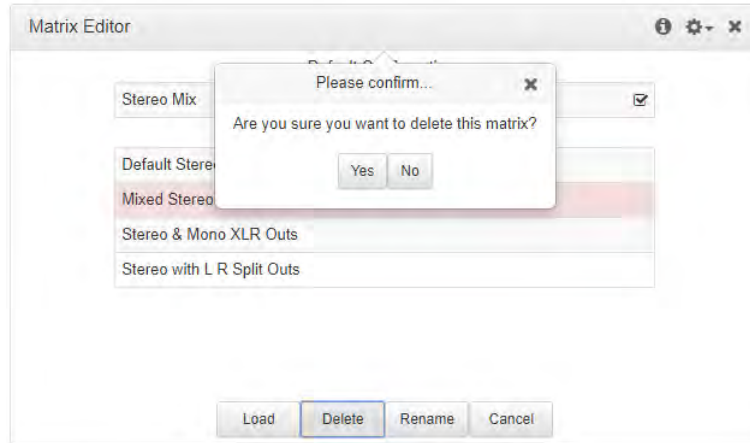
23.24   Adjusting the Connection Bit Rate

1. Open the HTML5 Toolbox Web-GUI and click **Connect**, then select **Connections** to open the **Connections panel**.

2. Click the **Connect/Disconnect** 📞 symbol for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate.
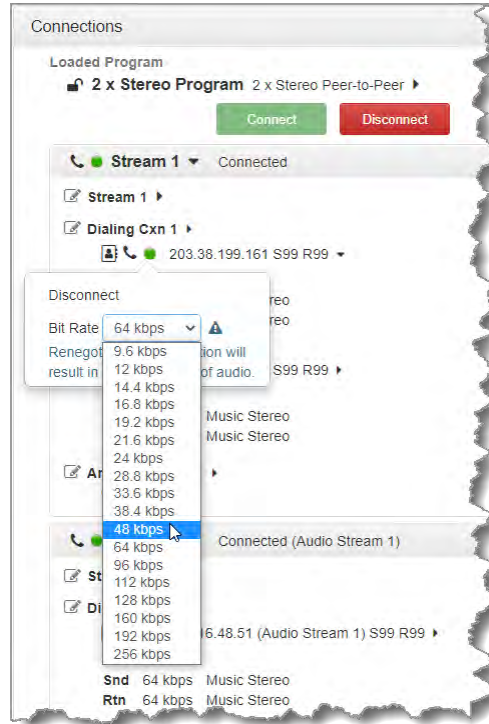
> **Important Notes:**
> - Renegotiation will result in a temporary loss of audio.
> - It is not possible to renegotiate the connection bit rate of a SIP connection.

23.25   Configuring ISDN

A single module slot is available to insert an optional Euro ISDN module into the codec for primary or backup connections. 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 is 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.

Tieline codecs fitted with a Gateway Euro ISDN module can provide high quality mono or stereo audio over a single B channel using the Tieline Music algorithm. With 2 B channels one can be used as a standby connection, or configure higher bandwidth stereo connections using 2 B channels with algorithms such as MusicPLUS and MPEG. It is also possible to bond two B channels to configure higher bandwidth mono connections.

Important Considerations

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

- Is a single B channel or 2 B channels required?
- Which network is being used?
- Is the ISDN line Point-to-Point or Point-to-Multipoint?
- What are the directory numbers (DN)?

ISDN configuration is influenced by the country in which you operate and a Service Profile ID (SPID) number does not need to be entered into a Tieline codec for operation outside North America.

S/T and U ISDN Interfaces

In Europe, the UK, most of Asia, Australia, Africa and parts of the Middle East, 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. 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. If an NT-1 device is connected to the U interface line then a Tieline Euro ISDN G6 module is required (S/T interface - model: TLISDNEUROG6). Note: The Gateway Euro ISDN module is suitable for Japan. Tieline does not support U interface module connections in North America as ISDN is either unreliable or unavailable in many markets.

> **Important Note:** Tieline Gateway Euro 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.

Use the HTML5 Toolbox Web-GUI to configure a dial and/or answer program with ISDN settings. It may be necessary to:

1. Configure ISDN module settings.
2. Configure ISDN Answering settings.

> **Important Note:** For detailed information about connecting with other brands of codec using ISDN visit www.tieline.com

23.25.1   Configuring ISDN Modules

ISDN settings in the **Modules panel** determine how each codec module operates at a particular site. A user may 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 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.
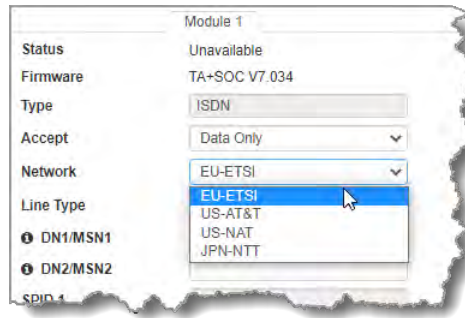
5. Click the drop-down **Line Type** arrow and select your preferred option. Ask your Telco whether your ISDN line is Point-to-Point or Point-to-Multipoint. By default select **Point-to-Multipoint**, unless your switch type is an AT&T 5ESS custom point-to-point.
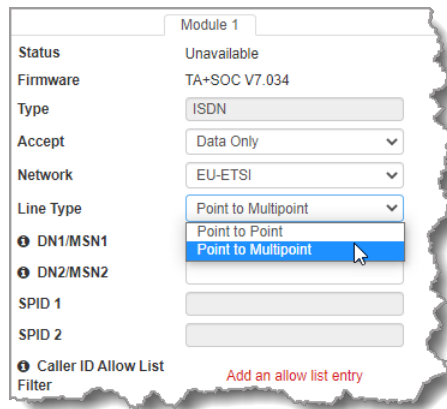
6. If you are in the US enter DN and SPID numbers as required, or in other regions enter DN or MSN numbers as required.
7. Click **Add an allow list entry** to only accept calls from numbers in the **Caller ID Allow List Filter**. All calls are accepted if there are no filter entries.
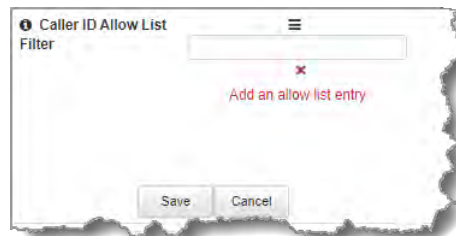
8. Click **Save** when configuration is complete.

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. Entering 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.

23.25.2   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.
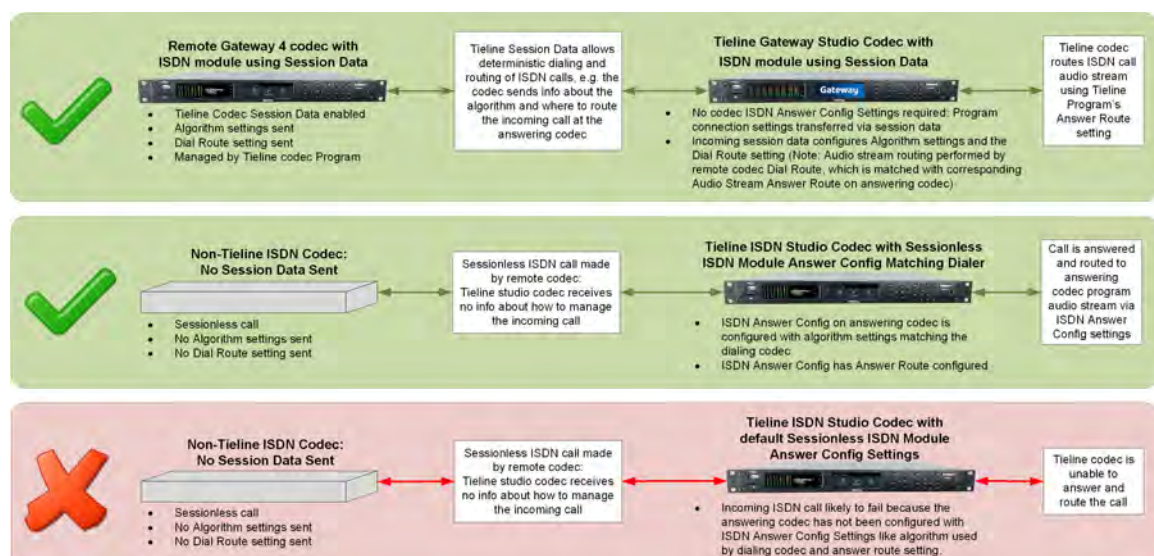
It is possible to save up to two different **ISDN Answer Configs**, which allows 2 ISDN B channels to be individually configured for unique answering behaviors. ISDN answering can be configured to suit:

- 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.

Each of the two available **Configs** visible at the top of the panel allow you to select which B channel or channels are used to answer a call or calls from incoming ISDN codecs.
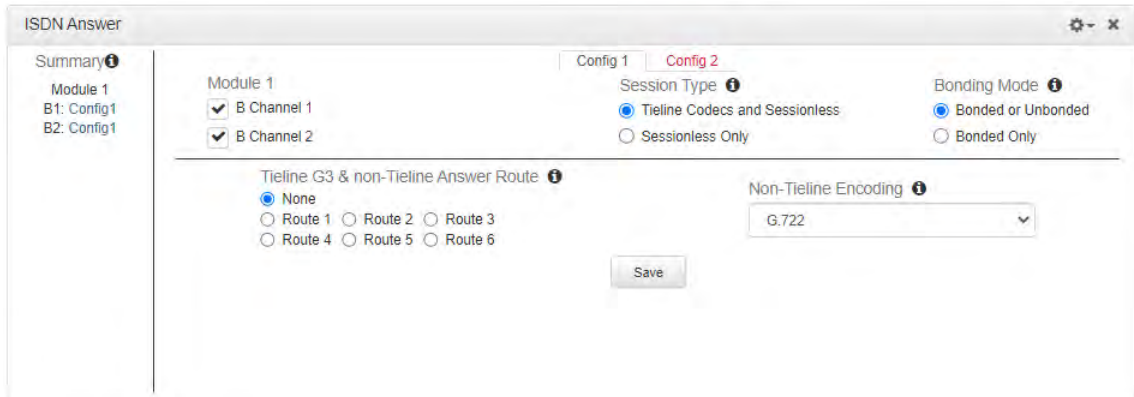
To reset ISDN answering to default settings click the **Options symbol** in the top right-hand corner of the panel and select **Reset all to factory defaults**.

> **Important Note:** A B channels can only be selected once and a **Summary** of allocated B channels is displayed on the left-hand side of the **ISDN Answer** panel.

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**.
2. Click to select a **Config**.
3. Select a B channel from those available and then click **Save**. The connection is not bonded if only one B channel is selected.
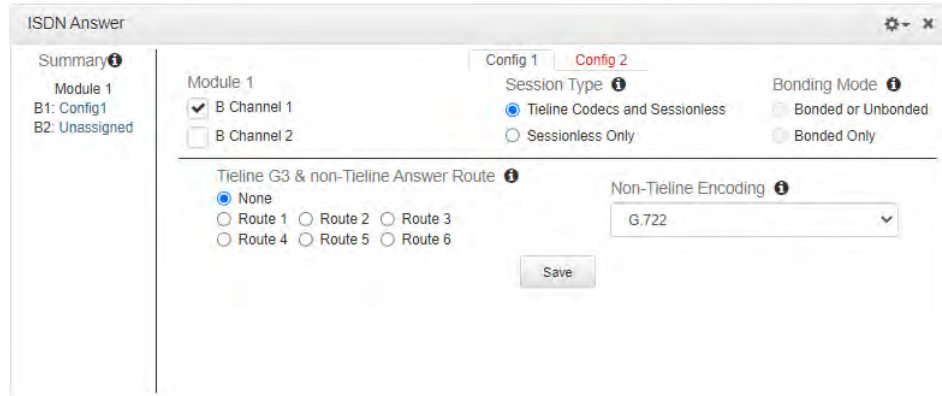
Multiple B Channel Bonding Config

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**.
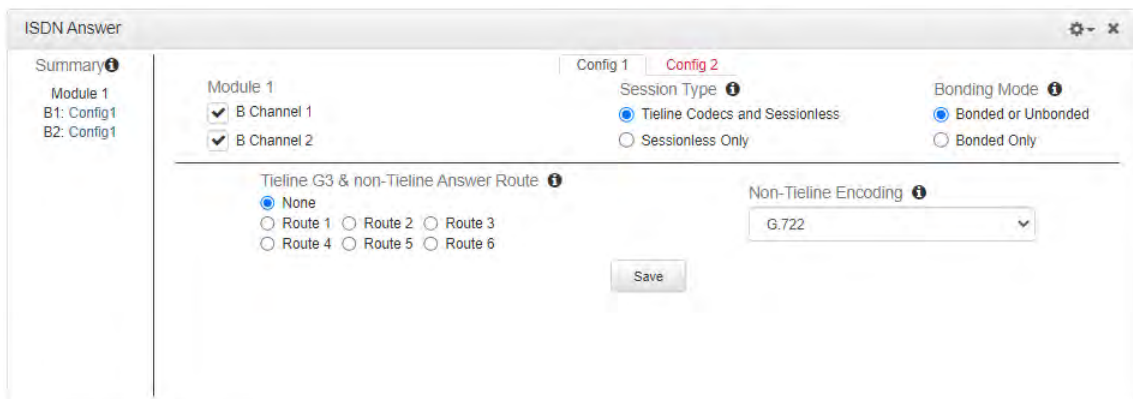
3. Configure the bonding setting that best suits the audio stream associated with this **Config**. **Bonded or Unbonded** is the best setting in most situations.

Bonding Setting	Behavior
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

4. 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 newer codecs (not G3), it is possible to configure a **Dial Route** in the dialing codec program and a corresponding **Answer Route** in the answering codec 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 receiving calls from these codecs.
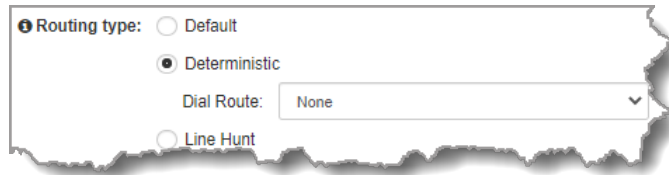
In principle, the concept of 'routes' operates similarly to how audio ports are used to route multiple audio streams over IP. Selecting different IP audio port numbers allows users to define which incoming IP audio stream is routed to a specific answering audio stream configuration on the codec. This ensures inbound calls from multiple codecs can be consistently routed to the same answering codec audio streams.

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**).

4. Click **Save** when configuration is complete 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** in the following image.
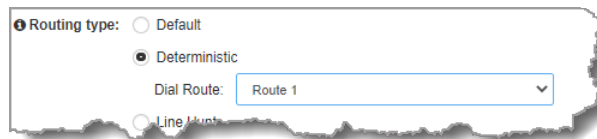
More detailed information about how to configure the codec to answer and route multiple sessionless ISDN calls is available in Using ISDN Answer Routes for Sessionless ISDN Calls. This includes examples to explain how to set up consistent deterministic routing of multiple incoming sessionless calls.

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.
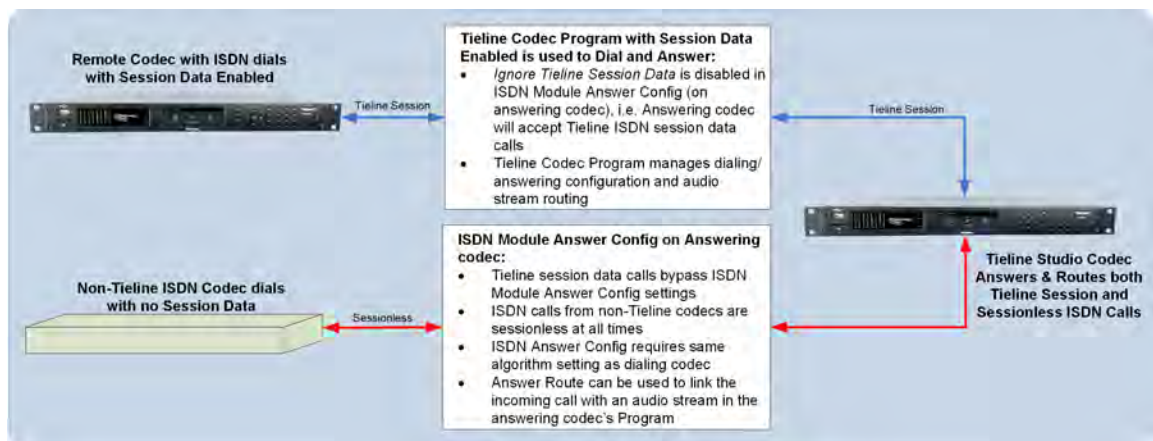
Allow Answering of Sessionless ISDN Calls Only

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.
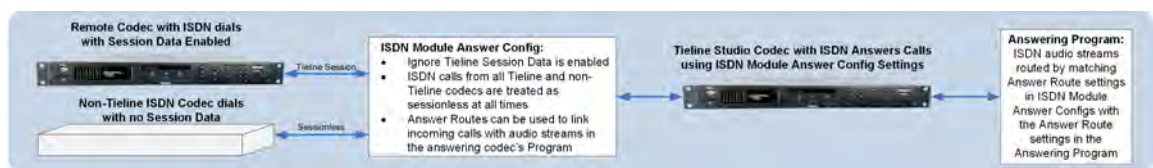
Default Answering Settings

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

23.25.3   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):
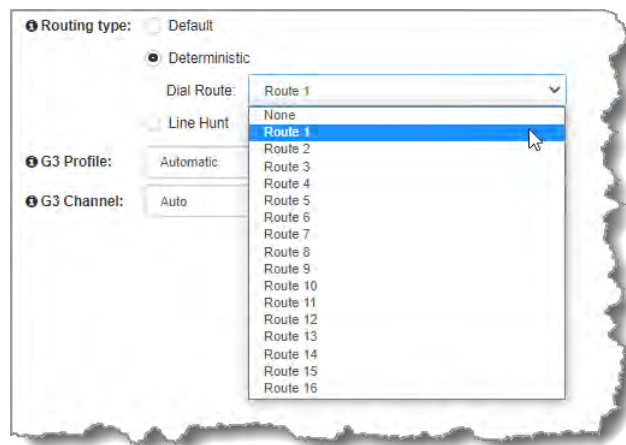
You can use any **Answer Route**, for example **Route 1** for ISDN Audio Stream 1 and **Route 2** for ISDN Audio Stream 2. The **Answer Route** number doesn't have to match the audio stream number because the route you select will be used by the incoming ISDN call. This is similar to how an "extension 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.
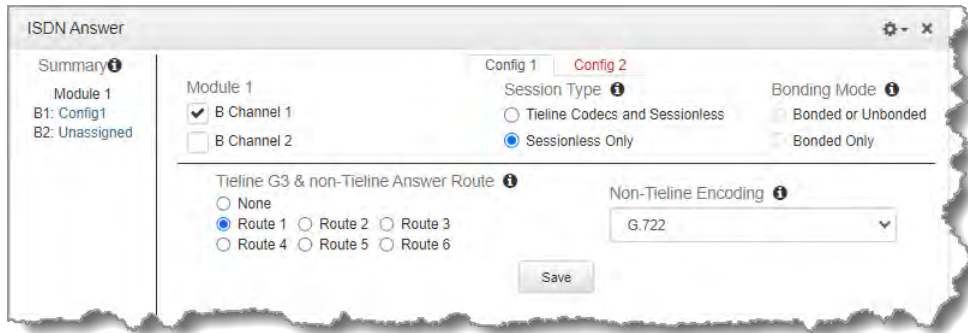
2. Next select **Config 2** and **Sessionless Only** and **Route 2**. Select your preferred algorithm, then click **Save**. This means that Module 1 B channel 2 will answer a sessionless ISDN call using these settings.
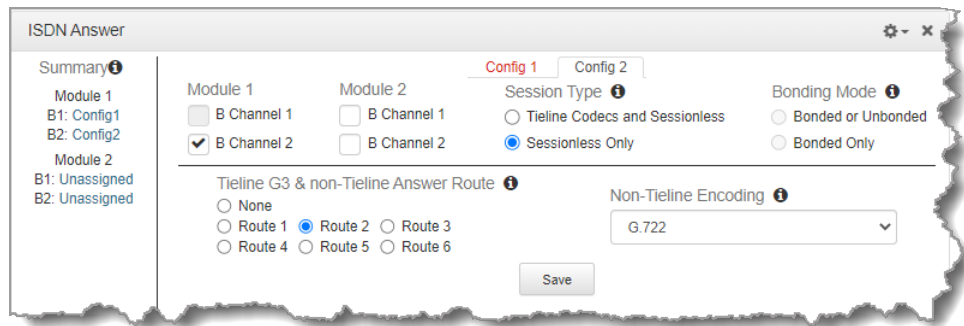
Both ISDN B channels can now answer incoming sessionless ISDN calls. If it doesn't matter which incoming codec call is answered by which B channel then that's all you need to do. If, however, you want each non-Tieline codec to use the same B channel and be routed to the same codec output consistently, 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**.
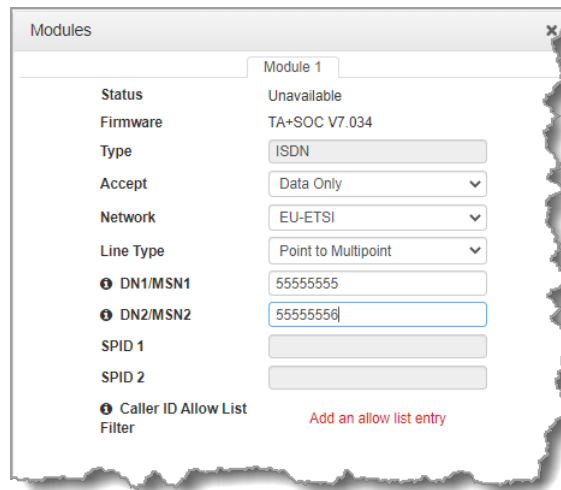
If codec 1 always uses the first directory number to call then it will always be routed via **Route 1** to the Answering Audio Stream Connection using **Answer Route 1** (configured in step 1). Codec 2 should always use the second directory number and then it will always be routed via **Route 2** to the Answering Audio Stream Connection configured with **Answer Route 2**.

