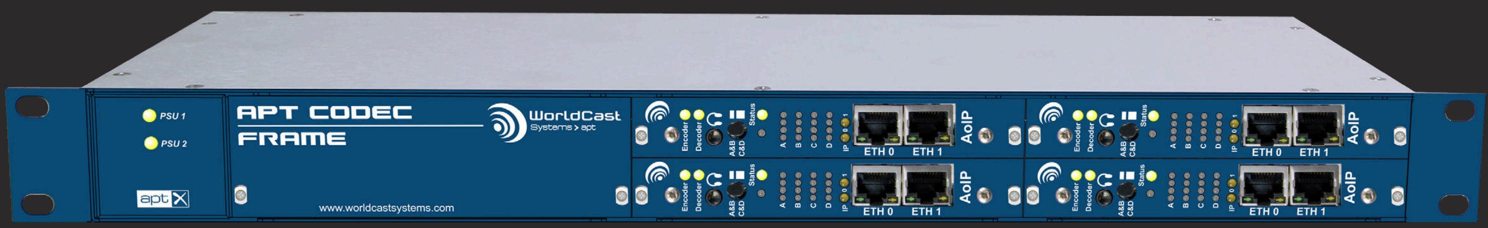
WORLDCAST SYSTEMS

POWERFUL PERFORMANCE CONTROL SAVINGS SERVICES



APT Multi-Channel Codec

Professional IP STLs for Content Delivery

The APT MULTI-CHANNEL CODEC is a compact and efficient solution for the transport of multiple channels of audio content over IP links. It supports up to 8 stereo channels of audio or MPX programs within a single unit of rackspace - and even more IP streams when using multicast or multiple unicast modes.

The modular 1RU frame can accommodate up to 4 AoIP codec modules, each equivalent to a stand-alone codec.

The hardware-based modularity and the redundant power supply assembly reliably exclude a single-point-of-failure.

The AoIP mudule offers the entire range of audio formats and modes meeting the broadcast industry's requirements.

It is equipped with AES/EBU and analog interfaces providing HI/LO or 600Ω impedance selection.

For the transmission of digital composite/MPX the AES192 mode is activated. Analog composite/MPX signals are fed through the alternative I/O interface, equipped with BNC connectors.

With the Dante/AES67 interface module, the 1u chassis becomes a multi-channel gateway codec, connecting your studio LAN to regional or global networks.

The APT modular codec chassis combines both proven and innovative technologies in the backbone of your broadcast network.

APTmpX SURESTREAM SCRIPTEASY

APT MULTI-CHANNEL CODEC Benefits:

Predictable IP Transport

APT AoIP Modules migrate the degree of reliability of an E1/T1 connection into the IP domain. SureStream reliably eliminates packet losses, and latency fluctuations are compensated by the NTP-based Content Time Alignment.

Pristine Audio Quality & Performance

Highest signal fidelity and lowest coding delay, which we established from the beginning with Enhanced aptX, are now avaiulable to composite/MPX transmissions with the new APTmpX algorithm.

Maximize your Cost Savings

The compact multi-channel system can save you money by scaling with your needs. SureStream, Enhanced aptX, and APTmpX for low bitrate composite/MPX transmissions form an ecosystem that enables highly-available and cost-effective audio distributions.

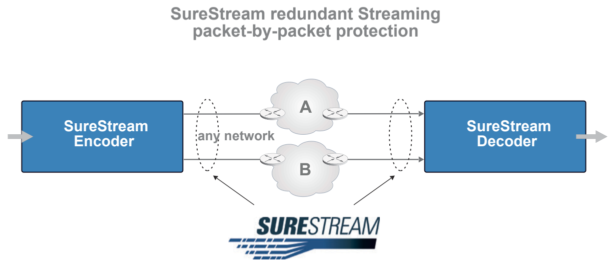
APT MULTI-CHANNEL CODEC **Key Technologies**

SURESTREAM

+10 Years Experience: Our team of engineers has extensive experience optimizing our algorithm for redundant streaming, making SureStream synonymous with reliable transmission in lossy IP networks.

Low Latency - Low Costs: SureStream enables the broadcaster to turn imperfect, but much cheaper services, into true broadcast-grade, low-latency IP connections.

