Managing the Transition from ISDN to IP

Tieline®

The Codec Company

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Introduction

If you're a radio engineer, there's a pretty good chance you know about how ISDN is being phased out by Telcos around the world. What is perhaps not as well known by some is how to seamlessly transition from ISDN network infrastructure to an IP network environment.

When the first murmurs of ISDN's imminent demise surfaced a few years ago, many radio engineers were fearful that there was nothing available to replace those trusty ISDN lines. Unknown to many engineers, Tieline had already been providing extremely reliable IP solutions for nailed-up STLs and remotes for years. So Tieline simply embraced the opportunity and embarked on educating customers about how IP actually has numerous advantages over ISDN.

Most broadcasters now clearly recognise that IP networks are more flexible, cheaper to upgrade and if configured correctly, can be just as reliable as ISDN. As a result, broadcasters are using IP audio codecs to design and operate more adaptable broadcast networks with streamlined work flows, reduced operating costs and the ability to remote control them from anywhere in the world.

Before we talk a bit more about how we as radio engineers can transition from ISDN to IP with a minimum of fuss, let's reminisce a little about why ISDN has been so prevalent, and why we need to transition to IP.

Historical Perspective of ISDN

ISDN (Integrated Services Digital Network) is an international communications standard for sending voice, video, and data.

Since the 1990s, radio broadcasters around the world have relied upon circuit switched ISDN to deliver highly reliable low latency voice and data streams. Broadcast stations and networks quickly adopted the service for point-to-point transmission and for many networks ISDN has been the preferred means of connection for over 25 years.

Traditionally, the only drawbacks were the initial expense of the transmission equipment, line rental costs, and the advanced planning required to install ISDN circuits for remote broadcasts.

Amongst broadcasters, the Basic Rate Interface (BRI) has been the most popular. It consists of two 64-Kbps B-channels and one D-channel for transmitting control information.

So why is ISDN being phased out?

Telcos recognized long ago that IP presents new opportunities and have been decommissioning ISDN infrastructure and replacing it with IP network infrastructure capable of higher bandwidth connections, at a lower cost.

Throughout the USA, Europe and Australasia, circuit-switched ISDN networks are either being scaling down or phased out completely and there has been a huge expansion of terrestrial IP and wireless IP broadband infrastructure. There are a few key reasons driving the demise of ISDN as a broadcast transport:

Telcos have stopped installing new ISDN infrastructure

If you can't get a line installed it's just not viable to use ISDN for your remotes or fixed line installations. This is also exacerbated by a lack of qualified ISDN service technicians.

Telcos are removing existing ISDN infrastructure

It's cheaper for Telcos to install and maintain IP infrastructure and it is also far superior in its data capabilities.

ISDN Circuits are Expensive

ISDN is more expensive to install and maintain than IP. ISDN line costs exceed the cost of DSL data plans which are adequate for reliable IP connections.

" IP Presents new opportunities "

IP is Not a New Paradigm for Tieline

It may surprise some people, but thousands of Tieline codec owners have been using IP successfully over the public internet for over a decade. In fact, Tieline was the first codec manufacturer to offer solutions over IP, ISDN and POTS way back in 2004. Amazingly, this was all possible using a single codec!

Many codec manufacturers implemented IP with the proviso that your network needed to have Quality of Service (QoS), or guaranteed bandwidth. However, Tieline has always recognized that for true broadcast flexibility you must be able to successfully broadcast over imperfect IP networks like the public internet. As a result, Tieline's wired and wireless solutions have always been engineered to perform over all IP networks – managed and unmanaged. When you're using Tieline codecs over IP you can be sure you are backed by the technological expertise of a competent and highly experienced manufacturer.

Why is this important? Well for IP audio to be sent reliably over unmanaged IP networks like the public internet, it's crucial that a codec employs IP network management strategies to address latency or "jitter" and lost packets. But we'll explain more about that later.

Tieline: A Leadership Role in Developing IP Solutions

Tieline assumed a leadership role in IP technology development from the beginning. In 2006, Tieline was the first non-European member of the Audio-via-IP Experts Group, along with AETA, Orban and Mayah. Tieline and these other manufacturers took a leadership role in developing SIP (Session Initiation Protocol) interoperability between Tieline IP audio codecs and other codec manufacturers.

This body of work ensured interoperability between manufacturers who have implemented EBU specifications for interoperability over IP using SIP (EBU N/ACIP Tech 3326). All Tieline IP codecs are EBU N/ACIP Tech 3326 compatible and can connect to other codec brands compliant with these standards.

Innovative and Modular Wireless IP Solutions

Ever since Tieline first offered 2G GSM data codec broadcasting solutions way back in 2004, the company has been at the forefront of wireless technology solutions. Tieline also delivered the first 3G wireless IP codec solution way back in 2005!

Tieline has always understood that for technology to be agile, a modular approach to hardware and software development is required. The same Tieline codecs purchased over 10 years ago can still be upgraded and connected to the latest 4G-LTE networks today. Why? Tieline delivers modular solutions which can be adapted over time. So, rather than have to purchase a new codec, customers only need to purchase a new wireless module to continue to keep pace with wireless broadband networks as they are upgraded.