23.26 Reset Factory Default Settings

There are several options which allow you to restore 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**.
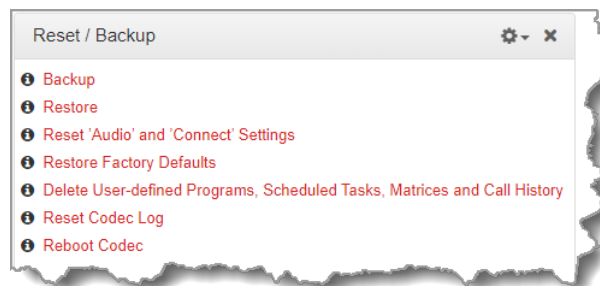
2. Click one of the available reset options to adjust codec settings, or reboot the codec. Note: Hover with the mouse pointer over the **Information** ⓘ symbol to view a tool-tip for each reset option.
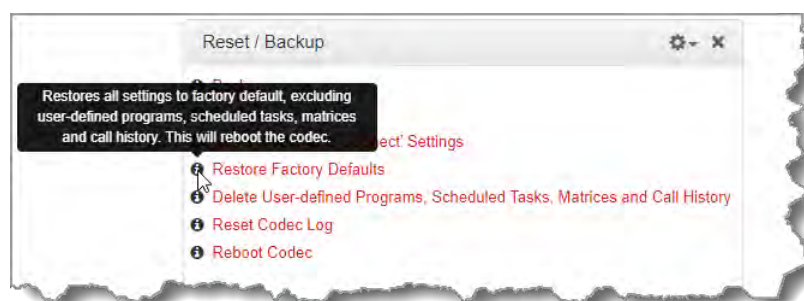
3. A confirmation dialog appears for each option; click **Yes** to proceed.
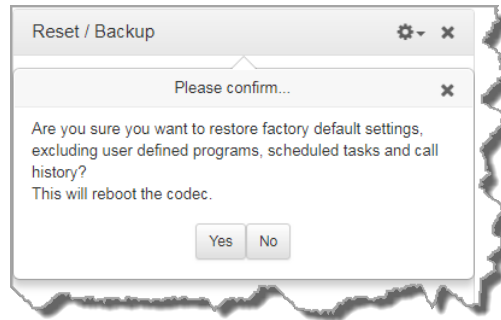
23.27 Backup and Restore Functions

The HTML5 Toolbox Web-GUI can be used to backup and restore codec settings, including:

- Programs containing a variety of connection settings and scheduler configuration settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

Files can also be used to copy configurations onto other similar codecs. Programs are essentially connection profiles that may include:

- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

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 / Backup panel**.
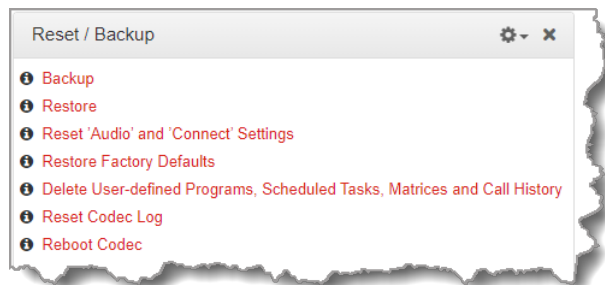
2. Click **Backup**.
3. Click to select the check-boxes to confirm your backup requirements, then click **Backup**.
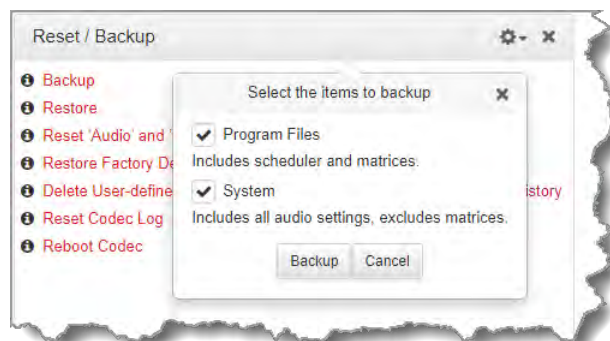
4. Select a location on your PC to save the configuration file. Note: You may need to "allow" your browser to display the pop-up dialog.

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 to only copy programs onto codecs.
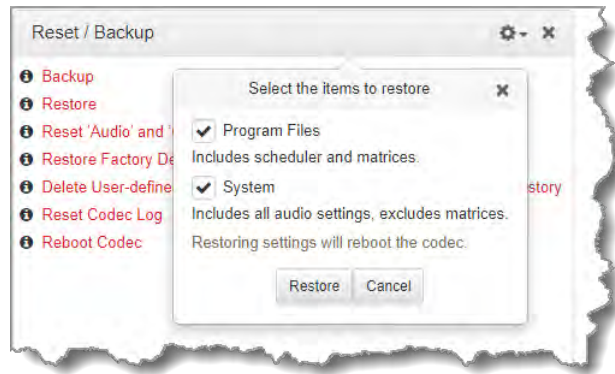
4. Click **Restore** and select the .tgz file you want to load onto the codec. A **Success** dialog confirms the files have been restored.
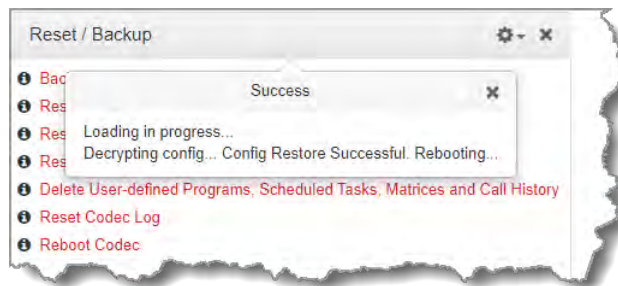
Note: The codec will automatically reboot when restoring system settings.

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**.
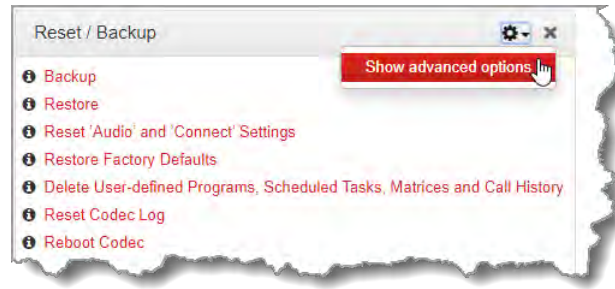
3. Click to select **Import/Export editable XML config** as required, or force the codec to reboot.
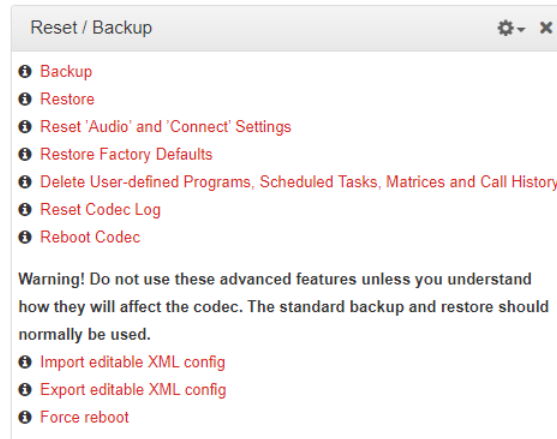
23.28   Import and Export Programs

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 to be exported.

3. Click the **Options symbol** ⚙ in the top right-hand corner of the **Program Manager** and select **Export Selected Programs**.
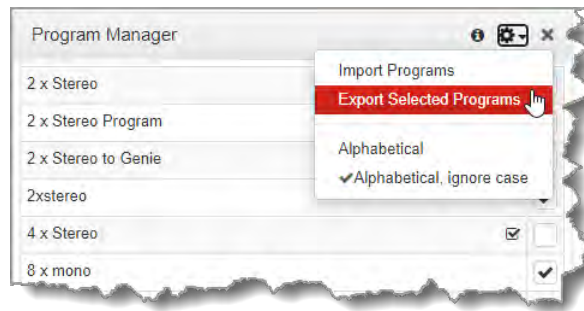
4. Navigate to a folder and save the program .zip file.
5. Click **Save** to save the program file.

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. Click the **Options symbol** ⚙ in the top right-hand corner of the **Program Manager** and select **Import Programs**.
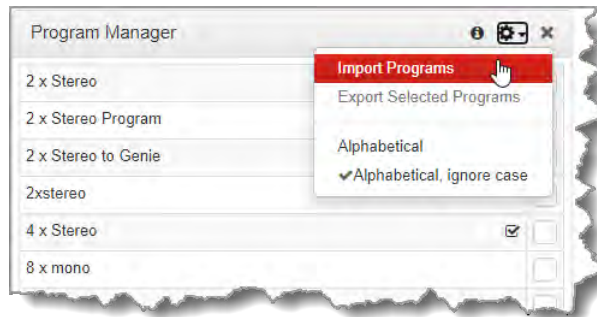
3. Navigate to the file folder containing the program .zip file to be imported. Click to select the .zip file and click **Open** to import it.

23.29 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 click **Edit**.
3. Click the **Lock Loaded User Program** check-box to lock or unlock a user program in the codec, then click **Save**.
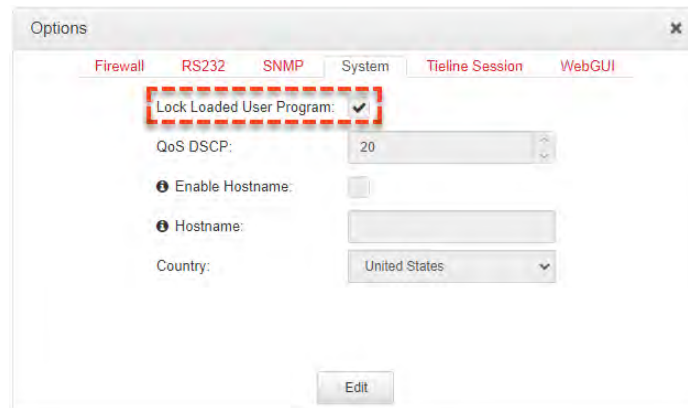
> **Important Note:**
> - A black **Padlock** symbol appears next to the program name in the **Connections panel** and in the **Program Loader panel** (in the Quick Connect web-GUI), to indicate a program is locked in the codec.
> - It is only possible to lock custom programs in a codec.
> - If **Lock 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.

23.30   Configure Country Setting

The **Country** setting in the codec configures country-specific settings like the 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 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 **Edit**.
3. Click the **Country** drop-down menu arrow to select the country of operation.
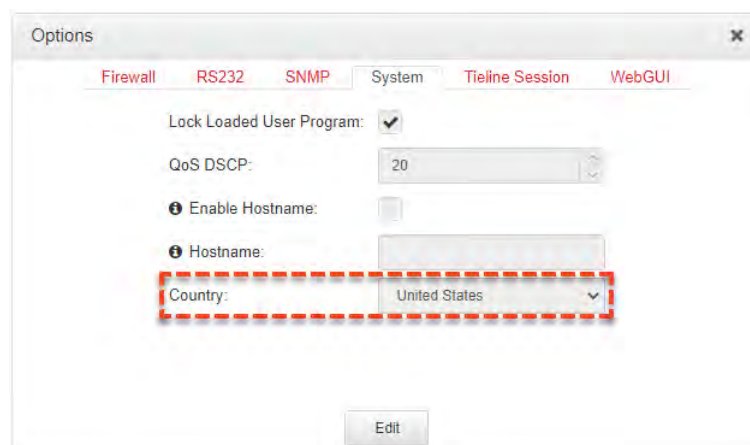
4. Click the **Save** button to store the new configuration.

23.31   Configuring SNMP in the Codec

The codec supports Simple Network Management Protocol (SNMP) for managing devices on IP networks. There are two elements to configuring SNMP in your codec:

1. Configure SNMP Device settings in your codec.
2. Configure SNMP Traps via the **Alarms Panel** in the Web-GUI (see Configuring Alarms, or to configure settings and traps using the codec front panel see Configuring SNMP Settings).

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.
Trap Destination	SNMP provides the ability to send traps (notifications or alerts), which are packets containing data relating to a system component. The destination is the end-point to which notifications and alerts are sent. See SNMP Trap Configuration.

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. Click the **Download MIB files** button to download the MIB .zip file to a PC and import the contents into the MIB browser used to manage SNMP-enabled network devices.

MIB files can also be downloaded from the codec using the following link in a web browser on a device connected to the same network as the codec:

- http://<YOUR_CODEC_ADDRESS>/mibs/tieline-mibs.zip

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/.
- Tieline recommends SNMP is disabled if a codec is connected to a public network like the internet. Adjust settings using the Toolbox HTML5 Web-GUI **Options panel** in the **Firewall** tab, or see Firewall Configuration.

23.32 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**.

Support

Resources
Online user manual
Visit the Tieline Support website
Email Tieline Support

System Diagnostics
Event History
🛈 Download System Logs
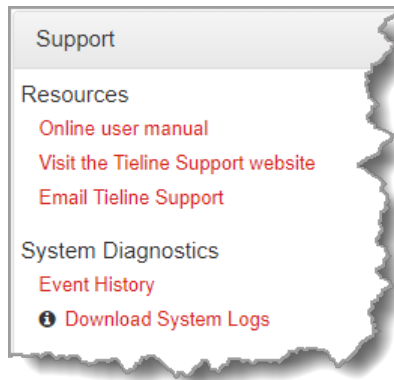
3. Save the file to your computer and then send the .zip file to Tieline support at support@tieline.com

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.

23.33   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.
- Automatically connect and disconnect individual audio streams, including editing stream settings like IP address, port and encoding details.
- 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.
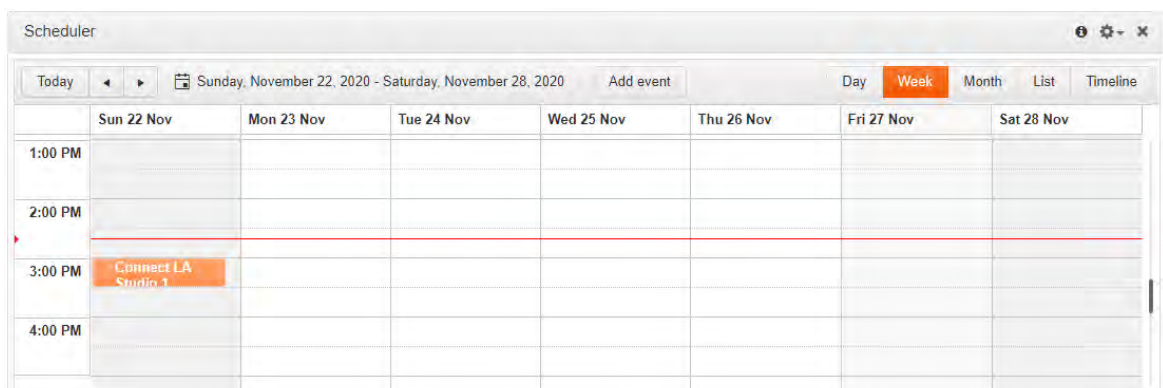
3. Add a **Title**, select a **Task**, **Program** and **Edit Stream** settings as required. Adjust the **Start** and **End** time for each event. It is also possible to adjust the frequency of an event, e.g. daily, weekly or monthly.

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.
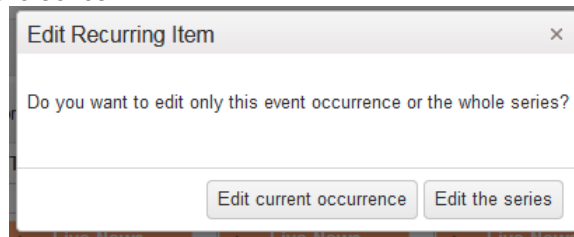
3. Select the preferred option and edit settings in the Event, then click **Save** to store all changes. Click **Delete** to delete an Event.
4. A confirmation dialog is presented to confirm the deletion of an event.
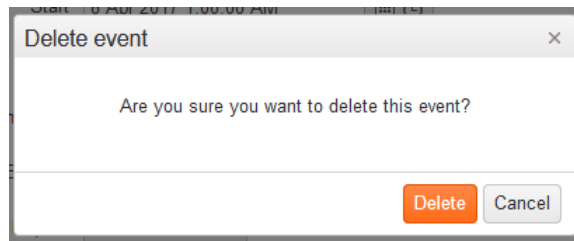
View Scheduler Events

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	Stream Name	Codec Time (UTC)	Browser Time	Action
Connect LA Studio 1	2 x Stereo		Fri, 27 Nov 2020 07:00:00 GMT	11/27/2020, 3:00:00 PM	Load & connect
Connect LA Studio 1	2 x Stereo		Fri, 27 Nov 2020 07:30:00 GMT	11/27/2020, 3:30:00 PM	Disconnect program
Connect LA Studio 1	2 x Stereo		Mon, 30 Nov 2020 07:00:00 GMT	11/30/2020, 3:00:00 PM	Load & connect
Connect LA Studio 1	2 x Stereo		Mon, 30 Nov 2020 07:30:00 GMT	11/30/2020, 3:30:00 PM	Disconnect program
Connect LA Studio 1	2 x Stereo		Tue, 01 Dec 2020 07:00:00 GMT	12/1/2020, 3:00:00 PM	Load & connect

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.

Scheduler History							✕
Drag a column header and drop it here to group by that column							
Title	Program Name	Stream Name	Codec Time (UTC)	Browser Time	Action	Result	
Connect LA Studio 1	2 x Stereo		Fri, 27 Nov 2020 07:00:00 GMT	11/27/2020, 3:00:00 PM	Load & connect	Invoked	

Purge Scheduler History

Other Scheduler Options

Click the **Options symbol** ⚙ in the top right-hand corner of the Scheduler panel to reveal a drop-down menu displaying other available options, including:

1. Generate a PDF view of scheduled events.
2. Adjust the Scheduler Timezone.
3. Enable / Disable the Scheduler.
4. Clear all Scheduled Events.
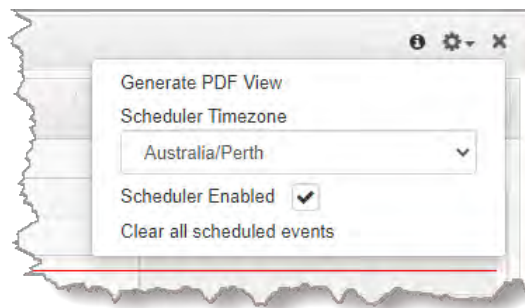
23.34   Configuring Alarms

Open the HTML5 Toolbox Web-GUI and click **Alarms** in the **Menu Bar** to open and view panels used to configure and monitor a range of alarms.
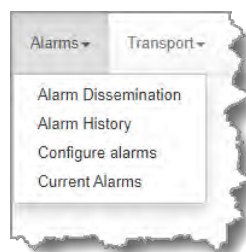
Configure and Enable Alarms

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then select **Configure alarms** to open the panel.
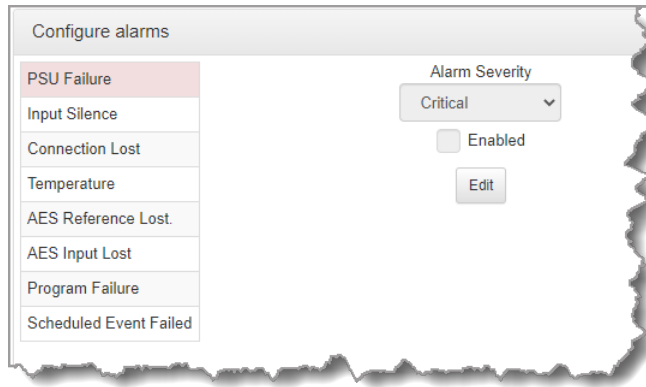
2. Click to select an alarm from the list on the left side of the panel.
3. Click **Edit** to configure alarm settings.
4. Click the **Enabled** check-box to activate the alarm and then select an **Alarm Severity** level from the drop-down menu.
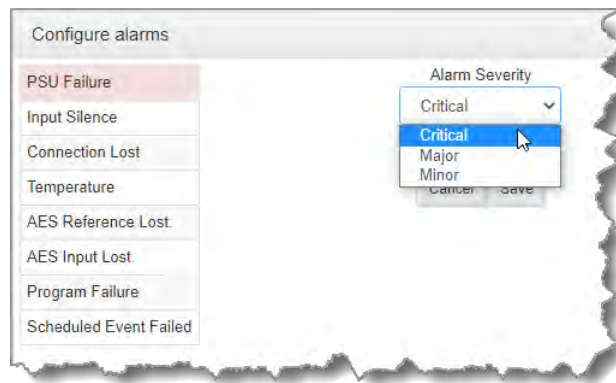
5. Click **Save** to store the new settings.