Scalability and Flexibility: SureStream is the most flexible and scalable solution for content transmission protection, able to combine multiple paths from any combination of MPLS, Satellite, Microwave, xDSL and/or Cellular (4G/5G), creating a unified super robust connection to get your audio from point A to B.



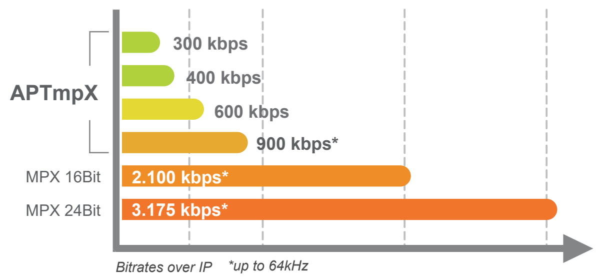
Two or more streams provide the decoder with redundant packets

APTmpX

Compressed Composite/MPX: APTmpX is the industry's best MPX/composite compression algorithm, that delivers the highest sound transparency over low-bitrate IP transmissions.

Lowest Bitrate, Lowest Delay: With tthe lowest bandwidth requirements at 300/400/600 and 900kbps, broadcasters no longer need to compromise between low bit rate and high audio quality.

APTmpX thus eliminates the two barriers that usually discourage migration to FM MPX transmission.



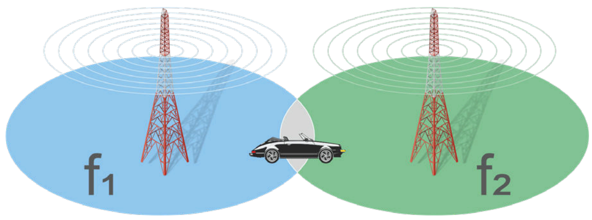
Data rates of linear MPX and compressed APTmpX

Content Time Alignment

Stable IP Latency: The NTP-based Content Alignment feature eliminates variable latencies of an IP network within narrow limits. For program transmissions in multi-frequency networks (MFN), this ensures a seamless program transition between frequencies.

Target Latency Control: The timestamp-based transmission requires only a single setting on the IP Encoder to define the general target latency to each Decoder at the transmitter sites.

Time Aligned Content: The temporal fine-adjustment at the decoder allows the optimal overlay of the modulations in the transition areas.



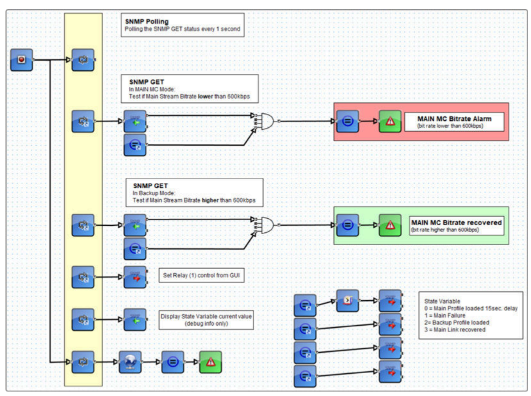
Time aligned overlapping area in an MFN network

SCRIPTEASY

Advanced Telemetry & Facility Management

ScriptEasy is a revolutionary facility control software for connected devices, enabling the automatic correction of any critical errors that may occur. Across its intuitive web interface, ScriptEasy includes management of the GPIO, serial communications, SNMP, logic operators, live user inputs, timers, and more.

Integrated in the AoIP Modules, ScriptEasy is the core technology that provides the device with its inherent "intelligence".



Graphical script application controlling a backup scenario

Support Level Agreement

To make sure you reap all the benefits of your broadcast investment, you can rely on the WorldCast Systems' Support Agreement program. The range of services available and with the support of our team of experts, you will benefit from maximum uptime, better performance, and overall improve your Total Cost of Ownership!

