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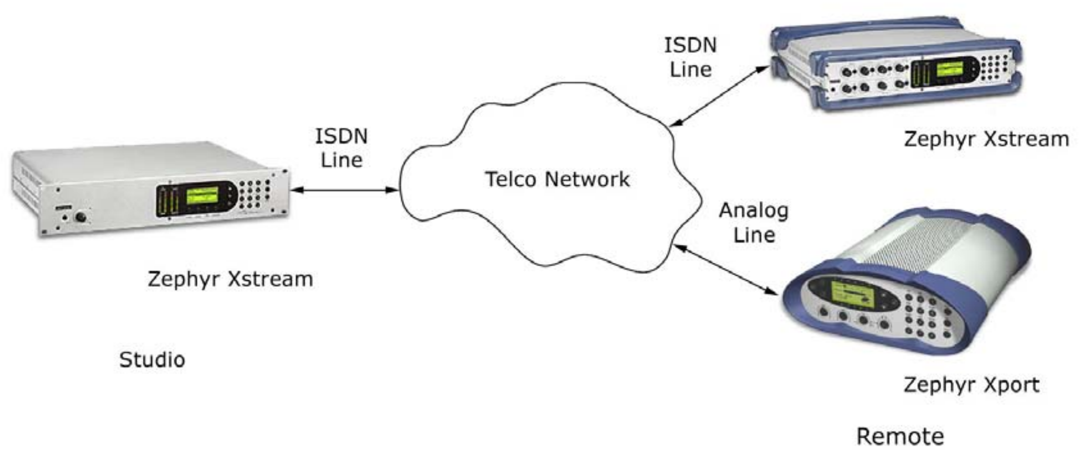
Omnia

Introducing the Zephyr Xport:
Ordinary Phone Line – Extraordinary Audio

ISDN makes for great sounding remotes, but you can't always get ISDN where and when you want it. Telos' new Xport lets you use an ordinary analog telephone line in the field to connect with the Zephyr Xstream ISDN codec in your studio.

The Xport features the highest fidelity low-bitrate coding method on Earth: aacPlus (MPEG AAC + the ground-breaking Spectral Band Replication enhancement). For the first time, you will experience FM-like audio over analog telephone lines with detailed highs and fuzz-free clarity – for both speech and music.

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.



To extract maximum reliability from real-world analog Telco lines, Xport has a custom DSP-based modem that lets us optimize for maximum performance with audio signals. Normal modems are designed for non-realtime data, where a bad packet may be re-transmitted without much consequence and "retraining" is not a major problem. With audio, these result in serious drop-outs. A modem for live audio requires a different set of trade-offs that are not possible with off-the-shelf consumer modem chips.

Your studio Zephyr Xstream becomes a universal codec, connecting with both Xport and ISDN codecs. This saves you money, rack space, operator training, telephone lines, and console/router audio inputs and mix-minus outputs.

The Xport's integrated mixer handles two inputs and includes a return mixer to combine local audio with the remote mix-minus feed. We've also included a multi-band automatic gain control and limiter designed by the Omnia processing gurus.

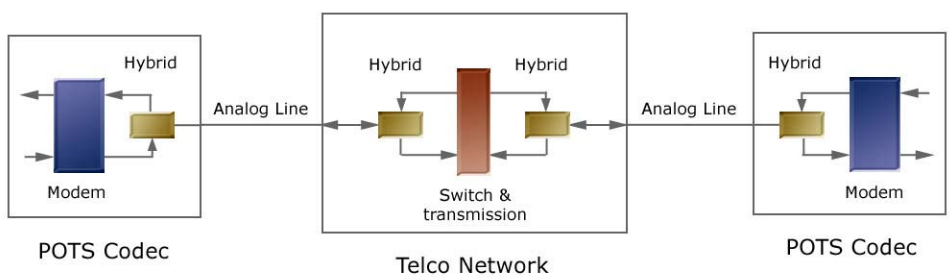
This was crafted to work in harmony with the audio codec, and is another reason audio is the smoothest, cleanest possible.

The Xport/Xstream combo is the best-sounding, easiest to use and most reliable analog Telco codec system ever offered to broadcasters. Because it lets you get double-duty out of your ISDN codec and line, it is also cost-effective.

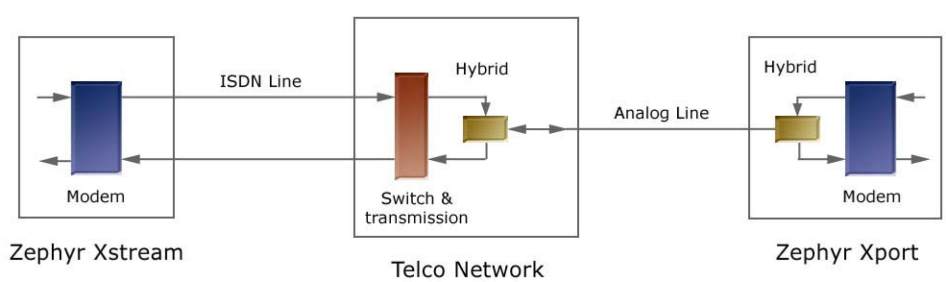
Q&A

Is Xport a "POTS codec"?
Yes and no. 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.

