Zephyr Xstream

Telos

In 1993, we had an idea. We envisioned a way to send CD-quality audio over standard digital phone lines — the perfect marriage of advanced audio coding and digital telephone technologies. The realization of that dream resulted in the Telos Zephyr™, which transformed broadcasting by making ISDN an easy-to-use, effective tool for sending and receiving high-quality audio.

Since its introduction, broadcasters and audio professionals worldwide have made Zephyr the most successful digital broadcast product ever. Its name has become synonymous with easy, instantaneous point-to-point audio transfer: "Just Zephyr it to me!"

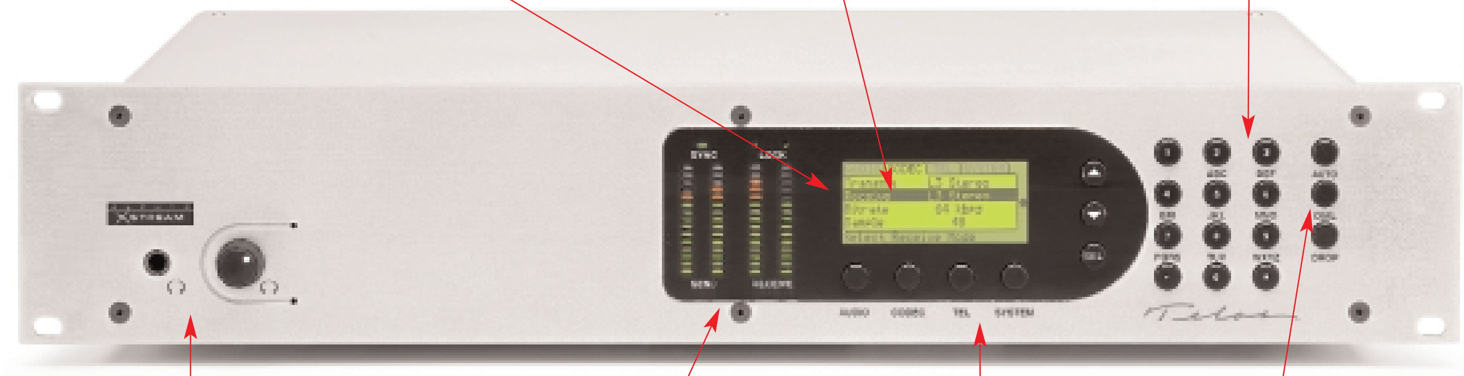
Zephyr Xstream™ continues this tradition of excellence. It includes all the tools and features Zephyr users have come to rely on — ISO/MPEG Layer III, Layer II and G.722 coding, straightforward front-panel controls, full-duplex, 20kHz stereo audio, analog and digital I/O — as well as new capabilities, such as MPEG-2 AAC (Advanced Audio Coding), Ethernet connectivity for remote control and IP-audio streaming, a straightforward, friendly graphical user interface, and exclusive DSP-based audio processing by Omnia (on models with built-in mixers).

IN 1993 WE HAD AN IDEA

High-resolution backlit graphical display screen with friendly, logical control structure makes setup easy.

The Zephyr Xstream is the first family of broadcast codecs with MPEG-2 AAC (Advanced Audio Coding) for delivering stunning true-to-life audio with reduced delay.

Integrated dialing keypad controls built-in US and EURO-ISDN compatible Terminal Adapter.



Convenient front-panel headphone jack with level control.

At-a-glance metering of send and receive channels. Two-channel flexibility over a single ISDN circuit – or two synchronous links – can be used to transmit and receive stereo audio from a single location, or two mono channels to and from separate locations.

Fast Access Menu Keys quickly call up system information and settings.

Auto-Dial up to 50 stored Preset Numbers, each with its own bitrate and transmit/receive coding settings. 30 Location settings permit quick recall of ISDN line and audio settings for your most commonly visited sites.

Zephyr Xstream is a full-featured ISDN transceiver with high-fidelity MPEG-2 AAC coding, along with standard Layer 3, Layer 2 and G.722. It includes AES/EBU I/O and features full duplex stereo operation of up to 20kHz audio on a single ISDN line; broadcast quality mono audio at 15kHz or 20kHz is possible on a single ISDN “B” channel or other 56/64 kbps channel. Includes built-in US and EURO-ISDN Terminal Adapter. There’s also a 10Base-T Ethernet port for remote control and IP audio streaming.

