Zephyr

Xstream • Xport

In 1993, we had an idea. We envisioned a way to send CD-quality audio over standard digital phone lines by marrying advanced audio coding and digital telephone technologies. The result: the Telos Zephyr, which transformed broadcasting by making ISDN an easy-to-use, trouble free tool for sending and receiving high-quality audio.

Broadcasters and audio professionals worldwide have since 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!" And the Zephyr family has grown — from a single model to a complete studio-to-studio and field audio transmission system.

A Zephyr Xstream at your studio becomes a "universal codec." It can connect with every popular ISDN codec for full-duplex, 20kHz stereo audio, transmit and decode streaming MP3 audio over Ethernet, and even connect with the revolutionary Zephyr Xport portable codec via analog (POTS) telephone lines, so you can simpify your life. You won't need a proliferation of codecs in your rack, so you save space, operation hassle, phone lines — and money.

In the field, Zephyr Xstream and Zephyr Xport are powerful remote tools, perfect for on-location broadcasts, news gathering, interviews and remote studio linkups. New MPEG AAC-LD coding lets you transmit Layer 3-quality audio with greatly reduced transmission delay, enabling smooth, natural, high-quality two-way audio. Zephyr models with built-in mixers and Phantom power help reduce equipment inventory and setup time; intuitive operation and simple user interfaces make operation easy even for non-technical personnel.

Exclusive new capabilities in the Zephyr Xport now give you unparalleled flexibility for remote broadcasts. Break-through technology developed by Telos lets Zephyr Xport communicate with your studio's ISDN Zephyr Xstream, from a standard POTS line — perfect for remote locations where ISDN is unavailable. Zephyr Xport's reliable digital connection, coupled with aacPlus audio coding, delivers clear, clean audio and rock-solid connections never before possible with a POTS codec. (An ISDN option lets your Zephyr Xport connect over digital phone lines, too.)

Now more than ever, Telos Zephyr is truly The Best Way To Hear From There.

IN 1993 WE HAD AN IDEA

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

The Zephyr Xstream family of ISDN codecs features MPEG-2 AAC (Advanced Audio Coding) for true CD-quality audio transmission; Low-Delay AAC-LD delivers Layer 3 quality with up to 70% less 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 data 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 100 stored Preset Numbers, each to store its own bitrate and transmit/receive 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 and low-delay MPEG4 AAC-LD 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 ISDN Terminal Adapter for worldwide use. 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 switch-able 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, net-work 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, plus easy software upgrades via FTP.

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 head-phone jacks, and a pair of balanced XLR line outputs.

Rugged shock-resistant case helps Zephyr Xstream MXP stand up to the rigors of the road. Flip-up stand tilts unit up for best viewing angle.

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.

STUDIO QUALITY AUDIO ON THE GO

5 MODELS

The Zephyr Xstream family of ISDN codecs transmit high-quality stereo and mono audio over single ISDN circuits, low bit-rate transmission paths and IP connections, and can be ordered with ISDN, V.35/X.21 or Ethernet-only connections. Models include:

Zephyr Xstream. Transmit two-way, 20kHz stereo audio plus ancillary data anywhere in the world using a single ISDN circuit, transmit 15 or 20kHz mono audio on a single ISDN "B" channel, stream or decode MP3 audio via Ethernet. Includes AES/EBU I/O.

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.

• XSTREAM Q&A

- *It seems like everyone I know uses Zephyr. Why is Zephyr so popular?*
  Ease of use, along with Telos Systems' coupling of MPEG Layer 3 and ISDN, made Zephyr the #1-selling broadcast codec around the world. Zephyr Xstream has won even more acclaim by incorporating MPEG-2 AAC (Advanced Audio Coding) for even greater fidelity with less delay at lower bit-rates.

- *Our facilities are IP networked. Can I control Zephyr Xstream over my network?*
  Yes. A 10Base-T Ethernet port allows remote control using a Web browser over a LAN, WAN or the Internet. TCP/IP connectivity also lets you upgrade system software easily via FTP. Local control options include TCP/IP (Telnet), RS-232, or "panic dial" contact closures.

- *Does Zephyr Xstream have a cooling fan?*
  No. Advanced technology means less heat; no cooling fan means less studio noise.

