# GATESAIR®

# 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

## Intraplex® Ascent

### 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, <br> 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. <br> - 20 destinations per transmit stream <br> - Total 100 streams (Transmit & Receive) per system <br> - Selectable codec algorithm and coding rate per stream <br> - Programmable egress interface and time offset per stream <br> - Programmable FEC scheme per stream <br> - Programmable encapsulation selection: RTP or SRT <br> - Programmable Jitter buffer – up to 512 packets per receive stream |
| Reliability | - Network diversity <br> - Time diversity with programmable offset <br> - FEC <br> - SRT (automatic re-transmission of lost packets) |
| Security | - Stream: Encrypted SRT (AES-128/256) <br> - 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 <br> 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<br>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<br>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.<br>(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) |