Intraplex® IP Link 100c

Compact, portable audio over IP STL codec

Made in USA

A member of the award-winning Intraplex® IP Link audio codec family, the IP Link 100c is a powerful compact AoIP model, designed to provide state-of-the-art capabilities for remote contribution, studio-to-transmitter, and studio-to-studio applications at an affordable price.

With a full-duplex AES3, Analog, AES67 input and output along with 3 GigE network ports, the IP Link 100c packs full-featured IP audio codec capabilities into a compact portable half-rack unit. The optional Mic level input further adds to the capabilities for reliable remote broadcast capabilities. Interoperable with other IP Link codecs and Ascent, it is also compatible with industry- standard AoIP formats, including support of FM-MPX signal transport. With the support of payload privacy using built-in 128/256 key encryption via Secure Reliable Transport (SRT), the IP Link 100c is extremely secure and flexible. Stream splicing and automatic multi-source audio switching, including USB playlist as backup, further ensures reliability with constant and successful signal transport for your station.

Product Features

- Full-duplex, single stereo channel with selectable AES3, Analog and AES67 input and output
- Support for Line and Mic level input
- AoIP formats include Linear, Compressed, AES67, and Icecast
- Standard audio coding: Linear; Opus
- Optional audio coding: AAC-LC; AAC-HE; AAC-HEv2; AAC-ELD; AES192; MPEG2; MPEG3; Icecast; Shoutcast
- Protocol Encapsulation: RTP; Icecast (requires optional audio coding pack); SRT
- Three independent IP interfaces for redundant network operation
- Built-in silence sensor
- Automatic backup to audio playout from USB drive
- Multicoding allows the input Line or Mic level signal to be sent to multiple destinations with different encoding formats and protocols
- Prioritized stream sources at the decoder with automatic switch over and switch back between primary, secondary, and backup sources
- Programmable RTP-level Forward Error Correction (FEC) scheme
- Integrated with Intraplex LiveLook (network analytics and monitoring software)
- Support for IP multicast and multi-unicast
- Web browser user interface and SNMP network management
- Four multipurpose contact closure inputs and outputs provide:
  - Transport of logic signals with time-alignment to audio
  - Alarm notification
- Additional options:
  - µMPX transport a full FM composite MPX signal, including pilot and RDS, with perfect peak control with bitrate of 320 kbit/s. Compatible with Intraplex SynchroCast.
  - 10-band high-precision audio processing
  - Digital FM-MPX format support with compatibility with IP Link MPXp
  - Secure Reliable Transport (SRT) with 128/256-bit encryption and automatic packet re-transmission
  - Dynamic Stream Splicing for both RTP and SRT formats, providing “hitless” operation over diverse network paths
  - SynchroCast™ provides dynamically managed precision delay for Single Frequency Network (SFN) broadcasting and simulcasting
  - Internal plug-in module supports two microphone level inputs

Product Details

The IP Link 100c is a full-duplex, single stereo-channel codec for simultaneous reception and transmission of AoIP streams. The codec is designed to be compatible with the IP Link codec family and Ascent, including the IP Link MPXp codec for FM-MPX signal transport.

The IP Link 100c is designed to provide an unprecedented level of full-featured reliability in an affordable compact form factor.

At the streaming layer, the combination of SRT and Dynamic Stream Splicing (DSS) provides a set of networking tools for signal reliability even over challenging IP connections. The SRT transport protocol can simultaneously re-transmit lost packets in real-time while encrypting the full payload. When using the traditional RTP transport format, the packet loss protection is provided using Forward Error Correction (FEC) and time diversity of packets. The optional Dynamic Stream Splicing (DSS) provides “hitless” protection against packet or link losses using diverse network paths for both SRT and RTP transport formats.

Product Details (continued)

With full-duplex capability, the codec enables simultaneous operation of multiple transmit streams and receive streams for reliability. Using Multicoding capability, the transmit streams can be sent to multiple destinations with different formats and encoding. For instance, feeding multiple transmitter sites with different network bandwidths or feeding an Icecast server simultaneously, while sending an RTP or SRT streams to remote studio or transmitter locations.

In addition to the various built-in "hitless" packet loss protection techniques (e.g. FEC, re-transmission and DSS), the decoder also provides for three prioritized sources for switching. The source switching protects against failure of either an encoder or the main network connection. The codec lets the user define Primary, Secondary and Backup sources of streams or local USB audio.

An example configuration:

- Primary source stream: DSS protected from the studio encoder
- Secondary source: an Icecast stream
- Backup source: a local USB file

The switching between these sources can be configured to be fully automatic or user initiated.

For control and status, the IP Link 100c provides an intuitive web interface and comprehensive SNMP interface.