Tieline was also the first manufacturer to deliver an iPhone codec in 2009 via the Report-IT application. Report-IT has revolutionised smartphone reporting and is the world's most popular app for live and recorded reports.

Combining Tieline's Merlin PLUS codec with Report-IT has delivered our stations greater remote flexibility. The workflow for all of our simple remotes has been streamlined and we have freed up significant resources for other purposes.

Mike Hutchens, Townsquare Media

IP Performance Compared to ISDN

As an engineer, your first questions when looking for an alternative to ISDN are:

1. Will IP deliver the same quality as ISDN?

2. Will IP perform as reliably as ISDN?

The answer to these questions is an emphatic yes and yes! The transition to IP provides a range of new opportunities for broadcasters to upgrade their networks and make them more flexible.

Following are 10 reasons why you should consider migrating from ISDN to IP.

1. Broadcasting over IP is cheaper:

   • IP network broadband costs are much cheaper than synchronous data networks. IP network infrastructure is cheaper because you can distribute broadcast quality audio for a low monthly fee over DSL/ADSL networks.

   • If you use an IP codec over wireless broadband there's no need to pay for the installation of expensive ISDN lines for your remotes.

   • In general, the cost advantages of packet-switched IP networks over circuit-switched networks have become too great to ignore.

2. IP hardware is cheaper and delivers economies of scale:

   • IP offers higher bandwidth connections in most circumstances when compared to ISDN. This means a single IP audio codec can send large numbers of IP audio streams to multiple end-points via multicast and multi-unicast technologies, so less hardware is required than over synchronous networks like ISDN.

   • In many cases savings in monthly line rental costs will pay for the cost of new IP hardware in just months.

   • The cost of IP data is continually decreasing as Telco network infrastructure is upgraded.

3. Broadcasting over IP is more flexible:

   • Routing audio over IP is more flexible because a single IP audio codec can deliver a choice of unicast, multicast and multi-unicast IP streams for network audio distribution.

   • Wireless cellular broadband offers more flexibility than traditional leased line solutions.

4. IP networks can be scaled to suit individual installations:

   • Internet Service Providers and Telcos offer a range of competitively priced data plans that provide flexible bandwidth to suit each installation - minimising data costs and maximising efficiency.

   • Engineers can incrementally purchase additional network data as demands increase over time.

5. Wireless IP networks deliver flexible broadcast connections from anywhere at any time a range of wireless networks are available to broadcast audio over IP, including:

   • Wireless cellular broadband via 4G-LTE and 3G networks.

   • Long-range WiMAX wireless IP networks.

   • Wireless BGAN satellite IP connections.

6. IP networks are widely available:

   • Terrestrial IP networks are continually being upgraded and expanded in most regions.

   • Cellular broadband networks are widely available in most regions of the world.

7. Setting up remote IP broadcasts is extremely simple:

   • With wireless IP there's no longer any lead time required to have ISDN lines installed for remote broadcasts.

   • There is no longer any need for remote vans and cumbersome microwave links when a wired or wireless IP codec can do the same job at a fraction of the cost.

   • An IP codec can be preconfigured to connect over an IP network at the touch of a single button.

> Universal availability is the best part about broadcasting over IP. We can send a feed from anywhere with IP connectivity, to anywhere with IP connectivity. We only connect over IP and integrate both DSL and cable, plus some locations have fiber available.
>
> **Cris Alexander, Director of Engineering, Crawford Broadcasting**

8. Audio over IP integrates seamlessly with new broadcast technologies:

   - Packet-based IP audio integrates seamlessly with other studio technologies when broadcasting audio streams over the internet.

9. Integration of audio over IP into large radio networks creates economies of scale:

   - Opportunities to consolidate and centralise the distribution of audio around radio networks and affiliates is facilitated by the flexible and scalable nature of IP codec hardware and broadband network infrastructure.

   - Smartphone codecs allow radio networks to manage live streaming, recording and file uploads to servers.

10. Audio over IP is the future of broadcasting

    - Many major networks around the world have already migrated to IP.

    - Circuit-switched ISDN networks are either being scaling down or phased out and there has been a huge expansion of terrestrial and wireless broadband infrastructure.

    - Audio over IP can adapt to meet the changing needs of technology.

    - Regular DSL/ADSL data plans are sufficient to deliver 20Hz to 22kHz audio quality for audio distribution or studio-to-transmitter link applications.

Without hesitation, if it is smartphone or hardware codecs you are looking for, it is worth trying Tieline codecs and Report-IT for high quality mobile reporting.

Jerome Bellini, President of NIP Productions and Technical Director for Radio Monaco

Using IP Codecs

IP is a packet-based computer network protocol used to send data packets over Local Area Networks (LAN) and Wide Area Networks (WAN). To send broadcast audio over IP you need a broadcast codec at the remote site to encode the audio into data packets and another one at the studio to decode the data packets.

