*Telos*

# *ZephyrExpress*

**Portable ISDN
Digital Audio Transceiver / Mixer**

*The Best Way to Hear from There*™

# User's Manual

**Manual Version: 1.0 / April, 1999**

(v1.2.1 software)

# What you've got here:

ZephyrExpress is a portable, one-box solution for high-quality broadcast audio remotes via ISDN. It combines a flexible stereo digital mixer, two stereo monitor mixers, coder and decoder for international-standard audio data compression, and an ISDN interface.

- You can plug microphones and a headphone in one end, an ISDN line in the other, and you're ready for instant two-way 20kHz stereo transmission virtually anywhere in the world.

ZephyrExpress communicates with Telos' Zephyr or another ZephyrExpress, or with any other brand of codec using MPEG Layer 2, MPEG Layer 3, or G.722 algorithms. ISDN and audio setups can be pre-programmed for one-button connection. ZephyrExpress can even call standard telephones via ISDN, eliminating the need for a separate circuit and phone at the remote location.

(ISDN, by the way, is a telephone company technique for stuffing two bi-directional 64kpbs channels on the same copper pair that was originally designed for a single voice circuit. In many cases it can use existing wiring. ISDN requires special terminators and synchronized terminal adapters, built in to ZephyrExpress.)

- Actually, what you've got here is the Users' Manual. It'll tell you everything you need to operate ZephyrExpress, how to order ISDN circuits and connect to them, how audio coding work and how to choose its options, how to maintain ZephyrExpress, and lots of tips and shortcuts for using the system efficiently.

- There's also a 40-page Mini-Manual with abbreviated operating instructions and common questions about using ZephyrExpress. It's designed to be kept with the unit and taken to the remote broadcast.

- And there's a 4-page laminated Field Guide, also intended to be kept with ZephyrExpress.

## Customer Service

We support you...

- **By phone/Fax in the USA.**

Customer service is available from 9:30 AM to 6:00 PM USA Eastern Time, Monday through Friday at +1 216.241.7225. We're often here at times outside of these, as well – please feel free to try at any time!

Fax: +1 216.241.4103.

- **By phone/Fax in Europe.**

Service is available from Telos Europe in Germany at +49 81 61 42 467.

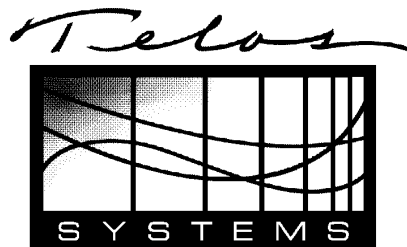Fax: +49 81 61 42 402.

- **By E-Mail.**

The address is: support@zephyr.com.

- **Via World Wide Web.**

The Telos Web site has a variety of information which may be useful for product selection and locating other compatible users. The URL is: http://www.zephyr.com.

## Feedback

We welcome feedback on any aspect of ZephyrExpress or this manual. In the past, many good ideas from users have made their way into software revisions or new products. Please contact us with your comments.



| Telos Systems | Telos Europe |
|---:|:---|
| 2101 Superior Avenue | JohannisstraBe 6 |
| Cleveland, OH 44114 | 85354 Freising |
| USA | Germany |
| +1 216.241.7225 | +49 81 61 42 467 |
| Fax: +1 216.241.4103 | Fax: +49 81 61 42 402 |

## Blank Screen?

If ZephyrExpress' LEDs light when you first apply power but the LCD screen appears dark, or the LCD screen lights but no characters appear, the last user may have left its brightness or contrast at an extreme setting. To reset:

> **DROP**
>
> • Press the DROP button and hold that button in while turning the Edit knob clockwise to reset the backlighting. It may take a few turns.
>
> **DIAL**
>
> • Press the DIAL button and hold that button in while turning the Edit knob either direction to change the LCD Contrast.

## Test Lines

Telos Systems maintains ISDN test lines with continuously-playing music. Feel free to dial into them from your ZephyrExpress to test your setup or line.

- USA: +216 781.9310, +216 781.9311 (Layer 3 Dual transmit @ 32 kHz sample rate. Dial just one number to test Layer 3 Mono, or dial both numbers to test Layer 3 Dual. You can also test Layer 3 Stereo by dialing both numbers; a slight distortion of the stereo image may result because you're receiving our Dual test signal rather than true stereo. Your ZephyrExpress will work correctly with a stereo transmission.)

- Germany: (49) 81 61 42 061 (Layer 3 Dual transmit @ 32 kHz sample rate; dial once to test Layer 3 Mono, or dial twice to test Layer 3 Dual. The warning about Layer 3 Stereo, above, applies.)

- Germany: (49) 81 61 42 062 (Layer 2 Mono @ 48 kHz; ZephyrExpress must be set to Layer 2 receive. Only one line is required.)

> **HOT TIP:**
>
> ZephyrExpress will dial a test line for you:
>
> 1) Verify that the word `Ready` appears twice on the LCD screen. If it doesn't, you have an ISDN problem and can't place a call. ISDN setup instructions appear in this booklet.
> 2) Press the DIAL button
> 3) Turn the EDIT knob to select `Auto...` and tap the EDIT knob in toward the panel to confirm that choice
> 4) Turn the EDIT knob to select a test line and tap the knob again
> 5) The word `Go!` will be highlighted on the screen. Tap the knob one more time, and ZephyrExpress will place the call.

## Updates

How ZephyrExpress works is almost entirely determined by software. This booklet is based on software version 1.2.1.

## Power Supply

ZephyrExpress is powered by a separate modular power supply, which is self-adjusting to any voltage between 100 - 250 VAC, 50 - 60 Hz. It connects to a 4-pin XLR connector on the rear panel. The supply must be turned off at its switch, or disconnected from the AC line, before plugging or unplugging this XLR connector.

**WARNING:**

Do not use the XLR connector as an on/off switch!

Connecting or disconnecting the power supply while it is powered can erase current ISDN setup and other user settings.

## Trademarks

Telos Systems, the Telos logo, Zephyr, ZephyrExpress, and "The Best Way to Hear from There" are trademarks of TLS Corporation. All other trademarks are property of their respective holders.

## Copyright

Copyright © 1999 by TLS Corporation. Published by Telos Systems, who reserves the right to make improvements or changes in the products described in this manual, which may affect the product specifications, or to revise the manual without notice. All rights reserved.

## Notice

All versions, claims of compatibility, trademarks, etc. of hardware and software products not made by Telos mentioned in this manual or accompanying material are informational only. Telos Systems makes no endorsement of any particular product for any purpose, nor claims any responsibility for operation or accuracy.

## Warranty

This product is covered by a one year limited warranty, the full text of which is in the Appendix section.

## Repairs

You must contact Telos before returning any equipment for repair. Telos Systems will issue a Return Authorization number **which must be written on the exterior of your package.** Be sure to adequately insure your shipment. Packages without proper authorization may be refused. US customers should contact Telos customer support at +1 216.241.7225. All other customers should contact their local Telos Dealer who will verify the problem and will contact Telos and arrange for repair.

**CAUTION:** The installation and servicing instructions in this manual are for use by qualified personnel only. To avoid electric shock, do not perform any servicing other than that contained in the operating instructions unless you are qualified to do so. Refer all servicing to qualified personnel.

**WARNING:** To reduce the risk of electrical shock, do not expose this product to rain or moisture. Avoid underwater remote broadcasts without adequate protection.

This symbol, wherever it appears, alerts you to the presence of uninsulated, dangerous voltage inside the enclosure — voltage which may be sufficient to constitute a risk of shock.

