TECHUPDATES

TELOS ALLIANCE INTRODUCES OMNIA MPX NODE

CLEVELAND — New to the Omnia product line, the Omnia MPX Node from the Telos Alliance provides new options for STL applications. Like its namesake — the classic Axia xNode — Omnia MPX Node is a building-block technology that helps stations leverage the growing power and capability of data networking.

The Omnia MPX Node is the first purpose-built hardware codec capable of sending or receiving full FM signals at data rates as low as 320 kbps, utilizing the Omnia µMPX algorithm, ideal for networks with limited capacity (including IP radios). MPX Node makes peak-controlled L/R baseband, stereo pilot, and RDS data routable from a studio to one or many FM transmitters.

By transporting an FM composite signal rather than left/right audio, broadcasters can keep their on-air processing and RDS encoding at the studio, then deliver a transmission-ready, peak controlled FM multiplex signal directly to an FM transmitter without the need for transmitter-side peak limiting or stereo generation.

The MPX Node is available as either an encoder or decoder, and a pair of units creates a complete system. Alternatively, the Omnia.9 audio processor running MKII software with an Omnia MPX Encoder License can be used with a single Omnia MPX Node Decoder to create an end-to-end system.

For information, contact the Telos Alliance in the United States at +1-216-241-7225 or visit www.telosalliance.com.

DIGIGRAM IQOYA X/LINK RANGE EASES LIVE REMOTES

MONTBONNOT, FRANCE — Digigram's Iqoya X/Link Range (AES, DUAL, ST and LE) is a stereo- to multi-stereo IP audio line of codecs. It is possible to use them in both legacy audio facilities and in full-IP AES67, Ravenna, or Livewire audio infrastructures.

Based on low energy consumption and a fan-less hardware platform, the company recommends the Iqoya X/Link range for mission-critical and 24/7/365 use.

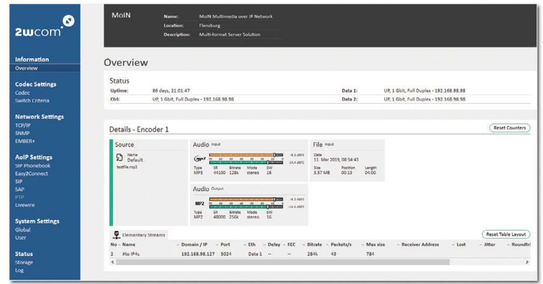
The processing power of Iqoya X/Link promises low-latency audio connections for live remote needs. In addition, multiple levels of redundancy ensure audio service continuity. The solution features two internal power supply units, four network ports for traffic separation (WAN, LAN, management), stream redundancy, audio failovers, audio hardware bypass, and 1+1 hot-device redundancy.

Iqoya X/Link is EBU/ACIP-compliant for interoperability with third-party codecs and any SIP infrastructure. For easy integration with codecs and network management/monitoring systems, it comes with SNMP, and web services.

Iqoya Serv/Link is a high-density 1U rack that supports four to 64 stereo (eight to 128 mono) input and output channels with the possibility to simultaneously encode, decode and transcode IP audio streams. It supports various audio I/O formats, such as AES/EBU, MADI, AES67 and Dante.

For information, contact Digigram in France at +33-4-76-52-47-47 or visit www.digigram.com.

2WCOM'S MOIN OFFERS INTEROPERABILITY



FLENSBURG, GERMANY — 2wcom says the recently launched Linux-based MoIN multimedia over IP network server eases the day-to-day work of network operators.

The server is available as hardware or virtualized software. Its new and advanced technological concept assures outstanding flexibility in application by supporting studio-to-studio and studio-to-transmitter links as well as broadcasters increasing cross-media tasks, the company says.

In addition, according to the company, MoIN's high-level compatibility is especially important in mixed networks and expanded infrastructures. For this purpose, the server supports all major protocols for internet interoperability like Ravenna, Livewire+, Dante and SRT.

Harmonization of data exchange between the protocols is carried out by standards such as EBU N/ACIP Tech 3326, AES67, SMPTE ST 2110, SMPTE 2022-7 and Ember+. To easily exchange data in mixed networks MoIN can transform protocols, e.g. from Livewire+ to Ravenna, when transmitting from a studio site to the headend.

Eying future crossmedia tasks, customers benefit from MoIN's transcoding feature, which supports all main codec algorithms. For high-quality, low-latency real-time applications, that might mean PCM to Opus. For economical needs, like dealing with low bandwidth or uploading files to a station website transcoding, for example, AAC profiles (even AAC xHE), Enhanced aptX or Ogg Vorbis is possible.

For live events the server efficiently supports the on-site team thanks to features such as the SIP phone book, temporary channel activation, combining individual audio streams into multichannel streams, as well as precise synchronization (PTPv2).

For information, contact 2wcom in Germany at +49-461-662830-0 or visit www.2wcom.com.

APT MOBILE SURESTREAMER IMPROVES QUALITY OF LIVE REMOTE BROADCASTING

MERIGNAC, FRANCE — The APT Mobile SureStreamer (MSSr), a WorldCast Systems technology, is a new mobile network access solution designed for live remotes and OBs.

It works with most portable codec types and brands to deliver clear, uninterrupted radio and video directly from the field.

APT says that the Mobile SureStreamer is a user-friendly solution, in a lightweight shoulder bag, with zero-field configuration, so users can rely on immediate and simple connection between the field and the station.

It works with any codec type and brand to improve the quality of remote broadcasting and adds no additional latency according to WorldCast.

The company says it provides a link so reliable that the codec's automatic buffering is redundant and latencies can be trimmed to a minimum. It adds that courtesy of the core technology, SureStream ensures flawless audio and it reduces operating costs by distributing content over affordable 3G/4G public internet links rather than ISDN, MPLS or satellite.

APT says Mobile Surestreamer has proven reliable in field tests of applications for sports broadcasters, remote desktops and journalists at major events. It quoted Conor Ewings, broadcast engineer at Bauer Media in Northern Ireland, as saying latency was "beyond expectations … rock-solid at 60 ms and not a single packet has been dropped."

The company says that overall, the product improves the quality and reliability of IP remotes to deliver audio and video for live and local remotes and OBs.

For information, contact WorldCast Systems in France at +33-557-928-928 or visit www.worldcastsystems.com.