The codec at the studio needs to be connected to a reliable broadband connection, preferably with a static public IP address, so that any remotely located codec can dial and connect. The remote codec can connect over a LAN or cellular wireless broadband connection.

LAN connections can connect directly to the public internet, whereas cellular broadband phone networks provide data links to internet connections managed by the Telco/Internet Service Provider (ISP).

Codec at Studio

6 simultaneous bidirectional mono remotes

You don't need to be an expert in IT to set up an IP codec

Just like you used to order a leased line from your Telco and get them to install it, your Telco will set up your fixed line IP broadband connection - preferably with a static public IP address.

You or your IT administrator can also configure Network Address Translation (NAT) between the public Internet, private Local Area Networks (LANs), Port Forwarding and any other routing required. Tieline also has support personnel available to ensure you get up and running with a minimum of fuss.
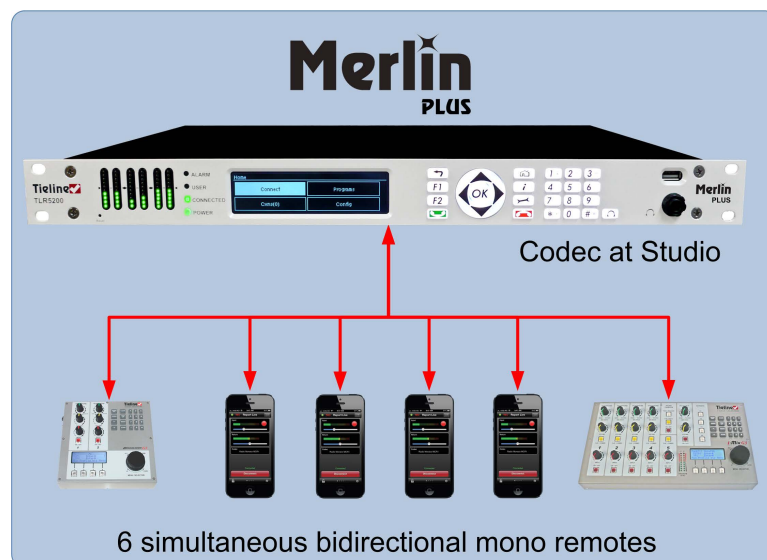
IP Issues to Consider

What's all the fuss about IPv4 and IPv6?

With IPv4 addresses nearing exhaustion, and the world moving inexorably towards IPv6 becoming the dominant internet infrastructure, IP audio codecs and other devices should be capable of operating in both IPv4 and IPv6 environments. All Tieline G5 codecs support both IPv4 and IPv6 (Dual Stack) protocols, so broadcasters can be assured that they have made a smart investment for the future as new networks emerge over time.

What do I need to know about IP packet loss?

We often hear customers say they have heard about IP packet loss and worry this will lead to loss of audio. While no technology is perfect, there are numerous ways to mitigate packet loss over IP networks to ensure reliable audio streaming. Tieline has been working with IP technologies for over a decade and can deliver consistently high quality audio over imperfect IP networks.

Although you don't need to be an expert in IT to broadcast over IP, it's important to understand what can affect the quality of an IP connection when broadcasting. The public internet is not a perfect network and during the course of a broadcast it is not uncommon to experience:

- Variable Packet arrival times – commonly referred to as latency or jitter.
- Changes in bandwidth availability – often referred to as congestion or contention.

As an IP audio pioneer, Tieline was the first major codec manufacturer to integrate features such as Automated Jitter Buffer management, Forward Error Correction (FEC) and error concealment techniques in all IP codecs. Some manufacturers have only deployed similar strategies recently. These features dynamically respond to variable conditions over unmanaged IP networks like the public internet to ensure reliable streaming.

More recently, Tieline's SmartStream PLUS revolutionised IP broadcasting by delivering the rock solid and reliable STL-grade audio quality you would expect over a T1/E1 link by using inexpensive unmanaged IP networks like the internet for STLs, audio distribution and remotes.

Latency and Jitter Management

There is always some inherent latency in IP networks due to the fact data packets take time to travel between encoding and decoding codecs. However, with appropriate strategies packet arrival times can be minimized to deliver very low latency and high reliability – even over the public internet!

Tieline was the first professional codec manufacturer to introduce automated jitter buffer management over 10 years ago. A jitter buffer is a temporary storage buffer in codec software which captures incoming data packets to ensure the continuity of audio streams is maintained. Data packets travel independently and arrival times can vary greatly depending on network congestion and the type of network used, e.g. LAN versus wireless networks.

Tieline's automated jitter buffer solution dynamically analyzes network 'health' and packet delivery times over a connection and then automatically adjusts settings to minimize delay and maximize reliability. This makes the job of managing network contention much simpler. You can tailor the setting to minimise delay, which is necessary for live connections requiring bidirectional communications. The selection of a low delay algorithm also assists in minimizing latency.

Forward Error Correction (FEC)

Forward Error Correction (FEC) increases the stability of UDP/IP connections. FEC works by sending a secondary stream of audio packets, so that if your primary audio packets are lost or corrupted, then packets from the secondary stream can be inserted into the primary stream.