CD-QUALITY AUDIO OVER STANDARD DIGITAL PHONE LINES

Inputs 1 & 2 provide switchable 48-volt phantom power to ease mic setup.

Four-input stereo DSP mixer directly feeds the codec section. Mic/line switchable inputs with pan include preset mic-limiter & AGC processing by Omnia®.

Layer 3 Dual Mode lets you receive 2 separate mono feeds from independent far end sites. Great for sporting events, network operation centers or other split-feed situations.

Back panel Ethernet port on all Zephyr Xstream models allows TCP/IP remote control and streaming output of IP audio.

Adjustable front panel headphone jack for Local Mix 1 monitors Send or Receive audio, or a mixture of the two.

Local Mix 2 has separate front-panel controls for three rear-panel headphone jacks, and a balanced XLR line output.

Rugged shock-resistant case helps Zephyr Xstream MXP stand up to the rigors of the road.

Alpha-numeric dialing pad also generates DTMF tones for navigation through voice menu systems.

The portable Zephyr Xstream MXP includes all the features of the Zephyr Xstream, plus a digital four-channel stereo mixer with two local mixes into a road-ready case for on-the-go broadcasting. Zephyr Xstream MX (not pictured) offers the flexibility of these mixing and remote-control features in a convenient rackmount package.

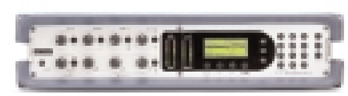
IDEAL SOLUTION FOR REMOTE BROADCASTS

5 MODELS

The Zephyr Xstream family of ISDN codecs are specifically designed to transmit high-quality audio over single ISDN circuits, low bit-rate transmission paths and IP connections. Models include:

Zephyr Xstream. Transmit two-way, 20kHz stereo audio plus ancillary data virtually anywhere in the world using just a single ISDN circuit, or transmit 15kHz or 20kHz mono audio on a single ISDN “B” channel.

Zephyr Xstream MX adds the utility of a digital 4-channel stereo mixer to the rackmount Zephyr Xstream. Mixer features switchable mic/ line inputs, selectable audio processing presets by Omnia®, and two local mix outputs.



Zephyr Xstream MXP, the portable version of the Xstream MX, packs the power of Zephyr Xstream into a rugged, carry-anywhere chassis.

• FEATURES

- Telos Systems’ coupling of MPEG Layer 3 and ISDN has made Zephyr the #1-selling broadcast codec around the world. Zephyr Xstream continues this tradition of innovation by incorporating MPEG-2 AAC (Advanced Audio Coding) for even greater fidelity at low bit-rates.
- 10Base-T Ethernet port allows remote control and streaming of MP3-coded audio over a LAN, WAN or the Internet. TCP/IP network connectivity also allows easy upgrade of system software via FTP.
- New Auto-Receive mode answers inbound calls and automatically determines correct decoder settings for the incoming audio stream.
- Built-in ISDN terminal adapter makes fast work of connection to US and EURO-ISDN phone lines via a single modular cable.
- Zephyr Xstream’s friendly, straightforward control panel with its high-resolution graphical interface will have you up and running in minutes. Programming is simple and intuitive using onscreen menus, and there’s an adjustable headphone jack for monitoring audio.
- An available V.35/X.21 port allows Zephyr Xstream to be connected to one or two Switched 56 lines, dedicated lines, satellite links, and other data paths.
- The ISDN Telephone feature (G.711) allows Zephyr Xstream to dial any standard analog telephone line for voice communications. DTMF tones are provided for navigation through automated menu systems.
- Advanced technology means less heat; no cooling fan means less studio noise.

- Split-channel Layer 3 transmit mode allows individual mono signals to be transmitted to separate far-end sites. This feature is ideal for bilingual programming, as the audio on each channel is completely separate.

- Unique Dual-Receive mode in Layer 3 allows reception of independent audio streams arriving from two distant codecs. Ideal for network operation centers and shared equipment facilities.

- 50 Auto-dial storage positions include the codec section settings (such as bit rates and transmit and receive coding choices) and the numbers of the remote location you wish to dial.

- 30 Location storage positions allow you to store and retrieve ISDN line information and audio parameters for up to 30 different "on-the-road" sites.

- Local control options include TCP/IP (Telnet), RS-232, or "panic dial" contact closures.

- End-to-end ancillary data options in Layer 3 and AAC include 9600bps serial data stream and eight bi-directional contact closures.