Are there advantages to using ISDN, in addition to being able to use the same Telco line and Zephyr codec equipment at the studio?
Yes, certainly. Modems work best on lines that are noise and distortion-free. They are also sensitive to the echo and noise inevitably caused by the hybrids and converters needed to adapt analog lines to the digital switching and transmission used within the Telco network. When you have a digital ISDN connection at the studio side, you get rid of a lot of the problem by removing two hybrids and eliminating any possible noise problems from half of the link.



Older POTS codecs confined to using analog at both ends require a total of 4 hybrids in the path, each of which can cause modem problems from reflections and from the analog/digital conversion noise that is added each time. Noise can be induced into two analog lines.



Zephyr opens a pure digital connection on the studio side, so half the trouble is eliminated.

The combination of ISDN + the special DSP-based modem means that Xport can cut through line problems that other codecs cannot.

Is there anything I need to know about using ISDN for this application?

Nothing special. The telephone network automatically translates the signals on the analog line to the digital form carried over ISDN.

How does the studio Zephyr know what kind of call is being received?

First, we look at the ISDN "set-up message," which tells us if the incoming call is from a digital device or from an analog phone line. Then we wait for the modem tone. Then we initiate Xport-specific handshaking to be sure.

Will it work for both local and long-distance calls? Anything special I need to do?

Within the USA and most other countries, all switching and transmission of telephone calls are done digitally, so performance should be nearly as good for long distance as for local. In both cases, only the "last mile" analog part should have any effect on modem performance.

What about international calls?

You should experience better quality and fewer drop-outs with the Xport approach compared to POTS-only codecs. Most international calls are switched and transmitted over digital links, so the only analog part will be the last mile at the remote site. On some connections, however, there may be some compression used to conserve bandwidth. The most common is ADPCM, which reduces the usual Telco 64 kbps rate to 32 kbps. There may also be "echo-suppressors" at some point within the connection. While the presence of modem tones is supposed to turn this off, we have noticed that sometimes this doesn't happen. If either of these occurs, you may want to try a different long-distance carrier to get a good connection. Note that these problems are possible with POTS-only connections also. Actually, ISDN makes them *less* probable because the Telco network is likely to assign better facilities to digital calls.

What happens when the line is very bad?

In this case, modem bitrate will be too low to be useful, and the Xport will switch to a non-modem "audio coupler" mode. Because we use digital hybrid technology borrowed from our telephone interface products, send/receive isolation is quite good. You will not get high-fidelity audio, but you will get the best possible.

What is the reasoning behind your choice of the aacPlus audio coding method?

It is, without much doubt, the best low-bitrate codec there is. MPEG AAC has been independently tested using a double-blind procedure and found to be superior to any other scheme at rates down to 16 kbps. aacPlus uses a powerful spectral band replication method to take this already excellent performance to jaw-dropping amazing. Because it is an enhancement designed specifically for very low bitrates, it is perfect for POTS codecs. (As a side note, SBR added to MP3 is called *MP3 Pro*, and is catching on quickly for Internet applications.)

aacPlus is being considered for inclusion within MPEG, and will probably be adopted within the next few months. XM Satellite Radio has announced that they will be using it.

Other POTS codecs use proprietary coding methods that have not been tested independently and are likely to be very much worse than aacPlus. Some use CELP, which is a voice-only codec – basically a scaled-up version of the codec in mobile phones. These don't work very well for music or for voice combined with background sounds such as from sports spectators, traffic, applause, etc. CELP does have the advantage of lower delay, but it comes at too much cost to audio quality, in our view.

So, what about that delay?

About 250ms – somewhat less than was our Zephyr classic stereo MP3 delay, but more than the mono delay. You cannot listen to yourself live with this much because the threshold for live monitoring is about 20ms. Talent must use a local mixer so that they hear themselves un-delayed. The Xport has such a mixer included. (This delay number is subject to change.)

Will other vendors be able to make products to interoperate with Xport and Xport-enabled Zephyr Xstreams?

