Hx1

Hx2

Telos Hx1 & Hx2 Digital Hybrids

Sweeten and control caller audio better than ever before

Harnessing and taming the random nature of analog phone lines has never been easier. Consider the Hx1 and Hx2. Two digital hybrids that present superb digital telephone hybrid performance to broadcast, teleconferencing, and communications applications. Proven Telos processing technologies perform all hybrid functions, gain control, and filtering completely in the digital domain. This edge-of-the-art approach makes the Hx very affordable and assures consistently superior performance regardless of telephone line characteristics.

Fast, precise, and automatic digital nulling allows smooth, natural, simultaneous conversation. All without the speakerphone upcutting, voice distortion, and level matching problems of other hybrid-type interface devices. Hx hybrids ship standard with features you won't find in other POTS hybrids – Auto-Answer, caller disconnect detection, sophisticated audio-leveling and anti-feedback routines for enhanced open speaker applications, call screening and line-hold features, and front-panel send and receive audio metering and much more. Take a look at the specs below for full details.

Incoming Line Capacity	• (1) POTS Analog for Hx1 • (2) POTS Analog for Hx2
Hybrids	• (1) High Performance, Digital for Hx1 • (2) High Performance, Digital for Hx2
Audio Interfaces	• Analog • AES (optional)
Ancillary	• Single 9-pin D-Sub connector to remotely turn each hybrid ON/OFF and to return Ringing and On-Air status for each hybrid

Features

- Our best, most advanced hybrid algorithms.
- New symmetrical wide-range AGC and noise gate by Omnia, with adjustable gain settings.
- Studio adaption and pitch shifter help prevent feedback in situations where open speakers are required.
- Adjustable caller override improves performance and allows you to individualize the degree to which the announcer ducks the caller audio.
- DDEQ-Digital dynamic EQ™ keeps audio spectrally consistent from call to call.
- New EQ High and EQ Low display meters for each hybrid.
- Separate Send level and Receive level meters for each hybrid.
- Status Symbols™ make life easier for producers and talent with their animated high-contrast icon display of line status.
- Place caller on-hold via front panel button.
- Auto-Answer with selectable ring count.
- Worldwide disconnect signal detection. (loop drop, dial tone, or reorder tone)
- Input/output via analog XLR. (Optional AES3 I/O module available)
- Input switchable between MIC or LINE levels.

Analog Telephone Line Connectivity

- Universal POTS interface for worldwide application.
- Auto-Answer with selectable ring count.
- Worldwide disconnect signal detection (loop drop, dial tone, or reorder tone).

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Hx1 back view

Hx2 back view

Processing Functions

General
- Telos 3rd-generation Adaptive Digital Hybrids
- Telos Exclusive Feedback Reduction Functions, including Acoustic Echo Cancellation

Send (to caller) Processing
- Sample Rate Conversion
- High-pass Filter
- Frequency Shifter
- AGC/Limiter

Receive (from caller) Processing
- High-pass 'Hum' Filter
- Smart AGC / Platform Leveler
- Noise Gate
- Caller Ducking
- Telos' DDEQ (Digital Dynamic Equalization) 3-band Adaptive Spectral Processor
- Sample Rate Conversion

Analog Inputs

Send Analog Inputs
- 1 for Hx1
- 2 for Hx2

Connector	XLR Female, Pin 2 High (Active Balanced with RF Protection)
Input Range	Select between MIC and LINE levels
Input Level	Adjustable from -10 to +4 dBu (nominal)
Impedance	Bridging > 50 KOhms
Analog Clip Point	+21 dBu
Analog-to-Digital Converter Res	24 bits

Analog Outputs

Receive Analog Outputs
- 1 for Hx1
- 2 for Hx2

Connector	XLR Male, Pin 3 High
Output Level	Nominal at +4 dBu
Impedance	< 50 Ohms
Digital-to-Analog Converter Res	24 bits
Headroom Before Clipping	20 dB headroom from 4 dBu nominal levels

AES Digital Input / Output (Optional)

Overview	Plug-in module converts the XLR inputs and outputs to AES3 (one input or output on Left channel of AES stream)
Rate Conversion	Sample Rate Converters on all inputs and outputs. Inputs can accept 32, 44.1, and 48 KHz rates. Clock for outputs may be sourced from the AES inputs or internally-generated 48 KHz
Input Level	Nominal at -20 dBFs
Output Level	Nominal at -20 dBFs

Audio Performance

Frequency Response	200 to 3400 Hz, +/- 1 dB
THD+N/Input	< 0.5% THD+N using 1 KHz sinewave
Signal to Noise	> 90 dB

Control Ports

General purpose Input/Output	Single 9 pin D-Sub connector with 2 status outputs (Ringing and ON-AIR) and 2 control inputs (ON and OFF) per hybrid

Physical

Dimensions	Standard rack mount, one rack unit high. 12 1/4" deep (31.1cm)
Weight	6 pounds (11 pounds shipping weight) 2.7 kilograms (5.0 kilograms shipping weight)

Questions

What is the advantage digital signal processing brings to telephone interface equipment?