- *Some codecs are very complicated to set up. Is Zephyr Xstream easy to use?*
  All Zephyr models are intuitive and user-friendly. The straightforward control panel, graphical menus and on-screen help will have you up and running in minutes. There's even an auto-receive mode that answers inbound calls and automatically determines the correct decoder settings for the incoming audio stream.

- *Can I use my Zephyr Xstream with digital connections other than ISDN?*
  Yes. An available V.35/X.21 port works with Switched 56 lines, satellite links, and other synchronous data paths. Xstream supports bit rates of up to 384 kbps, making it ideal for STLs, studio-to-studio links and other critial applications.

- *Does Zephyr Xstream work with DSL?*
  Yes, Zephyr Xstream can send audio over DSL and other IP services via Ethernet, although ISDN and

(continued)

dedicated synchronous data channels are still the most reliable choices due to the variable latency and non-guaranteed delivery inherent in packet-based data transmission. Please see our Technical Paper entitled “DSL vs. ISDN,” available on our website at www.telos-systems.com/techtalk/dsl/ .

- *I was told I can connect to an ISDN Zephyr Xstream using a POTS codec. Is this true?*
  Using custom modem technology, the Zephyr Xport uses ordinary POTS phone lines in the field to connect to your Zephyr Xstream at the studio. See the Zephyr Xport section later in this brochure for details.

- *I've heard lots of "buzz" about MPEG AAC. What is it?*
  “AAC” stands for “Advanced Audio Coding.” It's the newest high-performance audio coding standard, with approximately 100% more coding power than Layer 2 and 30% more power than the former MPEG performance leader, Layer 3, It was specifically developed to meet the ITU BS.1115 specification that calls for “indistinguishable source from output” in a 128kbps stereo stream. Using MPEG AAC, Zephyr Xstream can transmit and receive true CD-quality audio over ordinary ISDN lines.

- *What's Low-Delay MPEG AAC coding?*
  Zephyr users have known for years that Layer 3 offers all the fidelity needed in most broadcast situations. However, the delay that results from Layer 3 can be un-nerving to talent, particularly if high fidelity is needed in both directions. “Low Delay AAC” or “AAC-LD” for short, offers quality equivalent to Layer III with about 75% less delay.

- *We do several remote broadcasts each week. Can we store our most frequently dialed numbers?*
  Yes, Zephyr Xstream provides 100 auto-dial locations for your most frequently dialed numbers. There are also 30 Location storage positions that let you store and retrieve ISDN and audio parameters for the remote locations you visit most.

- *Can I send digital audio directly from a Zephyr to my studio equipment?*
  Yes. The non-mixer version of Zephyr Xstream features AES/EBU digital I/O for connections to digital studio equipment. Sample rates of 32, 44.1 and 48kHz are supported on both input and output paths. Zephyr Xstream accepts external sync clock, or can generate clock when required.

- *We broadcast sporting events in two languages, and I want to send both mono feeds simultaneously.*
  Zephyr Xstream has this capability. A Split-Channel Layer 3 transmit mode lets you send individual mono signals to separate far-end sites. This feature is ideal for bilingual programming, as the audio on each channel is completely separate.

- *Can I receive split feeds, too?*
  Yes, Dual-Receive modes in Layer 3 and G.722 allow reception of independent audio streams arriving from two distant codecs. Ideal for network operation centers and shared equipment facilities.

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

Friendly graphical display with back-light makes Zephyr Xport as easy to use as a cell phone. During transmission, operators can monitor send and receive levels simultaneously with modem performance.

Unique ergonomic shape fits easily under the arm. It's stackable, too.

Rugged shock-absorbing bumpers protect Zephyr Xport from jolts and jars.

Navigation keys surrounding the display help speed your way through graphical menu selections. NAV key provides fast access to menu settings; function, scroll and SEL keys make it easy to choose the options you want.

Stowable control knobs allow you to "set it and forget it," preventing accidental level changes. Push knobs in to lock settings; push again to extend for use.

Easy-to-use mixer section with Mic & Line inputs. MIX control lets you blend IFB audio with send audio for headphone and PA feeds. Front-panel headphone level adjustment controls rear-panel output.

AUTO key gives one-button access to 100 stored Autodial numbers and 30 Location setups.

