Tieline®
The Codec Company

MPX I and MPX II
Codec User Manual

Software Version: 3.8.xx
Manual Version: 1.3_20250122
January, 2025

Table of Contents

1 Warnings & Safety Information

1. Both appliance power cables must be removed from the device for Power Disconnection.
2. Remove 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 https://tieline.com/warranty-repairs/ 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:

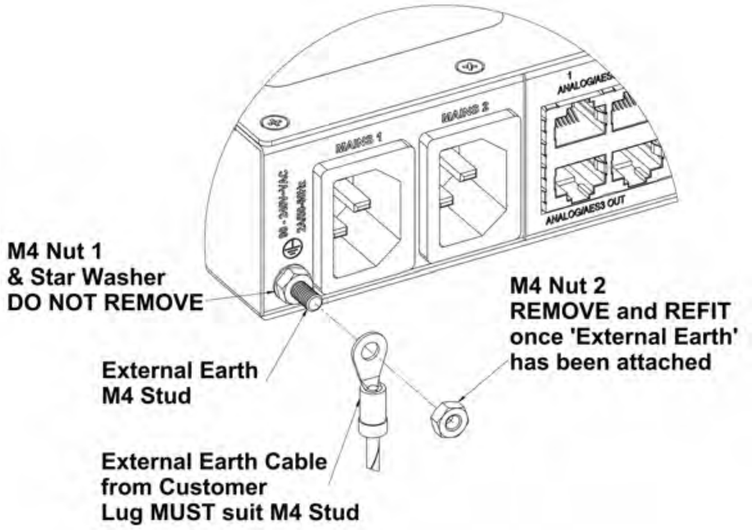
Before connecting this device to the power line, where required, the protective earth terminal screws of this device must be connected to the protective earth in the building installation.

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)
- Use shielded Ethernet Cable for connections.

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 https://tieline.com/warranty-repairs/ 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:

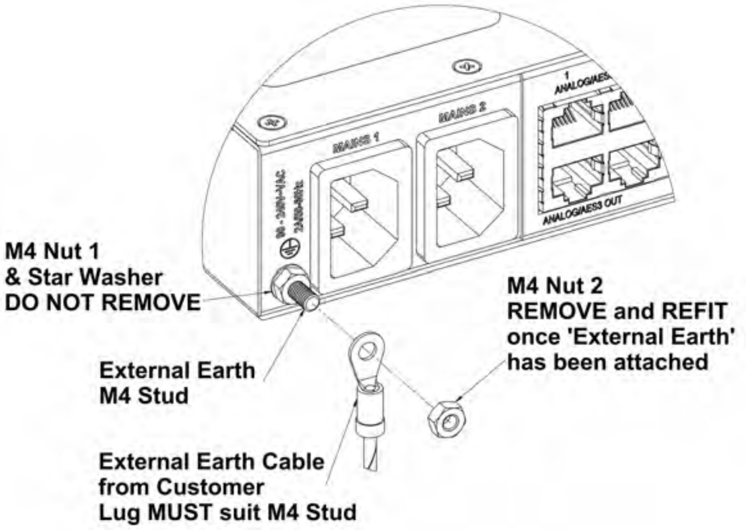
Avant de brancher cet appareil au réseau électrique, assurez-vous que la tension de la source d'alimentation correspond aux exigences de l'appareil. Reportez-vous aux spécifications de l'appareil pour plus d'informations sur la puissance nominale correcte de l'appareil.

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).
- Utilisez les câbles Ethernet blindés pour les connexions.

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 https://tieline.com/warranty-repairs/ 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:

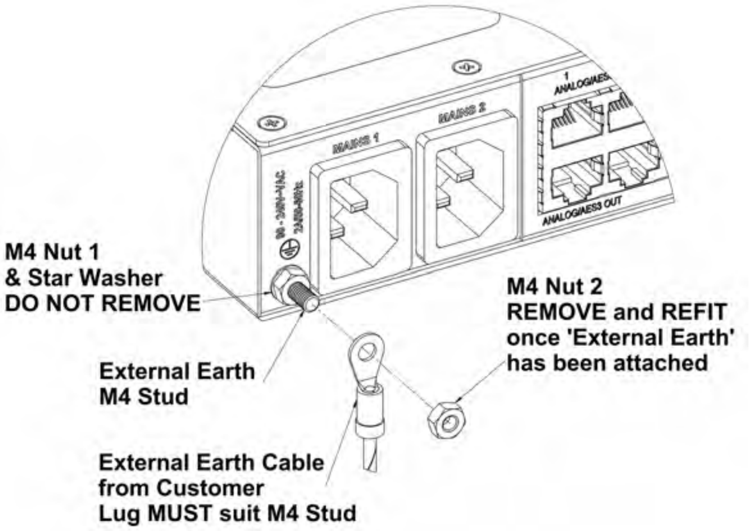
Bevor Sie dieses Gerät an das Stromnetz anschließen, stellen Sie sicher, dass die Netzspannung den Spezifikationen des Geräts entspricht. Informationen zur Stromaufnahme des Geräts finden Sie in den technischen Daten.

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).
- Verwenden Sie abgeschirmte Ethernet-Kabel für Verbindungen.

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

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. MPX I versus MPX II.

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 (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.
DVB-S and DVB-S2	DVB-S is the original Digital Video Broadcasting standard for satellite television. DVB-S2 is the second generation based on DVB-S with additional features.
Ember+	An open standard control protocol developed by Lawo which allows a third party application to gain access to device parameters.
Failover	Method of switching to an alternative backup Audio Stream if the primary connection is lost.
GUI	Graphical User Interface
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
Livewire+	Livewire+ is proprietary audio-over-Ethernet system created by Axia Audio (Telos Alliance) to route and distribute broadcast-quality audio throughout broadcast stations. (See the AoIP User Manual for more info)
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).
MicroMPX	MicroMPX transports a full FM composite signal, including pilot and RDS, at low bit-rates with perfect peak control.
MPE	Multi-protocol Encapsulation: MPE is a technique used within MPEG-TS to encapsulate data from different protocols (like IP data) for transmission.
MPEG-TS	MPEG-TS is a standard format for transmission and storage of audio, video, and data, and is used in broadcast systems such as DVB and ATSC.
MPX	A high bandwidth composite stereo multiplex signal containing the Main Channel (L+R), pilot tone, and the (L−R) difference signal.
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. (See the AoIP User Manual for more info)
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
PES	Packetized Elementary Streams: A method of encapsulating elementary streams into packets for transmission and processing in a standardized way.
PID	Packet Identifier: A PID uniquely identifies each packetized elementary stream within the transport stream. It helps demultiplexers and decoders to separate and process different streams correctly.
Primary Leader Clock	The primary source of synchronization for clock distribution via PTP. (See the AoIP User Manual for more info)
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.
RAVENNA	A technology for real-time transport of audio and other media data over IP networks. (See the AoIP User Manual for more info)
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.
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
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
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

6 Overview of MPX Codec Family

This user manual documents how to use Tieline's MPX I and MPX II codecs. The MPX I has a different front and rear panel compared to the MPX II.

MPX I and MPX II

Tieline's MPX I and MPX II codecs deliver composite FM multiplex (MPX) codec solutions for real-time network distribution of FM-MPX or MicroMPX* (µMPX) signals to transmitter sites. The MPX I is ideal for transmitting a composite STL signal from a single station with return monitoring, whereas the Tieline MPX II can transport two discrete composite FM-MPX signals from the studio to transmitters with return monitoring. Both units support analog MPX on BNC, MPX over AES192, and multipoint signal distribution, to deliver a wide range of flexible composite encoder and decoder solutions for different applications.

The MPX I and MPX II support sending the full uncompressed FM signal, or high quality compressed MicroMPX at much lower bit-rates. MicroMPX is an optional feature available for purchase. Each unit can operate as either an encoder or decoder. You can also order an optional DVB-S2 satellite tuner card at purchase to support decoding DVB-S or DVB-S2 signals.

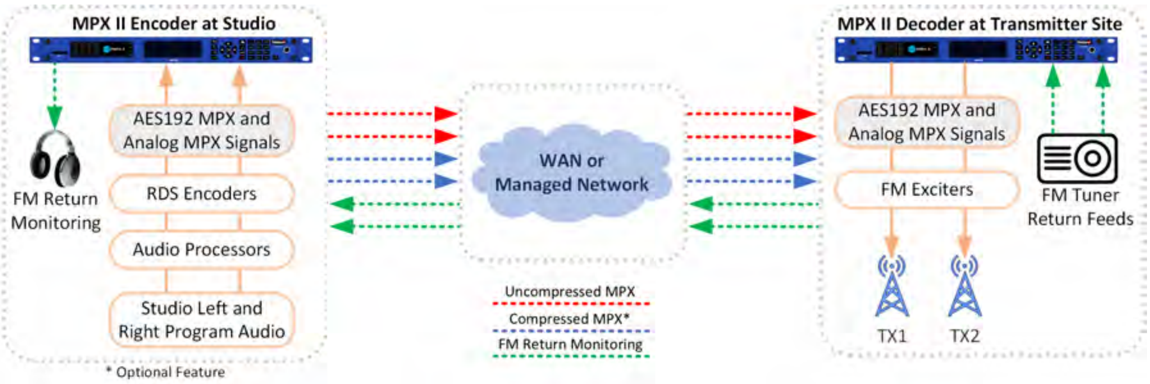
*Optional Feature

MPX I Codec

MPX II Codec

Applications

- Support for Tieline session IP audio streams: MPX I supports dual mono, stereo, multi-unicast and multicast. MPX II supports up to 4 mono, or dual stereo, or a mix of mono and stereo connections, plus multi-unicast and multicast modes.
- Single or Dual MPX Composite Encode or Decode: Encode/Decode up to two point-to-point MPX/µMPX composite signals.
- MPX/MicroMPX Composite Encode or Decode with Encoder or Decoder FM Monitoring: Encode/Decode two point-to-point MPX/MicroMPX composite signals; at the encoder monitor the demodulated local MPX input or the return feed. At the decoder monitor the MPX output or the secondary MPX feed, which is either the local MPX input or the secondary stream.



Compressed μMPX and uncompressed FM-MPX encoding and decoding with return confidence monitoring

Key Features

- Transport uncompressed MPX or reduce bandwidth requirements by transmitting compressed MicroMPX Composite signals to sites. Support for GPIOs in MPX or MicroMPX modes.
- Monitor demodulated MPX at the encoder or decoder and configure return FM confidence monitoring as required.
- Use a single MPX I or MPX II codec to multicast uncompressed MPX or compressed MicroMPX signals, or Multi-unicast MicroMPX signals to reduce CAPEX and OPEX at the studio and TX sites
- Both the MPX I and MPX II can operate as an encoder or decoder.
- Support for both analog and digital MPX signals allows networks to support transmissions using analog transmitters as networks transition to newer all-digital setups over time.
- Redundant streaming with hitless packet switching, RIST and Forward Error Correction (FEC)
- SD card backup delivers file failover.
- Dual internal PSUs for redundancy.
- Full remote control using the HTML5 Toolbox Web-GUI, Cloud Codec Controller, plus comprehensive automated alarms and SNMP monitoring.

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 AoIP User Manual from www.tieline.com.

7 MPX Front Panel Controls

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

MPX I Front Panel

MPX II Front Panel

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
F2	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
	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.
User	The **User LED** is a configurable LED.
CXN (Connection)	When dialing a connection the **CXN LED** flashes until the codec connects. Note: It also flashes until all streams / transports are connected when multiple streams / transports are connecting.
Power	The **POWER LED** indicates power is connected to the codec.

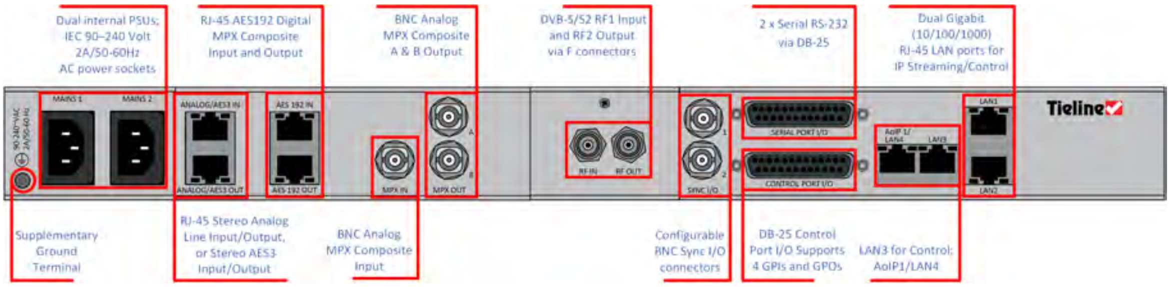
Stereo RTS Headphone Output

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

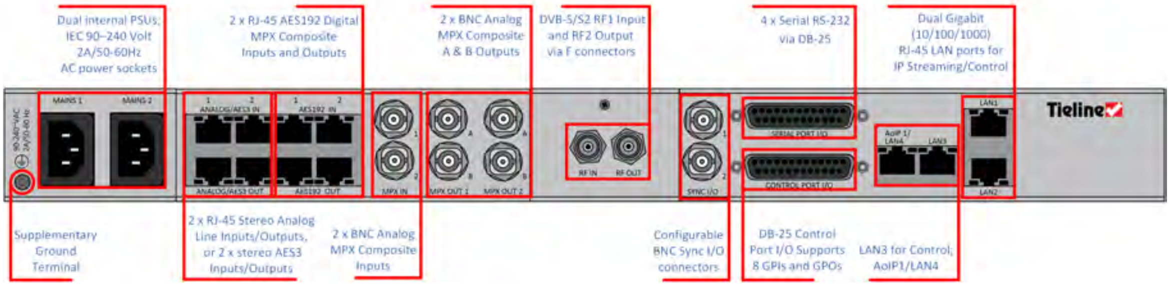
SD Card Slot

The SD card slot supports file failover backup audio and firmware upgrades.

8 MPX Rear Panel Connections



MPX I Rear Panel



MPX II Rear Panel

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 internal AC power supplies with standard IEC sockets.

Balanced Analog and AES3 Input/s and Output/s

MPX I: 1 Shared RJ-45 Stereo Analog Line Input or stereo AES3 Input and 1 Shared Stereo Analog Line Output or stereo AES3 Output.

MPX II: 2 Shared RJ-45 Stereo Analog Line Inputs or 2 x stereo AES3 Inputs and 2 Shared Stereo Analog Line Outputs or 2 x stereo AES3 Outputs.

AES192 Digital Composite Input/s and Output/s

MPX I: 1 RJ-45 Stereo AES192 Digital MPX Composite Input and Output.

MPX II: 2 RJ-45 Stereo AES192 Digital MPX Composite Inputs and Outputs.

BNC Analog MPX Composite Input/s and Outputs

MPX I: 1 BNC Analog MPX composite Input and 2 BNC Analog MPX Outputs (1 A & B output)

MPX II: 2 BNC Analog MPX composite Inputs and 4 BNC Outputs (2 A & B outputs)

Module Slot

A single module slot supports an optional satellite tuner card with MPEG-TS and MPE support which can receive DVB-S or DVB-S2 signals.

BNC Sync I/O Connectors

2 configurable BNC sync connectors.

DB-25 Serial Port I/O

MPX I: 2 RS-232 Serial I/O Ports (2 sync with audio).

MPX II: 4 RS-232 Serial I/O Ports (4 sync with audio).

Up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel; supports hardware and software flow control. DB-25 pinouts are available in Appendix A.

DB-25 Control Port I/O

MPX I: One DB-25 connector supporting 4 CMOS solid state relays for the control of equipment, consisting of 4 relay closures and 4 opto-isolated outputs.

MPX II: One DB-25 connectors supporting 8 CMOS solid state relays for the control of equipment, consisting of 8 relay closures and 8 opto-isolated outputs.

DB-25 pinouts are available in Appendix A.

LAN/AoIP Ports

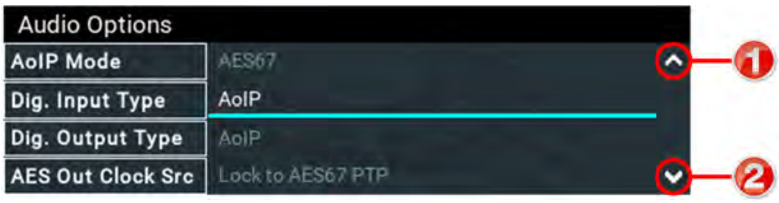
Dual Gigabit (10/100/1000) RJ-45 LAN 1 and LAN 2 ports for IP streaming and control. Gigabit RJ-45 LAN3 Port for control and Gigabit RJ-45 AoIP/LAN4 port.

9 Menu Navigation

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

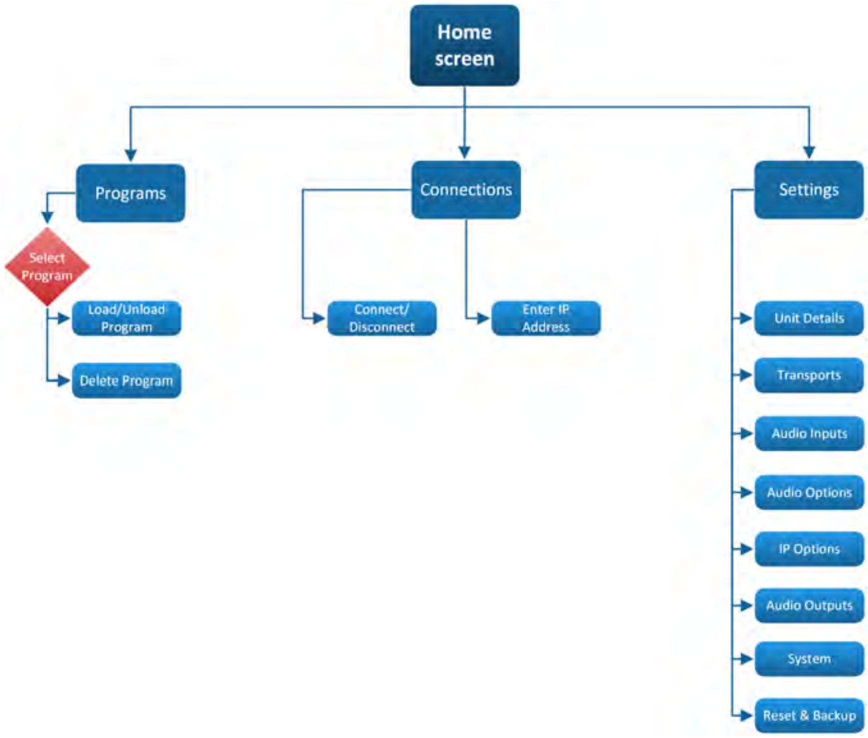
	Features	Codec Home Screen Elements
1	Screen Name	The name of the current screen
2	Programs	View and edit Program configurations
3	Connections	Adjust connection settings and show connections and connection details.
4	Settings	Menus for configuring a wide range of codec settings

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.

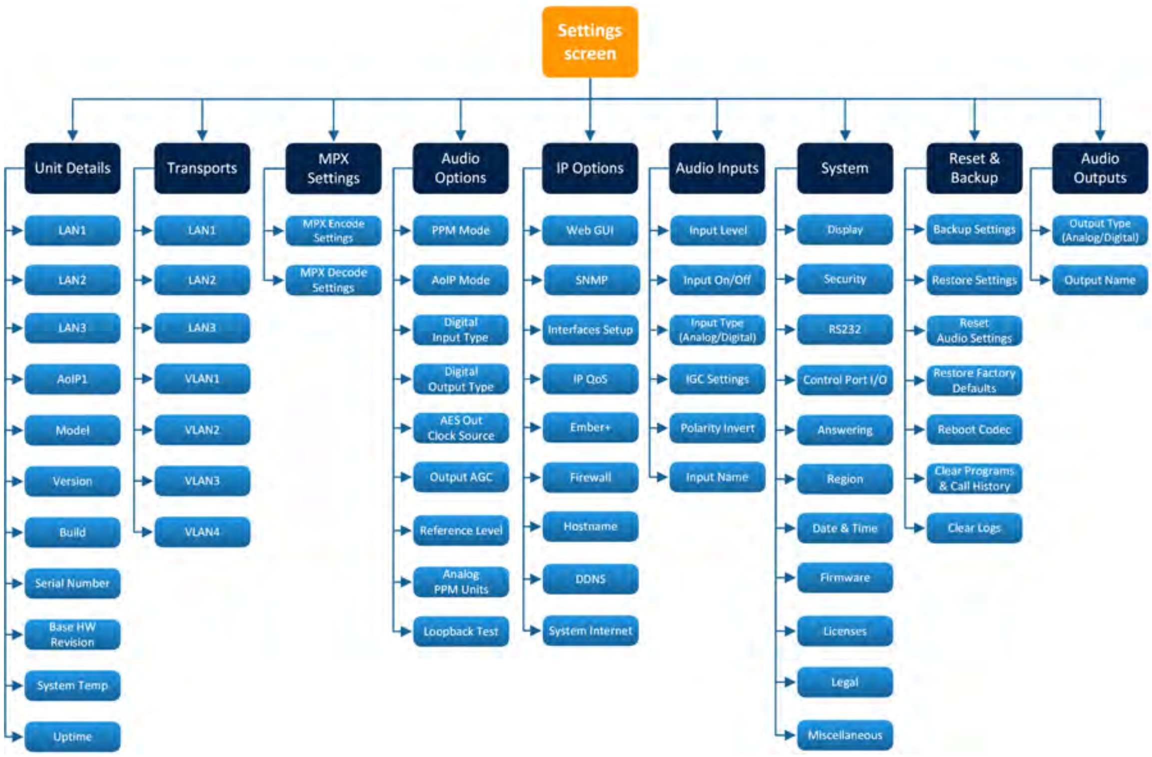


	Features	Codec Home Screen Elements
1	Up Arrow	Arrow indicating menus can scroll upwards
2	Down Arrow	Arrow indicating menus can scroll downwards

Home Screen Menus



Settings Menu Navigation



10 MPX Input and Output Settings

Product	Feature	Notification
MPX I	RJ-45 AES192 MPX Input / Output	**AES192 IN / AES192 OUT** provide 1 stereo AES192 digital MPX composite input and output.
MPX I	BNC Analog MPX Composite Input; BNC Analog MPX Composite A & B Output	**MPX IN / MPX OUT A/B** provide 1 BNC Analog MPX composite Input and 2 BNC Analog MPX Outputs (1A & B output)
MPX II	2 RJ-45 AES192 MPX Inputs / Outputs	**AES192 IN 1 & 2 / AES192 OUT 1 & 2** provide 2 stereo AES192 digital MPX composite inputs and outputs.
MPX II	2 BNC Analog MPX Composite Inputs; 2 BNC Analog MPX Composite A & B Outputs	**MPX IN 1 & 2 / MPX OUT 1A/B & 2A/B** provide 2 BNC Analog MPX composite Inputs and 4 BNC Analog MPX Outputs (2A & B outputs)

MPX codecs support analog MPX using BNC input and output connectors, or digital AES192 MPX inputs and outputs using RJ-45 connectors. Attach either analog or digital MPX connections to the codec and then adjust settings using the **MPX Settings panel**. MicroMPX encodes a full FM composite or MPX signal, including audio, stereo pilot and RDS, over a low bitrate connection. It currently supports bit-rates from 320 up to 900 kbit/s. Bit-rates down to 176 kbit/s are supported in MicroMPX+ mode. Note: MicroMPX+ is not currently supported.

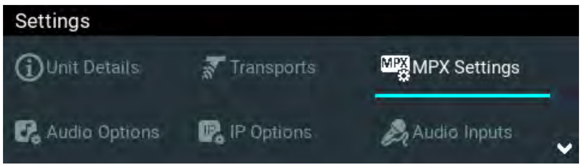
Important Notes:
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

Configure MPX Input (Encoding) and the Output Source

Important Note: Configure an MPX Encode program and load it in the codec as this will configure the **MPX Settings panel** for encoding MPX.

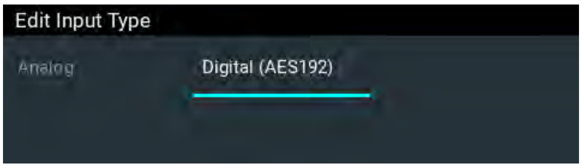
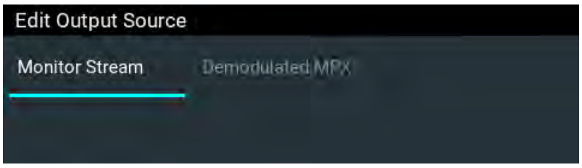
To configure the codec for MPX encoding:

1. Attach either an analog or digital MPX signal to the codec.
2. Press the **SETTINGS** button.
3. Navigate to **MPX Settings** and press the button.



4. Navigate to **Gain** and use the left and right navigation buttons to adjust levels up and down.

5. A summary of the settings available is outlined in the following table.

	Feature	Description
1	Input Type	Select either **Analog** or **Digital (AES192)**. The signal input level needs to be at exactly 0 dB for MicroMPX to encode correctly. Adjust the input gain until the signal peaks on the meter at 0dB. 
2	Output Source	Configure the preference for monitoring either **Demodulated MPX** audio locally, or an incoming remote codec IP audio **Monitor Stream**.  Note: A remote MPX codec will normally feed back IP audio from an FM monitor at the transmitter site to ensure audio transmissions are maintained.
3	GPI Encode Delay	Configure the GPI encode delay required to match audio latency as required.
4	µMPX Version	There are 4 different versions of µMPX: • Version 1: supports bit-rates from 320 up to 400 kbit/s. • Version 2 and 3: support bit-rates from 320 upto 576 kbit/s. • Version 4: supports bit-rates from 320 up to 900 kbit/s, and down to 176 kbit/s in MicroMPX+ mode.
5	µMPX Bitrate	Select a bit-rate that can be supported by the IP transport being used. When bandwidth is limited, 320 kbit/s is recommended as the minimum bit-rate as to use if possible.
6	µMPX Reset Interval	This setting determines how often a keyframe packet is sent.
7	µMPX Mono Mode	Press the OK button to enable this setting and force the codec to encode in mono. This will reduce the bit-rate requirement and increase the quality of the signal. This is useful when available network bandwidth is limited, and in situations where content is mainly mono, e.g. talk radio stations.
8	µMPX FEC	Press the OK button to toggle enabling and disabling FEC and then configure **µMPX FEC Delay** and **µMPX FEC Overhead** settings. MicroMPX sends about 94 packets per second not counting recovery packets. If you want to keep your delay to at most 1 second, you can set the delay to around 64 packets. A setting of 64 (delay) to 8 (error correction overhead) means decoder latency must be at least 0.7 seconds approximately. With this configuration every 64 packets the codec will send 8 extra packets as overhead. As long as at least 64 of the resulting 72 packets arrive, the decoder can reconstruct all missing packets. If latency is not an issue the delay and decoder latency can be increased as required.

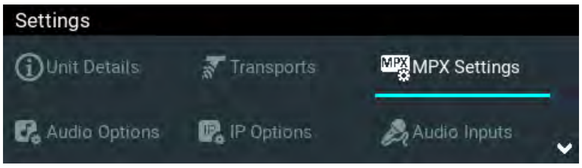
9	Stop on Pilot Loss	Press the OK button to configure the codec to stop encoding if the stereo pilot is lost. Configure the **Pilot Loss Threshold** and **Pilot Loss Timeframe** as required. This can trigger an alarm at the encoder or decoder as required.
10	Stop on Input Loss	Press the OK button to configure the codec to stop encoding if the MPX input is lost. Configure the **Input Loss Threshold** and **Input Loss Timeframe** as required. This can trigger an alarm at the encoder or decoder as required.
11	Stop on AES192 Input Loss	Press the OK button to configure the codec to stop encoding if the AES192 input is lost. Configure the **AES192 Input Loss Timeframe** as required.This can trigger an alarm at the encoder or decoder as required.

Configure MPX Output (Decoding) and the Output Source

Important Note: Configure an MPX Decode program and load it in the codec as this will configure the **MPX Settings panel** for decoding MPX.

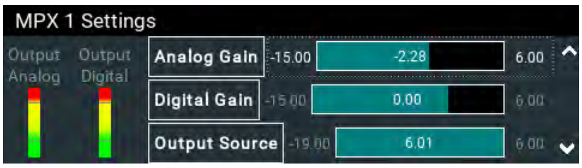
To configure the codec for MPX decoding:

1. Load an MPX Decode program in the codec.
2. Press the **SETTINGS** button.
3. Navigate to **MPX Settings** and press the OK button.

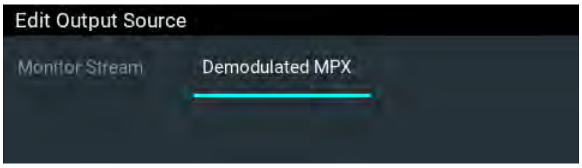


4. Navigate up to **Ref Level** to adjust the dBu reference level to match the expected reference level of devices fed from the analog MPX output.

5. Select **Source 1 Gain** or **Source 2 Gain** and adjust the level as required.
6. Navigate down to **Analog Gain** or **Digital Gain** and use the left and right navigation buttons to adjust levels up and down.



7. Navigate to **Output Source** and press the OK button to select either **Monitor Stream** or **Demodulated MPX**.
8. A summary of the decoder settings available is outlined in the following table.

	Feature	Description
1	Output Source	Configure the preference for monitoring either **Demodulated MPX** audio locally, or an incoming remote codec IP audio **Monitor Stream**. 
2	Source Switch Trigger	Press the button to select a local or remote GPI that will act as the trigger to switch between output sources.
3	Output Delay Source 1	Press the button to configure the output delay for Audio Source 1 in milliseconds. This setting relates to the MPX Source 1 in the **Output Audio Source** section of the **Program Manager panel**. It also configures the buffer setting for the decoded signal. Note: The maximum buffer setting for RAW uncompressed MPX is 1,000ms.
4	Output Delay Source 2	Press the button to configure the output delay for Audio Source 2 in milliseconds. This setting relates to the MPX Source 2 in the **Output Audio Source** section of the **Program Manager panel**. It also configures the buffer setting for the decoded signal. Note: The maximum buffer setting for RAW uncompressed MPX is 1,000ms.
5	Decode RDS Replacement	Press the button to enable insertion of an alternative RDS stream instead of using decoded RDS data.
6	Decode Loss Detection	Press the button to enable the codec to detect when the decoder stops decoding (potentially from loss of network or encoder). Configure the **Decode Loss Timeframe** and **Decode Resume Timeframe** as required. Note: If the decoder buffer remains empty for the given timeframe, the codec will set the AES192 output valid bit to false.

11 Configure Analog/AES Inputs and Outputs

Product	Feature	Notification
MPX I	Analog / AES3 Inputs / Outputs (1 Shared Stereo Analog / AES3 in/out)	**ANALOG/AES3 IN** RJ-45 connector provides shared stereo analog line input, or stereo AES3 digital input. **ANALOG/AES3 OUT** RJ-45 connector provides shared stereo analog line output, or stereo AES3 digital output.
MPX II	Analog / AES3 Inputs / Outputs (2 Shared Stereo Analog / AES3 in/out)	**ANALOG/AES3 IN** RJ-45 connectors provide shared stereo analog line inputs 1 and 2, or stereo AES3 digital inputs 1 and 2. Balanced **ANALOG/AES3 OUT** RJ-45 connectors provide shared stereo analog line outputs 1 and 2, or stereo AES3 digital outputs 1 and 2.

The codecs features stereo analog or AES inputs and outputs on the rear panel of the codec. These inputs for regular IP streaming connections and for an audio source that is used to send MPEG-TS, which uses MP2 encoding and is then sent over a sessionless MPEG-TS connection.

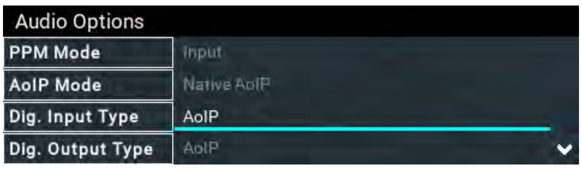
AES3 (AES/EBU) inputs are balanced 110 ohm inputs that can operate effectively over distances of up to 100 meters and accept both mono and stereo AES3 signals.

Important Note: MPX codecs support analog MPX using BNC input and output connectors, or digital AES192 MPX inputs and outputs using RJ-45 connectors. Attach either analog or digital MPX connections to the codec and then adjust settings using the **MPX Settings panel**. See Configuring MPX Inputs and Outputs for more info.

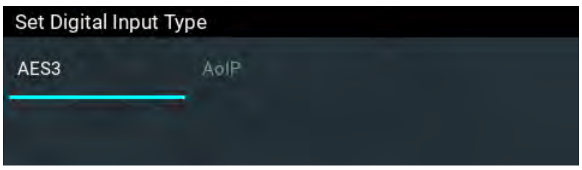
Configuring AES3 Inputs and Outputs Globally

Analog or digital audio is supported via RJ-45 connectors. To set up codec inputs and outputs for digital AES3 requires two settings to be configured. First, the codec needs to be configured globally for AES3 input and output audio, instead of digital AoIP 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.



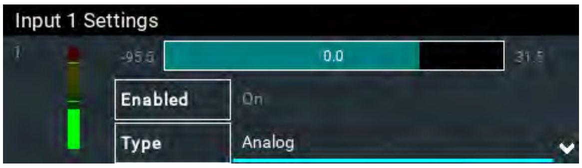
4. Select **AES3** and press the button.



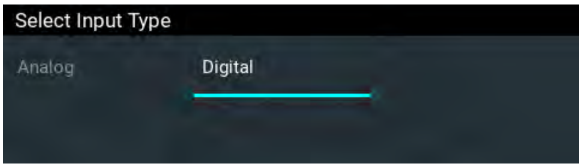
Configuring Inputs for Digital or Analog Audio

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. Inputs and outputs are configured in pairs for analog or digital audio:

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



4. Select **Digital** or **Analog** and press the button.



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.

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.

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 AoIP Clock
AES67/ST2110-30/Livewire derives the sampling clock from a primary leader clock over the network or a Livewire clock. This clock can be selected as the **AES3 Output Clock Source**. The **AES3 Output Clock Source** uses the **Lock to AoIP Clock** setting if either the **Digital Input Type** or **Digital Output Type** is set to **AoIP**. 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.

12 Input Levels and PPMs

Product	Feature	Notification
MPX I	PPM Meters	4 PPMs on front panel display inputs/outputs, encoders/decoders, or MPX+Monitor
MPX I	Inputs	2 Channels can be configured as analog or digital AES3 in/out
MPX II	PPM Meters	8 PPMs on front panel can display inputs/outputs, encoders/decoders, or MPX+Monitor
MPX II	Inputs	4 Channels can be configured as analog or digital AES3 inputs in pairs

Important Note: See Configure Analog/AES Inputs and Outputs for more information about digital in/outs. Input audio functions can also be configured using the Toolbox Web-GUI; see Configuring Inputs and Reference Levels for more information. See Configuring MPX Inputs and Outputs for configuring MPX signal inputs and outputs.

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 for non-MPX IP stream connections. 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.

The audio reference level settings in the codec are:

	Reference Level	Description	dBu	dBFS
1	Tieline G6 (Gateway, Gateway 4, MPX, Bridge-IT II, Bridge-IT XTRA II)	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, ViA)	PPM meter low point	-16dBu	-38dBFS
		Nominal 0vu reference level	+4dBu	-18dBFS
		Level at which audio will clip/distort	+22dBu	0dBFS
3	Tieline G3	PPM meter low point	-11dBu	-29dBFS

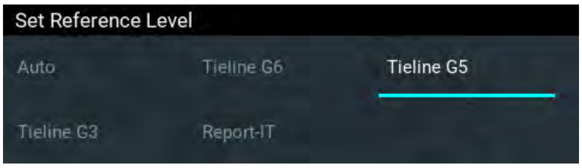
	(Commander and i-Mix)	Nominal 0vu reference level	+4dBu	-14dBFS
		Level at which audio will clip/distort	+18dBu	0dBFS
4	Report-IT	PPM meter low point	-9dBu	-23dBFS
		Nominal 0vu reference level	+4dBu	-10dBFS
		Level at which audio will clip/distort	+14dBu	0dBFS

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 button.
3. Navigate to **Reference Level** and press the button.
4. Select Tieline G5 and press the 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 button.
3. Navigate to **Reference Level** and press the button.
4. Select **Tieline G3** and press .

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 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.

PPM Metering for IP Connections

The codec LCD screen displays PPMs indications for both MPX and non-MPX IP audio streams. PPM metering options can be adjusted via **Settings > Audio Options > PPM Mode**. Note: The MPX I has 2 inputs, 2 outputs, 2 encoders, 2 decoders, and 4 PPMs only. The MPX II has has 4 inputs, 4 outputs, 4 encoders, 4 decoders, and 8 PPMs only.

MPX I

	PPM Mode	MPX I PPM Mapping Description
1	Input/Output (default)	Maps inputs 1 and 2 with PPM meters 1 and 2; maps outputs 1 and 2 with PPM meters 3 and 4
2	Encoder/Decoder	Maps encoders 1 and 2 with PPM meters 1 and 2; maps decoders 1 and 2 with PPM meters 3 and 4

MPX II

	PPM Mode	MPX II PPM Mapping 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 encoders 1 to 4 with PPM meters 1 to 4; maps decoders 1 to 4 with PPM meters 5 to 8

PPM Metering for MPX and Monitoring Connections

In the following table the PPM metering for MPX and associated confidence monitoring connections is summarized. In addition, the table includes how PPM metering displays when a mix of MPX and non-MPX connections is configured. For more information on MPX Inputs see Configuring MPX Inputs and Outputs.

MPX Streams	PPM 1	PPM 2	PPM 3	PPM4	PPM 5	PPM 6	PPM 7	PPM 8
1 x MPX	MPX1 signal		Monitor 1L	Monitor 1R				
2 x MPX Stream	MPX1 signal		Monitor 1L	Monitor 1R	MPX2 signal		Monitor 2L	Monitor 2R
1 x MPX + 1 x Stereo IP Audio	MPX1 signal		Monitor 1L	Monitor 1R	Input / Encoder 1L	Input / Encoder 1R	Output / Decoder 1L	Output / Decoder 1R
1 x IP Audio + 1 x MPX	Input / Encoder 1L	Input / Encoder 1R	Output / Decoder 1L	Output / Decoder 1 R	MPX1 signal		Monitor 1L	Monitor 1R

Important Note: When not sending an MPX signal there is a 19kHz pilot tone displayed on the PPMs. The decoder will generate this when the MPX signal is lost. The exact level will depend on the level that was being received before the stream was lost, but typically this will be about 9% of full-scale, i.e. -20.92 dBFS.

Adjusting Input Settings

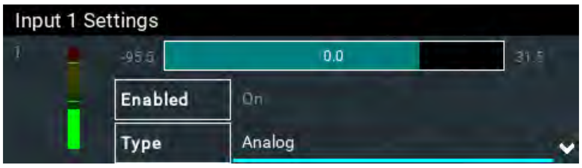
1. Press the **SETTINGS** button.
2. Navigate to **Audio Inputs** and press .
3. Adjustable input settings include:

- Input on/off.
- Input level: ganging of inputs is not currently supported.
- Input Type: Analog (line level input), Digital (AES3 or AoIP).

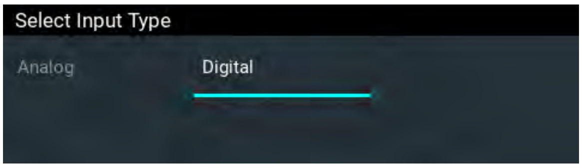
- IGC.
- Polarity/phase inversion
- Input Name

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 Notes:
- Gain adjustments can be made in 0.5dB increments.
- There is a maximum of 6dB of additional gain available when adjusting a digital input.

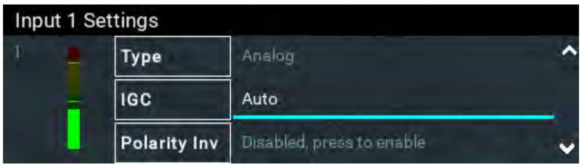
Intelligent Gain Control (IGC)

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.

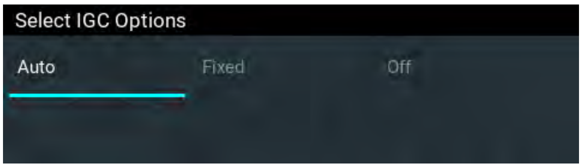
There are three settings; **Auto**, **Fixed** and **Off**. If **Auto** is configured the codec will detect when incoming audio levels have reduced sufficiently and automatically return input levels to the gain setting prior to IGC being activated. The codec takes just 250 milliseconds to detect audio 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 .
3. Select an input to adjust and press .
4. Navigate to **IGC** and press .



5. Select the preferred setting and press .



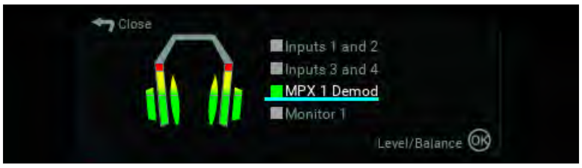
13 Headphone Output

The codec has a 6.35mm (1/4") RTS stereo **HEADPHONE** output on the front panel. IP audio streams and analog inputs or digital inputs can be selected and monitored. It is also possible to select and monitor a demodulated MPX signal in an encode or decode program.

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 the preferred inputs, MPX audio stream, or demodulated MPX monitor stream connection.



Important Note: Demodulated MPX audio cannot be monitored using the **HEADPHONE** output until the program is connected.

Codec Model Note: The number of inputs and 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.

14 Inserting a Satellite Module

A single module slot on the codec rear panel supports an optional satellite tuner card with MPEG-TS and MPE support which can receive DVB-S or DVB-S2 signals.

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. Launch the HTML5 Web-GUI and navigate to **Transport > Satellite Tuner Status panel** and verify that the DVB Satellite Receiver is locked with appropriate signal strength etc.

Important Note: If the module does not appear to be receiving a signal then it is possible that the connector on the module has not lined up correctly with the connector inside the codec. Power the codec down, then remove the module and reinsert it carefully to resolve this issue.

15 IP Streaming, Programs, and TieLink

Tieline codecs support high quality IP 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. Sessionless: The codec does not send session data when attempting to connect. Requires the "send" audio port and "return" audio port to be configured.

Connections that use either uncompressed MPX or compressed µMPX are sessionless connections.

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. A monitoring audio stream can also be configured to use Tieline session data to make it easier to connect and answer these connections.

Programs

A **Program** configures a Tieline codec to send or receive one or more **Audio Streams** or MPX signals based upon the particular application the codec is being used for at any given time. The attributes of the audio or MPX connection are embodied within a program when it is created, including the configuration, dialing and answering parameters. Tieline codecs can send proprietary session data over non-MPX connections in order to establish, manage and terminate connections. In this situation, 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, non-MPX connections within 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.

Program Types

Only simple peer-to-peer (point-to-point) non-MPX audio stream connections can be created using the codec front panel. The HTML5 Toolbox Web-GUI contains a **Program Manager panel** with a wizard for configuring program settings and backup connections. There are two program types:

1. User Defined Programs: Configure MPX, MPX monitoring, mono, stereo, multicast and multi-unicast connections in any order without selecting a standard template option. Note: Options available depend on how many channels the codec supports.

2. Standard Template Programs:Configure connections by selecting a standard program template and then remove or add streams and adjust configuration settings as required.

Defining Audio Streams within Programs

Each MPX connection or IP 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.

Multi-Unicast Programs

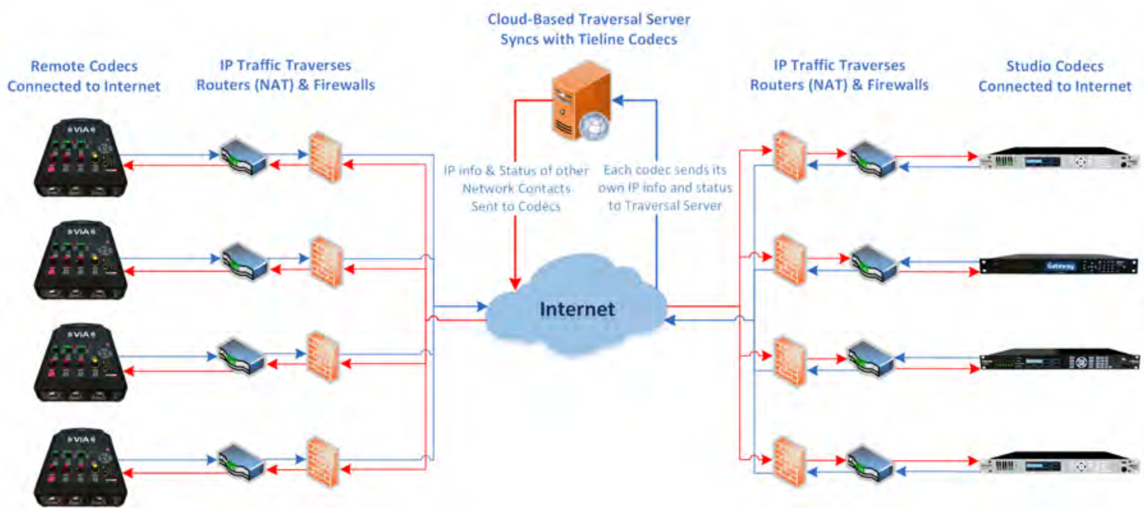
A multi-unicast program (also known as multiple unicast) can transmit a single MPX signal or IP audio stream to several different endpoints. All connections that are part of a multi-unicast connection will automatically attempt to reconnect if they are terminated remotely. Multi-unicasts can only be 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.

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.

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 MPX, Bridge-IT or G3 codecs.



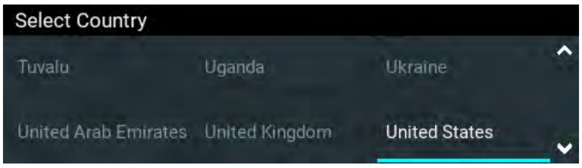
16 Configure Language and Country

Configure Country Setting

To configure the country setting in the codec:

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

4. Use the navigation buttons to select your country of operation and press the button.



Important Note: The country setting affects whether G.711 µ-Law (North America/Japan) or A-Law (Europe/Australasia) coding is used over IP 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 button.
3. Navigate to **Language** and press the button.
4. Use the navigation buttons to select a preferred language and press the button.

17 Connecting Quickly

Product	Feature	Notification
MPX I	Programs	Supports program connections using up to 2 channels of audio
MPX II	Programs	Supports program connections using up to 4 channels of audio

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 button.
   iii. Navigate to **Region** and press the button.
   iv. Use the navigation buttons to select your country of operation and press the 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) is used over IP connections.

Creating a New Connection Program

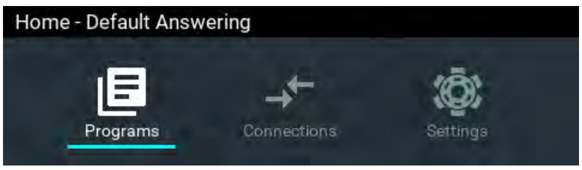
The codec supports a large number of different connection programs. To load a program it is necessary to launch the Toolbox HTML5 Web-GUI and perform the following steps:

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.
   - Select a program from the list of program types available.
3. Configure and save the new program using the procedures outlined in Configuring IP Streaming Programs, or the following sections About MPX Connections.

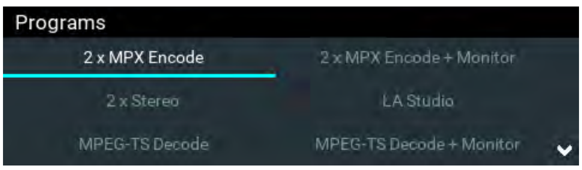
Load and Connect a Program

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

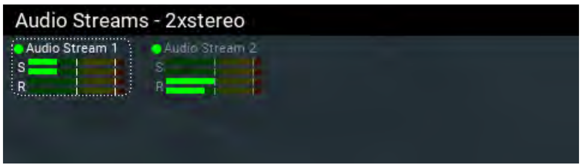
1. Press the **HOME** button to return to the **Home** screen.
2. Use the navigation buttons to select **Programs** and press the button.



3. 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.



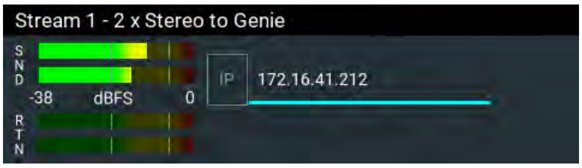
Edit the IP Address in a Loaded Program

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.

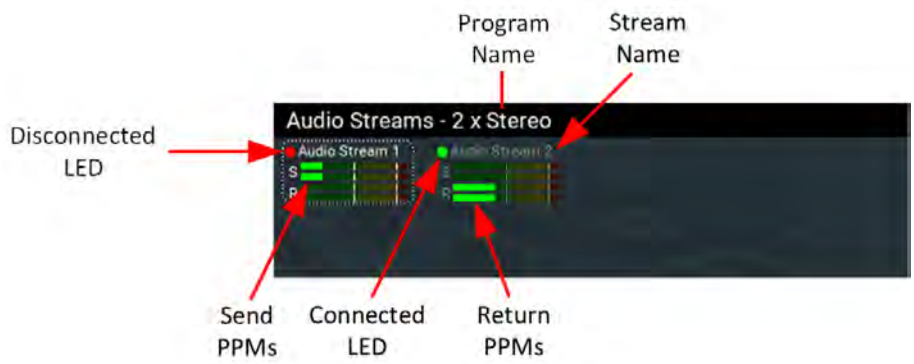
3. Select a disconnected stream and press the button.



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.

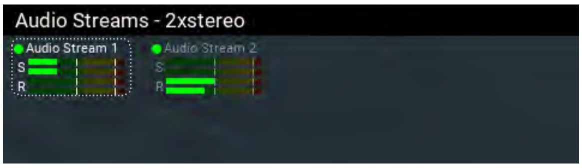
17.1 Monitoring Connections

Connections Screen Explained



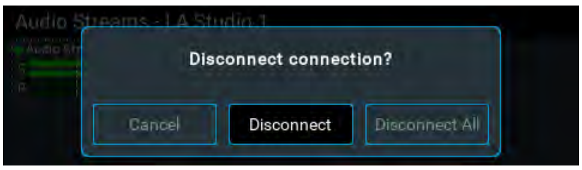
Connection status can be monitored using the **Connections** screen.

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

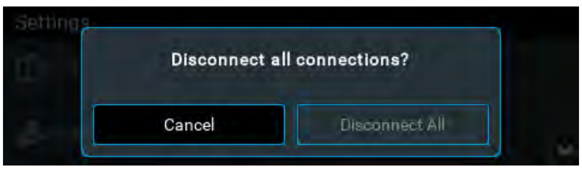


17.2 Disconnecting a Connection

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.

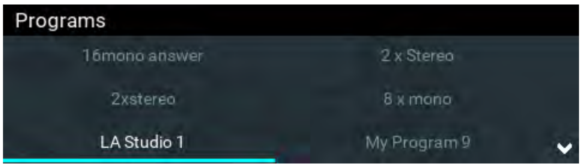


Note: Press the red **DISCONNECT** button on the numeric **KEYPAD** from other menus at any time to disconnect all active connections.

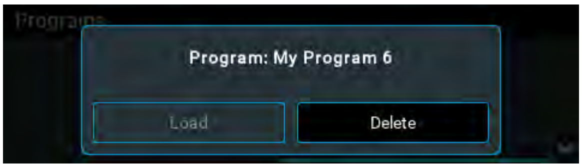


17.3 Deleting Programs

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.

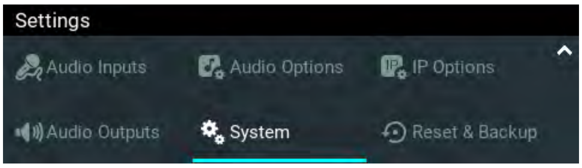


5. Confirm the deletion and press the button.

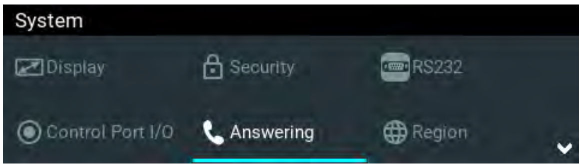
17.4 Lock or Unlock a Program in the Codec

It is possible to lock a customized single stream program, or a multistream program in a codec, to ensure the currently loaded program type cannot be unloaded by a codec dialing in with a different program type. It is essential to lock multistream 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 to lock a program in the answering codec is to always use a particular jitter buffer or FEC setting.