This symbol, wherever it appears, alerts you to important operating and maintenance instructions. Read the manual.

### CAUTION
**TO PREVENT RISKS OF ELECTRIC SHOCK, DISCONNECT POWER CORD BEFORE SERVICING.**

**USA CLASS A COMPUTING DEVICE INFORMATION TO USER. WARNING:** This equipment generates, uses, and can radiate radio-frequency energy. If it is not installed and used as directed by this manual, it may cause interference to radio communication. This equipment complies with the limits for a Class A computing device, as specified by FCC Rules, Part 15, Subpart J, which are designed to provide reasonable protection against such interference when this type of equipment is operated in a commercial environment. Operation of this equipment in a residential area is likely to cause interference. If it does, the user will be required to eliminate the interference at the user's expense. **NOTE:** Objectionable interference to TV or radio reception can occur if other devices are connected to this device without the use of shielded interconnect cables. FCC rules require the use of only shielded cables.

**CANADA WARNING:** "This digital apparatus does not exceed the Class A limits for radio noise emissions set out in the Radio Interference Regulations of the Canadian Department of Communications." "Le present appareil numerique n'emet pas de bruits radioelectriques depassant les limites applicables aux appareils numeriques (de les Class A) prescrites dans le Reglement sur le brouillage radioelectrique edicte par le ministere des Communications du Canada."

# Table of Contents

More comprehensive tables of contents, with page numbers, are at the start of each section.
A complete index follows section 11.

# Section 7: ISDN Reference

## ISDN BASICS
Background
The Basic Rate Interface (BRI)

## HOW TO ORDER ISDN
ZephyrExpress ISDN Compatibility
Ordering: Central Office Switches and Protocols

# Section 8: System Functions

## Software Updates

## ZephyrExpress data ports
The Serial Port
The Parallel Port

# Section 9: Troubleshooting

## First Steps

## GENERAL
Thinking About Problem Solving
Diagnostic Aids
Some Error Conditions

## ISDN Problems

# Section 10: Schematics and Data Sheets

# Section 11: Appendix

## ABOUT AUDIO LEVELS

## CODEC COMPATIBILITY INFORMATION
Comprehensive ZephyrExpress Compatibility List

## FINDING PUBLIC ISDN SITES
List of Known Working SPIDs by Telephone Company

## ISDN BRI, ZEPHYR, AND YOU
Ordering ISDN

## The Faxable ISDN Order Form (for use in US only)

## A Note From the President...

Waaay back in 1984, Telos' first product was being designed on a Radio Shack TRS-80 and the attached modem was considered to be respectably state-of-the-art, operating at the impressive speed of 300 bits per second. This was nearly three times the speed of the 110 bps, twenty-five pound, phone company-issue boxes I had been using over at the local college campus to talk to the hulking IBM in the bomb-shelter basement.

The PC revolution had begun a few years back, so there were quite a few of them around, but they were, except for by grace of these modems and mainframes, islands. No one as yet had figured a way to link them up in any practical way.

How the world has changed! Politicians talking up the "Information Super Highway" have made the phrase a parody, and 10 Million bits per second LANS are starting to be thought of as kind of slow.

We radio broadcasters have until recently been only just a bit ahead of computer users a decade ago. Our stations mostly exist as islands, with what audio we get from elsewhere coming from the mainframe-like satellites and networks.

But now come the liberating technologies that do for us what networking is doing for computing: digital telephony and high-power audio data coding. These make possible the instant dial-up transportation of audio from and to anywhere in the modern world. Digital telephone interconnection is being delivered to us via ISDN, and ISO/MPEG Layer III is the perfect coding method to exploit it for high-fidelity audio.

ZephyrExpress is the next logical extension of our original high-quality audio network transceiver, the Zephyr.

With Zephyr, we tried to bring together gracefully these technological pieces to permit you to easily do that which was previously difficult or impossible. It is my hope that it becomes, in your hands, an empowering tool for the creation of a more interesting audio future. It's a result of what were once my personal passions, and what have now become what MBA types call our "core corporate competencies," Digital Signal Processing and telephones for broadcast. It feels as if everything we've done until now have lead to this. Plug it in, dial it, listen, and see if you, too, don't share the excitement we felt in the lab when we got the first prototype going, listening to a Zappa CD being played from our partner lab in Europe. It was absolutely mind-boggling – we were hearing CD-quality audio from the other side of the planet ... *on a phone line.*

God, I love this technology!

<div align="center">Steve Church</div>

# Notes About this Manual

ZephyrExpress and the studio Zephyr unit share a lot of features in common, particularly in how they code audio and are connected to ISDN lines. In general, things you already know or learn here about one unit will apply to the other.

We use the following symbols to draw your attention to particularly important points.

### DANGER!

This symbol calls attention to the presence of dangerous voltages inside ZephyrExpress.

### IMPORTANT!

This information is essential to getting ZephyrExpress to work, or to prevent damage to it. Read these notes to avoid major headaches.

### HOT TIP!

This information will probably come in handy at some point. It's a good idea to read these tips.

### ISDN TIP!

Important information about ISDN and ZephyrExpress' built-in Terminal Adapter. Recommended for all users.
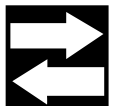
### COMPATIBILITY TIP!

Tips about operating modes, how to get the best results with other Telos units, and particularly how to connect ZephyrExpress to other brands of codec. You should read these.

### DEEP TECH NOTE!

Details about the intricacies of ZephyrExpress and ISDN and coding technologies. Useful information for troubleshooting, and fascinating reading for the technically-minded.

### CURIOSITY NOTE!

Amaze your friends with your grasp of ISDN and Coding trivia! Or just ignore these notes. It's up to you.

# Block Diagram

# Menu Structure

Utility Menu

**Codec**
Transmit
Receive
Bitrate
Sample Rate
Compatibility
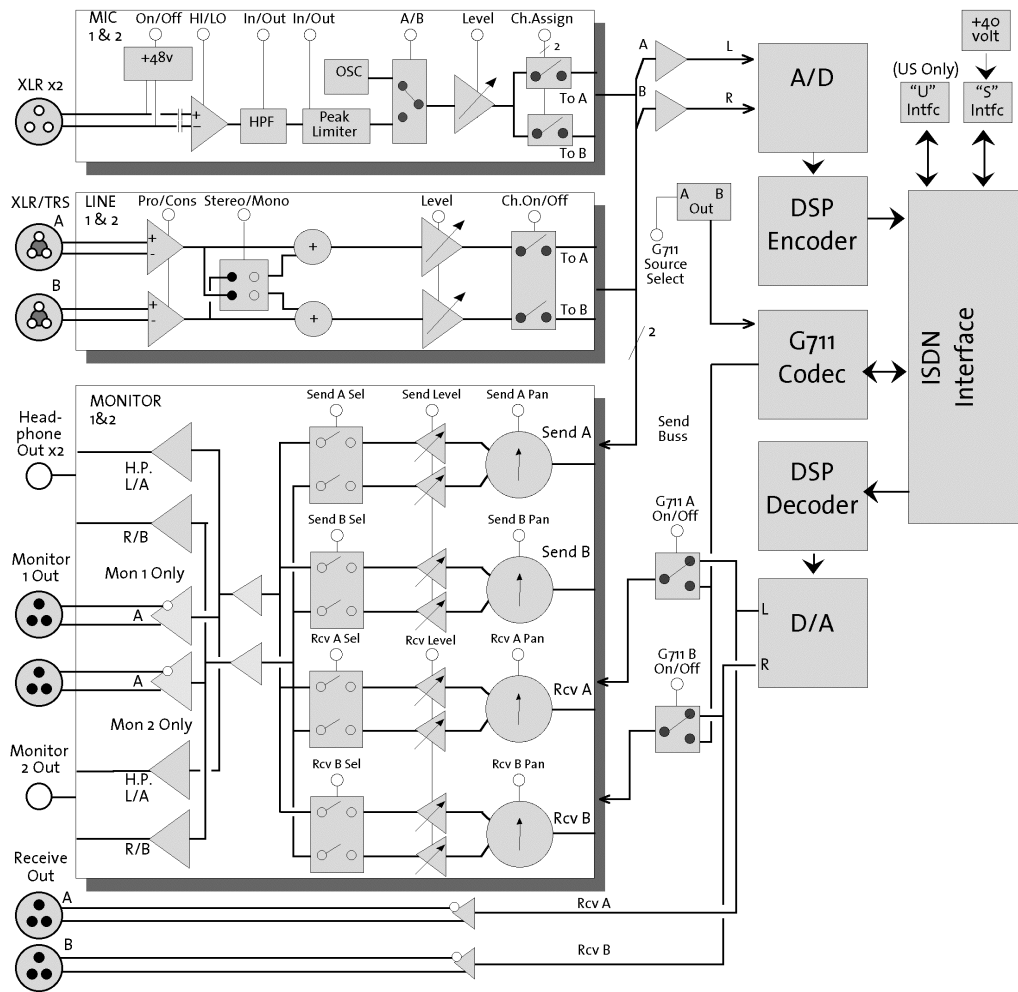Ancill Data
Status Out
Loop Mode
Contacts Out
Codec Setups