Note: The following **System** and **Audio** alarms are available:

Alarm	Alarm Type	Explanation
PSU Failure	System	Raises an alarm if one or both PSUs fail
Input Silence	Audio	Raises an alarm if input audio is lost based on whether silence is detected on a single input, or pair of inputs (according to preconfigured silence detection threshold parameters)
Connection Lost	Audio	Triggers and alarm whenever a streaming connection is lost
Temperature	System	Raises an alarm if the temperature is too low or too high
AES Reference Lost	Audio	Raises an alarm if the AES reference clock signal is lost (not available in WheatNet-IP capable codecs)
AES Input Lost	Audio	Raises an alarm if the AES input signal is lost (not available in WheatNet-IP capable codecs)
Program Failure	Audio	An element within a configured program has failed and has not been terminated manually. Note: the alarm is cleared if the program is unloaded.
Scheduled Event Failed	Audio	A scheduled event could not be started or stopped at the specified time. Note: the alarm is only present for the duration of the event.

Configuring Alarm Dissemination Severity Alerts

Codec alarms can be configured for three different severity levels:

1. Click to select an alarm from those displayed in the **Configure alarms** panel.

2. Click **Edit** to configure alarm settings.
3. Click the **Alarm Severity** drop-down menu and select the preferred severity level.
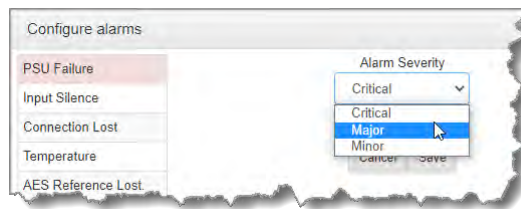
4. Click **Save** to store the new settings for the selected alarm.

Configuring Alarm Dissemination Severity Alerts

Alerts for each alarm severity level are configured using the **Alarm Dissemination** panel.

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Alarm Dissemination** to open the panel.
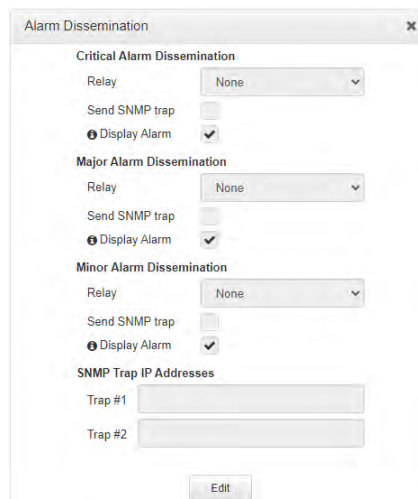
2. Click **Edit** to configure notification settings.
3. Select and configure relay, SNMP trap and alarm display settings for each **Alarm Severity** level. Enter **SNMP Trap IP Addresses** as required at the bottom of the panel.
4. Click **Save** to store the new settings.

Important Note: Simple Network Management Protocol (SNMP) is a protocol used to manage devices on IP networks. SNMP provides the ability to send traps (notifications or alerts), which are packets containing data relating to a system component that may be either statistic or status related. See Configuring SNMP in the Codec for more info, or ask your system administrator.

Configuring Input Silence Detection Parameters

When configuring an **Input Silence** alarm it is also necessary to configure the audio silence thresholds and timeout duration.

1. Click **Input Silence** to select the alarm.

2. Click **Edit** to configure alarm settings.
3. Configure the dBFS threshold and timeout duration in seconds and ensure the input **Enabled** check-boxes are selected. An alarm will be raised when these thresholds are breached.

4. Click **Save** to store the new input silence alarm settings.

23.34.1 Managing Alarms

Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Current Alarms** to view active alarms.

Viewing Current Alarms

Active alarms are indicated by:

1. The red Alarm Symbol flashing in the toolbar of the HTML5 Toolbox Web-GUI screen.

2. All new alarms being listed in the **Current Alarms panel**.
3. Other alerts as per **Alarm Dissemination panel** settings.
4. The codec front panel **ALARM LED** flashing red.

> **Important Note:** When a connection is active the front panel **CONNECTED LED** is illuminated solid green. Illumination will ceases if a connection is lost.

Acknowledging Alarms

To acknowledge an alarm in the **Current Alarms panel**:

1. Click to select the alarm in the **Current Alarms** panel.

2. Click **Acknowledge selected alarm**.

After acknowledging the alarm:

1. The **State** will change from **Active** to **Acknowledged**.
2. The red **Alarm Symbol** will stop flashing but remain visible in the toolbar of the HTML5 Toolbox Web-GUI screen.
3. The codec front panel **ALARM LED** will stop flashing and illuminate solid red.
4. The state of other alerts may change, as per **Alarm Dissemination panel** settings.

Alarm State	Front Panel Alarm LED	Web-GUI Alarm Symbol
Active	Flashing red	Flashing
Acknowledged	Solid red	Stops flashing, remains solid red

Deactivating Alarms

An alarm is deactivated automatically when the alarm state is reversed. E.g. if power is restored after a **PSU Failure** alarm, or if audio is restored after an **Input Silence** alarm.

Deactivating Input Silence Alarms

An **Input Silence** alarm is activated when the configured audio and duration thresholds have been breached. To recover from this alarm state the codec must detect input audio higher than the failure threshold. When audio at this level is detected, the codec monitors input audio to ensure it doesn't drop below the recovery threshold setting more than 5 times within the nominated **Input Silence** duration time. The alarm is then deactivated automatically.

Alarm History

1. Open the HTML5 Toolbox Web-GUI and click **Alarms**, then click **Alarm History** to display a record of all system alarms which have been raised.

Alarm History					✕
Drag a column header and drop it here to group by that column					
Severity	Time raised	Time cleared	Type	Asset	Description
Critical	Mon, 30 Apr 2018 20:35:23 GMT	Mon, 30 Apr 2018 23:50:54 GMT	PSU Failure	Power Supply: 2	Power supply failure
Major	Mon, 30 Apr 2018 23:38:00 GMT	Mon, 30 Apr 2018 23:50:54 GMT	AES Input Lost	AES Inputs: 1,2	AES input lost
Major	Tue, 01 May 2018 00:12:06 GMT	Tue, 01 May 2018 00:12:15 GMT	AES Input Lost	AES Inputs: 1,2	AES input lost

Purge alarm history

Click the **Purge alarm history** button to clear all alarms from the **Alarm History** panel.

23.35   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 IP and ISDN. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP	ISDN
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 (e.g. Genie, Merlin, ViA) and G6 (e.g. Gateway) codecs employing relay reflection minimizes latency • Tieline Music and MusicPlus 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, enable auxiliary data using the **Setup** menu (see Enabling RS232 Data).

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 use 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.
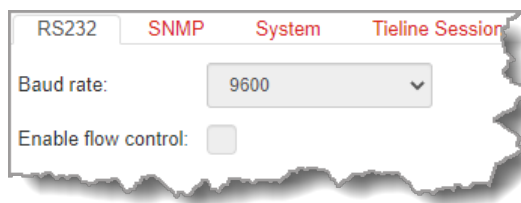
Important Notes:
- When connecting to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms.
- Use firmware higher than 2.8.xx in the Bridge-IT, Genie and Merlin families of codecs to enable auxiliary data over multicast connections.
- It is important to enable serial port flow 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 there may be 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.
- RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast or multicast connection if a codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

23.36   Creating Rules

Product	Feature	Notification
Gateway 4	Control Port GPIOs	Supports 8 control port inputs and 8 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connectors
	Software LIOs	3 virtual inputs, 56 logical outputs, and 64 WheatNet Logic Inputs/Outputs
Gateway 8/16	Control Port GPIOs	Supports 16 control port inputs and 16 opto-isolated outputs via dual DB25 **CONTROL PORT I/O** connectors
	Software LIOs	3 virtual inputs, 48 logical outputs, and 64 WheatNet Logic Inputs/Outputs

Codec 'rules' configure events based on specific codec actions. A range of default rules are preprogrammed into 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 logic IO, or a codec program being connected or disconnected. The Gateway 4 codec has 8 hardware GPIOs and 56 logical outputs, and the Gateway 8/16 has 16 hardware GPIOs and 48 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Inputs/Outputs. WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.

Gateway 4 GPIOs

GPI	Gateway 4 Input	GPO	Gateway 4 Output
8 (1-8)	Physical hardware inputs	8 (1-8)	Physical hardware outputs
3 (9-11)	Virtual input		
		56 (9-64)	Logical outputs
64	WNET Logic input	64	WNET Logic output

Gateway 8/16 GPIOs

GPI	Gateway 8/16 Input	GPO	Gateway 8/16 Output
16 (1-16)	Physical hardware inputs	16 (1-16)	Physical hardware outputs
3 (17-19)	Virtual input		
		48 (17-64)	Logical outputs
64	WNET Logic input	64	WNET Logic output

Note: WheatNet logic inputs are only available if the codec has a WheatNet-IP card installed.

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.
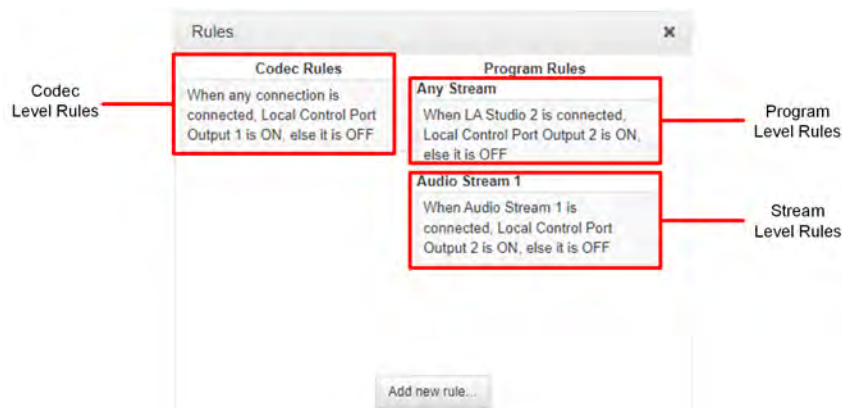
Enabling Data

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. For more information please see Enabling Relays & RS232 Data.

> **Important Notes for Rules:**
> - 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.

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.
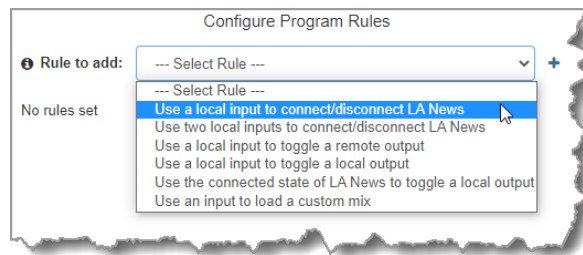
Note: Rules intended to activate dialing will not be valid in **Answer only** programs or audio streams.

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.
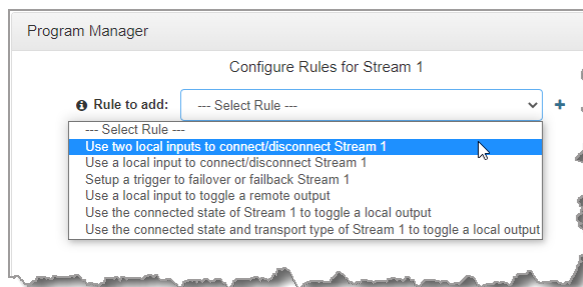
Note: A subset of filtered rules will be displayed for an **Answer only** audio stream connections.

Configuring Rules with the Rules Panel

Use the **Rules panel** to configure codec level rules related to programs and/or hardware and software I/O states.

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.
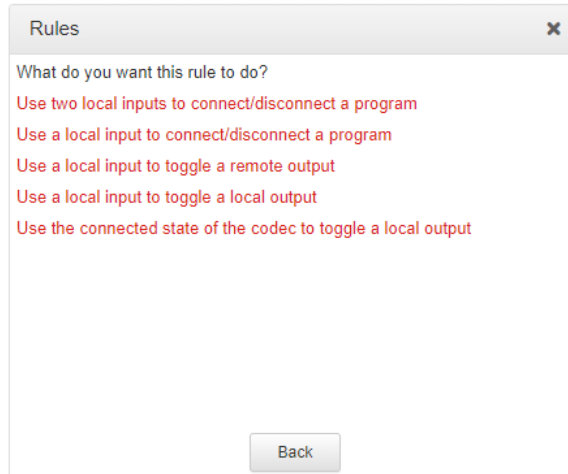
Note: When rules have been configured previously they are displayed when the **Rules panel** is opened.

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.
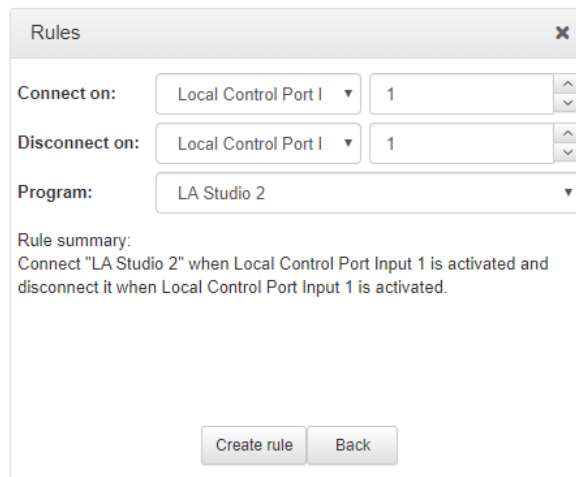
4. Check the **Rule summary** and click **Create Rule** to save the settings.

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.
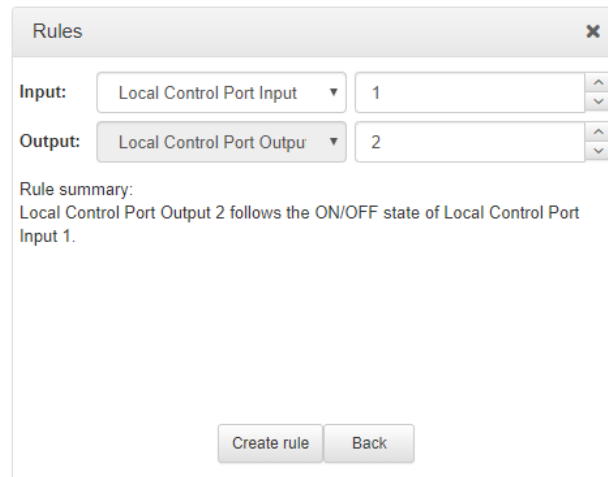
3. Check the **Rule summary** and click **Create Rule** to save the settings.

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.
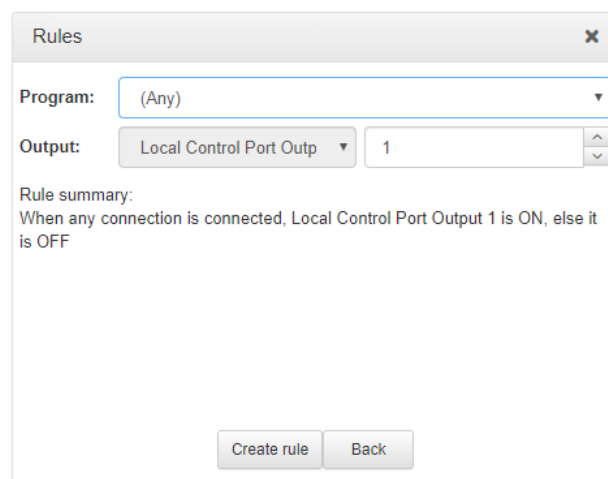
3. Check the **Rule summary** and click **Create Rule** to save the settings.

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 fourth 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 Toolbox HTML5 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.
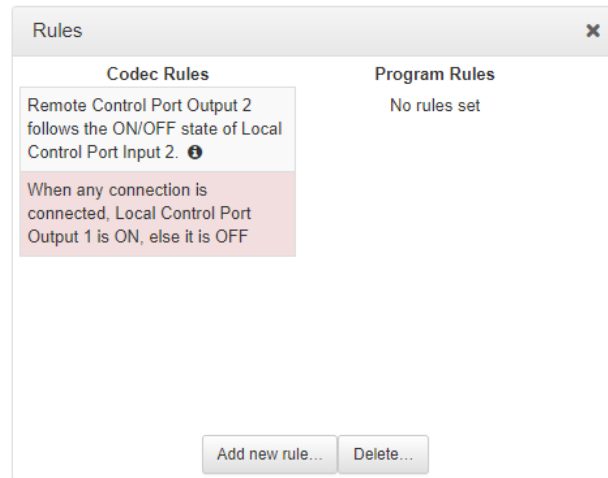
4. Click **Yes** in the confirmation dialog.

23.36.1 Rule Examples

Load a New Matrix on a Remote Codec

Rules can be configured to load different custom matrices in a codec on demand. There is also a solution to allow a studio codec to activate a remote codec matrix change:

1. On the studio codec, create a program and configure a new rule with **Use a local input to toggle a remote output** and then configure a local input (e.g. **Local Control Port Input 1**) to activate a remote output (e.g. **Remote Output 2**).
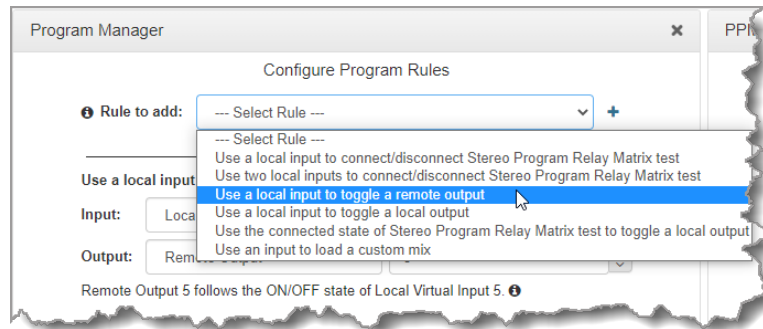
2. On the remote codec, configure a new rule with **Use an input to load a custom mix** and for the **Input** select **Remote Input** and the same **Remote Output** number as configured previously on the studio codec rule (e.g. number 2 in this example).
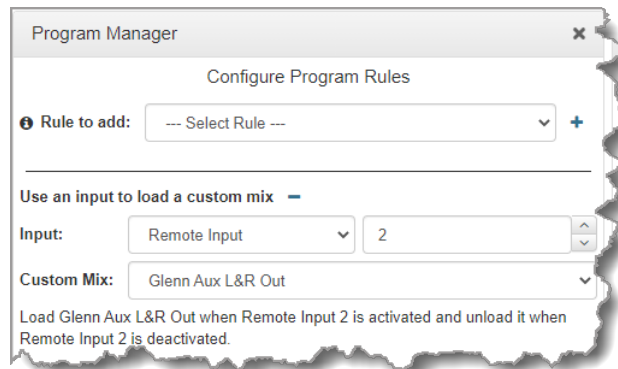
It may seem a bit counter-intuitive to select a **Remote Input** in the rule on the remote codec to perform the matrix change, which is actually referring to the **Remote Output** at the studio codec, but this is correct.

23.37 Monitoring Control Port I/O Status

Product	Feature	Notification
Gateway 4	Control Port GPIOs	Supports 8 control port inputs and 8 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connectors
Gateway 8/16	Control Port GPIOs	Supports 16 control port inputs and 16 opto-isolated outputs via dual DB25 **CONTROL PORT I/O** connectors

Monitor the status of control port inputs and opto-isolated outputs using the **Control Port I/O panel**.