Going on the road? Zephyr Xport is the perfect companion. With Xport as part of your remote system, your studio's Zephyr Xstream becomes a universal codec, connecting with both Xport and ISDN codecs, saving money, rack space, training time, telephone lines, and conserving on console/router audio inputs and mix-minus outputs.

IDEAL SOLUTION FOR REMOTE BROADCASTS

Zephyr Xport provides two bi-directional contact closures for remote operation of connected devices.

Low-heat operation means no cooling fan for noise-free operation.

Auto-switching internal power supply means no “wall warts” to carry around or lose.

Telephone interface includes connections to an analog (POTS) phone line, a telephone handset for making voice calls, and ports for ISDN connections (when equipped with one of the ISDN transmission options).

10/100Base-T Ethernet port facilitates remote control of Zephyr Xport using a local computer, LAN or WAN connection and Web browser. Feed PCM audio directly into Xport from any Windows®-based computer.

Full-featured I/O includes mic input with provision for 12-volt Phantom power, direct output of received audio, headphone and monitor mix outputs and line-level input.

AUX INTERFACE provides convenient input/output to a cellular phone’s headset jack for wireless transmission, or to send audio cues from auxiliary sources (such as a secondary mixer) directly to talent’s headphones. May also be used as a dedicated record output.

At the heart of Zephyr Xport is a custom DSP-based modem, optimized for maximum performance with audio codecs. Exclusive Telos technology lets Xport use a standard analog phone line to connect with any Zephyr Xstream ISDN codec; ground-breaking aacPlus coding with Spectral Band Replication delivers stunning reproduction of voice, music or both. A full-featured digital mixer with mic and line inputs (and selectable audio processing by Omnia) completes the package.

JAW-DROPPING AUDIO WITH ROCK-SOLID CONNECTIONS

Zephyr Xport codecs utilize ordinary analog phone lines to communicate with ISDN-connected Zephyr Xstream studio units for transmission of high-quality mono audio; optional ISDN capability allows connection to Zephyr Xstream by POTS or ISDN.

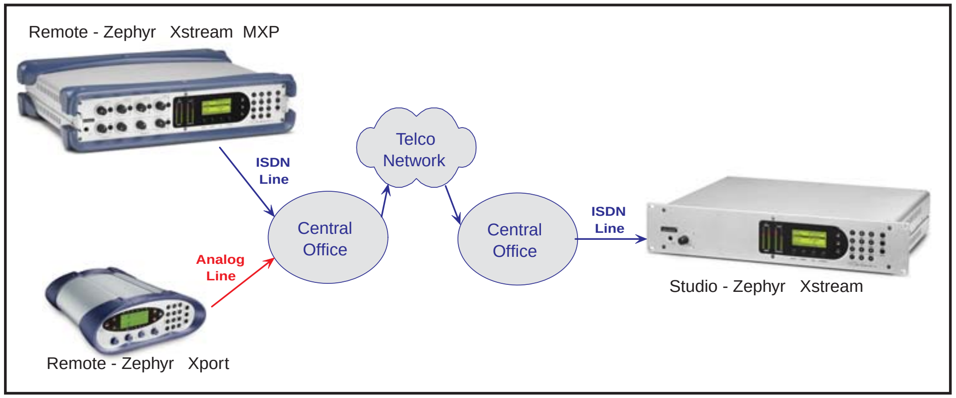
Zephyr Xport POTS. Custom Telos modem optimized for stable audio connections enables transmission of 15kHz bidirectional audio plus ancillary data to your studio's Zephyr Xstream, using a common analog telephone line. Employs Coding Technologies' aacPlus algorithm for spectacular fidelity at real-world connect rates. Includes Ethernet remote control and a full-featured mixer with voice & music processing by Omnia.

Zephyr Xport POTS+ISDN. What type of phone lines await you at your next remote? Analog? Digital? There's no need to carry along a codec for each kind when Zephyr Xport POTS+ISDN keeps you ready for both. ISDN mode transmits audio using powerful MPEG4 AAC-LD coding for high quality audio with minimal delay. Available with "S" interface for international use, or "S+U" interface for use in USA.

• XPORT Q&A

- *What is Zephyr Xport? A miniature Zephyr Xstream?*
  No, Zephyr Xport is the field side of a system that has Zephyr Xstream at the studio. Because the studio side is connected digitally with ISDN, modem performance is considerably more reliable than with POTS-only schemes.