**ISDN**
Switch type
SPID #1
SPID #2
DN #1
DN #2
MSN #1
MSN #2
Outside Line
Panic Dial

**System**
LCD Contrast
LCD Backlight
Click Volume
POTS Volume
Menu Timeout
Baud Rate
Set Time
Software
About...

Audio

Microphones

**Mic 1**
Sensitivity
Phantom Power
Low Cut Filter
Test Tone

**Mic 1**
Sensitivity
Phantom Power
Low Cut Filter
Test Tone

Limiter Bypass

**Safe Mode**
Codec
Audio
ISDN
System
Channel Assign
Manual Dial
Autodial

Line Sensitivity

**Mon 1 Xmt Pan**
A Setting
B Setting
AB setting A
AB setting B

**Mon 1 Rcv Pan**
A Setting
B Setting
AB setting A
AB setting B

**Mon 2 Xmt Pan**
A Setting
B Setting
AB setting A
AB setting B

**Mon 2 Rcv Pan**
A Setting
B Setting
AB setting A
AB setting B

Master

Autodial

# SECTION 1:

# QUICK RESULTS

This part of the manual will get you started quickly. It presents answers to the questions that come up often when using ZephyrExpress. It's not a thorough technical guide — that's what the rest of this book is about — but a way to get you through most operating situations.

If you need to put the unit in service immediately, or have experience with Zephyr or other similar ISDN codecs, this section may be all you need. It's in two parts:

- Step-by-step instructions for configuring ZephyrExpress and dialing our test site. You'll be able to hear music in high-quality stereo as well as verify that ZephyrExpress and the ISDN line are working properly.

- Answers to common questions about using ZephyrExpress in the field. This material can also be found in our Mini-Manual.
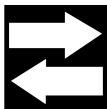
> FAST TRACK!
>
> If you already know how to use a Zephyr, we recommend you read through the next five or six pages. ZephyrExpress' controls and menus work differently, and the specific steps on these pages may be the fastest way to get acquainted with them... even if you're not interested in "Instant Gratification".

Don't ignore the rest of this book. Aside from the technical references, we've included interesting and useful information on audio coding and digital telephony. They'll help you get the best results from your ZephyrExpress, teach you about a technology that's becoming more important with the advent of digital broadcasting and Internet webcasts, and provide you with tons of tidbits you can use to impress others.

## Part I:
# Instant Gratification, or
# "How to hear stereo music over a phone line"

This section is for you if you:

- Know how to dial a telephone call.

- Are not intimidated looking at the front panel of ZephyrExpress.

- Have Basic ISDN[1], a switched digital service provided by the telephone company. ISDN allows two simultaneous connections on a single circuit — like having two simultaneous analog calls on a single copper pair. The circuit may show up as a standard telephone jack, an 8-pin RJ-45 jack, or even a pair of screw terminals or copper wires. US users should contact their telephone company to verify that the line

---

[1] Also known as a BRI or a 2B+D line. ZephyrExpress is not designed to be used with PRI, Switched 56, or fractional T-1 lines.

was installed exactly as specified in the ISDN order instructions found at Telos' web site and at the end of this manual.

- Have circuit information supplied by the phone company. This must include the type of switch on the circuit and the telephone number(s)[1]. In the US, it will probably also include two SPID numbers. In Europe, it may include MSN numbers.

- Have a ZephyrExpress, a microphone, and a pair of stereo headphones or other audio monitor.

- Understand that these instructions are just to get you started and will not answer all of your questions or demonstrate all of the features of ZephyrExpress.

- Promise to read the full manual later.

Now, let's get started.

## Gather information about your ISDN line

Analog telephones are simple: all a phone needs to do is tap into a line, and it'll probably start working immediately.

ISDN is more complicated. Before any ISDN device can receive or transmit signals, it has to be matched to central office settings for that particular circuit. These settings include line identification numbers and a communication protocol.

Your ISDN circuit consists of two digital lines, and both can operate at the same time on a single copper pair. Each line has an identification number, which may (or may not) be its telephone number. Sometimes, both lines will use the same identification number.

The phone company's equipment expects ZephyrExpress — or any other ISDN terminal equipment — to know the right identification numbers for the circuit it's connected to, and to transmit them on demand. If the proper identification numbers aren't entered into the equipment, you won't be able to place or receive calls.

In North America, these numbers take the form of Service Profile Identification (SPID) numbers: If you're given SPIDs, they must be programmed into ZephyrExpress.

The two lines also have standard telephone numbers, which users of other ISDN equipment will dial to call your ZephyrExpress. Unlike regular telephone lines, both lines may have the same telephone number assigned to them.

European telephone systems don't use SPID. They may use optional Multiple Subscriber Numbers (MSNs) to limit which lines will be answered. If you put two MSNs into ZephyrExpress, the unit will answer calls only on those lines.

Confused? Here are some rules of thumb:

---

[1] There can be one or two standard phone numbers for a US circuit, and may be more than two for a European one.

1. If you're given two SPIDs by the phone company, enter just those numbers into ZephyrExpress. This is the most common case in North America.

2. If you're given just one SPID, don't enter it: instead, use the `PTP` setting for `Switch Type`.

3. If you're in Europe you don't have to enter anything. However, if you're given MSNs and want to limit which numbers ZephyrExpress will answer, enter them for those numbers.

There are also several different types of ISDN switching devices that can be used at the central office, each with a different set of protocols. You must know what type you have.
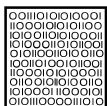
```
ISDN TIP!

If you have AT&T Point-to-Point protocol or European ISDN, the phone company
might provide just one number for the MSN or Directory Number.
```

These numbers, and possibly also the protocol, will be different for each ISDN circuit you connect to. Since ZephyrExpress is portable, it's likely to be used in multiple locations. You can enter the complete information for each location into ZephyrExpress, and recall its complete setting at the touch of a button.