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.
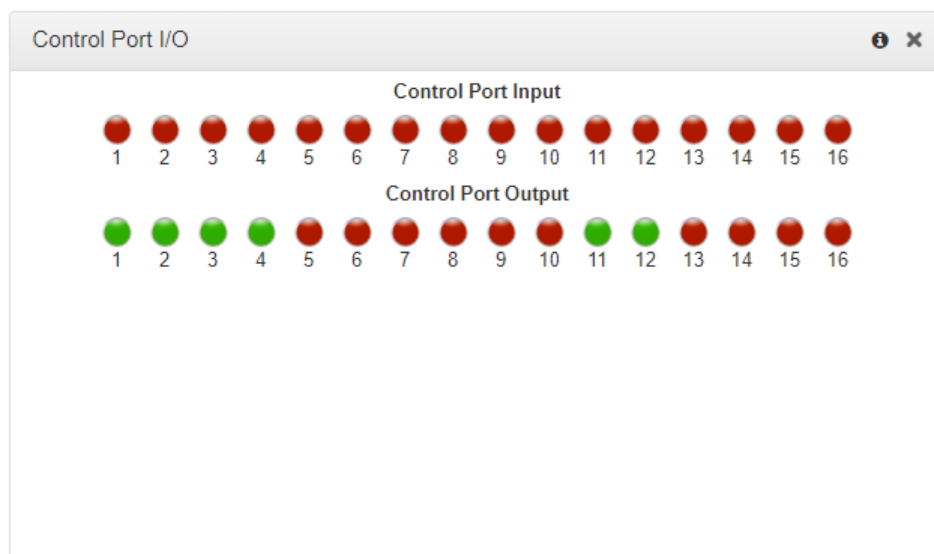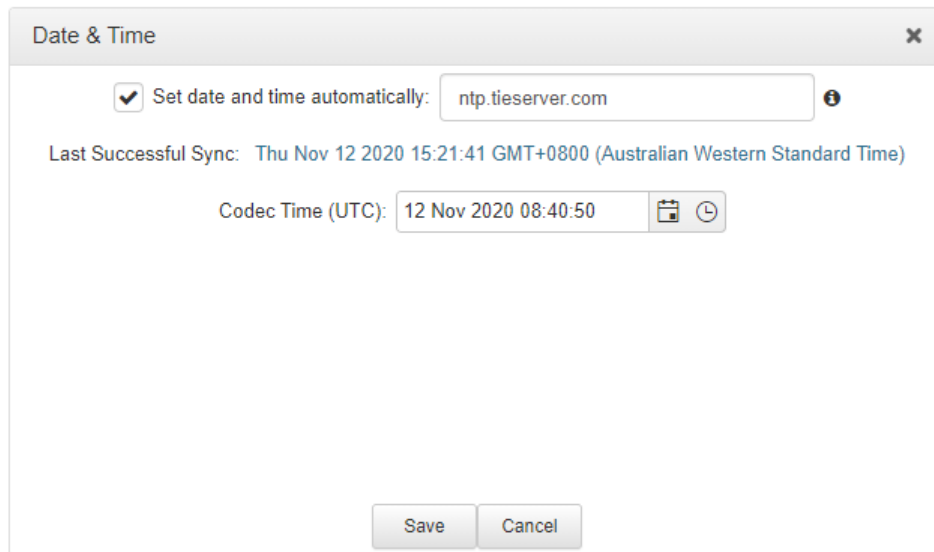
23.38 Adjusting Codec Time and Date

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.

Date & Time	✕
✔ Set date and time automatically: `ntp.tieserver.com` ℹ	
Last Successful Sync: Thu Nov 12 2020 15:21:41 GMT+0800 (Australian Western Standard Time)	
Codec Time (UTC): `12 Nov 2020 08:40:50` 🗓 🕐	
Save Cancel	

ℹ **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 synchronized to the server. This may cause scheduled events to be missed.

23.39   Upgrading Codec Firmware

To download the latest codec firmware visit www.tieline.com. See Upgrading Firmware via USB 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.
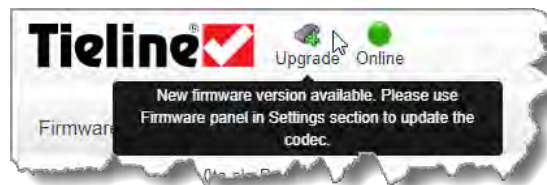
3. Click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel** to perform the firmware upgrade.

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 Toolbox HTML5 Web-GUI and click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel**.
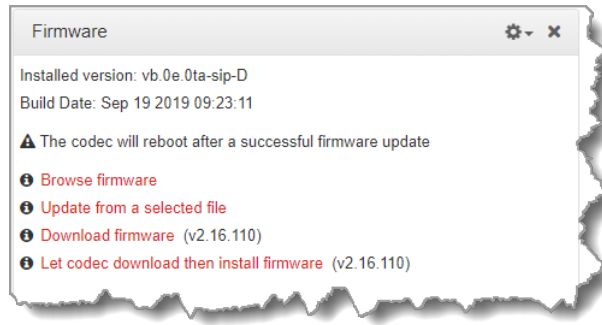
2. Click **Browse firmware** to search for the latest firmware and download it to your computer. Note: the **Download firmware** and **Let codec download then install firmware** links are only visible if firmware more recent than the currently installed version is available.
3. Once the firmware has 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.
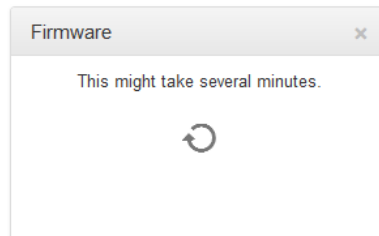
5. We recommend clearing your browser cache after the upgrade is complete. The short cuts for this are:
    *   Google Chrome: shift+Ctrl+delete
    *   Mozilla Firefox: Ctrl+shift+delete
    *   Internet Explorer: Ctrl+shift+delete
    *   Safari: Ctrl+alt+e

23.40 Software License Installation

Product	Feature	Notification
Gateway 4	Licensing	The Gateway 4 codec supports 4 audio channels only and this cannot be upgraded or expanded.
Gateway 8/16	Licensing	The Gateway codec supports between 8 and 16 audio channels. If a codec supports fewer than 16 channels, it can be upgraded to support more channels by purchasing a channel license upgrade from Tieline and installing it in the codec.

The Gateway codec supports between 8 and 16 channels and to view channel **License Status** in a codec open the HTML5 Toolbox web-GUI in a web-browser and select **Settings** and open the **Licensing panel**.
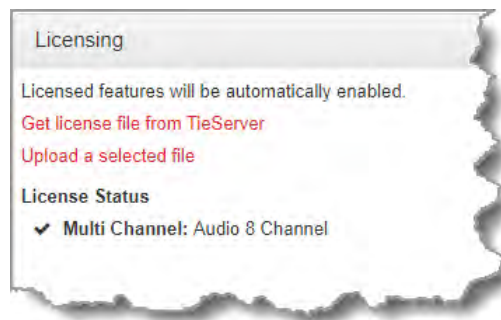
For more details about channel licensing see Channel Licenses.

Perform an Automatic Software License Install with the HTML5 Toolbox Web-GUI

To perform an automatic software license update after purchasing a new license file it is necessary to connect the codec to a PC and ensure it is connected to the internet. Check for a notification by email from Tieline that your new license file is ready to download from TieServer.

1. Open the HTML5 Toolbox web-GUI in a browser on your PC by typing either the IP address of the codec (LAN connection), or the USB address of the codec (USB connection) into the address bar.
2. Ensure you have unloaded any currently loaded program in the codec via the **Program Loader panel**.
3. Click **Settings** in the **Menu Bar**, then click **Licenses** to display the **License Manager**.
4. Click **Get license file from TieServer**.
5. A **Success** dialog in the web-GUI **Licensing panel** confirms when installation is complete.
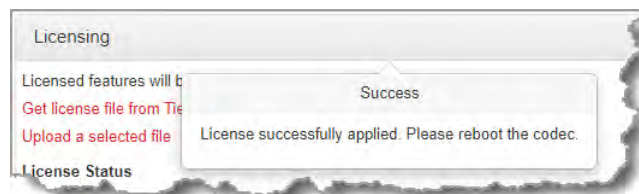
6. The codec screen should also display a confirmation message.
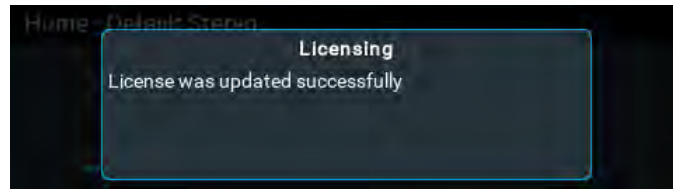
7. Reboot the codec via **Settings > Reset & Backup > Reboot Codec** and press the ⊙K button. Note: do not reboot by removing the power cable from the codec.

Download a License File and Install Manually

If the automatic license update fails, or the codec is not connected to the internet, it is possible to install a previously downloaded license file, or a license file copied to a PC. Save the license to a PC connected to the codec and use the following procedure to install it:

1. Open the HTML5 Toolbox web-GUI in a browser on your PC.
2. Click **Settings** in the **Menu Bar**, then click **Licensing** to display the **Licensing panel**.
3. Click **Upload a selected file**.
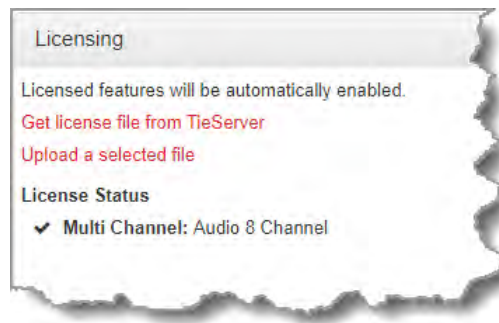
4. Navigate to the ".lcf" license file on your PC, then click the **Open** button to commence license installation.
5. A **Success** dialog in the web-GUI **Licensing panel** confirms installation is complete.
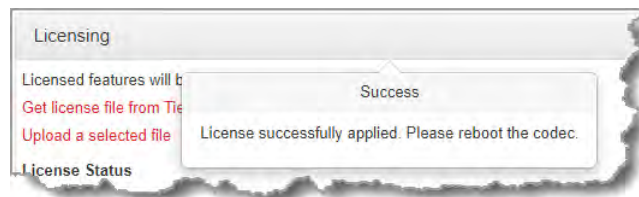
6. Reboot the codec via **Settings > Reset & Backup > Reboot Codec** and press the ⊙K button. Note: do not reboot by removing the power cable from the codec.
7. Clear the web-browser cache and then it may also be necessary to reset factory defaults to display the new channel assignment. To reset factory defaults select **Settings > Reset & Backup > Restore Factory Defaults**. Note: Please take note of network related settings before restoring factory defaults. Do not restore factory defaults in situations where the codec can only be accessed remotely. I.e. it is not possible to access the codec front panel to adjust network settings and restore connectivity.
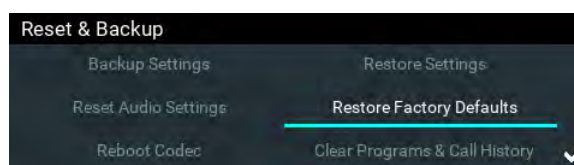
8. Verify that the license settings have been installed in the **Licenses panel**.
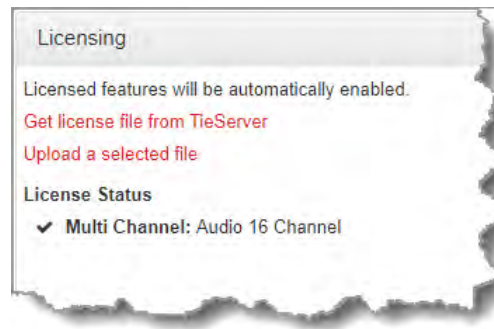
24 Front Panel Configuration Tasks

The following sections explain how to configure codec settings using the front panel **LCD SCREEN** and **KEYPAD**.

24.1 Configuring IP via the Front Panel

Checking IP Address and Unit Details in the Codec

1. Press the **SETTINGS** ⚙ button.
2. Select **Unit Details** and press the 🆗 button.
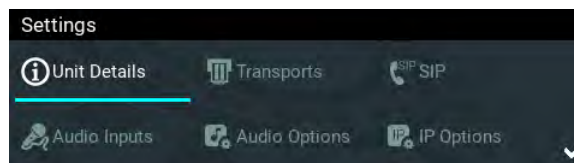
3. IP address information, serial number, and other unit details are listed. Use the arrow up 🔺 and down 🔻 buttons to scroll and view all details listed.
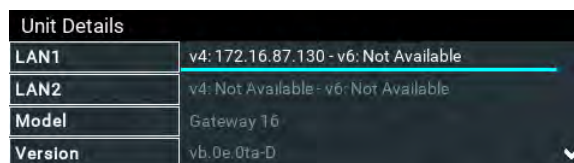
4. Press the 🆗 button for **LAN1** and **LAN2** to display **Network Details** including the **MAC Address**, which is used to add the codec to a domain along with the serial number.
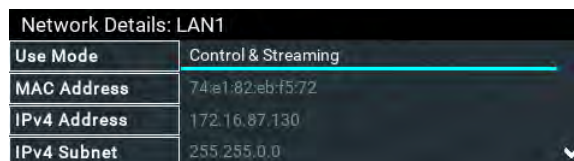
> ℹ️ **Important Note:** For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

Ethernet, Wi-Fi and VLAN Configuration Options

The codec features two physical Ethernet port interfaces and up to four additional VLAN interfaces. The codec also supports a Wi-Fi* connection via the USB port on the front panel.

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, Wi-Fi or VLANs can be used to operate SmartStream PLUS, or to separate codec Control and Streaming functions if required.

A range of options which can be viewed and configured via **Settings > Transports > LAN1 / LAN2**.
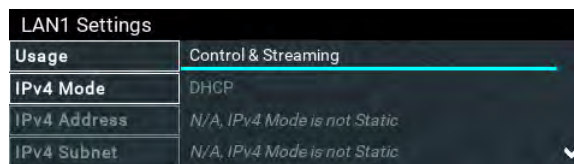
After completing configuration press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings..
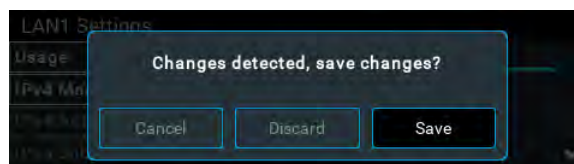
* WiFi modem support not included in the first firmware release.

Configure an IPv4 DHCP Address

By default the codec is programmed for DHCP-assigned IP addresses. DHCP IP addresses are automatically assigned and can change each time you connect to an Internet Service Provider or a router on a local area network (LAN).

1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.
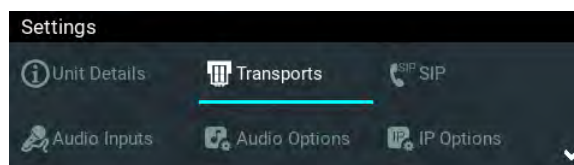
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Configuration** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the button.
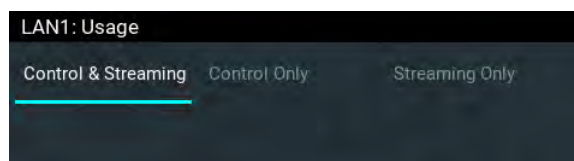
5. Select **IPv4 Mode** and press the OK button.
6. Select **DHCP** and press the OK button.
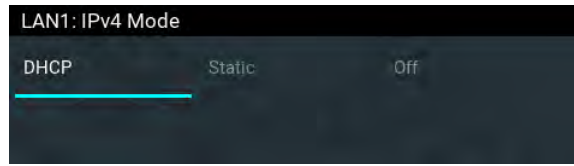
7. Press the **Return** button, then select **Save** in the confirmation dialog and press the OK button to confirm the new settings.

Configure a Static IPv4 Address

Static IP addresses are fixed addresses which are 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 **SETTINGS** button.
2. Select **Transports** and press the OK button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Configuration** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the OK button.
5. Select **IPv4 Mode** and press the OK button.
6. Select **Static** and press the OK button.
7. Navigate to **IPv4 Address** and enter the IP address (using the keypad or onscreen keyboard), then press the OK button (or select **Enter** using the onscreen keyboard).
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the OK button.
9. Navigate to **IPv4 Gateway** and enter the Gateway details, then press the OK button.

10. Press the **Return** button, then select **Save** in the confirmation dialog and press the OK button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

IPv6 Address Assignment

There are three IPv6 settings available for each Ethernet port, a Wi-Fi connection via the **USB PORT** on the codec and any VLANs which are configured.

1. Auto: 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.

To adjust this setting:

1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Configuration** and then **IPv6 Mode** and press the button.
5. Select **Auto**, **Manual** or **Off** and press the button.
6. Press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings.

By default the codec is configured to allow the codec to automatically receive IPv6 address information from an IPv6 enabled router.

Manual IPv6 Address Assignment

Select **Manual** mode using the previous procedure and enter information into the **IPv6 Address, IPv6 Prefix Size** and **IPv6 Gateway** fields in the codec to manually configure address details.

DNS Server

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 LAN codec menu.

1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface, then press the button.
4. Select **Configuration** and press the button.
5. Use the down navigation button to scroll to **Set DNS**.
6. Press the button to toggle between **Auto** and **Manual**.
7. Enter DNS Address and Domain details as required.
8. Press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings.

Link Mode Configuration

It is possible to configure the Ethernet link speed (10/100/1000/Auto) and whether each interface will operate in Full-Duplex or Half-Duplex modes.

1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface, then press the button.
4. Select **Configuration** and press the button.
5. Use the down navigation button to scroll to **Link Mode**.
6. Press the button to select a preferred setting. Note: Default setting is **Auto**.
7. Press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings.

VLAN ID (VLAN configuration only)

The **VLAN ID** is encapsulated in IP packets to facilitate routing throughout your network.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Transports** and press the button.
3. Use the down navigation button to select a **VLAN** interface and press the button.
4. Navigate to **Configuration** and press the button.
5. Select **Usage** and press the button.
6. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
7. Use the down navigation button to scroll to **VLAN ID**.
8. Press the button to enter a number between 1-4094 inclusive.
9. Press the button to confirm this setting.
10. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

VLAN Priority (VLAN configuration only)

VLAN Priority is a prioritization scheme for forwarding data packets throughout Virtual LANs.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Transports** and press the button.
3. Use the down navigation button to select a **VLAN** interface and press the button.
4. Navigate to **Configuration** and press the button.
5. Select **Usage** and press the button.
6. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
7. Use the down navigation button to scroll to **VLAN Priority**.
8. Press the button to enter a number from 0 to 7 inclusive.
9. Press the button to confirm this setting.
10. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

VLAN Interface (VLAN configuration only)

This setting applies the VLAN settings to a physical Ethernet port in the codec.

1. Press the **SETTINGS** button.
2. Use the navigation buttons on the front panel to select **Transports** and press the button.
3. Use the down navigation button to select a **VLAN** interface and press the button.
4. Navigate to **Configuration** and press the button
5. Select **Usage** and press the button.
6. Select the mode of operation for this VLAN (e.g. Control & Streaming, Streaming only, Control Only) and press the button.
7. Use the down navigation button to scroll to **Interface**.
8. Press the button to select **LAN1** or **LAN2**, then press the button.
9. Press the **Return** button, then select **Yes** in the confirmation dialog and press the button to confirm the new settings.

24.2   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 **SETTINGS** 🔧 button.
2. Use the navigation buttons to select **IP Options** and press the 🆗 button.
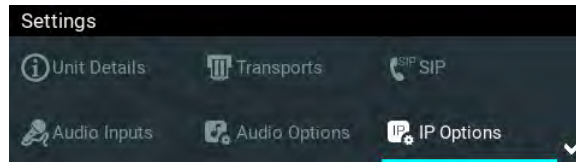
3. Navigate to **Hostname** and press the 🆗 button to enable this feature.
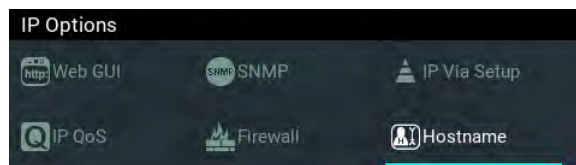
4. Select **Hostname Enable** and press the 🆗 button to enable this feature. Next, navigate to **Hostname String** and press the 🆗 button to display the **Set Hostname** screen keyboard and then enter the hostname. Next, navigate to **Enter** and press the 🆗 button to save the settings. Note: Press the **Return** 🔙 button to exit the screen without saving changes.
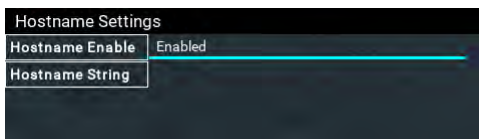
5. The new hostname will be displayed.
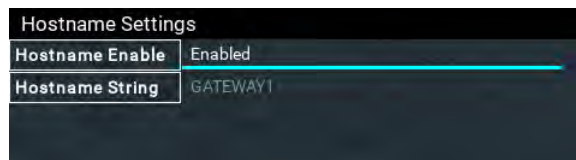
> ℹ️ **Important Note:**
> • Modifying hostname settings requires a codec restart before they take effect
> • In the **Hostname** only enter the characters a-z, A-Z, 0-9 and - and the first or last character cannot be a hyphen/dash.

24.3   IP Via Setting

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. **Wi-Fi** (default **Tertiary** Via interface)

> **Important Notes:**
> * If an interface is not available it is not listed in the **Via** interface selection screen. E.g. if Wi-Fi is not configured it will be unavailable.
> * 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** (Ethernet 1), **Secondary** (Ethernet 2) 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 an Ethernet port and for another it may be a Wi-Fi interface. This allows you to configure site-specific settings to suit available network interfaces at different remote locations.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the OK button.
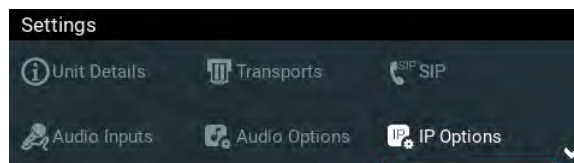
3. Navigate to **IP Via Setup** and press the OK button.
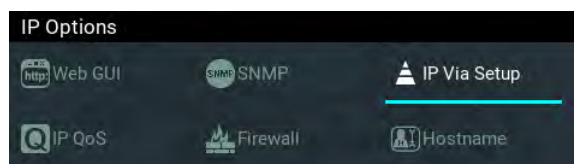
4. Navigate to the **Primary**, **Secondary** or **Tertiary** IP Via interface setting and press the OK button.
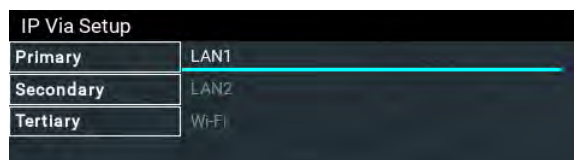
5. Select an alternative default interface and press the OK button.
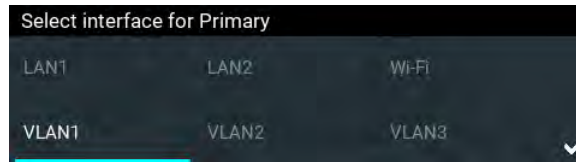
6. Press the **Return** ⬑ button to navigate out of the menu, then navigate to **Save** in the confirmation dialog and press the ⓞⓚ button to save all changes.
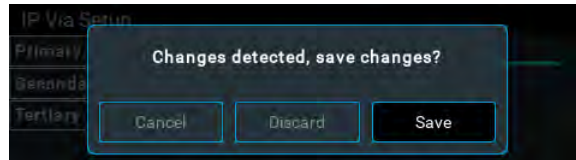
ⓘ **Important Note:** Fuse-IP cannot be configured as a default Primary, Secondary or Tertiary interface.