- *But how can a POTS codec place a call to a codec on an ISDN phone line? How do they communicate?*
  We developed custom modem technology for Zephyr Xport that takes advantage of the telephone network's ability to cross-connect analog and digital phone lines (Circuit-Switched Voice technology) — the same principle Telos uses in our TWOx12 and Series 2101 ISDN talkshow systems. Xport is for the field, and uses POTS (Plain Old Telephone Service) analog lines. But you connect with a Zephyr Xstream at the studio using an ISDN line - the same one you probably already have.



The majority of the public switched telephone network is now completely digital. As the illustration above shows, a POTS call from a Zephyr Xport is converted from analog to digital at the nearest Central Office, where it continues on a digital path all the way to the studio.

- *I read about a new coding method you're using in Zephyr Xport. Why not use MPEG Layer 3?*

  Layer 3 is perfect for ISDN bit rates, but less suitable for bit rates encountered with modem transmission over analog phone lines. Zephyr Xport uses the highest fidelity low-bitrate coding method on Earth: Coding Technologies' aacPlus, the combination of MPEG's AAC (Advanced Audio Coding) and SBR (Spectral Band Replication). This method improves the efficiency of the codec by 30% over "plain" AAC – which, itself, is 30% more efficient than Layer 3.

- *Okay, the technology is cool, but how does it sound?*

  It sounds fantastic! Companies like XM Satellite Radio employ aacPlus to achieve superior results with digitally transmitted audio. Our implementation of aacPlus has been specially tuned and optimized for transmission of audio at very low bit rates, resulting in FM-like audio over analog telephone lines, with detailed highs and fuzz-free clarity for both speech and music.

- *I use my Zephyr Xstream to do ISDN remotes, and they sound great. What advantage does Zephyr Xport give me that I don't already have?*

  Using ISDN makes for great sounding remotes, but you can't always get ISDN where and when you want it. With Zephyr Xport, you can use a standard analog phone line in the field to connect with the Zephyr Xstream ISDN codec in your studio. This capability gives you much more freedom to choose when and where you can do remote broadcasts, without sacrificing fidelity or stability. You'll also enjoy significantly more stable connections than with ordinary POTS codecs, due to the ISDN link at the studio end of the call.

- *Some POTS codecs re-train a lot when line quality drops. Does your POTS-to-ISDN method eliminate this?*

  No transmission method that relies even a little on an analog signal path can completely eliminate re-training. However, our custom DSP-based modem is optimized for stability and audio (rather than data) transmission. This, along with the ISDN connection at the studio end, results in rock-solid connections that POTS codecs restricted to analog on the studio side can't provide. Even if you do have to re-train, Zephyr Xport lets you decide when to do so, avoiding the unexpected dead air that can occur with other POTS codecs.

- *What happens if line quality degrades?*

  An on-screen connection quality display constantly informs you of line conditions. If you need to re-train, a message alerts the operator; a simple button push during a break completes the process. There's also a built-in telephone hybrid; if line performance drops dramatically, the system automatically switches to this mode so that your broadcast can continue without interruption or audio loss.

CONNECT

USE A STANDARD POTS LINE TO TO ANY ZEPHYR XSTREAM ISDN CODEC

• *How hard is it to operate Zephyr Xport?*
We understand that sometimes remote gear is operated by non-technical users, so Zephyr Xport is designed to be very easy to use, with a friendly graphical display and online help that makes Xport as easy to operate as a cell phone. There's one-touch access to presets and space to store up to 100 auto-dial numbers and 30 locations (to recall settings for your most frequently visited remote sites).

• *It sure would be convenient to have a field unit like Zephyr Xport that could connect over POTS or ISDN.*
Actually, Zephyr Xport is available with an optional ISDN interface so you can connect to your studio over ISDN or POTS, depending on your needs. Zephyr Xport's ISDN option let you use Low Delay MPEG-4 AAC-LD to connect with your studio's Zephyr Xstream, providing Layer 3 audio quality with greatly reduced encoding delay.

• *Can I upgrade my POTS-only Xport to allow ISDN use?*
Yes. You can choose this option at the time of purchase, or easily install the ISDN option yourself at a later date.