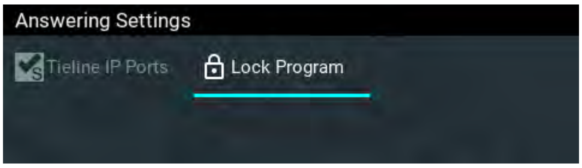
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.



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.

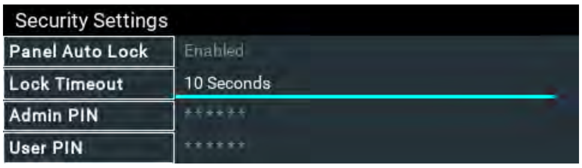
17.5 Locking the Front Panel

The codec features a front panel lock feature for tamper-proof operation which is disabled by default. There are two levels of panel lock and each requires a user to enter a PIN to access different features:

1. **Admin PIN**: Required to change codec connection or configuration settings accessed via **SETTINGS** button. (Default PIN is: 456789)
2. **User PIN**: Required to use the codec front panel buttons and dial/hangup a connection (Default PIN is: 123456)

Enabling the Front Panel Lock Feature

1. Press the **SETTINGS** button.
2. Navigate to **System** and then **Security** and press .
3. Navigate to **Panel Auto Lock** and press to toggle from **Disabled** to **Enabled**.



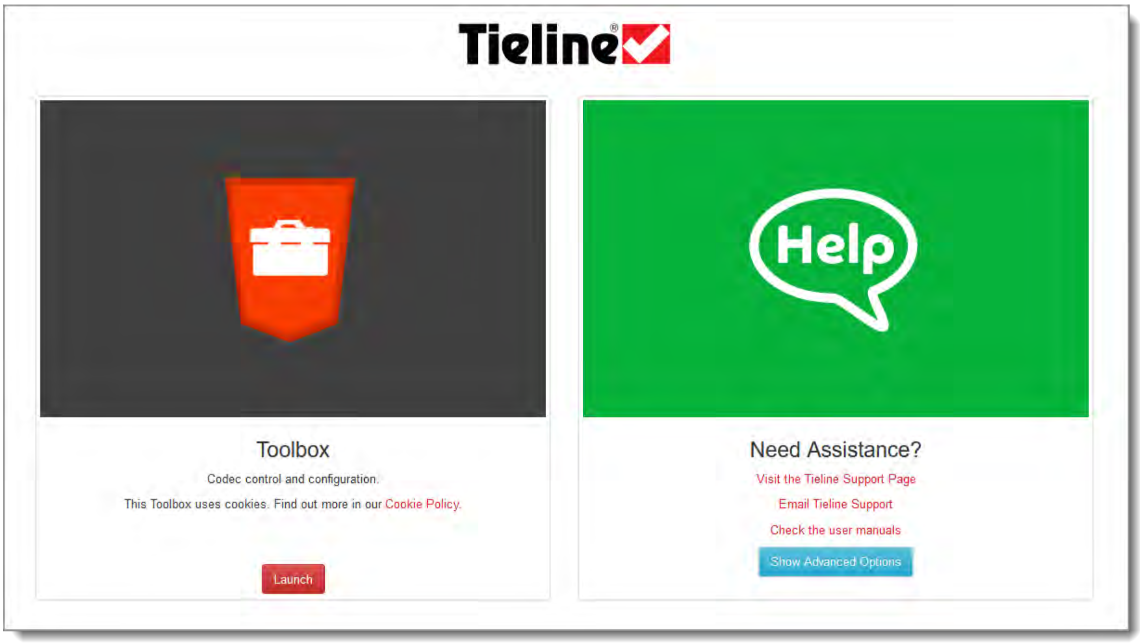
4. Navigate down to the panel **Lock Timeout** field and press to enter the desired timeout period in seconds. Note: The timeout period is the time in seconds before the codec front panel is relocked after being used.
5. If you want to change the default **Admin PIN** or **User PIN**, navigate down to each in turn and press to enter a new PIN.

18 Connecting to the ToolBox Web-GUI

The HTML5 Toolbox Web-GUI can fully configure codec settings, create dialing programs and dial, hangup and monitor connections. It improves the user experience with codec command and control and runs seamlessly on modern browsers. It runs on Mac, Windows and Linux computers.

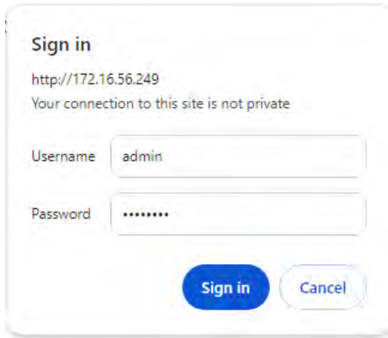
18.1 Opening the HTML5 Web-GUI & Login

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 default password immediately (see Changing the Default Password). This will provide better network security to maintain reliability during live broadcasts.



Launching the AoIP Web-GUI

To access the AoIP Web-GUI type the IP address of the AoIP interface into a browser. IP address details can be found in the codec via the **AoIP 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 AoIP/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.

Port Selection

By default port 80 is used by your PC to communicate with the codec and launch the Web-GUI. If port 80 cannot be used across your network for some reason, type the IP address of your codec into your browser with a full colon and the port number 8080. E.g. 192.168.0.176:8080

It is also possible to specify a different port for connecting the Toolbox web-GUI to your codec.

1. Press the **SETTINGS** button.
2. Use the navigation buttons to navigate down to **IP Options > WebGUI** and press the button.
3. Select **Alternate Port** and press .
4. Use the **KEYPAD** to enter a new port number and press the button to save the new setting.
5. Type the codec IP address into your browser with a full colon and then the new port number.

Note: Any new port specified must be within the range 2000 to 65535 inclusive.

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**.

18.2 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

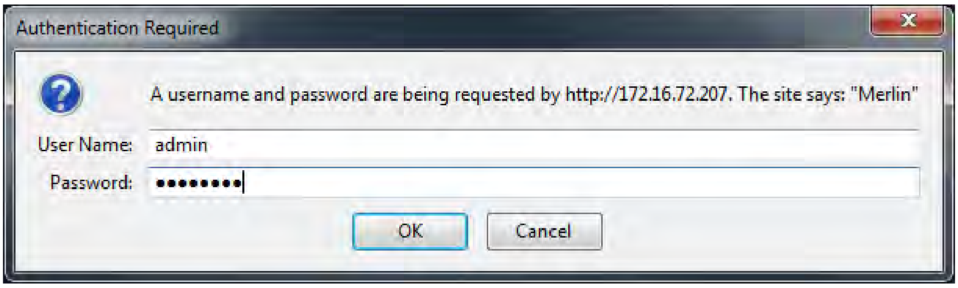
Adequate security is a major factor in ensuring your codecs and your broadcast network remain secure. There are several layers of security available in Tieline codecs to maintain secure connections. These include:

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. 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.
5. 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.
6. Implementation of CSRF protection (Cross-Site Request Forgery). Enable and disable this setting using the **Options panel** in the Toolbox HTML5 Web-GUI, 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.

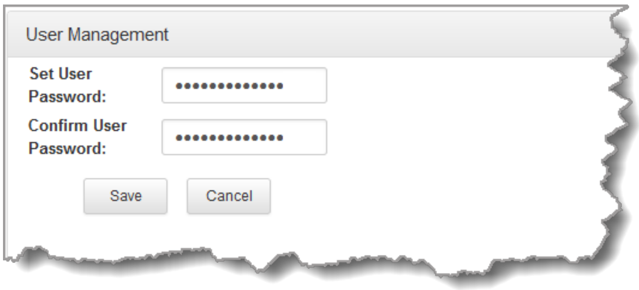
Important Note: The **Username** in the codec menu is permanently set to **admin** and cannot be changed; only the **Password** can be changed.

Changing the Password Using the Web-GUI

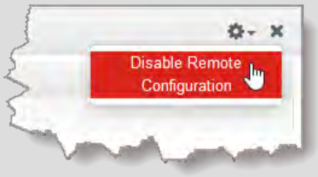
It is also possible to change the codec password with the **User Management panel** in the Toolbox Web-GUI. This feature is disabled by default. Use the codec front panel to enable this feature by navigating to **Settings > IP Options > Web GUI > Change Remotely** and then press the button.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **User Management** to display the **User Management panel**.

2. Click **Password** and then enter and confirm the new password, then click **Save**.



Important Note: This remote password change feature can only be enabled using the codec front panel. However, it can be disabled in the **User Management panel** by selecting the **Options symbol** and then **Disable Remote Configuration**.



19 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.

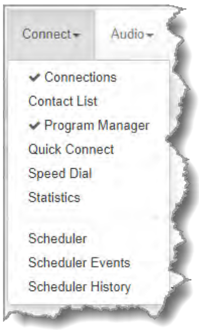
Important Note: The AoIP menu and associated panels are for managing AoIP streaming (e.g. AES67, Livewire+. Please see the AoIP Streaming Manual for configuration and streaming information.

Adjusting the Theme

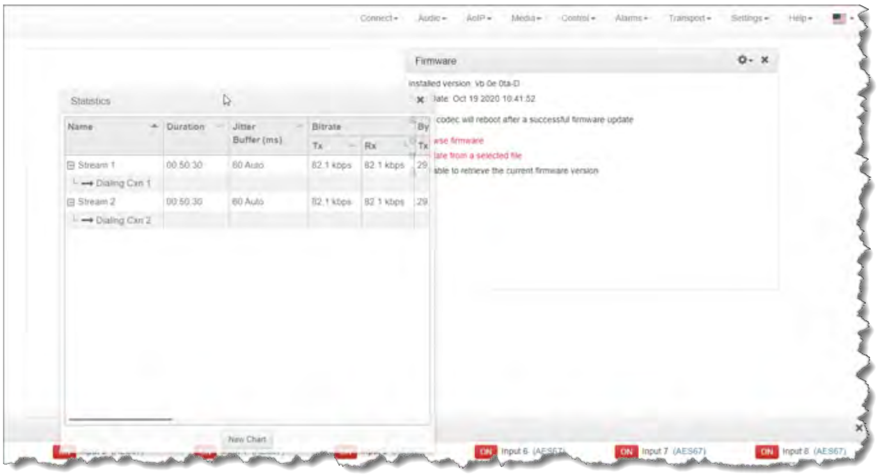
To adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI, navigate to the **Menu bar** at the top of the screen and click **Settings**, then click to select the preferred option. Note: this manual uses the **White** theme for most images.

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.

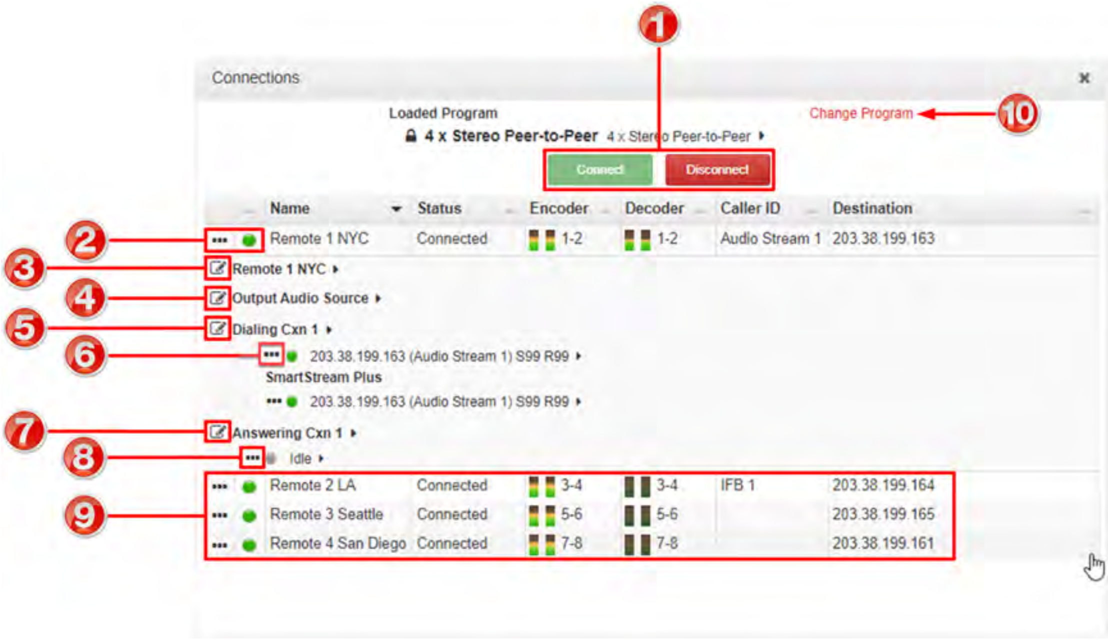


Position the mouse pointer over a panel's **Title bar** and click and drag to move a panel and reposition it in a preferred screen position.



Connect Panels: Load & Connect Programs & Manage Audio Streams

Connections Panel

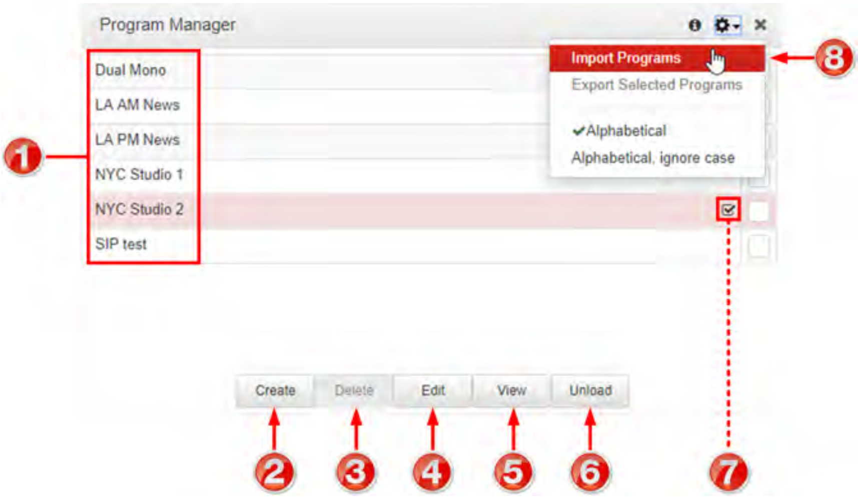


Note: The preceding image illustrates settings in the **Connections panel** only and may display connection options that are not available in all products, e.g. more connections than are supported.

	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 over stream details to expand stream information. Click again to collapse stream details. Click the **3 dot** ··· icon to connect or disconnect all connections in an audio stream. The **Connection State LED** displays status of an audio stream connection as follows: • Gray: Not in use • Green: Connected and stable (LQ greater than 70) • Yellow: Connected but connection quality is not stable (LQ 50 to 70) • Red: Connection establishing or problem with connection (LQ less than 50). LED flashes when establishing connection. Note: ISDN connections do not display the yellow LED state.
3	Audio Stream **Edit**	Click the **Edit** button to edit audio stream settings.
4	Output Audio Source **Edit**	Click the **Edit** button to edit backup audio settings to maintain program audio.
5	Connection **Edit** button (dialing connection)	Click the **Edit** symbol to edit audio stream settings, including the IP address.
6	Connection **Connect / Disconnect** button	Click the **3 dot** ··· icon to connect/disconnect an individual connection and adjust the bit-rate when a connection is active.
7	Answering connection **Edit** button	Click to edit answering connection settings
8	Answering Connection **Disconnect** button	Click to disconnect an answering connection

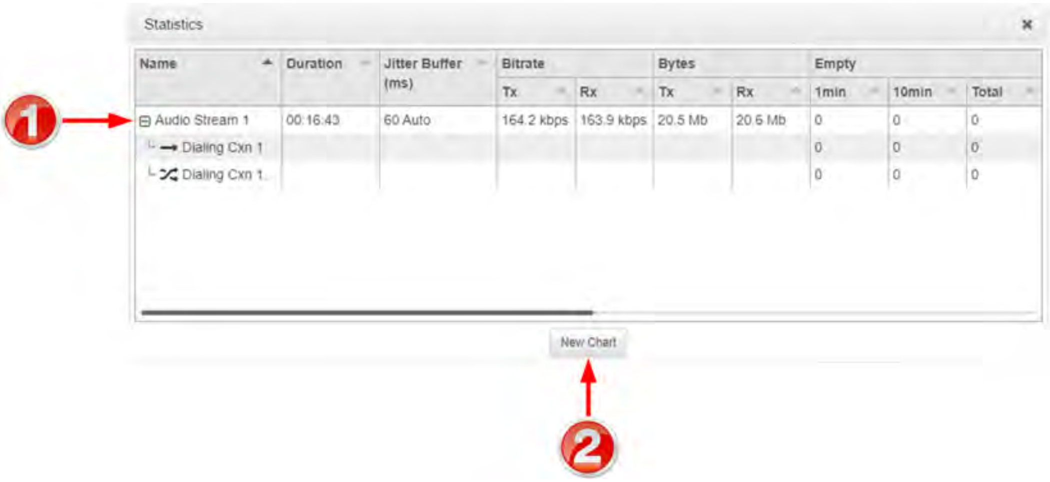
9	Additional Connections	Other connections listed
10	**Change Program**	Click to unload the current program and load a new program.

Program Manager Panel



	Feature	Description
1	Program list	The list of saved programs in the codec
2	Create New Program button	Click to create a new program using the program wizard.
3	Delete Selected Programs button	Click to delete all selected programs
4	Edit Selected Program button	Click to edit the selected program
5	View Selected Program button	Click to view configuration settings for a selected program
6	Unload/Load program button	Click to load or unload a program
7	Loaded program symbol	Symbol identifies the currently loaded program
8	Import Programs and Export Selected Programs	Click the **Options** menu to select and import previously saved programs or export selected programs as a .zip file

Statistics Panel

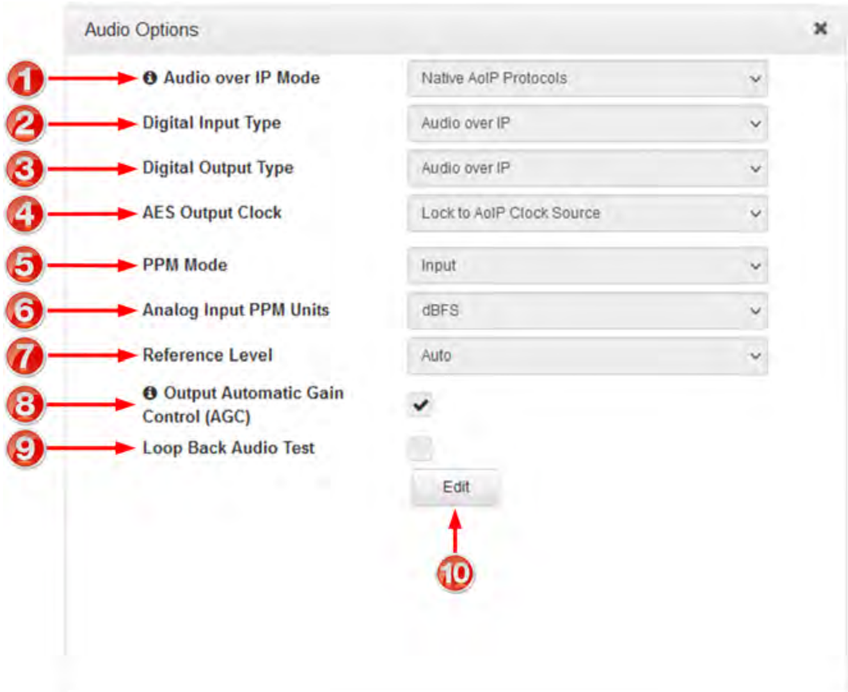


	Feature	Description
1	Expand/Collapse	Click to show/hide audio stream statistics, including packet arrival data info.
2	**New Chart** button	Click the **New Chart** button to select a **Data Series** and create a customized **Statistics panel**.

Audio Menu Panels

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

Audio Options



	Feature	Description
1	Audio over IP Mode	**Native AoIP Protocols** is the AoIP mode when streaming with natively supported protocols, e.g. AES67, ST 2110-30, Livewire+ and RAVENNA.
2	Digital Input Type	Select either **AES3** or **Audio over IP** as the **Digital Input Type** for the codec. This setting is global for all digital inputs.
3	Digital Output Type	Select either **AES3** or **Audio over IP** 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. AES67/ST2110-30/Livewire derives the sampling clock from a primary leader clock or Livewire clock over the network and the **AES3 Output Clock** source uses the **Lock to AoIP Clock Source** setting if either the **Digital Input Type** or **Digital Output Type** is set to **AoIP**.
5	PPM Mode	PPM metering can be adjusted to display inputs/outputs, encoders/decoders, or MPX+Demodulated MPX.
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.
8	Output Automatic Gain Control (AGC)	Check-box to enable/disable AGC. AGC is independent of IGC (Intelligent Gain Control) on each input.

9	Loop Back Audio Test	Check-box to enable/disable an input/output loopback test of audio. E.g. **Input 1** is routed to analog **Output 1**, **Input 2** is routed to **Output 2**.
10	**Edit** button	Click to edit settings in the **Audio Options panel**.

Headphones Panel

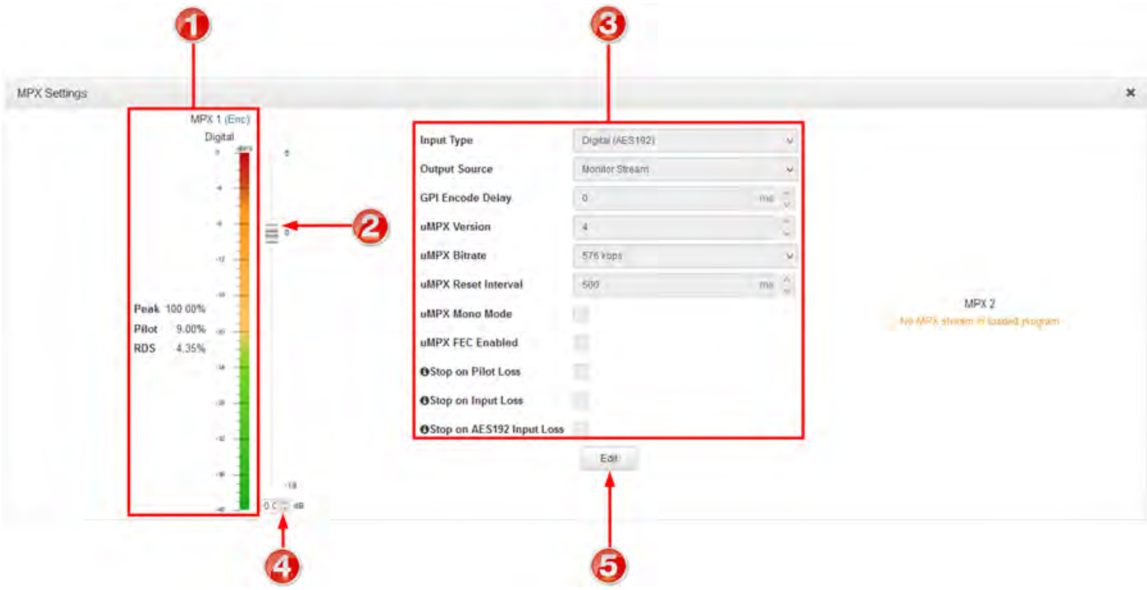
	Feature	Description
1	Headphone Volume Slider	Slider used to increase and decrease headphone output volume. Headphone gain expressed as a percentage. **WARNING: This may adjust the headphone volume of someone listening to a headphone output at a remote location. Extreme caution is recommended when this setting is enabled. High headphone levels can cause hearing loss!**
2	**Headphone Adjust Lock**	Click the **Headphone Adjust Lock symbol** to enable adjustment of headphone output volume. **WARNING: This may adjust the headphone volume of someone listening to a headphone output at a remote location. Extreme caution is recommended when this setting is enabled. High headphone levels can cause hearing loss!**
3	Headphone Level PPMs	View PPM levels of input/decoder audio fed to the headphone output when monitoring a non-MPX IP audio stream. Note: Demodulated MPX signals may peak near 0 dbFS on the **Headphone panel** PPMs but this is normal due to the nature of MPX signals which are heavily processed.
4	**Monitor Input**	Select sources to monitor outgoing and incoming audio, e.g. inputs, monitor streams, and demodulated MPX signals.
5	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 the headphone output. Monitor return audio when a demodulated MPX signal is monitored.

6	**Information symbol**	Hover over the **Information symbol** to view **Headphones panel** information
7	**Options** menu	Click the **Options symbol** to view timezone options and generate a PDF view, or enable/disable the scheduler
8	**Close** button	Click to close the panel

Inputs Panel

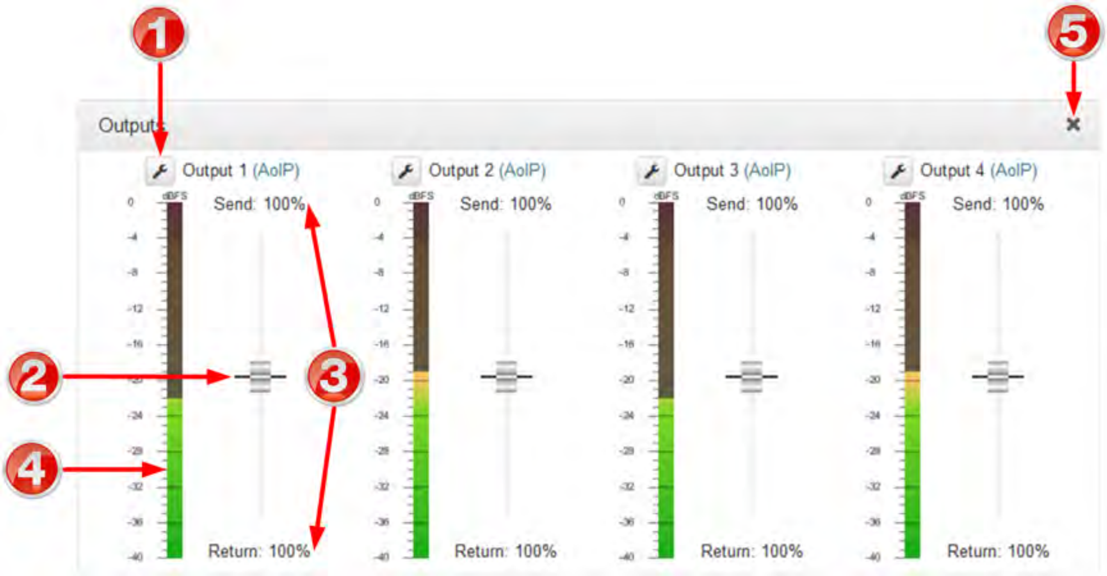
	Feature	Description
1	**Settings** button	Click to adjust input **Name**, **Type**, **Polarity Inverted** and **IGC** settings
2	**On/Off** button	Click to toggle an input on or off
3	Input Sliders/Faders	Input gain control sliders/faders. Adjust input gain in +/- 0.5dB increments
4	Input PPM meter	Input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

MPX Settings Panel



	Feature	Description
1	MPX PPM meter	Displays the level of the MPX signal on the PPM meter.
2	MPX Gain slider	Adjust the input gain slider until the signal peaks on the meter at 0dB. Note: The input level has to be at exactly 0 dB for MicroMPX to work correctly.
3	Input gain adjustment	Enter a dB value to adjust input gain in 0.5dB increments
4	MPX Configuration Settings	MPX configuration settings that are displayed in Encode and Decode mode. Load an MPX encode program to view encoding settings, or load a decode program to view decoding settings.
5	**Edit** button	Click to edit settings in the **MPX Settings panel**.

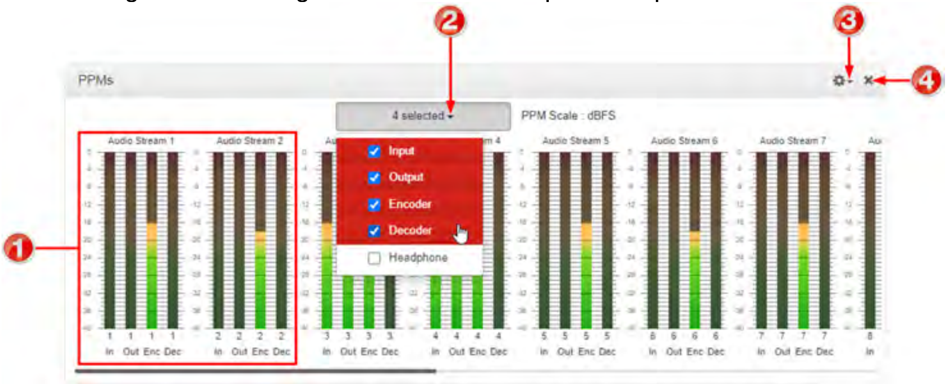
Outputs Panel



	Feature	Description
1	Settings	Select analog or digital as the output type. Note: configure the global digital output type via **Settings > 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	**Send/Return** indications	Balance of send and return audio included in the mix for each output (expressed as a percentage)
4	Output PPM meter	Output PPM meter
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 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, decoder and headphone.
4	**Close** button	Click to close the panel.

Media Menu

File Manager

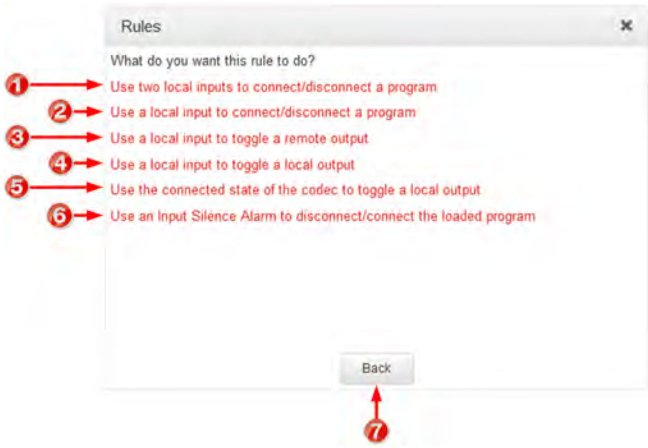
	Feature	Description
1	Folder and File View	View all folders and files on an external drive / removable media
2	**New Folder**	Select the drive or a folder and then click to create a new sub-folder
3	**Upload**	Click to select and upload a file onto removable media
4	**Download**	Click to download a file 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

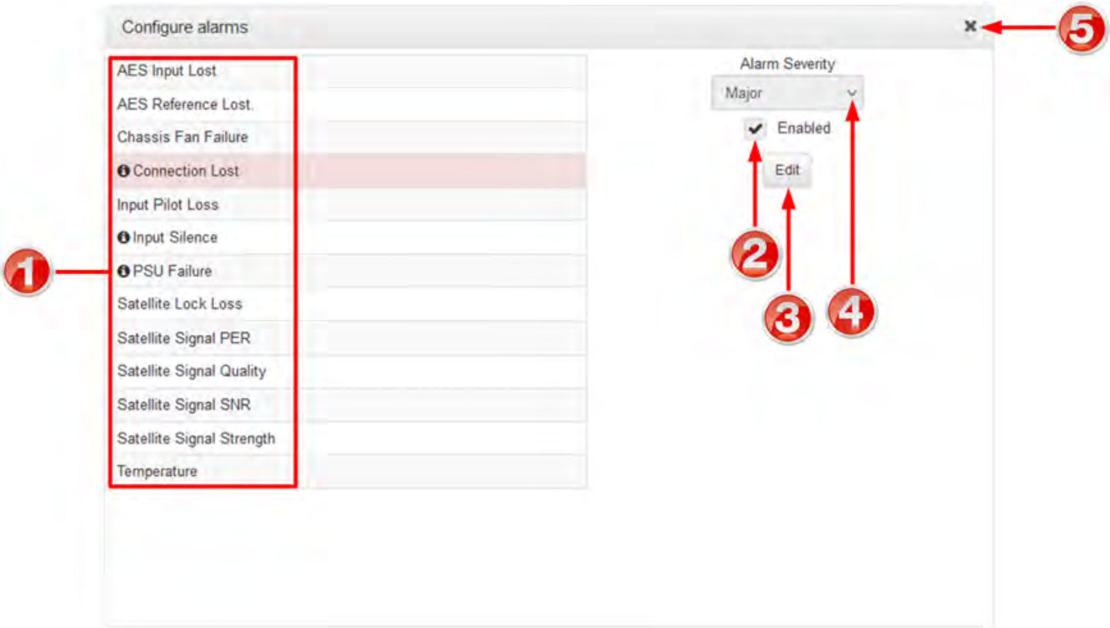
Rules Panel



	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	Use an Input Silence Alarm to disconnect/connect the loaded program	Click to configure a silence alarm to disconnect or connect the loaded program
7	**Back** / **Add New Rule** button	Click to add a new rule, or exit the rule creation function.

Alarms Panels: Configure & Monitor Alarms

Configure Alarms Panel



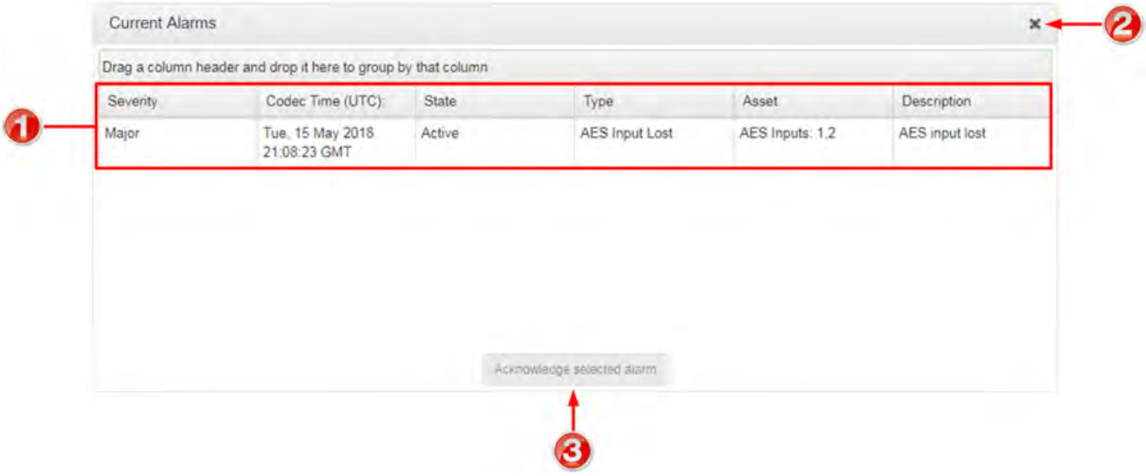
	Feature	Description
1	List of alarm types	Click to select an alarm type to configure.
2	Enable Alarm check-box	Click the **Enabled** check-box to enable the currently selected alarm.
3	**Edit** / **Save** button	Click to edit an alarm, or save configured alarm settings when in edit mode.
4	Alarm Severity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

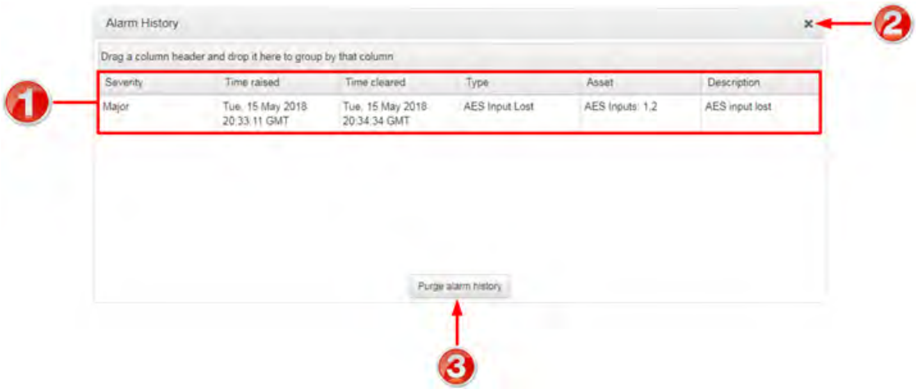
	Feature	Description
1	List of alarm severity levels	Click to select an alarm severity level to configure it.
2	**Front panel alarm & Toolbox** check-box	Select the check-box (default enabled) to deliver front panel **ALARM LED** notifications and HTML5 Toolbox Web-GUI alarm notifications.
3	**Close** 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.

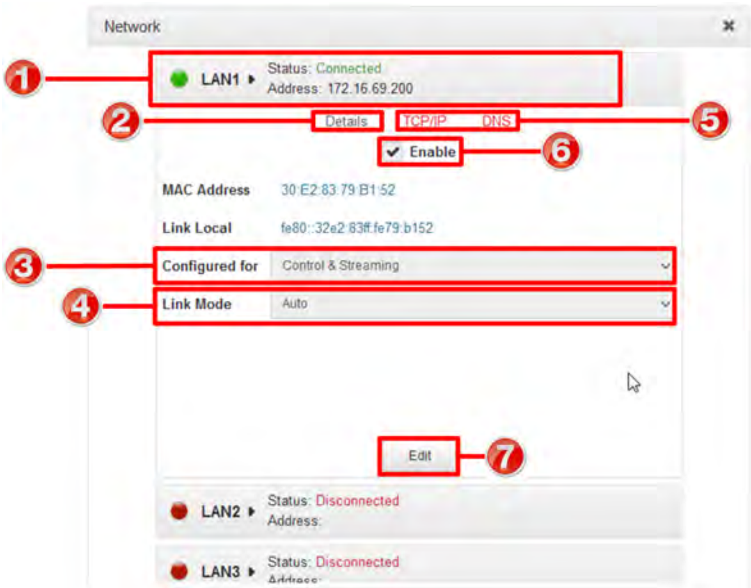
Alarm History

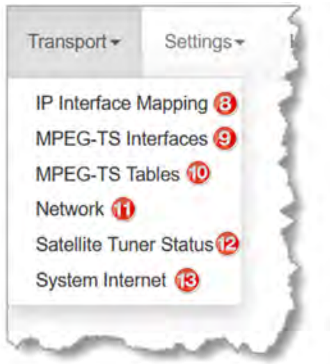


	Feature	Description
1	Alarm history description	View the history of previous alarms in the codec.
2	**Close** button	Click to close the **Alarms** panel.
3	**Purge Alarm History** button	Click to clear the alarm history.

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.





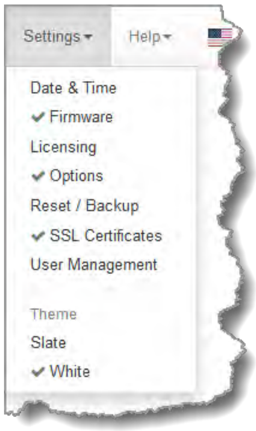
	Feature	Description
1	Network interface	Click to select, expand and view configuration, and edit network configuration settings for Ethernet/LAN and VLAN interfaces.
2	**Details** tab	Displays configuration options for the selected network interface, plus other device details.
3	Control/Streaming	Select **Control** and/or **Streaming** options for the selected interface.
4	**Link Mode**	Configure the Ethernet or VLAN 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/Cancel** button	Click **Save** to store settings, or click **Cancel** to revert to previously configured settings.
8	**IP Interface Mapping**	Configure default **Primary**, **Secondary** and **Tertiary** interfaces.
9	**MPEG-TS Interfaces**	This panel allows you to configure interfaces to receive MPEG-TS packets. These interfaces may include LAN1, LAN2 and the DVB Satellite Receiver when installed.
10	**MPEG-TS Tables**	This panel displays the contents of the DVB SI tables received on the enabled MPEG-TS interfaces.
11	**Network**	Click to open the **Network panel** and configure network settings.
12	**Satellite Tuner Status**	This panel displays the quality measurements and settings for the satellite signal currently being received.
13	**System Internet**	Click to configure the preference for interfaces used to connect to the internet. For example, when using DDNS services. Or perhaps where the default setting is LAN1, however this may be connected

		to a closed network and LAN2 is the interface connected to a WAN, i.e. the internet.

Settings Panels

There are several **Settings** panels which can be opened in the Web-GUI. Each panel provides different codec configuration settings and options. Click to select and open each panel. It is also possible to change the HTML5 Web-GUI theme.

As an example, the **Options panel** is displayed with a brief description of the other panels available.



Settings panels

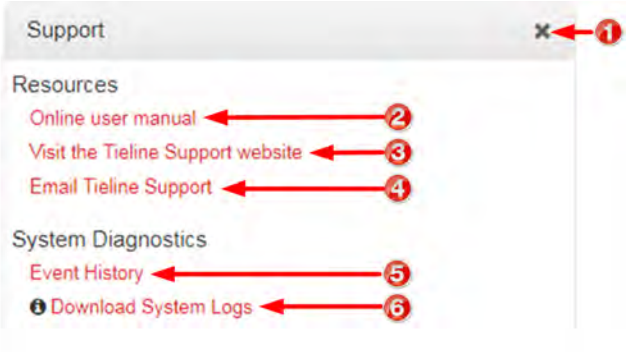
	Feature	Description
1	**Date and Time**	Click to open the panel view and sync the codec to NTP time.
2	**Firmware tab**	Click to open the panel; view software versions, download firmware and perform an upgrade.
3	**Licensing tab**	Click to open the panel; select a license file and install it in the codec.
4	**Options tab**	Click to open the panel and adjust a wide range of codec audio, firewall, RS232, SNMP, system, session data and Web-GUI settings.
5	**Reset / Backup**	Click to open the panel; reset codec default settings and perform backup/restore of codec programs and settings.
6	**SSL Certificates**	Click to open the panel and update digital SSL certificates remotely. Note: The first certificate must be installed using the front panel. Certificates cannot be updated if SSL is not enabled.
7	**User Management**	Click to open the panel and change the Web-GUI password. Note: This feature must first be enabled in the front panel via **Settings > IP Options > Web GUI > Change Remotely**.
8	**Theme**	Adjust the **Theme** or 'skin' of the HTML5 Toolbox Web-GUI; options include **White** or **Slate**.

Options panel

	Feature	Description
1	**DDNS** Settings	Configure Dynamic DNS to use a hostname to connect to a codec when using a dynamic IP address.
2	**Ember+** Settings	Enable and disable Ember+, adjust the **Local Port** and configure **Send Keep Alive**.
3	**Firewall** Settings	Enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. Ember+ can also be enabled.
4	**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**.
5	**SNMP** Settings	Configure SNMP settings in the codec.
6	**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.
7	**Tieline Session**	Edit the Tieline session and alternative session port used by the codec.
8	**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**.
9	**Edit button**	Press to edit settings in the **Options panel**.

Help Panels

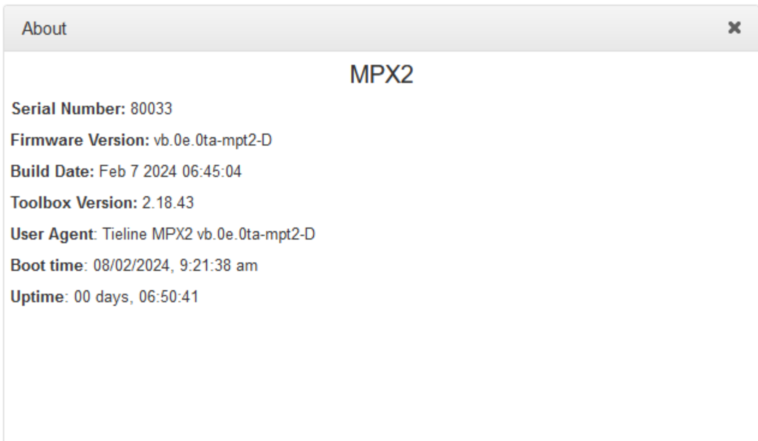
Support Panel



	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.



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.

19.1 Configuring IP Settings

Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **Network** to view and configure Ethernet and VLAN interface settings in the Web-GUI.

Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

IPv4 versus IPv6

An IP address is a unique address to identify a device on a TCP/IP network. Your codec uses dual IP protocol stacks to allow your codec to work on both IPv4 and IPv6 networks. Tieline codecs support both DHCP (default) IP addressing and static IP addresses for dialing IPv4 connection endpoints.

If you want to dial a codec with a public IP address you simply dial the IP address to connect. If you want to dial a codec with a private IP address you need to perform network address translation (NAT). NAT allows a single device, such as a broadband router, to act as an agent between the public internet and a local private LAN. Usually this will be set up at the studio end so you can dial into the studio from the remote codec.

Support for IPv6 connections allows you to use IPv6 infrastructure to connect to other codecs globally.

Configuring Ethernet Ports and VLANs

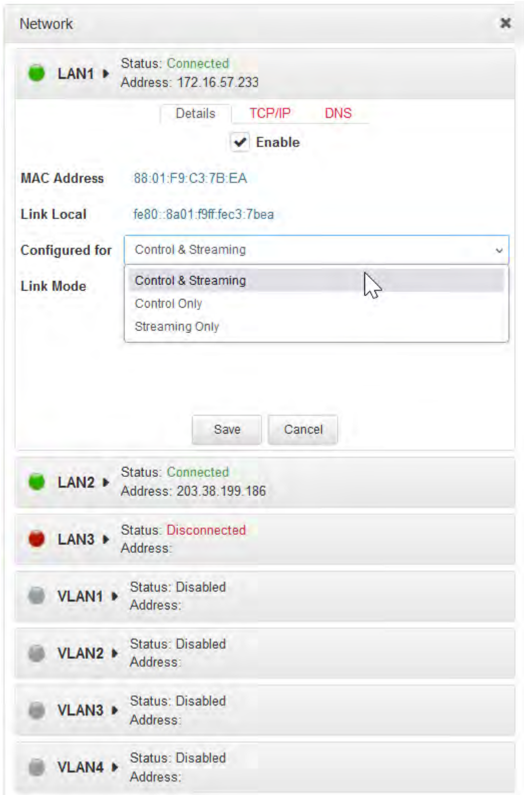
LAN1 and **LAN2** are physical Ethernet port interfaces and **LAN3** is an Ethernet port for control only. **LAN4** is configurable as either an additional Ethernet port, or as an AoIP port for AES67-ST2110-10 streams. Up to four additional VLAN interfaces can be configured. VLAN interfaces have features 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. In the **Details** tab click the **Edit** button.
3. Select the **Details** tab and configure control and streaming as required.
4. Click **Save**.



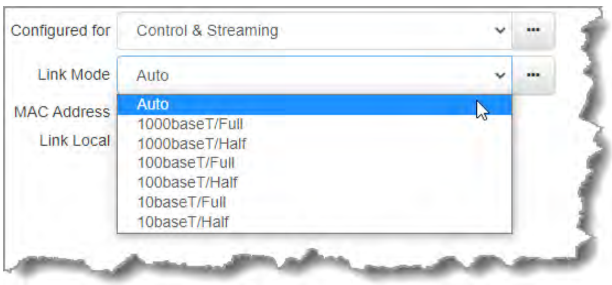
To enable or disable an interface select the **Enable** check-box when in **Edit** mode, then click **Save**. **LAN1** and **LAN2** are enabled by default. Please note that the **Status LED** color indicates the status of each network interface.

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.

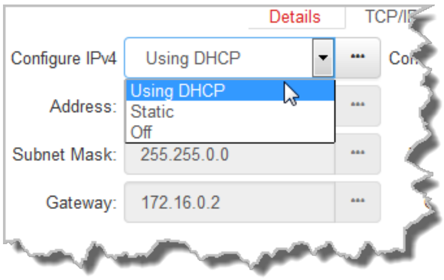
1. Click the drop-down **Link Mode** arrow and select the preferred setting.



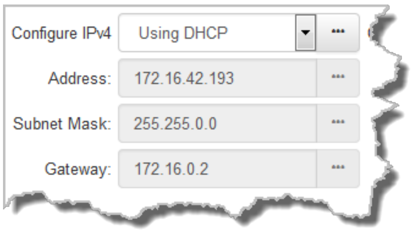
2. Click **Save** to store the new setting.

IPv4 Address Configuration

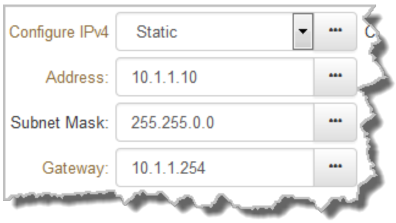
Click to select the **TCP/IP** tab in the **Network** panel to configure settings. The codec is capable of automatic DHCP address assignment, or manually configured static IPv4 address configuration via the **Configure IPv4** drop-down menu. To ignore IPv4 settings select **Off**.



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. Note: DNS settings must be specified when a static IP address is configured.



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.

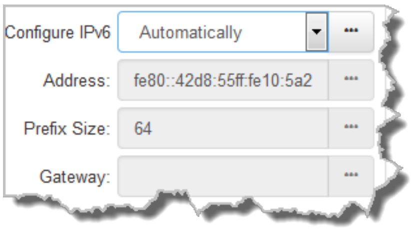
Types of IPv6 Addresses

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

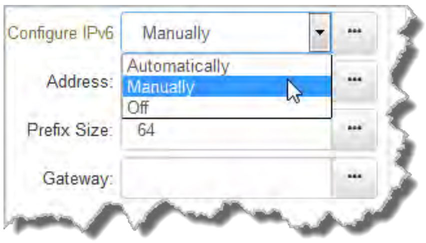
1. By default the codec is configured for connecting to an IPv6 router which automatically allocates IPv6 address details, as displayed in the following example.



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.

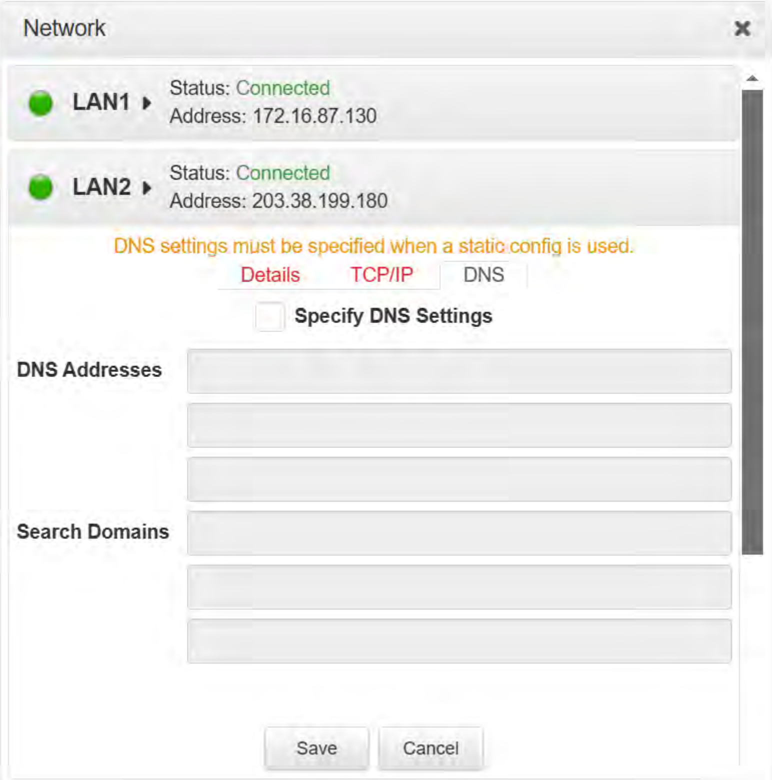


2. Click **Save** to store all configuration settings.

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**. Note: When installing codecs at the studio it is important to ensure that DNS settings are configured correctly. This is particularly important when using tools like the Cloud Codec Controller. Correct DNS settings are also required for NTP, DDNS, use of host names, and if the **Download and Install** option is used to upgrade codec firmware.

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 Interface 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 interfaces in order of use when available are:

1. **LAN1** Ethernet port (default **Primary** interface)
2. **LAN2** Ethernet port (default **Secondary** interface)
3. Internal Wi-Fi (default **Tertiary** interface)
4. **VLAN1** (default **Quaternary** interface)

If none of the primary, secondary, tertiary or quaternary interfaces are available, the codec will look for the first available interface in the following order:

1. LAN1
2. LAN2
3. Wi-Fi
4. VLANs

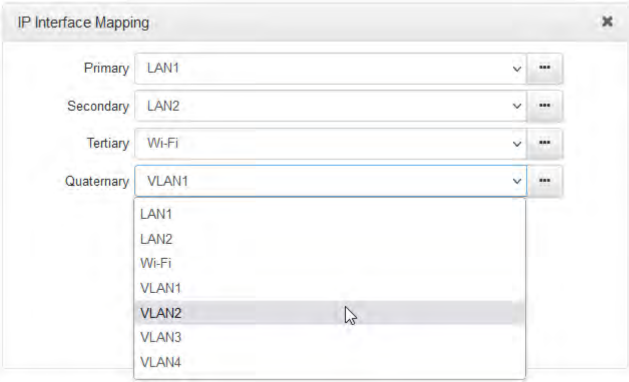
This is determined at the time of connection. For example, if **Any** is selected and VLAN1 is unavailable but VLAN2 is available, then VLAN2 will be used to connect. If a subsequent connection is dialed and VLAN1 becomes available then VLAN1 will be used for the second connection as it has higher priority.