24.4   Configure Audio Options

A range of input, output, PPM, synchronization, metering and reference options are configurable within the **Audio Options** menu.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the button.
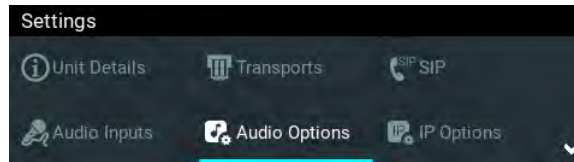
PPM Mode

PPM metering can be adjusted to display inputs, outputs, encoders and decoders and can be adjusted via **Settings > Audio Options > PPM Mode**. Note: Gateway 4 has 4 inputs, 4 encoders, 4 decoders, and 8 PPMs only; Gateway has 16 PPM meters and between 8 and 16 inputs depending on configuration purchased; encoders and decoders available will match input availability/configuration. See Input Configuration, Levels and PPMs for more details.
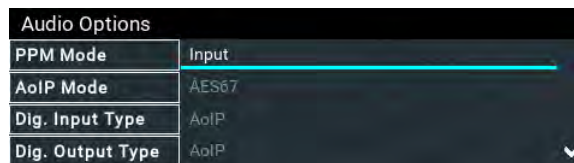
AoIP Mode

All codecs support AES67 AoIP streaming. This setting selects whether the codec will stream using either AES67 or WheatNet-IP when an optional WheatNet-IP card is installed.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the button.
3. Select **AoIP Mode**.
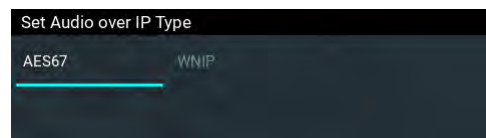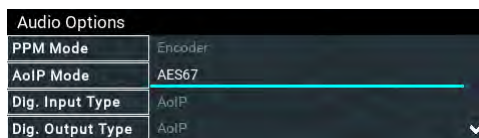
Digital Input Type

The codec supports configuration of all inputs as digital AES3, or digital AoIP inputs.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the button.
3. Select **Dig. Input Type**.
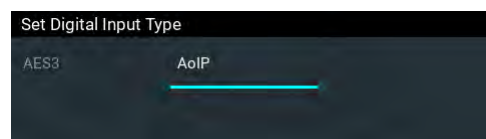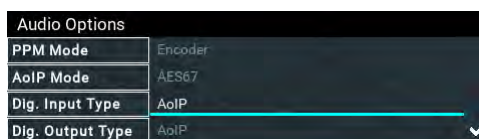
Digital Output Type

The codec supports configuration of all outputs as digital AES3, or digital AoIP outputs.

1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the button.
3. Select **Dig. Output Type**.
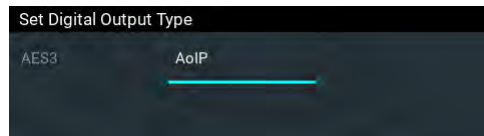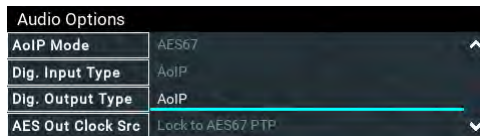
Audio Options	
AoIP Mode	AES67
Dig. Input Type	AoIP
Dig. Output Type	AoIP
AES Out Clock Src	Lock to AES67 PTP

Set Digital Output Type	
AES3	AoIP

AES3 Out Clock Source

The codec contains two sample rate converters.

Input Sample Rate Converter

The codec implements an Asynchronous Sample Rate Converter (ASRC) to convert the sample rate of an AES3 input to the sample rate set in the codec. The codec sample rate is determined by the selected algorithm. For example, if you select the Music algorithm, the sample rate will be set to 32kHz. By default the codec will up-sample all AES3 input sources to 96kHz sampling and then convert to match the AES output sample rate setting.

> **Important Note:** All AES3 inputs must have the same sample-rates and must be synchronized to a common clock. See Appendix B for pin-outs of AES3 inputs.

Output Sample Rate Converter

The sample rate of the AES3 output is configured using the clock source setting via the **SETTINGS** button and then **Audio Options > AES3 Out Clock Src**. This configures the sample rate frequency of all AES3 output signals.

Wordclock Sync In

This setting configures the codec for a word clock source via the **SYNC I/O 2** BNC connector on the codec rear panel (this is the same as the **External Word Clock** setting in Tieline G3 codecs). Often this will be a studio reference signal. In television broadcasting facilities the audio reference signal should be locked to the video reference if there is one available. The sample rate being received is recognized by the codec and automatically adjusted. Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Lock to AES3 Input

With this setting the codec uses AES3 input sync information to set the codec output sample rate (Note: this is the same as the AES Rx Clock setting in Tieline G3 codecs). The codec initially tries to use the signal on AES inputs 1 and 2 as the clock to which the AES outputs are synchronized. If unavailable, it then attempt to use inputs 3 and 4, or inputs 5 and 6 and so on in that order. If you select this option, all AES inputs must always be synchronized to the same clock source, e.g. if AES3 inputs 1 and 2 use 48kHz sampling then all other inputs must also be synchronized to the same clock. Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Lock to AES11 Input

AES11 is a standard AES3 signal with an accurate clock reference (DARS, or Digital Audio Reference Signal), usually without the audio data. To use this input attach a female BNC connector to the SYNC I/O 1 connector on the codec rear panel. Supported sample rates include 32 kHz, 44.1 kHz, 48 kHz and 96 kHz. Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Fixed Sample Clock

Select from a range of fixed output sample rates including:

1. 32 kHz
2. 44.1 kHz
3. 48 kHz
4. 88.2 kHz
5. 96 kHz
6. 192 kHz

Note: The reference clock must be within +/- 50ppm of the listed sample rates.

Reference Level

The default Tieline G6 audio reference scale displayed on the PPMs when connecting to a Tieline G6 codec is -40dBFS to 0dBFS. Using this reference scale audio peaks can safely reach 0dBFS without clipping, providing 20dB of headroom from the nominal 0vu point. The comparison table below outlines the reference scales for G6, G5, G3 codecs and Report-IT in dBFS, as well as the equivalent dBU scale.

	Reference Level	Description	dBu	dBFS
1	Tieline G6 (Gateway)	PPM meter low point	-16dBU	-40dBFS
		Nominal 0vu reference level	+4dBU	-20dBFS
		Level at which audio will clip/distort	+24dBu	0dBFS
2	Tieline G5 (Genie, Merlin, Bridge-IT)	PPM meter low point	-16dBu	-38dBFS
		Nominal 0vu reference level	+4dBu	-18dBFS
		Level at which audio will clip/distort	+22dBu	0dBFS
3	Tieline G3 (Commander and i-Mix)	PPM meter low point	-11dBu	-29dBFS
		Nominal 0vu reference level	+4dBu	-14dBFS
		Level at which audio will clip/distort	+18dBu	0dBFS
4	Report-IT	PPM meter low point	-9dBu	-23dBFS
		Nominal 0vu reference level	+4dBu	-10dBFS
		Level at which audio will clip/distort	+14dBu	0dBFS

The default setting in the codec is **Auto**. To reconfigure this:

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **Audio Options** and press the 🆗 button.
3. Select **Reference Level**.

ℹ️ **Important Note:** If a codec supports multiple stream programs and the **Auto** (default) reference level is selected, the first codec to connect will configure the reference level used for all subsequent connections. I.e. If a G5 codec connects first then the G5 Audio Reference Level will be configured for all connections.

Analog PPM Units

It is possible to switch the analog input PPM meter unit of measurement from dBFS (default) to dBU:

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **Audio Options** and press the OK button.
3. Select **Analog PPM Units**.

Loopback Test

Test mode is used by the codec to perform an input/output loopback test of audio. E.g. Input 1 is routed to Output 1, Input 2 is routed to Output 2 etc.

1. Press the **SETTINGS** 🔧 button.
2. Navigate to **Audio Options** and press the OK button.
3. Select **Start Loopback** and press the OK button for **Loopback Test**.

Audio Options	
AES Out Clock Src	Lock to AES67 PTP
Reference Level	Auto
Analog PPM Units	dBFS
Loopback Test	Start Loopback
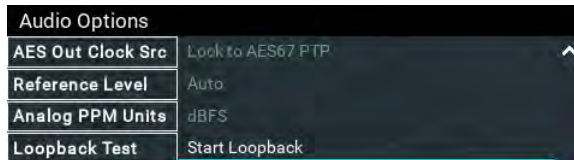
4. Press OK or the **Return** ↩ button to end the loopback test.

Audio Loopback Test

Press OK or Back to stop audio loopback test
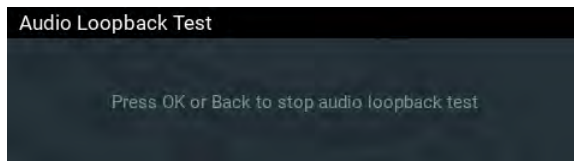
24.5   Enabling the Cloud Codec Controller

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. Press the **SETTINGS** button.
2. Navigate to **IP Options** and press the OK button.
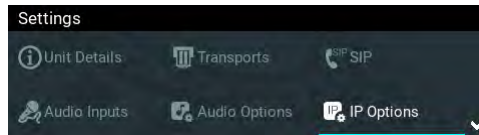
3. Select **Web-GUI** and press the OK button.
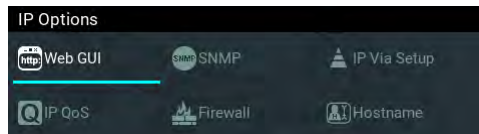
4. Navigate down to **CC Controller** and press the OK button to toggle between **Enabled** and **Disabled**.
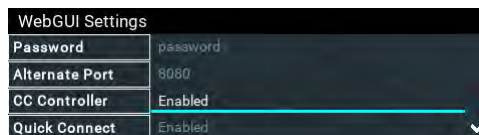
> **Important Notes:**
> - Ensure **CSRF Protection** is disabled in the codec or it will not be able to connect to the CCC. This setting is **[Disabled]** by default and is also available in the codec menu via **Settings > WebGUI**, and in the **Options** panel in the HTML5 Toolbox Web-GUI.
> - Locally 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 data.

24.6   Inverting Input Polarity

The codec supports inverting input polarity.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Audio Inputs** and press the OK button.
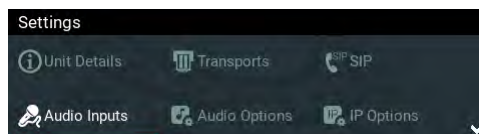
3. Navigate to the input for which polarity is to be inverted and press the OK button.
4. Navigate down to **Polarity Inv** and press the OK button to toggle between **Enabled** and **Disabled**.
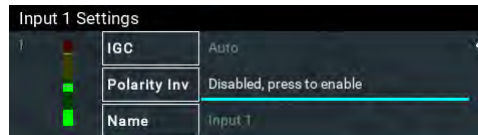
24.7 Connecting with Fuse-IP Bonding

Fuse-IP is a proprietary Tieline IP bonding technology which aggregates 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:

- Dual Ethernet LAN ports.
- An Ethernet LAN port and a Wi-Fi* dongle. (*Not in first release)

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 **Via** interface you can use to dial, similar to selecting a LAN interface.
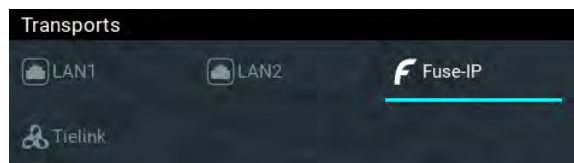
Fuse-IP requires one codec to be a server and the other a client. Normally the remote codec is configured as the client and the studio codec is the server, because it's easier to dial static IP addresses configured at the studio than cellular, Wi-Fi or LAN interfaces at the remote site. Like SmartStream PLUS redundant streaming, two IP interfaces can be used at the studio for additional redundancy.

Prerequisites

Version 2.16.xx firmware is required to use Fuse-IP. 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 a Fuse-IP Server at the Studio

1. Press the **SETTINGS** button.
2. Select **Transports** and press the OK button.
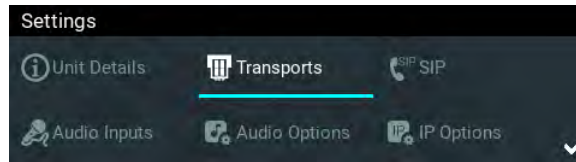
3. Select **Fuse-IP** and press the ⓞⓚ button.
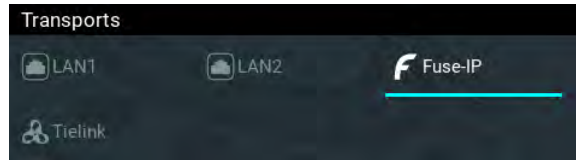
ⓘ **Important Note:** Ensure Fuse-IP is disabled prior to configuration. Navigate to **Fuse-IP Status** and disable Fuse-IP if it is enabled.
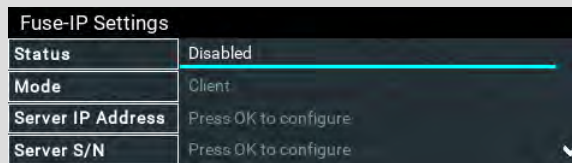

4. Navigate down to **Mode** and press the ⓞⓚ button to select **Server** if the codec is at the studio.
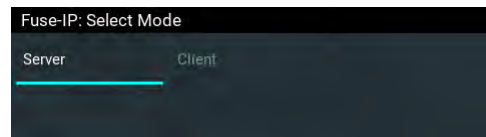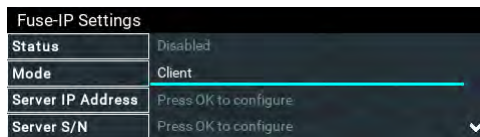
Note: the server codec serial number will be displayed and needs to be entered into the Fuse-IP client codec.
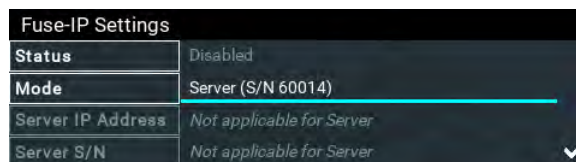
5. Navigate to **Bonded Interfaces** and press the ⓞⓚ button, then navigate to each interface in turn and press the ⓞⓚ button to select or deselect interfaces. Confirm the changes when exiting the menu.

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

Fuse-IP Settings	
Server S/N	*Not applicable for Server*
Bonded Interfaces	LAN1,LAN2
Port	8999
Inactivity Timeout	*Not applicable for Server*
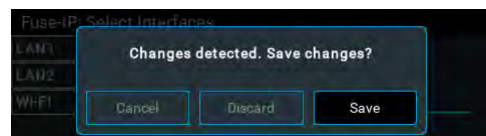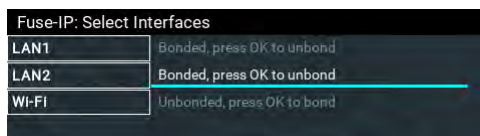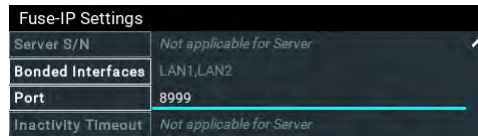
7. The codec is now configured to connect with a client codec over a **Fuse-IP** tunnel.

Configuring a Fuse-IP Remote Client

1. Press the **SETTINGS** button.
2. Select **Transports** and press the OK button.

Settings		
(i) Unit Details	Transports	SIP SIP
Audio Inputs	Audio Options	IP Options

3. Select **Fuse-IP** and press the OK button.

Transports		
LAN1	LAN2	F Fuse-IP
Tielink		

> (i) **Important Note:** Ensure Fuse-IP is disabled prior to configuration. Navigate to **Fuse-IP Status** and disable Fuse-IP if it is enabled.

Fuse-IP Settings	
Status	Disabled
Mode	Client
Server IP Address	Press OK to configure
Server S/N	Press OK to configure

4. Navigate to **Mode** and press the OK button to select **Client** if it is the remote codec.

Fuse-IP Settings	
Status	Disabled
Mode	Client
Server IP Address	Press OK to configure
Server S/N	Press OK to configure
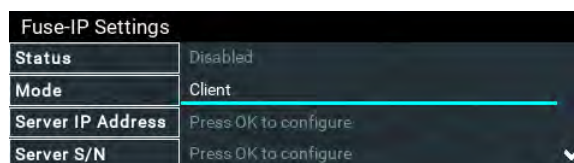
5. Navigate to **Server IP Address** and press the OK button to enter a public static IP address associated with the bonded interfaces at the studio, then press the OK button. Note: if the bonded interfaces have private addresses behind a firewall then port forwarding needs to be configured. See [Installing the Codec at the Studio](#) for more details on port forwarding.

Fuse-IP Settings	
Status	Disabled
Mode	Client
Server IP Address	203.38.199.186
Server S/N	Press OK to configure

6. Navigate to **Server S/N** and press the 🆗 button to enter the serial number of the server codec to which you are connecting (or select a previously entered serial number via History), then press the 🆗 button.
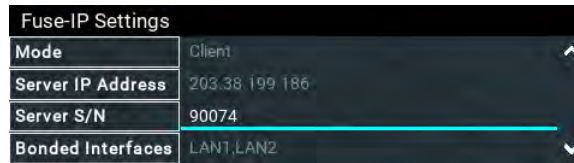
7. Navigate to **Bonded Interfaces** and press the 🆗 button, then navigate to each interface in turn and press the 🆗 button to select or deselect interfaces. Confirm the changes when exiting the menu.
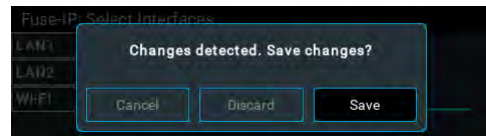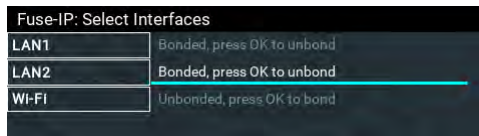
5. Leave the default **Fuse-IP Port** as **8999** in most situations unless this port is already in use, e.g. there are multiple codecs behind a firewall using Fuse-IP, therefore it is necessary to allocate a different port for each Fuse-IP tunnel. Note: the port number on the client and server codecs must be the same.

6. Navigate to **Inactivity Timeout** and press the 🆗 button if you want to adjust the predetermined time period for turning the Fuse-IP tunnel off. Adjust the setting and press the 🆗 button to store the new setting. Note: **Inactivity Timeout** can be configured from 0 to 1440 minutes. Enter 0 to disable the timeout.

9. Navigate up to **Status** and then press the 🆗 button to create a Fuse-IP tunnel between the server and client codecs.
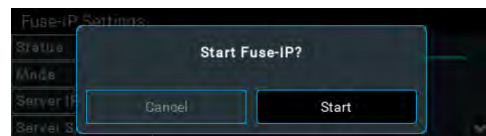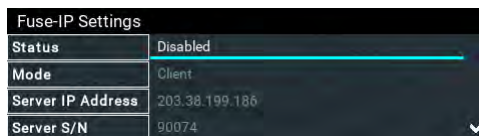
10. **Connected** is displayed when a connection is successful. Remember Fuse-IP must be enabled on both codecs.

Please note: double-check all settings on both the server and client codecs if the message **Started, dialing server** persists after starting Fuse-IP.

11. Select **Fuse-IP** as the **Via** interface with which to connect when creating a program using the front panel codec menus, or the HTML5 Toolbox Web-GUI **Program Manager panel**.

> **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, 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.

24.8   Selecting an Algorithm

The codec offers PCM uncompressed linear 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. There is a range of pre-programmed connection profiles to simplify codec configuration.

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-ELDv 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.

aptX® Enhanced is supported over ISDN at the following sample and bit rates:

Encoding	Bit rate Required	B Channels Required
aptX® Enhanced Mono 16 bit, 32 kHz	128 kbps	2

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)

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 creating multi-unicast connections.
- 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 connection 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 (16bit;32kHz) to 4.6 Mbps (24bit; 96 kHz)	80kbps	• Full bandwidth, perfect audio quality for voice and music • No error concealment/correction or artifacts	• Extremely high quality PCM linear uncompressed 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	• 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 between 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 when bidirectional communication between 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 where limited connection bandwidth is available • Suitable when bidirectional communication between 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 where high connection bandwidth is available and very low delay is highly desirable. • Resilient with multiple encodes/decodes when 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
TxTran / RxTran					NOT FOR BROADCAST USE	NOT FOR BROADCAST USE

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.