---

*Here's a summary of circuit information. You may want to copy this form for each location.*
*Item 1 and 2 are required. Use just one set of numbers for item 3. Item 4 is optional.*

Location_____  Date _____

**1      ISDN type, check one:**

❒   National ISDN-1 (The most frequent choice in the US. Will always  have SPIDs. Also use this choice if your ISDN protocol is DMS Custom.)

❒   AT&T Point-to-Point or PTP. (Will not have SPIDs.)

❒   European ISDN (Euro-ISDN or ETS300. Will not have SPIDs.)

❒   Other _____ *(Contact Telos Customer Support.)*

**2      ISDN telephone numbers with area code:**

Line 1 _____      Line 2 _____

**3      SPID numbers, <u>if</u> supplied (US users only with National ISDN-1):**

Line 1 _____      Line 2 _____

*o r*   **MSN numbers, <u>if</u> supplied (Euro ETS300 only):**

Line 1 _____      Line 2 _____

**4**      Prefix required by internal phone system for outside lines: _____

---

# Set up the hardware

You'll need:

- A ZephyrExpress

- The ZephyrExpress Modular Power Supply with an appropriate power cord for your local AC supply. The supply is self-adjusting for any voltage between 100 - 250 VAC, 50 - 60 Hz. It connects to a 4-pin XLR connector on the rear panel. **The modular supply must be turned off at its switch, or disconnected from the AC line, before plugging or unplugging its XLR connector.**

  You may connect the power supply to ZephyrExpress now, and then apply AC power and turn it on. The screen and all the LEDs will light up. After a few seconds, you should see the word `Status` on the top of the screen. If the screen goes dark or totally blank, see page 1-15 of this chapter.

> **WARNING!**
>
> Do not use the XLR connector as an on/off switch!
>
> Connecting or disconnecting the power supply while it is powered can erase current ISDN setup and other user settings.

- An NT1 (also called a Network Termination Unit) with connection cables. All of the connection cables have telephone-type modular connectors and come packaged with your Zephyr. There are three ways your NT1 can be present:

  1. ZephyrExpress units sold in North America have a built-in NT1. If you see two modular-type telephone jacks on the back of your ZephyrExpress, the lower jack is the NT1. It's marked U and LINE DIRECT. It uses a standard analog telephone line cord (type RJ-11 or RJ-14).

     If you have this kind of NT1 and a North American ISDN installation, you may connect them now. If the telco provided a large 8-pin (RJ-45) jack, you should still use the standard line cord: plug it into the middle of the jack, and it'll find the right conductors.

  2. Most European and Asian telephone companies provide the NT1 as part of their installation. It uses a large eight-conductor telephone plug (RJ-45), the same as computer 10Base-T ethernet.

     If you have this kind of NT1 and a European ZephyrExpress with only one telephone jack on the back panel, you may connect them now.

You can tell if a ZephyrExpress will supply power by looking at the ISDN connections: if there are two telephone jacks, it's a North American unit and you must follow the above precaution. If there's only one telephone jack, it's a European unit and can be connected safely to powered NT1s.

European units are also marked with this Euro Telecom logo on the rear panel.

If you're using a North American ZephyrExpress with two telephone jacks, and a European or Asian NT1, **disconnect the NT1's power supply!** Once you have done this, you may connect an eight-conductor cable to the NT1. If you can't disconnect the NT1's power supply, contact Telos Customer Service before proceeding.

3. If you're using a European ZephyrExpress in North America, you'll need an external NT1. It's about the size of a small book, has some telephone-type connectors, and usually has some little indicator lights. You'll need two cables: The first is a large, telephone-type modular cable with eight wires that fits into the single jack on the Zephyr and a jack on the NT1 that is labeled "Terminal." The second is a telephone-type modular cable that fits into the jack on the NT1 that is labeled "Line" and the jack provided by the telephone company.

If you're using this kind of NT1, you may connect the two telephone cables now.

The external NT1 has its own power supply. **Do not use the external NT1's power supply if your ZephyrExpress has two telephone jacks on the back.** If your ZephyrExpress has only one telephone jack, you may connect the NT1's power supply now.

• You'll also need a microphone, a mic cable with an XLR male plug, and a stereo headphone with a standard 1/4" tip/ring/sleeve plug.

# Configure ZephyrExpress for your ISDN line.

HOT TIP!

There are two boxes on the bottom of ZephyrExpress' screen.

If you see the word `Ready` in both of them, your unit is probably already configured for the ISDN line. Skip this step, and go right to "Calling Yourself" on page 1-9.

If you see the word `inact` in both of them, your unit is not connected to a working ISDN circuit. Recheck your connections.

ISDN TIP!

You can check ISDN continuity with an ordinary analog telephone or a pair of 600Ω headphones. Disconnect ZephyrExpress, and briefly place the phone (or phones) across the line. You should hear either a rhythmic clicking — about once per second — or a loud, continuous white noise. If you hear a dial tone, it's not an ISDN circuit. If you don't hear anything, the line is dead.

ZephyrExpress is configured by pressing the UTIL button, using the EDIT knob to select menu items, and tapping the EDIT knob in towards the panel to confirm choices. The process is intuitive, and you should be able to configure your unit with a minimum of instructions. But if you get lost, see the instructions on page 1-16.

To configure ZephyrExpress for a particular line, first make sure you have the information about that line from page 1-4.

UTIL

Press the UTIL button.

The LCD display will change to look like this:

```
Utility Screen
  System Default*
Codec              →
Audio              →
ISDN               →
 ISDN settings submenu
```

*Lost already?*

If you don't see something that says `Utility Screen` at the top, don't panic. Just turn the EDIT knob counterclockwise until it appears.

If the second line of the screen says something other than `System Defaults`, don't worry.

If you don't see anything at all on the screen, but you know ZephyrExpress is getting power (because some LEDs are on):

> If the screen is dark, press the DROP button and hold it in while turning the EDIT knob clockwise to reset the backlighting. This may take a few turns.

> If the screen is lit, press the DIAL button and hold it in while turning the EDIT knob either direction to chance the LCD contrast.
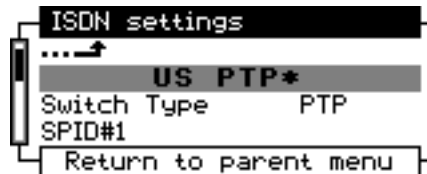
Turn the EDIT knob three clicks clockwise, to highlight ISDN.

Then tap the knob in toward the panel.

The display will change to look like this:

```
ISDN settings
...⤴
          US PTP*
Switch Type        PTP
SPID#1
     Return to parent menu
```

Turn the edit knob clockwise to highlight either SPID#1, DN#1, or MSN#1, depending on which type of number you entered in the form on page 1-4. The menu scrolls down as you keep turning the knob.

Tap the edit knob in toward the panel again to start entering that number.

The display will change to look something like this, depending on what kind of number you'll be entering.

```
ISDN settings
        US  Ntl.I-1*
Switch Type      Ntl.I-1
SPID#1
SPID#2
          Line 1 SPID
```
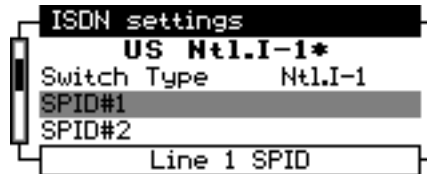
Use the numeric keys to enter the number.

Enter exactly the digits the phone company supplied you with. Leading and trailing zeros are important. If you make a mistake, you can turn the EDIT knob counterclockwise to back up.

Once the number is completely entered, tap the EDIT knob toward the panel to confirm.

Turn the knob one click clockwise to highlight SPID#2, DN#2, or MSN#2. Tap the knob, then enter that number the same way. When you're done entering the number, tap EDIT to confirm.