When Telos first invented the digital adaptive telephone hybrid in the mid-1980s, the primary advantage was vastly improved "trans-hybrid loss". The Hx uses the most advanced DSP not only to produce superior trans-hybrid loss, but to provide consistent caller level and tone and address the problem of feedback when callers are monitored through open speakers.

What are the problems caused by poor trans-hybrid loss?

Trans-hybrid loss is the announcer's voice signal (or send audio) that leaks through the hybrid to its output. Ideally, the output should consist of the caller audio only. The reason is that, in a broadcast studio, the announcer audio is mixed at the console with the hybrid (caller) output to create the "on-air" mix. When you use a poor hybrid, its output includes a distorted, phase shifted version of the announcer signal. When this "leakage" is combined with the clean announcer audio, a "hollow" or "tinny" sound is produced as some frequencies are more effected by phase cancellation than others. One way to evaluate hybrid performance is to listen to the announcer's voice. If announcer voice deteriorates when the hybrid is turned on, there is excessive leakage and inferior trans-hybrid loss. In systems using multiple hybrids to conference several callers, poor trans-hybrid loss will cause a serious "singing" feedback, especially on low-level callers.

These problems result from the nature of phone lines, right?

You've got it. Hybrids must deal with complex and erratic phone line impedance characteristics across the phone line's frequency range. Impedance variations are caused by nearly every piece of equipment and run of cable between your studio and the callers telephone. To cancel the send audio, primitive analog hybrids use simple resistor-capacitor "balancing networks" to attempt to match the impedance of the phone line. It is a rare phone line that has a smooth, unvarying characteristic, so analog hybrids are often hopelessly ineffective.

How does Telos' digital processing hybrid work?

Telos digital hybrids use a very advanced time domain convolutional adaptive filter algorithm to synthesize a transfer function for the balancing network. A feedback loop continuously adjusts the filter to compensate for changing line impedances. In the Telos Hx, a gradient search technique is used to minimize the discrepancies between the synthesized transfer function and the telephone line's transfer function. The result is a very close match to the phone line impedance curve for optimum rejection.

You said that the Hx includes more than a digital adaptive hybrid. What else have you got in there?

Our full digital approach provides very smart, automatic dynamics control on both the announcer send signal and the caller signal. The input gain section compensates for widely varying levels without bringing up noise. The output gain section is cross-coupled to the input section so that it will not compress up hybrid leakage. And the downward expander subtly reduces phone line noise while distinguishing and passing low level callers. When used in combination with our DDEQ digital dynamic equalizer, this advanced gain control system produces caller audio that is clear and undistorted.

Why is this dynamics control so critical?

Levels from caller to caller can vary as much as 30 dB. The smart response of our gain control uses the least amount of processing required so that the natural characteristics of the caller's voice are retained. To achieve this end, the Hx has a digital, logarithmic (dB linear) compressor that uses a feed-forward topology. This sophisticated compressor provides level-independent operation for outstanding performance regardless of caller level.

How do you address the problem of feedback when listening to callers through open speakers?

The Hx reduces the likelihood of feedback several ways. One cause of feedback is poor trans-hybrid loss and the excellent trans-hybrid loss of the Hx serves as the first line of defense. In addition, we use digital dynamics processing to intelligently and instantaneously determine appropriate gain values in the send and receive paths. The Hx also applies a subtle, inaudible digital downward pitch shift to the input audio before it is sent to the telephone line. The pitch shift further reduces the potential build-up of feedback.

Do you provide a way to control the balance between the talent and the caller?

We certainly do. While the Hx makes full duplex operation possible, programmers may want the announcer to "override" the caller by reducing the level of the caller audio slightly when the announcer speaks. The Hx duplex depth subsystem inserts a controlled loss, or ducking, into the inactive audio path. When the DUPLEX DEPTH control knob is turned to full-duplex, simultaneity of conversation is at its maximum. When the knob is turned towards half-duplex, the system inserts more loss.

In addition to giving the announcer more "presence" over callers, moving the control towards half duplex increases gain-before-feedback. DUPLEX DEPTH may be adjusted using two, remotely selectable controls. This enables easy switching between setups. For example, you may have one setup for live programming and another for off air recording of calls.