- Zephyr Xstream MX and MXP models include a "four- in/ two- out" DSP-powered main mixer stage which feeds the codec directly. Four mic/line inputs include selectable AGC/limiter processing by Omnia®.

- Zephyr Xstream model includes standard AES/EBU digital input/output provides maximum flexibility for connection with digital studio equipment. Sample rate conversion is available on both input and output paths. Sample rates of 32, 44.1 and 48kHz are supported. Zephyr Xstream accepts external clock or can generate clock when required.

- Zephyr Xstream is the ideal solution for remote broadcasts, ad hoc networks, voiceovers, commercial distribution, backup to microwave and satellite links, and many other applications.

ZEPHYR XSTREAM MAKES ISDN AN EASY-TO-USE, EFFECTIVE TOOL FOR SENDING AND RECEIVING HIGH-QUALITY AUDIO

Zephyr Xstream Transmission Modes

Zephyr Xstream offers MPEG-2 AAC coding for highest fidelity audio transmissions, as well as industry-standard MPEG Layer 3, Layer 2 and G.722 coding for compatibility with the widest range of codecs from other manufacturers. With Zephyr Xstream, you are in touch with the world.

Stereo and Dual-Mono Modes using two ISDN "B" channels:

- MPEG-2 AAC Joint Stereo at 20kHz or 15kHz for maximum fidelity.
- MPEG-2 AAC Independent Stereo at 20kHz or 15kHz.
- Layer 3 Joint-Stereo at 20kHz or 15kHz for high fidelity and compatibility.
- Layer 3 Independent Stereo at 20kHz or 15kHz for compatibility and surround-sound transmission.
- Layer 3 Dual-Mono at 20kHz or 15kHz when each channel has unique audio.
- Layer 2 Joint Stereo at 20kHz for compatibility.
- Layer 2 Independent Stereo at 7.8 or 9.8kHz for compatibility.
- G.722 Dual-Channel at 7kHz for compatibility and/or lowest delay.

Mono Modes using one ISDN "B" channel:

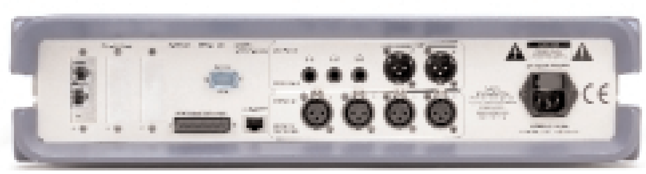
- MPEG-2 AAC at 20kHz or 15kHz for maximum fidelity.
- Layer 3 at 20kHz or 15kHz for high fidelity and compatibility.
- Layer 2 at 7.8kHz or 9.8kHz for compatibility.
- Layer 2 Mono at 8.6kHz (24kHz fs) for compatibility.
- G.722 at 7kHz for compatibility and/or lowest delay.

Split-Channel Modes using two ISDN "B" channels:

- Layer 3 Mono-Dual allows each “B” channel to independently transmit and receive 20kHz or 15kHz.
- G.722 Mono-Dual allows each “B” channel to independently transmit and receive 7kHz for compatibility and lowest delay.

ISDN Telephone Mode:

- G.711 is used to call a standard POTS telephone for low-grade voice communications at 300Hz-3400Hz. Two simultaneous G.711 calls are possible.



PERFECT MARRIAGE OF ADVANCED AUDIO CODING & DIGITAL TELEPHONE TECHNOLOGIES

About Audio Coding

What is coding and why is it required to transmit audio over ISDN?

Without data rate reduction, high-quality audio requires a transmission capacity of about 700kbps for each audio channel. Channels that can handle data rates that high are very expensive and hard to get. More affordable and accessible channels, such as the two 64kbps channels in each ISDN circuit, offer a rate of only about 9% of that of a compact disc. That means you must do some coding to get "12 gallons of water into a one-gallon container." (Note that some refer to coding as "compression" but are generally referring to the same process.)

How can coding be accomplished?

One might think that lossless, redundancy-reducing methods (such as those used for computer hard-disk compression) would be ideal for audio. Unfortunately, there is not enough redundancy in the audio signal for the significant reduction required by ISDN. Early coding schemes, such as Adaptive Delta Pulse Code Modulation (ADPCM), take advantage of the fact that it takes fewer bits to code the difference, or delta, between successive audio samples compared to using their individual values. Further efficiency is had by adaptively varying the difference comparator according to the nature of the program material. However, the reduction power of ADPCM is insufficient for transmitting full bandwidth audio over ISDN, as evidenced by 7kHz codecs that use G.722.