The amount of FEC required depends on the number of data packets lost over a network connection. It is only recommended over networks where bandwidth congestion is not an issue. A high quality broadcast codec should provide statistics of how many packets are being lost over the network. This allows you to gauge the amount of FEC that you require to maximise connection quality and stability.

> " It may surprise some people, but thousands of Tieline codec owners have been using IP successfully over the public internet for over a decade. "
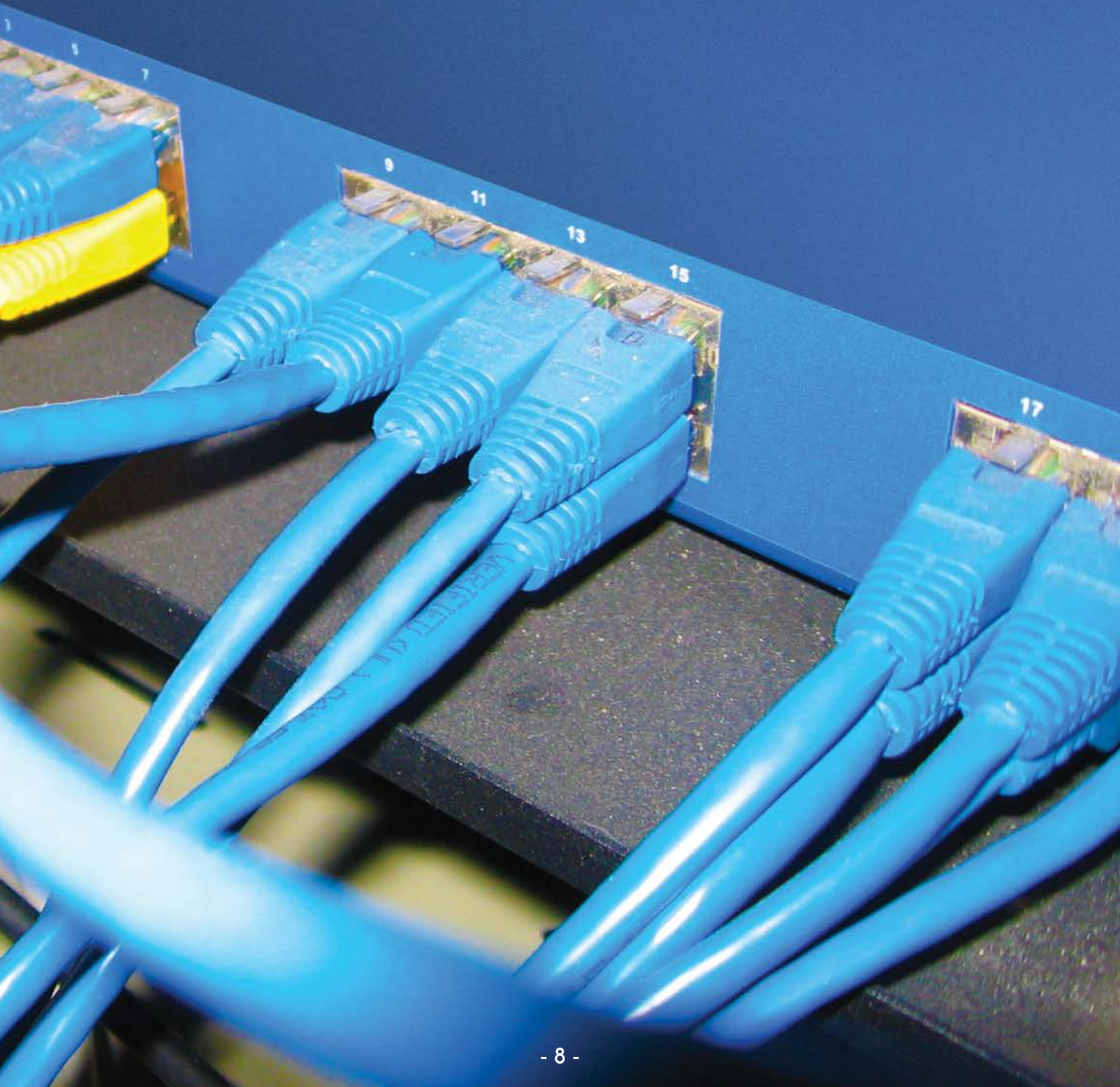
For example, if you are losing one packet in every five that is sent, and you have a FEC setting of 20%, the lost packets will be replaced by FEC to maintain the quality of the connection. If you are losing more packets than this, say one in three, it will be necessary to increase the FEC setting to 33% to compensate.

Error Concealment

Packet loss concealment can also be used to mask the effects of lost or discarded packets during an IP broadcast. This may include:

- Reproduction of the packet received prior to the lost packet.
- Estimation of the value of each dropped packet by interpolation and insertion of artificially generated packets into the bit-stream.

These methods are useful in disguising a few dropped packets here and there, but if several packets are lost in a row audio quality will become noticeably impaired.