If they want to, yes. As is Telos tradition, we prefer standards that do not lock you in. Remember the early ISDN codec days? In the face of repeated attempts to get users to take up proprietary schemes, we held fast to standards. And we sought the very best at the time – MPEG Layer 3. We were, in fact, the first in the world to license and offer what has come to be known as MP3, a decision well vindicated over time. When needed, we have worked co-operatively with competitors to help them achieve interoperability. Xport continues this tradition and approach. The aacPlus codec is licensable by all who choose it. We hope it catches on widely, because it really *is* very good.

Why did you wait so long to make an analog line codec? Seems like a natural extension of your Zephyr, and you guys *do* know coding, right?

Maybe it was *because* we know coding that we have been on the sidelines until now. We just hadn't heard a codec that sounded *broadcast quality* to our ears before this one. Those with long memories may recall that we demo-ed a POTS codec prototype using MP3 at an NAB radio show many years ago – before any others were marketed. We have been thinking and planning this step for some time.

Also, we didn't want to use an off-the-shelf modem owing to the problems with those, and we wanted to use ISDN on the studio side, so that had to wait for the companion Xstream. The pieces finally came together to let us give you all the right stuff.

Can Xport use an ISDN line as well as a POTS line?
Yes, there is an ISDN upgrade for Xport that can be ordered from the factory or field-installed. With this, Xport can be used with whatever phone line is readily available at your remote location.

Is Xport shipping now?
Yes!

Does an Xstream with Xport capability cost more than an ISDN-only unit?
No, the Xport feature is standard and included at no additional charge.

Can I upgrade my existing Xstream codec? Will the upgrade be free?
Of course! Telos has had a long history of enabling new applications with software enhancements to products. Because the Xstream has an Ethernet port, it may be upgraded over the Internet, without changing any parts or sending it to Telos. You will be able to do this for free. Or you may choose to upgrade by replacing a memory SIMM or by sending the unit to us. In these cases, there will be a modest charge.

Will you ever have some kind of universal box for the remote side that does ISDN and analog?
Yes. There will be an option for the Zephyr Xstream that will give it the same analog line capability as the Xport. Because this includes an additional plug-in board, there will be a cost for that upgrade. You will then have a one-box field solution that serves both your ISDN and analog needs.

Tell me more about the Xport's mixer.
There are two inputs and one output. One of the inputs is for mics and the other is for line-level sources. The output lets you mix local audio with the studio mix-minus to prevent problems with talent hearing themselves delayed. Normally, this goes to headphones. Because the send and receive paths are fully isolated, the receive channel may also be used for cuing information from the studio. The controls can push-in to protect from damage during transport.

Anything interesting 'round back?

Have a look…

There are two modular jacks, one for the Telco line and the other for a phone set.

These days, field work usually includes a Laptop PC. We believe that Ethernet will come to be regarded as an essential way to interface most broadcast gear to computers and networks, so all of our new equipment has it. The Xport will support configuration and control via a web browser, and a streaming audio connection so that you can play files from a PC into one of the mixer inputs.

The Interface connector can be used for parallel control, or optionally to connect with mobile phones that have RS-232 capability.

The Receive Direct Out is balanced line-level unmixed.

The Monitor Mix output is the same as the headphone output, but not affected by the front panel gain control.

The Aux Interface connector has audio send/receive on a TRS connector, and is intended for analog hook-up to today's mobile phones via the often-included headset jack. You can achieve some benefit over normal phone quality by using a professional microphone and the Omnia processing – a little better than standard phone audio, anyway.

Anything better possible with mobile phones?

Not yet. Current generation phones have a maximum data rate of 14.4 kbps, not enough for anything approaching high fidelity.

We expect to interface digitally to next-generation phones with fast rate capability. This will be via the Ethernet or Interface connectors, depending on what the phone makers bring us. Japan's I-Mode system, based on 3G Qualcomm technology, has a 64 kbps ISDN-like channel option today. If that catches on around the world, you will be ready to use your Xport for remotes with quality equal to landline ISDN codecs. Since the fixed end of I-Mode is via ISDN, your studio side Xstream will be ready. (This mode will not be possible with POTS codecs that don't use ISDN at the studio side because there would be no way to get the 64 kbps down an analog line.)

Why do you include a multi-band digital AGC and limiter? Maybe that is a little over-the-top?

Because they are under the same roof, the Telos and Omnia designers often share ideas and work together. Omnia has been making a streaming audio dynamics processor for some time, which was designed in partnership with the Telos codec guys to sound good with coded audio. So it really was natural to get some of this Omnia work back into Telos products – hence the integration into Xport. The bitrate is very low, and we want every practical tool to smooth and clarify the audio. That is why we went “whole hog” with a multi-band DSP approach – it really makes a difference to the quality. And we figure most of the time you are using a POTS codec, you want some control over dynamics anyway. (Think sports announcers.) Of course, you can switch it off if you prefer a more purist approach.

Do you have more information available?

Yes, on our web site at www.telos-systems.com.