To develop coding algorithms with sufficient power to achieve the desired reduction, the audio industry has turned to psychoacoustics. Using carefully researched psychoacoustic principles, coding processes have been designed to reflect the way in which human hearing interprets audio information.

How does perceptual coding work?

With perceptual coding, only information that can be perceived by the ear and the brain is retained. It has been discovered that certain audio creates a "mask" that hides other audio. The masking depends on the frequency, the level, and the spectral distribution of both the masker and the masked sounds. These masks occur in both the frequency and time domains. Perceptual coding takes advantage of masking by reducing the resolution of signals that fall below the mask.

Find out more about audio coding by visiting us online at: www.telos-systems.com/techtalk/coding

About MPEG-2 AAC

(Advanced Audio Coding)

The MPEG-2 AAC system is the newest audio coding method selected by MPEG and became an International standard in April 1997. It is a fully state-of-the-art audio compression tool that provides performance superior to any known approach at bit rates greater than 64 kbps and excellent performance relative to the alternatives at bit rates reaching as low as 16 kbps.

AAC is the first codec system to fulfill the ITU-R/EBU requirements for indistinguishable quality at 128 kbps/stereo. It has approximately 100% more coding power than Layer 2 and 30% more power than the former MPEG performance leader, Layer 3. AAC takes advantage of such tools as temporal noise shaping, backward adaptive linear prediction and enhanced joint stereo coding techniques. Zephyr Xstream is the first broadcast codec to incorporate the power of AAC coding, resulting in superior high-fidelity audio at lower bitrates and with less delay than Layer 3 or Layer 2.

More details about AAC are available online at: www.telos-systems.com/techtalk/aac

About MPEG Layer 3 and Layer 2

The international standards group ISO/IEC established the ISO/MPEG (Moving Pictures Expert Group) in order to develop a universal standard for encoding moving pictures and associated audio for use with digital storage and transmission media. The standard was finalized in November 1992, with three related algorithms, called Layers, being defined for encoding of audio, taking advantage of psychoacoustic effects.

Telos pioneered the use of Layer 3 for transmission of broadcast-quality audio with the introduction of the original Zephyr in 1993. Layers 2 and 3 have found widespread use in broadcast and professional audio applications. Zephyr Xstream harnesses the power of both Layer 3 and Layer 2 efficiently and economically for high-fidelity audio and compatibility with a wide range of other codecs.

More details about MPEG Layer 3 and Layer 2 are available online at:
www.telos-systems.com/techtalk/mpeg

About ISDN

Integrated Services Digital Network (ISDN) is an international standard that defines a worldwide, completely digital switched telephone network. ISDN is designed to carry large amounts of information and has a number of potential uses, such as high-speed modem communications and desktop videoconferencing. For broadcast and professional audio, ISDN offers unique opportunities for the transmission of high-quality audio.

The form of ISDN of most interest to broadcasters and audio professionals is called Basic Rate Interface (BRI) or S0. On a single pair of ordinary phone wires, BRI offers two “bearer” channels at a 64kbps transmission rate and one “data” channel at 16kbps. This configuration is often referred to as 2B+D. When ISDN BRI is installed in your facility, each line is brought in on only one pair of wires.

ISDN is full duplex and calls are dialed and routed just like analog calls. Zephyr Xstream uses the two “B” channels for bi-directional audio (transmitted as digital data) and ancillary RS-232 data. The “D” channel is reserved exclusively for telephone network signaling.

There is also ISDN Primary Rate Interface (PRI) or S2M. In the Western Hemisphere, PRI offers 23 “B” channels and one “D” channel. In Europe and Asia, this service offers 30 “B” channels and one “D” channel.

Zephyr Xstream connects directly to ISDN BRI lines. When equipped with an optional V.35/X.21 data port and an external CSU/DSU, Zephyr Xstream can also connect to fractional T-1, Switched 56 or any other synchronous digital transmission path.

More information about ISDN is available online at:
www.telos-systems.com/techtalk/isdn

JUST
ZEPHYR IT
TO ME

GENERAL

Full duplex, high-fidelity codec using MPEG-2 AAC, ISO/MPEG Layer 3, ISO/MPEG Layer 2, and G.722, fully compliant with international standards. Stereo or mono transmit configurations, both with stereo receive.