SmartStream PLUS: Dual Redundant IP Streaming

Tieline introduced SmartStream PLUS dual redundant IP streaming into its latest IP codecs. Why? To ensure that two totally independent streams of IP packets are sent to a destination codec. This allows you to maintain perfect audio, even if you lose an entire stream of audio packets.

How does it work? SmartStream PLUS technology lets you connect two inexpensive IP links to a codec and stream simultaneous redundant data streams from Ethernet ports or VLANs. If an audio stream is lost for some reason, the codec delivers seamless redundancy by switching back and forth, without loss of audio, from the nominated primary data link to the backup link if one fails and then subsequently recovers. Tieline recommends using IP links from two different Internet Service Providers to deliver optimal redundancy over mission critical STL connections.
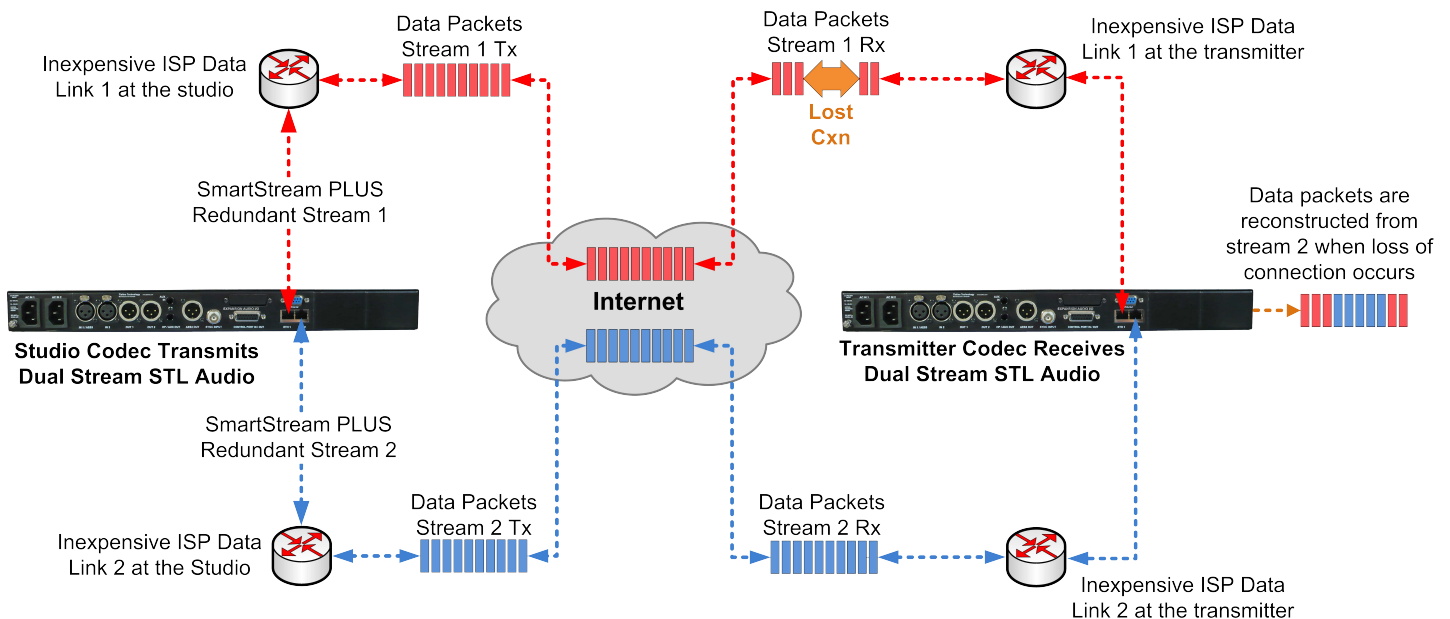
SmartStream PLUS: How it works.

Automated Backup Connections

Tieline IP codecs provide peace of mind by offering fully automated backup connection options. Tieline supports connecting simultaneously via two separate network interfaces and using automated fail over to a backup IP connection or alternative transport.

For example, you can configure a primary IP connection over one ISP for transmitting program audio. Then configure a backup connection via a second ISP to send backup program audio. ISDN, POTS and even cellular broadband IP can also be configured as backup connections.

Tieline codecs can also automatically fail over to backup file audio if required.

IP Network Monitoring

Tieline's Toolbox web-browser Graphical User Interface can be used to configure individual codec settings over a LAN, WAN or the internet. Tieline G5 codecs also support Simple Network Management Protocol (SNMP) for managing devices on IP networks.

Managing the Transition from ISDN to IP with Tieline

Unless you are starting a new station or network, it's likely that the transition to IP will occur gradually over time. Expensive network equipment is rarely replaced totally in one fell swoop, which means it is likely that many engineers will require flexible solutions, which offer connectivity over IP, as well as ISDN, for the foreseeable future.

The Tieline Genie and Merlin families of codecs deliver flexibility by allowing you to connect over IP, ISDN and POTS networks as required. This means you can upgrade your network over time.

Simply purchase optional ISDN G5 or POTS G5 modules, insert them in the codec and you are ready to connect over ISDN, POTS or IP on demand.