• *Tell me about Xport's mixer section.*
There's a line input for external sources, a headphone level control, and a "mix" control that lets you blend send and receive audio for headphones or PA feeds. The mic input can be configured for use with most popular condenser mics using the built-in 12-volt Phantom power supply. The mixer's output is fed directly to the codec and includes selectable multi-band processing from Omnia; outputs include separate receive audio and monitor mixes for easy foldback setup.

• *Why do you include a multi-band AGC and limiter? Isn't this a little over the top?*
When sending audio with low bit rates, we want every practical tool to smooth and clarify the audio. That is why we went all out with a multi-band DSP approach — it really makes a difference to the quality. And we figure that during most of the time you're using a POTS codec, you 'll want some control over dynamics (think sports announcers). Of course, you can switch it off if you prefer a more "purist" approach.

• *Does Zephyr Xport include an Ethernet connection?*
Yes. Like Zephyr Xstream, Xport includes an Ethernet connector for remote control via LAN or WAN, and for one-button software updates. Xport's 10/100Base-T port also provides a direct input for PCM audio from a personal computer.

• *I can send audio from my laptop right into the mixer?*
Yes! A supplied driver lets Windows® (98 and up) see Zephyr Xport as a sound card, so you can send 48kHz PCM audio directly into the mixer from your computer — great for including actualities, pre-recorded interviews or drop-ins in a live remote or field report.

• *Does Zephyr Xport require an external power supply?*
No, Xport features an efficient, fanless internal power supply. All it needs is line cord suitable for your locale.

BUILT-IN MIXER MAKES IT PERFECT FOR GRAB-AND-GO REMOTES

About Audio Coding

What is coding and why is it required?

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. POTS offers reliable connections only at rates less than 30 kbps. That means you must do some coding to get "12 gallons of water into a one-gallon container."

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, these coding methods are not powerful enough to produce the significant reduction required.

To develop coding algorithms with sufficient power to achieve the desired reduction, the audio industry has turned to psychoacoustics. Using carefully researched psychoacoustic principles, processes referred to as "perceptual audio coding" 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, level, and 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 AAC & AAC-LD

(Advanced Audio Coding)

Developed by Fraunhofer Institute IIS (the inventors of MPEG Layer 3, a.k.a. MP3), the MPEG-2 AAC system is the newest audio coding method selected by MPEG. It became an International standard in April 1997, and 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 at bit rates as low as 16 kbps.

AAC is the first codec system to fulfill the ITU-R/EBU BS.1115 requirements for indistinguishable quality at 128 kbps/stereo. It has approximately 100% more coding power than MPEG 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.

Years of use in the broadcasting community have proven that Layer 3 offers all the fidelity needed in most broadcast situations. However, broadcasters also know that the delay of Layer 3 can be frustrating, particularly if high fidelity is needed in both directions.

Scientists at Fraunhofer were aware of these factors, and developed an extension to AAC called "Low Delay AAC" (AAC-LD for short). AAC-LD offers quality equivalent to Layer 3, with about 75% less delay! Zephyr Xstream is the first codec to implement AAC-LD, and we think you will find it very useful for high-quality interactive audio transmission.

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

About aacPlus

Highly efficient perceptual audio coders like MPEG Layer 3 or AAC have proven extremely effective in coding audio for transmission over ISDN and other digital data paths where reduced bit rates are necessary. But as powerful as these codecs are, even greater data reduction is needed to deliver high-quality audio when bandwidth is severely limited, as with POTS.

To address this issue, researchers at Coding Technologies (www.codingtechnologies.com) developed Spectral Band Replication. SBR can be used to extend existing coding methods, actually improving their coding efficiency. SBR added to MP3 is called MP3 Pro, and has become very popular in both professional and consumer audio applications.

When SBR is added to MPEG AAC, the result — called aacPlus — is, with little doubt, the best low-bitrate codec there is. By itself, MPEG AAC has been independently tested and found to be superior to any other coding at rates down to 16 kbps. The addition of Spectral Band Replication takes this already excellent performance to jaw-dropping amazing. Because of this, aacPlus has been selected by digital broadcasters such as XM Satellite Radio and Digital Radio Mondiale to deliver true CD-quality audio to their listeners.

The capability to deliver extremely high-quality audio using very low bit rates makes aacPlus ideally suited for POTS codecs. The version of aacPlus used in Zephyr Xport has been specially optimized for very low bitrates.