24.9   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.
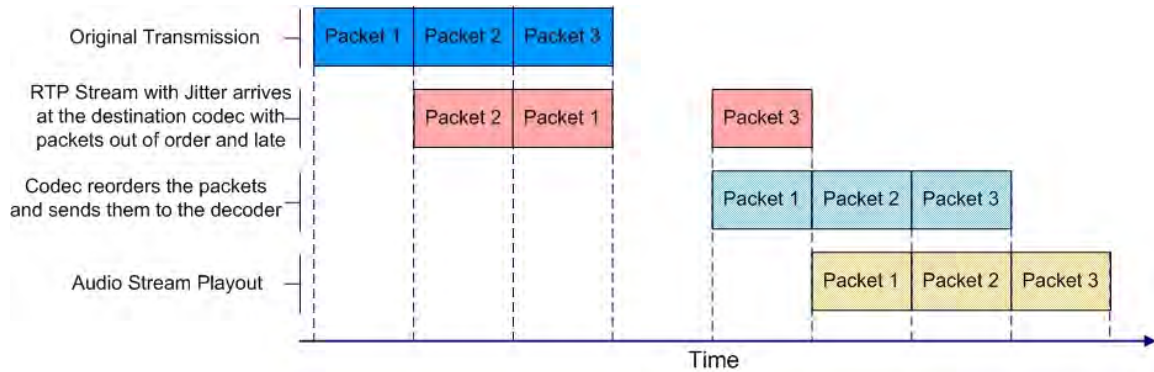
Jitter-buffer management is encompassed within Tieline's SmartStream IP technology which can:

- 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 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 programmed into 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 Support 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	Tieline Session Data Connections	SIP Connections
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	✘	✘

How to get the Best 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 adjusted to suit prevailing 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 statistics can be viewed via **HOME** 🏠 **> Cxns** and use the down ▼ and up ▲ navigation buttons to scroll through connection statistics. The first four stages are observed in "auto" jitter buffer mode.

1. **Stabilization period (a1):** A few seconds during which a stable connection is established.
2. **Stage 2 (a2):** A compatibility check occurs.
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.

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.

If you change the jitter buffer setting in a codec it will only adjust to the new level when link quality is high (e.g. above 70%). This is done to ensure audio quality is not compromised. 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 programming 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 - 999 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.

Important Notes: There is no jitter buffer setting on a multicast server codec because it only sends and never receives audio packets.

24.10   Configuring Forward Error Correction

There are two modes of Forward Error Correction (FEC) available in the codec:

1. Tieline FEC: Transmits a secondary stream of audio data packets over a single connection.
2. RFC 2733 compliant FEC (Sessionless connections only): Transmits audio packets over a separate connection.

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 the codec before dialing. FEC should only be used if link quality displayed on the codec is below **S:99 R:99**, as it is of no benefit otherwise. Tieline and RFC2733 compliant FEC settings are outlined in the following table with their bit rate ratios.

FEC Setting	Bit rate Ratios	Connection Use	Tieline FEC	RFC 2733 FEC
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.	Yes	Yes
50%	Additional data is sent by FEC in a ratio of 2:1.	Recommended for international & national connections	Yes	Yes
33%	Additional data is sent by FEC in a ratio of 3:1.	Recommended for national and local connections.	Yes	Yes
25%	Additional data is sent by FEC in a ratio of 4:1.	Recommended for national and local connections.	No	Yes
20%	Additional data is sent by FEC in a ratio of 5:1.	Recommended for local and LAN connections.	Yes	Yes
10%(Highest delay)	Additional data is sent by FEC in a ratio of 10:1.	Recommended for local and LAN connections.	No	Yes
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	Yes	Yes

		connections for STLs with connections that aren't shared & have quality of service (QoS).		

> **Important Notes:**
> - The **FEC Delay** configured should take into account the packet arrival (jitter buffer) strategy at the remote codec. For example, if the maximum jitter buffer at the remote codec is 1000 ms, the FEC Delay setting should be lower, to ensure there is enough time for FEC packets to arrive and replace lost packets prior to audio playout.
> - By default, the codec will use the audio stream IP address as the remote FEC IP address as well. This can be adjusted in the **Program Manager panel** in the HTML5 Toolbox web-GUI.
> - The default local and remote UDP audio FEC ports are 9002.
> - Any of the available algorithms can be selected when configuring RFC 2733 FEC in the codec.

How does FEC work?

If you enter 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 the maximum 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 maximum upload speed at the remote end too. 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 give you 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 configure different

FEC settings on each codec to match connection bandwidth capabilities 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 and increases the overall bandwidth available for the incoming broadcast signal from the remote site.

24.11   Enabling Relays & RS232 Data

Product	Feature	Notification
Gateway 4	Control Port GPIOs	Supports 8 control port inputs and 8 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connectors
	Software LIOs	3 virtual inputs, 56 logical outputs, and 64 WheatNet Logic Inputs/Outputs
Gateway 8/16	Control Port GPIOs	Supports 16 control port inputs and 16 opto-isolated outputs via dual DB25 **CONTROL PORT I/O** connectors
	Software LIOs	3 virtual inputs, 48 logical outputs, and 64 WheatNet Logic Inputs/Outputs

Data must be enabled to activate contact **CONTROL PORT** closure 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. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

Algorithm Selected	IP	ISDN
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.

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 switch inputs and relay 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 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 Note:**
> - WheatNet logic I/Os 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.

See the section titled Creating Rules for more information.

Configuring RS232 Data

Once **Data** is enabled, the codec can be connected to external devices and transport RS232-compatible data via the serial port on the rear panel of the codec.

1. Press the **SETTINGS** button.
2. Navigate to **System** and press the OK button.
3. Select **RS232 Config** and press OK.
4. Use the navigation buttons to select the correct baud rate.
5. Next, press OK to toggle flow control **On** to enable flow control.

> **Important Notes:**
> - When connecting to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms. See RS232 Data Configuration for more details.
> - It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
> - Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. 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 to another Tieline G5 or G6 codec. Session data sent from the dialing codec configures the answering codec.
> - RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast audio stream if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

24.12   Monitor Control Port I/O Status

Product	Feature	Notification
Gateway 4	Control Port GPIOs	Supports 8 control port inputs and 8 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connectors
Gateway 8/16	Control Port GPIOs	Supports 16 control port inputs and 16 opto-isolated outputs via dual DB25 **CONTROL PORT I/O** connectors

To monitor the status of DB25 **CONTROL PORT I/O** relay inputs and opto-isolated outputs:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System > Control Port I/O** and press the OK button.
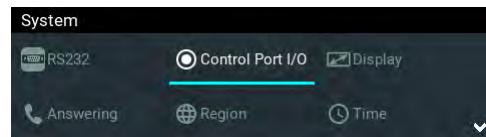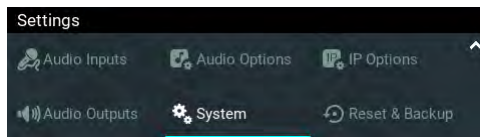
3. Select **Inputs** and then press the OK button to view input status.
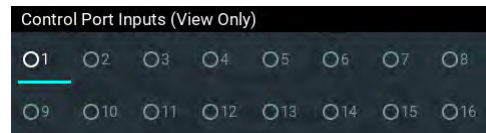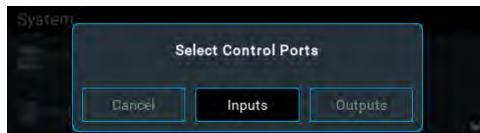
4. Select **Outputs** and then press the OK button to view output status. Select an output and press the OK button to toggle the output state from **Off** to **On**. Note: Input states cannot be changed in this way.
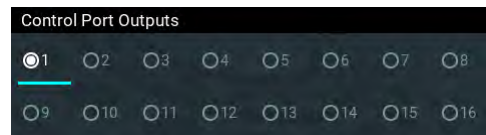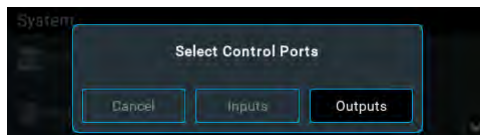
24.13   Configuring TCP/UDP Ports

In TCP and UDP networks the codec port is the endpoint of your connection. Software network ports are doorways for systems to communicate with each other. For example, several codecs in your studio may use the same public static IP address. Unique port numbers can be used to port forward audio to each codec.

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. The session port uses the TCP protocol because it is most likely to get through firewalls – ensuring critical session data (including dial, connect and hang-up data) will be received reliably. The default session and audio port settings in Tieline codecs, for both TCP and UDP connections, are outlined in the Default Program Ports section of the manual.

Changing Codec Port Numbers

Reasons for adjusting the port setting on your codec include:

- Creating a path through gateways and firewalls.
- 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.

Changing Tieline Session Ports when Answering

To adjust the local Tieline session data port used by your codec:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the OK button.
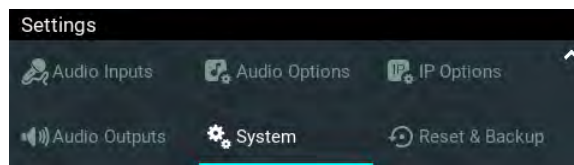
3. Navigate to select **Answering** and press the OK button.
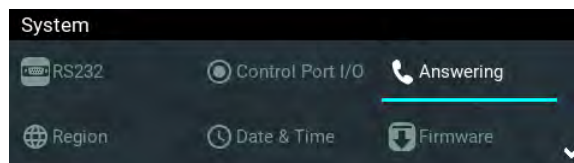
4. Select **Tieline IP Ports** and press the OK button.
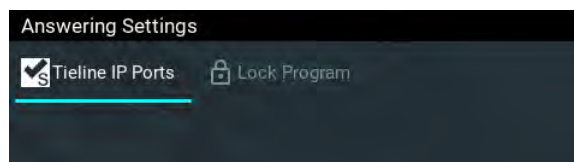
5. Navigate to **Session Port** or **Alt. Session Port** and press the OK button.

6. Adjust the port number and press **Enter** on the onscreen keyboard to save the new setting.

Audio Port Settings for Tieline Session Data and Sessionless IP Calls

The codec supports sessionless IP streaming, whereby the codec does not send Tieline session data when attempting to connect. When using this mode you need to configure 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).

It is also possible to configure the remote and local audio ports for a codec using Tieline session data to establish IP connections. This may be required because some firewalls require symmetric port configuration.

Sessionless Audio Port Configuration

When you select **Sessionless** as the **Session Protocol**:
- The default value for both the **Local** and **Remote** (audio) **Ports** is 9000
- The range of values for the audio ports is 2000 to 65535
- The audio port values can be set independently
- Both audio ports can always be configured, i.e. there is no dependency on encode/decode direction

"Tieline Codec" Port Configuration

If using the **Tieline Codec** setting for call establishment (i.e. Tieline session data is enabled), you can also change the default audio ports if required.
- The default value for the **Remote Audio Port** is 9000
- The range of values for the **Remote Audio Port** is 2000 to 65535
- The default port value for the **Local Audio Port** is **Automatic**. Note: **Automatic** indicates that the codec will allocate the local port value and send this information to the codec to which you are dialing
- The range of values for the **Local Audio Port** is 2000 to 65535

Sessionless Multicast Connections

For a sessionless multicast server connection:
- Only the **Remote Audio Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

For a sessionless multicast client connection:
- Only the **Local Audio Port** is available
- The default value for the port is 9000
- The range of values for the port is 2000 to 65535

24.14   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 modern data networks many different IP services like email, voice, web pages, video and streaming music coexist within the same network infrastructure.

Prioritizing IP Data Packets when Broadcasting

IP audio data packets can be programmed for expedited or assured forwarding (Quality of Service or QoS) when traversing different networks. Routers can also be programmed to ignore these forwarding priorities so they are not assured across all networks.

The codec can be programmed 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. Check with your IT administrator before changing this setting. By default the codec is programmed for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

Configuring QoS DSCP

1. Press the **SETTINGS** [icon] button.
2. Use the navigation buttons to select **IP Options** and press the [OK] button.
3. Navigate to **IP QoS** and press the [OK] button.
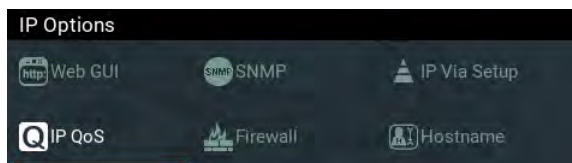
4. Press the [OK] button to edit the setting and use the **RETURN** [icon] button to delete the DSCP value already entered.
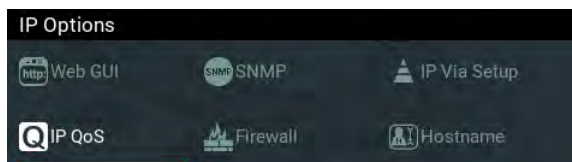
5. Next, use the numeric **KEYPAD** or onscreen keyboard to enter the new setting, then press the [OK] button to save the new setting.

> **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.

24.15 Reset and Restore Factory Default Settings

There are several options in the **Reset & Restore** menu allowing restoration of factory default settings within the codec.

	Function	Description
1	Backup Settings	Select to backup custom Program and Scheduler data, and/or System data.
2	Restore Settings	Select to restore custom Program and Scheduler data, and/or System data.
3	Reset Audio Settings	Select to restore factory default settings for Audio and connection-related menu settings
4	Restore Factory Defaults	Select to restore factory default settings, excluding user defined programs and call history
5	Reboot Codec	Select to reboot the codec
6	Clear Programs & Call History	Deletes custom programs and recent calls in the codec; speed dial contacts are retained
7	Clear Logs	Deletes codec event and log history. Note: This should only be performed if instructed to by Tieline support staff.

Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **Reset & Backup** and press the button.
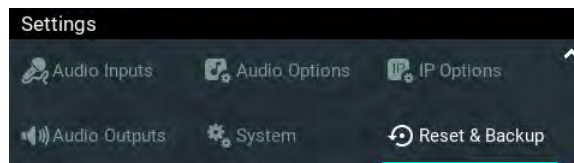
3. Navigate to the preferred option and press the button.
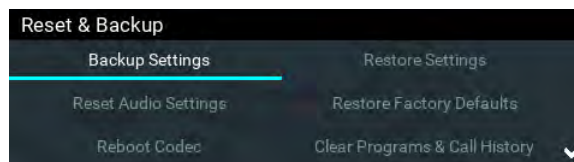
4. Press the button to confirm the function using the confirmation dialog.
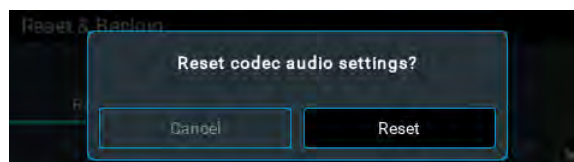
Reset and Restore Factory Defaults using the Web-GUI

See Reset Factory Default Settings to use the HTML5 Toolbox Web-GUI to reset and restore factory defaults.

24.16 System Backup and Restore

The **Reset & Backup** menu allows users to backup and restore Program and Scheduler information and/or system data to external media like an SD card or USB stick. Note: A single partition SD card must be used.

> **Important Note:** Initial firmware only supports an SD card for saving backup data. Therefore it is only possible to restore codec data files using an SD card backup.

Creating a Backup File

1. Insert a single partition FAT 32 formatted SD card into the **SD CARD SLOT** on the front of the codec.
2. Press the **SETTINGS** button.
3. Use the navigation buttons to select **Reset & Backup** and press the OK button.
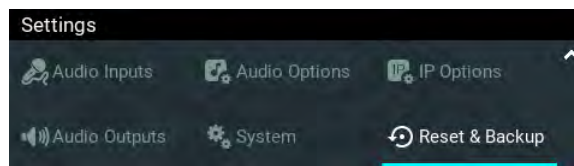
4. Select **Backup Settings** and press the OK button.
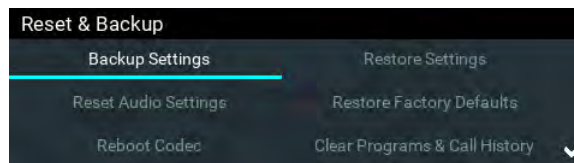
5. Select the preferred backup option and press the OK button.
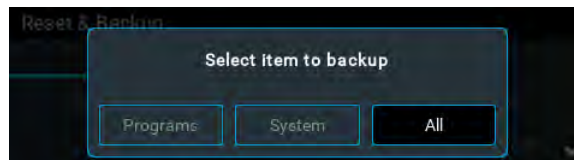
6. Edit the file name if required and then navigate to **Enter** and press the OK button to save the file.
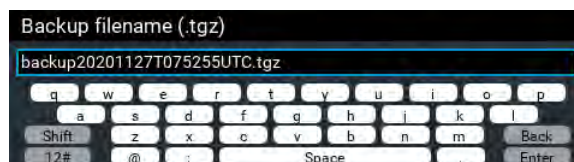
7. A confirmation dialog is displayed when the file is saved successfully.
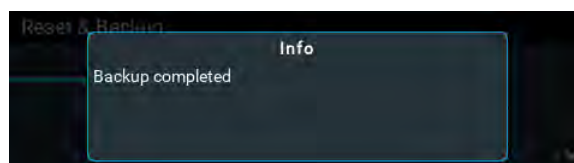
Restoring Data from a Backup File

1. Insert a single partition FAT 32 formatted SD card with a backup .tgz file into the **SD CARD SLOT** on the front of the codec.
2. Press the **SETTINGS** button.
3. Use the navigation buttons to select **Reset & Backup** and press the OK button.

4. Select **Restore Settings** and press the OK button.
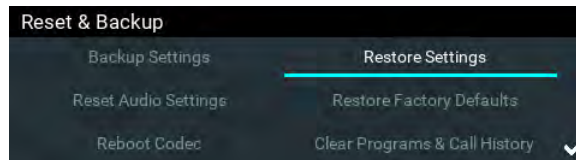
5. Select the preferred restore option and press the OK button.
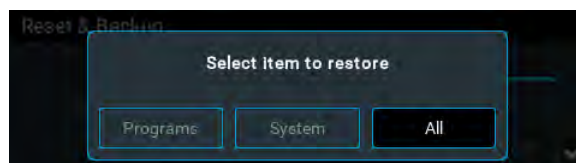
6. Select the file to restore and press the OK button.
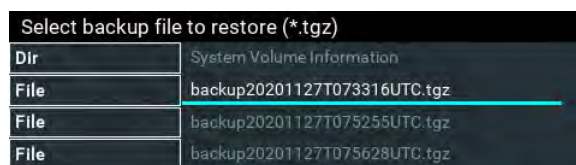
7. A dialog confirms data has been restored successfully.
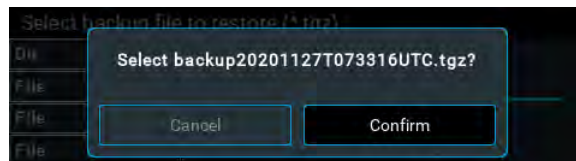
24.17   Configuring SNMP Settings

The codec supports Simple Network Management Protocol (SNMP) for managing devices on IP networks. To configure SNMP settings:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the OK button.
3. Navigate to **SNMP** and press the OK button.
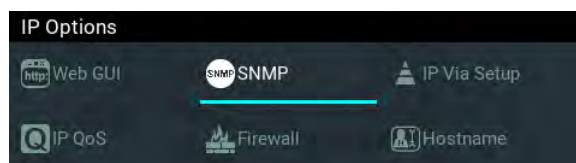
4. Navigate to each setting in turn and press the OK button to adjust and save each new setting.

> **Important Note:** For more information on SNMP codec settings see <u>Configuring SNMP in the Codec</u>.

24.18   Adjusting Time Settings

By default **Use NTP** time is enabled in the codec. 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 adjust time settings in the codec:

1. Press the **SETTINGS** button and navigate to **System**.
2. Use the navigation buttons to select **Date and Time** and press the button.

3. Navigate to **Use NTP** and press the button to enable or disable this feature.

4. Navigate to **NTP Synchronize Now** and press the button to synchronize the codec time with the designated **NTP Server**.

24.19   Upgrading Firmware

To download the latest codec firmware visit www.tieline.com. Copy the firmware file onto a FAT32 formatted SD card and then use the following procedure to perform a firmware upgrade. Note: A single partition FAT32 formatted SD card must be used.