Specifications

Specifications and designs are subject to change without notice

Overview	
Channels	One full-duplex stereo (or two mono) program audio channel or one MPX channel, encode and decode
Audio Coding	■ Standard: Linear Uncompressed, Opus, AES67 ■ Optional Package 1: MPEG2, MPEG3, AAC-LC, AAC-HE, AAC-HEv2, AAC-ELD, AES192 ■ Optional µMPX: MicroMPX Encoder & MicroMPX Decoder (sold separately). Transport a full FM composite MPX signal, including pilot and RDS, with perfect peak control with bitrate of 320 kbit/s. Compatible with Intraplex Syncrocast feature.
FM MPX via AES192 (BB192)	Support for transport of FM MPX via AES192 (BB192) composite signal. Sampling rates and sample size compatible with IP Link MPXp.
Streaming Format	RTP (EBU N/ACIP Tech 3326), SRT, Icecast (requires optional audio coding algorithms)
SynchroCast	Optional: Audio delay programmable up to 2.5 seconds with 1 microsecond accuracy
Multicoding	Allows the input to be encoded and streamed out using multiple different algorithms simultaneously
Webcasting	Can receive and transmit Icecast streams (requires optional audio coding algorithms)
Backup	■ Configurable for automatic backup to secondary incoming audio stream ■ Playout of audio from USB drive
Aux Data Channel	RS-232 data transport programmable to 2400, 4800 & 9600, and 19200 bps with time-alignment to audio streaming
Contact Closures	■ Four input and four output opto-isolated contact closures, with time-alignment to audio streaming ■ Contact inputs can transport state to peer within the stream packet ■ Contact outputs can receive state from peer or be linked to system alarms
Connectors	■ Ethernet: Three 10/100/1000 Base-T, RJ-45 connector ■ XLR: Four for analog L&R audio and AES/EBU inputs and outputs. User selectable for either analog or AES/EBU ■ RS-232: One full-duplex port, RJ-45 connector ■ Contact Closures: RJ-45 connector ■ USB: One type A connector ■ Audio Headphone: One ¼" stereo headphone jack
Digital Audio	
Accepted Audio Sampling Rates	Accepts AES/EBU sample rates between 32 and 192 ksps to support both discreet (L&R) audio and AES192 (BB192) signals
Sample Rate Conversion	Automatic sample rate conversion at input with a THD of 120 dB
Digital Gain	AES/EBU output has micro adjustable gain between +6 and -6 dB
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
Microphone	Optional internal plug-in module supports up to two microphone level inputs from -26 to -50 dBu, user-settable in 1 dB steps. Module supports user-selectable 48 VDC microphone power
FM MPX using AES192 (BB192)	
Interoperation	Interoperation with Intraplex IP Link MPXp
Sample Rate	132, 162, or 192 ksps
Sample Size	12, 14, 16, 20, or 24 bit
Ethernet	
Ethernet Data Rate	10/100/1000 Base-T full duplex, auto-negotiation
Network Connections	■ Two WAN ports plus management port ■ Three network ports all available for both streaming and management
Network Protocols	IPV4, TCP, SRT, UDP, RTP, HTTP, NTP, DNS, DHCP, SNMP, ARP, ICMP, Ultravox (v1, v2) for Icecast
Remote Management	■ Web browser user interface HTTP/HTTPS with multiple levels of user accounts ■ SNMP with SNMPv2C/SNMPv3
Streaming	
RTP / SRT Streams	■ Total of 8 streams with multiple IP destinations for the transmit streams ■ Setup: Static ■ Unicast, multi-unicast, multicast (RTP only) ■ Standard RFC payload formats, auto configuration ■ Source IP address and UDP port verification at the receiver for security ■ Audio plus meta-data format to support GPIO and RS-232 alignment
TCP Streams	Multiple Icecast streams not exceeding the total count of 8
Redundancy	Automatic failover mode between Primary, Secondary and Backup streams
Backup Audio Source	USB playlist
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 512 packets. Static or automatic jitter buffer adjustment
Reliability	Secure Reliable Transport (SRT) for automatic retransmission of lost packets, FEC, time diversity
Security	■ Stream encryption supported in SRT (AES-128/256) ■ Access control with user settable firewall configuration per network interface
Time Diversity	Time delay configured on per stream basis, used with redundant streams for burst packet loss protection
Diagnostics	
Test Tone Generator	1 kHz test tone at -12 dBFS
Loopbacks	Input to output channel equipment loopback while simultaneously sending
Network Performance Statistics Tracked	■ Per stream and group statistics for packets received, packet lost, packets recovered by FEC and packets sent ■ Send and receive stream bandwidth
Network Tools	Ping
Alarms	
Alarm Reporting	■ Major/minor alarms, normally open relay contacts, SNMP traps ■ Maintains internal and syslog messages alarm log
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 8.5 x 6 in. (4.45 x 21.6 x 15.2 cm)
EIA Rack Mountable Weight	2.2 lbs (1 kg) typical
Power Supply	External 100-240 VAC, 50/60 Hz, AC/DC
Power Consumption	10 Watts, typical
Cooling	Convection cooled
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, UL 60950, RoHS