Intraplex® IP Link 200

Dual Bidirectional IP Audio Codec

Made in USA

The Intraplex® IP Link family of IP audio codecs provides high-end features at an affordable price

Offering an array of audio coding options along with IPConnect technology for data tunneling, the IP Link codecs are suitable for use in Studio to Transmitter Links (STLs) as well as audio contribution and distribution networks. The IP Link 200 is ideal for STL applications requiring two separate stereo channels, and its support for IP multicast and multiple unicast streams enables one encoder to feed multiple decoders.

By incorporating three IP Interfaces that can be used for streaming and management, the IP Link systems can provide a level of reliability not seen in comparably priced codecs.

As the latest additions to the Intraplex line of data transport products, the IP Link family of audio codecs bring legendary Intraplex reliability to the IP codec market.

Product Features

- Two bidirectional stereo audio channels
- Standard: Linear, AAC-LC, Opus and G.722 audio coding
- Optional: AAC-HE, AAC-HEv2, AAC-ELD, MPEG2, MPEG3 and Enhanced aptX audio coding
- Optional: Automatic audio loudness leveling and metering compliant with EBU R-128 and ITU-R
- Optional: IPConnect capability to reliably transport external IP packets
- Other transport modes: Transparent AES up to 192 ksps to support composite FM multiplex signal over AES
- Protocol Encapsulation: RTP, SHOUTcast/Icecast, MPEG-TS
- Three independent IP interfaces for redundant network operation
- Optional redundant power supply: 12VDC or 48VDC
- Built-in silence sensor with optional stream switch over
- Automatic backup to audio playout from USB drive or external audio source
- Multicoding can encode the same audio source in multiple formats for STL, backup, and web streaming
- Optional Dynamic Stream Splicing providing “hitless” operation and T1/E1 circuit like performance on less predictable IP networks
- Prioritized stream sources at the output with automatic switch over and switch back between primary and secondary streams and backup sources (streams, USB, external audio source)
- Programmable RTP level Forward Error Correction (FEC) scheme
- Programmable time diversity and interleaving of streams to combat burst packet losses
- Integrated with Intraplex IP Link Scheduler for automated scheduled program switching
- Integrated with Intraplex LiveLook (network analytics and monitoring software)
- N+1 redundancy with integrated control of external switching equipment
- SynchroCast™ option provides dynamically managed precision delay for Single Frequency Network (SFN) broadcasting and simulcasting
- Support for IP multicast and multi-unicast
- Web browser user interface and SNMP network management
- Eight multipurpose contact closure inputs and outputs provide:
  - Transport of logic signals with time- alignment to audio
  - Stream control
  - Alarm notification

Product Details

The Intraplex IP Link 200 audio codec was designed to provide unprecedented level of reliability from ground up. At the hardware level, the N+1 redundancy with built-in control for data switches provides automatic synchronization of configuration and switch-over capability. This reliability is further enhanced with an optional hot-standby power supply.

At the streaming layer, Dynamic Stream Splicing provides a set of networking tools for reliability, such as redundant streams with network and time diversity. The support of Forward Error Correction (FEC) and interleaving further enhances these capabilities. IP Links can be intelligently combined to achieve reliability generally associated with T1/E1 circuit over less robust IP networks. Hitless operation can be achieved when multiple networks are available. The use of time diversity on redundant streams along with FEC and interleaving can provide protection against burst packet losses.

The IP Link 200 also offers Multicoding, the ability to simultaneously encode the same audio program using multiple different algorithms. Multicoding can, for example, allow the user to send linear uncompressed audio on a main STL, while sending the same program with AAC coding on a lower-bandwidth backup link and MP3 to feed a streaming Web server such as SHOUTcast.

An optional built-in Audio Loudness Leveling capability ensures that the loudness of incoming audio is kept at a consistent level based on the EBU R128 and ITU-R standard.

A built-in silence sensor and alarm enable IP Link codecs to offer a variety of automatic backup options. If the main link is lost, the IP Link 200 can switch to a secondary feed from a lower bandwidth link. In the event of total IP connectivity loss, the system can switch to playout from a plug-in USB drive or from any local audio source connected to the audio inputs on the rear panel. A comprehensive Web browser interface makes the IP Link codecs easy to monitor, configure, and operate.

IPConnect capability enables transport and tunneling of external IP packets protected with Dynamic Stream Splicing and FEC.

The IP Link 200 provides optional SynchroCast capability to dynamically align the playback of audio at geographically dispersed transmitter sites for SFN broadcasting. SynchroCast can be used with compressed or linear audio formats.

The IP Link 200's front panel has VU meters for each channel, and a convenient front-panel user interface to access key configuration settings and status information.

Specifications

Specifications and designs are subject to change without notice

Overview	
Channels	Two stereo (or four mono) program channels, encode and decode
Front Display	Graphical front panel user interface - 3.2 inch display; 256 x 64 pixel, white monochrome OLED; six-button keypad; VU meters for each audio channel
Audio Coding	Standard: Linear, AAC-LC, Opus, G.722 Optional: MPEG2, MPEG3, AAC-HE, AAC-HEv2, AAC-ELD, Enhanced aptX and transparent AES
Audio Loudness Leveling	Optional: Leveling, metering with true-peak measurement and brick-wall limiter applied to the input analog and digital audio in conformance with ITU-R BX1770-3 and EBU R 128
Streaming	EBU N/ACIP Tech 3326, SHOUTcast/Icecast, MPEG Transport Stream over IP, Transparent AES, SynchroCast
SynchroCast	Optional: Audio delay programmable up to 1 second with 1 microsecond accuracy
Multicoding	Allows the input to be encoded and streamed out using multiple different algorithms simultaneously
Digital/Analog Operation	Dual domain: AES/EBU and analog For audio input, AES/EBU / analog is auto-detected For audio output, AES/EBU and analog are simultaneous
Webcasting	Can provide a TCP stream to a SHOUTcast or other Webcasting server
Backup	Configurable for automatic backup to secondary incoming audio stream, playout of audio from USB drive, or playout of audio from a local device connected to the rear panel inputs
Aux Data Channel	RS-232, in- or out-of-band data transport programmable to 2400, 4800 and 9600, and 19200 bps with time-alignment to audio streaming
Contact Closures	Eight input and eight output opto-isolated contact closures, with time-alignment to audio streaming Contact inputs can transport state to peer or control stream state Contact outputs can receive state from peer or be tied to system alarms
Hardware Redundancy	N+1 with integrated support of external switching equipment