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 Interfaces

It is possible to reconfigure the default **Primary** (Ethernet 1), **Secondary** (Ethernet 2), **Tertiary** (Wi-Fi) and **Quaternary** (VLAN1) IP 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. Open the HTML5 Toolbox Web-GUI and click **Transport** and then click **IP Interface Mapping** to open this panel.
2. Click the drop-down arrow for each interface to select the preferred default setting.

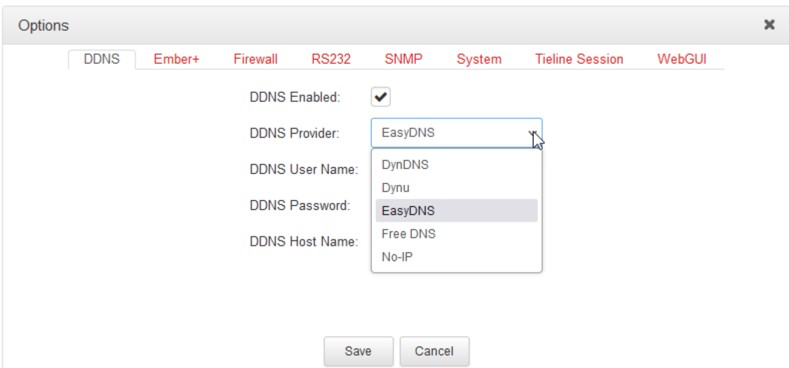


3. Click **Save** to store the configuration.

Configure Dynamic DNS (DDNS)

Dynamic DNS can be configured in the codec to allow the use of a host name to connect when using a dynamic IP address. To facilitate this a DDNS service is used to register a device's IP address to a host name, allowing a remote codec to dial to a host name. The host name will periodically be updated automatically as required, or when the dynamic IP address changes.

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 **DDNS** and then click **Edit** to adjust settings.
3. Click **Save** to store the new configuration settings.



Important Note:

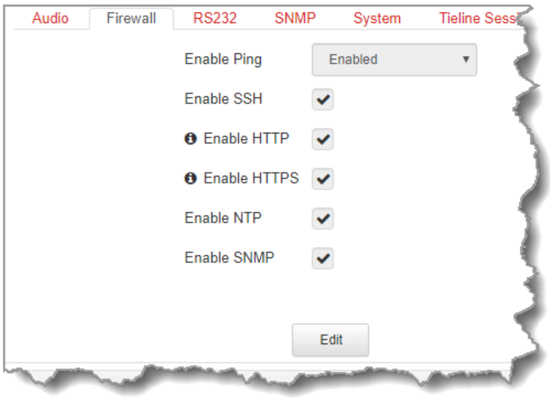
- Supported DDNS providers are listed in the **DDNS Provider** drop-down menu.
- Devices should be DDNS registered to public IP addresses.

- The codec will utilize the **System Internet** interface order when contacting DDNS service providers.
- Does not support connection to remote G3 codecs.
- DDNS Host Name settings are unrelated to the **Hostname** setting accessed via the **Options panel** under **Settings**.
- Codec DNS settings must be specified when configuring DDNS.

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.

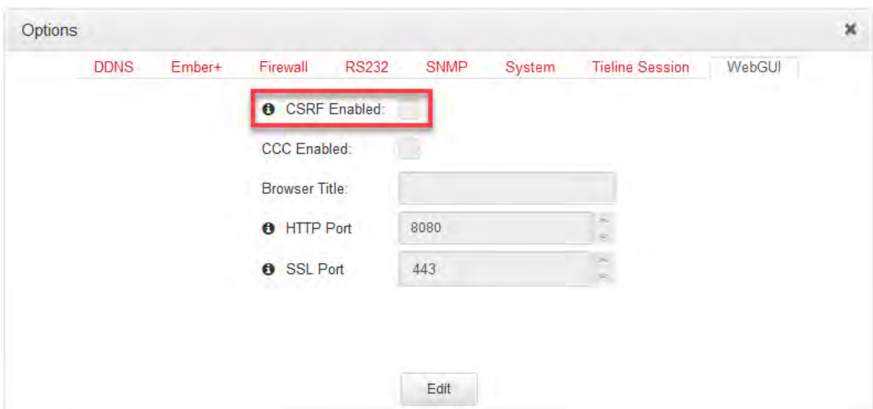
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Options** to display the **Options panel**.
2. Select **Firewall** and then click **Edit** to adjust settings.
3. Click **Save** to store the new configuration settings.



Configure Cross-Site Request Forgery

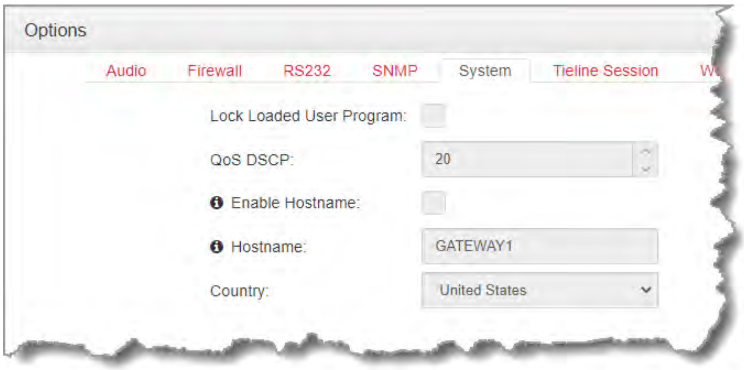
CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

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.

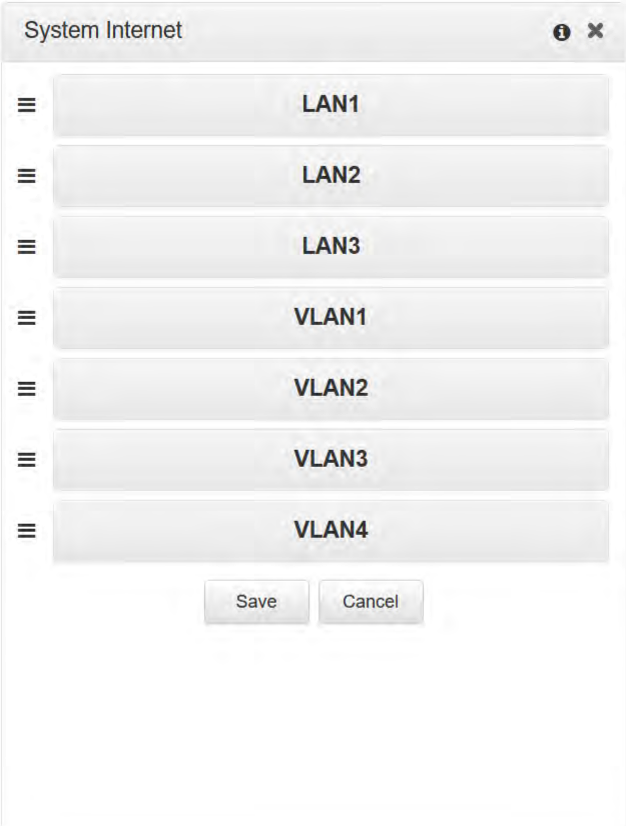


Important Note: Check with your IT administrator before changing this setting. By default the codec is configured for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

19.2 Configure System Internet Settings

In some situations when multiple interfaces are connected to the codec it may be necessary to select the preference for interfaces used to connect to the internet. For example, when using DDNS services. Or perhaps where the default setting is LAN1, however this may be connected to a closed network and LAN2 is the interface connected to a WAN, i.e. the internet. To adjust the default settings:

1. Open the HTML5 Toolbox Web-GUI and click **Transport** at the top of the screen, then click **System Internet** to display the **System Internet panel**.
2. Click **Edit** and drag the handles on the left-hand side to adjust the order of the listed interfaces.



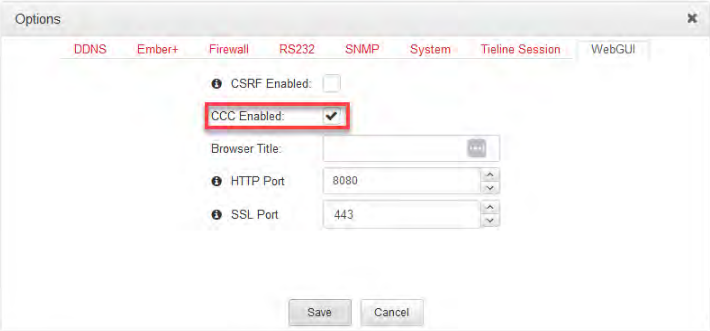
3. Click **Save** to store the new configuration.

Important Notes:
- The **System Internet** setting is only related to internet connectivity. It is not related to the IP Interface Setup for IP streaming interfaces.
- If interface 1 is not connected to the internet, the codec will attempt to use interface 2 and other interfaces listed thereafter in order to access the internet.
- If the codec is connecting to the Cloud Codec Controller and has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. See Enabling the Cloud Codec Controller for more details.

19.3 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.

DNS and the Cloud Codec Controller (CCC)

If a codec has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. When using the Toolbox HTML5 Web-GUI to configure DNS network settings:

1. Select **Transport** and then **Network** to open the **Network panel**.
2. Select an interface and then select the **DNS** tab to enter details into the **DNS Addresses** fields as required.
3. Ensure that the **Specify DNS Settings** check-box is selected. Note: This will ensure the DNS servers are used.

Note: These settings can also be configured using the front panel of the codec.

Where can I find my DNS settings?

DNS settings for static public IP addresses can be sourced from your internet provider/Telco. Sometimes the Gateway setting in the TCP/IP tab in the Network panel will work as the DNS Address to enter. Another common scenario encountered is that static DNS settings are configured to override DHCP settings when they are not compatible with the network being used. When using DHCP IP settings ensure the **Specify DNS Settings** check-box is disabled in the **Network panel**.

19.4 About MPX Connections

Important Notes:
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

A range of different MPX encode and decode programs can be configured, including:

- MPX Encode Programs.
- MPX Decode Programs.
- MPX Multi-Unicast Encode Programs
- MPX Multicast Server Encode Programs
- MPX Multicast Client Decode Program

Several encoding and decoding options are supported for MPX and MicroMPX streams.

1. µMPX over UDP: This is the native packetization format for MicroMPX and offers the lowest data overhead.
2. µMPX over RTP: This mode provides encapsulation in RTP because there are network monitoring tools that can track packet loss, network jitter etc for RTP packets. However, It does increase the data overhead.
3. Raw MPX over RTP: This provides uncompressed MPX (PCM 192kHz).
4. µMPX + MPEG-TS over RTP: This mode provides encapsulation in MPEG-TS as this can be fed directly into an MPEG-TS multiplexor, to avoid having to use a separate MPEG-TS encapsulator. Note: RTP provides network monitoring tools that can track packet loss, network jitter etc for RTP packets. However, It does increase the data overhead.
5. µMPX + MPEG-TS over UDP: This mode provides encapsulation in MPEG-TS as this can be fed directly into the MPEG-TS multiplexor, to avoid having to use a separate MPEG-TS encapsulator.
6. µMPX Forwarding: This mode allows a MicroMPX (UDP) stream to be received and then forwarded. For example, it is possible to receive a unicast and forward the signal as a multicast or visa versa. Along with MicroMPX forwarding, the user can send local GPIO data while maintaining synchronization with the MPX signal.
7. µMPX forwarding + MPEG-TS over RTP: This mode allows a MicroMPX (UDP) stream to be received and then encapsulated and forwarded as MPEG-TS over RTP.
8. µMPX forwarding + MPEG-TS over UDP: This mode allows a MicroMPX (UDP) stream to be received and then encapsulated and forwarded as MPEG-TS over UDP.

19.5 Configuring MPX Inputs and Outputs

Product	Feature	Notification
MPX I	RJ-45 AES192 MPX Input / Output	**AES192 IN / AES192 OUT** provide 1 stereo AES192 digital MPX composite input and output.
MPX I	BNC Analog MPX Composite Input; BNC Analog MPX Composite A & B Output	**MPX IN / MPX OUT A/B** provide 1 BNC Analog MPX composite Input and 2 BNC Analog MPX Outputs (1A & B output)
MPX II	2 RJ-45 AES192 MPX Inputs / Outputs	**AES192 IN 1 & 2 / AES192 OUT 1 & 2** provide 2 stereo AES192 digital MPX composite inputs and outputs.
MPX II	2 BNC Analog MPX Composite Inputs; 2 BNC Analog MPX Composite A & B Outputs	**MPX IN 1 & 2 / MPX OUT 1A/B & 2A/B** provide 2 BNC Analog MPX composite Inputs and 4 BNC Analog MPX Outputs (2A & B outputs)

MPX codecs support analog MPX using BNC input and output connectors, or digital AES192 MPX inputs and outputs using RJ-45 connectors. Attach either analog or digital MPX connections to the codec and then adjust settings using the **MPX Settings panel**. MicroMPX encodes a full FM composite or MPX signal, including audio, stereo pilot and RDS, over a low bitrate connection. It currently supports bit-rates from 320 up to 900 kbit/s. Bit-rates down to 176 kbit/s are supported in MicroMPX+ mode. Note: MicroMPX+ is not currently supported.

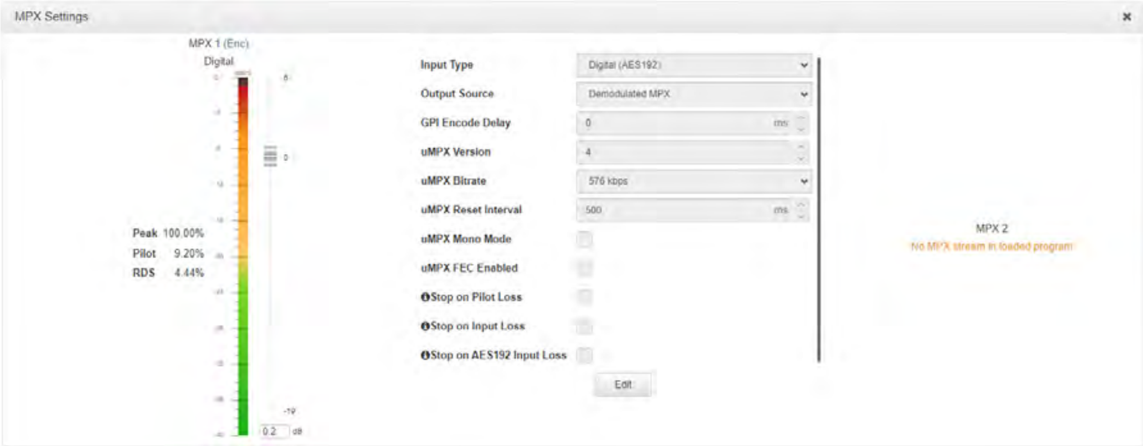
Important Notes:
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

Configure Input Type (Encoder) and the Output Source

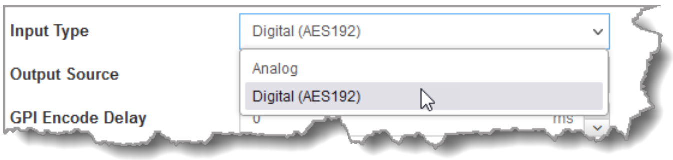
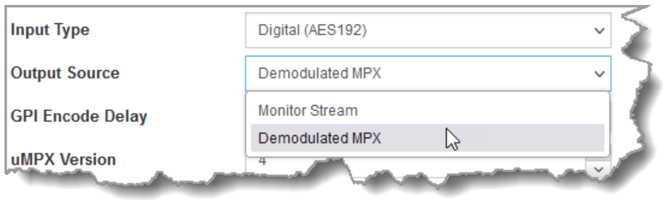
Important Note: Configure an MPX Encode or Decode program first and load it in the codec as this will configure the **MPX Settings panel** for either encoding or decoding MPX.

To configure the codec for MPX encoding:

1. Attach either an analog or digital MPX signal to the codec.
2. Open the HTML5 Toolbox Web-GUI and ensure an MPX encode program is loaded via the **Connections panel** or the **Program Manager panel**. Click **Audio** in the **Menu Bar**, then click **MPX Settings** to display the **MPX Settings panel**.
3. If an analog MPX source is being configured, click the **Plus symbol** + to expand the view and adjust the analog gain slider and the **dBu Ref** slider as required. Note: The **dBu Ref** slider can be adjusted to match the input source reference level being fed into the analog MPX input.



4. Click **Edit** to adjust other encoder settings, then click **Save**.

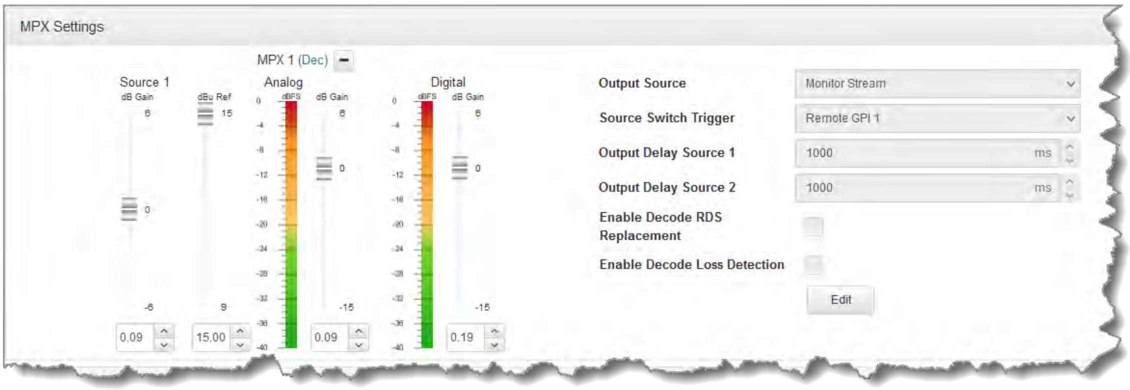
	Feature	Description
1	Input Type	Select either **Analog** or **Digital (AES192)**. The signal input level needs to be at exactly 0 dB for MicroMPX to encode correctly. Adjust the input gain slider until the signal peaks on the meter at 0dB. 
2	Output Source	Configure the preference for monitoring either **Demodulated MPX** audio locally, or an incoming remote codec **Monitor Stream**.  Note: A remote MPX codec will normally feed back audio from an FM monitor at the transmitter site to ensure audio transmissions are maintained.
3	GPI Encode Delay	Configure the GPI encode delay required to match audio latency as required.
4	µMPX Version	There are 4 different versions of µMPX: • Version 1: supports bit-rates from 320 up to 400 kbit/s. • Version 2 and 3: support bit-rates from 320 upto 576 kbit/s. • Version 4: supports bit-rates from 320 up to 900 kbit/s, and down to 176 kbit/s in MicroMPX+ mode.
5	µMPX Bitrate	Select a bit-rate that can be supported by the IP transport being used. When bandwidth is limited, 320 kbit/s is recommended as the minimum bit-rate as to use if possible.
6	µMPX Reset Interval	This setting determines how often a keyframe packet is sent.

7	µMPX Mono Mode	Select the check-box to force the codec to encode in mono. This will reduce the bit-rate requirement and increase the quality of the signal. This is useful when available network bandwidth is limited, and in situations where content is mainly mono, e.g. talk radio stations.
8	µMPX FEC Enabled	Select the check-box to enable FEC and then configure FEC Delay and Overhead settings. MicroMPX sends about 94 packets per second not counting recovery packets. If you want to keep your delay to at most 1 second, you can set the delay to around 64 packets. A setting of 64 (delay) to 8 (error correction overhead) means decoder latency must be at least 0.7 seconds approximately. With this configuration every 64 packets the codec will send 8 extra packets as overhead. As long as at least 64 of the resulting 72 packets arrive, the decoder can reconstruct all missing packets. If latency is not an issue the delay and decoder latency can be increased as required.
9	Stop on Pilot Loss	Select this check-box to stop encoding if the stereo pilot is lost. This can trigger an alarm at the encoder or decoder as required.
10	Stop on Input Loss	Select this check-box to stop encoding if the MPX input is lost. This can trigger an alarm at the encoder or decoder as required.
11	Stop on AES192 Input Loss	Select this check-box to stop encoding if the AES192 input is lost. This can trigger an alarm at the encoder or decoder as required.

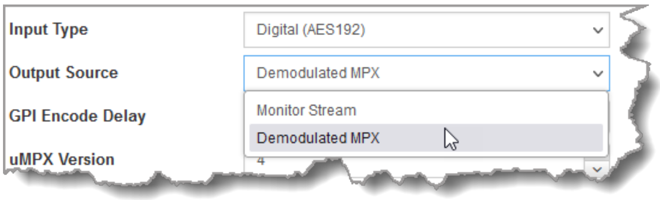
Configure Decoder and the Output Source

To configure the codec for MPX decoding:

1. Attach either an analog or digital MPX input to the codec.
2. Open the HTML5 Toolbox Web-GUI and ensure an MPX decode program is loaded via the **Connections panel** or the **Program Manager panel**. Click **Audio** in the **Menu Bar**, then click **MPX Settings** to display the **MPX Settings panel.**
3. Click the **Plus symbol** + to expand the view and adjust the **Source** gain slider and the **dBu Ref** slider as required. Note: The **dBu Ref** slider can be adjusted to match the expected reference level of devices fed from the analog MPX output.



4. Next click **Edit** to adjust other decoder settings as required.

	Feature	Description
1	Output Source	Configure the preference for monitoring either **Demodulated MPX** audio locally, or an incoming remote codec **Monitor Stream**. 
2	Source Switch Trigger	Select a local or remote GPI that will act as the trigger to switch between output sources.
3	Output Delay Source 1	Output delay for Audio Source 1 in milliseconds. This setting relates to the MPX Source 1 in the **Output Audio Source** section of the **Program Manager panel**. It also configures the buffer setting for the decoded signal. Note: The maximum buffer setting for RAW uncompressed MPX is 1,000ms.
4	Output Delay Source 2	Output delay for Audio Source 2 in milliseconds. This setting relates to the MPX Source 2 in the **Output Audio Source** section of the **Program Manager panel**. It also configures the buffer setting for the decoded signal. Note: The maximum buffer setting for RAW uncompressed MPX is 1,000ms.
5	Enable Decode Loss Detection	Select the check-box to detect when the decoder stopped decoding (potentially from loss of network or encoder). If the decoder buffer remains empty for the given timeframe, the codec will set the AES192 output valid bit to false.

Connecting a Decoder to an FM Transmitter

If a transmitter has a digital MPX input (MPX over AES/EBU), that is the preferred way of connecting a MPX decoder. For analog MPX inputs, the transmitter might feature an XLR or BNC MPX input connector. Only one connector needs to be attached for an MPX data signal. Attach the MPX connection and then adjust the level so that the output of the transmitter complies to your local laws (75 kHz modulation – or sometimes a bit more).

It is recommended that the MPX/MicroMPX encoder sends a test tone signal at 100% peak during initial setup to assist with line-ups at the decoding codec. Select and transmit a 1kHz tone and adjust the MicroMPX or MPX output level, or the transmitter input gain, to match the maximum allowed modulation.

19.6 Configure MPX Encode Programs

Product	Feature	Notification
MPX I	Program Manager panel	• Supports 1 peer-to-peer MPX encode stream. • Supports one MPX multi-unicast to up to 5 endpoints, or one multicast audio stream. • Supports an MPX stream and a confidence monitoring connection.
MPX II	Program Manager panel	• Supports 1 or 2 peer-to-peer MPX encode streams • Supports two MPX multi-unicasts to up to 10 endpoints in total. For example, 2 separate audio streams can each be streamed to 5 endpoints. 2 different multicast streams also supported. • Supports 2 MPX encode streams and 2 confidence monitoring connections

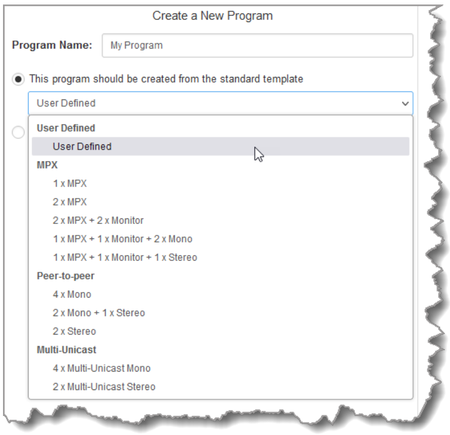
User Defined programs let you configure connections depending on how many channels the codec supports. Configure MPX connections by selecting a standard program template and then remove or add MPX connections and adjust configuration settings as required. This delivers flexibility for distribution of MPX signals over wide area networks. Before configuring an MPX connection consider if it will also have an associated confidence monitoring connection, which streams audio back to the studio/encoding codec.

Important Notes:
- Connections using uncompressed MPX or compressed µMPX are sessionless connections.
- A program cannot be edited in the **Program Manager panel** when loaded in the codec.
- Lock a loaded custom program or multistream program in a codec receiving a call 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 single connection dialing into the codec.
- 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 using default settings can save configuration time.
- SmartStream PLUS redundant streaming is not available with sessionless IP connections.
- Redundant MicroMPX connections can be configured by sending a stream to multiple destinations.
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

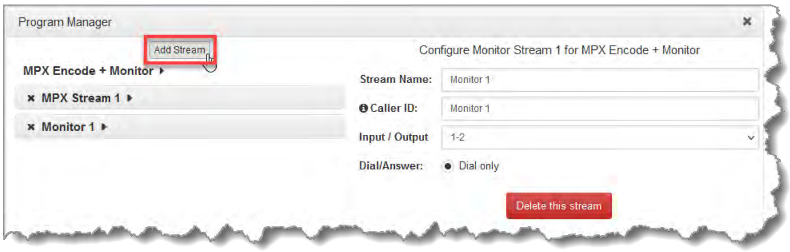
Creating MPX Encode Connections

Use the following info to configure uncompressed MPX connections, or compressed MicroMPX connections.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option to create one or more new audio streams, then click **Next**.

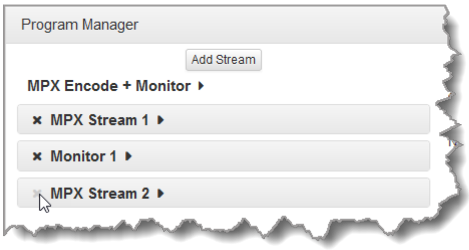


5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: the **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.

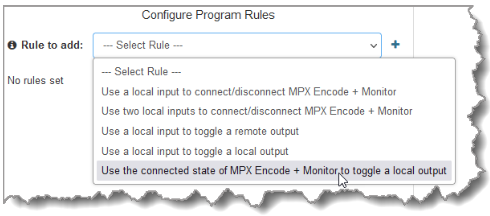


Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

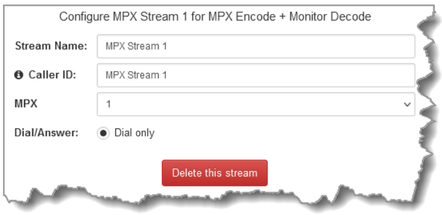
6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**.



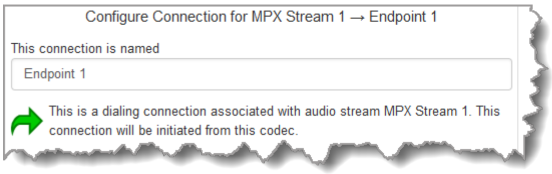
7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule. Note: Enlarge the width of the **Program Manager panel** to view and edit streams using the configuration settings available on the left-hand side of the panel. This makes it simpler to navigate to points in the wizard without navigating through multiple screens.



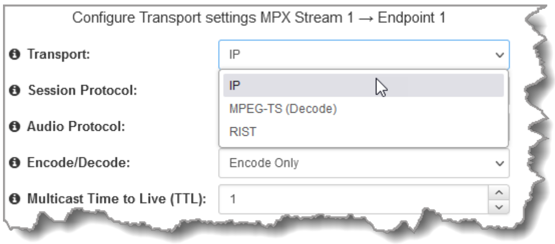
8. Enter the **Stream Name** and select an **MPX** input, then click **Next**.



9. Enter the name of the connection in the text box and then click **Next**. Note: If this MPX encode program will be distributed to multiple endpoints as a multi-unicast, there will be an opportunity to add additional connections later in the **Program Manager panel**.

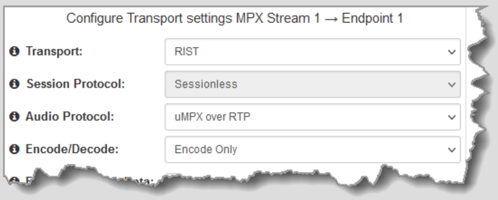


10. Configure the transport settings for the connection. Select **IP** or **RIST** (Reliable Internet Stream Transport) as the transport and configure the preferred audio protocol (e.g. **µMPX over UDP**, **µMPX over RTP**, or **RAW MPX over RTP**. Then select **Encode Only** and click **Next**.



Important Notes:

- If uncompressed **Raw MPX over RTP** encoding is selected it is necessary to select either 16 bit or 24 bit sampling.
- See Configure MPEG-TS Encode Programs for details on streaming MicroMPX encapsulated as MPEG-TS.
- It is also possible to select **RIST** as the transport and encapsulate a MicroMPX signal over UDP. RIST provides error recovery by supporting retransmission of lost packets. The sender keeps track of which packets were successfully received at a decoder, and if missing packets are detected, the lost packets are retransmitted.



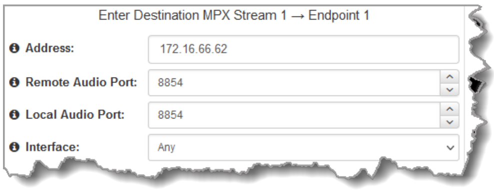
- See RS232 Data Configuration for detailed information on relay and RS232 data.

11. Configure destination codec dialing like the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN1**) or **Secondary** (**LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable. Add an additional MPX connection if required, then click **Next**.

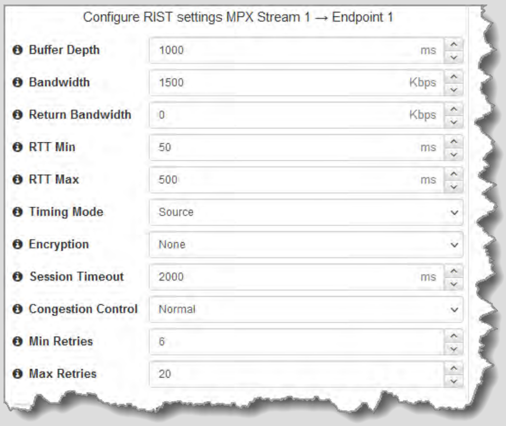
Important Notes:

- For both uncompressed MPX and MicroMPX connections the default audio port for MPX Stream 1 is 8854 and for MPX Stream 2 is 8874. There is no session port as these connections are sessionless.

- The default monitoring connection audio ports for MPX Stream 1 and MPX Stream 2 are 9010 and 9030 respectively.
- Additional encode streams may be redundant encode streams, or in a traditional multi-unicast use case can be used to distribute a stream to multiple locations. If there are multiple interfaces connected between the encoder and decoder codecs, e.g. different ISPs, satellite connection etc., this provides additional layers of redundancy.
- To create a multicast server or client program, simply enter a multicast address in the IP address field, e.g. 239.xxx.xxx.xxx and the connection will automatically be configured for multicasting. The **Multicast Time to Live (TTL)** setting can also be configured for a server program.



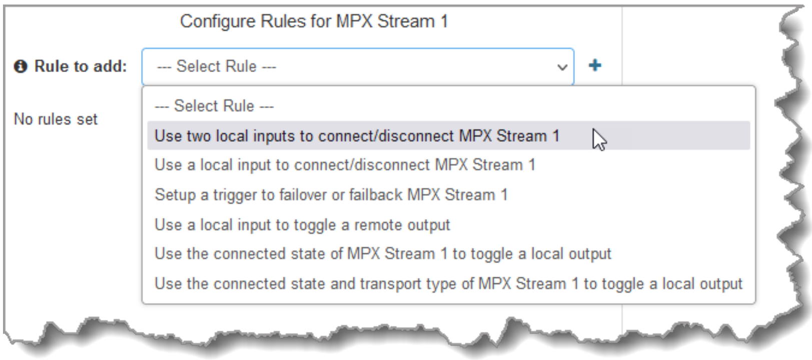
Important Note: If MicroMPX packets are encapsulated using **RIST** there is a range of RIST settings that can be configured.



- **Buffer Depth**: Indicates the jitter buffer size in milliseconds.
- **Bandwidth**: Maximum bandwidth in kbps. Note: Must be higher than the maximum stream bandwidth, plus headroom, and re-requested packets. Analyze stats and add 10% for constant bit-rate safety, or add 100% for variable bit-rate.
- **Return Bandwidth**: Maximum bandwidth in kbps for receiver to sender data streams.
- **RTT Min**: Minimum wait time in milliseconds before re-requesting packets. Adjust for network conditions.
- **RTT Max**: Maximum wait time in milliseconds before re-requesting packets. Note: Usually RTT Min and RTT Max should be equal to one another.
- **Timing Mode**: Synchronize playback, e.g. using NTP plus the buffer size as a guide.
- **AES Type**: Specify encryption, e.g. 128 for AES-128, or 256 for AES-256.
- **Secret**: Set a passphrase for encryption.
- **Session Timeout**: Timeout in milliseconds for terminating the RIST connection when no keep alive response is received.
- **Key Rotation**: The Key rotation period in milliseconds when aes and a passphrases are specified.
- **Congestion Control**: Built in congestion control for situations in which a sender drops off the connection, but the receiver still sends re-requests. The three options include: **Disabled**, **Normal** and **Aggressive**. Note: Don't configure **Aggressive** unless congestion is definitely a problem.
- **Min Retries**: The minimum number of re-requests for a lost packet. Note: setting this too high can lead to congestion.
- **Max Retries**: The maximum number of re-requests for a lost packet.

- SRP Username: The srp-auth username credentials defined (globally) on the "other" side when it is in listen mode. An srp-auth file holds the username credentials.
- SRP Password: The srp-auth password credentials defined (globally) on the "other" side when it is in listen mode. An srp-auth file holds the password credentials.

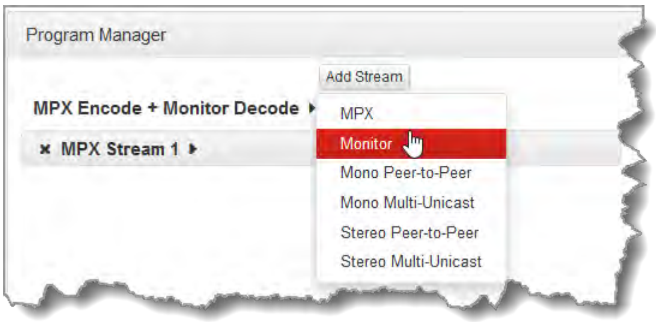
12. Configure stream level rules if required, then click **Next**.



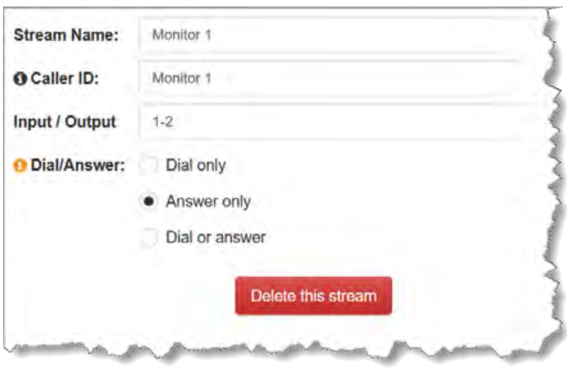
Add a Monitoring Stream

At this point is is possible to add/configure a monitoring stream if required to receive a confidence monitoring stream.

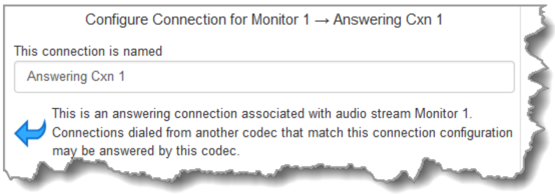
1. Click the **Add Stream** button to add a **Monitor** stream if it is not already visible in the program.



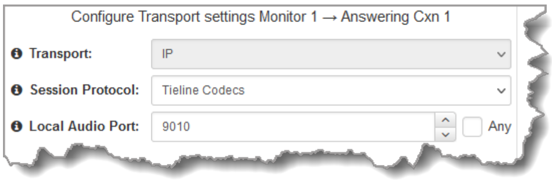
2. Usually this will be an answering connection that decodes confidence monitoring audio sent back by the codec at a transmitter site. Enter a **Stream Name**, **Caller ID** if required, and select the codec **Inputs/Outputs**. Then click **Next**.



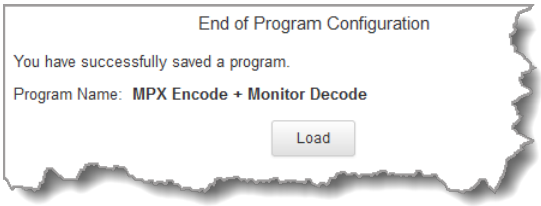
3. Enter a name for the incoming monitoring connection and click **Next**.



4. See Configure IP Streaming Programs if required for more details on configuring Tieline IP stream connections. Otherwise continue through the wizard to the end and click **Save**. Note: Only Opus encoding is supported for sessionless or Tieline session Monitor connections.



5. The newly created program can be loaded from within the **Program Manager panel** or the **Connections panel**. Load, Unload and Dial a Program using the **Connections panel**, or connect the program using the codec front panel.



19.7 Configure MPX Decode Programs

Product	Feature	Notification
MPX I	RJ-45 AES192 MPX Input / Output	**AES192 IN / AES192 OUT** provide 1 stereo AES192 digital MPX composite input and output.
MPX I	BNC Analog MPX Composite Input; BNC Analog MPX Composite A & B Output	**MPX IN / MPX OUT A/B** provide 1 BNC Analog MPX composite Input and 2 BNC Analog MPX Outputs (1A & B output)
MPX II	2 RJ-45 AES192 MPX Inputs / Outputs	**AES192 IN 1 & 2 / AES192 OUT 1 & 2** provide 2 stereo AES192 digital MPX composite inputs and outputs.
MPX II	2 BNC Analog MPX Composite Inputs; 2 BNC Analog MPX Composite A & B Outputs	**MPX IN 1 & 2 / MPX OUT 1A/B & 2A/B** provide 2 BNC Analog MPX composite Inputs and 4 BNC Analog MPX Outputs (2A & B outputs)

When configuring an MPX Decode program consider if a confidence monitor stream IP stream will also be configured to send back to the studio/sender.

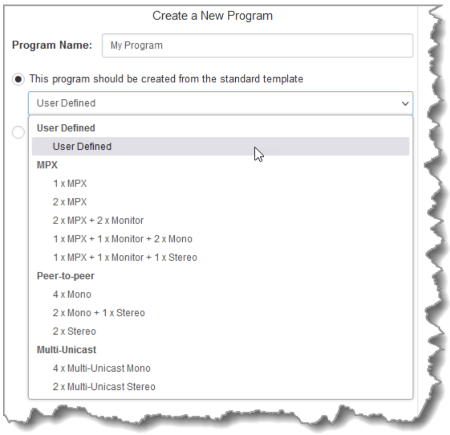
Create an MPX Decode Program

Important Notes: Before configuring a program please note:

- Connections using uncompressed MPX or compressed µMPX are sessionless connections.
- 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 single stream dialing into the codec.
- 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 using default settings can save configuration time.
- SmartStream PLUS redundant streaming is not available with sessionless IP connections.
- The maximum buffer setting for RAW uncompressed MPX connections is 1,000ms. This setting is configured by the **Output Delay Source 1/2** setting in the **MPX Settings panel**.
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

User Defined programs let you configure connections depending on how many channels the codec supports. Alternatively, configure connections by selecting a standard program template and then remove or add streams and adjust configuration settings as required. This delivers flexibility for distribution of audio streams over wide area networks.

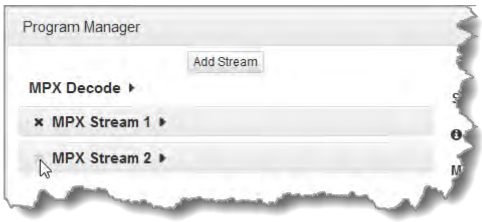
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option to create one or more new audio streams, then click **Next**.



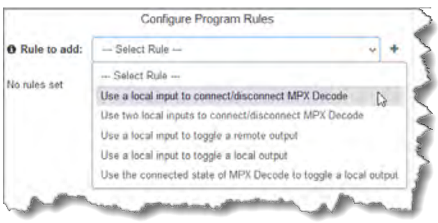
5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: The **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.

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.

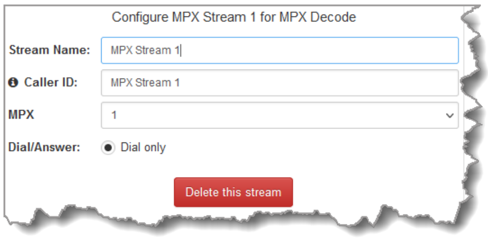
6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**. Note: Enlarge the width of the **Program Manager panel** to view and edit streams using the configuration settings available on the left-hand side of the panel. This makes it simpler to navigate to points in the configuration wizard without navigating through multiple screens.



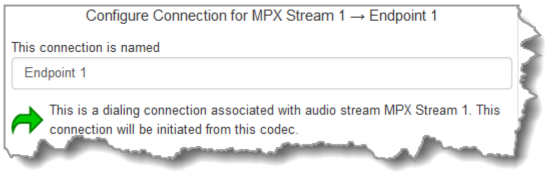
7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule.



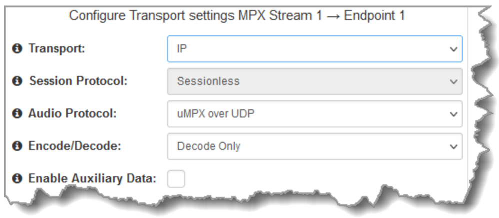
8. Enter the **Stream Name** and select an **MPX** input, then click **Next**.



9. Enter the name of the connection in the text box and then click **Next**. Note: If this MPX encode program will be distributed to multiple endpoints as a multi-unicast, there will be an opportunity to add additional connections later in the **Program Manager panel**.

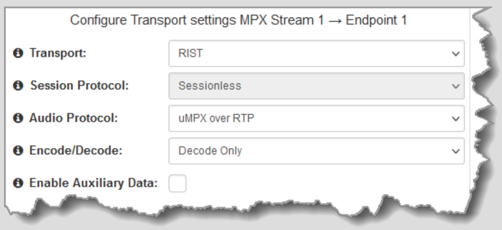


10. Configure the transport settings for the connection. Select **IP** or **RIST** (Reliable Internet Stream Transport) as the transport and configure the preferred audio protocol (e.g. **µMPX over UDP**, **µMPX over RTP**, or **RAW MPX over RTP**. Then select **Encode Only** and click **Next**.



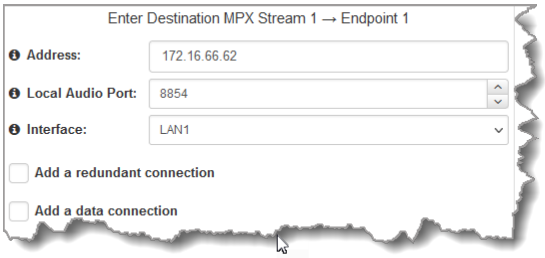
Important Notes:

- If uncompressed **Raw MPX over RTP** decoding is selected it is necessary to select either 16 bit or 24 bit sampling.
- See Configure MPEG-TS Encode Programs or Configure MPEG-TS Decode Programs for details on streaming µMPX encapsulated as MPEG-TS.
- It is also possible to select **RIST** as the transport and receive a MicroMPX signal over UDP that support RIST. RIST provides error recovery by supporting retransmission of lost packets. The sender keeps track of which packets were successfully received at a decoder, and if missing packets are detected, the lost packets are retransmitted.



- See RS232 Data Configuration for detailed information on relay and RS232 data and supported data connections.

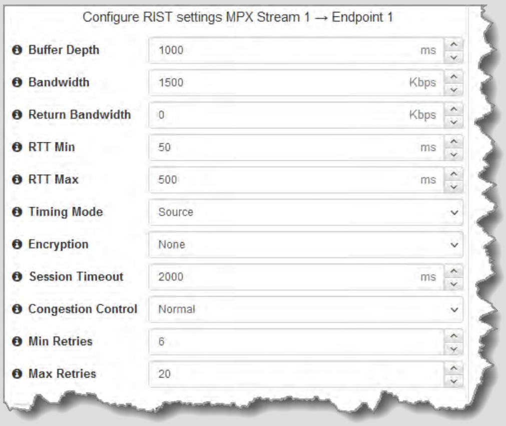
11. Configure destination codec dialing like the IP address, port, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN1**) or **Secondary** (**LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable. Select **Add a redundant connection** to add a redundant decode stream if required, then click **Next.**



Important Notes:

- For both uncompressed MPX and MicroMPX connections the default audio port for MPX Stream 1 is 8854 and for MPX Stream 2 is 8874. There is no session port as these connections are sessionless.
- The default monitoring connection audio ports for MPX Stream 1 and MPX Stream 2 are 9010 and 9030 respectively.
- To create a multicast server or client program, simply enter a multicast address in the IP address field, e.g. 239.xxx.xxx.xxx and the connection will automatically be configured for multicasting. The **Multicast Time to Live (TTL)** setting can also be configured for a server program.

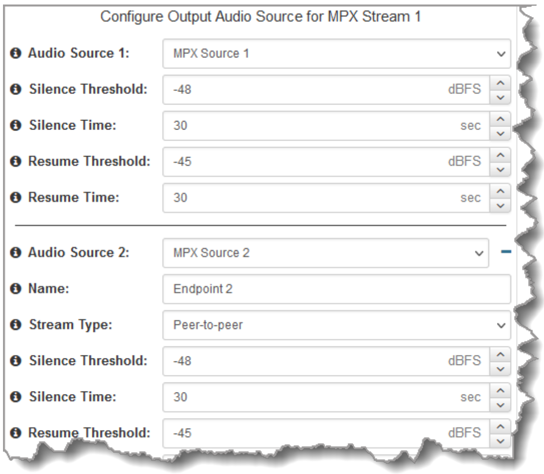
Important Note: If MicroMPX packets are encapsulated using **RIST** there is a range of RIST settings that can be configured.



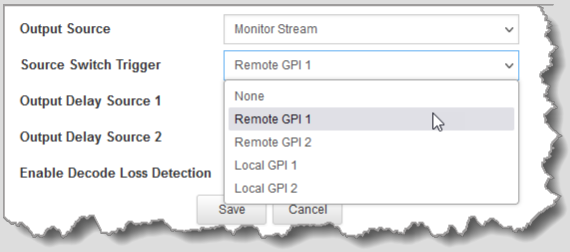
- **Buffer Depth**: Indicates the jitter buffer size in milliseconds.
- **Bandwidth**: Maximum bandwidth in kbps. Note: Must be higher than the maximum stream bandwidth, plus headroom, and re-requested packets. Analyze stats and add 10% for constant bit-rate safety, or add 100% for variable bit-rate.
- **Return Bandwidth**: Maximum bandwidth in kbps for receiver to sender data streams.
- **RTT Min**: Minimum wait time in milliseconds before re-requesting packets. Adjust for network conditions.
- **RTT Max**: Maximum wait time in milliseconds before re-requesting packets. Note: Usually RTT Min and RTT Max should be equal to one another.
- **Timing Mode**: Synchronize playback, e.g. using NTP plus the buffer size as a guide.
- **AES Type**: Specify encryption, e.g. 128 for AES-128, or 256 for AES-256.
- **Secret**: Set a passphrase for encryption.
- **Session Timeout**: Timeout in milliseconds for terminating the RIST connection when no keep alive response is received.
- **Key Rotation**: The Key rotation period in milliseconds when aes and a passphrases are specified.
- **Congestion Control**: Built in congestion control for situations in which a sender drops off the connection, but the receiver still sends re-requests. The three options include: **Disabled**, **Normal** and **Aggressive**. Note: Don't configure **Aggressive** unless congestion is definitely a problem.

- **Min Retries**: The minimum number of re-requests for a lost packet. Note: setting this too high can lead to congestion.
- **Max Retries**: The maximum number of re-requests for a lost packet.
- SRP Username: The srp-auth username credentials defined (globally) on the “other” side when it is in listen mode. An srp-auth file holds the username credentials.
- SRP Password: The srp-auth password credentials defined (globally) on the “other” side when it is in listen mode. An srp-auth file holds the password credentials.

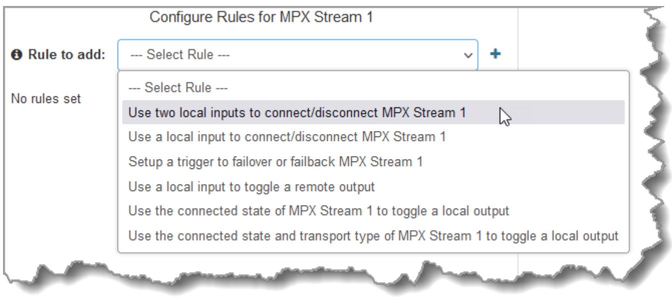
12. If two MPX inputs are attached to an MPX II codec which supports two input sources, you can configure an alternate **Output Audio Source** for the audio stream being configured. For example, **Audio Source 1** could be a satellite source and **Audio Source 2** could be local station content. Then click **Next**.



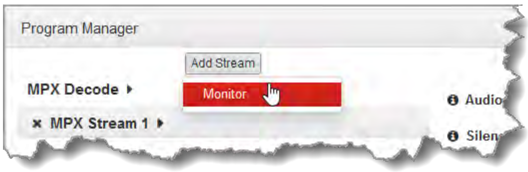
Important Note: In the **MPX Settings panel** you can select a local or remote GPI that will act as the trigger to switch between output sources. Note: MPX I codecs only support a single MPX input source.



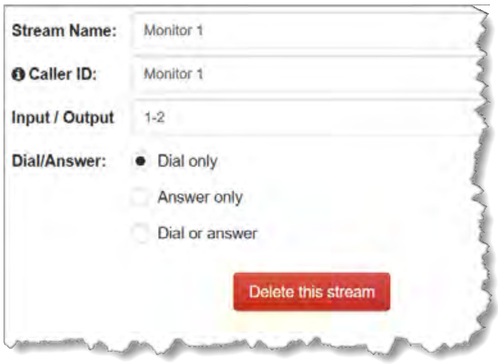
13. Continue configuring the transport settings for **MPX Source 2** like previously for **MPX Source 1**. Then configure any rules if required and click **Next**.



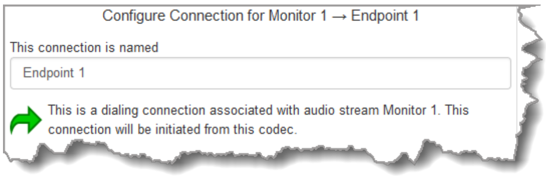
14. If configuration is complete click **Save**. If a confidence monitoring connection is required, click the **Add Stream** button to add a new **Monitor** stream.



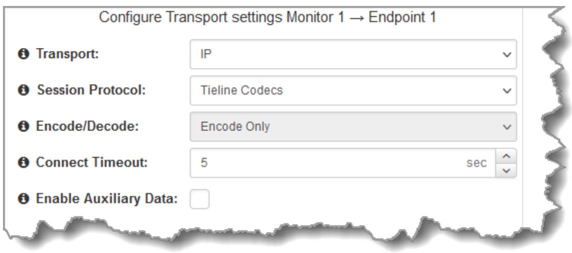
15. Then configure a dialing connection to encode and send the stream back to the studio or monitoring site, e.g. Master Control. Then click **Next**.



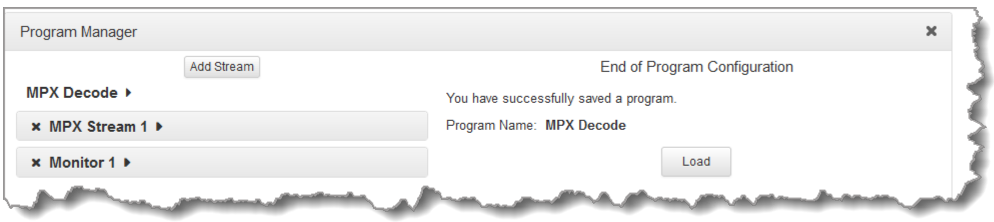
16. Enter a name for the connection and click **Next**.



