Intraplex® Ascent

Multichannel Audio Over IP Codec

Built on the proven, award-winning Intraplex® IP Link codec technology, GatesAir's Intraplex Ascent platform supports up to 32 audio channels with high-end features.

The scalability of Ascent makes it ideal for applications that require multiple channels of audio encoding and decoding at head-end sites. The high-density solution reduces cost and provides a path for convergence of IT and broadcast infrastructure. The interoperability between Ascent and IP Link codecs provides a complementary solution for remote contribution and distribution use cases.

Product Features

- 4, 8, 16 stereo channels. Each channel can be full-duplex, encode-only or decode-only
- Supports AES3, Analog and AES67 audio input and outputs
- Standard Coding: Linear, Opus
- Optional: AAC-HE, AAC-HEv2, AAC-ELD, AAC-xHE, MPEG 2 and MPEG 3 audio coding
- Protocol Encapsulation: Real-time Transport Protocol (RTP), Secure Reliable Transport (SRT)
- Streams: Multicast, unicast, and multi-unicast
- 10 streams per channel with up to 20 destinations per transmit stream; maximum of 100 streams per system
- Three independent IP interfaces for redundant network operation
- Built-in silence sensor with optional stream switchover
- Automatic backup to audio playout from USB drive or external audio source
- Multicoding: can encode the same audio source in multiple formats
- 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 scheduler for automated scheduled program switching
- Integrated with Intraplex LiveLook (network analytics and monitoring software)
- Web and SNMP for management
- Multiple web account types to restrict access
- GPIOs: up to 32 in, 8 out. Available for stream transport and alarm assignment
- Network reliability
  - Dynamic Stream Splicing with network and time diversity for "hitless" packet loss protection
  - Programmable RTP level Forward Error Correction (FEC)
  - Secure Reliable Transport (SRT): new protocol provides automatic packet retransmission method
- IP Security
  - Access control per interface
  - Encryption of streams with AES-128

Product Details

The Ascent platform continues the Intraplex tradition of providing an unprecedented level of reliability for audio over IP application. Built with enhanced network reliability and security in mind, Ascent supports the Dynamic Stream Splicing technology of IP Link codec, which provides "hitless" protection for packet losses using the combination of time and network diversity for packet transmission, and FEC.

To further enhance reliability and security, the platform supports the SRT protocol encapsulation. SRT (Secure Reliable Transport) is an open-source protocol which provides low-latency, reliable and secure streaming of audio and video data. The reliability in SRT is accomplished by a built-in packet re-transmission scheme; the security of streams is handled by the built-in AES-128/256 payload encryption.

In addition to "hitless" protection, like the IP Link codecs, the Ascent also supports automatic failover between different incoming streams or stored audio.

This combination of capabilities gives Intraplex codecs a market-leading position in STL over IP technology and unmatched network transport reliability.

Ascent supports both physical audio interfaces (AES3, Analog) and AES67 (AoIP), with a maximum of 16 full-duplex or a total of 32 channels (combination of encode and decode). Physical audio interfaces can be ordered as 4- or 8-channel cards, supporting both AES3 and analog audio signals.

The Ascent product is built on the Commercial-Off-The-Shelf (COTS) x86 architecture to leverage the scalability and cost of the technology. The product is available in a 1RU branded hardware server and as a software-only option.

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Specifications

Specifications and designs are subject to change without notice

Overview	
Channels	4,8,16 selectable from combination of AES3, analog, AES67
Audio Coding	Standard: Linear, Opus, Optional: MPEG2, MPEG3, AAC-HE, AAC-HEv2, AAC-ELD, AAC-XHE
Streaming	Standard RTP (EBU N/ACIP Tech 3326), GatesAir RTP (Audio + GPIO + Metadata), SRT. - 20 destinations per transmit stream - Total 100 streams (Transmit & Receive) per system - Selectable codec algorithm and coding rate per stream - Programmable egress interface and time offset per stream - Programmable FEC scheme per stream - Programmable encapsulation selection: RTP or SRT - Programmable Jitter buffer – up to 512 packets per receive stream
Reliability	- Network diversity - Time diversity with programmable offset - FEC - SRT (automatic re-transmission of lost packets)
Security	- Stream: Encrypted SRT (AES-128/256) - Access Control: User settable firewall setting per interface
Multicoding	Allows the input to be encoded and streamed out using multiple different algorithms simultaneously
Backup	Configurable sources for each output channel: Primary, Secondary, Backup. Support for using stored audio files as a backup source
Scheduler	Built-in scheduler to switch output channel sources based on user defined time-table
Contact Closures	Up to 32 In and 8 Out. Contact closure can be used with stream transport and assignment to alarms
Management	Web: HTTP/HTTPS with multiple level of users SNMP: SNMPv2C/SNMPv3

Network Interfaces	3 x 10/100/1000Mbps Ethernet (RJ-45)
Digital Audio	
Sample Rates	Accepts AES/EBU sample rates of 32, 44.1, 48, and 96 kHz. Sample rate conversion on inputs
Input/Output	110 Ohms
Connector	68pin VHDCI (XLR breakout cable available)
Word clock	Dedicated word clock input and output for audio sample rate timing
Analog Audio	
Input Impedance	Balanced, 10 k Ohms
Output Impedance	Balanced, 50 Ohms
Load impedance	600 Ohms or greater
Audio Frequency Response	20Hz to 20kHz +0/-0.1dB 20Hz to 40kHz +0/-3dB
Audio Level	-10 to +24 dBu in 0.5 dBu steps
Total Distortion	THD+N <= -96 dB (0.0015%)
Dynamic Range	>=110 dB
Sample Size	24 bit
Connector	68pin VHDCI (XLR breakout cable available)
Alarms	
Alarm Reporting	Major/minor alarms, normally open relay contacts, SNMP traps Maintains internal and sys-log 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 18.1 in. (4.45 x 48.3 x 46.0 cm)
EIA Rack Mountable Weight	5 lbs (2.27 kg) typical
Power Supply	Main: AC 100-240 VAC, 50/60 Hz
Humidity	10% to 90% non-condensing
Operating Temperature	50º to 122º F (10º to 50º C)