Frequency Response (+0/-1dB, swept sine tone procedure)

AAC and Layer 3 all modes:	20-20,000Hz at 48kHz fs., 20-15,000Hz at 32kHz fs.
Layer 2 mono, dual-mono:	20 Hz-7.8kHz/9.8kHz at 48kHz fs.
Layer 2 joint stereo:	20-20,000Hz at 48kHz fs.
G.722: 20-7,000Hz.	

THD+N

Far loopback (end-to-end), 48kHz fs, analog I/O, input at 1kHz +20dBu, .004%

Dynamic Range

A Weighting, AAC, Layer 3 or 2 end-to-end: 101dB typical

SEND INPUT

Active balanced with RF protection.

Zephyr Xstream:

LINE:	-12 to +4dBu nominal level.
Clip point:	+21dBu.
Bridging:	< 10K ohms (x2)

XLR female/quarter-inch TRS combo connector.

Zephyr Xstream MX and MXP:

LINE:	-12 to +4dBu nominal level.
Clip point:	+21dBu.
-68 to -35dBu level.	
Bridging:	< 10K ohms (x2)

XLR female/quarter-inch TRS combo connector.

LIMITER - Zephyr Xstream MX and MXP only:

Internal DSP-based AGC/limiter with Omnia® audio processing. Includes presets for music & voice, selectable per channel.

LINE BIT RATES

56 or 64kbps per channel, front panel selectable.

RECEIVE OUTPUT

Active differential

Level:	Front panel selectable for -10 or +4dBu, nominal.
Impedance:	33 ohms (x2)
XLR male	

AES/EBU DIGITAL I/O (Zephyr Xstream only)

Sample rates supported:	32, 44.1 and 48kHz
Rate conversion:	Input and output independently selectable. Can be bypassed.
Input clock:	From external source or Zephyr telco clock.
Output clock:	From sample frequency, external source, or AES/EBU input.

MULTIPLEX/DEMULTIPLEX

Internal channel splitting/combining of two network channels for stereo modes.

MPEG-2 AAC:	Telos Zephyr protocol.
Layer 3:	FHG/Telos Zephyr protocol.
Layer 2:	CCS CDQ™ protocol compatible.

OPTIONAL V.35/X.21 DIRECT DIGITAL INTERFACE

Two ports, both V.35/X.21. Automatically selected when the appropriate cable is connected.

ISDN INTERFACE

Compatible with National ISDN, AT&T 5ESS, Northern Telecom DMS-100, Siemens EWSD, INS 64(Japan) and EURO-ISDN. Compatibility and approval pending in some countries; contact Telos for current status.

LAN INTERFACE

10Base-T Ethernet port using RJ-45 connector. Supports TCP/IP (Telnet and FTP).

ISDN VOICE TELEPHONE MODE

Two channels using G.711 standard, µLaw or A-Law. 300–3,400Hz. DTMF signaling provided (CCITT standard).

REMOTE CONTROL AND ANCILLARY DATA

RS-232 9-pin D-Sub female (DCE): Asynchronous; 8 data, no parity, 1/2 stop, 2400-57,600 baud.
10Base-T Ethernet port using RJ-45-style connector.
Bi-directional ancillary data: Serial connection at 9600bps; eight contact closures.

SOFTWARE UPGRADES

Downloadable from Telos; uses FTP over 10Base-T Ethernet connection.

CONTROL PORTS

Eight bi-directional inputs/outputs for end-to-end contact closure emulation.

Inputs:	Open collector, closure to ground.
Outputs:	Sink up to 125 mA to ground.

RESOLUTION

Send Input:	MPEG, 24 bits; G.722, 16 bits
Receive Output:	MPEG, 24 bits; G.722, 16 bits

POWER SUPPLY

90-240 VAC (50/60 Hz) auto-configuring.
100 watts peak
CE approved

DIMENSIONS, ZEPHYR XSTREAM AND ZEPHYR XSTREAM MX

19" (48cm) standard rackmount front panel
17 1/8" (43cm) wide behind front panel
12 1/2" (32cm) deep
3 1/2" (9cm) high

DIMENSIONS, ZEPHYR XSTREAM MXP

18 1/4" (46cm) wide
14" (36cm) deep
4 1/4" (11cm) high

SHIPPING WEIGHT

17 pounds (7.7kg)

Telos

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