17. See [Configure IP Streaming Programs] if required for more details on configuring Tieline IP stream connections. Otherwise continue through the wizard to the end and click **Save**. Note: Only Opus encoding is supported for sessionless or Tieline session Monitor connections.



18. The newly created program can be loaded from within the **Program Manager panel** or the **Connections panel**. [Load, Unload and Dial a Program] using the **Connections panel**, or connect the program using the codec front panel.



19.8 MPEG-TS Satellite Module Settings

There are three Web-GUI panels for monitoring and configuring satellite signals being received via the optional DVB satellite module.

Optional satellite module

Satellite Tuner Status Panel

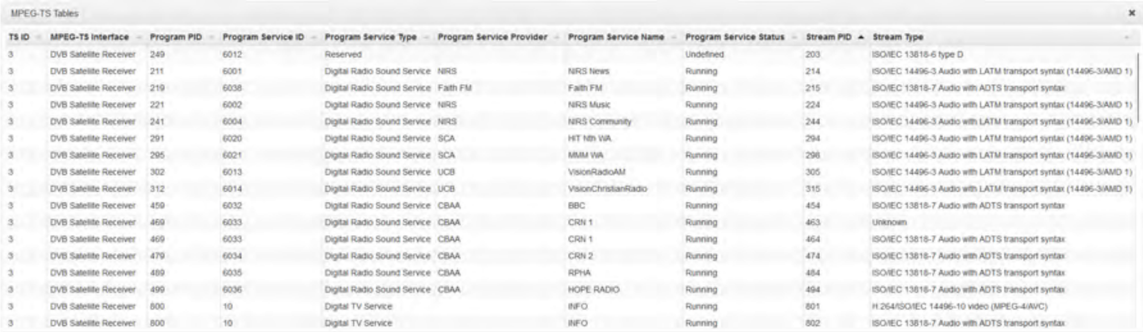
The **Satellite Tuner Status panel** displays the quality measurements and settings for the satellite signal that is currently being received.

MPEG-TS Interfaces Panel

The **MPEG-TS Interfaces panel** allows you to configure interfaces to receive MPEG-TS packets. These interfaces may include LAN1, LAN2 and the DVB Satellite Receiver (if installed).

MPEG-TS Tables Panel

The MPEG-TS Tables panel displays the contents of the DVB SI tables that are received on enabled MPEG-TS interfaces.


MPEG-TS Tables

TS ID	MPEG-TS Interface	Program PID	Program Service ID	Program Service Type	Program Service Provider	Program Service Name	Program Service Status	Stream PID	Stream Type
3	DVB Satellite Receiver	249	6012	Reserved			Undefined	203	ISO/IEC 13818-6 type D
3	DVB Satellite Receiver	211	6001	Digital Radio Sound Service	NIRS	NIRS News	Running	214	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	219	6038	Digital Radio Sound Service	Faith FM	Faith FM	Running	215	ISO/IEC 13818-7 Audio with ADTS transport syntax
3	DVB Satellite Receiver	221	6002	Digital Radio Sound Service	NIRS	NIRS Music	Running	224	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	241	6004	Digital Radio Sound Service	NIRS	NIRS Community	Running	244	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	291	6020	Digital Radio Sound Service	SCA	HIT Nth WA	Running	294	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	295	6021	Digital Radio Sound Service	SCA	MMM WA	Running	298	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	302	6013	Digital Radio Sound Service	UCB	VisionRadioAM	Running	305	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	312	6014	Digital Radio Sound Service	UCB	VisionChristianRadio	Running	315	ISO/IEC 14496-3 Audio with LATM transport syntax (14496-3/AMD 1)
3	DVB Satellite Receiver	459	6032	Digital Radio Sound Service	CBAA	BBC	Running	454	ISO/IEC 13818-7 Audio with ADTS transport syntax
3	DVB Satellite Receiver	469	6033	Digital Radio Sound Service	CBAA	CRN 1	Running	463	Unknown
3	DVB Satellite Receiver	469	6033	Digital Radio Sound Service	CBAA	CRN 1	Running	464	ISO/IEC 13818-7 Audio with ADTS transport syntax
3	DVB Satellite Receiver	479	6034	Digital Radio Sound Service	CBAA	CRN 2	Running	474	ISO/IEC 13818-7 Audio with ADTS transport syntax
3	DVB Satellite Receiver	489	6035	Digital Radio Sound Service	CBAA	RPHA	Running	484	ISO/IEC 13818-7 Audio with ADTS transport syntax
3	DVB Satellite Receiver	499	6036	Digital Radio Sound Service	CBAA	HOPE RADIO	Running	494	ISO/IEC 13818-7 Audio with ADTS transport syntax
3	DVB Satellite Receiver	800	10	Digital TV Service		INFO	Running	801	H.264/ISO/IEC 14496-10 video (MPEG-4/AVC)
3	DVB Satellite Receiver	800	10	Digital TV Service		INFO	Running	802	ISO/IEC 13818-7 Audio with ADTS transport syntax

19.9 Configure MPEG-TS Encode Programs

Product	Feature	Notification
MPX I	MPEG-TS Encode	Configure 1 MPEG-TS Encode stream.
MPX II	MPEG-TS Encode	Configure 1 or 2 MPEG-TS Encode streams.

MPEG-TS is a standard format for transmission and storage of audio, video, and data used in broadcast systems such as DVB and ATSC. MPEG-TS allows for the multiplexing of multiple elementary streams into a single transport stream. MPEG-TS supports tables which provide essential information about the programs and services within the transport stream. These tables help receivers in demultiplexing and decoding the transmitted content.

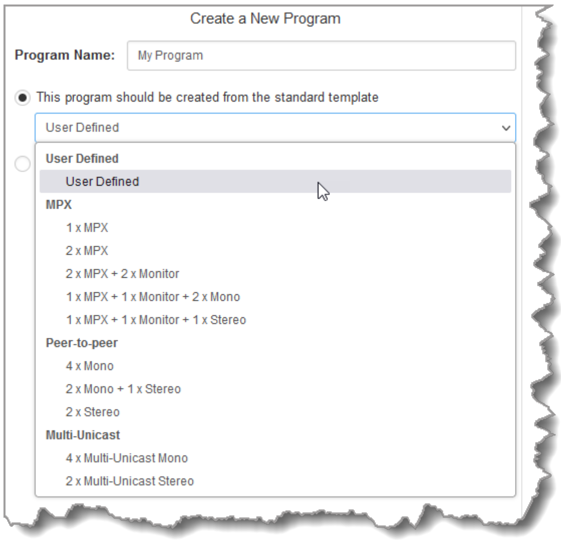
Configure MPEG-TS Encode Programs

MPEG-TS encode programs allow the codec to encapsulate a MicroMPX signal in MPEG-TS IP packets for transmission over IP to a destination codec. The configuration workflow is generally similar to setting up a standard IP audio stream.

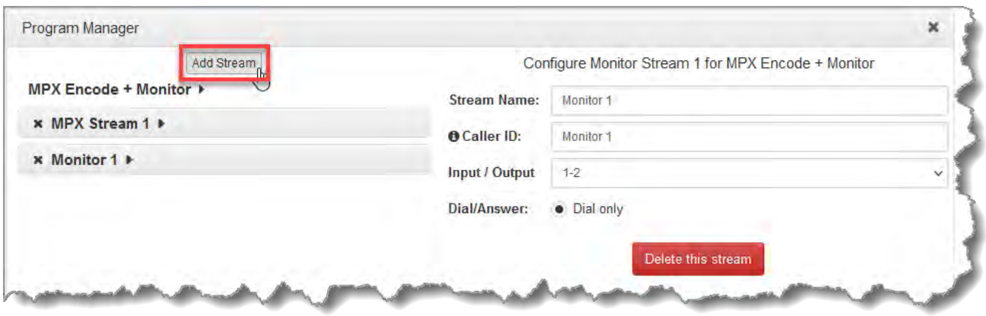
Important Notes:

- Connections using uncompressed MPX or compressed µMPX are sessionless connections.
- A program cannot be edited in the **Program Manager panel** when loaded in the codec.
- Lock a loaded custom program or multistream program in a codec receiving a call 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 single connection dialing into the codec.
- 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 using default settings can save configuration time.
- SmartStream PLUS redundant streaming is not available with sessionless IP connections.
- Redundant MicroMPX connections can be configured by sending a stream to multiple destinations.
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option to create one or more new audio streams, then click **Next**.

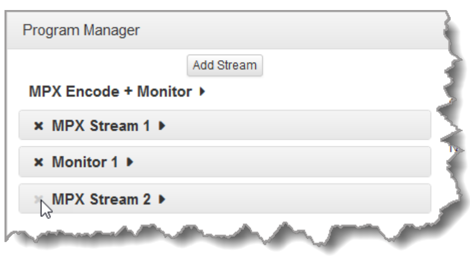


5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: the **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.



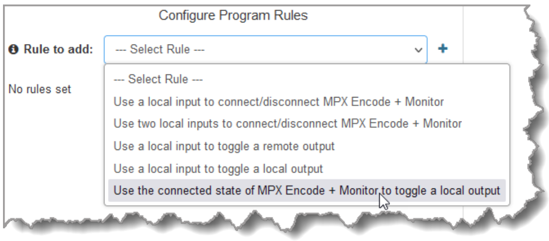
Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**.

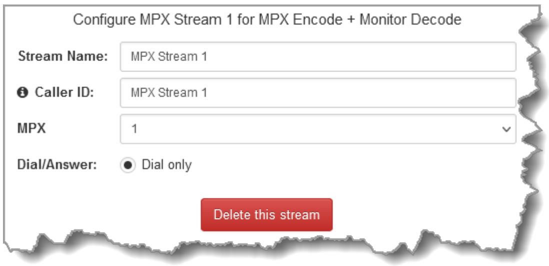


7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule. Note: Enlarge the width of the **Program Manager**

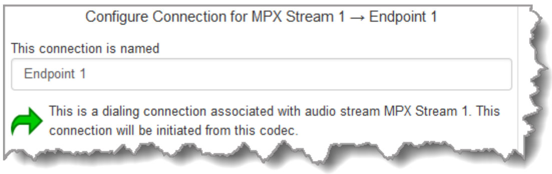
panel to view and edit streams using the configuration settings available on the left-hand side of the panel. This makes it simpler to navigate to points in the wizard without navigating through multiple screens.



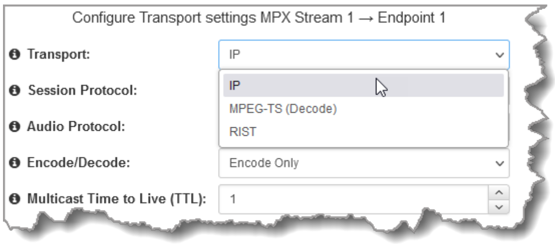
8. Enter the **Stream Name**, then click **Next**.



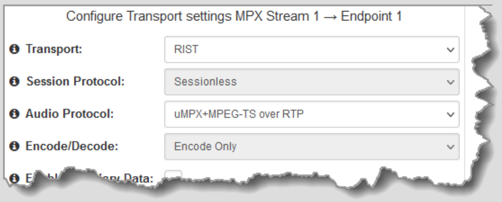
9. Enter the name of the connection in the text box and then click **Next**.



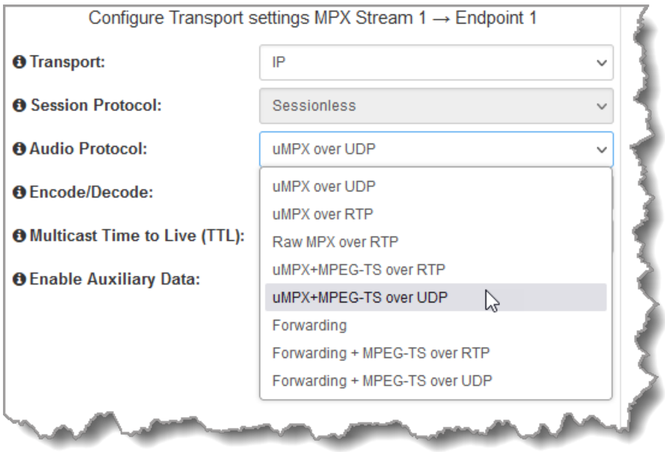
10. Select **IP** as the transport, then click **Next**. Note: Select **Enable Auxiliary Data** to enable synchronized out-of-band RS-232 data in separate packets which can be configured in the **MPEG-TS Settings** screen.



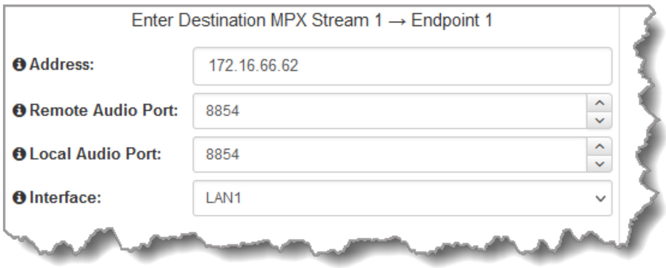
Important Note: It is also possible to select **RIST** as the transport and encapsulate a MicroMPX signal in MPEG-TS IP packets. RIST provides error recovery by supporting retransmission of lost packets. The sender keeps track of which packets were successfully received at a decoder, and if missing packets are detected, the lost packets are retransmitted.



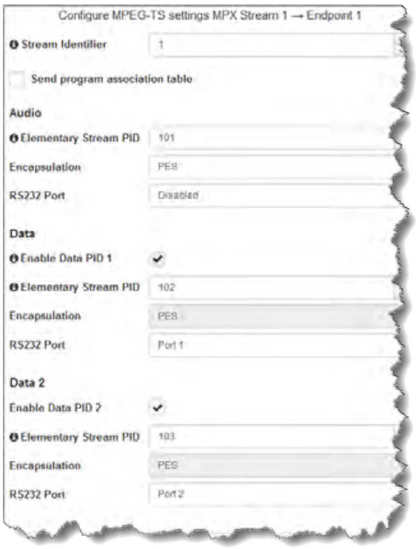
11. Configure the Audio Protocol as **uMPX+MPEG-TS over RTP** or **uMPX+MPEG-TS over UDP**. Note: RTP can be selected if the remote device supports RTP headers. RTP provides network monitoring tools that can track packet loss, network jitter etc for RTP packets. However, It does increase the data overhead.



12. Enter the IP address, remote and local ports and choose an interface.



13. Configure MPEG-TS settings where both PES and MPE encapsulation are supported. In-band data is also supported, or if **Enable Auxiliary Data** was configured previously, it is possible to configure up to two separate RS-232 data streams with separate Stream PIDs.

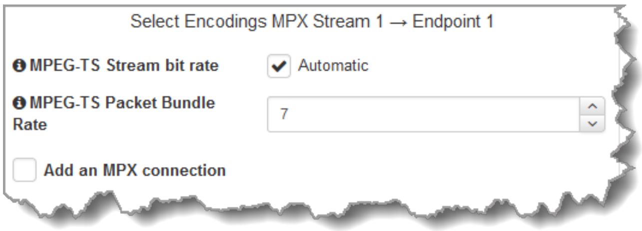


Important Notes:

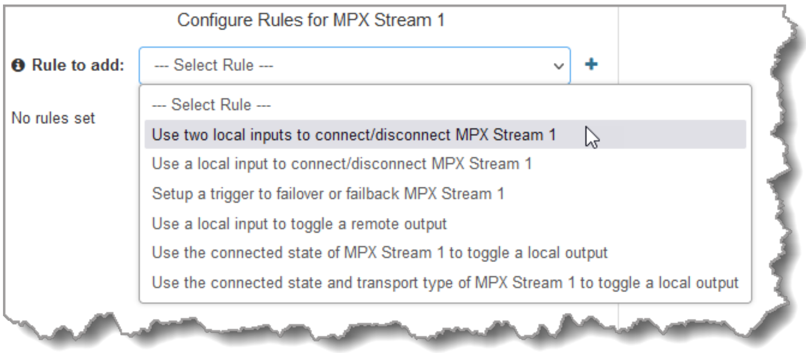
- The **Stream Identifier** relates to the **TS ID** in the **MPEG-TS Tables panel**. Range from 1 - 65535.
- The **Audio Elementary Stream PID** or **MPE PID** relates to the **Stream PID** in the **MPEG-TS Tables panel**. Range from 32 - 8190.

- The **Data Elementary Stream PID** relates to the **Stream PID** in the **MPEG-TS Tables panel**. This data stream requires a different Stream PID to the audio stream. Range from 32 - 8190.

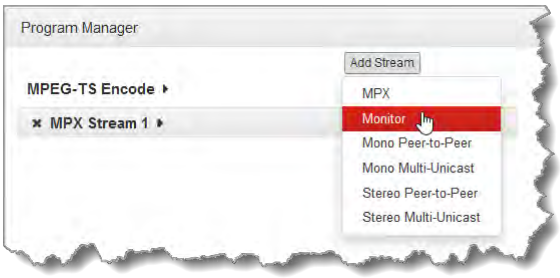
14. Configure Encoding settings like a specific **MPEG-TS Stream Bit Rate** if required, or adjust the **MPEG-TS Packet Bundle Rate**. Add additional MPX connections if required, then click **Next**. Note: Additional encode streams may be redundant encode streams, or in a traditional multi-unicast use case can be used to distribute a stream to multiple locations. If there are multiple interfaces connected between the encoder and decoder codecs, e.g. different ISPs, satellite connection etc., this provides additional layers of redundancy.



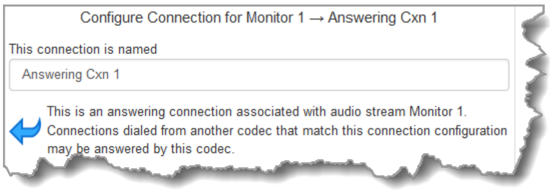
15. Configure stream level rules if required, then click **Next**.



16. If configuration is complete click **Save**. If a confidence monitoring connection is required, click the **Add Stream** button to add a new **Monitor** stream. Then configure an answering connection to receive the stream.

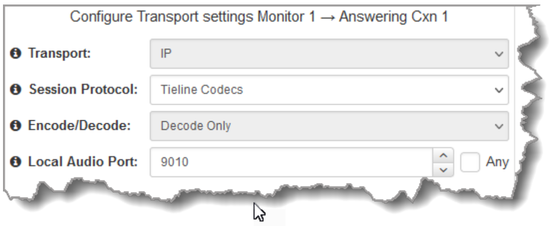


17. Enter a name for the incoming monitoring connection and click **Next**.

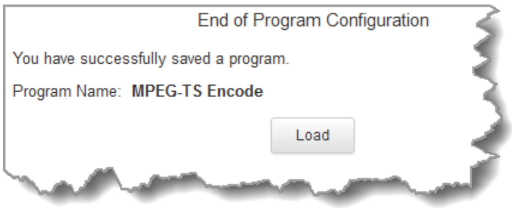


18. See [Configure IP Streaming Programs]() if required for more details on configuring Tieline IP stream connections. Otherwise continue through the wizard to the end and click **Save**.

Note: Only Opus encoding is supported for sessionless or Tieline session Monitor connections.



19. The newly created program can be loaded from within the **Program Manager panel** or the **Connections panel**. Load, Unload and Dial a Program using the **Connections panel**, or connect the program using the codec front panel.



19.10 Configure MPEG-TS Decode Programs

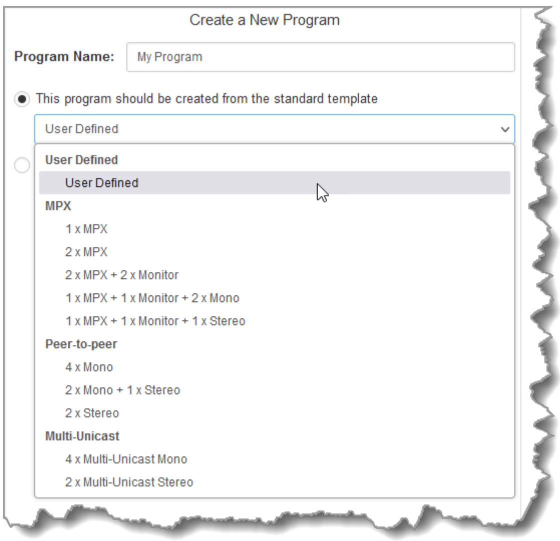
Product	Feature	Notification
MPX I	MPEG-TS Decode	Configure 1 MPEG-TS Decode stream.
MPX II	MPEG-TS Decode	Configure 1 or 2 MPEG-TS Decode streams.

Configure MPEG-TS Decoding Programs

MPEG-TS programs encapsulate a MicroMPX signal in MPEG-TS IP packets for transmission over IP to a destination codec. An MPEG-TS decode program can be configured to decode the MPEG-TS stream when it is received either over an IP interface, e.g. LAN1 or LAN2, or a DVB Satellite Receiver module connection.

Important Notes:
- Connections using uncompressed MPX or compressed µMPX are sessionless connections.
- A program cannot be edited in the **Program Manager panel** when loaded in the codec.
- Lock a loaded custom program or multistream program in a codec receiving a call 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 single connection dialing into the codec.
- 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 using default settings can save configuration time.
- SmartStream PLUS redundant streaming is not available with sessionless IP connections.
- Redundant MicroMPX connections can be configured by sending a stream to multiple destinations.
- A MicroMPX license needs to be purchased and installed to support MicroMPX.
- If a MicroMPX license is not installed then MicroMPX features will be unavailable or greyed out in menus.

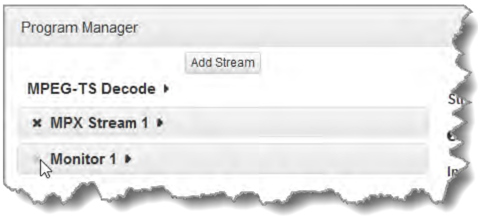
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option to create one or more new audio streams, then click **Next**.



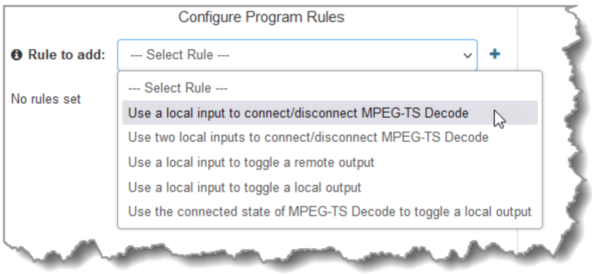
5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: the **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.

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.

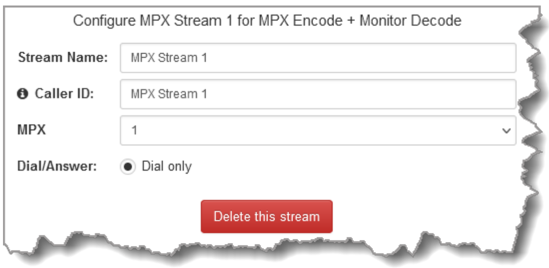
6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**.



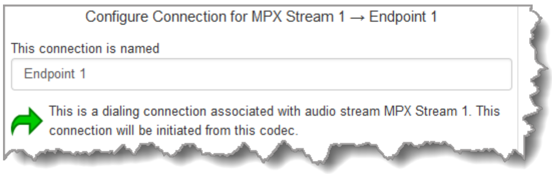
7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule. Note: Enlarge the width of the **Program Manager panel** to view and edit streams using the configuration settings available on the left-hand side of the panel. This makes it simpler to navigate to points in the wizard without navigating through multiple screens.



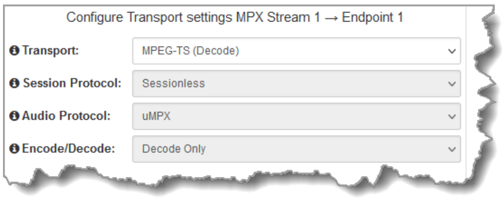
8. Enter the **Stream Name**, then click **Next**.



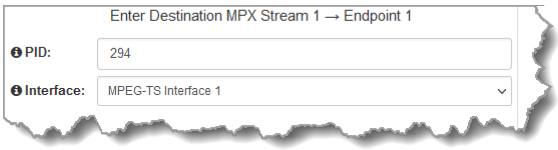
9. Enter the name of the connection in the text box and then click **Next**.

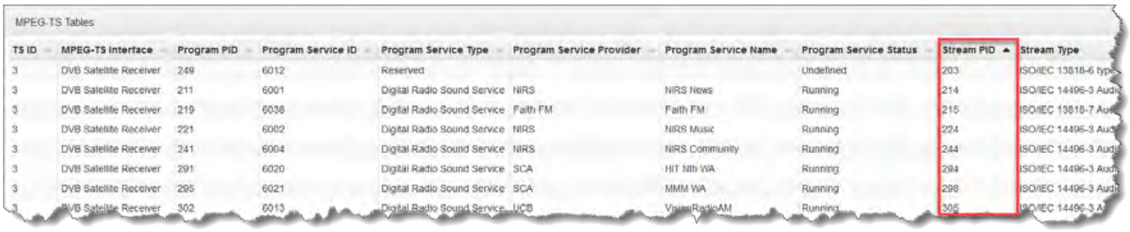


10. Select the **Transport** as **MPEG-TS (Decode)**, then click **Next**.



11. Enter the stream **PID** that can be sourced from the **MPEG-TS Tables panel** in the Toolbox Web-GUI and select the **Interface** from which to decode the MPEG-TS signal.




MPEG-TS Tables

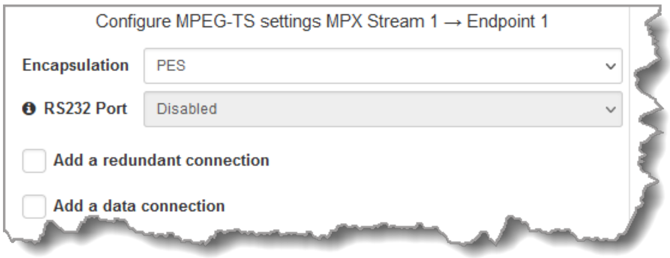
TS ID	MPEG-TS Interface	Program PID	Program Service ID	Program Service Type	Program Service Provider	Program Service Name	Program Service Status	Stream PID	Stream Type
3	DVB Satellite Receiver	249	6012	Reserved			Undefined	203	ISO/IEC 13818-6 type
3	DVB Satellite Receiver	211	6001	Digital Radio Sound Service	NIRS	NIRS News	Running	214	ISO/IEC 14496-3 Audi
3	DVB Satellite Receiver	219	6038	Digital Radio Sound Service	Faith FM	Faith FM	Running	215	ISO/IEC 13818-7 Au
3	DVB Satellite Receiver	221	6002	Digital Radio Sound Service	NIRS	NIRS Music	Running	224	ISO/IEC 14496-3 Au
3	DVB Satellite Receiver	241	6004	Digital Radio Sound Service	NIRS	NIRS Community	Running	244	ISO/IEC 14496-3 Audi
3	DVB Satellite Receiver	291	6020	Digital Radio Sound Service	SCA	HIT Nth WA	Running	294	ISO/IEC 14496-3 Aud
3	DVB Satellite Receiver	295	6021	Digital Radio Sound Service	SCA	MMM WA	Running	298	ISO/IEC 14496-3 Audi
3	DVB Satellite Receiver	302	6013	Digital Radio Sound Service	UCB	VisionRadioAM	Running	305	ISO/IEC 14496-3 A

MPEG-TS Tables panel

Important Notes: In addition to supporting an optional DVB Satellite Receiver module, the MPX II codec includes two **MPEG-TS Interfaces** in the **MPEG-ST Interfaces panel**.This allows MPX II codecs to receive 2 streams on different interfaces, e.g. **LAN1** and **LAN2**, when decoding two MPEG-TS streams.

12. Configure MPEG-TS settings and both PES and MPE encapsulation are supported. In addition:

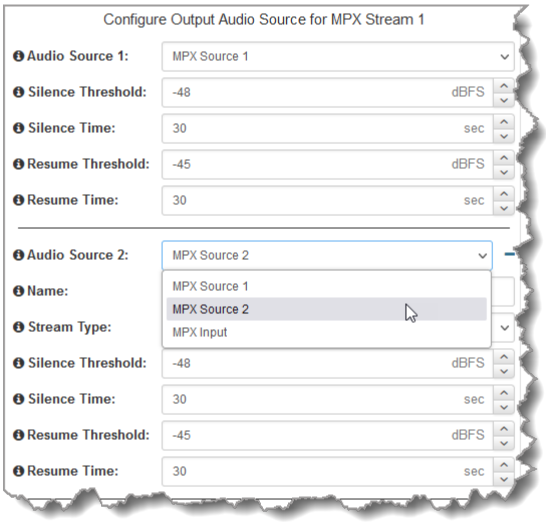
- A redundant connection can be decoded over a different **MPEG-TS Interface**. Select the **Add a redundant connection** check-box and click **Next** to configure this redundant connection.
- Select the check-box for **Add a data connection** to add up to two separate MPEG-TS data streams (RS-232), which can be decoded using separate Stream PIDs over different MPEG-TS interfaces. Data can be routed to specific codec RS232 ports. For more information on relay and RS-232 data options see RS232 Data Configuration.



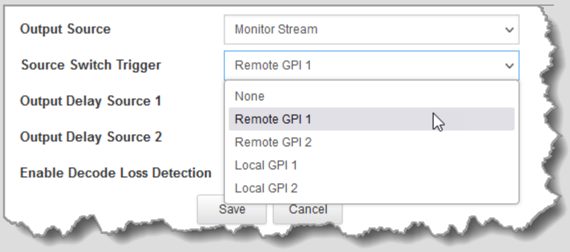
Important Notes

- The **Stream Identifier** relates to the **TS ID** in the **MPEG-TS Tables panel**. Range from 1 - 65535.
- The **Audio Elementary Stream PID** or **MPE PID** relates to the **Stream PID** in the **MPEG-TS Tables panel**. Range from 32 - 8190.
- The **Data Elementary Stream PID** relates to the **Stream PID** in the **MPEG-TS Tables panel**. This data stream requires a different Stream PID to the audio stream. Range from 32 - 8190.
- When a redundant connection is configured ensure that the **Output Delay Source 1/2** setting in the **MPX Settings panel** takes into account the transport latency. In other words, ensure the latency/buffer is higher than the latency over the network with the most jitter. There could be large differences in the latency of different paths if a satellite interface connection and a LAN interface are configured. As long as you keep the delay setting in the decoders the same, and there are no large differences in delays in some of the paths all decoders will play the audio at the same moment in time within a few milliseconds, which is good enough for seemless RDS AF frequency switching.
- The latency settings for **Output Delay Source 1/2** refer to **Audio Source 1** and **Audio Source 2** in the **Output Audio Source** settings in the **Program Manager panel**.

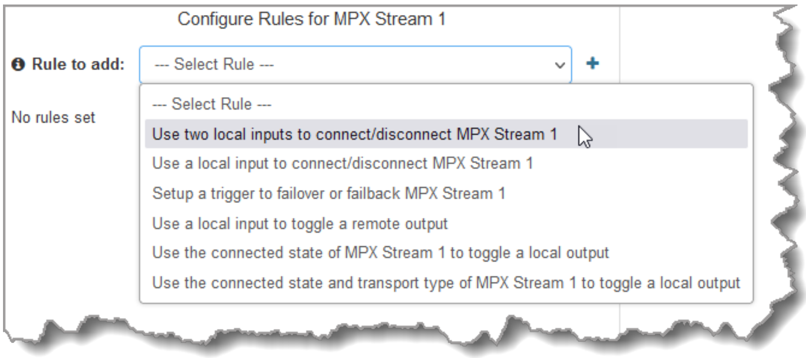
13. If two MPX inputs are attached to a codec you can configure an alternate **Output Audio Source** for the audio stream being configured. For example, **Audio Source 1** could be a satellite source and **Audio Source 2** could be local station content. Then click **Next**.



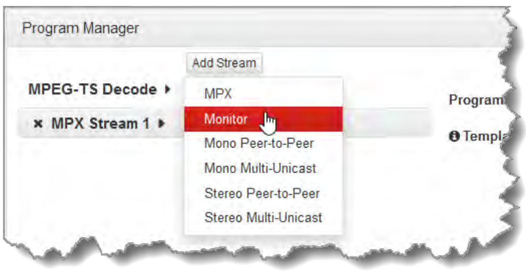
Important Note: In the **MPX Settings panel** you can select a local or remote GPI that will act as the trigger to switch between output sources.



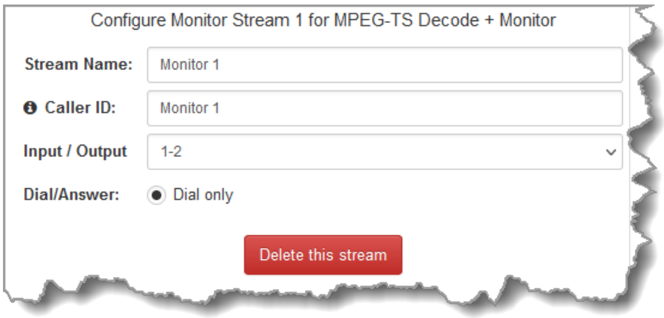
14. Continue configuring the transport settings for **MPX Source 2** like previously for **MPX Source 1**. Then configure any rules if required and click **Next**.



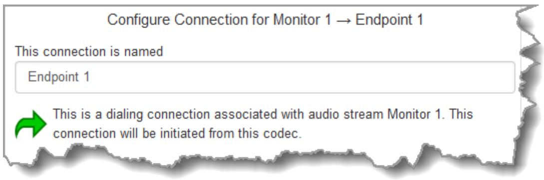
15. If configuration is complete click **Save**. If a confidence monitoring connection is required, click the **Add Stream** button to add a new **Monitor** stream.



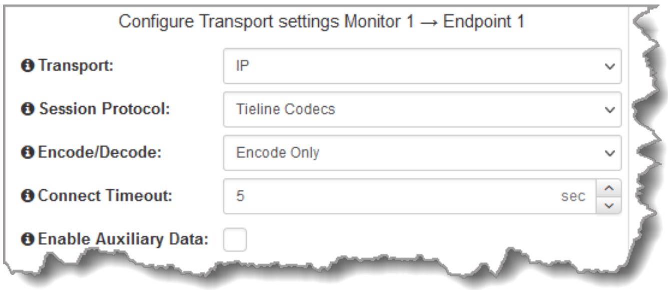
16. Then configure a dialing connection to encode and send the stream back to the studio or monitoring site, e.g. Master Control. Then click **Next**.



17. Enter a name for the connection and click **Next**.



18. See [Configure IP Streaming Programs] if required for more details on configuring Tieline IP stream connections. Otherwise continue through the wizard to the end and click **Save**. Note: Only Opus encoding is supported for sessionless or Tieline session Monitor connections.



19. The program can be loaded from within the **Program Manager panel** or the **Connections panel**. Load, Unload and Dial a Program using the **Connections panel**, or connect the program using the codec front panel.

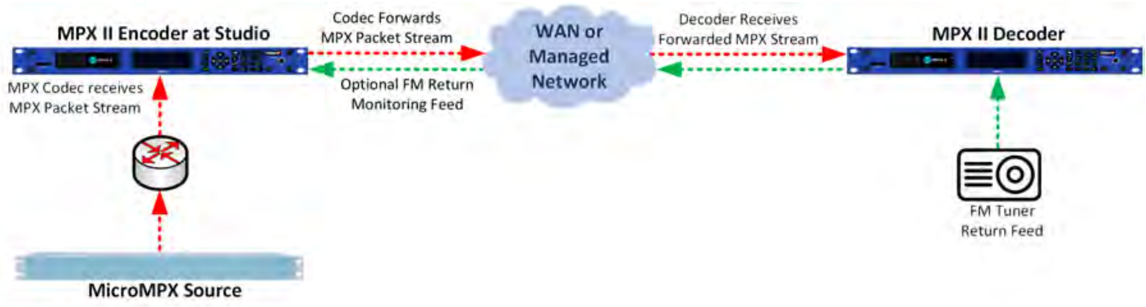
19.11 MicroMPX Forwarding Programs

Product	Feature	Notification
MPX I	MicroMPX Forwarding	Configure 1 MicroMPX Forwarding stream.
MPX II	MicroMPX Forwarding	Configure 1 or 2 MicroMPX Forwarding streams.

The codec supports MicroMPX packet forwarding in the following modes:

1. µMPX Forwarding: This mode allows a MicroMPX (UDP) stream to be received and then forwarded. For example, it is possible to receive a unicast and forward the signal as a multicast or visa versa. Along with MicroMPX forwarding, the user can send local GPIO data while maintaining synchronization with the MPX signal. Packets are given a new header and then forwarded.
2. µMPX forwarding + MPEG-TS over RTP: This mode allows a MicroMPX (UDP) stream to be received and then encapsulated and forwarded as MPEG-TS over RTP.
3. µMPX forwarding + MPEG-TS over UDP: This mode allows a MicroMPX (UDP) stream to be received and then encapsulated and forwarded as MPEG-TS over UDP.

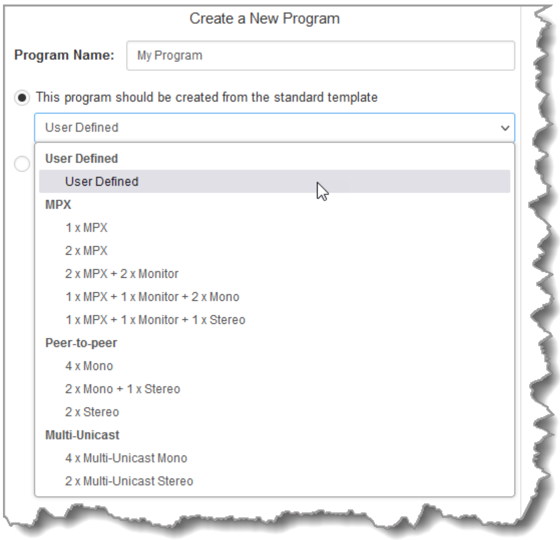
MicroMPX packet forwarding using MPEG-TS encapsulation delivers an internationally standardized and widely accepted format for distributing MPX signals over IP networks. It leverages the efficiency and reliability of MPEG-TS packetization, while ensuring the integrity of the MPX signal through the inclusion of MicroMPX headers.



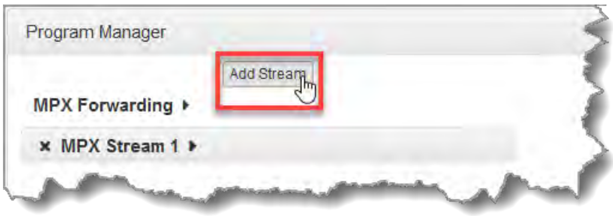
Example of MicroMPX forwarding between MPX codecs

Configure MicroMPX Forwarding Programs

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option to create one or more new audio streams, then click **Next**.

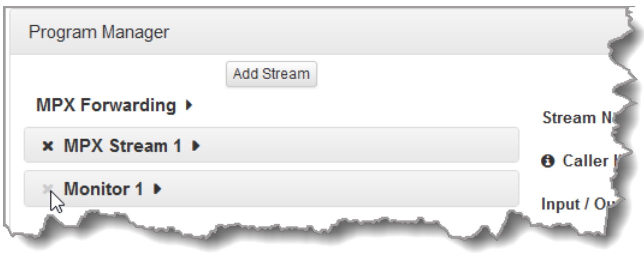


5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: the **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.

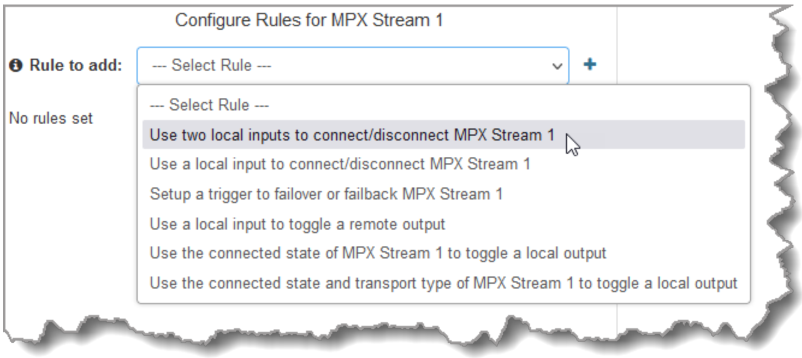


Important Note: When you decide to use an existing program as a template, the new program inherits all the settings of the template program and you can adjust these settings as required by continuing through the program wizard.

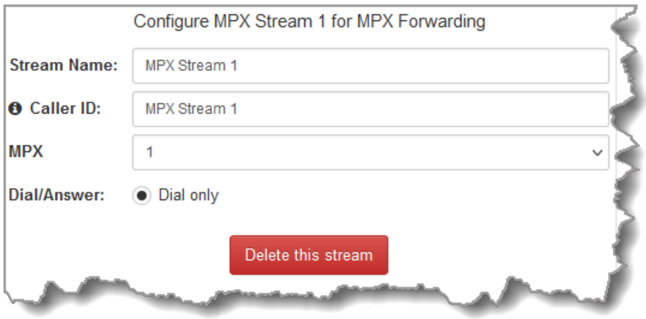
6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**.



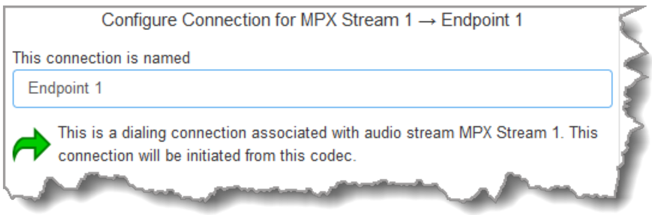
7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule. Note: Enlarge the width of the **Program Manager panel** to view and edit streams using the configuration settings available on the left-hand side of the panel. This makes it simpler to navigate to points in the wizard without navigating through multiple screens.



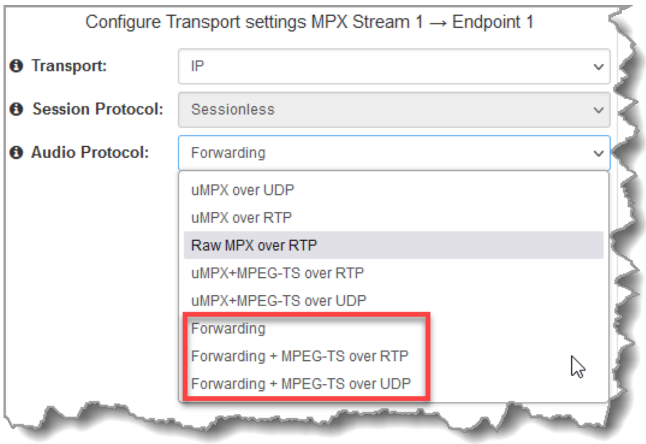
8. Enter the **Stream Name**, then click **Next**.



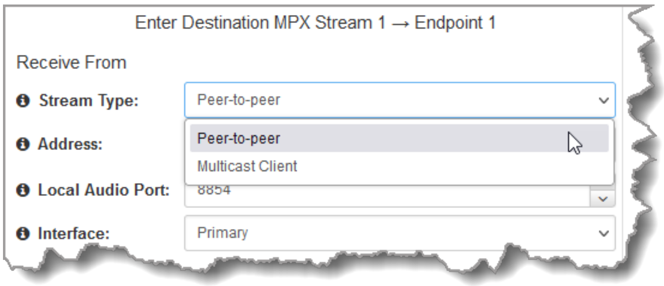
9. Enter the name of the connection in the text box and then click **Next**.



10. Select **IP** as the transport and select one of the available forwarding options, then click **Next**. Note:The **Audio Protocol** setting configured should match on both ends of the link.



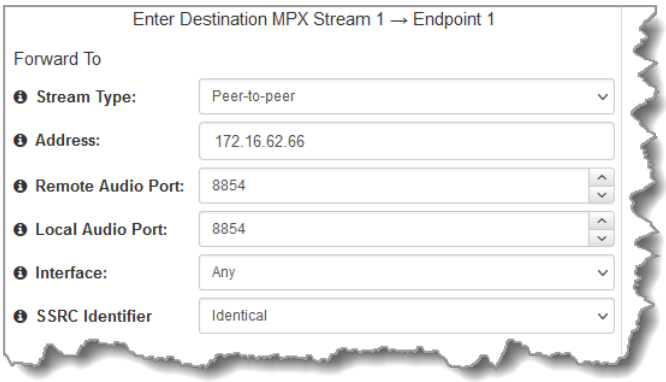
11. Configure the **Stream Type** to allow the codec to receive packets from a peer-to-peer source or as a multicast client, then click **Next**.



Important Notes:

- All connections using uncompressed MPX or compressed MicroMPX are sessionless.
- When the **Stream Type** is **Peer-to-peer** the configured **Interface** is used to receive packets.

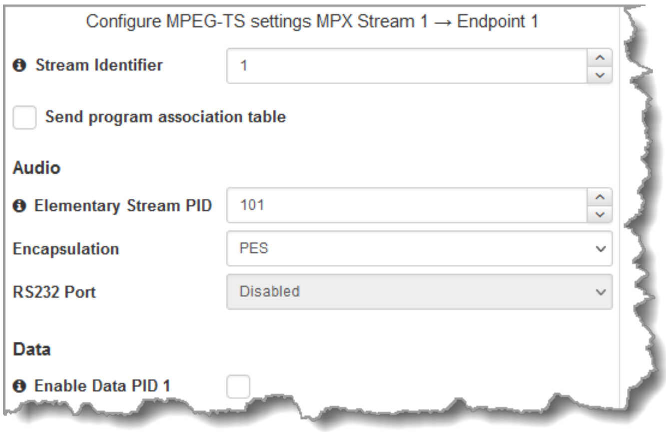
12. Configure the **Stream Type** in peer-to-peer or multicast server mode for forwarding the packet stream. In peer-to-peer mode configure a destination IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (**LAN1**) or **Secondary** (**LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable. The SSRC also needs to be configured correctly, then click **Next**.



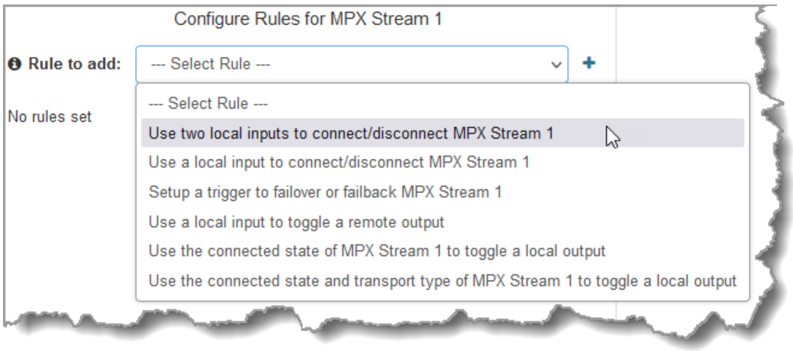
Important Notes:

- For both uncompressed MPX and MicroMPX connections the default audio port for MPX Stream 1 is 8854 and for MPX Stream 2 is 8874. There is no session port as these connections are sessionless.
- The default monitoring connection audio ports for MPX Stream 1 and MPX Stream 2 are 9010 and 9030 respectively.
- SSRC Identifier options include:
  - o **Unique**: Each connection in this stream uses a different SSRC. This follows the RFC7198 specification.
  - o **Identical**: Each connection in this stream uses the same SSRC. This follows the ST2022-7 specification.
- To create a multicast server or client program, simply enter a multicast address in the IP address field, e.g. 239.xxx.xxx.xxx and the connection will automatically be configured for multicasting. The **Multicast Time to Live (TTL)** setting can also be configured for a server program in an earlier screen.

13. If the **Audio Protocol** selected is **Forwarding + MPEG-TS over RTP** or **Forwarding + MPEG-TS over UDP**, additional settings should be configured, then click **Next**.



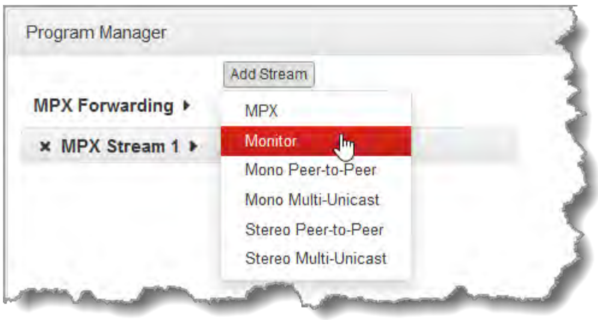
14. Configure stream level rules if required, then click **Next**.



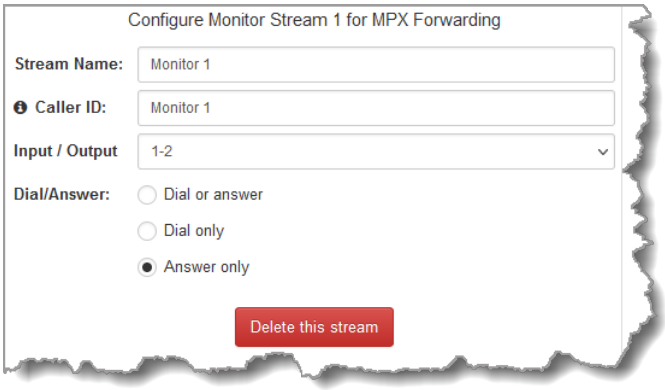
Add a Monitoring Stream

At this point is is possible to add/configure a monitoring stream if required to receive a confidence monitoring stream.

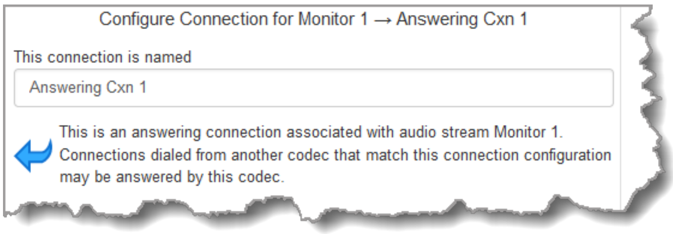
1. Click the **Add Stream** button to add a **Monitor** stream if it is not already visible in the program.



2. Usually this will be an answering connection that decodes confidence monitoring audio sent back by the codec at a transmitter site. Enter a **Stream Name**, **Caller ID** if required, and select the codec **Inputs/Outputs**. Then click **Next**.

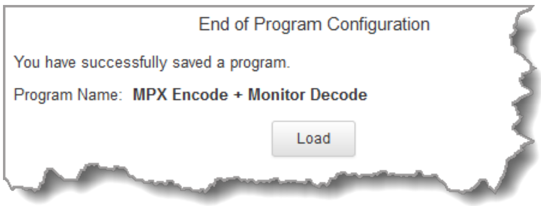


3. Enter a name for the incoming monitoring connection and click **Next**.



4. See Configure IP Streaming Programs if required for more details on configuring Tieline IP stream connections. Otherwise continue through the wizard to the end and click **Save**. Note: Only Opus encoding is supported for sessionless or Tieline session Monitor connections.

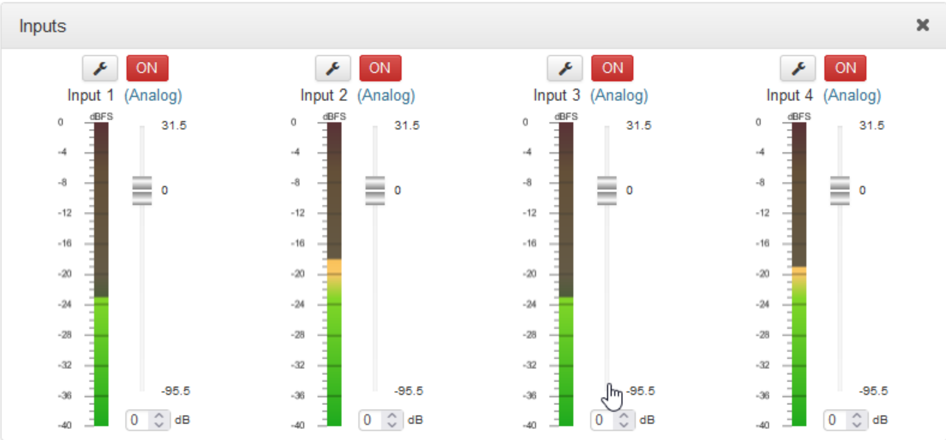
5. The newly created program can be loaded from within the **Program Manager panel** or the **Connections panel**. Load, Unload and Dial a Program using the **Connections panel**, or connect the program using the codec front panel.