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

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.

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

ZEPHYR XPORT GENERAL SPECIFICATIONS

Full duplex, high-fidelity POTS (ISDN optional) field transceiver using aacPlus coding (and MPEG-4 AAC-LD with ISDN option); Digital Adaptive Hybrid fallback mode. Fully compliant with international standards. Monaural transmit and receive.

Frequency Response (+0/-3dB):	
aacPlus (POTS):	20Hz - 15kHz
MPEG-4 AAC-LD (ISDN):	20Hz - 15kHz
THD+N	
Mixer:	Loop Audio: 0.0043% @ 1kHz, 22Hz - 22kHz
Codec:	End-to-end connection at 24 kbps, aacPlus (line in to direct receive output), 0.04% @ 1kHz, 22Hz - 22kHz
Delay	
aacPlus (POTS):	<700 ms.
MPEG-4 AAC-LD (ISDN):	< 90 ms.

ZEPHYR XPORT TRANSMISSION MODES

- ISDN: MPEG4 AAC-LD @ 15kHz
- POTS: aacPlus @ 15kHz
- POTS Hybrid Mode: 300 - 3.4kHz

MICROPHONE INPUT

Balanced XLR female

Clip Point:	
Gain High:	-37dBu
Gain Low:	-27dBu
Available Gain:	
Gain High:	50dB
Gain Low:	60dB
Impedance:	≥ 3.8K ohms
CMMR @ 1kHz:	75dB
Phantom Power:	12 volt @10mA

LINE INPUT

Balanced ¼" TRS

Level:	+4dBu (+20 clip)
Impedance:	≥ 19K ohms
CMMR @ 1kHz::	62dB

AUXILIARY INTERFACE

Unbalanced input/output, ¼" phone jack; Tip input, Ring output

Input:	
Level:	Unity Gain to mix out, 20dBu clip point
Impedance:	≥ 9k ohms
Output:	
Level:	-26dBu nominal
Impedance:	≤ 12k ohms

RECEIVE DIRECT OUTPUT

Balanced XLR male jack

Level:	+4dBu, 20dBu clip
Digital Operating Level:	-18dBfs, nominal
Impedance:	≤ 50 ohms
Noise Floor:	-75.5dBu (referenced to +4dBu, no decoder lock)
Dynamic Range:	No decoder lock: 94dB Encoder/Decoder loopback: 91dB
THD	(Line input, 100 Hz to 10 kHz @ +4dBu): 0.05%

HEADPHONE OUTPUT

Level (level control @ maximum and nominal input level at line input): ±103dBa SPL

MONITOR MIX OUTPUT

Floating, balanced ¼" TRS

Level:	+4dBu, 20dBu clip
Impedance:	≤ 50 ohms
Noise floor:	-85dBu (referenced to +4dBu, Mic & Line inputs off)
THD:	0.01% (Line input, 100 Hz to 10 kHz @ +4dBu)

LIMITER

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

POTS INTERFACE

6-Position/4-Pin miniature modular connector (RJ-11 style) with connections on the center pins (3 & 4). User-selectable configuration allows International operation.

ISDN INTERFACE (OPTIONAL)

Supports National ISDN-1, DMS Custom, 5ESS Custom PTP, ETS 300 (Euro ISDN), INS 64 (Japan) protocols. International version supports 4-wire ISDN "S" interface on an 8-position/8-pin miniature modular connector (RJ-45 style); USA & Canada version supports 4-wire ISDN "S" interface on an 8-position/8-pin miniature modular connector (RJ-45 style) and 2-wire ISDN "U" interface on a 6-position/4-pin connector (RJ-11 style).

LAN INTERFACE

10/100Base-T Ethernet port using RJ-45 connector. Supports TCP/IP (Telnet and FTP).

REMOTE CONTROL AND ANCILLARY DATA

Remote control supported using popular web browsers over 10/100 Base-T Ethernet port using RJ-45-style connector.

Bi-directional ancillary data: Serial connection at 9600bps; two contact closures.

SOFTWARE UPGRADES

Downloadable from Telos; uses FTP over 10/100Base-T Ethernet connection.

CONTROL PORTS

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

Inputs:	Closure to ground; integral 10k ohm pull-up to 5 volts. External pull-ups can be used to support higher voltages.
Outputs:	Open collector, sinks up to 250mA to ground.