If you already own a Tieline codec, upgrading it for ISDN use is a simple process. Simply purchase an ISDN module, plug it into the codec and you are ready to connect. Tieline ISDN enabled codecs provide ultra-reliable, high-speed digital leased line connections that can connect over either 'U' or 'S/T' ISDN interfaces.
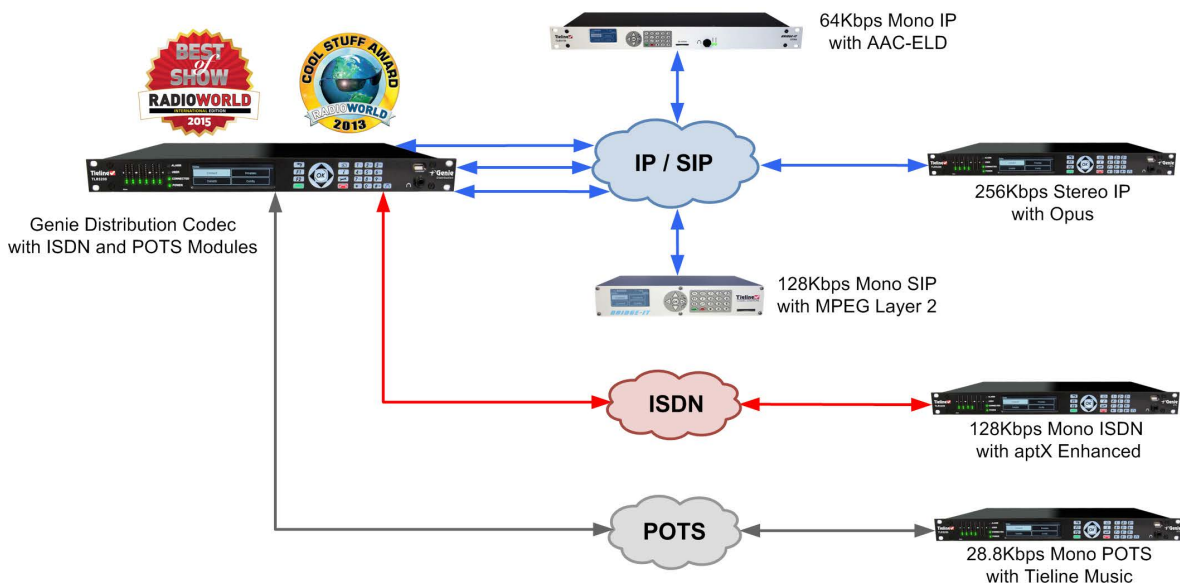
Save with IP

Like many other engineers, you will probably discover that the cost of upgrading your equipment to IP will actually save you money in the short to medium term. How can that be? Well as long as you use a dedicated DSL, fiber or cable broadband link for your IP connection, you can save yourself the cost of expensive ISDN, satellite and MPLS network access. This can save hundreds or even thousands of dollars every month. At that rate it won't take you long to pay for a couple of codecs!

Tieline IP and ISDN Interoperability

Tieline codecs are the only ones to support compatibility with other brands of codecs over five different network types – IP, cellular wireless IP, ISDN, POTS and X.21.

Over IP Tieline is compatible with all major brands of codecs that have implemented EBU N/ACIP Tech 3326 compatible specifications for interoperability over IP using SIP.

Using ISDN modules, the Tieline Genie and Merlin families of codecs, and G3 codecs, are also compatible over ISDN with all major

Genie Distribution Codec
with ISDN and POTS Modules

64Kbps Mono IP
with AAC-ELD

IP / SIP

256Kbps Stereo IP
with Opus

128Kbps Mono SIP
with MPEG Layer 2

ISDN

128Kbps Mono ISDN
with aptX Enhanced

POTS

28.8Kbps Mono POTS
with Tieline Music

codec brands using the popular G.711, G.722 and MPEG Layer 2 algorithms. Genie and Merlin codecs can also connect to Mayah, Prodys and APT Worldcast codecs over ISDN using aptX® Enhanced encoding.

All Tieline POTS enabled codecs can connect with Comrex Matrix, Access, Vector and Blue codecs.

Incredibly, Tieline codecs can connect 6 simultaneous connections over any combination of IP, SIP, ISDN and POTS using different algorithms, bit rates and codecs!