19.12 Configure Non-MPX Inputs and Reference Levels for IP Links

Product	Feature	Notification
MPX I	Inputs	2 Channels can be configured as analog or digital inputs (AES3 or AoIP)
MPX II	Inputs	4 Channels can be configured as analog or digital inputs (AES3 or AoIP) in pairs

Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Inputs** to display the **Inputs panel**, which can be used to adjust analog and digital inputs attached to the codec. To adjust MPX signal inputs see Configuring MPX Inputs and Outputs.



Adjusting Non-MPX Audio Input Levels

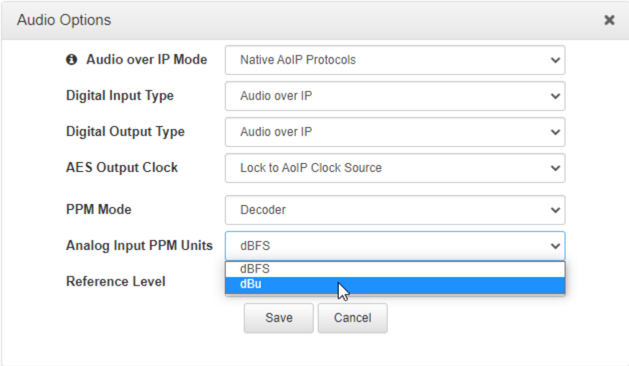
To adjust input audio levels, click on the input slider and drag it to the desired input gain level. Alternatively, type a value below a PPM meter to increase or decrease the input level value. Note: 0.5dB increments are accepted for increasing and decreasing audio levels. Input levels on the **Inputs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

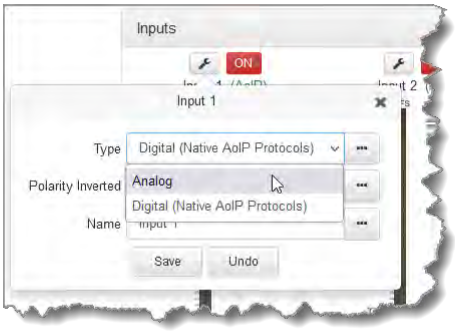


Configuring Input Settings

Selecting Analog and Digital Audio Sources

Codec inputs can be configured for AES3, digital (e.g. AES67, ST 2110-30, Livewire+, RAVENNA), 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 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 **Digital** (**AES3** or **Native AoIP Protocols**).

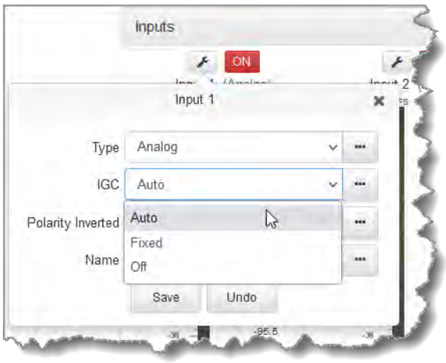


3. Click **Save** to confirm the new setting.

Adjusting IGC

Input IGC (Intelligent Gain Control) for analog inputs 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

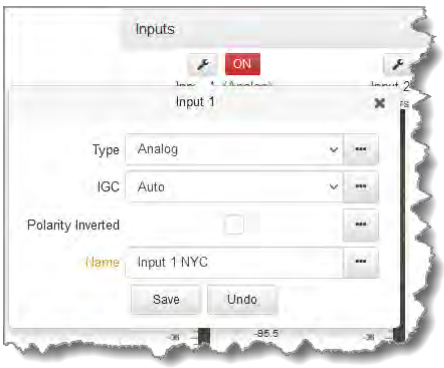
Select the **Polarity Inverted** check-box to reverse the polarity of an analog or digital input.

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.



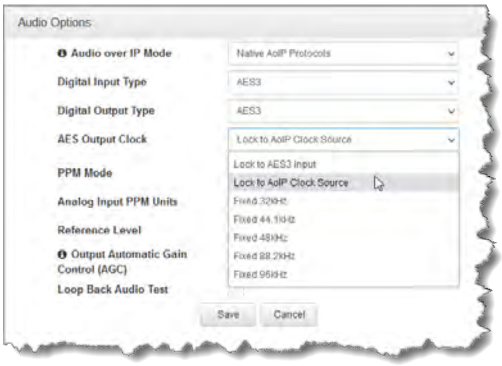
3. Click **Save** to confirm the name change.

Important Note: Audio signal processing occurs in the following order: IGC/Limiter (Intelligent Gain Control limiting on the input) > 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 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 for non-MPX connections. 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. For more information on reference levels see Input Levels and PPMs for IP Connections.

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.

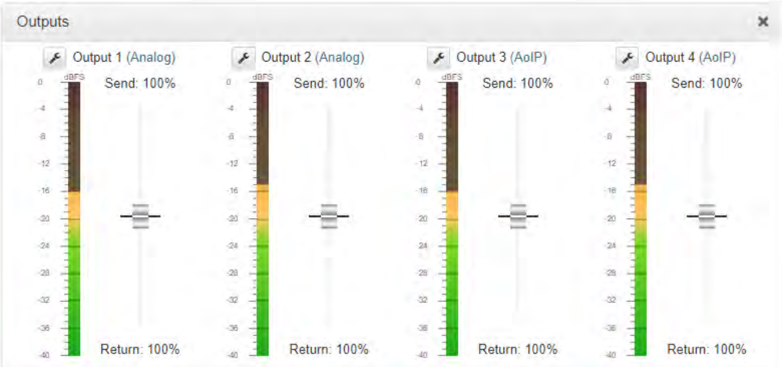
To adjust this setting:

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**.

19.13 Configuring Non-MPX Output Settings

Product	Feature	Notification
MPX I	Outputs	Outputs 1 and 2 can be configured as a pair of analog or digital inputs (AES3 or AoIP).
MPX I	Web-GUI Outputs panel	2 Outputs and PPMs displayed
MPX II	Outputs	Inputs 1-4 can be configured as analog or digital inputs (AES3 or AoIP) in pairs
MPX II	Web-GUI Outputs panel	4 Outputs and PPMs displayed

The **Outputs panel** can be configured to support either analog or digital output types. The digital output type (AES3 or AoIP) setting in the **Audio Options panel** configures all outputs. To adjust MPX signal outputs see Configuring MPX Inputs and Outputs.

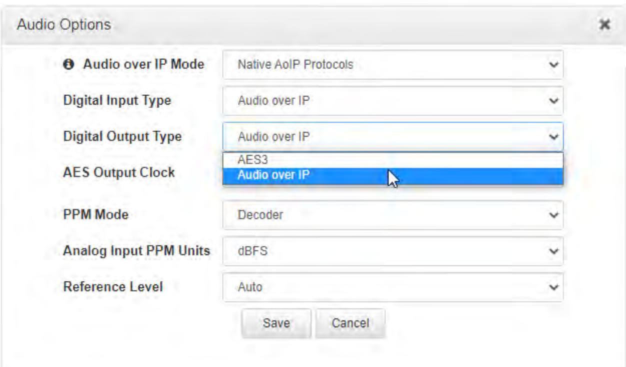


MPX II Outputs panel

Selecting the Non-MPX Digital Output Type

To select the digital output type for all outputs:

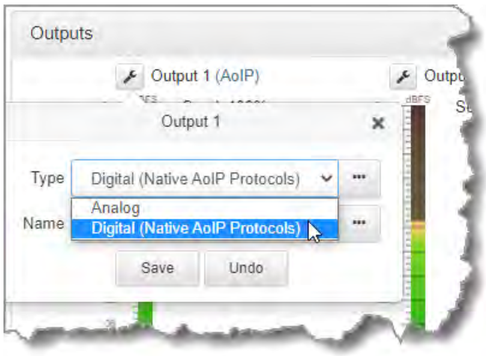
1. Open the HTML5 Toolbox Web-GUI and click **Audio** in the **Menu Bar**, then click **Audio Options** to display the **Audio Options panel**.
2. Click the **Digital Output Type** drop-down menu and select either **Audio over IP** or **AES3**, then click **Save**.



Select Output Type as Analog or Digital

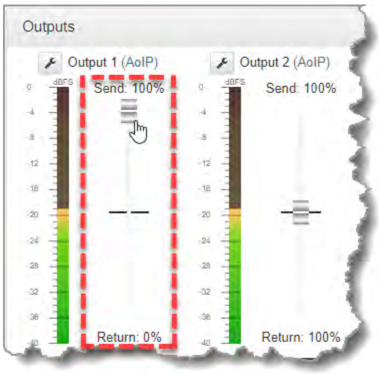
To select either analog or digital output types for outputs 1 to 8 (in pairs):

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** (**AES3** or **Native AoIP Protocols**).



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/ST 2110-30 streaming.

Output Assignments for AoIP Source Audio Streams

When streaming in AoIP mode (e.g. AES67, ST 2110-30, Livewire+, RAVENNA), the hard coded **Matrix Editor** output channel assignments depend on the stream configuration mode, as displayed in the following table.

AoIP Audio Stream	Mode	Mix Outputs Used
Source 1	Mono	1
	Stereo	1 and 2
	8 Channel	1 to 4
Source 2	Mono	2
Source 3	Mono	3
	Stereo	3 and 4
Source 4	Mono	4

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

19.14 Configure Non-MPX IP Streaming Programs

Product	Feature	Notification
MPX I	Programs	Supports program connections using up to 2 channels of audio, e.g. stereo, or dual mono.
	Program Manager panel	Displays program types that support 2 channels in/out
MPX II	Programs	Supports program connections using up to 4 channels of audio, e.g. 2 x stereo.
	Program Manager panel	Displays program types that support 4 channels in/out

The **Program Manager panel** incorporates a wizard to configure a new program and all audio stream settings. There is a range of preconfigured program templates which assist in configuring MPX, MPX Monitor, and non-MPX IP audio streams. Please note: The options available depend on how many channels the codec supports. Before you configure a new codec program consider whether:

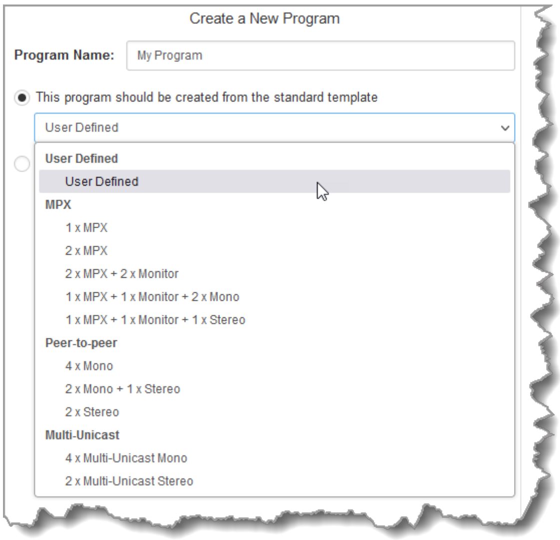
- An MPX connection will also have an associated Monitoring stream.
- Any IP connection is dialing and answering, dialing only, or answering only.
- A backup connection is required.

Create an IP Streaming Program

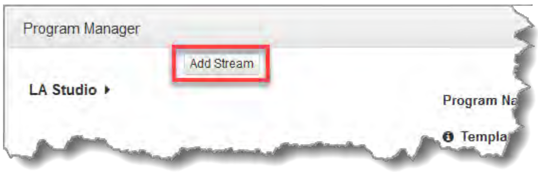
Important Notes: Before configuring a program please note:
- A program cannot be edited in the **Program Manager** when loaded in the codec.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different program type. For example, if a multistream program is not locked it will be unloaded by a single stream dialing into the codec.
- 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 using default settings can save configuration time.
- SmartStream PLUS redundant streaming is not available with sessionless IP connections.
- Redundant streaming for sessionless connections uses RFC 7198 or ST 2022-7 compliant specifications.

User Defined programs let you configure connections depending on how many channels the codec supports. Configure connections by selecting a standard program template and then remove or add streams and adjust configuration settings as required. This delivers flexibility for distribution of audio streams over wide area networks.

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option to create one or more new audio streams, then click **Next**.



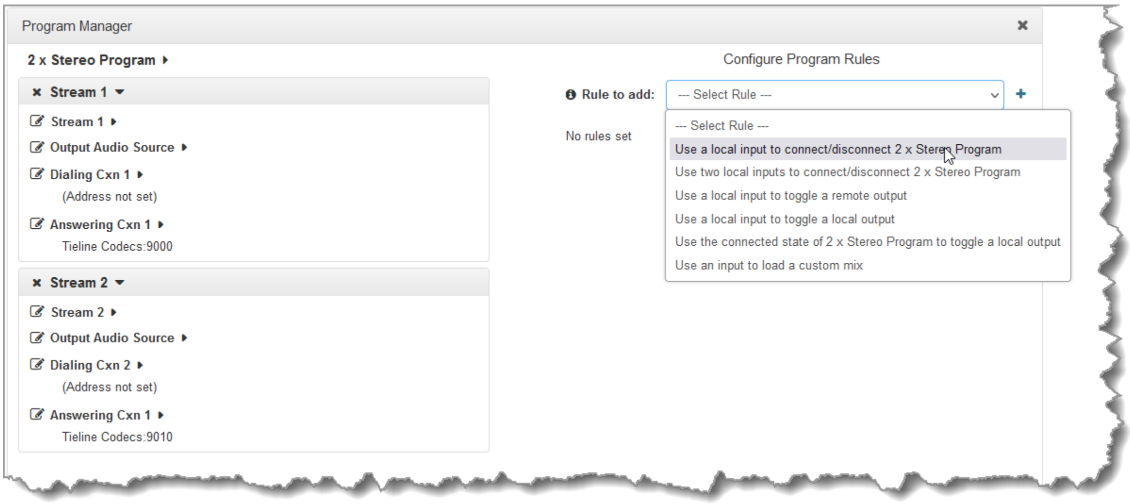
5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: the **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.



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.

6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**.

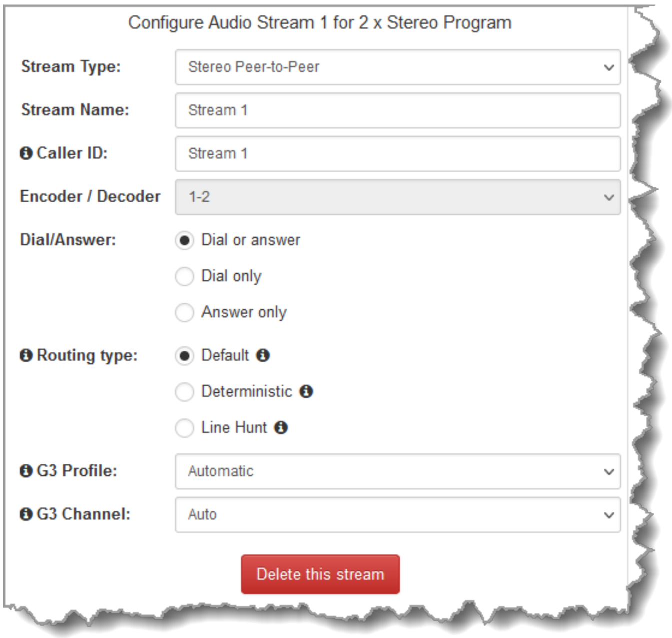
7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule. Note: Enlarge the width of the **Program Manager panel** to view and edit streams using the configuration settings available on the left-hand side of the panel. This makes it simpler to navigate to points in the wizard without navigating through multiple screens.



Important Notes for Rules:

- The MPX I codec has 4 hardware GPIOs and 60 logical outputs, and the MPX II has 8 hardware GPIOs and 56 logical outputs; both codecs also have 3 virtual inputs, and 64 Livewire GPIOs, or 64 WheatNet Logic Outputs. (WheatNet logic inputs are not supported in MPX codecs. WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.).
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- For more details about rules see Creating Rules.

8. Edit the **Stream Type**, 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. Note: The rest of the configuration wizard process is exactly the same as selecting a different preconfigured program type, e.g. **2 x Stereo Peer-to-Peer**.



Routing Type Options:	
Default	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	Use of **Dial** and **Answer Routes** is not usually necessary over IP because dedicated ports or **Line Hunt** mode call answering is used.
Line Hunt IP	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.

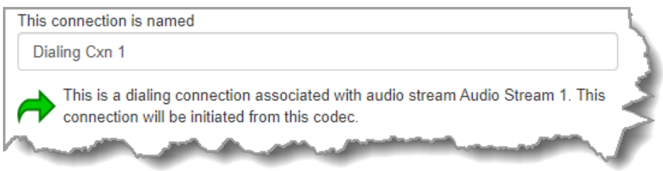
Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 or G6 codec.

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.

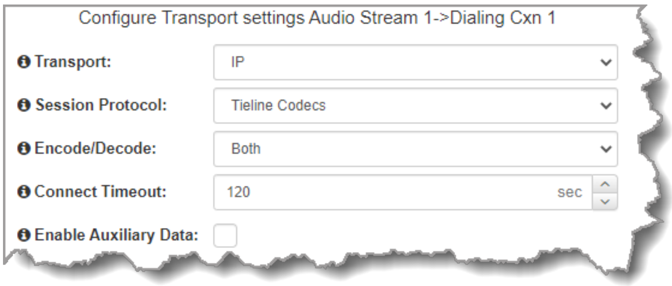
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

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).

9. 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**.



10. 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.

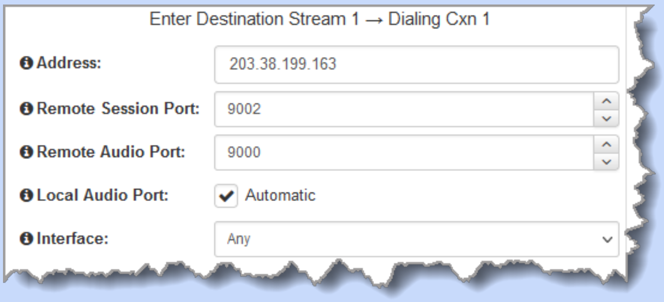


Important Note: See RS232 Data Configuration for detailed information on relay and RS232 data.

11. Configure destination codec dialing and encoding settings:

IP

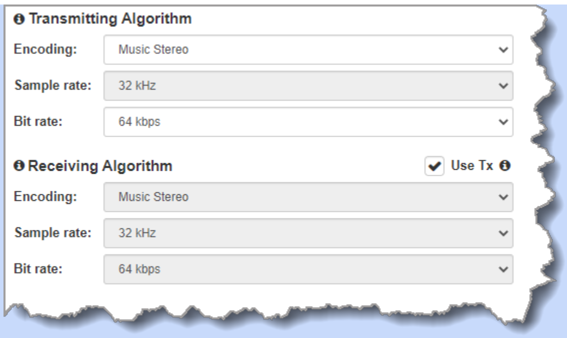
For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.



Important Notes:

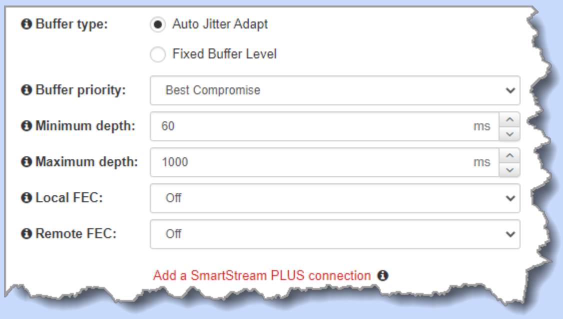
- To create a multicast server or client program, simply enter a multicast address in the IP address field, e.g. 239.xxx.xxx.xxx and the connection will automatically be configured for multicasting. The **Multicast Time to Live (TTL)** setting can also be configured for a server program.
- The **Remote 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**.

Click **Save** to save the program with the default algorithm, jitter and FEC settings. 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.



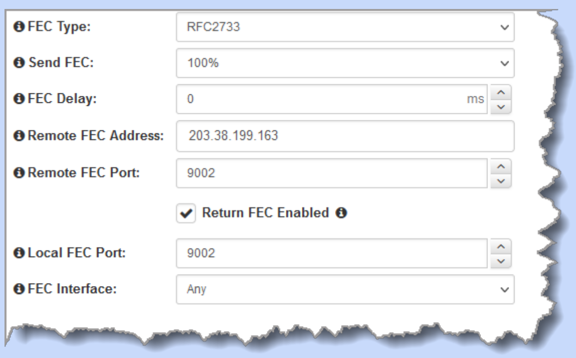
For IP connections click to configure:

- **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.



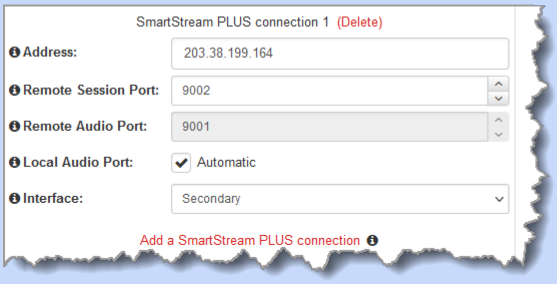
Important Notes:

- If you select **Sessionless** 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.
- To learn more about SmartStream PLUS visit https://tieline.com/smartstream-plus/

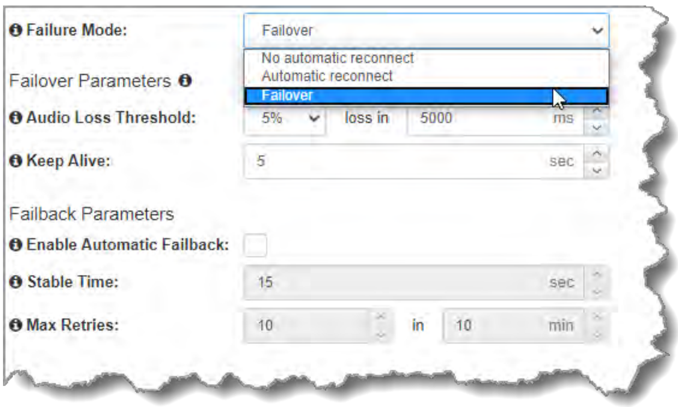
Configuring a Failover Connection or Auto Reconnect

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**.

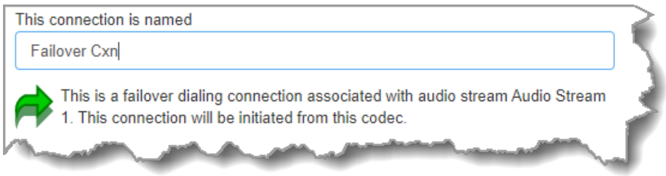
The codec will always attempt Failback if any of the following conditions are met:

- 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

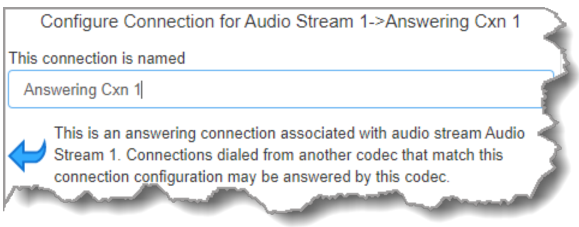
2. Enter a name for the failover connection and click **Next**.



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

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

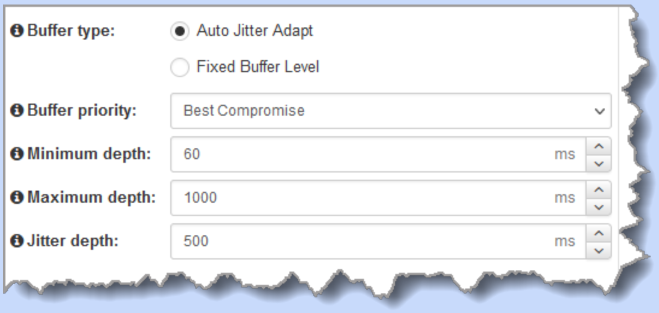


2. Configure the transport settings:

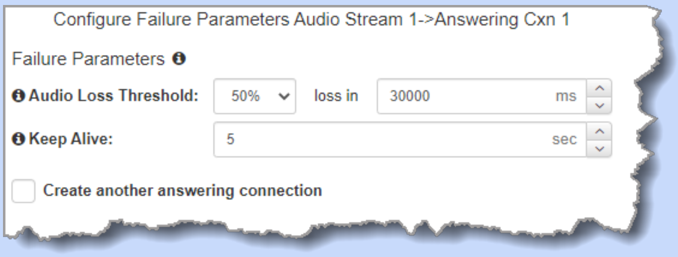
Select the **Session Protocol** and **Local Audio Port**.

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.



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.



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.

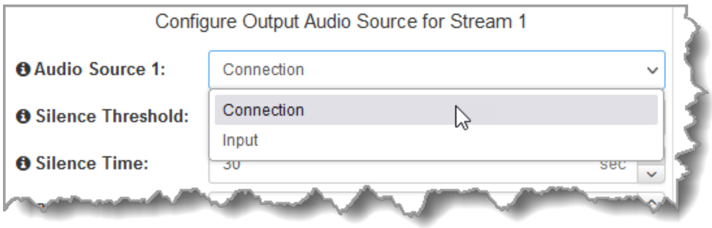
Configuring Output Audio Source Options

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

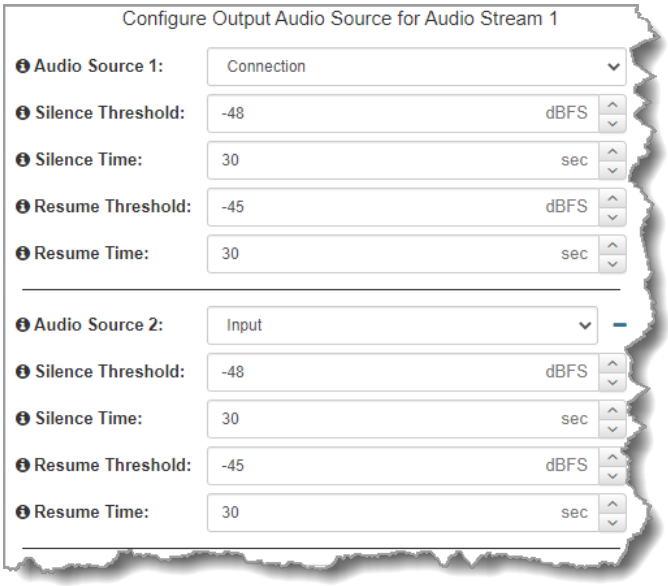
- **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
- **Input**: Input audio looped to the physical codec outputs.

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.

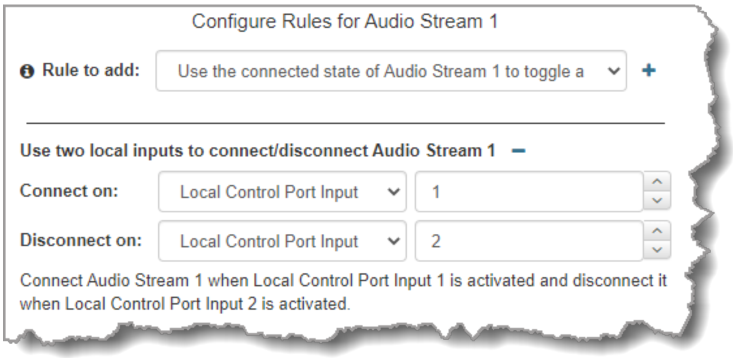


5. After configuring **Output Audio Source** options you can:

   i. Click **Save** to save the program at this point.
   ii. Click **Next** to configure rules options.
   iii. Click **Next Stream** to configure another audio stream (if available).

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.



2. Click **Save** to save the program.
3. Click **Next Stream** to configure another audio stream (if available).
4. Click **Finish** to exit the wizard.
5. The newly created program can be loaded from within the **Program Manager panel** or the **Connections panel**. Load, Unload and Dial a Program using the **Connections panel**, or connect the program manually using the codec front panel.

19.15 Configure Non-MPX IP Multi-Unicast Programs

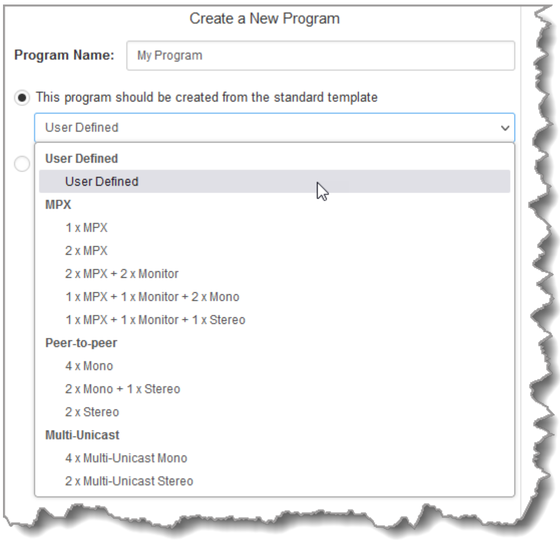
Product	Feature	Notification
MPX I	Multi-unicasting	Supports 1 multi-unicast IP audio stream being sent to up to 5 endpoints
MPX II	Multi-unicasting	Supports 2 multi-unicast IP audio streams being sent to up to 10 endpoints in total. For example, 2 separate audio streams can each be streamed to 5 endpoints.

Important Notes: Before you start program configuration please note:

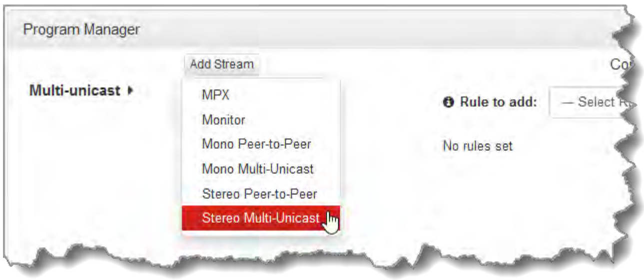
- You cannot edit a program when it is currently loaded in the codec.
- Lock a loaded custom program or multistream program in a codec to ensure it cannot be unloaded by a codec dialing in with a different type of program. For example, if a multistream program is not locked it will be unloaded by a mono 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.
- 24 bit, 48kHz PCM audio is 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 counted and included in the stream limits for the MPX I and MPX II codecs.
- 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.

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 **Create** button to open the wizard.
3. Enter a **Program Name**.
4. Click the drop-down menu and select the preferred option, e.g. to create one or more new multi-unicast audio streams, then click **Next**.



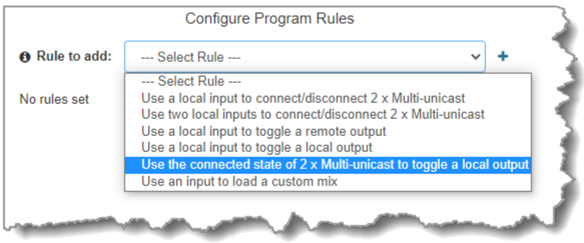
5. Alternatively, select **User Defined** and use the **Add Stream** button to add new connections as required. Note: the **Program Manager panel** needs to be expanded in size to see the **Add Stream** button.



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.

6. To remove a stream click the cross symbol next to the stream on the left-hand side of the **Program Manager panel**.

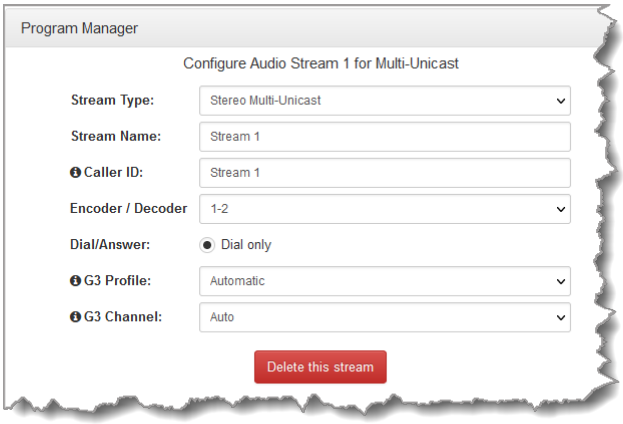
7. To configure new program level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** to add a new rule and click the **Minus symbol** to remove a rule. Then click **Next**.



Important Notes for Rules:

- The MPX I codec has 4 hardware GPIOs and 60 logical outputs, and the MPX II has 8 hardware GPIOs and 56 logical outputs; both codecs also have 3 virtual inputs, and 64 Livewire GPIOs, or 64 WheatNet Logic Outputs. (WheatNet logic inputs are not supported in MPX codecs. WheatNet logic I/Os allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.).
- The **Enable Livewire GPIO** check-box must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- For more details about rules see Creating Rules.

8. Enter the **Stream Name** and add a **caller ID**. Then click **Next**. Note: The caller ID is used to identify calls.



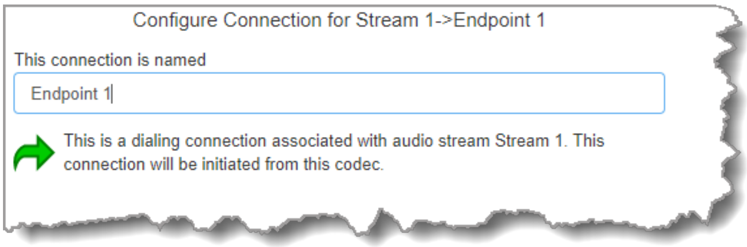
Important Notes on G3 Profile Settings:
The G3 profile setting supports maintaining specific G3 codec settings when answering a call from a G5 codec.

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.

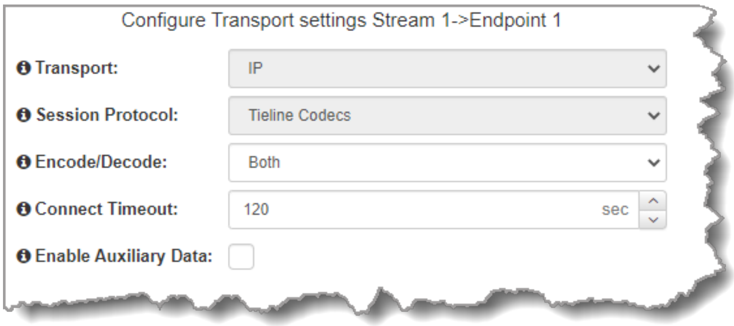
Important Notes on G3 Channel Settings:
This setting is for compatibility with the **Dual Mono** profile in Tieline Commander G3 and i-Mix G3 codecs. It is designed to configure routing of the audio stream to a specific G3 codec channel consistently.

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).

9. 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**.



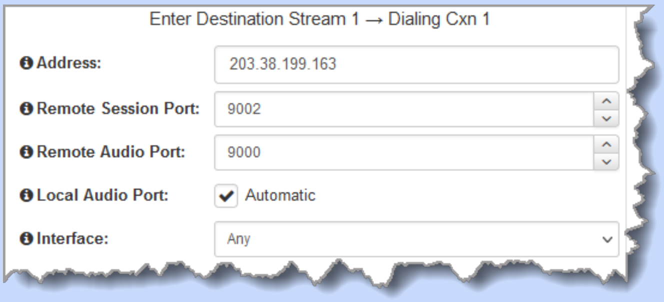
10. Configure the transport settings for the connection, either **IP** or **RIST**, then click **Next**. Note: only the first connection dialed in a multi-unicast audio stream can encode and decode audio. All other connections are unidirectional and encode only.



Important Note: See [RS232 Data Configuration]() for detailed information on relay and RS232 data.

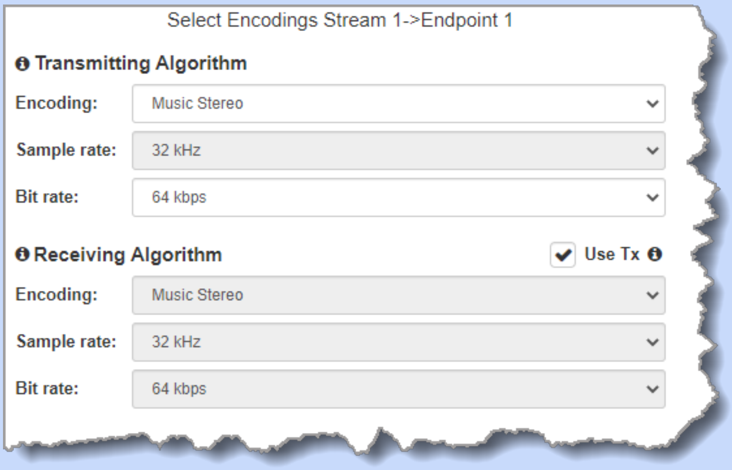
11. Configure destination codec dialing and encoding settings:

For IP connections configure the IP address, ports, and then specify which streaming interface is used to dial this connection, e.g. **Primary** (port **LAN1**) or **Secondary** (port **LAN2**). Note: By default **Any** will select **LAN1** if it is available and **LAN2** if it is unavailable.

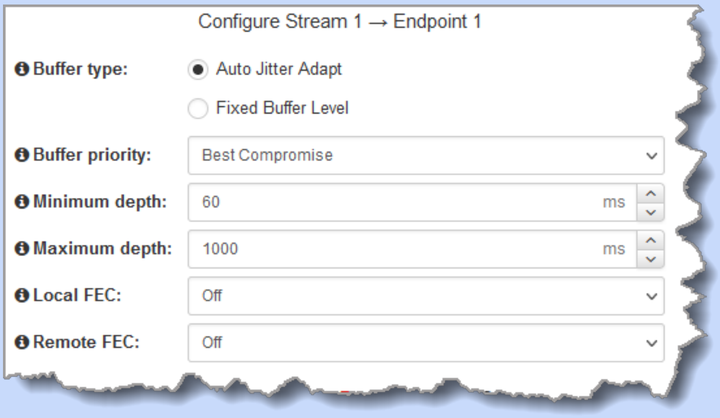


Important Note: The **Remote Audio Port** is the codec port at the remote end of the link to which you are sending audio. The **Local Audio Port** is used by the local codec to receive audio from the remote codec. When **Tieline Codecs** is the **Session Protocol** selected (using Tieline session data), 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.

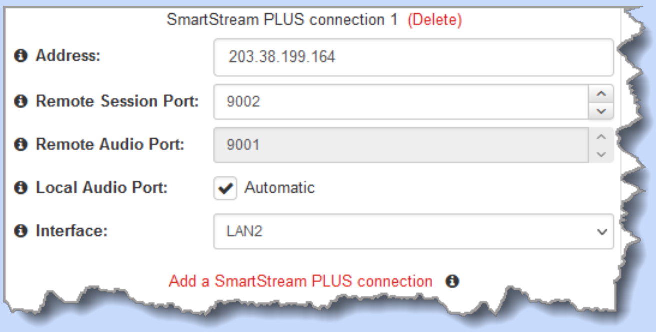
Click **Next Stream** to configure the first audio stream with default algorithm, jitter and FEC settings. Alternatively, click **Next** to specify individual algorithm, jitter buffer and FEC settings (recommended) and more connection endpoints. Click the drop-down arrows on the right-hand side of each 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.



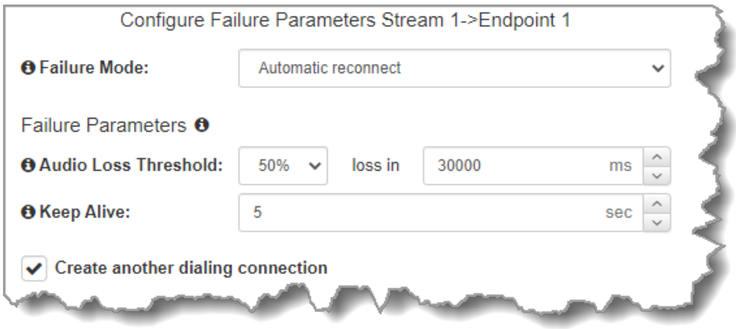
By default, primary IP streaming is via **LAN1**. To achieve the maximum level of redundancy select **Secondary** to configure redundant streaming from the secondary IP port **LAN2**. The redundant stream uses **Remote Audio Port 9001** by default and the **Local Audio**

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.

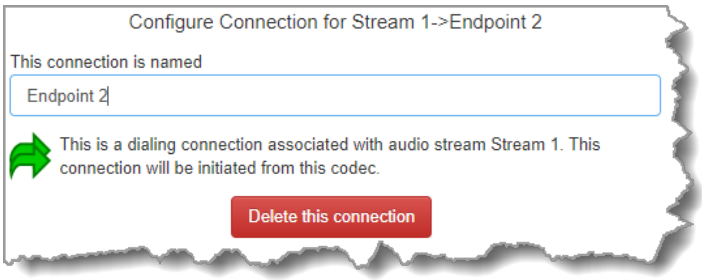
Important Notes:

- 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.

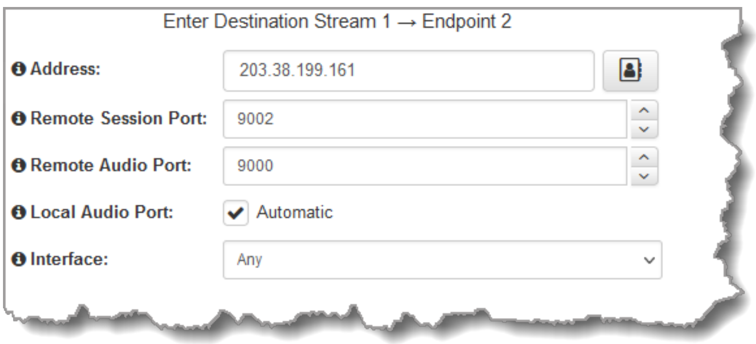
12. Click the drop-down **Failure Mode** arrow and select **Automatic reconnect** to enable this feature and configure **Failure Parameters**. To add another stream endpoint select the check-box for **Create another dialing connection**, then click **Next**.



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

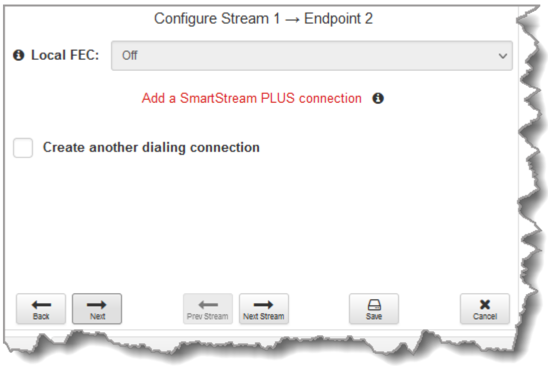


14. Enter destination settings for the dialing connection, then click **Next**.



15. There are four options at this point:

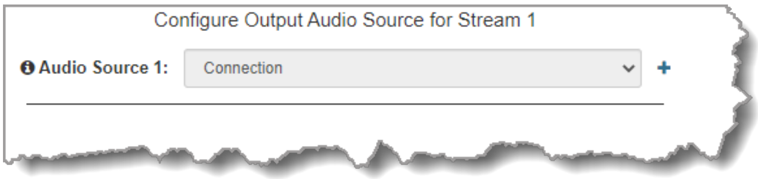
- Click **Add a SmartStream PLUS connection** to add a redundant stream for the current connection.
- Select **Create another dialing connection** to create another stream endpoint.
- Click **Next** to configure the **Output Audio Source** for the audio stream.
- Click **Next Stream** to configure the second multi-unicast audio stream.



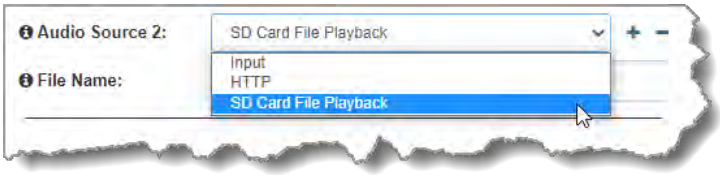
Configuring Output Audio Source Options

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:

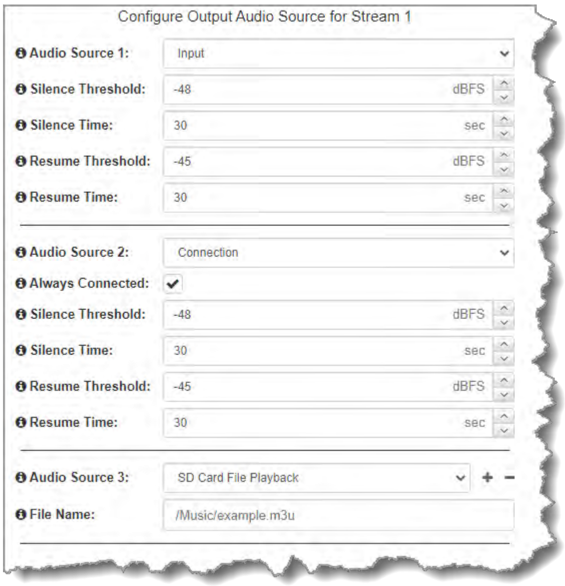
   - **Connection**: Decoded connection audio sent from a remote codec (Note: this must be selected as one of the configured sources).
   - **Input**: Input audio looped to the physical codec outputs.



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.



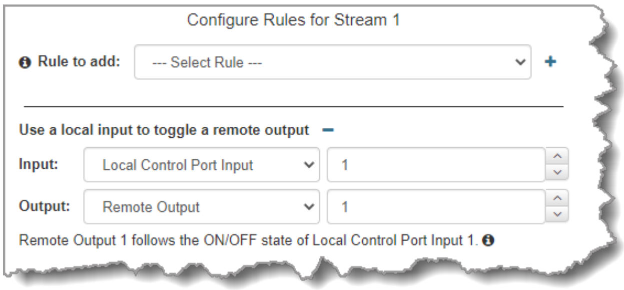
4. Configure silence threshold parameters for enabling a preferred backup option, as well as resume thresholds for reactivating a previous source.



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.

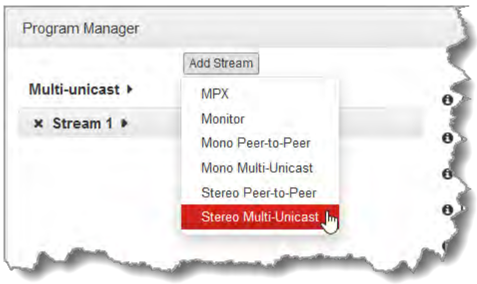
6. To configure new stream level rules click the drop-down arrow and select the preferred option from those available. Click the blue **Plus symbol** ➕ to add a new rule and click the **Minus symbol** ➖ to remove a rule.



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

Configure a Second Multi-unicast Audio Stream

1. Click the **Add Stream** button to add a second multi-unicast audio stream if required as this is supported in the MPX II codec.



2. Continue through the program wizard and configure all multi-unicast connections for this audio stream like the first 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.

3. Click **Save** to save the program settings, then click **Finish** or **Load** to load the new program.
4. The newly created program can be loaded from within the **Program Manager panel** or the **Connections panel**. Load, Unload and Dial a Program using the **Connections panel**, or connect the program manually using the codec front panel.

19.16 Configure Programs with the Connections Panel

The **Connections panel** allows:

- The configuration of new programs.
- Editing of existing programs.
- Loading and unloading of programs.

The **Connections panel** also delivers the ability to edit settings for an audio stream when other audio streams are connected, without unloading a program.

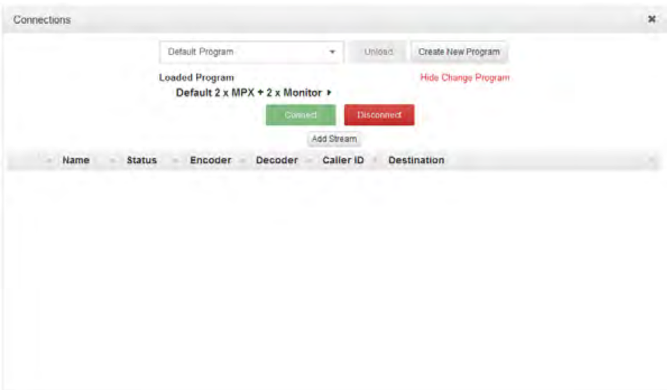
Configuring New Programs

Important Notes: 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:

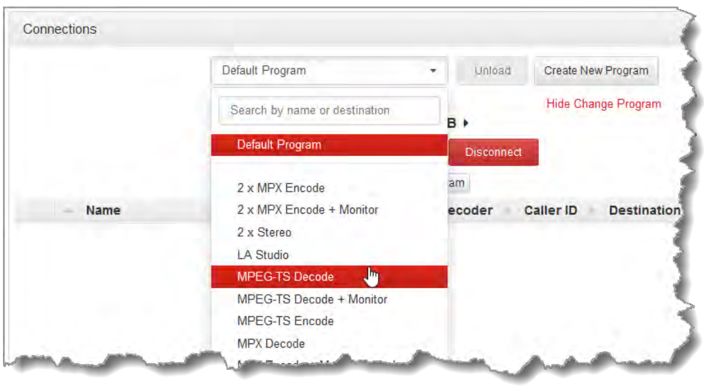
- Name or rename a program.
- Manage program level Rules.
- Create a program from a copy of an existing program.

To configure new programs using the **Connections panel**:

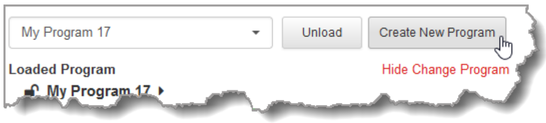
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**. By default a **2 x MPX + 2 x Monitor program** is loaded.



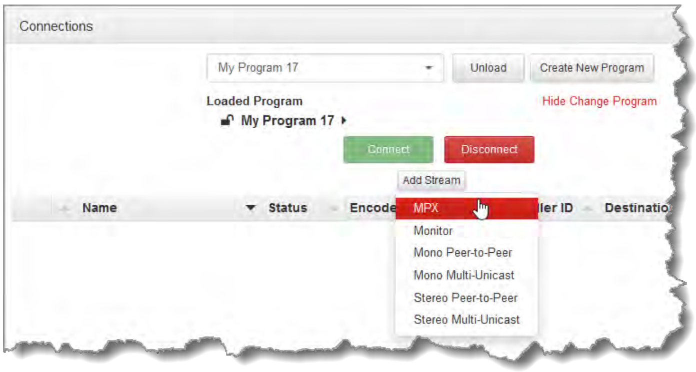
2. The default program can be edited as required, or if a custom program has been previously configured click the drop-down arrow to select it.



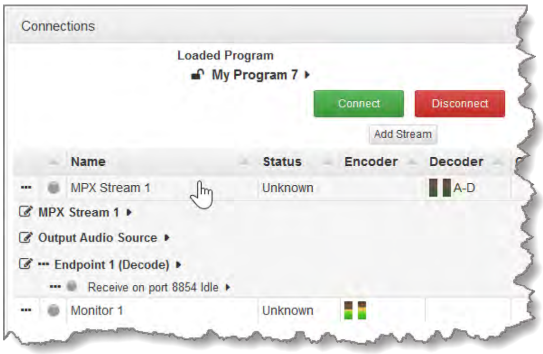
3. If you want to create a new program click the **Create New Program** button to create a completely new program.



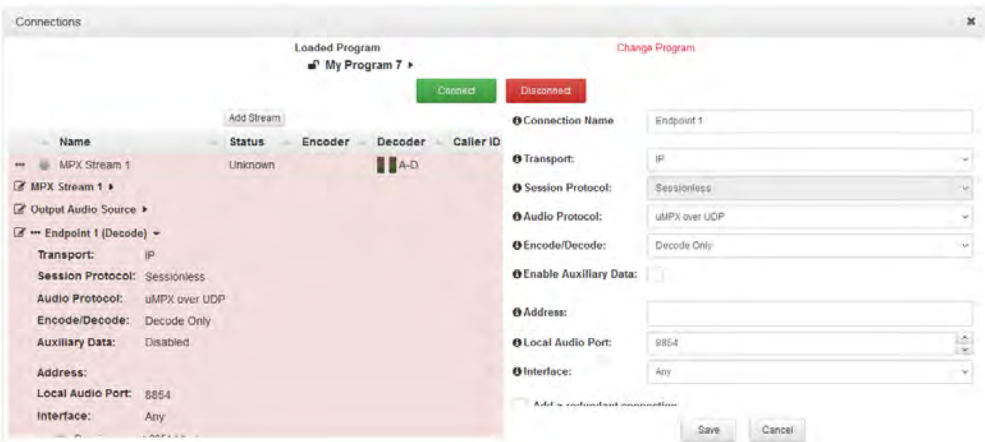
4. Click the **Add Stream** button to add a new stream to the program.



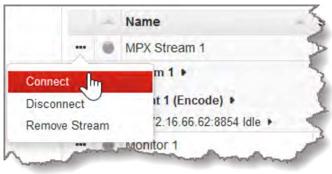
5. Add additional streams as required and then click on a stream to expand it.