1. Insert the SD card into the **SD CARD SLOT** on the front panel of the codec.
2. Press the **SETTINGS** 🔧 button.
3. Navigate to **System** and press the 🆗 button.
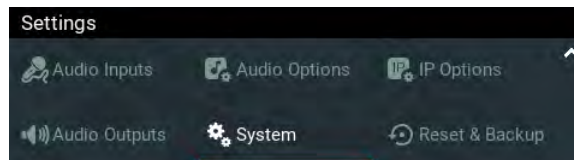
4. Navigate down to **Firmware** and press the 🆗 button.
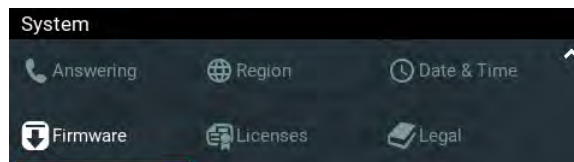
5. Navigate to the firmware file and press the 🆗 button to commence the upgrade.
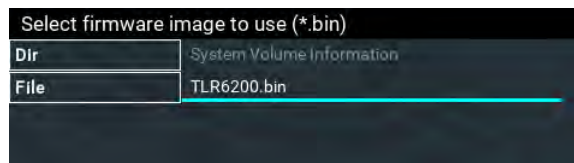
6. Progress of the update is displayed and when complete the codec will automatically reboot.

> ℹ️ **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

24.20   Installing a Security Certificate

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.

Certificate Installation

To install certificates purchased from a reputable vendor:

1. Press the **SETTINGS** [wrench icon] button.
2. Use the navigation buttons to select **IP Options** and press the [OK] button.
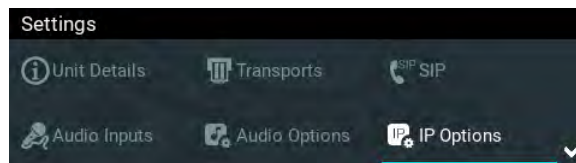
3. Select **Web-GUI** and press the [OK] button.
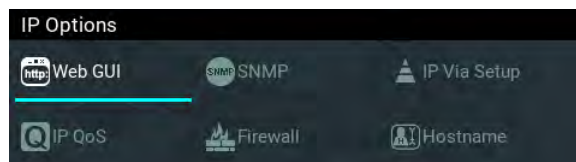
4. Navigate down to **SSL Certificates** and press the [OK] button.
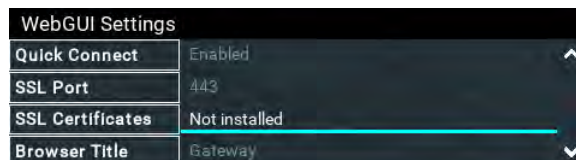
5. Ensure the Private Key, digital SSL Certificate and Intermediate Certificate (if required), are loaded onto an SD card and then insert it into the front panel **SD CARD SLOT**. Note: A single partition FAT32 formatted SD card must be used.
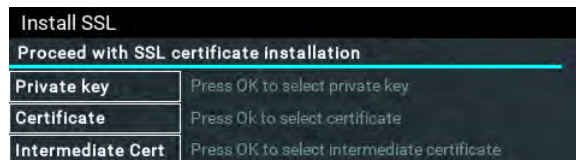
6. Select **Private Key** and navigate to the correct directory and .key (Private Key) file to install from the SD card and press the [OK] button.
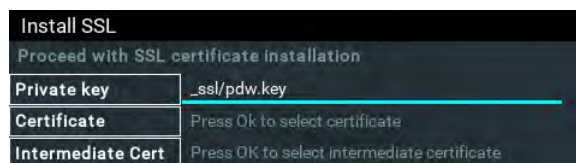
7. Select **Certificate** and navigate to the SSL Certificate (.crt) file on the SD card and press the ⓞ button.

Install SSL	
Proceed with SSL certificate installation	
Private key	_ssl/pdw.key
Certificate	_ssl/da7dd8c6e5bdcd9b.crt
Intermediate Cert	Press OK to select intermediate certificate
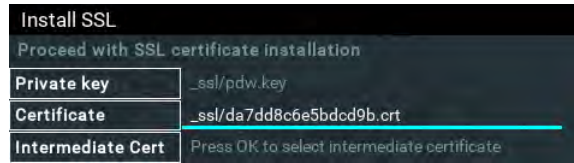
8. If an Intermediate Certificate has been supplied, select **Intermediate** and navigate to the Intermediate Certificate (.crt) file on the SD card and press the ⓞ button.

Install SSL	
Proceed with SSL certificate installation	
Private key	_ssl/pdw.key
Certificate	_ssl/da7dd8c6e5bdcd9b.crt
Intermediate Cert	_ssl/gd_bundle-g2-g1.crt
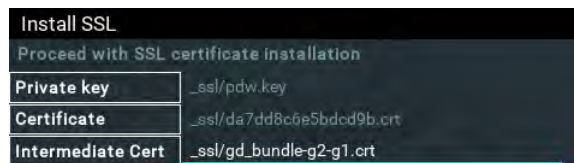
9. After adding the private key, SSL certificate, and intermediate certificate (if supplied), navigate up to **Proceed with SSL certificate installation** and press the ⓞ button to install the files.

Install SSL	
Proceed with SSL certificate installation	
Private key	_ssl/pdw.key
Certificate	_ssl/da7dd8c6e5bdcd9b.crt
Intermediate Cert	_ssl/gd_bundle-g2-g1.crt
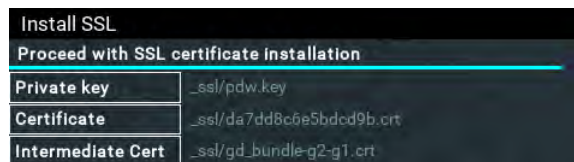
10. The **SSL Certificates** menu should confirm the files are successfully **Installed and enabled**.

WebGUI Settings	
SSL Port	443
SSL Certificates	Installed and enabled
Browser Title	Gateway
CSRF Protection	Disabled
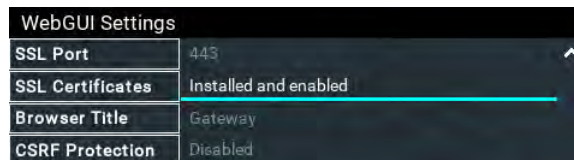
11. To access a codec via the HTML5 Toolbox Web-GUI in a browser after installing SSL security certificates ensure you type "https://" before the codec IP address. For example, https://172.16.0.100.

Disable or Remove SSL Security Certificates

To disable or remove installed SSL security certificates from a codec:

1. Press the **SETTINGS** 🔧 button.
2. Use the navigation buttons to select **IP Options** and press the ⓞ button.

Settings		
ⓘ Unit Details	🔳 Transports	📞 SIP
🎤 Audio Inputs	🔧 Audio Options	🔧 IP Options
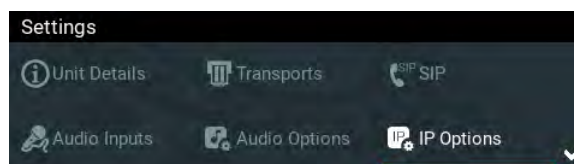
3. Select **Web-GUI** and press the ⊙ button.
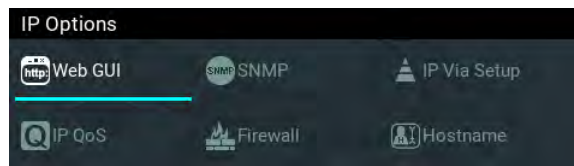
4. Navigate down to **SSL Certificates** and press the ⊙ button.
5. Select either **Disable** or **Remove** and press the ⊙ button.
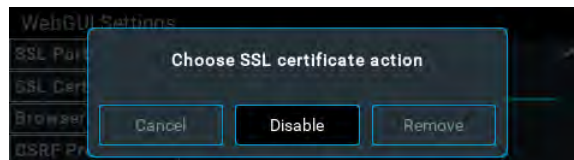
6. Confirm disabling or removal of the SSL files in the onscreen dialog.
7. A dialog confirms the certificates have been disabled or removed successfully.

Changing the Default SSL Port

The codec uses the standard TCP port 443 for SSL communications. The port number can be adjusted by navigating to **SETTINGS** 🔧 **> IP Options > Web-GUI > SSL Port**.

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.

24.21   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. Tieline also recommends that SNMP is disabled if a codec is connected to a public facing network like the internet.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the OK button.
3. Navigate to **Firewall** and press the OK button.
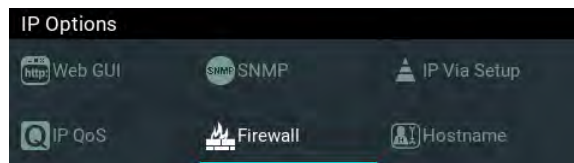
4. Select and configure **Ping**, **SSH**, **HTTP**, **HTTPS**, **NTP** and **SNMP** firewall options.
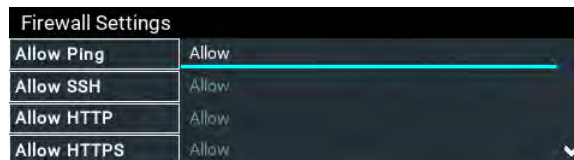
24.22   Enabling CSRF Security

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 **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the OK button.
3. Use the navigation buttons to select **Web-GUI** and press the OK button.
4. Navigate down to **CSRF Protection** and press the OK button to toggle between **Enabled** and **Disabled**.
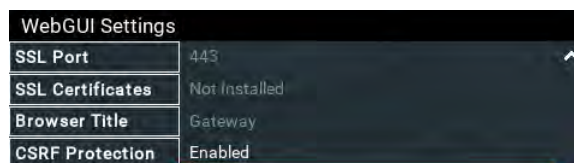
> **Important Note:** Ensure **CSRF Protection** is disabled in the codec when connecting to the CCC. This setting is **[Disabled]** by default and can also be configured using the **Options panel** in the HTML5 Toolbox Web-GUI.

25   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 Gateway and Gateway 4 codecs.
- All firmware versions in the Genie and Merlin codec families.
- All Bridge-IT 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.
   - Many Tieline 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.

6. Visit www.portforward.com and download the port checking application to verify your router's ports are open, or use a port checker like the one available at https://www.ipfingerprints.com/portscan.php.

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.
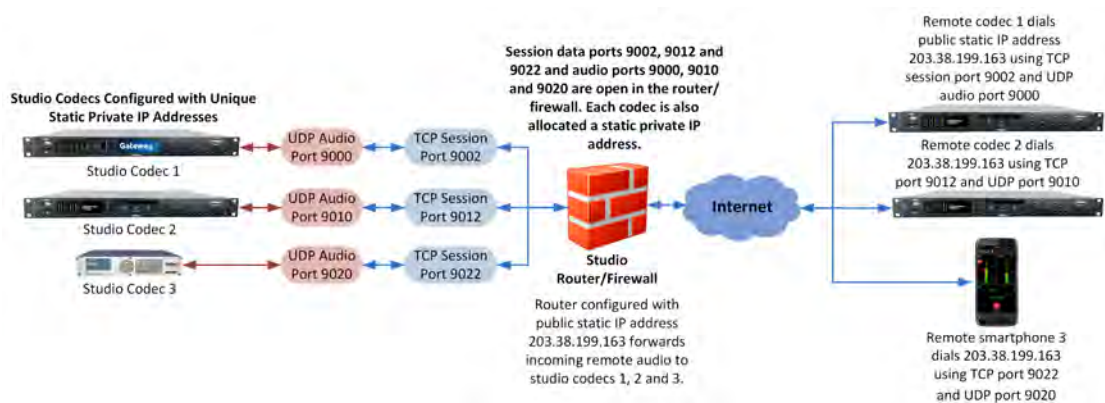
Port Forwarding to 3 Studio Codecs Sharing a Public Static IP Address

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		Gateway / Gateway 4 Codecs		Cloud Codec Controller
TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP	UDP	TCP
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	See Table Below		HTTP 80
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			HTTPS 443
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 Session: 5060	
					SIP Audio: 5004-5054				SIP Audio: 5004-5054	
					Fuse-IP 8999				Fuse-IP 8999	

Gateway 16 Channel Default Ports

Audio Stream	TCP Session Port	UDP Audio Port	SmartStream + 1 (UDP)	SmartStream + 2 (UDP)	SmartStream + 3 (UDP)
1	9002	9000	9001	9002	9003
2	9002	9010	9011	9012	9013
3	9002	9020	9021	9022	9023
4	9002	9030	9031	9032	9033
5	9002	9040	9041	9042	9043
6	9002	9050	9051	9052	9053
7	9002	9060	9061	9062	9063
8	9002	9070	9071	9072	9073
9	9002	9080	9081	9082	9083
10	9002	9090	9091	9092	9093
11	9002	9100	9101	9102	9103
12	9002	9110	9111	9112	9113

13	9002	9120	9121	9122	9123
14	9002	9130	9131	9132	9133
15	9002	9140	9141	9142	9143
16	9002	9150	9151	9152	9153

Configure a Static IPv4 Address in Gateway and Gateway 4

Static IP addresses are fixed addresses which are 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 **SETTINGS** button.
2. Select **Transports** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2** or a **VLAN** interface.
4. Select **Configuration** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the button.
5. Select **IPv4 Mode** and press the button.
6. Select **Static** and press the button.
7. Navigate to **IPv4 Address** and enter the IP address (using the keypad or onscreen keyboard), then press the button (or select **Enter** using the onscreen keyboard).
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the button.
9. Navigate to **IPv4 Gateway** and enter the Gateway details, then press the button.
10. Press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings.
11. Check the **Unit Details** menu to ensure the new static IP address has been entered correctly.

Configuring a Static Public or Private IP Address in Genie, Merlin and Bridge-IT (v.2.x firmware) Codecs

1. Press the **SETTINGS** button.
2. Select **Network** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2**, **Wi-Fi** or a **VLAN** interface.
4. Select **Config** and then **Usage** and then the appropriate control and/or streaming mode for the connection. Next, press the button.
5. Select **IPv4 Mode** and press the button.
6. Select **Static** and press the button.
7. Navigate to **IPv4 Static** and enter the IP address, then press the button.
8. Navigate to **IPv4 Subnet** and enter the Subnet Mask, then press the button.
9. Navigate to **IPv4 Gateway** and enter the Gateway details, then press the button.
10. Use the up navigation button to scroll to the top of the menu and select **Apply Setting**, then press the button to confirm the new settings.
11. Check the **Unit Details** menu 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	**(Not available in legacy Commander and i-Mix codecs)**
IP Address	: : : : : : :
IPv6 Prefix Size	
IPv6 Gateway	: : : : : : :

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 on a G3 codec.
> - The **Send** link quality reading is the same as the Remote (**R**) setting displayed 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" above).

Testing your Codec

- Visit www.tieline.com or contact Tieline 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.

26   Reference

The following sections contain reference and troubleshooting information.

26.1   Regular Maintenance

Tieline recommends the codec undergoes regular maintenance to ensure operational efficiency and prolong its life.

⚠️ **WARNINGS:** All work should be carried out by suitably qualified personnel. Remove both power leads from the codec before removing the cover. All parts are mounted on plugs and only a Philips screwdriver is required. Ensure that fan mounting lugs are not hooked out by the cover.

Maintenance Schedule

Tieline recommends a three year maintenance schedule which includes the following procedures to be completed:

1. Evacuate all dust from the unit and clean vents.
2. Replace both PSUs.
3. Replace the fan.

Controlled rack environments may allow a longer maintenance cycle. Uncontrolled environments, where temperatures are elevated, may require a shorter maintenance cycle.

Tieline recommends that the racks in which codecs are installed are thoroughly evacuated to ensure proper airflow from the bottom to the top. Where space is available, a 1RU gap between codecs will assist in minimizing internal temperature build up. Tieline has incorporated dual redundant PSUs and backup temperature alarm features to assist in maintaining reliable operations. The fan has been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. If rack temperatures are elevated above 40 degrees Celsius, the fan speed will increase to reduce CPU temperature. Maximum fan speed is reached when the temperature reaches 45 degrees Celsius. Fan speed control circuitry reduces the fan speed as internal rack temperatures fall below 45 degrees Celsius. This greatly extends the working life of the fan and the codec.

Tieline recommends that the racks in which codecs are installed are thoroughly evacuated to ensure proper airflow from the bottom to the top. Where space is available, a 1RU gap between codecs will assist in minimizing internal temperature build up. Tieline has incorporated dual redundant PSUs and backup temperature alarm features to assist in maintaining reliable operations. The fan has been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. The fan is activated to reduce the CPU temperature when the internal rack temperature reaches 40 degrees Celsius. Maximum fan speed is attained when the temperature reaches 45 degrees Celsius. Fan speed control circuitry reduces the fan speed as internal rack temperatures fall below 45 degrees Celsius. This greatly extends the working life of the fan and the codec.

26.2 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 http://www.ipfingerprints.com/portscan.php 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 web-tool for IPv4, plus TraceRoute and TracePath software tools.
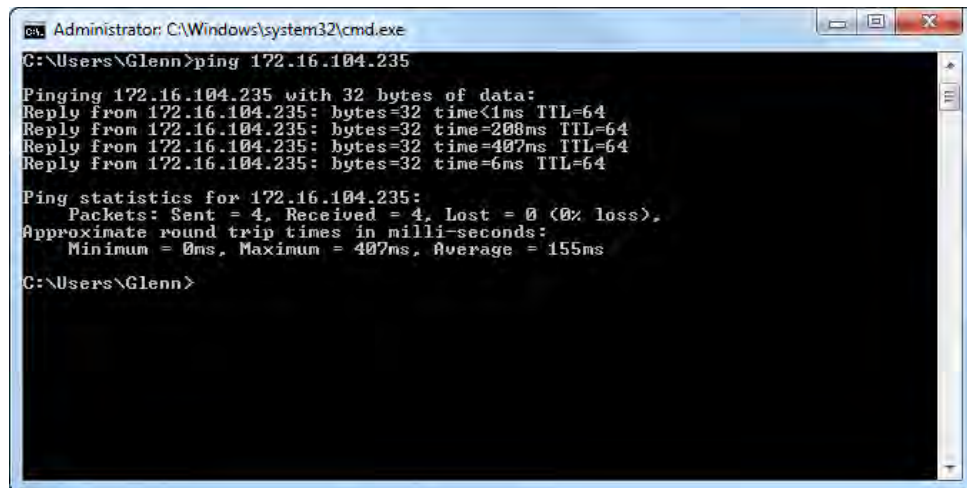
Ping the Codec

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.

26.3   Compliances and Certifications

Declaration of Conformity

This Gateway codec meets the requirements of directives for CE and C-Tick 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

This is to certify that Tieline Gateway is shielded against the generation of radio interference in accordance with the application of EN 55 022: 2006 Class A. 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.

Canadian Department of Communications Radio Interference Regulations

This digital apparatus (Tieline Gateway) 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 Gateway) 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.

26.4   FCC Compliance Statements

FCC Part 15

Compliance: Tieline Research Pty. Ltd., 4 Bendsten Place, Balcatta, Western Australia 6021.

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. Changes or modifications not expressly approved by Tieline Pty Ltd could void the user's authority to operate the equipment.

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 and correct the problem by one or more of the following measures:

1. Increase the separation between the equipment and the receiver;
2. Connect the equipment into an outlet on a circuit different to that used by the receiver;
3. Consult the dealer or an experienced radio/TV technician.

Tieline
The Codec Company

Supplier's Declaration of Conformity
47 CFR § 2.1077 Compliance Information

Product Name   Gateway
Regulatory Model Number   TLR6200
Product Options   ALL

Responsible Party

Tieline Research Pty Ltd
ACN 099-303-045
C/- Tieline America LLC
7202 E.87th Street
Suite #116
Indianapolis IN 46256
USA

FCC Compliance Statement

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.

Tieline Research Pty Ltd

William McLean
Chief Executive Officer

October 20, 2020

For regulatory topics only:

Tieline Research Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia

Ph: 61 8 94132000

Web: www.tieline.com

26.5   Declaration of Conformity

Tieline®
The Codec Company

Declaration of Conformity
according to ISO/IEC 17050-1 and EN 17050-1

Manufacturer's Name	Tieline Research Pty Ltd ACN 099-303-045
Manufacturer's Address	4 Bendsten Place Australia Balcatta WA 6021 Australia

declares, that the product

Product Name	Gateway
Regulatory Model Number	TLR6200
Product Options	ALL

complies with the essential requirements of the following applicable European Directives, and carries the CE marking accordingly:

Low Voltage Directive (2014/35/EU)
EMC Directive (2014/30/EU)

and conforms to the following product standards

SAFETY	IEC 60950-1:2005+AMD1:2009+AMD2:2013
EMC	CISPR32-2015 Class A IEC 61000-4-2:2008 / EN 61000-4-2:2008 IEC 61000-4-3:2010 / EN 61000-4-3:2010 IEC 61000-4-4:2012 / EN 61000-4-4:2012 IEC 61000-4-5:2014 / EN 61000-4-5:2014 IEC 61000-4-6:2013 / EN 61000-4-6:2013 IEC 61000-4-11:2004 / EN 61000-4-11:2004
	Canada: ICES-003 USA: FCC Title 47 CFR, Part 15 Class A

Supplementary Information:

This Device complies with the requirements of CISPR 32:2015 Class A, Electromagnetic compatibility of multimedia equipment – Emission requirements and EN 55035:2017 Electromagnetic compatibility of multimedia equipment – Immunity requirements.

This Device complies with the requirements of ICES-003 – Information Technology Equipment (Including Digital Apparatus) Limits and Methods of Measurement – Class A.

Tieline Research Pty Ltd

William McLean
Chief Executive Officer

October 20, 2020

For regulatory topics only:

Tieline Research Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia
Ph: 61 8 94132000
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.