If this line requires a prefix to reach an outside line (item 4 on the form on page 1-4), turn the knob clockwise to highlight Outside line prefix. Then enter the prefix the same way you've entered the line identification numbers, and tap the knob to confirm.

Turn the EDIT knob counterclockwise to go back up to Switch Type, and tap it toward the panel. The screen will look like this. If you've entered a SPID#1, you should see it also.

```
ISDN settings
...⤴
          US PTP*
Switch Type    Ntl.I-1
SPID#1
        ISDN switch type
```
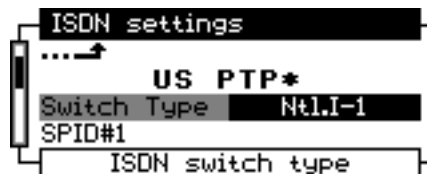
Turn the knob to select the switch type for this line, and tap to confirm.

> **HOT TIP!**
>
> The currently-selected switch type will have brackets around it (like `[PTP]`). If the current switch is correct for your line, *select a different type and tap the knob to confirm.* Then wait a few seconds, and select the proper one and confirm.
>
> This will force ZephyrExpress to resynchronize to the line using your circuit numbers.

Hold the edit knob in toward the panel for about five seconds. This will take you back up through the menu system, back to the `Status` screen. Then release the knob.

In a few seconds, you should see the word `Ready` in each of the boxes on the bottom of the screen.

- If you don't see `Ready` twice on the Status screen, go back and check to make sure you've entered all the numbers and the switch type correctly.

- If that doesn't work, turn to the Troubleshooting section of this manual.

# Call yourself (testing your unit and the line)

Before you call our test lines, make sure ZephyrExpress is working properly with the phone system. The easiest way to do this is to place a call on one of your ISDN lines, through the telco's central office, and back to yourself!

Verify that the word `Ready` appears twice on ZephyrExpress' screen. If it doesn't, go back through the previous few pages to configure the system for your ISDN line.

## Configure the Codec

Check the `Status` screen. If it shows `Transmit L3-DUAL` and `Receive L3-STEREO` like the picture below, you can skip the configuration steps and go to Place the Call on page 1-9.

```
Status    15.03.1999 17 28
Transmit           L3-DUAL
Receive          L3-STEREO
Rates        64kbps  32kHz
   Ready           Ready
```

If the `Transmit` or `Receive` modes are different from the picture above, follow the following steps:

**UTIL**  Press the UTIL button.

A screen like this will appear. If you don't see it, turn the EDIT knob counterclockwise.

```
┌─────────────────────────┐
│ Utility Screen          │
│    System Default*      │
│ Codec              ➞    │
│ Audio              ➞    │
│ ISDN               ➞    │
│  ISDN settings submenu  │
└─────────────────────────┘
```

Turn the EDIT knob clockwise, to select Codec.

Tap the knob towards the panel to confirm that choice. The display will change to look like this:

```
┌─────────────────────────┐
│ Codec Settings          │
│ ...➞                    │
│    [L3 stereo*]         │
│ Transmit      L3-STEREO │
│ Receive       L3-STEREO │
│  Activate codec setup   │
└─────────────────────────┘
```

Turn the knob one click clockwise, to select the first text line (L3 stereo* in the picture above, but the text may be different on your ZephyrExpress). Tap the knob in towards the panel to confirm.

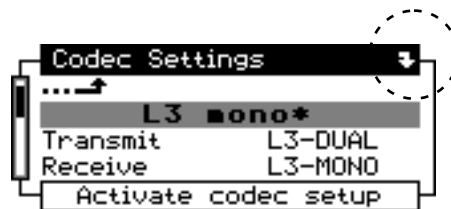Turn the knob to select L3 Stereo*. Tap to confirm.

┌────────────────────────────────────────────────────────────────────────┐
│ HOT TIP!                                                                 │
│                                                                          │
│ In the step above, make sure you are selecting L3 Stereo* in the centered line at │
│ the top of the screen.                                                   │
└────────────────────────────────────────────────────────────────────────┘

ZephyrExpress will show you a confirmation screen, like this:

```
┌─────────────────────────┐
│   Do you really want to │
│    activate this setup? │
│                         │
│   [ Yes ]      [ No ]   │
└─────────────────────────┘
```

Turn the knob to select Yes and tap to confirm. It may take about ten seconds for ZephyrExpress to change codec modes. During that time, you'll see a bent arrow, pointing down, in the upper right corner of the screen:

```
┌─────────────────────────┐
│ Codec Settings        ⬇ │
│ ...➞                    │
│    L3 mono*             │
│ Transmit      L3-DUAL   │
│ Receive       L3-MONO   │
│  Activate codec setup   │
└─────────────────────────┘
```

When the arrow disappears, hold the edit knob in towards the panel for a few seconds to return to the status screen. It should show Transmit L3-DUAL and Receive L3-STEREO, like the illustration at the top of this section. If it says L3-STEREO twice, that's okay.

## Set up an audio input and output

Plug the microphone into the MIC 1 INPUT jack on the back panel. Plug a stereo headphone into the MON 2 OUT headphone jack on the front panel.

Mic 1 Volume

Mic 1 Routing Switch

Monitor 2 Receive
routing switch

Monitor 2 Receive
volume

Monitor 2 Headphone Jack          Monitor 2 Send volume
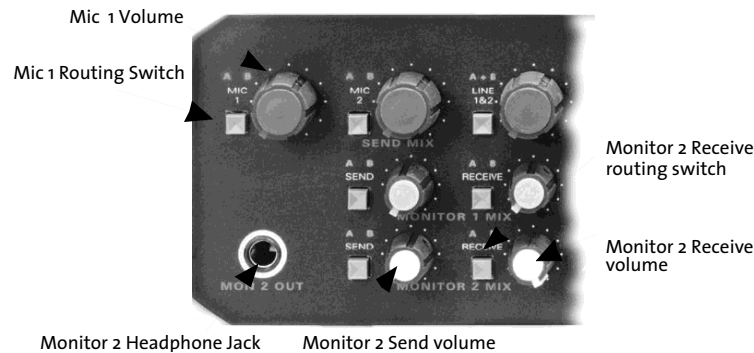
Tap the MIC 1 routing switch until the letters AB above it light up green.

Talk into the mic, and raise its volume control until you see a good level on the top two meters in the middle of the front panel. If you can't adjust the mic for a good level, or the letters AB are turning red on peaks, you'll have to change ZephyrExpress' input sensitivity[1]: see page 1-22.

Tap the MONITOR 2 RECEIVE routing switch until both letters A and B above it light up. You shouldn't hear anything in the headphones for now: if you do, make sure the MONITOR 2 SEND volume control is turned all the way down.

## Place the call to yourself

DIAL

Verify that you can see the word Ready twice in the Status screen. If you can't, check the configuration steps on page 1-7.

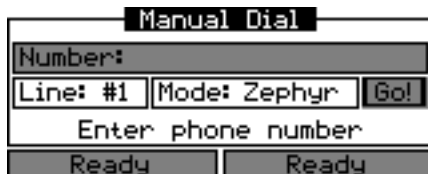Then press the DIAL button. A screen like this will appear:

```
   ▀▀▀▀▀▀ Select Dial Type ▀▀▀▀▀▀
   ┌────────┐  ┌───────┐  ┌─────────┐
   │Manual…│  │Auto…│  │Recent…│
   └────────┘  └───────┘  └─────────┘
   Press jog to enter number
   ┌──────────────┐  ┌──────────────┐
   │    Ready     │  │    Ready     │
   └──────────────┘  └──────────────┘
```

If the word Manual... is not already highlighted, turn the EDIT knob to select it.