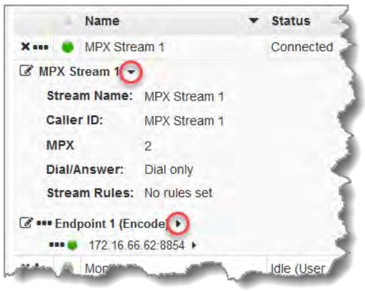
6. The **Edit symbol** expands each audio stream to reveal configuration settings. Hover over the **Information symbol** to view details of each configuration setting. Click **Save** to store changed settings. Note: Configuration of different MPX and non-MPX connections is explained in more detail in earlier sections.



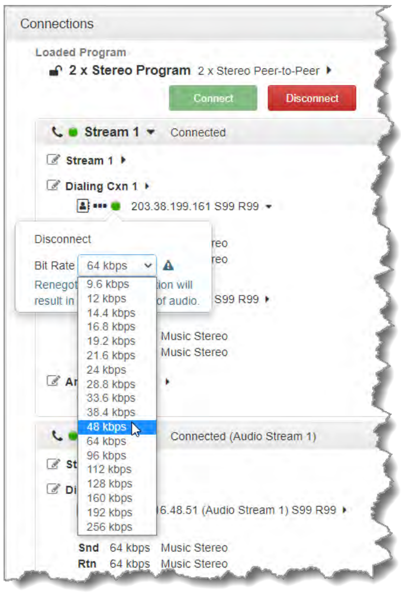
7. To connect an individual connection, click the **3 dot** ··· icon and then click **Connect**.



8. Click the right-facing arrows ▶ to expand stream settings. Click the down-facing arrows ▼ to collapse expanded stream settings. Audio streams can be connected, disconnected and managed from within the **Connections panel**.

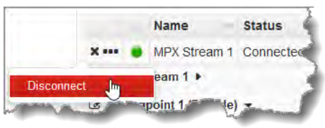


9. Click the **3 dot** ··· icon for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate for a non-MPX IP connection. Note: The bit-rate for a MicroMPX connection can be adjusted in the **MPX Settings panel**.



Important Note: Renegotiation will result in a temporary loss of audio.

10. To disconnect an individual connection, click the **3 dot** ··· icon for an active connection and then click **Disconnect**.



19.17 Monitoring PPMs

Product	Feature	Notification
MPX I	PPM Meters	4 PPMs on front panel display 2 channel inputs/outputs or encoders/decoders
MPX I	Web-GUI PPMs Panel	Capable of monitoring 4 inputs, 4 outputs, 4 encoders, 4 decoders, and headphone output simultaneously
MPX II	PPM Meters	8 PPMs on front panel display 4 channel inputs/outputs or encoders/decoders
MPX II	Web-GUI PPMs Panel	Capable of monitoring 4 inputs, 4 outputs, 4 encoders, 4 decoders, and headphone output simultaneously

The **PPMs panel** is able to monitor incoming and outgoing audio streams in the codec.

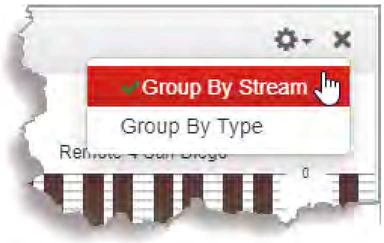
PPMs grouped by stream and identified by Stream Name

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

- Inputs
- Outputs
- Encoders
- Decoders
- Headphone (output)

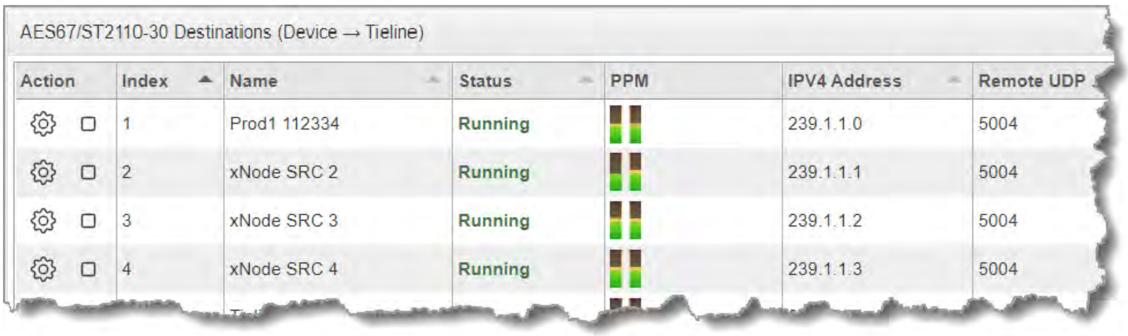
Select the check-boxes to select or deselect the PPMs displayed for each audio stream.

Click the **Options** menu in the panel to choose whether PPMs are grouped by stream or by type.

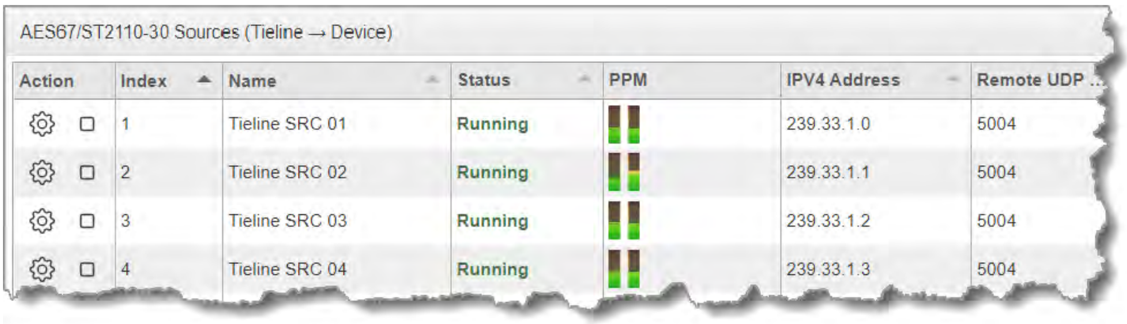


AoIP Stream PPMs

PPMs for AoIP (e.g. AES67, ST 2110-30, Livewire+, RAVENNA) streams are displayed in the **Destinations panel** and **Sources panel** in the Toolbox HTML5 Web-GUI and the AoIP Web-GUI. For more details please download the AoIP User Manual.



Toolbox Destinations panel displaying incoming codec audio

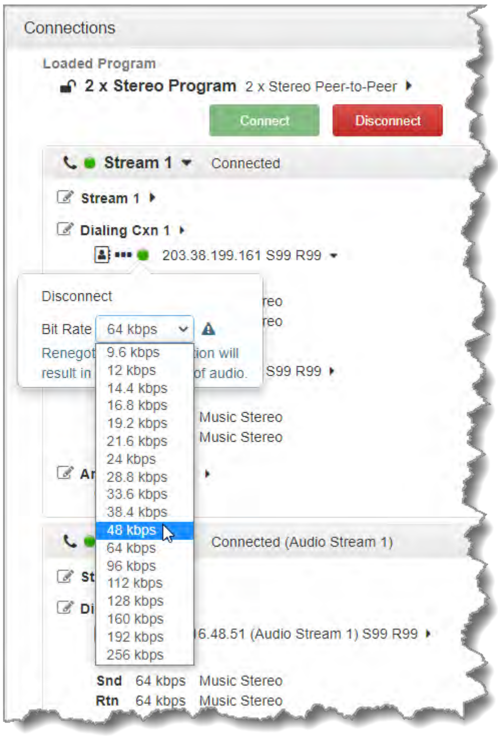


Toolbox Sources panel displaying outgoing codec audio

Important Note: PPMs on the AoIP **Destinations panel** are inactive if the **Digital Input Type** in the **Audio Options panel** is not set to **Audio over IP** and the corresponding **Inputs panel** input **Type** is not set as **Digital**.

19.18 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 **3 dot** ••• icon for an active connection and then click the drop-down **Bit Rate** arrow to select and renegotiate a new bit-rate for non-MPX connections.



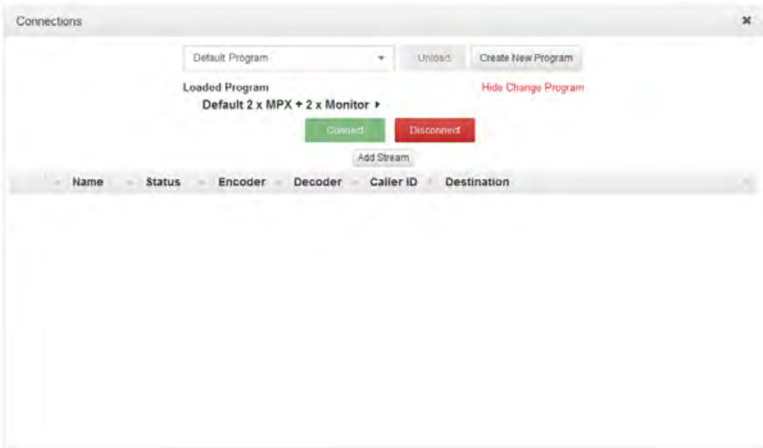
Important Note: Renegotiation will result in a temporary loss of audio.

19.19 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.

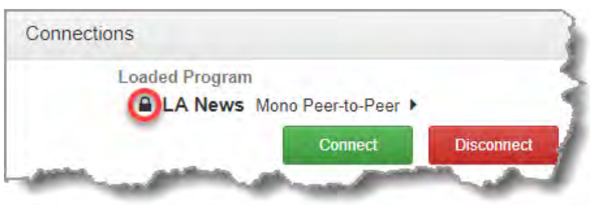
Load and Unload a Program

1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Connections** to launch the **Connections panel**.



2. Click the drop-down arrow to select one of the custom programs available in the codec, then click **Load**.

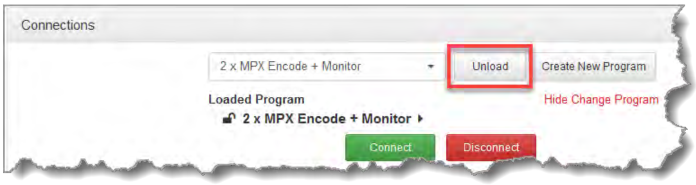
3. If **Lock Loaded User Program** has been configured in the **Options panel**, a black **Padlock** symbol appears next to the program name in the **Connections panel**, to indicate a program is locked in the codec.



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 **3 dot** ··· icon and then click **Connect**; this dials all connections associated with an audio stream.

Connection LED States

The **Connection State LED** displays status of an audio stream connection as follows:

- Gray: Not In use
- Green: Connected and stable (LQ greater than 70)
- Yellow: Connected but connection quality is not stable (LQ 50 to 70). Note: ISDN connections do not display the yellow LED state
- Red: Connection establishing or problem with connection (LQ less than 50). LED flashes when establishing connection

Disconnecting a Program

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

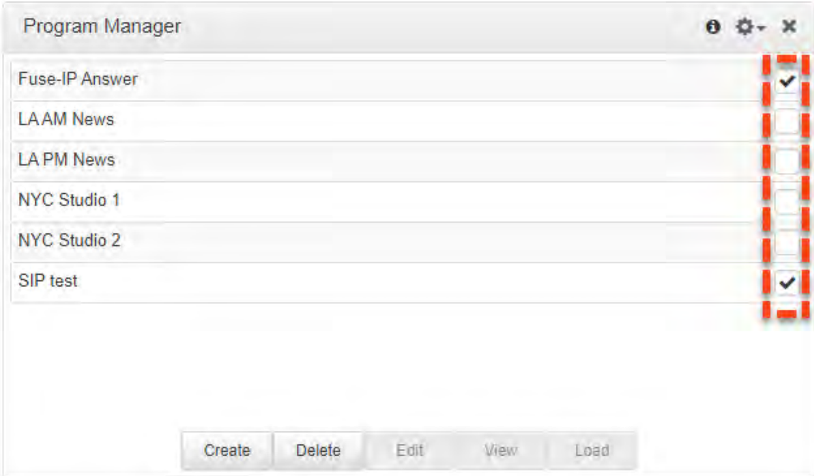
1. Click the **Disconnect** Disconnect button to disconnect all audio streams and connections configured in a program.
2. Click the audio stream **3 dot** ··· icon and then click **Disconnect** to disconnect an individual audio stream and all associated connections.

Change Destination Dialing Settings

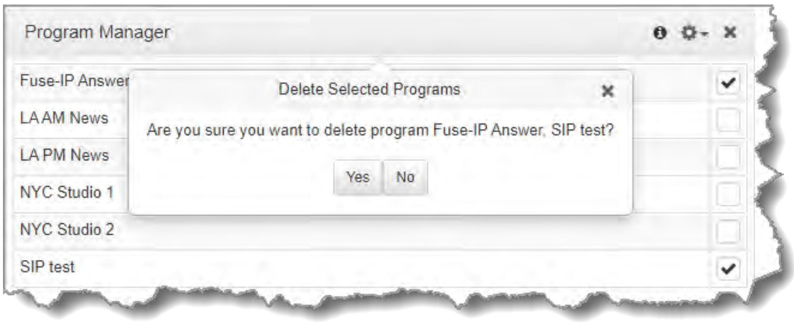
1. Click the **Edit symbol** adjacent to a connection. Note: The connection must be disconnected before editing.
2. Adjust dialing and connection settings and then click **Save** to change edited settings in the program.

19.20 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.



19.21 Default Program Ports

Product	Feature	Notification
MPX I	Program Manager panel	• Supports port configuration for 1 or 2 peer-to-peer audio streams • Supports port configuration for one multi-unicast and multicast audio stream • Supports port configuration for an MPX stream and a monitor connection
MPX II	Program Manager panel	• Supports port configuration for 1-4 peer-to-peer audio streams • Supports port configuration for 2 different multi-unicast audio streams, or 2 different multicast streams • Supports port configuration for 2 MPX streams and 2 monitor connections

The following table displays the default TCP and UDP ports used when configuring non-MPX 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 Web-GUI use and NMOS etc., 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

Default Multi-Unicast Audio Stream Ports

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

Default Multicast Audio Stream Ports

Multicast Audio Stream	TCP Session Port	UDP Audio Port
1	9002	9000
2	9002	9010

Default MPX Port Settings

MPX connections are sessionless so do not use a session port. However, monitoring connections associated with these connections that connect over regular IP stream connections require use of a session port. The default ports are as follows:

- UDP audio port 8854 is the default for MPX Stream 1 connections (Raw uncompressed and MicroMPX).
- UDP audio port 8874 is the default for MPX Stream 2 connections (Raw uncompressed and MicroMPX).

TCP port 9002 is the default session port for Monitor Stream 1 and Monitor Stream 2 connections. Ports 9010 and 9030 are the default audio ports.

19.22 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).

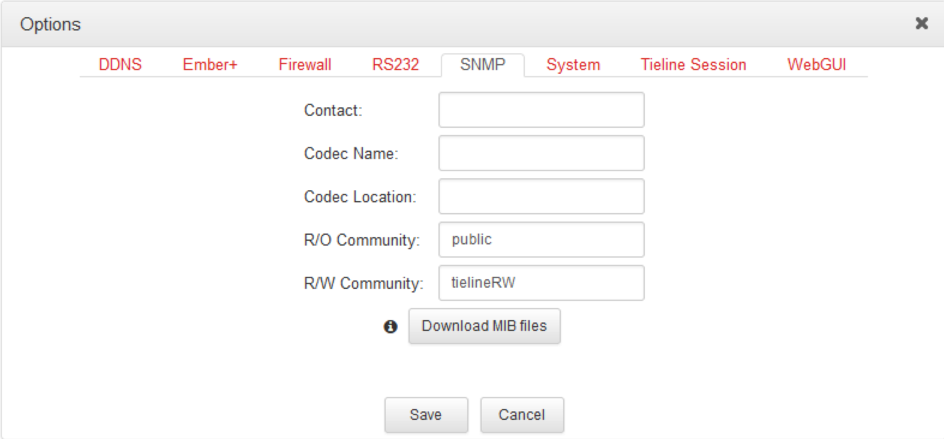
Description of SNMP Settings in the Codec

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.

Important Note: For more information on SNMP codec settings see Configuring SNMP in the Codec.

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:

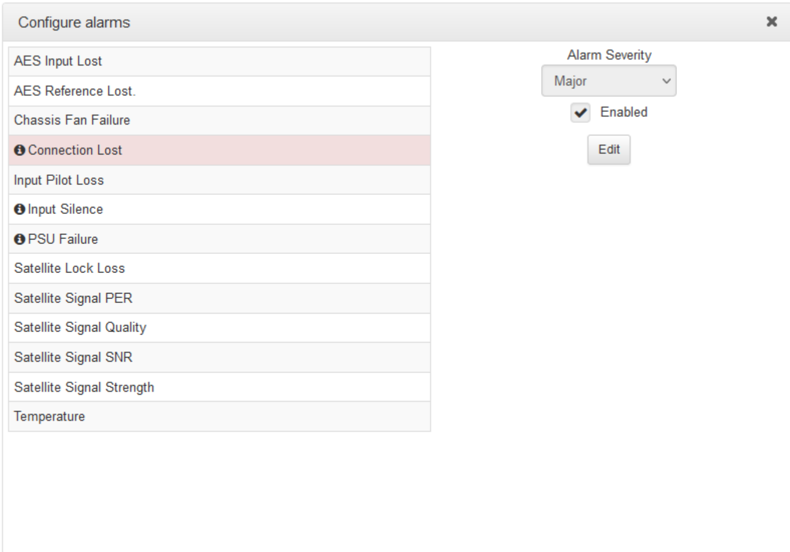
- The codec supports the attributes specified in the MIB-II standard. Please verify that your SNMP software contains the required files as specified in RFC 1213. An example of a free MIB browser is available at http://www.ireasoning.com/.
- Tieline recommends SNMP is disabled if a codec is connected 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.

19.23 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.

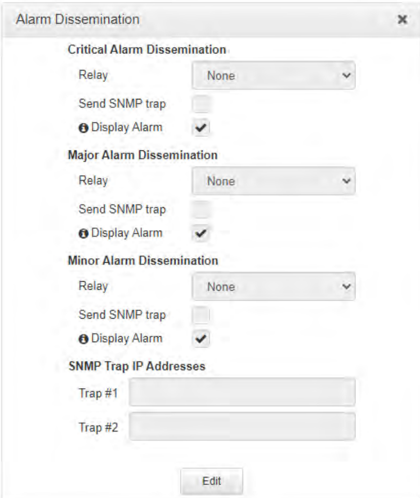
Note: The following **System** and **Audio** alarms are available:

Alarm	Alarm Type	Explanation
AES Input Lost	Audio	Raises an alarm if the AES input signal is lost
AES Reference Lost	Audio	Raises an alarm if the AES reference clock signal is lost
Chassis Fan Failure	System	Raises an alarm if a codec fan failure occurs
Connection Lost	Audio	Triggers an alarm whenever a streaming connection is lost
Input Pilot Loss	Audio	Raises an alarm if the stereo MPX pilot signal is lost.
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)
PSU Failure	System	Raises an alarm if one or both PSUs fail
Satellite Lock Loss	System	Raises an alarm if satellite lock is lost
Satellite Signal PER	System	Raises an alarm if the satellite signal Packet Error Rate (PER) is above the designated threshold
Satellite Signal Quality	System	Raises an alarm if the satellite signal quality is below the designated threshold
Satellite Signal SNR	System	Raises an alarm if the satellite signal SNR is below the designated dB threshold
Satellite Signal Strength	System	Raises an alarm if the satellite signal strength is below the designated threshold
Temperature	System	Raises an alarm if the temperature is too low or too high

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.



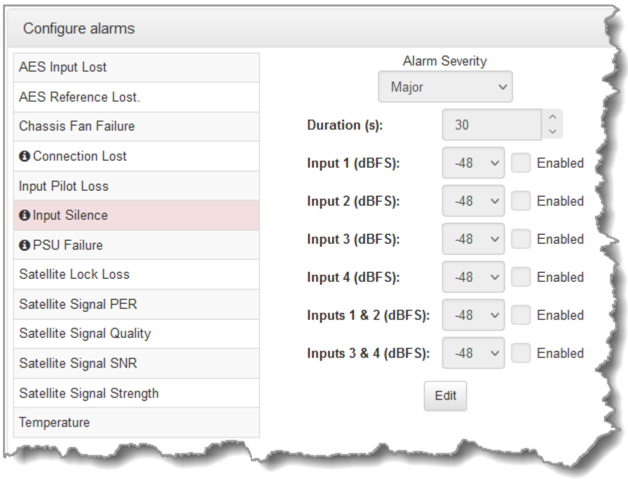
2. Click **Edit** to configure notification settings.
3. Select and configure relay, SNMP trap and alarm display settings for each **Alarm Severity** level. Enter **SNMP Trap IP Addresses** as required at the bottom of the panel.
4. Click **Save** to store the new settings.

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.

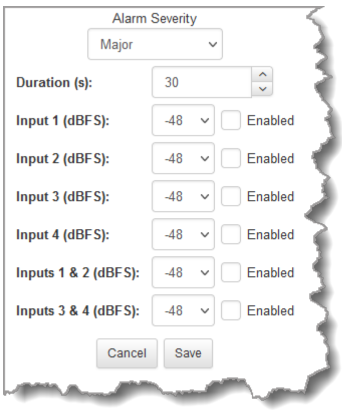
Configuring Input Silence Detection Parameters

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.



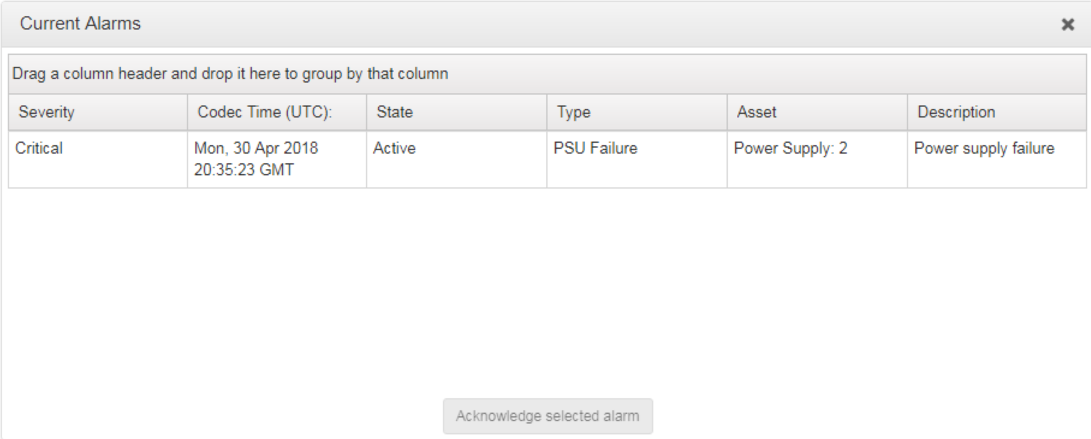
2. Click **Edit** to configure alarm settings.
3. Configure the dBFS threshold and timeout duration in seconds and ensure the input **Enabled** check-boxes are selected. An alarm will be raised when these thresholds are breached. Note: the number of inputs displayed will vary depending on the codec model, e.g. MPX I or MPX II.



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

19.23.1 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.

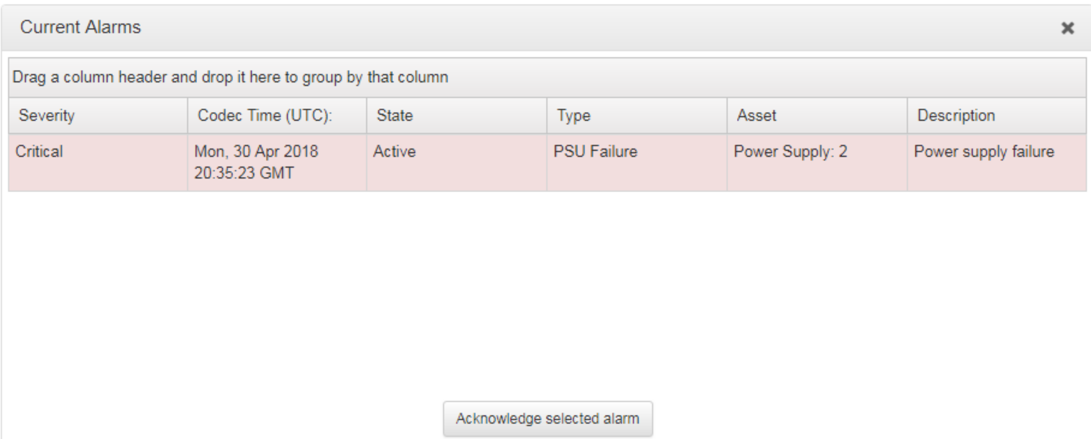
2. All new alarms being listed in the **Current Alarms panel**.
3. Other alerts as per **Alarm Dissemination panel** settings.
4. The codec front panel **ALARM LED** flashing red.

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

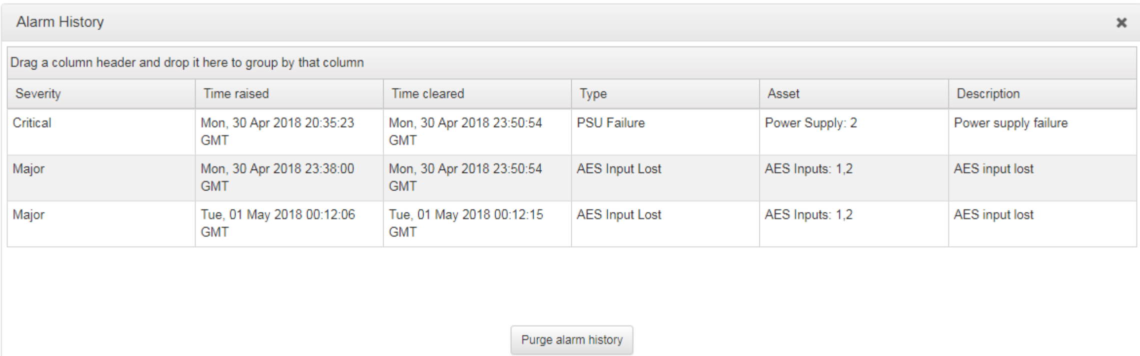
An alarm is deactivated automatically when the alarm state is reversed. E.g. if power is restored after a **PSU Failure** alarm, or if audio is restored after an **Input Silence** alarm.

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.



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

19.24 RS232 Data Configuration

Product	Feature	Notification
MPX I	RS-232	2 x RS-232 in MPX I (2 sync with audio); up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel; supports hardware and software flow control
MPX II	RS-232	4 x RS-232 in MPX II (4 sync with audio) up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel; supports hardware and software flow control

For non-MPX IP audio connections 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. 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 ports on the rear panel. Please see Appendix A for RS232 and Control Port Wiring information.

MPX Data Configuration

The following table outlines the data transport capabilities of the codec in MPX transmission modes.

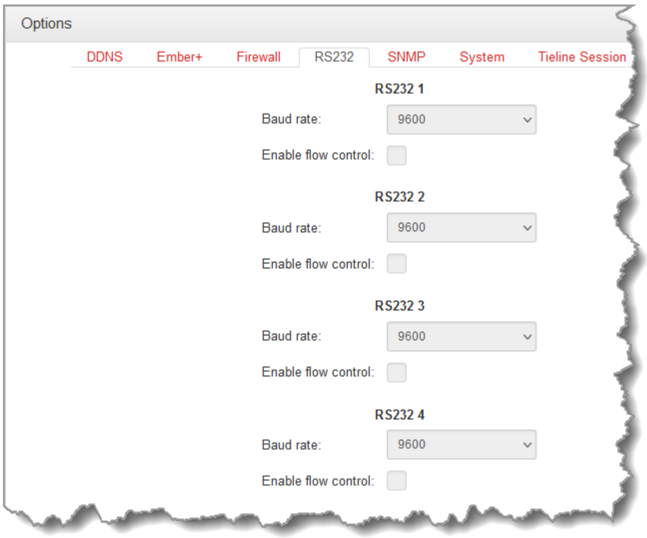
Program Protocol Type	TX Program: Encode Inband µMPX Data	RX Program: Decode Inband µMPX Data	TX Program: Encode Separate MPEG-TS Data Streams	RX Program: Decode Separate MPEG-TS Data Streams
µMPX over UDP	Yes	Yes	No	Yes: Decode up to 2 MPEG-TS data streams as separate connections
µMPX over RTP	Yes	Yes	No	Yes: Decode up to 2 MPEG-TS

				data streams as separate connections
Raw MPX over RTP	No: Data within RTP packets supported	No: Data within RTP packets supported	No	Yes: Decode up to 2 MPEG-TS data streams as separate connections
µMPX + MPEG-TS over RTP	Yes	Yes	Yes: Encode up to 2 MPEG-TS data streams in a single connection	Yes: Decode up to 2 MPEG-TS data streams as separate connections
µMPX + MPEG-TS over UDP	Yes	Yes	Yes: Encode up to 2 MPEG-TS data streams in a single connection	Yes: Decode up to 2 MPEG-TS data streams as separate connections
µMPX Forwarding	Proprietary Tieline MPX to Tieline MPX data supported	Proprietary Tieline MPX to Tieline MPX data supported	No	No
µMPX Forward + MPEG-TS over RTP	Yes	Yes: Decode up to 2 data streams as separate connections	Yes: Encode up to 2 MPEG-TS data streams in a single connection	No
µMPX Forward + MPEG-TS over UDP	Yes	Yes: Decode up to 2 data streams as separate connections	Yes: Encode up to 2 MPEG-TS data streams in a single connection	No

Important Notes:
- Control port relay data is always sent within the µMPX stream.
- Only RS-232 data is sent over MPEG-TS data stream connections.
- µMPX Forwarding in MPX codecs is not compatible with packet forwarding in Gateway codecs.
- Auxiliary data must be enabled to allow MPEG-TS data streams.

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.
- 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.
- In a non-MPX connection only the dialing codec needs to be configured to send RS232 data between 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 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 a non-MPX connection.

19.25 Creating Rules

Product	Feature	Notification
MPX I	Control Port GPIOs	Supports 4 control port inputs and 4 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connector
	Software LIOs	3 virtual inputs, 60 logical outputs, and 64 WheatNet Logic Outputs or 64 Livewire GPIOs.
MPX II	Control Port GPIOs	Supports 8 control port inputs and 8 opto-isolated outputs via a single DB25 **CONTROL PORT I/O** connector
	Software LIOs	3 virtual inputs, 56 logical outputs, and 64 WheatNet Logic Outputs or 64 Livewire GPIOs.

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 MPX I codec has 4 hardware GPIOs and 60 logical outputs, and the MPX II has 8 hardware GPIOs and 56 logical outputs; both codecs also have 3 virtual inputs, and 64 WheatNet Logic Outputs. WheatNet logic Outputs allow Tieline WheatNet-IP enabled codecs to activate functions across a WheatNet-IP network.

MPX I GPIOs

GPI	MPX I Input	GPO	MPX I Output
4 (1-4)	Physical hardware inputs	4 (1-4)	Physical hardware outputs
3 (5-7)	Virtual input		
		60 (5-64)	Logical outputs
		64	WNET Logic output
64	Livewire GPIs	64	Livewire GPOs

MPX II GPIOs

GPI	MPX II Input	GPO	MPX II 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 output
64	Livewire GPIs	64	Livewire GPOs

Note: WheatNet logic inputs are not supported in MPX codecs. WheatNet logic inputs and outputs cannot be used in conjunction with Livewire GPIOs. Virtual inputs (5 to 7 on the MPX I and 9 to 11 on the MPX II) can be activated by pressing the **F1** button and **KEYPAD** buttons **1-3**. There are three categories of rules:

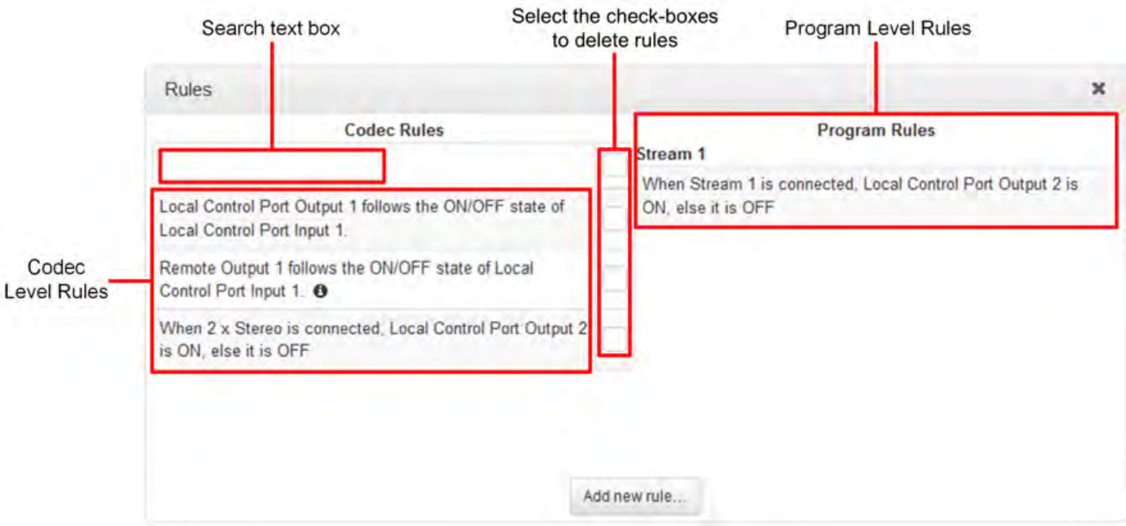
1. Codec level rules: Rules based on programs or codec hardware and software I/O states, e.g. Connect or disconnect a program when an input is toggled, or synchronize a local input to a remote relay.
2. Program level rules: Rules based on codec behaviors at the program level, e.g. Connect and disconnect 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:

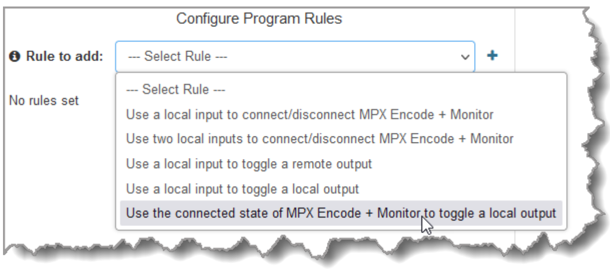
- A non-WheatNet-IP Tieline codec can be configured to trigger a WheatNet LIO in a Tieline WheatNet-IP codec.
- The **Enable Livewire GPIO** checkbox must be selected in the **Options panel** of the HTML5 Toolbox Web-GUI to use Livewire GPIOs. 64 Livewire GPIO ports are supported in the codec. These inputs and outputs are also mapped to 64 Tieline GPIs and GPOs.
- Relay reflection is not available for Multicast Client programs.
- Connection-related rules are not displayed in **Answer only** audio streams.
- Rules intended to activate dialing are not valid in **Answer only** programs or audio streams.

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

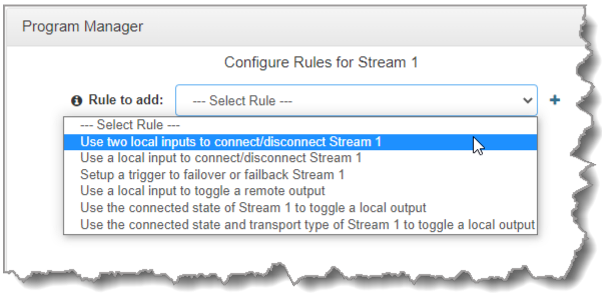
In the **Program Manager panel** wizard use the **Configure Program Rules** screen to configure program level rules.



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.

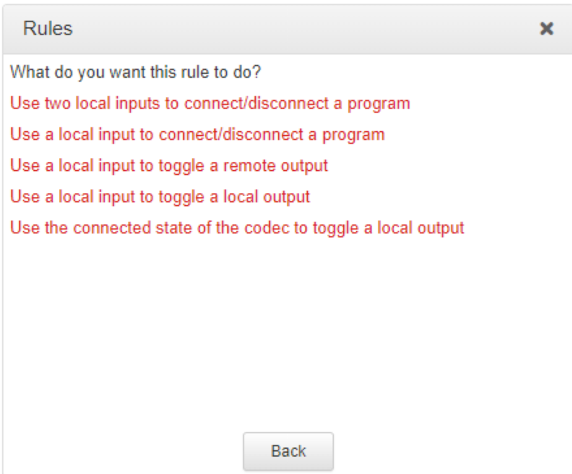


Note: A subset of filtered rules will be displayed for an **Answer only** audio stream connections.

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.

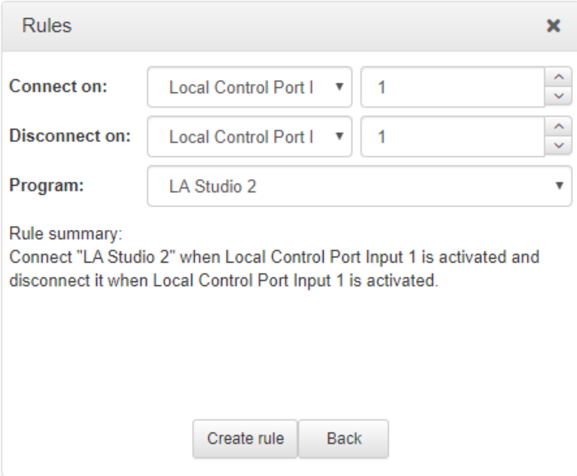


Note: When rules have been configured previously they are displayed when the **Rules panel** is opened.

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.

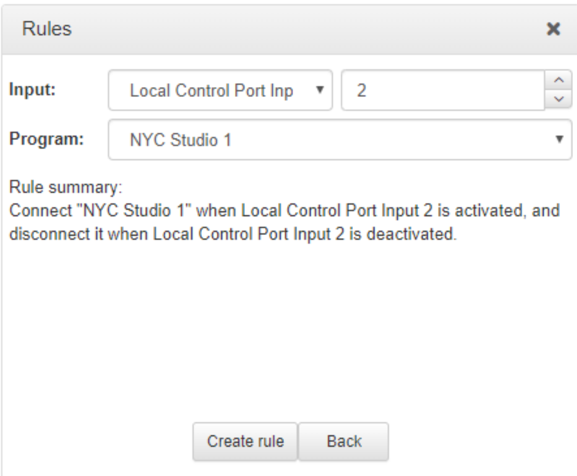


4. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

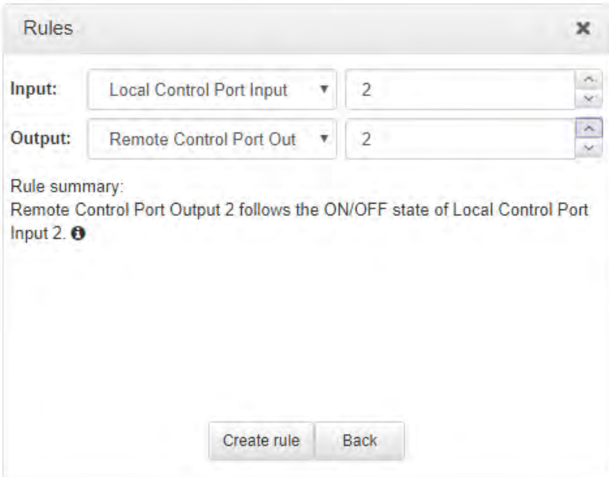


4. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

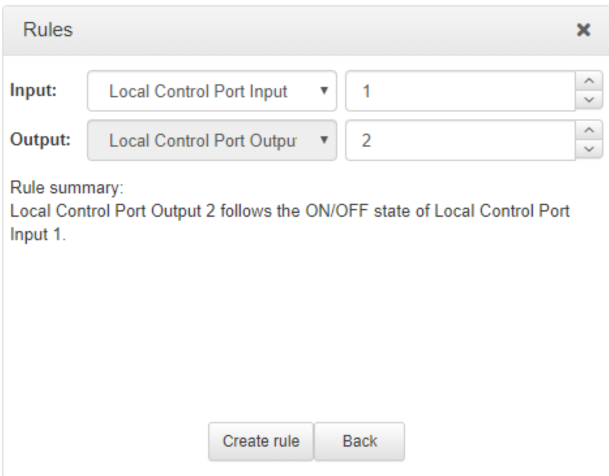


3. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

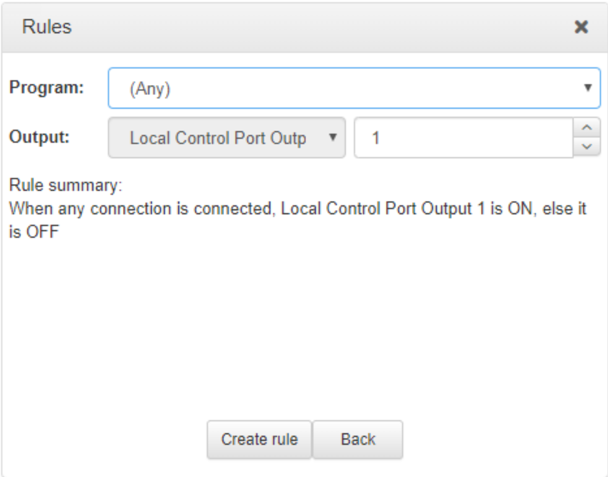


3. Check the **Rule summary** and click **Create Rule** to save the settings.

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.

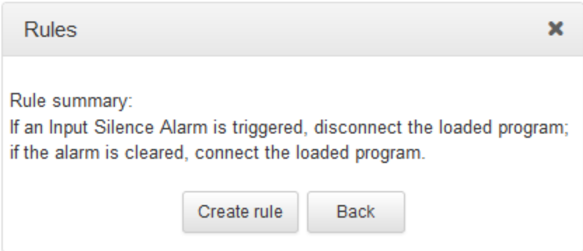


4. Check the **Rule summary** and click **Create Rule** to save the settings.

Rule 6: Use an Input Silence Alarm to disconnect/connect the loaded program

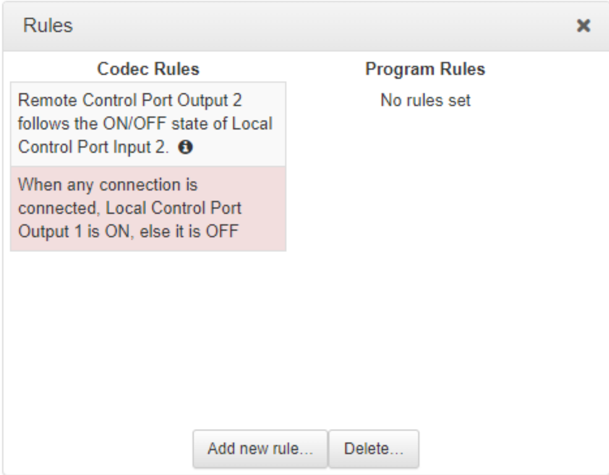
If an Input Silence Alarm is triggered, disconnect the loaded program; if the alarm is cleared, connect the loaded program.

1. Click the rule in the **Rules panel** titled **Use an Input Silence Alarm to disconnect/connect the loaded program**.
2. Click **Create rule**.



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.



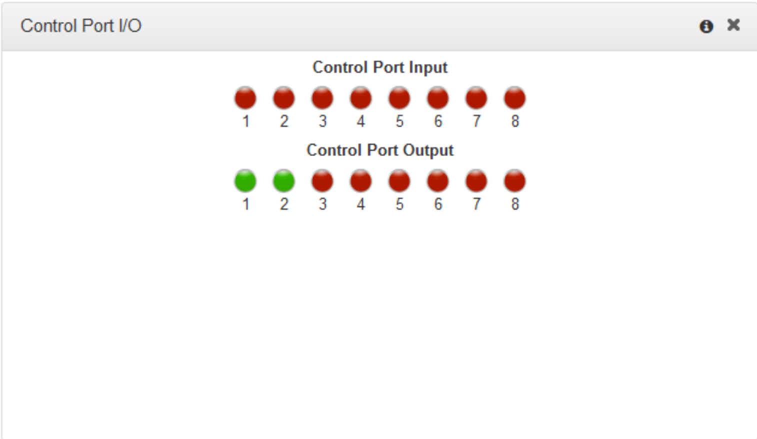
4. Click **Yes** in the confirmation dialog.

19.26 Monitoring Control Port I/O Status

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

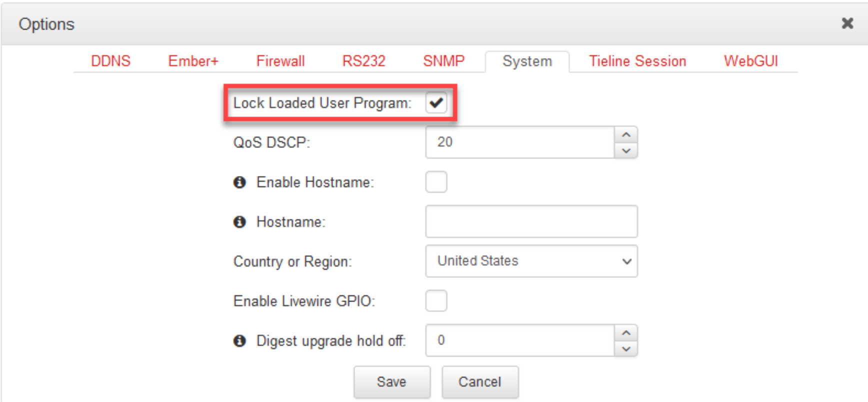
1. Open the Toolbox HTML5 Web-GUI and click **Control** in the **Menu Bar**, then click **Control Port I/O** to display this panel.
2. Click on an output to change the state from **Off** to **On**, then click **Save**. Note: Input states cannot be changed.



19.27 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, non-MPX programs configured to connect IP audio streams 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. Note: MPX signals are not mixed in this way.

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.

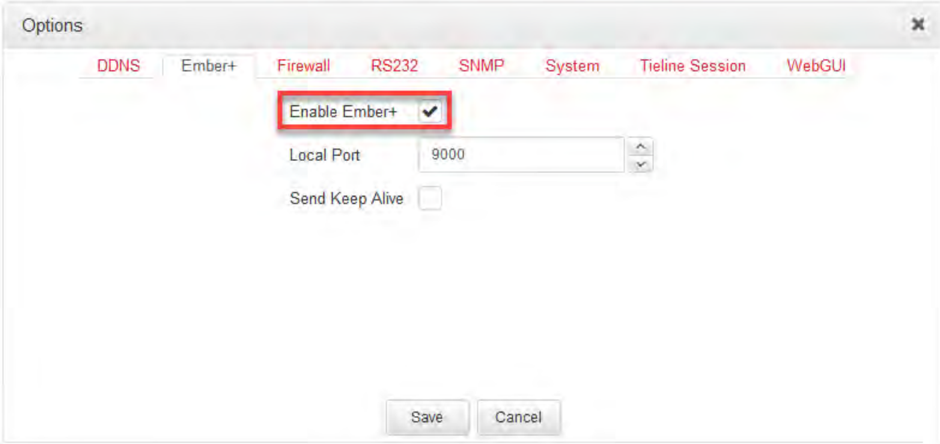
19.28 Enable Ember+

Ember+ is an open standard control protocol developed by Lawo which allows a third party application to gain access to device parameters. The codec supports:

- Physical GPIO: on/off.
- Inputs: Gain, on/off, name.
- Outputs: send/return, name.
- Connection Stream: name, connect/disconnect, caller ID.
- Connection Endpoint: name, connect/disconnect, address, remote audio port.

A single Ember+ provider on TCP port 9000 is configured by default and this can be changed.

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Options** to display the **Options panel**.
2. Select **Ember+**, then **Edit**.
3. Click to select the **Enable Ember+** check-box.
4. Adjust other settings as required, then click **Save**.

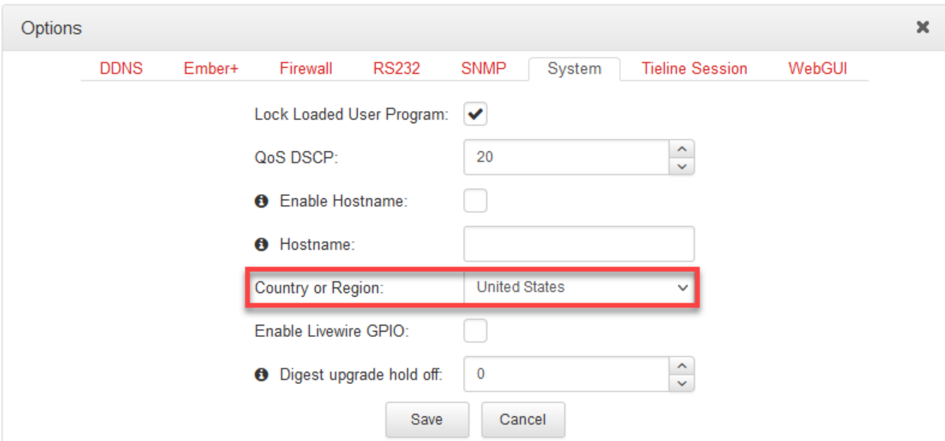


Important Note: Ember+ is currently only supported on **LAN1**, **LAN2**, **LAN3** and not the **AoIP 1** port.

19.29 Configure Country Setting

The **Country or Region** 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 connections. To configure the 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 or Region** drop-down menu arrow to select the country of operation.

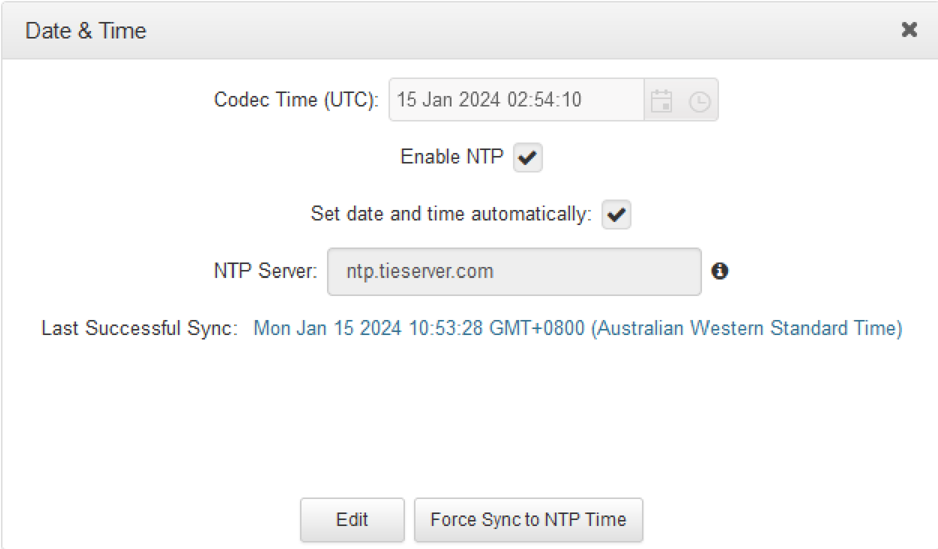


4. Click the **Save** button to store the new configuration.

19.30 Adjusting Codec Time and Date

Network Time Protocol (NTP) is a networking protocol for clock synchronization between computer systems over packet-switched, variable-latency data networks. By default **Use NTP** time is enabled in the codec and it will synchronize with **ntp.tieserver.com**. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected. To manually synchronize time settings in the codec:

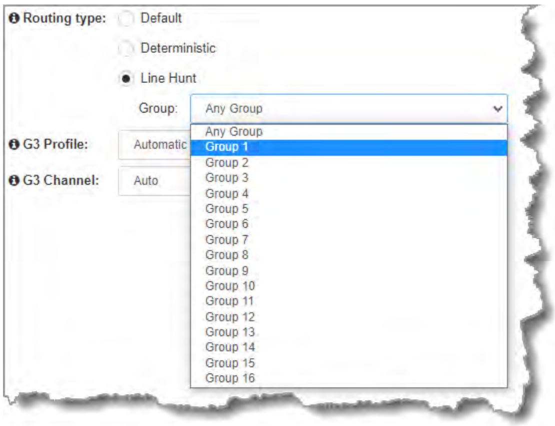
1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Date & Time** to display the **Date & Time panel**.
2. Click **Force Sync to NTP Time** to manually synchronize the codec to NTP time.



Important Notes:
- It may take more than one attempt to **Force Sync to NTP Time**.
- When NTP address settings are configured and enabled, the codec will immediately jump to the new time when synchronized to the server. This may cause scheduled events to be missed.
- Ensure DNS settings are configured correctly as this is required for NTP.

19.31 Line Hunt Call Answering

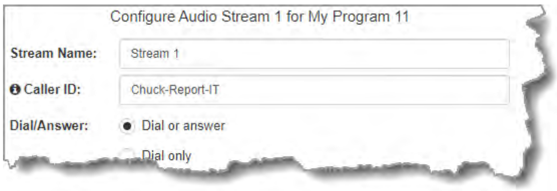
The codec supports line hunt call answering, whereby line hunt groups can be assigned for multiple incoming callers on a first come, first served basis. This is ideal for separating groups of inputs and outputs between different studios or stations. As an example, when dialing a codec using a multi-stream program with line hunt groups configured, select **Line Hunt** as the routing type. Then select the group to which the audio stream should be routed by the answering codec, e.g. **Group 1** in the following example.