Connectors	Rear panel	XLR for channel 1 analog L&R and digital AES/EBU inputs and outputs, RJ-45 connectors with StudioHub cabling format for channel 2 audio inputs/outputs Ethernet: Three 10/100 Base-T, RJ-45 RS-232 data: D-sub, 9 pin male Contact Closures: D-sub, 26-pin female USB: Type A DC Power: Two pin screw terminal AC Power: C14 power inlet
	Front panel	Ethernet: One 10/100 Base-T, RJ-45 Audio Headphone: One ¼" stereo headphone jack
GPS	External GPS: 10 MHz and 1 PPS BNC connectors Optional: GPS receiver plug-in board kit with SMA connector for external GPS antenna (provided with kit)	
Digital Audio		
Accepted Audio Sampling Rates	Accepts AES/EBU sample rates between 32 and 192 ksps	
Sample Rate Conversion	Automatic rate conversion at input with dynamic range of 128 dB	
Digital Gain	AES/EBU output has micro adjustable gain between +6 and -6 dB	
AES Transparent Transport		
Sample Rate	32, 44.1, 48, and 192 ksps	
Analog Audio		
Input Impedance	Balanced, greater than 10 k Ohms	
Output Impedance	Balanced, less than 52 Ohms	
Audio Frequency Response	48 ksps: 10 Hz to 22 kHz 44.1 ksps: 10 Hz to 20.5 kHz 32 ksps: 10 Hz to 15 kHz	
Audio Level	Full scale analog audio input/output: 9 to 24 dBu, user-settable in 1 dB steps	
Total Distortion	(THD+N) Less than 0.003% at 1 kHz, -1 dBFS input	
Dynamic Range	Greater than 91 dB	
Sample Size	16 or 24 bit	
Ethernet		
Ethernet Data Rate	10/100Base-T (10 or 100 Mbps) full duplex, auto-negotiation	
Network Connections	Two WAN ports plus management port. Mirror port on front panel Per port 802.1 pq configuration; Three network ports available for both streaming and management	
Network Protocols	IPV4, TCP, UDP, RTP, RTCP, SIP, HTTP, FTP Telnet, NTP, SNMPv2C, ARP, ICMP, Ultravox (v1, v2)	
Remote Management	Web browser interface SNMP	
Streaming		
RTP Streams	Total of 12 streams Setup: Static or SIP Unicast, multi-unicast, multicast Standard RFC payload formats, auto configuration Source IP address and UDP port verification at the receiver for security	
TCP Streams	Total 12 SHOUTcast/Icecast transmit or receive streams with selectable codec and coding rate for each stream	
IPConnect	Enables transport of external IP packets as payload of IP Link RTP streams	
SIP	Compliant with EBU N/ACIP Tech 3326 Works in peer to peer and proxy mode NAT traversal support	
Redundancy	Automatic failover mode between Primary, Secondary and Backup streams	

Backup Audio Source	USB Playlist, Local input channel, Other output channel
Dynamic Stream Splicing	Optional: Enables multiple identical audio streams to be sent across the IP network (or two separate IP paths, if available) and provides for hitless switching at the decoder
Jitter Buffer	Programmable jitter buffer depth up to 1024 packets. Static or automatic jitter buffer adjustment
Forward Error Correction	Multiple FEC schemes configured per stream with 25%, 50%, 66% and 100% overhead selection
Time Diversity	Time delay configured on per stream basis, used with redundant streams for burst packet loss protection
Interleaving	Configured per stream for mitigation of consecutive packet losses
Diagnostics	
Test Tone Generator	1 kHz test tone at -12 dBFS
Loopbacks	Input to output channel equipment loopback while simultaneously sending streams from the input channel
Network Performance Statistics Tracked	Burst packet loss statistics based on RFC 3611 Per stream and group statistics for packets received, packet lost, packets recovered by FEC and packets sent Send and receive stream bandwidth
Status Indicators	
LED Indicators	Stream activity and status Multi-LED bar graph audio level meters for channel 1 and channel 2 input and output
Alarms	
Alarm Reporting	Major/minor alarms, normally open relay contacts, SNMP traps Maintains internal and syslog messages alarm log Log files can be sent off to off-site server for storage User configurable per-stream packet loss threshold
Loss-of-Audio-Alarm	Built-in silence detection with ability to provide alarm and perform switch over of stream on loss of audio
Mechanical and Environmental	
Dimensions (H X W X D)	1RU: 1.75 x 19 x 10.1 in. (4.45 x 48.3 x 25.7 cm); EIA rack mountable
Weight	5 lbs (2.27 kg) typical
Power Supply	Main: AC 100-240 VAC, 50/60 Hz with type T2A 250 V AC input fuse Backup: Optional external module, AC to 12 VDC converter or internal module for -48 VDC
Power Consumption	15 watts
Humidity	10% to 90% non-condensing
Operating Temperature	32º to 122º F (0º to 50º C)
Compliance	
Regulatory Compliance	CE, FCC Part 15 Class A, IEC 60950, RoHS