RESOLUTION

Send Input:	24-bit.
Receive Output:	24-bit.

POWER SUPPLY

100-240 VAC (47/63 Hz) auto-configuring.
390 watts max
CE approved

DIMENSIONS

17 1/8" (43cm) wide
12 1/2" (32cm) deep
3 1/2" (9cm) high

SHIPPING WEIGHT

7.5 pounds (3.4kg)

MODELS AND ACCESSORIES

Please contact your Telos representative for information about Zephyr Xport models, pricing and accessories.

ZEPHYR XSTREAM GENERAL SPECIFICATIONS

Full duplex, high-fidelity ISDN transceiver using MPEG-2 AAC, MPEG-4 AAC-LD, 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
20 - 20kHz @ 48kHz fs (+0/-1dB, swept sine tone procedure)

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

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

ZEPHYR XSTREAM ISDN TRANSMISSION MODES

Stereo and Dual-Mono Modes Using Two ISDN "B" channels:
- MPEG-2 AAC Joint Stereo @ 20kHz or 15kHz for maximum fidelity.
- MPEG-2 AAC independent Stereo @ 20kHz or 15kHz.
- MPEG-4 AAC-LD Joint Stereo @ 14kHz for high quality with low delay
- MPEG-4 AAC-LD Independent Stereo @ 14kHz for high quality with low delay
- Layer 3 Joint Stereo @ 16 kHz or 15kHz for high fidelity and compatibility.
- Layer 3 Independent Stereo @ 16kHz or 15kHz for compatibility and surround-sound transmission.
- Layer 3 Indepenedent Dual-Mono allows each "B" channel to independently transmit and receive 15.8kHz.
- Layer 2 Joint Stereo @ 20kHz or 15kHz.
- Layer 2 Independent Stereo @ 7.8 or 9.8kHz.
- G.722 Mono-Dual allows each "B" channel to independently transmit and receive 7kHz.

Stereo Modes Using One ISDN "B" channel:
- MPEG-2 AAC Stereo 64 @ 7kHz or 10kHz.

Mono Modes Using One ISDN "B" channel:
- MPEG-2 AAC @ 20kHz or 15kHz for maximum fidelity.
- MPEG-4 AAC-LD @ 15.8kHz for high quality with low delay
- Layer 3 @ 20kHz or 15kHz for high fidelity and compatibility.
- Layer 2 @ 7.8kHz or 9.8kHz.
- Layer 2 Mono@ 8.6kHz (24kHz fs).
- G.722 @ 7kHz.

ISDN Telephone Mode:
- G.711 is used to call a standard POTS telephone for low-grade voice communications at 300Hz - 3.4kHz.

LINE BIT RATES
56 or 64kbps per channel, front panel selectable.

LIMITER
Zephyr Xstream:
Soft clipper prevents A/D converter overload without loss of dynamic range.

Zephyr Xstream MX and MXP only:
Internal DSP-based AGC/limiter with Omnia® audio processing. Includes presets for music & voice, selectable per channel.

SEND INPUT
Active balanced with RF protection.

Zephyr Xstream:

LINE:	-12 to +4dBu nominal level.
Clip point:	18dB above chosen nominal.
Bridging:	< 10K ohms (x2)

XLR female/quarter-inch TRS combo connector.

Zephyr Xstream MX and MXP:
XLR female/quarter-inch TRS combo connector.

LINE/MIC:	-12, +4/-55, -35dBu nominal level.
Clip point:	16dB above chosen nominal.
Bridging:	< 10K ohms (x2)

RECEIVE OUTPUT
XLR male, active differential

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

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. Supports "S" (4-wire) & "U" (2-wire) ISDN interfaces.

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

ISDN VOICE TELEPHONE MODE
2 channels using G.711 standard, µLaw or A-Law. 300–3,400Hz. DTMF 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 bps.
10Base-T Ethernet port using RJ-45-style connector.
Bi-directional ancillary data: Serial connection at 9600bps; eight contact closures.

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

Inputs:	Closure to ground; integral pull-up to 5 volts. External pull-ups can be used to support higher voltages.
Outputs:	Open collector, sinks up to 125mA 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)

MODELS AND ACCESSORIES
Please contact your Telos representative for information about Zephyr Xstream models, pricing and accessories.