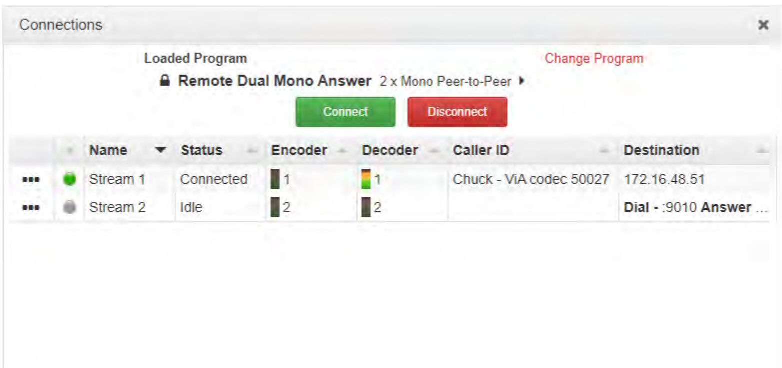
When **Any Group** is selected for all streams in **Line Hunt** mode, the codec will route all incoming calls on a first come, first served basis. The dialer doesn't have to specify a line hunt group, however they do need to select **Line Hunt** mode in the dialing program for this to function correctly.

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 or G6 codec can display a designated **Caller ID** in the **Connections panel** when dialing. In the following example, a ViA codec has called into the codec using the caller ID **Chuck - ViA 50027**.



19.32 Upgrading Codec Firmware

To download the latest codec firmware visit www.tieline.com. See Upgrading Firmware via SD Card to upgrade codec firmware using an SD card 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.

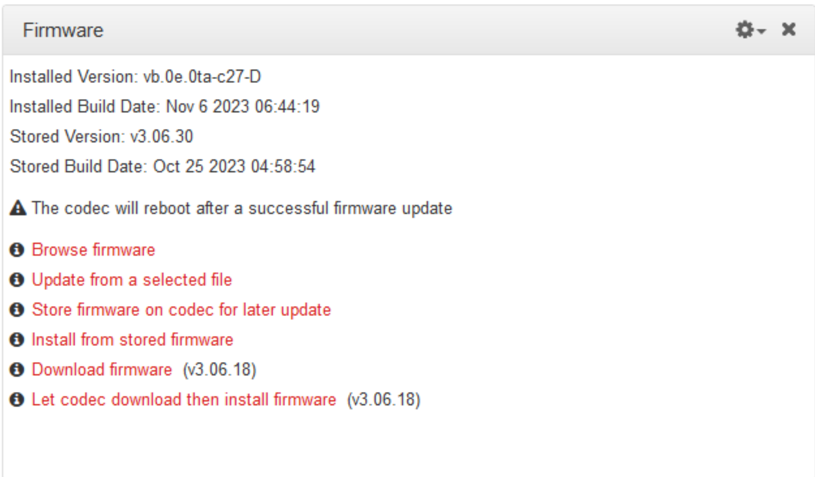
1. Connect the codec to a PC using a LAN connection and open the HTML5 Toolbox Web-GUI.
2. If new software is available the **Upgrade** symbol appears in the top-left of the screen.

3. Click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel** to perform the firmware upgrade.

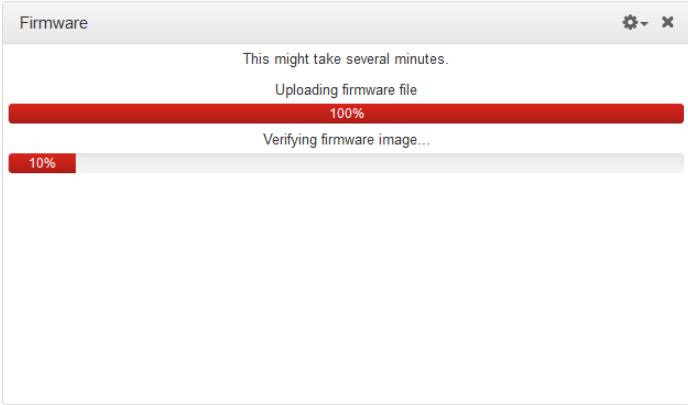
Performing a Firmware Upgrade

Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **Firmware** to display the **Firmware panel**. There are several firmware upgrade options available in the **Firmware panel**:

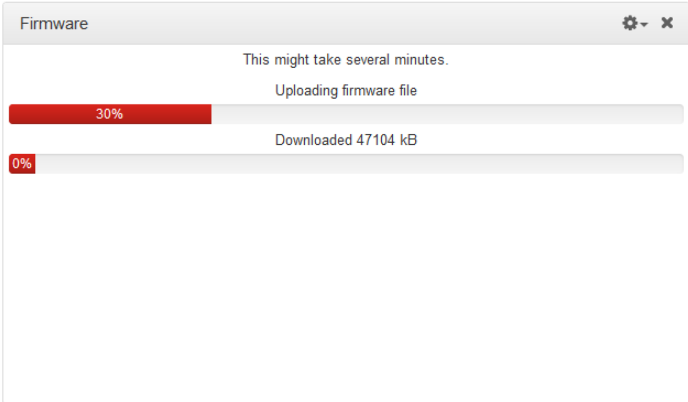
1. **Browse Firmware**: Click to navigate to the Tieline website and download the latest firmware for the codec to your computer. Once the firmware has been saved locally, click **Update from a selected file** in the **Firmware panel**.



2. **Update from a Selected file**: Click to navigate to a firmware file saved on a computer or network drive. 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.

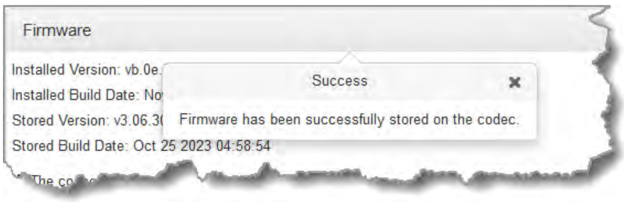


3. **Store firmware on codec for later update:** Click to navigate to a firmware file and download and store it on codec flash memory for installation when required.

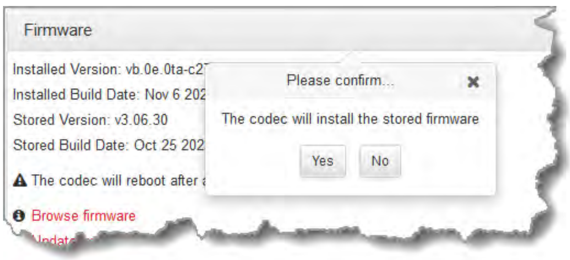


Firmware downloading onto a codec

A confirmation dialog is displayed when firmware has been downloaded successfully.



4. **Install from stored firmware**: Click and select **Yes** to install a stored firmware file on the codec.



5. **Download firmware**: Click to download a previous reliable firmware version. Note: Only visible when a new release is available.
6. **Let codec download then 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.

7. 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.

Refresh Browser Cache After Firmware Upgrade

Tieline recommends 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

19.33 Backup and Restore Functions

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

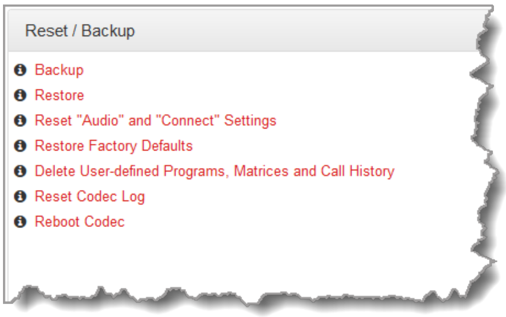
- Programs containing a variety of connection settings settings.
- All system settings that have been adjusted to change the factory default codec settings (current runtime settings).

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

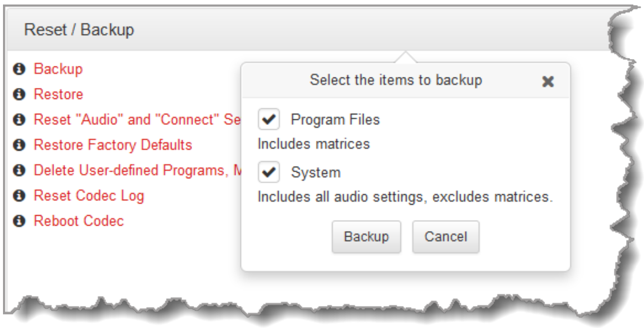
- Program, audio stream and connection names.
- IP address, port, algorithm, jitter buffer, FEC and bit rate settings (etc.) for audio stream connections.

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**.



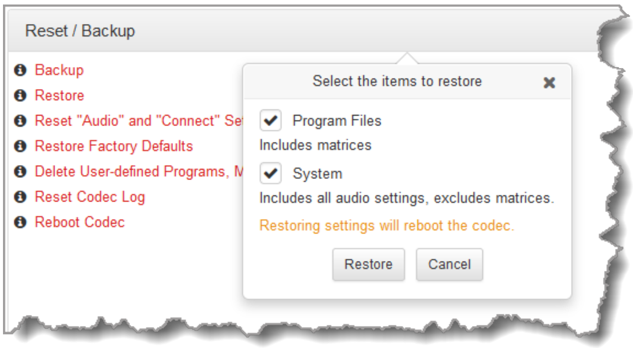
2. Click **Backup**.
3. Click to select the check-boxes to confirm your backup requirements, then click **Backup**.



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.

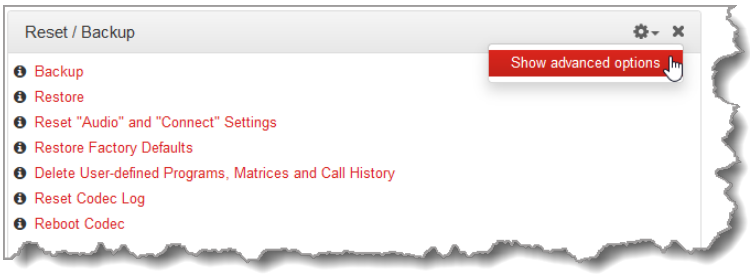
Note: The codec will automatically reboot when restoring system settings.

Advanced Settings: XML Config

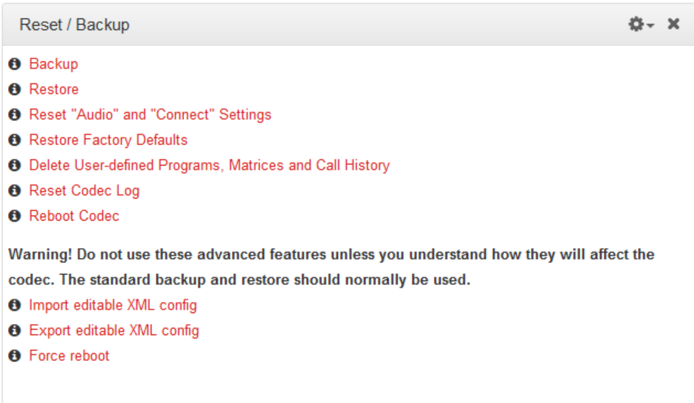
Caution: DO NOT use advanced XML configuration features unless you fully understand how they will affect the codec. The standard backup and restore function should normally be used. Damage to the codec may occur if this feature is used without fully understanding how it will affect the codec.

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.



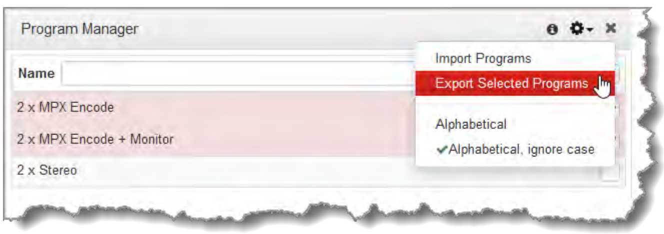
19.34 Import and Export Programs

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

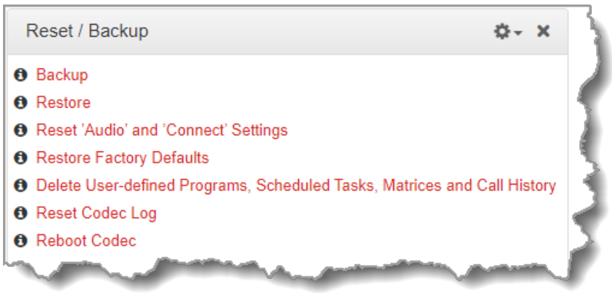
1. Open the HTML5 Toolbox Web-GUI and click **Connect** in the **Menu Bar**, then select **Program Manager** to launch the **Program Manager panel**.
2. Click the **Options symbol** in the top right-hand corner of the **Program Manager** and select **Import Programs**.

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.

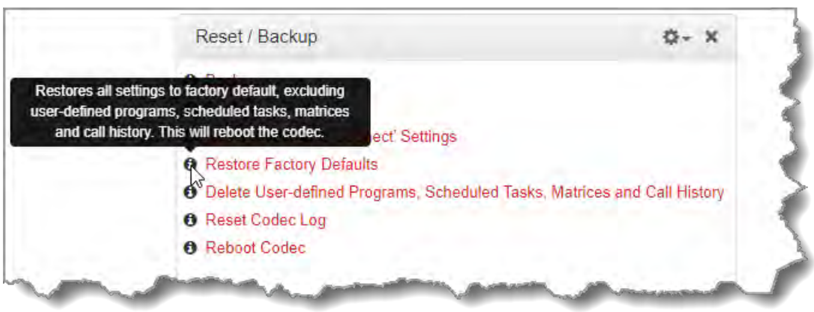
19.35 Reset Factory Default Settings

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.

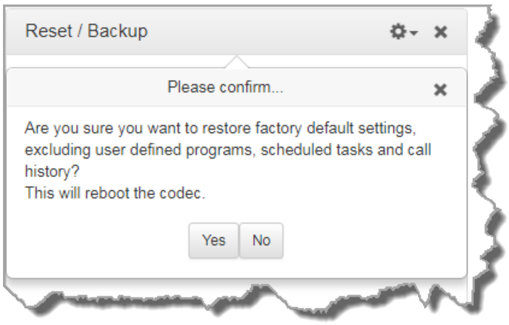
1. Open the HTML5 Toolbox Web-GUI and click **Settings** at the top of the screen, then click **Reset / Backup** to display the **Reset / Backup panel**.



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.



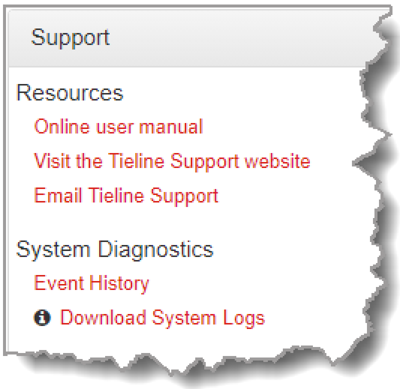
Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

19.36 Download Logs

The codec is capable of providing diagnostic information via user logs, which can either be sent to Tieline support, or downloaded for user diagnostics.

Procedure for Sending Logs to Tieline

1. Open the HTML5 Toolbox Web-GUI and click **Help** in the **Menu Bar**, then click **Support**.
2. Click **Download System Logs**.



3. Save the file to your computer and then send 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.

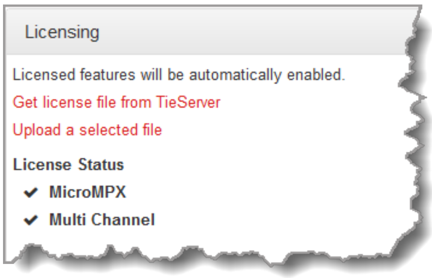
Clearing Logs

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.

19.37 Software License Installation

Product	Feature	Notification
MPX I	Licensing	The MPX I codec supports 2 audio channels only and this cannot be upgraded or expanded. MicroMPX is an optional license available to purchase.
MPX II	Licensing	The MPX II codec supports 4 audio channels. MicroMPX is an optional license available to purchase.

To view a codec's **License Status**, or to update licenses, open the HTML5 Toolbox web-GUI in a web-browser and select **Settings** and open the **Licensing panel**.

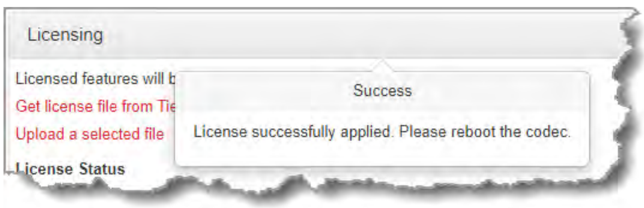


A MicroMPX license needs to be purchased and installed to support MicroMPX. To update licenses using the front panel of the codec see Updating Licenses.

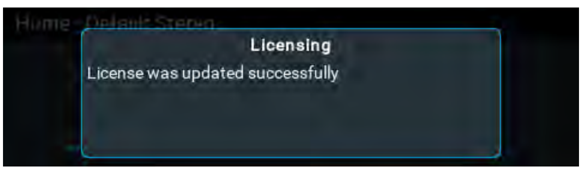
Perform an Automatic Software License Install with the HTML5 Toolbox Web-GUI

If a new license has been purchased, e.g. for MicroMPX encoding, check for a notification email from Tieline indicating that the new license file is ready to download from TieServer. To perform an automatic software license update using TieServer it is necessary to connect the codec to a PC and the internet.

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.

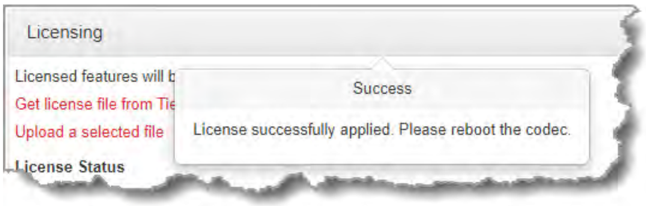


7. Reboot the codec via **Settings > Reset & Backup > Reboot Codec** and press the  button. Note: do not reboot by removing the power cable from the codec.

Download a License File and Install Manually

If the automatic license update fails, or the codec is not connected to the internet, you can install a previously downloaded license file, or a license file copied to a PC. Save the license on 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.



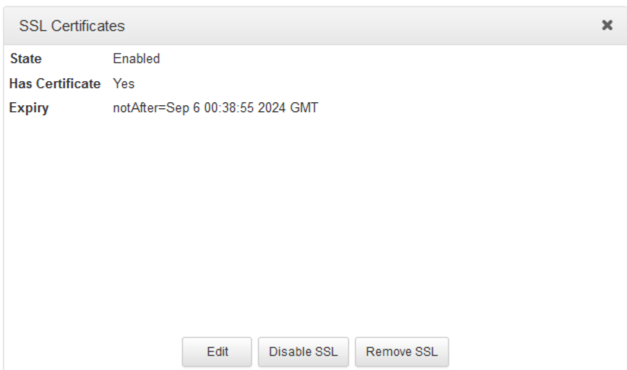
6. Reboot the codec via **Settings > Reset & Backup > Reboot Codec** and press the  button. Note: do not reboot by removing the power cable from the codec.
7. Clear the web-browser cache.
8. Verify that expected licenses have been installed in the **Licenses panel**.

19.38 Updating an SSL Certificate

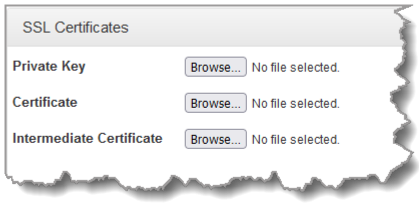
A security certificate can be updated using the **SSL Certificates panel** in the Toolbox Web-GUI. This is only possible after it has been installed locally on a codec using an SD card. The codec Web-GUI cannot be used to initially install an SSL certificate on a codec, it can only update it. To install a certificate using the codec front panel see Installing a Security Certificate for more details.

Updating SSL Certificates

1. Open the HTML5 Toolbox Web-GUI and click **Settings** in the **Menu Bar**, then click **SSL Certificates** to display the **SSL Certificates panel**.



2. Select **Edit**.

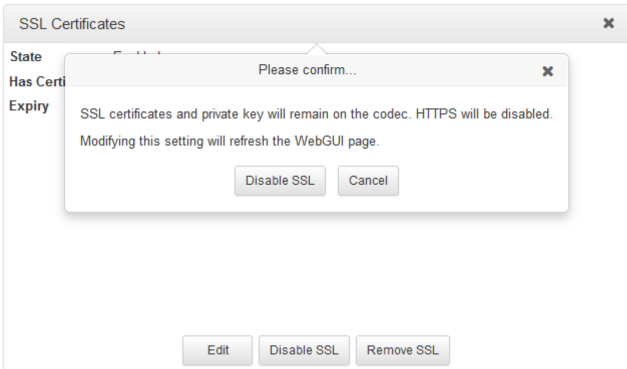


3. Click **Browse** to select relevant files to be installed from your connected device.

4. Click **Save** to update the selected SSL files in the codec.

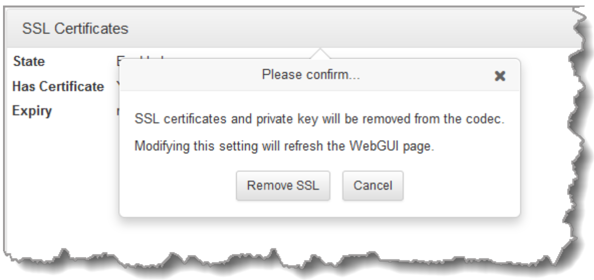
Disable SSL Certificates

1. Click the **Disable SSL** button to disable HTTPS without removing a private key or SSL certificates from the codec.
2. Click **Disable SSL** in the confirmation dialog to confirm the change.



Remove SSL Certificates

1. Click the **Remove SSL** button to remove a private key or SSL certificates from the codec.
2. Click **Remove SSL** in the confirmation dialog to confirm the change.

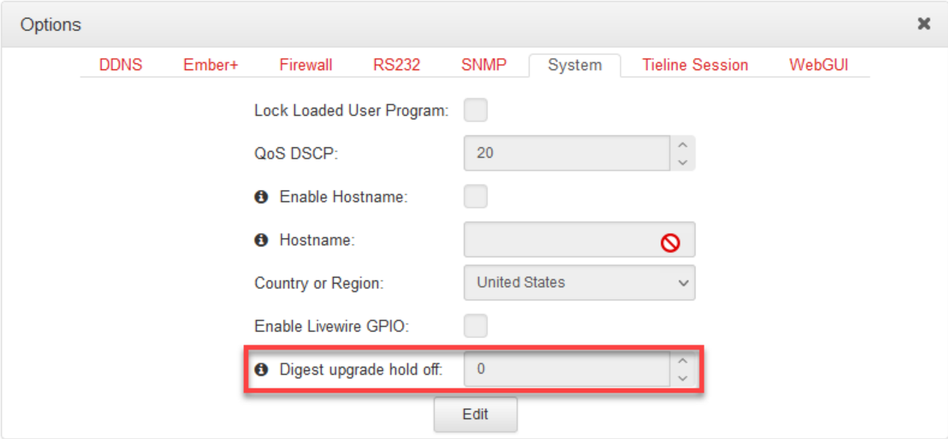


Troubleshooting Certificate Installation

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.

19.39 Digest Upgrade Hold Off

The **Digest Upgrade Hold Off** setting is accessed via the **System tab** in the **Options panel** in the Toolbox web-GUI. It is also accessible from the codec front panel via **Settings > Miscellaneous**.



The **Digest Upgrade Hold Off** value influences the **Connection panel** and Cloud Codec Controller (CCC) connection LEDs by maintaining the lowest LQ value that a connection, stream and program has been in for the last X seconds. The default setting is 0 and in nearly every situation this is the recommended configuration. DO NOT change this setting if you are unsure of its effect on the display in the Connection panel or CCC. LED display settings are as follows:

- Gray: Not In use
- Green: Connected and stable (LQ greater than 70)
- Yellow: Connected but connection quality is not stable (LQ 50 to 70)
- Red: Connection establishing or problem with connection (LQ less than 50). LED flashes when establishing connection.

Note: ISDN connections do not display the yellow LED state.

To adjust the setting in the **System tab** in the **Options panel**:

1. Click **Edit**.
2. Adjust the **Digest upgrade hold off** period in seconds.
3. Click **Save**.

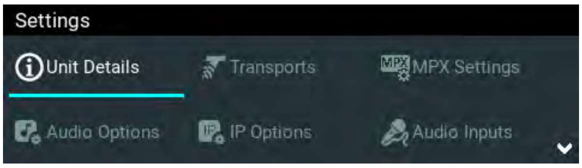
20 Front Panel Configuration Tasks

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

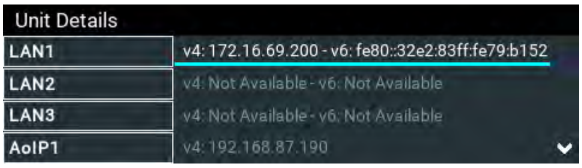
20.1 Configuring IP via the Front Panel

Checking IP Address and Unit Details in the Codec

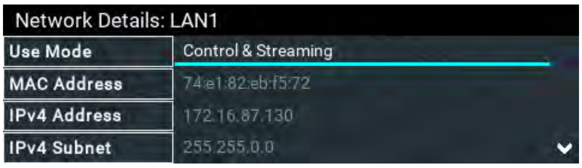
1. Press the **SETTINGS** button.
2. Select **Unit Details** and press the button.



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**, **LAN2, LAN3** or **AoIP1/LAN4** to display **Network Details** including the **MAC Address**, which is used to add the codec to a domain along with the serial number.

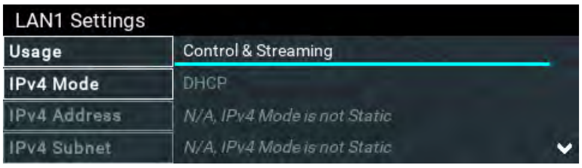


Important Note: For assistance with configuration of IPv4 or IPv6 network connections contact your IT Administrator.

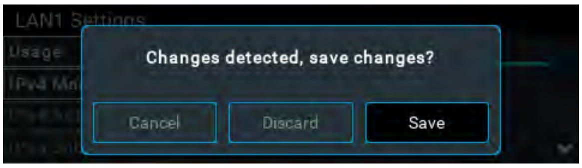
Ethernet and VLAN Configuration Options

LAN1 and **LAN2** are physical Ethernet port interfaces and **LAN3** is an Ethernet port for control only. **AoIP1/LAN4** is configurable as an AoIP port for AES67-ST2110-10 streams. Up to four additional VLAN interfaces can be configured.

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 settings can be viewed and configured via **Settings > Transports > LAN1 / LAN2 / LAN3 / AoIP1/LAN4**.



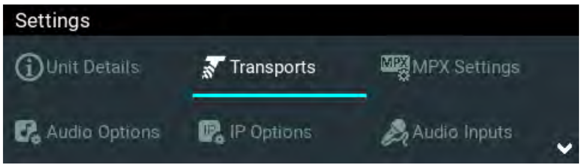
After completing configuration press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings..



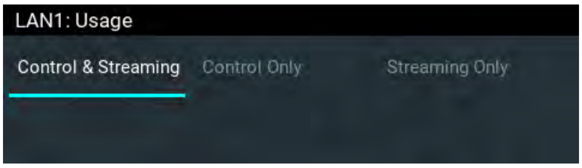
Configure an IPv4 DHCP Address

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).

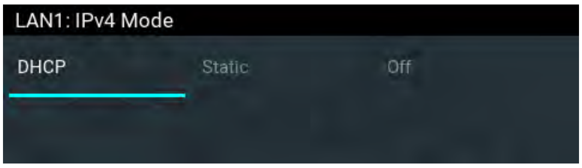
1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.



3. Use the down navigation button to select **LAN1**, **LAN2, LAN3** 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 **DHCP** and press the button.

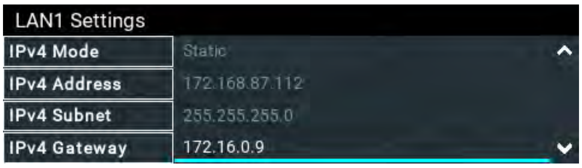


7. Press the **Return** button, then select **Save** in the confirmation dialog and press the button to confirm the new settings.

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. Note: DNS settings must be specified when a static IP address is configured.

1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2**, **LAN3** 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.

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**, **LAN3** 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.

Specifying DNS Server 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** section within the Web-GUI. This feature can be turned on or off in the LAN codec menu. When installing codecs at the studio it is important to ensure that DNS settings are configured correctly. This is particularly important when using tools like the Cloud Codec Controller. Correct DNS settings are also required for NTP, DDNS, use of host names, and if the **Download and Install** option is used to upgrade codec firmware.

1. Press the **SETTINGS** button.
2. Select **Transports** and press the button.
3. Use the down navigation button to select **LAN1**, **LAN2**, **LAN3** 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**, **LAN3** 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)

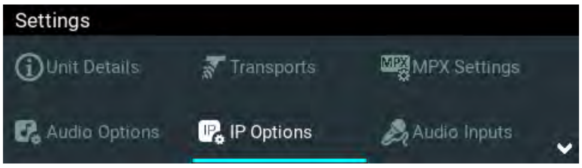
This setting applies the VLAN settings to a physical Ethernet port in the codec.

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**, **LAN2** or **LAN3**, 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.

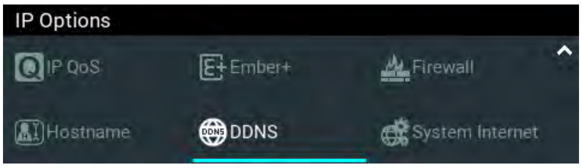
20.2 Configuring DDNS

Dynamic DNS (DDNS) can be configured in the codec to allow the use of a host name to connect when a dynamic IP address is used. To facilitate this a DDNS service is used to register a device's IP address to a host name, allowing a remote codec to dial to a host name. The host name will be updated automatically periodically as required, or when the dynamic IP address changes.

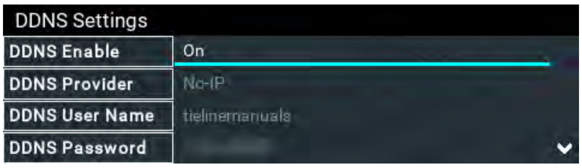
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.



3. Navigate to **DDNS** and press the button.



4. Select and configure each DDNS setting and enable DDNS. The codec is then able to accept a call using the specified DDNS Host Name.

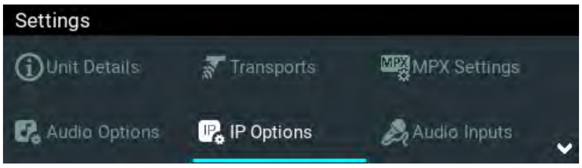


Important Note:
- Supported DDNS providers are listed in the **DDNS Provider** drop-down menu.
- Devices should be DDNS registered to public IP addresses.
- The codec will utilize the **System Internet** interface order when contacting DDNS service providers.
- Does not support remote G3 codecs connecting in this mode.
- DDNS Host Name settings are unrelated to the **Hostname** setting accessed via the **Options panel** under **Settings**.
- Codec DNS settings must be specified when configuring DDNS.

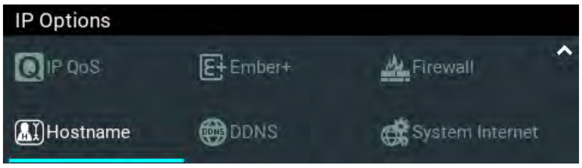
20.3 Configuring a Hostname

It is possible to assign a hostname to the codec to provide a flexible way of identifying the codec on a network.

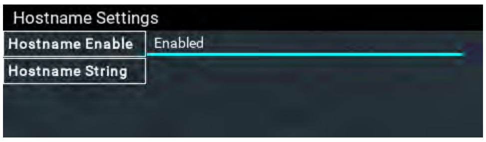
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.

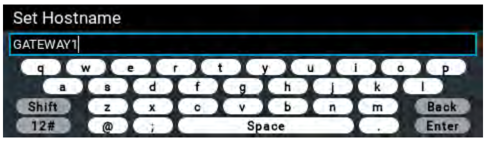


3. Navigate to **Hostname** and press the button to enable this feature.

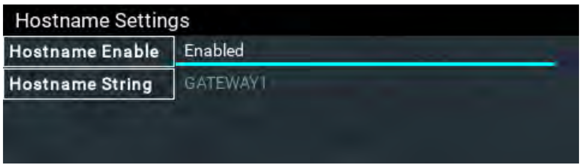


4. Select **Hostname Enable** and press the button to enable this feature. Next, navigate to **Hostname String** and press the button to display the **Set Hostname** screen keyboard and then enter the hostname. Next, navigate to **Enter** and press the button to save the settings. Note: Press the **Return** button to exit the screen without saving changes.





5. The new hostname will be displayed.



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.

20.4 Configure Default Interfaces

When dialing over IP you can select the preferred interface to use when establishing a connection. By default **Any** is selected, which means the first available interface will be used to dial a connection. The default interfaces in order of use when available are:

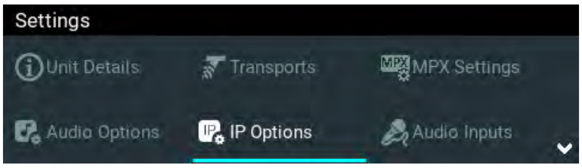
1. **LAN1** Ethernet port (default **Primary** interface)
2. **LAN2** Ethernet port (default **Secondary** interface)
3. **Wi-Fi** (default **Tertiary** interface)
4. **VLAN1** (default **Quaternary** interface)

Important Notes:
- If an interface is not available it is not listed in the interface selection screen. E.g. Wi-Fi is not enabled or a USB modem is not attached.
- If none of the 4 primary, secondary, tertiary and quaternary interfaces listed are available, the codec will look for the first available interface in the following order: LAN1; LAN2; VLANs.
- 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.

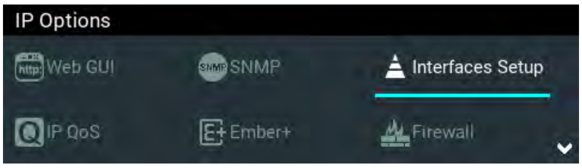
Reconfigure Default Interfaces

It is possible to reconfigure the default **Primary** (Ethernet 1), **Secondary** (Ethernet 2), **Tertiary** (Wi-Fi) and **Quaternary** (VLAN1) IP 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.

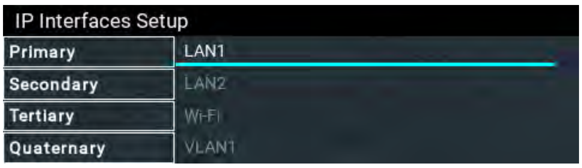
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.



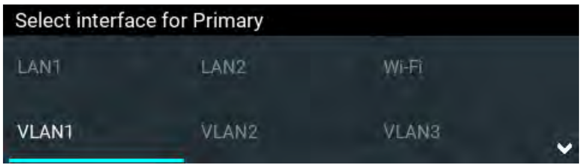
3. Navigate to **Interfaces Setup** and press the button.



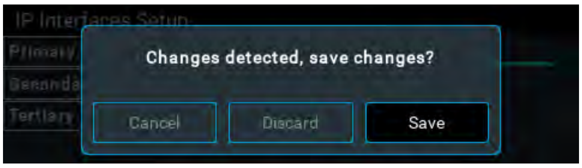
4. Navigate to the **Primary**, **Secondary**, **Tertiary** or **Quaternary** IP interface setting and press the button.



5. Select an alternative default interface and press the button.



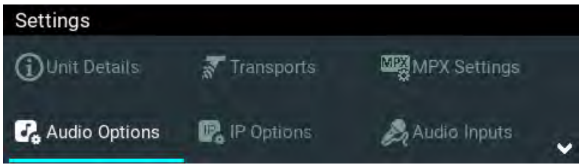
6. Press the **Return** button to navigate out of the menu, then navigate to **Save** in the confirmation dialog and press the button to save all changes.



20.5 Configure Audio Options

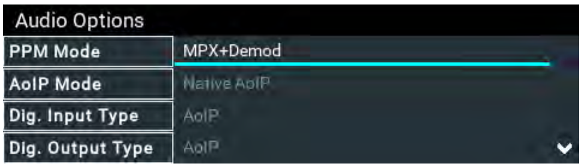
A range of input, output, PPM, synchronization, metering and reference options are configurable within the **Audio Options** menu.

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



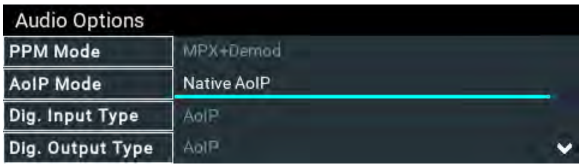
PPM Mode

PPM metering can be adjusted via **Settings > Audio Options > PPM Mode**. The MPX I codec has 2 inputs, 2 outputs, 2 encoders, 2 decoders, and 4 PPMs.The MPX II codec has 4 inputs, 4 outputs, 4 encoders, 4 decoders, and 8 PPMs. Both codecs support **Input/Output**, **Encoder/Decoder** and **MPX+Demod** modes. See Input Configuration, Levels and PPMs for more details.



AoIP Mode

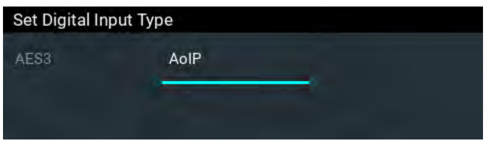
The codec can stream using native AoIP protocols (e.g. AES67, ST 2110-30, Livewire+, RAVENNA). This setting is not configurable.



Digital Input Type

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

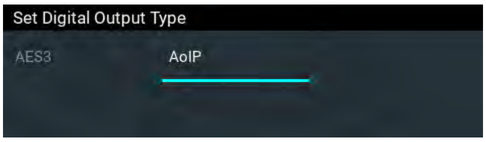
1. Press the **SETTINGS** button.
2. Navigate to **Audio Options** and press the button.
3. Select **Dig. Input Type**.
4. Adjust the setting and press the button.



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**.
4. Adjust the setting and press the button.



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.

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 attempts to use inputs 3 and 4 in the MPX II codec only. 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 other inputs in the MPX II codec 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

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, MPX I, MPX II, Bridge-IT II, Bridge-IT XTRA II)	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**.

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

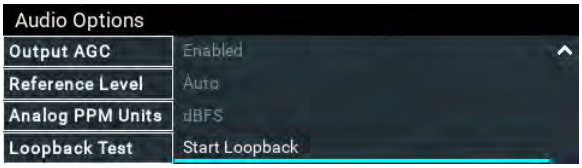
It is possible to switch the analog input PPM meter unit of measurement from dBFS (default) to dBU:

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

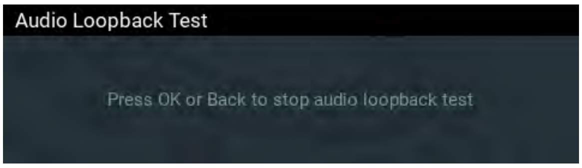
Loopback Test

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 button.
3. Select **Start Loopback** and press the button for **Loopback Test**.



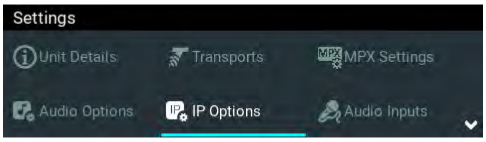
4. Press or the **Return** button to end the loopback test.



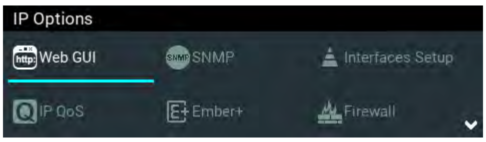
20.6 Enabling the Cloud Codec Controller

To allow the codec to be configured and managed by Tieline's Cloud Codec Controller over the public internet it needs to be enabled for CCC management:

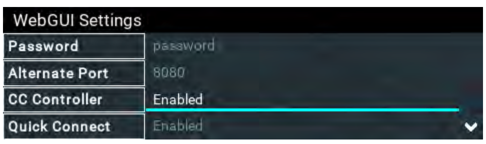
1. Press the **SETTINGS** button.
2. Navigate to **IP Options** and press the button.



3. Select **Web-GUI** and press the button.



4. Navigate down to **CC Controller** and press the 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.

DNS and the Cloud Codec Controller (CCC)

If a codec has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. When using the Toolbox HTML5 Web-GUI to configure DNS network settings:

1. Select **Transport** and then **Network** to open the **Network panel**.
2. Select an interface and then select the **DNS** tab to enter details into the **DNS Addresses** fields as required.
3. Ensure that the **Specify DNS Settings** check-box is selected. Note: This will ensure the DNS servers are used.

Note: These settings can also be configured using the front panel of the codec.

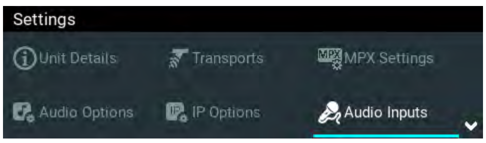
Where can I find my DNS settings?

DNS settings for static public IP addresses can be sourced from your internet provider/Telco. Sometimes the Gateway setting in the TCP/IP tab in the Network panel will work as the DNS Address to enter. Another common scenario encountered is that static DNS settings are configured to override DHCP settings when they are not compatible with the network being used. When using DHCP IP settings ensure the **Specify DNS Settings** check-box is disabled in the **Network panel**.

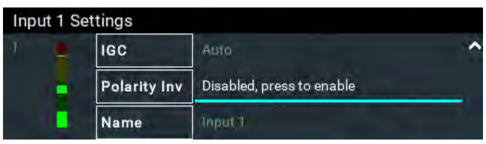
20.7 Inverting Input Polarity

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**.



20.8 Selecting an Algorithm

For regular non-MPX IP connections 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. For information on MPX encoding see Configuring MPX Inputs and Outputs. Note: Only Opus encoding is supported for sessionless or Tieline session MPX Monitor connections.

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.

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)

Stream Encoding Limits

All G5/G6 products:

- High frame-rate algorithms are limited to one SmartStream PLUS redundant stream (e.g. PCM, aptX Enhanced, G.711 and G.722).
- Music PLUS encoding limited to two SmartStream PLUS redundant streams
- All other algorithms support three SmartStream PLUS redundant streams

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.
- The uplink bandwidth capability of your broadband connection.

Important Notes: 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 moderate to high bit rates	• Highly compatible with other brands of audio codec • Very high quality audio for remotes, STLs and audio distribution
MPEG Layer 3	Up to 15kHz	100ms	64kbps	8.5 - 13.3kbps	• High quality voice and music • Moderate bit rates • High delay	• High quality remotes, STLs and audio distribution • Use when bidirectional communication between announcers is not required

LC-AAC	Up to 15kHz	64ms	64kbps	15kbps	• High quality voice and music at lowest bit rate; better quality at higher bit rates • Moderate delay at moderate to high bit rates	• Voice or music remotes as well as STLs and audio distribution where some delay is tolerable • Tieline Music or MusicPLUS deliver lower delay
HE-AAC v.1	Up to 15kHz	128ms	48kbps	7.4kbps	• High quality voice and music at the lowest bit rate; better quality at higher bit rates • Low to Moderate bit rates • High delay	• Live voice or music remotes as well as STLs and audio distribution with limited connection bandwidth • Use when bidirectional communication between announcers is not required
HE-AAC v.2	Up to 15kHz	128ms	Minimum 16kbps (Mono); 24kbps (stereo)	7.4kbps	• High quality voice and music • Low bit rates • High delay	• Used for DAB+ radio streaming • Ideal for low bit rate remotes • Use when bidirectional communication between announcers is not required
AAC-LD	Up to 20kHz	20ms at 48kHz	48kbps minimum	30kbps	• Very high quality voice and music • Very low delay at low to moderate bit rates	• Very high quality, very low delay STLs and audio distribution • Remote connections able to achieve 48kbps for each audio channel requiring • Suitable when bidirectional communication 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
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

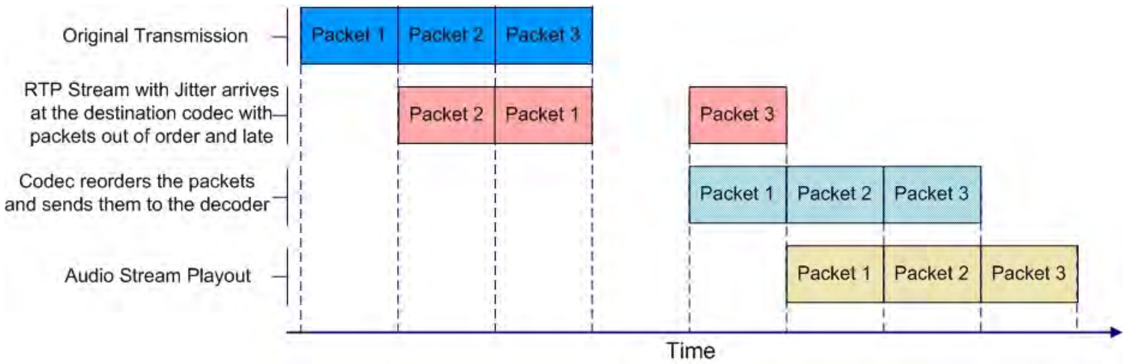
Algorithm Selection Guide

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.

20.9 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).

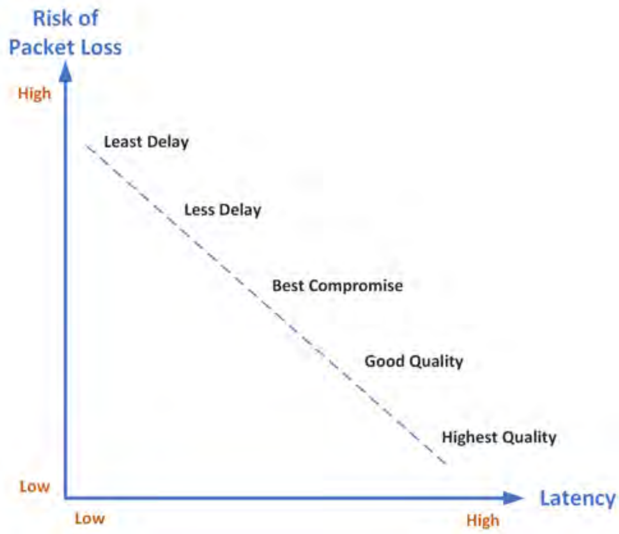
With Tieline codecs you can configure either a fixed or automatic jitter buffer and the settings you use depend on the IP network over which you are connecting. Over LANs, WANs and wireless networks the automatic jitter buffer generally works well. It adapts automatically to prevailing IP network conditions 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

For non-MPX connections, 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 between 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

For non-MPX connections, 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.

20.10 Configuring Forward Error Correction

For non-MPX connections there are two modes of Forward Error Correction (FEC) available in the codec:

1. In-band FEC: Transmits a secondary stream of audio data packets over a single in-band connection. This is relevant for Tieline Music and MusicPLUS encoding as well as Opus. Note: Opus encoding bit-rates are variable and therefore FEC percentages will vary slightly compared to those outlined in the following table.
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	In-band 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	Recommended for wired LAN	Yes	Yes

	to the connection bit rate setting in the codec.	connections & managed T1 & E1 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.

20.11 Enabling Relays & RS232 Data

Product	Feature	Notification
MPX I	RS-232	2 x RS-232 in MPX I (2 sync with audio); up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel; supports hardware and software flow control
MPX II	RS-232	4 x RS-232 in MPX II (4 sync with audio) up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel; supports hardware and software flow control

For non-MPX IP audio connections 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. Over IP it is also possible to enable synchronized out-of-band data in separate packets using any algorithm.

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 ports on the rear panel. Please see Appendix A for RS232 and Control Port Wiring information and see Creating Rules for details on Control Port Contact Closure operation.

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 button.
3. Select **RS232 Config** and press .
4. Select the RS232 port that needs to be configured and press .
5. Use the navigation buttons to select the correct baud rate.
6. Next, press to toggle flow control **On** to enable flow control.

Important Notes:

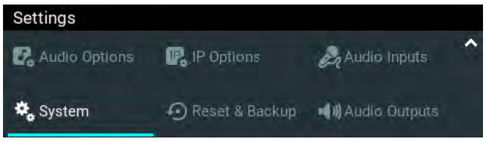
- When connecting to G3 codecs only in-band data is available via the Music and MusicPLUS algorithms. See RS232 Data Configuration for more details.
- It is important that you enable serial port flow control within the codec. Flow control regulates the flow of data through the serial port. If disabled, 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.

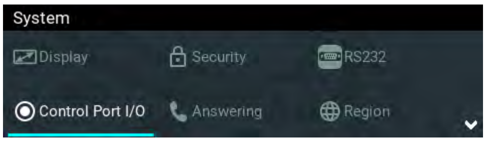
20.12 Monitor Control Port I/O Status

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

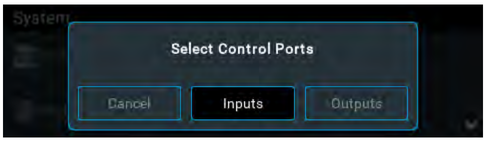
To monitor the status of DB25 **CONTROL PORT I/O** relay inputs and opto-isolated outputs:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System > Control Port I/O** and press the button.

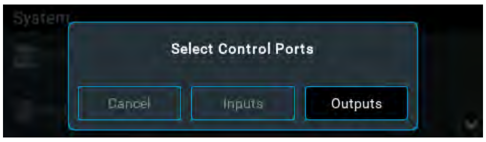




3. Select **Inputs** and then press the button to view input status.



4. Select **Outputs** and then press the button to view output status. Select an output and press the button to toggle the output state from **Off** to **On**. Note: Input states cannot be changed in this way.



20.13 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. Note: This section includes port information for non-MPX connections.

Tieline Codec Default Port Settings

By default, the codec uses a TCP session port to send session data and a UDP port to send audio for non-MPX connections. 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

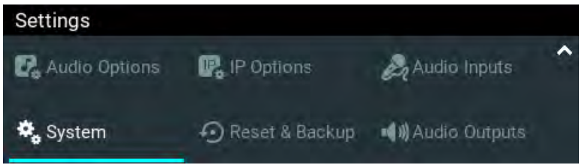
Reasons for adjusting the port setting on your codec include:

- 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.

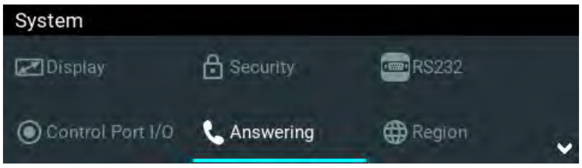
Changing Tieline Session Ports when Answering

To adjust the default local Tieline session data ports (9002 and 9012) used by the codec:

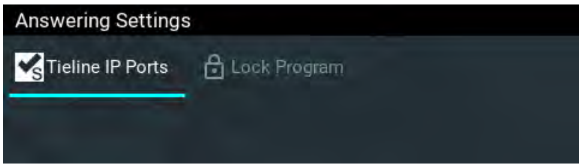
1. Press the **SETTINGS**  button.
2. Use the navigation buttons to select **System** and press the  button.



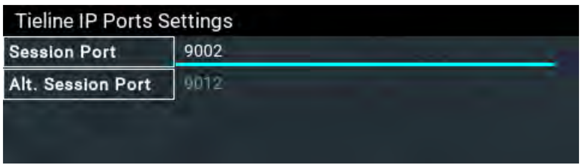
3. Navigate to select **Answering** and press the  button.



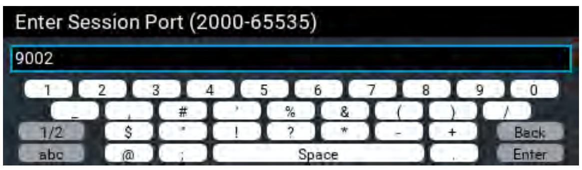
4. Select **Tieline IP Ports** and press the  button.



5. Navigate to **Session Port** or **Alt. Session Port** and press the  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

For non-MPX connections 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).

20.14 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.

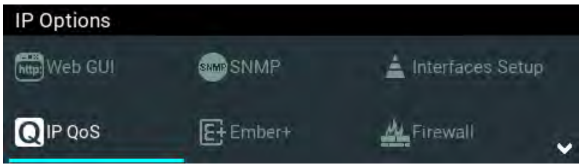
Prioritizing IP Data Packets when Broadcasting

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.