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27   Gateway Specifications

Input/Output Specifications	
RJ45 Analog Inputs 1-4/AES3 Inputs 1-4	Shared analog stereo line inputs 1-4, or stereo AES3 digital inputs 1-4
RJ45 Analog Outputs 1-4/AES3 Outputs 1-4	Shared analog stereo line outputs 1-4, or stereo AES3 digital outputs 1-4
RJ45 AES3 Inputs 5-8	Stereo AES3 digital inputs 5-8
RJ45 AES3 Outputs 5-8	Stereo AES3 digital outputs 5-8
Gigabit LAN Ports	2 x Gbit Ethernet ports for IP streaming over WANs (16 bidirectional mono or 8 bidirectional stereo)
Gigabit AoIP Streaming Ports	2 x Gbit Ethernet ports for AES67 or Optional WheatNet-IP streaming over LAN 1 (16 bidirectional mono or 8 bidirectional stereo); port 2 not operational in early releases. Note: Gateway codecs with a WheatNet card only support 8 streams maximum. To support more than 8 mono streams some sources need to be configured in mono mode.
USB 2.0 Host Port	Front panel host port (supports firmware upgrades, Wi-Fi USB modems and memory sticks)
USB 3.0 Host Port	Rear panel host port supports record and playback*
Front Panel Headphone Output	1 x 6.35mm (1/4") headphone Jack
Front Panel SD Card Slot	Full size push-push SD card slot for backup/restore config, firmware upgrades, file storage, and SDIO capability*
Internal SSD Storage	128GB Internal Solid-State Drive supports record, store and forward use cases for network syndication applications*
BNC Sync Input/Output	2 configurable input/output BNC sync connectors
Control Port In/Out	Dual DB-25 Control Ports support 16 CMOS control inputs and 16 opto-isolated solid-state relay outputs
Analog Input Impedance	> 10kW
Analog Output Impedance	< 30 ohm balanced
Clipping Level	Nominal output level +4dBu with maximum output + 24 dBu
A/D & D/A Converters	24 bit
AES3 (AES/EBU)	24 bit inputs with support for sample rates from 32kHz to 192 kHz*; 24 bit outputs
Frequency Response	20Hz to 22kHz
Total Harmonic Distortion	<0.002% at +22dBu unweighted
Signal to Noise Ratio	>100dB at +24dBu, unweighted, 20Hz – 20kHz
Crosstalk	< -95dB between adjacent channels
Data and Control	
Configuration and connectivity	HTML5 Toolbox Web-GUI, Cloud Codec Controller (CCC), TieLink Traversal Server*
Web Audio Monitoring	Selectable web audio monitoring of IP audio streams

Serial (DB-9)	RS232 up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel
Software Logic I/Os (SLIO)	16 hardware GPIOs, plus 3 virtual inputs, 48 logical outputs, and 64 WheatNet Logic Inputs/Outputs
Encoding and IP Streaming	
Encoding Formats	Tieline Music, Tieline MusicPLUS, Opus, G.711, G.722, MPEG Layer 2, MPEG Layer-3 LC-AAC, HE-AAC, HE-AACv.2, AAC-LD, AAC-ELD, 16/24 bit aptX® Enhanced algorithm. MPEG audio technologies licensed by Frunhofer IIS (http://www.iis.fraunhofer.de/audio)
Uncompressed IP	Linear PCM16/24 bit 32kHz, 44.1kHz, 48kHz, 96kHz sampling
IP Sample Frequencies	8kHz, 16kHz, 32kHz, 44.1kHz, 48kHz, 96kHz
Phase-locked Surround Sound Channels	Stream 4, 6 and 8 digital channels in phase-locked mode to deliver live content in both stereo and surround sound formats
Asymetric Encoding	Supports asymetric multi-format encoding
Protocols	RTP, DHCP, SNMP, DNS, HTTP, IGMP, IPv4/IPv6, RTCP, STUN, SSL Security Certificate
SmartStream PLUS	Primary plus 3 redundant streams supported (algorithm dependent)
Multi-unicasting	Support for Multi-unicast and 1, 2, 3, or 8 Multiple multi-unicast streams (80 connections in total)
Multicasting	Support for Multicast and Multi-multicast (8-16) streams over compatible IP networks
Fuse-IP	Bond IP interfaces to aggregate data
AoIP Standards & Specifications	
EBU N/ACIP Tech 3326	Audio Contribution Over IP Compliant
I3P EBU Tech 3347	Intercom Over IP Compliant
N/ACIP 3368	SIP Profiles Compliant
AES67 Compliant	44.1.kHz, 48kHz, 96kHz Sample rates; 16 & 24 Bit, 1-8 Channels, SDP
STMPE 2110-30 Compliant	Class A, Ax, B, Bx Sender and Receiver Compliant
ST 2022-7 Seamless Protection Switching[2]	Compliant with requirements for sending multiple redundant streams of RTP packets to enable seamless protection switching.
NMOS Compliant	NMOS IS-04 & IS-05 Discovery, Registration and Connection Management
Supported Audio Frames	125us, 250us, 333µs, 1ms, 4ms
Clock Modes Supported	Primary Leader* (Master), Follower (Slave), Follower Only (Slave Only)
Advanced Networking	
VLAN Tagging	IEEE 802.1Q,802.1p
Quality of Service (QoS)	Support for DiffServ (DSCP)
Synchronization	IEEE 1588-2008 (PTPv2)
Multicasting	IGMP v2 and v3
SAP	SAP v2 (Session Announcement Protocol) as defined in RFC 2974

Front Panel Interfaces	
Display	24-bit Color LCD Screen (480 x 128 pixels)
Keypad	26 button silicon keypad
Navigation	5 navigation and selection buttons
General	
Size	1U x 19" Rackmount
Dimensions	19" x 1 3/4" x 11 13/16" [482mm (W) x 44.45mm (H) x 300mm (D)] excluding rear connectors
Weight	8 lb 13.1 oz / 4 Kg
AC Power	Dual AC 90-240V IEC power inlets; 2A 50-60 Hz
Operating Temperature	0°C to 45°C (32°F to 113°F)
Humidity Operating Range	20% ≤ RH ≤ 70% (0 to 35°C/32°F to 95°F), non-condensing

* Future Release

28 Gateway 4 Specifications

Input/Output Specifications	
RJ45 Analog Inputs 1-4/AES3 Inputs 1-4	Shared analog stereo line inputs 1-4, or stereo AES3 digital inputs 1-4
RJ45 Analog Outputs 1-4/AES3 Outputs 1-4	Shared analog stereo line outputs 1-4, or stereo AES3 digital outputs 1-4
Gigabit LAN Ports	2 x Gbit Ethernet ports for IP streaming over WANs (4 mono, stereo and dual mono, or 2 stereo - full duplex)
Gigabit AoIP Streaming Ports	2 x Gbit Ethernet ports for AES67 or Optional WheatNet-IP streaming over LAN 1 (4 bidirectional mono or 2 bidirectional stereo); port 2 not operational in early releases
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Analog Input Impedance	> 10kW
Analog Output Impedance	< 30 ohm balanced
Clipping Level	Nominal output level +4dBu with maximum output + 24 dBu
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Signal to Noise Ratio	>100dB at +24dBu, unweighted, 20Hz – 20kHz
Crosstalk	< -95dB between adjacent channels
Data and Control	
Configuration and connectivity	HTML5 Toolbox Web-GUI, Cloud Codec Controller (CCC), TieLink Traversal Server*
Web Audio Monitoring	Selectable web audio monitoring of IP audio streams
Serial (DB-9)	RS232 up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel
Software Logic I/Os (SLIO)	8 hardware GPIOs, plus 3 virtual inputs, 56 logical outputs, and 64 WheatNet Logic Inputs/Outputs
Encoding and IP Streaming	

Encoding Formats	Tieline Music, Tieline MusicPLUS, Opus, G.711, G.722, MPEG Layer 2, MPEG Layer-3 LC-AAC, HE-AAC, HE-AACv.2, AAC-LD, AAC-ELD, 16/24 bit aptX® Enhanced algorithm. MPEG audio technologies licensed by Frunhofer IIS (http://www.iis.fraunhofer.de/audio)
Uncompressed IP	Linear PCM16/24 bit 32kHz, 44.1kHz, 48kHz, 96kHz sampling
IP Sample Frequencies	8kHz, 16kHz, 32kHz, 44.1kHz, 48kHz, 96kHz
Phase-locked Surround Sound Channels	Stream 4 digital channels in phase-locked mode to deliver live surround sound content
Asymetric Encoding	Supports asymetric multi-format encoding
Protocols	RTP, DHCP, SNMP, DNS, HTTP, IGMP, IPv4/IPv6, RTCP, STUN, SSL Security Certificate
SmartStream PLUS	Primary plus 3 redundant streams supported (algorithm dependent)
Multi-unicasting	Support for 1 or 2 Multi-unicast streams (20 connections in total)
Multicasting	Support for 1 or 2 Multicast streams to unlimited endpoints over compatible IP networks
Fuse-IP	Bond IP interfaces to aggregate data
AoIP Standards & Specifications	
EBU N/ACIP Tech 3326	Audio Contribution Over IP Compliant
I3P EBU Tech 3347	Intercom Over IP Compliant
N/ACIP 3368	SIP Profiles Compliant
AES67 Compliant	44.1.kHz, 48kHz, 96kHz Sample rates; 16 & 24 Bit, 1-8 Channels, SDP
SMPTE 2110-30 Compliant	Class A, Ax, B, Bx Sender and Receiver Compliant
ST 2022-7 Seamless Protection Switching[2]	Compliant with requirements for sending multiple redundant streams of RTP packets to enable seamless protection switching.
NMOS Compliant	NMOS IS-04 & IS-05 Discovery, Registration and Connection Management
Supported Audio Frames	125us, 250us, 333µs, 1ms, 4ms
Clock Modes Supported	Primary Leader* (Master), Follower (Slave), Follower Only (Slave Only)
Advanced Networking	
VLAN Tagging	IEEE 802.1Q,802.1p
Quality of Service (QoS)	Support for DiffServ (DSCP)
Synchronization	IEEE 1588-2008 (PTPv2)
Multicasting	IGMP v2 and v3
SAP	SAP v2 (Session Announcement Protocol) as defined in RFC 2974
Front Panel Interfaces	
Display	24-bit Color LCD Screen (480 x 128 pixels)
Keypad	26 button silicon keypad
Navigation	5 navigation and selection buttons
General	

Size	1U x 19" Rackmount
Dimensions	19" x 1 3/4" x 11 13/16" [482mm (W) x 44.45mm (H) x 300mm (D)] excluding rear connectors
Weight	8 lb 13.1 oz / 4 Kg
AC Power	Dual AC 90-240V IEC power inlets; 2A 50-60 Hz
Operating Temperature	0°C to 45°C (32°F to 113°F)
Humidity Operating Range	20% ≤ RH ≤ 70% (0 to 35°C/32°F to 95°F), non-condensing

* Future Release

29   Appendix A: RS232 and Control Port Wiring

Product	Feature	Notification
Gateway 4	Control Ports	One DB-25 connector supporting 8 CMOS solid state relays for the control of equipment, consisting of 8 relay closures and 8 opto-isolated outputs.
Gateway 8/16	Control Ports	Two DB-25 connectors supporting 16 CMOS solid state relays for the control of equipment, consisting of 16 relay closures and 16 opto-isolated outputs.

Relays

The Gateway range of codecs uses one or two DB-25 connectors to facilitate the use of CMOS solid state relays for the control of equipment. Operation functions can be configured using the **Rules panel** in the HTML5 Toolbox Web-GUI.

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 1-8

A closing contact across Inputs 1-8 to Ground will provide a closing contact on remote codec Outputs 1 to 8. If a Tieline 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.
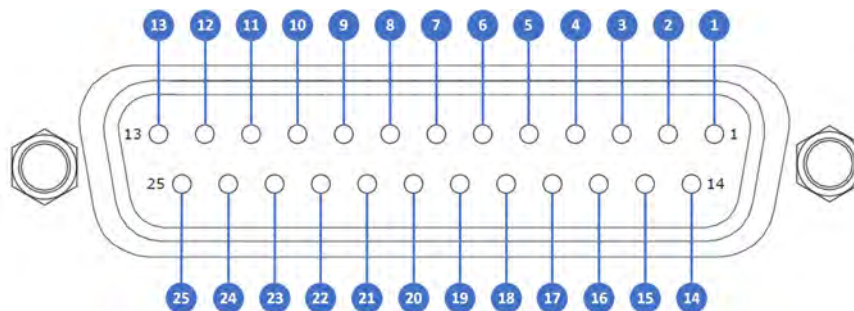
Female DB25
Codec Connector I/O 1-8

Pin	Pin Function
1	Output 1
2	Output 2
3	Output 3

4	Output 4
5	Output 5
6	Output 6
7	Output 7
8	Output 8
9	Input 1
10	Input 3
11	Input 5
12	Input 7
13	Ground
14	Output 1
15	Output 2
16	Output 3
17	Output 4
18	Output 5
19	Output 6
20	Output 7
21	Output 8
22	Input 2
23	Input 4
24	Input 6
25	Input 8

Important Note: For more information about how to program relay operations with a PC using the Toolbox Web-GUI, please see Creating Rules.

Control Port Pin-outs 9-16 (Gateway 8/16 Only)

A closing contact across Inputs 9-16 to Ground will provide a closing contact on remote codec Outputs 9 to 16.
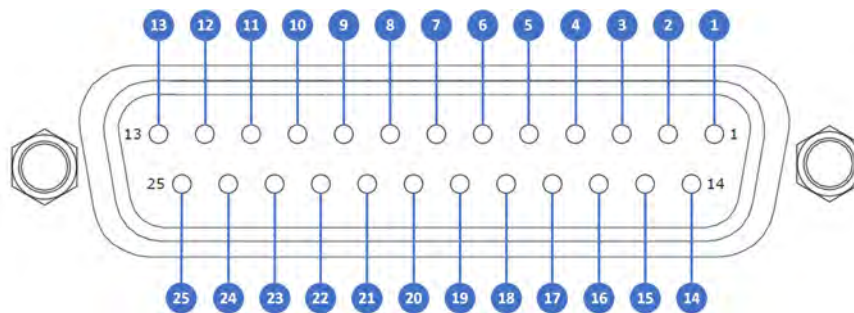
Female DB25
Codec Connector I/O 9-16

Pin	Pin Function
1	Output 9
2	Output 10
3	Output 11
4	Output 12
5	Output 13
6	Output 14
7	Output 15

8	Output 16
9	Input 9
10	Input 11
11	Input 13
12	Input 15
13	Ground
14	Output 9
15	Output 10
16	Output 11
17	Output 12
18	Output 13
19	Output 14
20	Output 15
21	Output 16
22	Input 10
23	Input 12
24	Input 14
25	Input 16

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
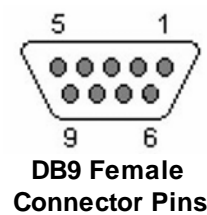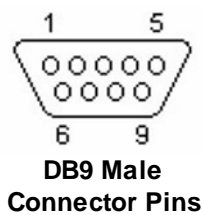
Important Notes:
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, data will flow unregulated and some may be lost.
- Ensure you match the serial port baud rate to match the rate of the external device you are connecting to. Ideally the settings on both codecs should match, or you could have data overflow issues.

- Only the dialing codec needs to be programmed to send RS232 data. Session data sent from the dialing codec will program all other compatible codecs (non-G3) when you connect.
- RS232 data can be sent from the dialing codec to all endpoints of a multi-unicast or multicast audio stream if your codec is capable of these connections. Note: Bidirectional RS232 data is only available on the first connection dialed when multi-unicasting.

30   Appendix B: RJ-45 Pinout Guide

Please see the following pinouts for RJ-45 adapters that can be used with the codec. Please note the codec does not use +/-15V on pins 7 and 8. Image courtesy of AngryAudio and StudioHub.

StudioHub+ Adapter Pinout Guide
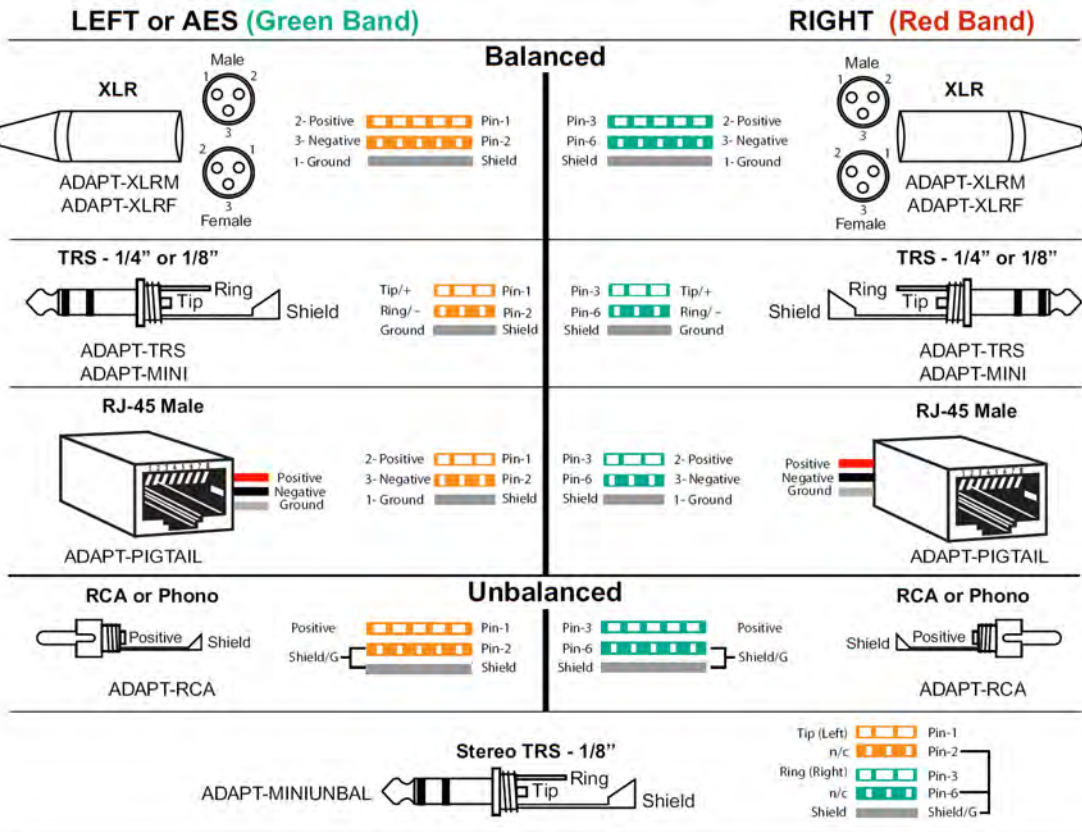
Channel	Color Pair	RJ-45 Pins	110 Pins	Ethernet
L+ / AES+	Wht/Org	1	3	TR+
L- / AES-	Org/Wht	2	4	TR-
R+	Wht/Grn	3	5	R+
R-	Grn/Wht	6	6	R-
nc	Wht/Blu	5	1	n/c
DC GND	Blu/Wht	4	2	n/c
15V-	Wht/Brn	7	7	n/c
15V+	Brn/Wht	8	8	n/c
Shield	Wht/Slt	Shield	9	x
Shield	Slt/Wht	Shield	10	x
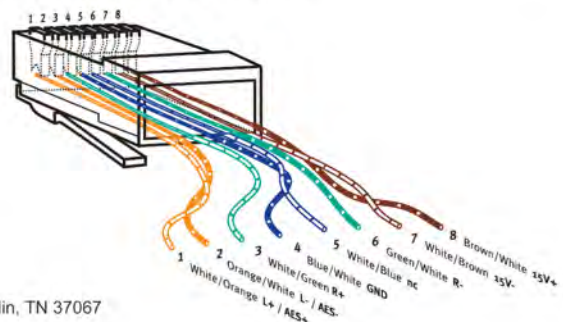
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+1 (615) 763-3033

31 Appendix C: Gateway 4 DB-25 Audio In/Out Pinout

Gateway 4 DB25 Analog and AES3 Audio Pinouts

In the following images:

- G = Ground
- C = Cold
- H = Hot

Balanced Analog DB-25 Wiring Pinouts
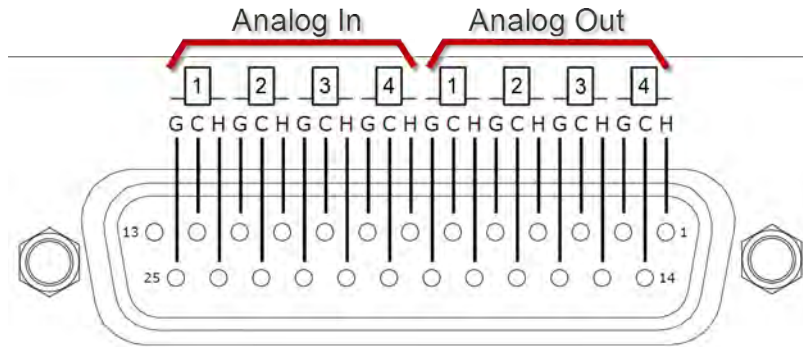
Digital AES3 (AES/EBU) DB-25 Wiring Pinouts

Index

- A -

- B -

- C -

- G -

- H -

- W -

- X -