Some screens prompt you to Press jog to enter data or confirm a choice. It means the same thing as tap the EDIT knob in toward the front panel… but we didn't have enough room to say all that.

---

[1] Or if you're eager to place your first call, leave the sensitivity alone for now — our audio circuits and codecs have a wide dynamic range — but be aware that any noise or distortion you hear during this test is because of that mismatch.
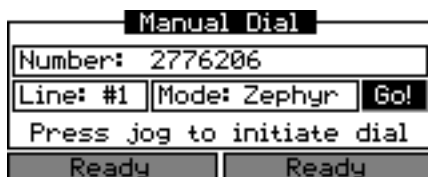
Now tap the EDIT knob toward the panel. The display will change to look like this:

```
┌───────■ Manual Dial ■───────┐
│┌───────────────────────────┐│
││Number:                    ││
│└───────────────────────────┘│
│┌──────────┐┌────────────┐┌───┐│
││Line: #1  ││Mode: Zephyr ││Go!││
│└──────────┘└────────────┘└───┘│
│      Enter  phone  number      │
│┌─────────────┐ ┌─────────────┐│
││   Ready     │ │    Ready    ││
│└─────────────┘ └─────────────┘│
└─────────────────────────────┘
```

Using the numeric keypad, enter the local number — usually without an area code — for line *2* of this ISDN circuit. You want to enter line 2, because you'll be calling it from line 1.

If you've placed a manually-dialed call since turning ZephyrExpress on, that number will already be entered. You can erase it, or delete any number you enter by mistake, by turning the EDIT knob counter-clockwise.

When the number is completely entered, tap the EDIT knob toward the panel. This confirms the number. The Go! box will now turn dark:

```
┌───────■ Manual Dial ■───────┐
│┌───────────────────────────┐│
││Number:  2776206           ││
│└───────────────────────────┘│
│┌──────────┐┌────────────┐┌───┐│
││Line: #1  ││Mode: Zephyr ││Go!││
│└──────────┘└────────────┘└───┘│
│   Press  jog  to  initiate  dial │
│┌─────────────┐ ┌─────────────┐│
││   Ready     │ │    Ready    ││
│└─────────────┘ └─────────────┘│
└─────────────────────────────┘
```

Tap the Edit knob to place the call.

In a few seconds, ZephyrExpress will beep twice to indicate the connection has been made. The two red LOCK lights towards the lower left of the display screen will light: if they don't, verify that the status screen shows Transmit L3-DUAL and Receive L3-STEREO (see page 1-9).

Talk into the microphone. You should see the signal, slightly delayed, on both lower volume meters. Turn up the MONITOR 2 RECEIVE volume until you hear the mic in the headphone.

> CURIOSITY NOTE
>
> While the signal you're hearing sounds almost as good as if the mic were connected directly to the headphones, you're actually listening to it after it's gone to the phone company's central office and been sent back to you. Both digital signals — sending and receiving — are multiplexed on the same copper ISDN pair.
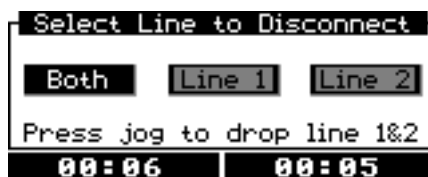>
> But the delay you're hearing is *not* because of the round trip. It's a result of the computer processing necessary to squeeze a high-fidelity signal into a very small amount of digital data.

ZephyrExpress includes features to minimize the effect of this delay during live broadcasts. These include low-delay modes for less critical signals and flexible monitoring to let talent hear their voices locally while receiving music and cues from the distant studio. You'll learn how to use these features later in this manual.

**DROP** When you're finished talking to yourself via ISDN, press the DROP button.

The screen will change to look like this:

```
┌──Select Line to Disconnect──┐
│                             │
│  ▆Both▆    │Line 1│  │Line 2││
│                             │
│ Press jog to drop line 1&2  │
├─────────────┬───────────────┤
│   00:06     │     00:05     │
└─────────────┴───────────────┘
```

Tap the EDIT knob to hang up. Now you're ready to *really* rock!

## Tune into one of our test lines

Our US test line originates less than a mile from the Rock 'n' Roll Hall of Fame, in Cleveland, Ohio. There's always something interesting playing through it.

We also maintain a European test line in Freising, Germany.
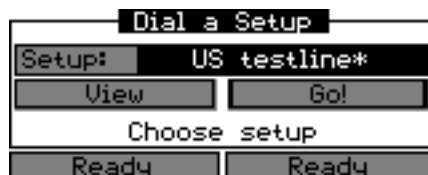
Here's how to connect to them:

**DIAL** Press the DIAL button.

A screen like this will appear:

```
┌──────Select Dial Type──────┐
│                            │
│ │Manual...│ ▆Auto...▆ │Recent...││
│                            │
│   Activate a master setup  │
├─────────────┬──────────────┤
│   Ready     │    Ready     │
└─────────────┴──────────────┘
```

If the word Auto is not highlighted, turn the EDIT knob until it is.

With Auto highlighted, tap the knob towards the  panel to confirm. The screen will change like this:

```
┌──────Dial a Setup──────┐
│                        │
│ │Setup:│  ▆US testline*▆│
│  │  View  │  │   Go!   │ │
│                        │
│     Choose setup       │
├──────────┬─────────────┤
│  Ready   │   Ready     │
└──────────┴─────────────┘
```

Turn the knob to select US testline* or Euro testline*. The asterisk means the number is pre-programmed at the factory. Tap the knob to confirm.

After you tap, the highlighting will switch to the Go! box. Tap again to dial.

If ZephyrExpress' codec is not already set to the correct mode to receive the testline, the unit will automatically change to match. This may take a few seconds.

Then ZephyrExpress will dial the test line. In a few seconds, you should see the red LOCK LEDs light, and music on the RECEIVE meters. If you see the words Busy or Conn once or twice on the status screen, try again later.

Press the small gray MONITOR 2 RECEIVE routing button until the letters A and B directly above it light up. This button is located along the bottom of the front panel, above the printed words MONITOR 2 MIX.
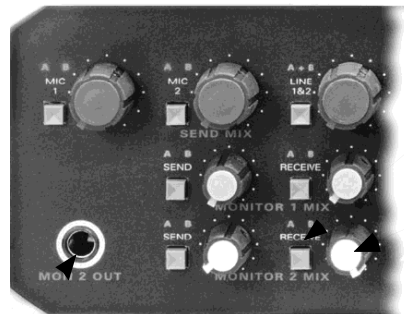


Monitor 2 Receive button

Monitor 2 volume
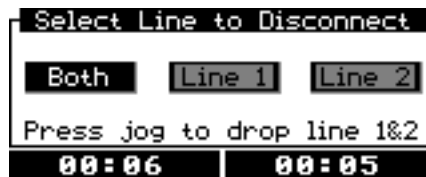
Monitor 2 Headphone Jack

Turn up the MONITOR 2 volume knob, and you should hear music through headphones plugged into the front panel headphone jack.

Enjoy.

When you're finished listening, press the DROP button.

You'll see a screen like this:



The word Both will already be highlighted. Tap the EDIT knob to drop the lines, and ZephyrExpress will hang up.

## Part II:
# Fast Answers to Frequently-Asked Questions

This section covers common operating issues. Chances are it'll tell you all you need to know about setting up ZephyrExpress and using it in the field. Scan through the topics now, so you'll know what you can find here.