Contact your Sales Manager for more information

Key Features of the APT Multi-Channel

- An AoIP module transports up to two stereo audio channels per module
- Simplex and duplex operational modes
- Point-to-Point and Point-to-Multipoint operation
- Packet redundancy provided by SureStream enables reliable transmissions on the Internet
- The AES192 interface supports digital MPX
- Supports analog MPX with the alternative I/O interface
- Wide range of MPX/APTmpX bandwidths are supported (300 kbps to 4.5 Mbps)
- Protection against carrier overmodulation due to interference from lost IP packets (OMC)
- NTP-based packet timestamping allows to set precise target latencies per stream
- Supports UPnP IGD protocol for configuration of UPnP enabled gateways (routers)
- The advanced NAT feature overcomes inherent port blockages in the network
- Forwarding and protecting of audio or non-audio UDP Streams, such as EDI or E2X data
- Supports “Diffserv” Quality of Service (QoS) on variable DSCP values
- VLANs and virtual IP interfaces enables multi-network integration
- Performance monitoring on each individual IP stream
- Configurable jitter buffer for each receive IP stream (1 ms to 5000 ms)
- Headphone socket for audio monitoring
- Easy connection setup with or without SIP

Audio Over IP Codec Module

The APT AoIP Codec Module includes audio encoding/decoding, IP transport, management and auxiliary data on a single plug-in module.

This enhances the Audio over IP performance of the APT Multi-Channel Codec System as well as increasing its scalability and flexibility.

Fully compatible with the many hundreds of existing units already deployed worldwide, each AoIP card can deliver two independent stereo audio channels on multiple IP streams using multiple unicast or multicast.

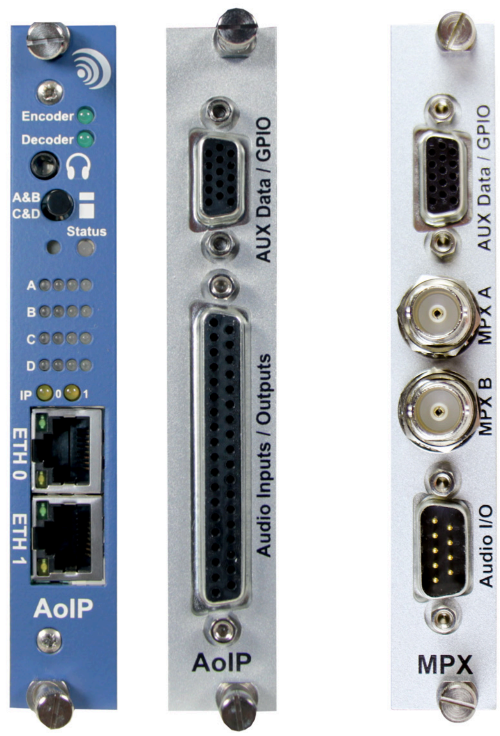
The APT AoIP codec module offers the entire range of audio formats and modes meeting the audio industry’s requirements: dual simplex, stereo-duplex, AES/EBU, AES192, analog connections with HI/LO or 600Ω impedance selection.

It provides broadcast quality audio with support for a variety of standards such as: Linear PCM 16/24 bit, Enhanced apt-X® 16/24 bit, MPEG 1/2 LII, LIII MPEG 2/4 AAC LC/LD/ELD, HE-AAC v1/v2, OPUS and APTmpX (compressed MPX).

The APT AoIP Codec Module is also capable to support up to 88 kHz of bandwidth and scale the sample rate to 192/128 kHz, interfacing to a digital MPX signal in the AES192 format. The output of the AoIP module can therefore be transferred directly to the modulator of the exciter. This 100% digital path eliminates D/A and A/D conversions which may cause degradation of signal quality and could introduce distortion.

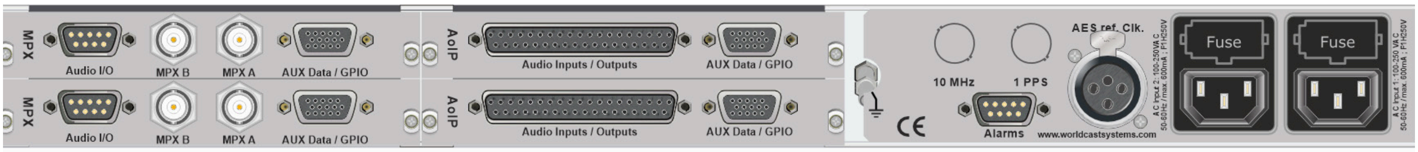
Supporting MPX over IP

The analog MPX interface (BNC sockets) can be used to output analog MPX signals that have been digitally fed in the studio, e.g. if the transmitter does not support AES192.