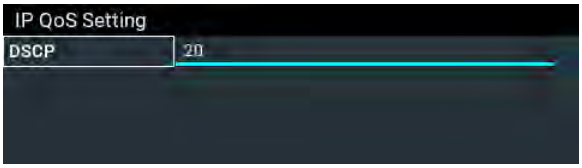
The codec can be programmed to tag IP data packets sent across a network by entering a value into the Differentiated Services Code Point (DSCP) field within the header of data packets transmitted. Check with your IT administrator before changing this setting. By default the codec is programmed for Assured Forwarding and more details about DSCP are available on Wikipedia at http://en.wikipedia.org/wiki/Dscp.

Configuring QoS DSCP

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.
3. Navigate to **IP QoS** and press the button.



4. Press the button to edit the setting and use the **RETURN** button to delete the DSCP value already entered.



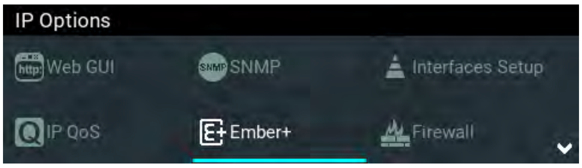
5. Next, use the numeric **KEYPAD** or onscreen keyboard to enter the new setting, then press the button to save the new setting.

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.

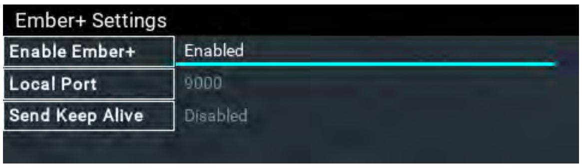
20.15 Configuring Ember+

A single Ember+ provider on TCP port 9000 is configured by default. To adjust this and other settings using the codec front panel:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.
3. Navigate to **Ember+** and press the button.



4. Select **Enable Ember+** and press the button to toggle enabling and disabling Ember+ on the codec.



5. Select **Local Port** to adjust the port number and select **Send Keep Alive** to enable this feature.

For more information on Ember+ see Enable Ember+.

Important Note: Ember+ is currently only supported on **LAN1**, **LAN2**, **LAN3** and not the **AoIP 1** port.

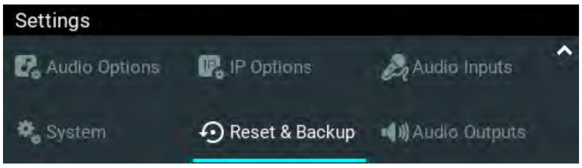
20.16 Reset and Restore Factory Default Settings

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.

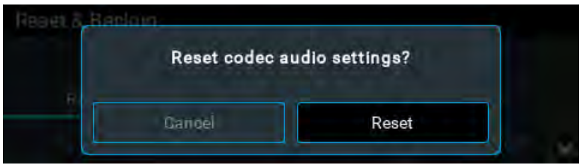
Important Note: After restoring factory defaults, always reboot the codec using the **Reboot Codec** function, not by removing power from the codec.

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

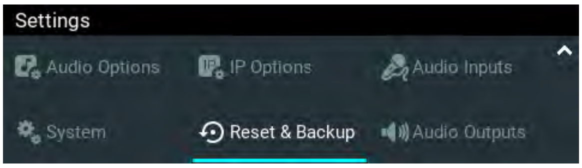
See Reset Factory Default Settings to use the HTML5 Toolbox Web-GUI to reset and restore factory defaults.

20.17 System Backup and Restore

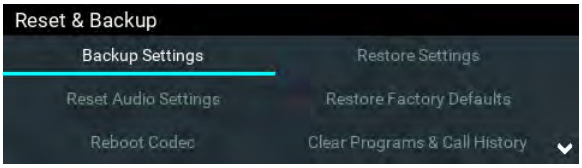
The **Reset & Backup** menu allows users to backup and restore Program information and/or system data to an SD card. Note: A single partition SD card must be used.

Creating a Backup File

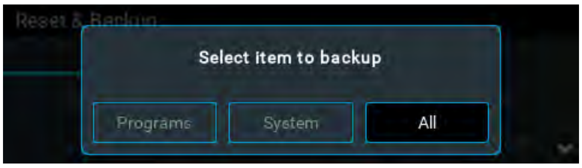
1. Insert a single partition FAT 32 formatted SD card into the **SD CARD SLOT** on the front of the codec.
2. Press the **SETTINGS** button.
3. Use the navigation buttons to select **Reset & Backup** and press the button.



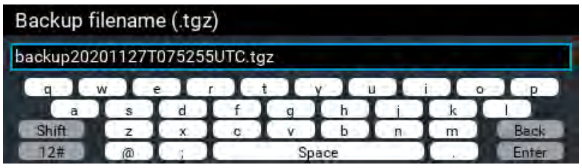
4. Select **Backup Settings** and press the button.



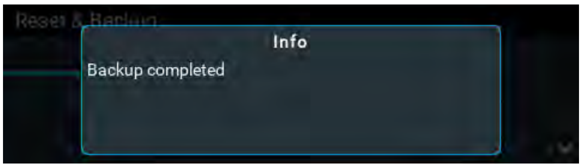
5. Select the preferred backup option and press the button.



6. Edit the file name if required and then navigate to **Enter** and press the button to save the file.



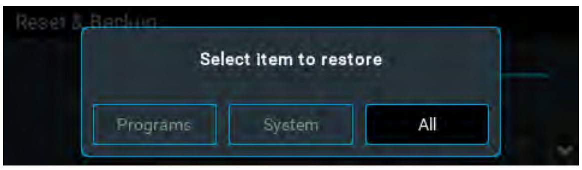
7. A confirmation dialog is displayed when the file is saved successfully.



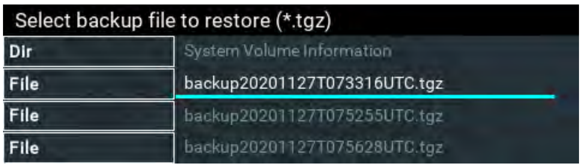
Restoring Data from a Backup File

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 button.
4. Select **Restore Settings** and press the button.

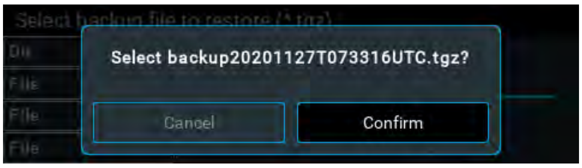
5. Select the preferred restore option and press the button.



6. Select the file to restore and press the button.



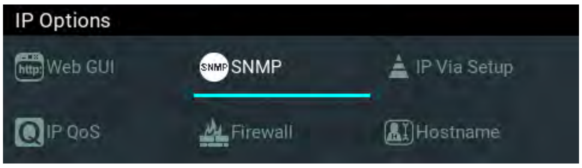
7. A dialog confirms data has been restored successfully.



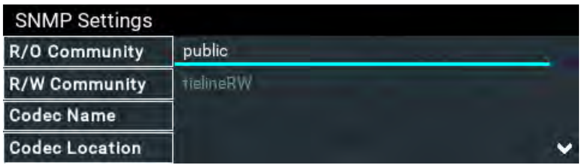
20.18 Configuring SNMP Settings

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 button.
3. Navigate to **SNMP** and press the button.



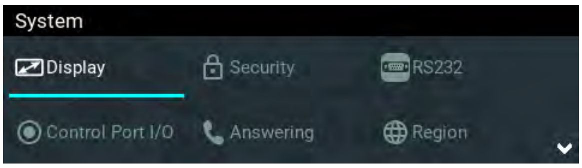
4. Navigate to each setting in turn and press the button to adjust and save each new setting.



Important Note: For more information on SNMP codec settings see Configuring SNMP in the Codec.

20.19 Adjusting the LCD Screen Display

1. Press the **SETTINGS** button and navigate to **System**.
2. Use the navigation buttons to select **Display** and press the button.



3. Use the navigation buttons to select from several options and configure: **Brightness**, **Backlight Sleep** (Enable/Disable), **Sleep After** (time in minutes 1-60), and **Sleep Mode** (screen dim mode or screensaver).

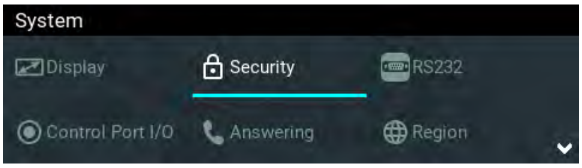
20.20 Locking the Front Panel

The codec features a front panel lock feature for tamper-proof operation. This feature is disabled by default. There are two levels of panel lock and each requires a PIN to access different features:

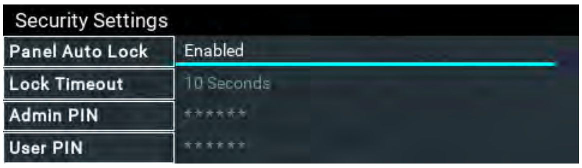
1. **Admin PIN**: Required to change codec connection or configuration settings accessed via the **SETTINGS** button. (Default PIN is: 456789)
2. **User PIN**: Required to use the codec front panel buttons and dial/hangup a connection (Default PIN is: 123456)

Enabling the Front Panel Lock Feature

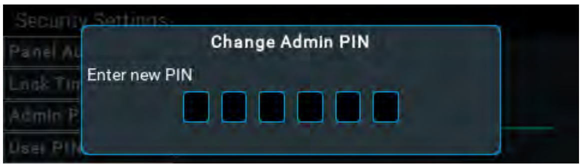
1. Press the **SETTINGS** button and navigate to **System**.
2. Navigate to **Security** and and press the button.



3. Navigate to **Panel Auto Lock** and press the button to toggle from **Disabled** to **Enabled**.



4. Navigate to the **Lock Timeout** field and press to enter the desired time-out period in seconds. Note: The time-out period is the time in seconds before the codec front panel is relocked after being used.
5. To change the default **Admin PIN** or **User PIN**, navigate to each in turn and press to set a new PIN.

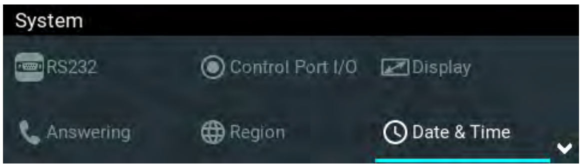


20.21 Adjusting Time Settings

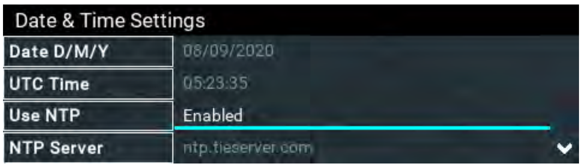
By default **Use NTP** time is enabled in the codec. When the codec is attached to a network it will automatically ping the selected NTP server every two hours and update the time in the codec. It is also possible to manually synchronize the time. Note: The codec will not ping a server while connected.

To adjust time settings in the codec:

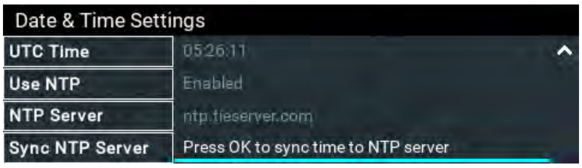
1. Press the **SETTINGS** button and navigate to **System**.
2. Use the navigation buttons to select **Date and Time** and press the button.



3. Navigate to **Use NTP** and press the button to enable or disable this feature.



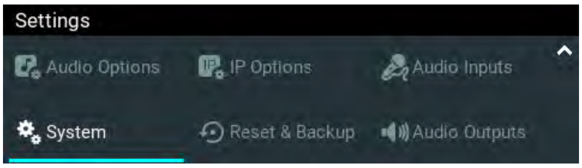
4. Navigate to **Sync NTP Server** and press the button to synchronize the codec time with the designated **NTP Server**.



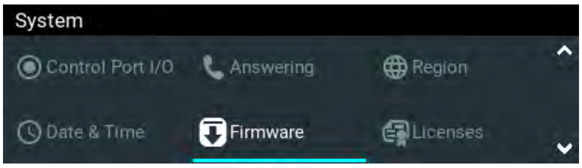
20.22 Upgrading Firmware

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.

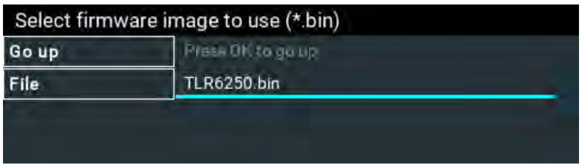
1. Insert the SD card into the **SD CARD SLOT** on the front panel of the codec.
2. Press the **SETTINGS** button.
3. Navigate to **System** and press the button.



4. Navigate down to **Firmware** and press the button.



5. Navigate to the firmware file and press the button to commence the upgrade.



6. Update progress is displayed and when complete the codec will automatically reboot.

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

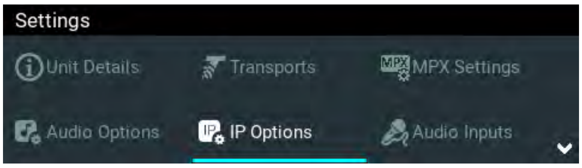
20.23 Installing a Security Certificate

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.

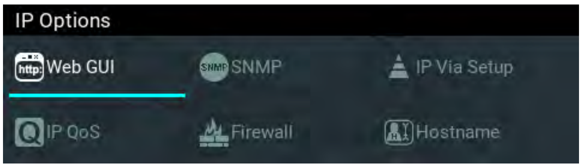
Certificate Installation

To install certificates purchased from a reputable vendor:

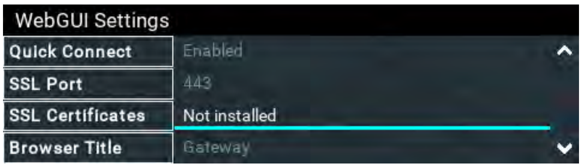
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.



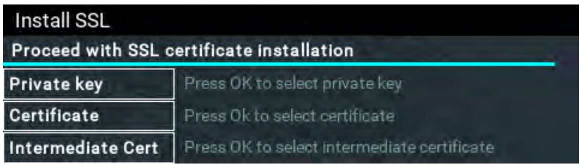
3. Select **Web-GUI** and press the button.



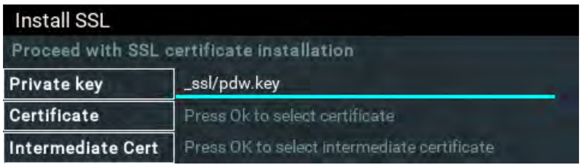
4. Navigate down to **SSL Certificates** and press the button.



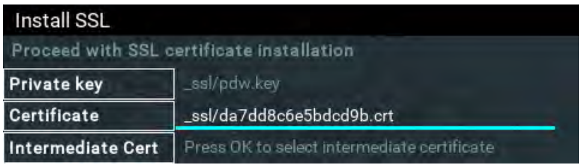
5. Ensure the Private Key, digital SSL Certificate and Intermediate Certificate (if required), are loaded onto an SD card and then insert it into the front panel **SD CARD SLOT**. Note: A single partition FAT32 formatted SD card must be used.



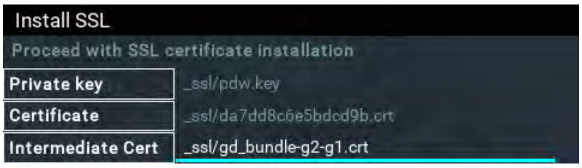
6. Select **Private Key** and navigate to the correct directory and .key (Private Key) file to install from the SD card and press the button.



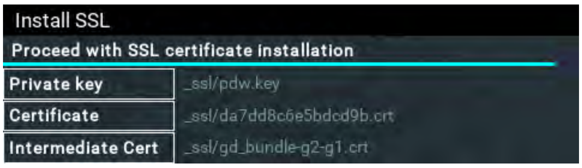
7. Select **Certificate** and navigate to the SSL Certificate (.crt) file on the SD card and press the button.



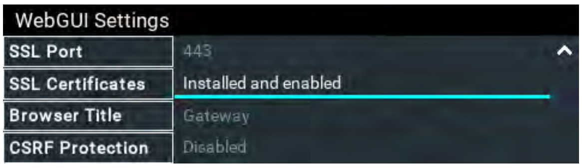
8. If an Intermediate Certificate has been supplied, select **Intermediate** and navigate to the Intermediate Certificate (.crt) file on the SD card and press the button.



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.



10. The **SSL Certificates** menu should confirm the files are successfully **Installed and enabled**.



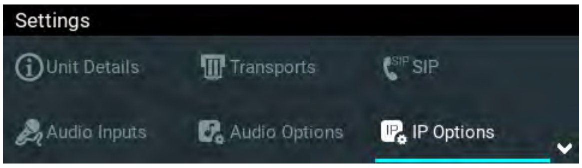
11. To access a codec via the HTML5 Toolbox Web-GUI in a browser after installing SSL security certificates ensure you type "https://" before the codec IP address. For example, https://172.16.0.100.

Important Note: A security certificate can be updated using the **SSL Certificates panel** in the Toolbox Web-GUI. This is only possible after it has been initially installed using an SD card. The codec Web-GUI can not be used to install an SSL certificate on a codec, it can only update it. See Updating an SSL Certificate for more details.

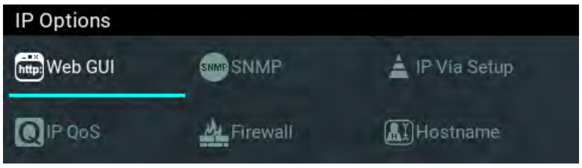
Disable or Remove SSL Security Certificates

To disable or remove installed SSL security certificates from a codec:

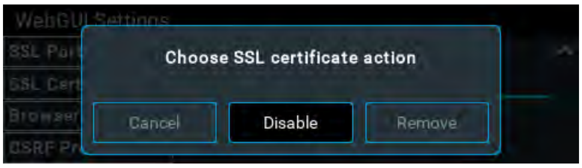
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.



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

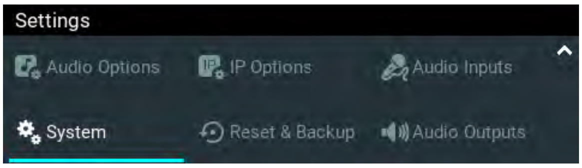
If you type "https://" before the codec IP address and can't open the Toolbox web-GUI, first uninstall the certificates and then reinstall them. Also double-check you are installing the correct certificates. If you continue to have issues, contact your certificate vendor to ensure the certificate is valid.

20.24 Updating Licenses

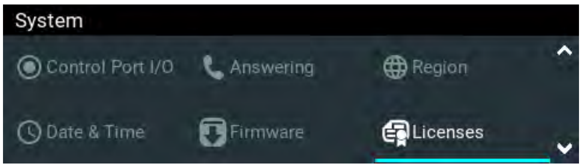
Automatic Software License Installs from TieServer

To automatically install or reinstall a license or licenses on a codec using the front panel of the codec:

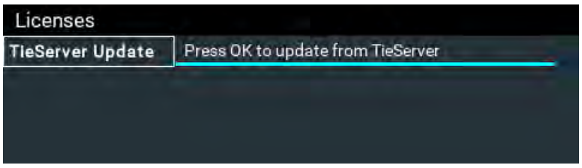
1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.



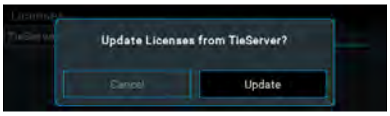
3. Select **Licenses** and press the button.

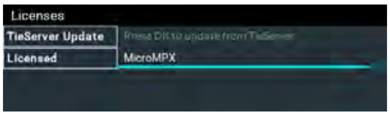


4. Press the button to update any licenses from TieServer that are installed on the codec.



5. Confirm the update to complete it.



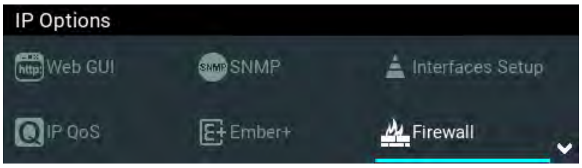


For additional information on license updates see Software License Installation.

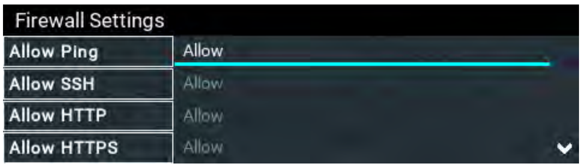
20.25 Firewall Configuration

The **Firewall** menu can be used to enable or disable a range of firewall-related network services, or limit ping to only work in a local subnet. 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 button.
3. Navigate to **Firewall** and press the button.



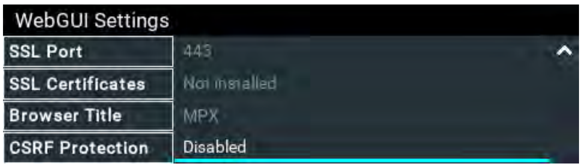
4. Select and configure **Ember+**, **Ping**, **SSH**, **HTTP**, **HTTPS**, **NTP** and **SNMP** firewall options.



20.26 Enabling CSRF Security

CSRF (Cross-Site Request Forgery) protection can be configured to protect the codec from CSRF attacks. To enable or disable this setting:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **IP Options** and press the button.
3. Use the navigation buttons to select **Web-GUI** and press the button.
4. Navigate down to **CSRF Protection** and press the button to toggle between **Enabled** and **Disabled**.

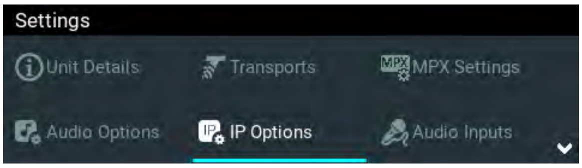


Important Note: Ensure **CSRF Protection** is disabled in the codec when connecting to the CCC. This setting is **[Disabled]** by default and can also be configured using the **Options panel** in the HTML5 Toolbox Web-GUI.

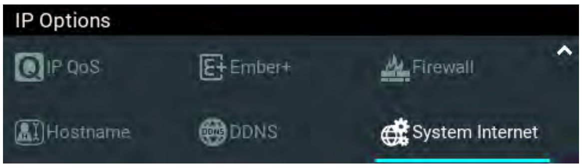
20.27 System Internet Setting

In some situations when multiple interfaces are connected to the codec it may be necessary to select the preference for interfaces used to connect to the internet. For example, when using DDNS services. Or perhaps where the default setting is LAN1, however this may be connected to a closed network and LAN2 is the interface connected to a WAN, i.e. the internet. To adjust the default settings:

1. Press the **SETTINGS**  button.
2. Use 2the navigation buttons to select **IP Options** and press the  button.

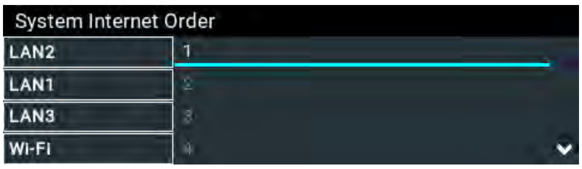


3. Navigate to **System Internet** and press the  button.



4. Navigate to an interface and press the  button to adjust the numeric order as required. Then navigate to **Enter** on the screen and press the  button. In the following example **LAN 2** has been selected as the main interface to use when connecting to the internet.

5. The new setting is reflected in the adjusted interface order.



Important Notes:

- The **System Internet** setting is only related to internet connectivity. It is not related to the IP Interface Setup for IP streaming interfaces.
- If interface 1 is not connected to the internet, the codec will attempt to use interface 2 and other interfaces thereafter to access the internet.
- If the codec is connecting to the Cloud Codec Controller and has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline's various CCC servers in order to be able to contact them. See Enabling the Cloud Codec Controller for more details.

20.28 Digest Upgrade Hold Off LEDs

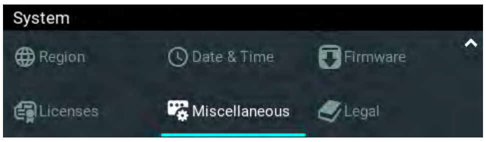
The **Digest Upgrade Hold Off** setting is accessed via the **System tab** in the **Options panel** in the Toolbox web-GUI. It is also accessible from the codec front panel via **Settings > Miscellaneous**.

This value influences the **Connection panel** and Cloud Codec Controller (CCC) connection LEDs by maintaining the lowest LQ value that a connection, stream and program has been in for the last X seconds. The default setting is 0 and in nearly every situation this is the recommended configuration. DO NOT change this setting if you are unsure of its effect on the display in the Connection panel or CCC. LED display settings are as follows:

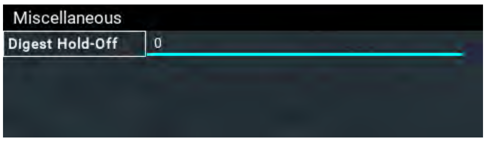
- Gray: Not In use
- Green: Connected and stable (LQ greater than 70)
- Yellow: Connected but connection quality is not stable (LQ 50 to 70)
- Red: Connection establishing or problem with connection (LQ less than 50). LED flashes when establishing connection.

To adjust the setting using the codec front panel:

1. Press the **SETTINGS** button.
2. Use the navigation buttons to select **System** and press the button.
3. Navigate to **Miscellaneous** and press the button.



4. Press the button and use the keypad to edit the **Digest upgrade hold off** period in seconds.



5. Press the button to save the setting.

21 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.

Getting Started at the Studio

To perform a typical codec installation at the studio you will need to:

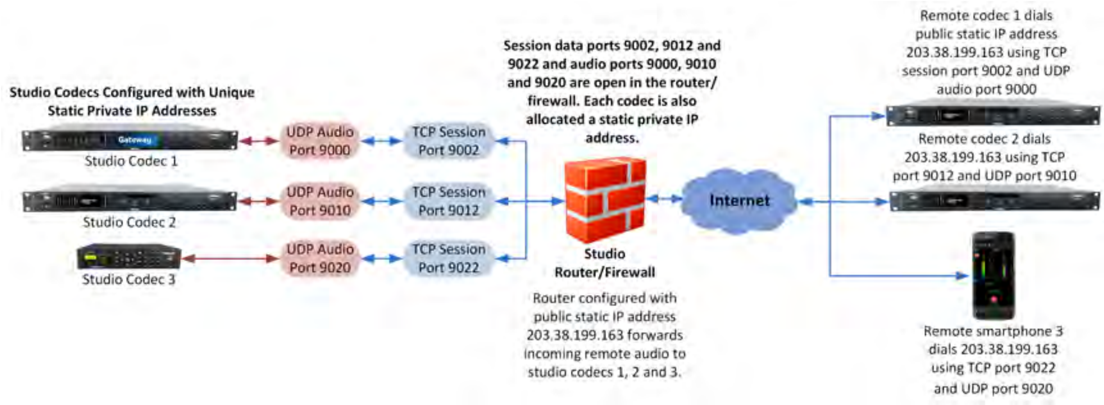
1. Contact your Internet Service Provider and organize a dedicated high speed broadband connection at the studio for your codec with a public static IP address. Do not share this connection with other devices.
2. Install your codec at the studio and attach an active RJ-45 LAN cable to the “LAN” or “Ethernet” 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		MPX I / MPX II Codecs		Bridge-IT II / Bridge-IT XTRA II		Cloud Codec Controller
TCP	UDP	TCP	UDP	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**		Session Port: 9002	Audio Port Stream 1: 9000	Session Port: 9002	Audio Port Stream 1: 9000	HTTP 80
IP2 Session Port: 9012	IP2 Audio Port: 9010	Web-GUI: 80	SIP Session: 5060	Alter-nate Session: 9012	Audio Port Stream 2: 9010	Alter-nate Session: 9012	Audio Port Stream 2: 9010			Alternate Session: 9012	Audio Port Stream 2: 9010	Alternate Session: 9012	Audio Port Stream 2: 9010	HTTPS 443
Toolbox Software: 5550	Toolbox Software: 5550	Alternate Session: 9012	SIP Audio: 5004	Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060			Web-GUI: 80	Audio Port Stream 3: 9020	Web-GUI: 80	SIP Session: 5060	
	SIP Session: 5060	Alternate Web-GUI: 8080	Fuse-IP 8999	Alternate Web-GUI: 8080	Audio Port Stream 4: 9030	Alternate Web-GUI: 8080	SIP Audio: 5004-5054			Alternate Web-GUI: 8080	Audio Port Stream 4: 9030	Alternate 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			TLS/SSL 443	Audio Port MPX1 Link 8854	TLS/SSL 443	Fuse-IP 8999	
					Audio Port Stream 6: 9050						Audio Port MPX2 Link 8874			
					SIP Session: 5060			NMOS 8081	SIP Session: 5060	NMOS 8081		NMOS 8081		
					SIP Audio: 5004-5054			Ember+ 9000	SIP Audio: 5004-5054	Ember + 9000				
					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, Gateway 4, MPX I and MPX II

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.

Critical DNS Network Setup to Complete when Installing Codecs

When installing codecs at the studio it is important to ensure that DNS settings are configured correctly. This is particularly important when using tools like the Cloud Codec Controller and TieLink. Correct DNS settings are also required for NTP, DDNS, use of host names, and if the Download and Install option is used to upgrade codec firmware.

DNS and the Cloud Codec Controller (CCC) and TieLink (if supported)

If a codec has a statically configured IP address, ensure that the DNS Server settings are also configured. This is necessary because the codec must be able to resolve with Tieline’s various CCC and TieLink servers in order to be able to contact them. When using the Toolbox HTML5 Web-GUI to configure DNS network settings:

1. Select **Transport** and then **Network** to open the **Network panel**.
2. Select an interface and then select the **DNS** tab to enter details into the **DNS Addresses** fields as required.
3. Ensure that the **Specify DNS Settings** check-box is selected. Note: This will ensure the DNS servers are used.

Note: These DNS settings can also be configured using the front panel of the codec.

Where can I find my DNS settings?

DNS settings for static public IP addresses can be sourced from your internet provider/Telco. Sometimes the **Gateway** setting in the TCP/IP tab in the **Network panel** will work as the **DNS Address** to enter.

Another common scenario encountered is that static DNS settings are configured to override DHCP settings when they are not compatible with the network being used. When using DHCP IP settings ensure the **Specify DNS Settings** check-box is disabled in the **Network panel**.

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.

22 Reference

The following sections contain reference and troubleshooting information.

22.1 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 fans.

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 internal PSUs and backup temperature alarm features to assist in maintaining reliable operations. The fans have been carefully chosen for long life operation and should not be replaced by a cheaper equivalent. If the internal codec temperature is above 45 degrees Celsius, the fan speed will increase to reduce CPU temperature. Maximum fan speed is reached when the temperature reaches 65 degrees Celsius. Fan speed control circuitry reduces the fan speed as internal temperatures fall below 65 degrees Celsius. This greatly extends the working life of the fans and the codec.

22.2 Compliances and Certifications

Declaration of Conformity

This MPX 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 MPX 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 MPX) 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 MPX) 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.

22.3 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 MPX1 and MPX2
Regulatory Model Number TLR6251 and TLR6252
Product Options ALL

Responsible Party

Tieline Research Pty Ltd
ACN 099-303-045
C/- Tieline America LLC
7202 E.87th Street
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 Pty Ltd

William McLean
Chief Executive Officer

September 19, 2023

For regulatory topics only:

Tieline Pty Ltd
4 Bendsten Place, Balcatta WA 6021, Australia

Ph: 61 8 94132000

Web: www.tieline.com

22.4 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	MPX1 and MPX2
Regulatory Model Number	TLR6251 and TLR6252
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 62368-1:2018/COR1:2020
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
September 19, 2023

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.

22.5 Software Licences

This product uses a combination of proprietary and open source software packages which are licensed as follows.

*** Tieline Proprietary License:

THE PROPRIETARY OR CLOSED SOURCE SOFTWARE USED IN THE PRODUCT IS PROVIDED IN BINARY FORM ONLY BY TIELINE RESEARCH PTY LTD "AS IS" AND ANY EXPRESS OR IMPLIED WARRANTIES, INCLUDING, BUT NOT LIMITED TO, THE IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE ARE DISCLAIMED. IN NO EVENT SHALL TIELINE RESEARCH PTY LTD, TIELINE AMERICA LLC, OR ITS VENDORS BE LIABLE FOR ANY DIRECT, INDIRECT, INCIDENTAL, SPECIAL, EXEMPLARY, OR CONSEQUENTIAL DAMAGES (INCLUDING, BUT NOT LIMITED TO, PROCUREMENT OF SUBSTITUTE GOODS OR SERVICES; LOSS OF USE, DATA, OR PROFITS; OR BUSINESS INTERRUPTION) HOWEVER CAUSED AND ON ANY THEORY OF LIABILITY, WHETHER IN CONTRACT, STRICT LIABILITY, OR TORT (INCLUDING NEGLIGENCE OR OTHERWISE) ARISING IN ANY WAY OUT OF THE USE OF THIS SOFTWARE, EVEN IF ADVISED OF THE POSSIBILITY OF SUCH DAMAGE. YOU MAY NOT MAKE ANY CHANGES OR MODIFICATIONS, DECOMPILE, DISASSEMBLE, OR OTHERWISE REVERSE ENGINEER THE SOFTWARE OR USE THE SOFTWARE ON ANY HARDWARE OR PRODUCT EXCEPT WHICH TIELINE RESEARCH PTY LTD LICENSES THIS SOFTWARE FOR.

*** Open Source Licenses:

The open source software included in this product contains copyrighted software is licensed under various open source licenses (e.g. GNU General Public License v2/v3, GNU Lesser GPL v2.1/v3, BSD 3/2/4 Clause, MIT, ISC, etc.). The licenses are listed on the product via the Settings menu, About, Legal. You may request a copy of the source code of the open source software used in this product on DVD, by contacting our support team on +61 (0)8 9413 2000. Tieline Research Pty Ltd will charge a small handling fee for distribution of this software.

* Attributions and Credits:

o This product includes software developed by the University of California, Berkeley and its contributors.
o This product includes software developed by the University of California, Lawrence Berkeley Laboratory and its contributors.
o This product includes software developed by the OpenSSL Project for use in the OpenSSL Toolkit. (http://www.openssl.org/).
o This product includes cryptographic software developed by Eric Young (eay@cryptsoft.com)
o This product includes software developed by Eric Young (eay@cryptsoft.com)
o This product includes software developed by Computing Services at Carnegie Mellon University (http://www.cmu.edu/computing/).
o This product includes software developed by Paul Mackerras <paulus@samba.org>
o This product includes software developed by Pedro Roque Marques <pedro_m@yahoo.com>
o This product includes software developed by Tommi Komulainen <Tommi.Komulainen@iki.fi>
o This product includes software developed by the Computer Systems Engineering Group at Lawrence Berkeley Laboratory.
o This product includes software developed by the Kungliga Tekniska Högskolan and its contributors.
o This product includes software developed by Yen Yen Lim and North Dakota State University o This product uses the FreeType library.

22.6 Trademarks and Credit Notices

1. Windows is a registered trademark of Microsoft Corporation in the United States and/or other countries.
2. Windows XP, Windows Vista, Windows 7, Windows 10 are either trademarks or registered trademarks of Microsoft Corporation in the United States and/or other countries.
3. Firefox is a registered trademark of Mozilla Corporation in the United States and/or other countries.
4. Solaris is a trademark of Sun Microsystems Inc. in the United States and/or other countries.
5. Linux is the registered trademark of Linus Torvalds in the U.S. and other countries.
6. Java is a trade mark Sun Microsystems Inc. in the United States and/or other countries.
7. Comrex® Vector, Matrix® and BlueBox® are registered trademarks of Comrex Corporation in the United States and/or other countries.
8. APT is a registered trademark of Audemat group in the United States and/or other countries.
9. Mayah is a registered trademark of MAYAH Communications GmbH in the United States and/or other countries.
10. Wheatnet and Wheatstone are trademarks of Wheatstone Corporation in the United States and/or other countries.
11. Telos, Zephyr and Xstream are registered trademarks of Telos Systems in the United States and/or other countries.
1. AXIA® and LIVEWIRE® are registered trademarks of The Telos Alliance in the United States and/or other countries.
2. LIVEWIRE is a registered trademark of Axia Audio in the United States and/or other countries.
3. xNode is a trademark or registered trademark of The Telos Alliance in the United States and/or other countries.
4. RAVENNA® is a registered trademark of ALC NetworX GmbH in the United States and/or other countries.
5. Dante® is a registered trademark of Audinate Pty Ltd in the United States and/or other countries.
6. Lawo is a registered trademark of LAWO AG in the United States and/or other countries.
12. Prodys and Prontonet LC are registered trademarks of Prodys S.L. in the United States and/or other countries.
13. MPEG Layer-3 audio coding technology licensed from Fraunhofer IIS and Thomson Licensing.
14. AAC audio coding technology licensed from VIia Licensing Corporation.
15. IOS is a trademark or registered trademark of Cisco in the U.S. and other countries and is used under license.
16. Wi-Fi® and Wi-Fi Protected Access® (WPA) are registered trademarks of Wi-Fi Alliance.
17. Other product names mentioned within this document may be trademarks or registered trademarks, or a trade name of their respective owner.
18. MPEG audio technologies licensed by Fraunhofer IIS. (http://www.iis.fraunhofer.de/audio)

23 MPX Specifications

Specifications	
MPX I Rear Panel Input and Outputs	
RJ-45 Analog/AES3 Inputs 1	1 Shared Stereo Analog Line Input or stereo AES3 Input
RJ-45 Analog/AES3 Outputs 1	1 Shared Stereo Analog Line Output or stereo AES3 Output
RJ-45 AES192 MPX Input/Output 1	1 Stereo AES192 Digital MPX Composite Input and Output
BNC Analog Input 1/Outputs 1-2	1 BNC Analog MPX composite Input and 2 BNC Analog MPX Outputs (1 A & B output)
DB-25 Serial Port	2 x Serial I/O Ports
MPX II Rear Panel Input and Outputs	
RJ-45 Analog/AES3 Inputs 1 & 2	2 Shared Stereo Analog Line Inputs or 2 x stereo AES3 Inputs
RJ-45 Analog/AES3 Outputs 1 & 2	2 Shared Stereo Analog Line Outputs or 2 x stereo AES3 Outputs
RJ-45 AES192 MPX Inputs/Outputs 1 & 2	2 Stereo AES192 Digital MPX Composite Inputs and Outputs
BNC Analog Input 1 & 2/Outputs 1-4	2 BNC Analog MPX composite Inputs and 4 BNC Outputs (2 A & B outputs)
DB-25 Serial Port	4 x Serial I/O Ports
Common MPX I and MPX II Rear Panel Input and Outputs	
Optional DVB Satellite card/module	DVB-S/S2 RF1 Input and RF2 Output via F connectors
Gigabit RJ-45 LAN1 and LAN2 Ports	Gbit Ethernet ports for Control/IP streaming over WANs
Gigabit RJ-45 LAN3 Port	Gbit Ethernet port for Control
Gigabit RJ-45 AoIP/LAN4 Port	Gbit Ethernet AoIP/LAN4 Port
BNC Sync Input/Output	2 configurable Input/Output BNC sync connectors
Female DB-25 Control Port In/Out	DB-25 Control Port I/O Supporting: 4 GPIOs in MPX I; 8 GPIOs in MPX II
Front Panel Inputs and Outputs	
Front Panel Headphone Output	1 x 6.35mm (1/4") headphone Jack
Front Panel SD Card Slot	Full size SD card slot for firmware upgrades and backup files
Data and Control	
Configure, Control, and Monitoring	HTML5 Toolbox Web-GUI, Cloud Codec Controller (CCC)
Serial (DB-25)	2 x RS-232 in MPX I; 4 x RS-232 in MPX II (one sync with audio) up to 115kbps with or without CTS/RTS flow control can be used as a proprietary data channel; supports hardware and software flow control
Hardware / Software Logic I/Os (SLIO)	Support for software logic I/Os
Monitoring and Alarms	
Demodulated Audio Monitoring	Demodulated monitoring of left and right composite audio from MPX signal encoder or decoder using Opus.
Front Panel PPMs	Front Panel PPMs to monitor MPX levels or input/outputs: 4PPMs in MPX I; 8 PPMs in MPX II
Front Panel LEDs	Configurable Alarm and User LEDs with Connection and Power LEDs
Embedded HTML5 Web Server	HTML5 Toolbox Web-GUI used for configuration and control of all functions including alarms

Comprehensive Alarms	Configurable alarms including automated silence detection, PSU, connection, temperature, AES input, AES reference, loss of pilot
SNMP	Support for SNMP monitoring and traps via SNMP v1 and SNMPv2c
Encoding, Decoding, and IP Streaming	
MPX Encoding/Decoding	• 16-bit (2.3 Mbps) and 24-bit (4.6Mbps) Raw MPX; • µMPX 320, 340, 360, 384, 400, 448, 512, 576, 640, 704, 768, 800, 900 kbps; • Support for µMPX v1, 2, 3 and 4 • Sampling frequency: 192kHz Digital
MPX Buffer	0 to 10 seconds
MPEG-TS Encoding/Decoding	DVB-S and DVB-S2 supported via optional DVB module; 16, 24 and 32-bit PCM at 96kHz; support for MPE forwarding of MPX via IP
IP (non-MPX) 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 Fraunhofer IIS (http://www.iis.fraunhofer.de/audio)
Uncompressed IP	Linear PCM 12/16/20/24 bit 32kHz, 44.1kHz, 48kHz, 96kHz sampling
IP Sample Frequencies	8kHz, 16kHz, 32kHz, 44.1kHz, 48kHz, 96kHz
FEC	Selectable FEC 15% -100% with time delay
Protocols	RTP, DHCP, SNMPv2c, DNS, HTTP, IGMPv3, IPv4/IPv6, RTCP, SSL Security Certificate, RIST
Redundant Streaming	Primary plus automatic failover to 1 MPX/µMPX redundant stream per stream with hitless packet switching at the decoder, RIST
Jitter Buffer Management	5 automated, or fixed/static jitter buffer settings for IP stream monitoring
Multi-unicasting	Support for IP or MicroMPX multi-unicast streams; up to 10 connections in total
Multicasting	Support for Multicast streams to unlimited endpoints over compatible IP networks
Satellite Tuner DVB-S – ETSI EN 300 421 (Optional)	
Frequency	950 to 2150 MHz
Input Level	-65 to -25 dBm
Standard Modulation	QPSK
Symbol Rate	1~54 Msps
FEC	Enhanced FEC decoder
Satellite Tuner DVB-S2/S2X – ETSI EN 302 307-1 (Optional)	
Standard Modulation/Symbol Rate	QPSK: 1~60 Msps; 8PSK: 1~60 Msps; 16APSK: 1~58 Msps ; 32APSK: 1~58 Msps ; 64APSK: 1~34 Msps
Modulation Type	VCM, ACM
Roll-off	Roll-off factors from 0.05 to 0.35
FEC	Normal (64800 bits) FECFRAME supported
Transport Stream Processing	Single Transport Stream, Multiple Transport Stream
Data Forwarding	IP
AoIP Standards Specifications	
AES67 Compliant	Compliant with Audio Engineering Society standards for AES67
ST 2110-30 Compliant	Class A, Ax, B, Bx Sender and Receiver Compliant
RAVENNA Compliant	Natively supports RAVENNA Stream Discovery and Advertisement

Livewire Compliant	Natively supports Livewire+ for AoIP streaming
NMOS Compliant	NMOS IS-04 & IS-05 Discovery, Registration and Connection Management
Ember+	Supports the Ember+ control protocol
Supported Audio Frames	125ms, 250ms, 333ms, 1ms, 4ms
Clock Modes Supported	Primary Leader, Follower, Follower Only
Advanced Networking	
VLAN Tagging	IEEE 802.1Q, 802.1p
Quality of Service (Q0S)	Support for DiffServ (DSCP)
Synchronization	EEEE1588-2008 (PTPv2)
Multicasting	IGMP v2 and v3
SAP	SAP v2 (Session Announcement Protocol) as defined in RFC 2974
General	
Display	24-bit Color LCD Screen (480 x 128 pixels)
Keypad	26 button silicon keypad
Navigation	5 navigation and selection buttons
Size	1U x19" Rackmount
Dimensions	19" x 1 3/4" x 11 13/16" [482mm (W) x 44.45mm (H) x 300mm (D)] excluding rear connectors
Weight (including Satellite module)	4.08kg/8.99 pounds
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	10% ≥ RH ≤ 90% (0 to 45°C/32°F to 113°F), non-condensing

24 Appendix A: Control Port / RS-232 Pinouts

Product	Feature	Notification
MPX I	Control Ports	One DB-25 connector supporting 4 CMOS solid state relays for the control of equipment, consisting of 4 relay closures and 4 opto-isolated outputs.
MPX II	Control Ports	One DB-25 connectors supporting 8 CMOS solid state relays for the control of equipment, consisting of 8 relay closures and 8 opto-isolated outputs.
MPX I	RS-232 Serial Ports	2 RS-232 Serial I/O Ports (2 sync with audio).
MPX II	RS-232 Serial Ports	4 RS-232 Serial I/O Ports (4 sync with audio).

Relays

A DB-25 connector facilitates the use of CMOS solid state relays for the control of equipment. 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.

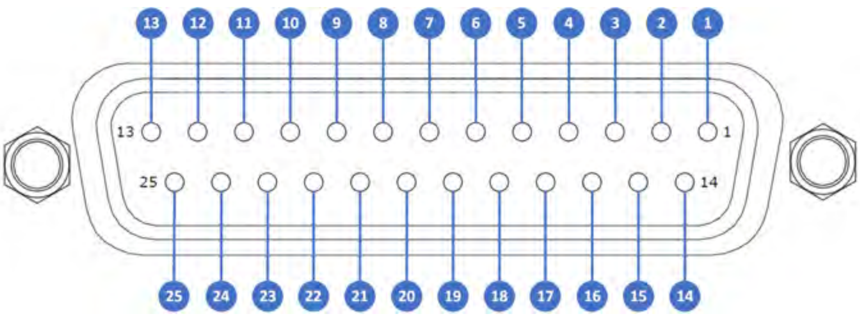
Operation is as simple as joining the input pin to the ground terminal. This can be via a remote relay contact or the open circuit collector of a transistor or FET. DO NOT feed voltages into the inputs.

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.



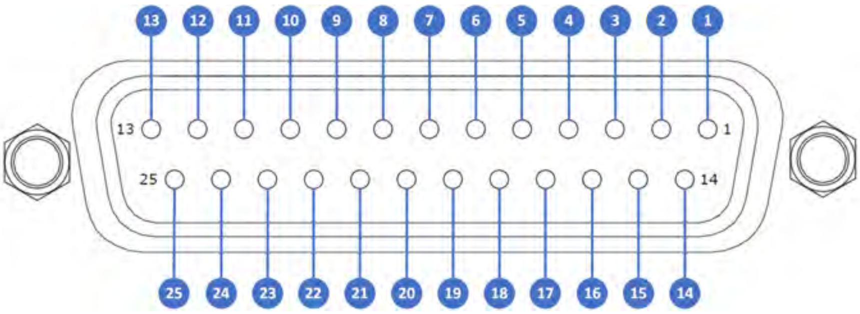
Female DB-25
Codec Connector I/O 1-8

Pin	Pin Function
1	Output 1A
14	Output 1B
2	Output 2A
15	Output 2B
3	Output 3A
16	Output 3B
4	Output 4A
17	Output 4B
5	Output 5A
18	Output 5B
6	Output 6A
19	Output 6B
7	Output 7A
20	Output 7B
8	Output 8A
21	Output 8B
9	Input 1
22	Input 2
10	Input 3
23	Input 4
11	Input 5
24	Input 6
12	Input 7
25	Input 8
13	Ground

Important Note: For more information about how to program relay operations with a PC using the Toolbox Web-GUI, please see Creating Rules.

RS232 Pin-outs and Data Connections

One rear panel DB-25 connector supports 2 RS-232 serial ports in MPX I codecs and 4 RS-232 Serial I/O Ports in MPX II codecs. Pin-outs are as follows:



Female DB-25

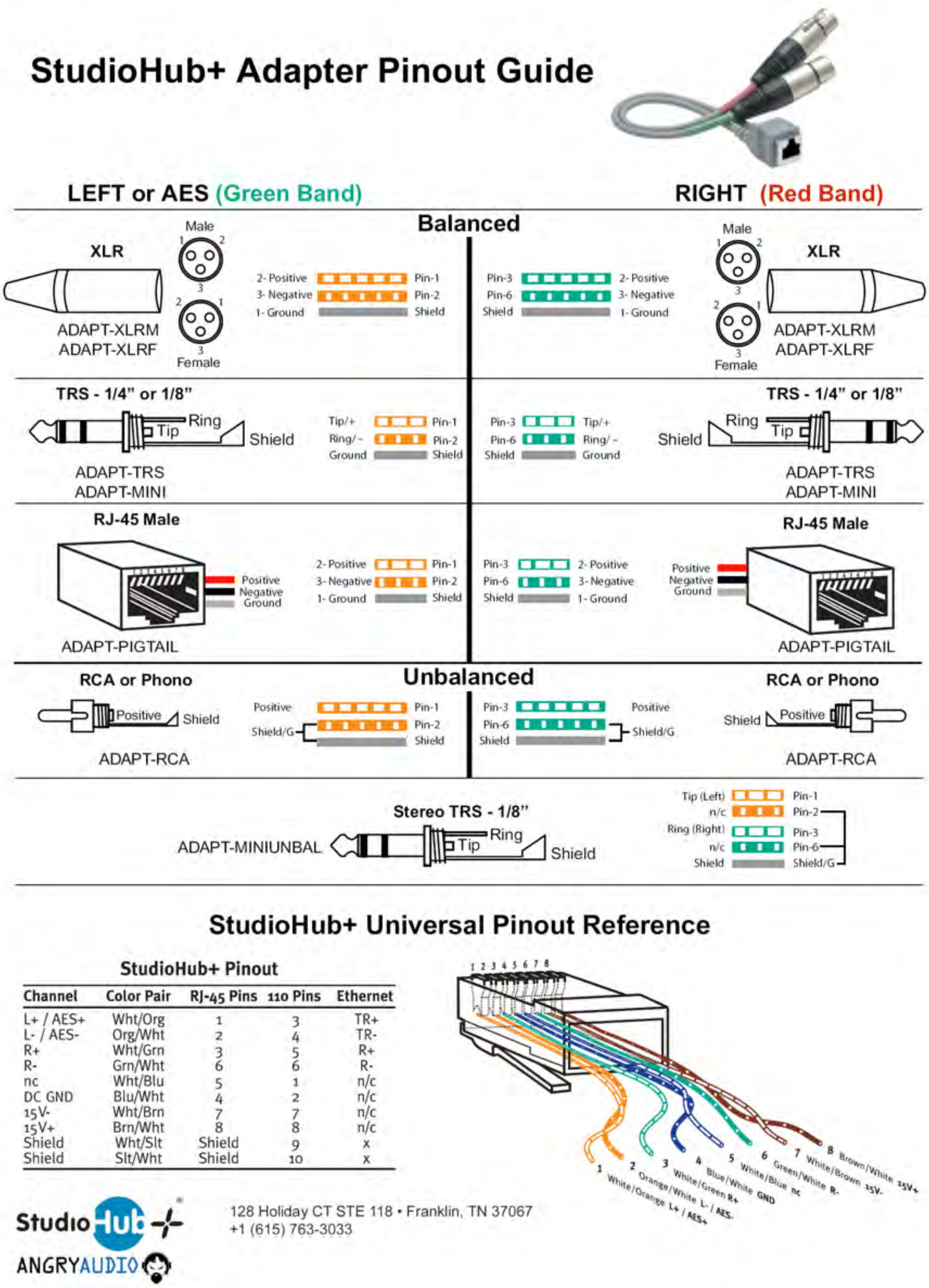
Pin	DATA Male DB25 (RS232) DTE	DATA Female DB25 (RS232) DCE
1	Ground	Ground
14	No connection	No connection
2	RX1	TX1
15	RTS1	CTS1
3	TX1	RX1
16	CTS1	RTS1
4	Ground	Ground
17	No connection	No connection
5	RX2	TX2
18	RTS2	CTS2
6	TX2	RX2
19	CTS2	RTS2
7	Ground	Ground
20	No connection	No connection
8	RX3	TX3
21	RTS3	CTS3
9	TX3	RX3
22	CTS3	RTS3
10	Ground	Ground
23	No connection	No connection
11	RX4	TX4
24	RTS4	CTS4
12	TX4	RX4
25	CTS4	RTS4
13	Ground	Ground

Important Notes:

- It is important to 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.

25 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.



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

Index

- A -

- B -

- C -

- D -

- I -

- J -

- L -

- Q -

- R -

- S -

- T -

- U -

- V -

- W -

- X -