Of course, the rest of this manual will tell you a lot more about ZephyrExpress, teach you how to use the system more efficiently, and turn you into an all-around ISDN and audio coding guru. Please don't ignore it.

> **HOT TIP!**
>
> This section also appears in the handy Mini-Manual that came with ZephyrExpress. We recommend you keep that booklet with the unit, and bring it along on remote broadcasts.

## Blank Screen?

If ZephyrExpress' LEDs light when you first apply power but the LCD screen appears dark, or the LCD screen lights but no characters appear, the last user may have left its brightness or contrast at an extreme setting. To reset:

**DROP**

> • Press the DROP button and hold that button in while turning the Edit knob clockwise to reset the backlighting. It may take a few turns.

**DIAL**

> • Press the DIAL button and hold that button in while turning the Edit knob either direction to change the LCD Contrast.

## The Basics

> **TURN or TAP**
>
> The EDIT knob can be turned to scroll through menus or enter choices. It can also be tapped in toward the front panel, just like a push button, to confirm a choice.
>
> When the Status screen is showing, turning or tapping the EDIT knob brings up the help screen.

### 1:  What do I do if I'm totally lost?

**HELP**
Press the HELP button on the right side of the front panel, near the Edit knob. Different Help screens will appear, depending on what you were doing when you pressed it.
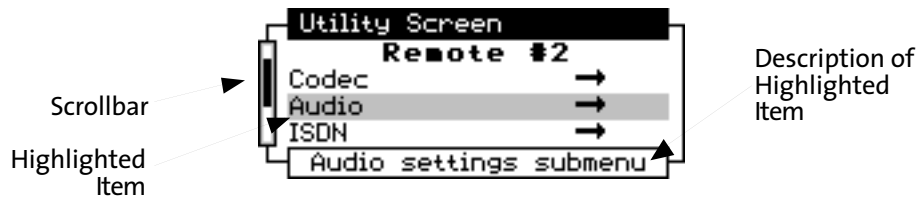
To exit Help, tap the EDIT knob in towards the panel and you'll return to the screen that was displayed when you pressed help.

To exit any other screen and return to the main Status screen, press and hold the EDIT knob in for at least three seconds.

## 2: How do I use the menus?

UTIL

ZephyrExpress uses a series of nested menus to set how it works. To open the menu system, press the UTIL button to the right of the Edit knob. The Utility Screen, with a list of submenus, will appear:

```
Utility Screen
     Remote #2
Codec          →
Audio          →
ISDN           →
 Audio settings submenu
```

Scrollbar →

Highlighted Item

Description of Highlighted Item

To access any menu item, turn the EDIT knob until that item is highlighted. The scrollbar moves up and down to show how far through the list you've gone.

Once you've highlighted an item, tap the EDIT knob toward the panel and the item's submenu will appear. Some screens refer to this knob as JOG to save space. It's the same thing.

Takes you up to next higher menu

Pre-programmed setup

```
Audio settings
...↵
   Symphony Hall
  Microphones       →
  Line sens   Profess ▲
   Activate audio setup
```

Indicates there's a submenu

This item is directly settable and doesn't have submenus.

The first text entry in any menu is the current Setup for that menu (in this picture, it's a user-programmed setup called Symphony Hall). Setups instantly set every parameter in every submenu below them. Some setups are pre-programmed at the factory, and appear with an asterisk in their name. You can also create your own setups, following instructions on page 1-18.

If an item displays its current value (such as Line sens Profess in the picture) you can change its value with the EDIT knob:

1.  Highlight the item and tap the knob once.

2.  Turn the knob to cycle through its possible values. The currently-active setting has brackets around it.

3.  When you see the value you want, tap the knob to set it.

If you can't change a value, make sure the SAFE light isn't lit. See page 1-17 for more information about Safe Mode.

When you change a Setup, every menu item below it is changed to match the stored value for that setup.

If a menu item has a → icon, there's a submenu. Highlight it and tap the Edit knob to open the submenu.

HOT TIP!

To jump to the top of any menu, hold the Edit knob in and turn it one click to the left. To jump to the bottom, hold the knob in and turn in one click to the

right.

To exit any menu, hold the EDIT knob in for at least three seconds…

- or, select the ▪▪▪◢ icon in a submenu and tap the EDIT knob.

- or, hold the EDIT knob in and turn it to the left: each click will take you one step back through the menu structure, until you return to the Status screen.

- or ignore the menu. After a while (which can be set in the System menu) it will time out and revert to the next-highest level.

> **HOT TIP:**
>
> There's a complete map to the menus in the front of this manual.

## 3: How do I use a menu that appears locked?

Critical parameters can be locked to protect against accidental changes. When Safe Mode is on, SAFE appears in red on the front panel and the message This menu is disabled in SAFE mode appears when you try to open a blocked submenu. Use the SAFE MODE switch on the rear panel to turn this mode on or off. A small screwdriver blade will help you reach the switch. *Don't use a pencil to change the setting — the graphite tip might break off, and cause a short within ZephyrExpress.*

> **NOTE:**
>
> Don't turn off Safe Mode unless you know what you're doing. Codec settings have to match the distant studio and ISDN ones have to agree with the phone company. With the wrong ones, transmission will be impossible.

## 4: How do I lock a specific menu?

Obviously, turning the SAFE MODE switch to its On position will lock *something*. But before you use it, enter the Utility: Safe Mode submenu. You'll see a list of ZephyrExpress activities that can be locked. Each can be set to Allow or Block.

## 5: How do I enter numbers or names?

To enter a number for any menu item or when dialing, use the keypad.

To enter individual letters in a text field, press the number key with that letter printed on it. Tap the key repeatedly to cycle through the choices. When you see the one you want, move to the next character by tapping another number key or turning the Edit knob clockwise.

If you make a mistake, turn the Edit knob counter-clockwise to back up. Once you've finished entering, tap the knob to confirm.
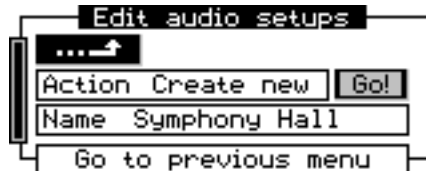
## 6: How do I create my own Setups and Master Setups?

> HOT TIP!
>
> You can capture a "snapshot" of all your current settings by using the `Auto-Create` function, described in step 4 below.

1. Make sure all the parameters within a menu page are set the way you want them for this Setup. Then enter the `Setups` submenu at the bottom of that page. It will be similar to this:

```
┌─────────────────────────┐
│   Edit audio setups     │
│ ...↵                    │
│ Action  Create new  Go! │
│ Name   Symphony Hall    │
│   Go to previous menu   │
└─────────────────────────┘
```

2. Enter the `Action` box and select `Create new`. This will take you to the `Name` box.

3. Using the keypad, enter a name for this setup. This will take you to the `Go!` box.

4. Tap the `Edit knob` and the parameters will be stored.

The main `Utility` menu has two setup choices at the bottom:

- `Master Setups` activate all the individual menu setups at once:

  If you select `Action: Create New`, you'll be able to enter a name for this setup. Then you'll see a list where you can choose individual Setups that will be activated for each of ZephyrExpress' menus. After you've selected these setups, scroll back to the `Go!` field to create the new master setup.

  If you select `Action: Auto-Create`, after you enter the name the `Go!` field will be automatically selected. If you then tap EDIT again, ZephyrExpress will scan every current menu setting and create a full set of new Setups that include them. For example, Auto-creating a new "`Arena`" master setup will also create a "`Arena-ISDN`" setup, a "`Arena-audio`" setup, and so on. You can use these individually, or just leave them alone to be invoked by the maser setup.