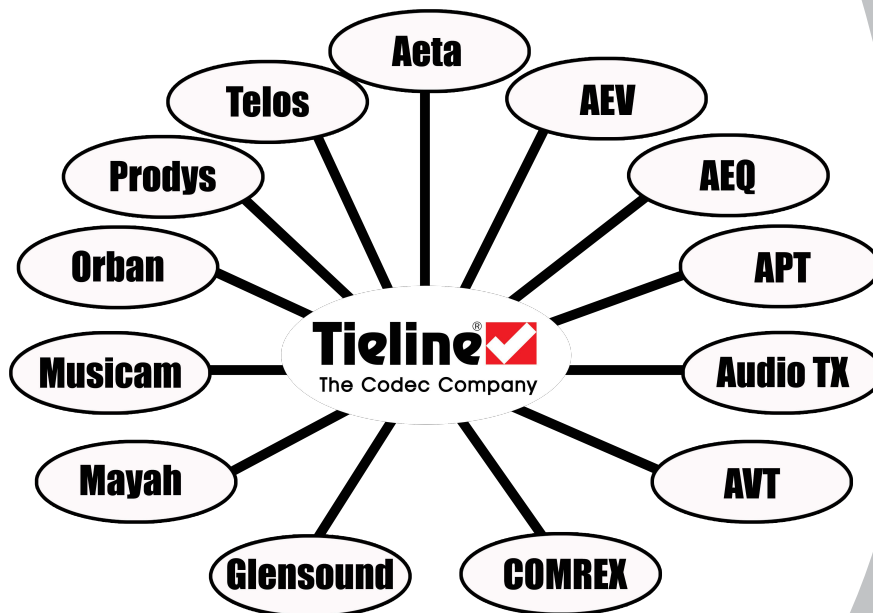## Compatibility with all Major Codec Brands over ISDN

With Tieline ISDN you can connect to AEQ, AETA, APT, AudioTX, Comrex, Glensound, Mayah, Orban, Prodys, and Telos codecs - to name just a few.

Higher Bit-rates = Higher Quality Connections

Each Tieline ISDN plug-in module is capable of connecting using two ISDN B channels and you can use two modules to bond up to 4 x 64kbps B channels.

Tieline ISDN delivers up to 20kHz quality FM audio when connecting to all major codec brands. Connect using the popular aptX Enhanced, MPEG Layer 2, G.722 and G.711 algorithms.

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Alternatively, you can connect between two Tieline codecs using Tieline MusicPLUS and achieve 20kHz stereo at 128kbps, with only 20ms of encoding delay. Use the Tieline Music algorithm to achieve 15kHz stereo over a single 64kbps ISDN B channel, with a low 20ms encode delay. Low delay aptX Enhanced coding is supported at up to 256kbps over ISDN.

Checklist for IP Broadcasting

The following 10 tips will help to obtain the best possible IP connection between two codecs, without paying for Quality of Service (QoS).

1. Always use the best quality Internet Service Provider (ISP). Tier 1 service providers are best as their infrastructure actually makes up the internet 'backbone'.

2. You will get the best quality connection if both the local (studio) and remote codecs use the same ISP. This can substantially increase reliability, audio bandwidth and reduce audio delay. Using the same service provider nationally can give better results than using different local service providers. This is especially true if one of the service providers is a cheap, low-end domestic service provider, which buys its bandwidth from other ISPs. Second and third tier providers sublease bandwidth from first tier providers and can result in connection reliability issues due to multiple switch hops. We also highly recommend using First Tier ISPs if connecting two codecs in different countries.

3. Sign up for a business plan that provides better performance than domestic or residential plans. Business plans typically have a fixed data limit per month with an additional cost for data beyond that limit. In addition, Service Level Agreements (SLA) will often provide better support and response times in the event of a connection failure. If you choose to purchase a domestic plan, check the data limitations and be aware that these plans are often speed-limited or "shaped" when usage exceeds a predefined limit.

4. Ensure that the speed of the connection for both codecs is adequate for the job. Generally, the minimum upload speed recommended is 256 kbps for a studio codec and 64 kbps for a field unit connection.

5. Use good quality equipment to connect your codecs to the internet. (Tieline successfully uses Cisco® switching and routing equipment.):

   • If you are using a DSL or ADSL connection make sure you purchase a high quality modem that can easily meet your speed requirements. This is especially important if you are over 4 kms from an exchange.

   • If you have multiple codecs connected to a local area network (LAN) please ensure that your network infrastructure is designed for media streaming and not domestic usage. Tieline has tested several cheap 8-port switches that lose more packets between local computers than an international IP connection between Australia and the USA!

   • If using a wireless connection ensure that the antenna signal strength reception is strong. The type of antenna used and the amount of output gain also affects connection quality.

6. Once your internet connection is installed at the studio check that the connection performance corresponds with what you ordered and are paying for. A connection can perform below advertised bit rates if:

   • There is an error in ISP configuration.

   • There is an error in modem configuration.

   • There is a poor quality line between the studio and the exchange.

   • There are too many phones or faxes connected to the phone line.

   • Line filters have been connected incorrectly.

7. Use a dedicated DSL/ADSL line for your codecs. Do not share a link with PCs or company networks. The only exception to this rule is if an organization has network equipment and engineers that can implement and manage quality of service (QoS) on its network.

8. Use UDP as the preferred audio transport protocol.

9. When using UDP ensure the total bit rate (audio bit rate plus header bit rate) is no more than 80% of the ISP connection rate

10. Wireless IP connections can easily become congested and result in packet loss and audio drop-outs. It is very difficult to guarantee connection quality when there is no way of knowing how many people are sharing the same wireless connection, particularly at large sporting events etc.

I calculated that in a single year we transmitted more than 100,000 hours of continuous live broadcasts throughout the world using a wide range of Tieline codecs. It is important that the codecs are flexible enough to allow us to connect over POTS, ISDN, IP, wireless 3G and 4G-LTE, BGAN Inmarsat and KA-SAT Tooway.