The AoIP Codec Module with the standard I/O and the analog MPX interface

APT MULTI-CHANNEL CODEC **Technical** Specifications



analog composite/MPX
rear panel interfaces

analog/digital audio/AES192
rear panel interfaces

Dual power supplies
AC or DC or combined

AUDIO	
Asymmetric Audio	Independent audio modes for sent and receive, Tx and Rx or dual Tx or dual Rx; 4 clock domains and auto-detection
Analog I/O	Electronically balanced, capacitive isolated for Left/ Right, Imp. Hi/Lo and 600 Ω, level adjustment in 0.1 dBu steps
Digital Audio I/O	AES-3, AES192, 24 Bit, transformer balanced, Imp. 110 Ω, XLR-Connectors
AUDIO FORMATS	
Multi Algorithm Suite	Eapt-X 16/24 bit, lin. PCM 16/24 bit, MPEG2/4 AAC LC/ LD/ELD, HE-AACv1/2, MPEG1/2 L1/2, OPUS
Digital MPX (AES192)	Lin. MPX 16/24 bit, FS 192/128 kHz or compressed, APTmpX @300/400/600 & 900 kbps
Analog MPX	Lin. MPX 16/24 bit, FS 192/128 kHz or compressed, APTmpX @300/400/600 & 900 kbps
STREAMING MODES	
Stream Types	Multiple stereo Audio, UDP and RTP forwarding, Reply-to-Sender, NAT traversal mode
SIP Modes	Peer-to-peer & SIP-Server mode, multiple SIP user accounts, sym. and asymm. SIP profiles
Unit Clock Modes	Asymmetric, master, slave, or NTP-based
Jitter Buffer	2 - 5000 ms with packet re-sequenzer
QoS	DiffServ (RFC 2474) per stream
Redundant Streaming	SureStream, multi-stream packet-by-packet redundnacy
Backup Feature	SD Card for audio file storage
MANAGEMENT	
Web Browser GUI	
APT NMS	
WCS Kybio (SNMP-based Manager)	
SNMPv2c	
API	
ScriptEasy	
MONITORING & ALARMS	
Adjustable Silence Detectors (Inputs & Outputs)	
Event Logs	
Alarm Relays	
SNMP Traps/Notifications	
PHYSICAL INTERFACES	
Audio on XLR (breakout cable included)	2 analog In-Outputs, 2 digital In-Outputs, 1x ext. AES11 reference Input
Headphone	Mini Jack Socket (front)
AUX Data	HD15-way connector
GPIO	D15-way connectors
Network	2x RJ45

NETWORK	
IP Interfaces	2x 10/100BaseT/Tx, Ethernet IEEE 802.3x, IP4, Auto MDI-X
Port Configuration	Flexible WAN and/or LAN (Management) configuration
VLAN Tagging (IEEE 802.1q)	
Virtual IP Interfaces (IP Aliasing)	
Dynamic DNS	multiple clients
Standard Protocols	DHCP, FTP, HTTPS, ICMP, IGMP v2/3, SMTP, SNMPv2c, NTP, SMTP
Security	TLS 1.1 and higher, Service Filter and Firewall
DATA	
Serial Data per stereo	1x RS232 embedded up to 9600 Baud via UDP stream up to 115.200 Baud
GPIO per stereo	2 switch Inputs and 2 relays embedded (E-aptX) and via UDP stream
Telemetry	Script Easy “distributed intelligence”
MAIN CHARACTERISTICS (CHASSIS)	
Dimensions (l x h x d) 19”, 1u rack mount	483 mm x 44 mm x 370 mm 19” x 1.75” x 14.5”
Weight	5 kg / 11 lbs
Mains power supply	90-264 VAC / 47-63Hz
DC power supply	36 - 75 VDC
PSU population options	Dual AC, dual DC, or AC & DC
Power consumption	10VA per AoIP module
Env. Temperatures Operation Storage Humidity	 0 °C - +45 °C -30 °C - +80 °C 95 % (non-condensing)

Order information

REF	DESCRIPTION
TF01250-AC-AC	APT Codec Frame with AC/AC PSU
TF01250-AC-DC	APT Codec Frame with AC/DC PSU
TF01250-DC-DC	APT Codec Frame with DC/DC PSU
STP00034	AoIP Module for APT Codec Frame 1U with XLR-Breakout Cable
SPP00049	AUX/GPIO Breakout Cable for AoIP Module
CD00123	SureStream Technology license (per AoIP Module)
LC00074	Digital MPX over IP option for APT Codecs

This document is not contractual. All specifications are subject to change without notice.

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