Marco Cech, AF Manager for the World Family of Radio Maria

Selecting a Data Plan

For an STL over IP we recommend ordering a business data plan over fiber, cable or DSL for both the studio and transmitter site. The cost of these plans has decreased substantially and they are now cheaper than other forms of leased line networks.

We recommend using a dedicated broadband connection which is not shared, so that your connection will never be compromised by other people downloading over the same Internet connection. Sharing a connection quite often will result in packet loss and therefore loss of audio.

Business SHDSL plans send guaranteed symmetrical data (e.g. 512 kbps downlink and 512 kbps uplink) as opposed to ADSL connections, which send asymmetrical data (e.g. 512 kbps downlink and 256 kbps uplink). Symmetrical data is preferred in most IP broadcast situations because you are more likely to achieve higher uplink speeds than with ADSL connections. Higher bandwidths increase the likelihood of stability and high quality for your connections.

Data Plan Costs

IP network data costs vary depending on the network you are connecting to and the number of channels you need to broadcast. However, in general IP networks are much cheaper to operate than synchronous data networks like ISDN. There are a wide range of IP networks to choose from when broadcasting over IP and some of the factors that affect the selection of a network include:

- Your program content: Are you performing a simple remote broadcast or distributing high bandwidth audio around a network, e.g. STL or audio distribution.

- The number of audio channels you are sending: Do you need a simple point-to-point IP audio connection, are you multicasting, or do you need to send multi-unicast IP audio streams to different studios?

- Your broadcasting region: Depending on where you are situated, you may have access to different infrastructure like DSL/ADSL, cable or fiber; similarly, you may have access to different types of wireless networks.

- Your budget: A community radio station may be looking for a cost effective hardware and data solution, whereas a large network may be looking to integrate flexible and high quality hardware with innovative software management solutions.

Calculating Data Requirements and Costs

To calculate your total IP data requirements you need to:

- Determine how many channels you are sending: is your connection mono, stereo, multicast or multi-unicast?

- Calculate the bit rate requirement per channel; this will depend on the compression algorithm you select and will need to include packet overhead data requirements.

As a general rule of thumb, ensure the total bit-rate (audio bit-rate plus header bit-rate) is no more than 50% of the ISP connection rate. IP headers generally require around 20 kbps in addition to the audio bit rate. For example, with a 64 kbps connection the audio bit rate should be (64-20) x 0.8 = 31.2 kbps or lower.

Once you have calculated the total connection bit-rate and how high the ISP connection bit-rate needs to be, you can shop around for the most suitable and competitive data plan to suit your needs.

Contact Tieline to receive a spreadsheet that will tell you how much data your codec will consume per hour of broadcasting to assist with plan purchases.

" Tieline was the first professional codec manufacturer to add the Opus algorithm to its range of codecs. **"**

Which Algorithms are best over IP?

Tieline codecs offer a huge range of algorithm options for IP broadcasting. Your choice of encoder will depend upon the bandwidth of your internet connection and the type of codec you are connecting to, as well as the task you are performing.

Tieline was the first professional codec manufacturer to add the Opus algorithm to its range of codecs. When other manufacturers were not quite sure about the advantages of Opus, Tieline led from the front and since then interest in Opus encoding has exploded.

Opus Encoding

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.

Tieline Algorithms

Tieline also offers its very own Music and Music PLUS algorithms, which are optimised to deliver high quality audio at low bit rates.

The following table lists and summarises the algorithm options in Tieline G5 codecs.

Algorithm	Very Low delay	Moderate to high delay	Excellent Performance at Low Bit rates	Preferred for Live Remotes	Preferred for STLs and Audio Distribution	High Compatible with other Codecs
Linear/PCM	X				X	X
Opus	X		X	X	X	X
Tieline Music	X		X	X		
Tieline Music PLUS	X		X	X	X	
aptX Enhanced	X				X	X
LC-AAC		X			X	X
HE-AACv1		X			X	
HE-AACv2		X	X	x*		
AAC-LD	X			X	X	
AAC-ELD	X		X	X		
MPEG Layer 2	X				X	X
MPEG Layer 3		X				X
G.722	X					X
G.711	X					X

* Use with caution for remotes due to high delay; not suitable when bidirectional communications is required.

The Future of IP Broadcasting

The power of IP broadcasting is finally being realised. Engineers and programmers love the flexibility of IP codecs and there is no doubt they have the potential to open new revenue streams and deliver cost savings. Cellular wireless technologies still present challenges in some situations, however with networks improving all the time, the future looks bright.

The sky really is the limit in terms of IP codec broadcast opportunities and the radio industry can expect to see on-going innovation over time, as the quality of wired and cellular IP connections increases and codecs become more feature-rich.

Tieline Technology is the leading manufacturer of digital broadcast audio codecs that can connect over IP, 4G, POTS and ISDN networks. They have offices in Australia & the United States and a global distribution network spanning the Americas, Europe, U.K, Africa, Asia, Middle East and Australasia.

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