- `Autodial Setups` let you enter phone numbers and call type (`Zephyr` or standard `phone`) for each line, and select an optional `Master Setup` to apply whenever the call is placed.

  Autodial Setups can override Safe Mode. Even if a user is blocked from manually changing a parameter or setup in an individual menu, Autodialing will automatically change all the menus to match its master setup.

## ISDN connections

### 7:    How do I connect ZephyrExpress to an ISDN circuit?
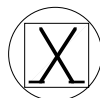
There are two telephone jacks on the left side of the rear panel. Only one of them should be connected to the ISDN line.

The upper jack is an S/T interface, for lines that have an NT1 terminator (frequently supplied by phone companies in Europe). Use the supplied 8-conductor cable, or a 10Base-T ("Category 5") cable, to the NT1. This jack also provides power to the terminator.

The lower telephone jack on a North American ZephyrExpress is a U interface marked LINE DIRECT. This is designed for direct connection to the unterminated lines supplied by American telephone companies. This jack uses ZephyrExpress' internal NT1. The phone company may have supplied an 8-pin jack (RJ-45) or a standard four-conductor jack (RJ-14 or RJ-11); you can plug a standard telephone line cord into either jack and it will work properly. If the phone company has supplied screw terminals or bare wires, use a modular cord with spade lugs, available from electronics stores, and connect to red and green.

The green LED below the LINE DIRECT label will blink rapidly when first contacting the central office, and then blink slowly while handshaking is going on. When the connection is complete, the LED will light steadily. If this doesn't appear to be happening, see section 9 of this manual for troubleshooting tips.

## 8: How do I manually enter a network setup?

You'll need to know the type of ISDN protocol your local phone company uses for the ISDN circuit, and any other numbers that are needed: SPIDs or Directory Numbers are required by some US installations, and MSNs by some European ones (your local phone company will have to tell you what you need). Open the `Utility: ISDN` menu. Select `Telco`, and set the protocol being used. Then enter the two `SPIDs`, `Directory` numbers, or `MSNs`.

If these menus appear locked, check Safe Mode (page 1-17).

## 9: How do I test that I've set things up properly?

Connect to the ISDN line and turn ZephyrExpress on. After about twenty seconds, `Ready` should appear in both boxes on the bottom of the status screen. If it doesn't, see the next question.
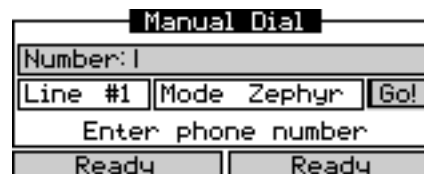
**DIAL**

Then call yourself. You'll actually be calling one ISDN number with the other over the same copper pair, to check that you're properly connected to the phone company switch and that ZephyrExpress is programmed correctly.

1. Set `Codec` for `L3-DUAL` transmit and `L3-STEREO` receive.

2. Tap the DIAL button. A screen like this will appear:

```
┌──────Select Dial Type──────┐
│                            │
│ Manual... ▊Auto...▊ Recent...│
│                            │
│   Activate a master setup  │
│ ┌────Ready────┐┌────Ready────┐│
└──────────────────────────────┘
```

3. Select `Manual`. When you tap the EDIT knob, a screen like this will appear.

```
┌──────Manual Dial──────┐
│ Number:│              │
│ Line #1│Mode Zephyr│Go!││
│    Enter phone number  │
│ ┌───Ready───┐┌───Ready───┐│
└─────────────────────────┘
```

4. Enter your *second* ISDN number. Since this is a local call, you probably won't need "1" or an area code. If you make a mistake, turn the `Edit` knob counter-clockwise to back up.

5. When your second number is completely entered, tap the `Edit` knob. This selects the `Go!` box. Tap again, and ZephyrExpress will use line 1 to call its own line 2.

Once you're successfully connected, play some audio routed to both channels or use the built-in tone generator (page 1-26). You'll see it on the Send meters, and — with a slight delay — on the Receive meters.

**DROP**

Whether or not you managed to connect, you must drop both lines before continuing. Tap the DROP button and turn the Edit knob to select `Both`. Then tap the Edit knob once.

### *Alternate test procedure*

This won't test whether you're set up properly to send audio, but does test the ISDN setup and ZephyrExpress receive circuits.

1.  Press the DIAL button and select `Auto...`

2.  In the `Dial a Setup` screen that appears, select either `US testline*` or `Euro testline*`.

3.  Tapping the Edit knob at the Setup will select the Go! box. Tap again to configure ZephyrExpress for the test line and make the connection.

Our test lines play continuous music. Since they're intended for testing both mono and stereo connections with both Switched56 and ISDN connections, you may hear some distortion of the stereo image. This happens with the test line only: your ZephyrExpress will have correct imaging with a stereo connection.

> ⚠️ **NOTE:**
>
> Auto-dialing a testline will change your `Codec` menu settings. Don't do it unless you know how to restore these settings, or have a pre-programmed Codec Setup that's appropriate for the current broadcast.

## 10: What do I do if I can't connect?

Try again.

If you're calling yourself, try dialing your first ISDN number instead of the second; the installer might have reversed them. Then check that the `ISDN Settings` were entered correctly, including any extra digits at the end of the SPIDs.

If you're calling yourself, see the two call timers (`00:00`) on the status screen, but don't hear audio and no LOCK LED is lit, make sure the transmit and receive coding methods match (see page 1-9).

If you're calling a Telos test line, try dialing manually using the numbers on page 3 of this booklet.

If you can't connect to a distant studio, try having them call you.

More troubleshooting tips appear in Section 9 of this manual.

---

⁻* An asterisk in a setup's name indicates that it was pre-programmed at the factory. But we'll also use it here to point out that these calls may involve a long-distance connection at ISDN rates: the US test site is in Ohio, and the European test site is in Freising, Germany.

## Audio setup

### 11:  How do I set ZephyrExpress for my microphones?

From the Utility screen, select the `Audio` menu and then the `Microphones` submenu. Each microphone has its own submenu, where you can:

- set `sensitivity` (`60 dB`, `45 dB`, or `30 dB` for various ribbon, dynamic, and condenser microphones; or `line` to use that input with a line-level source). If you can't get a good meter reading no matter how high you turn a mic's GAIN knob, try a higher sensitivity; if a mic sounds distorted at any setting of the GAIN knob, try a lower one.

- turn 48 volt `phantom power` on or off. Phantom power is not available when sensitivity is set to `line`.

> **NOTE:**
>
> Phantom power may damage improperly-wired or ribbon microphones. Leave it off unless you're sure a condenser mic needs it.

- turn a `low cut` filter on or off, to reduce rumble and wind noise. The filter is 12 dB/octave, 3 dB down at 100 Hz.

- Substitute a test oscillator for that microphone's signal. If a microphone's audio is missing and replaced by a steady tone, make sure the mic's `test tone` is set `Off`.

- The `Microphones` submenu also lets you turn a dual-channel `limiter` on or off.

The phantom power selection is remembered even while ZephyrExpress is unplugged. When you finish a remote broadcast, turn phantom power off as a courtesy to the next user.

> **HOT TIP:**
>
> When phantom power is turned on, it's supplied continuously — regardless of how the routing switch is set. That's so you can use the routing switch as a microphone mute, without worrying about a turn-on "thunk".

Turn each microphone on or off by pressing the small gray routing switch next to its gain knob, on the upper left of the front panel. LEDs indicate whether the mic is being sent to codec channel A, channel B, both, or is disconnected. If you're sending a mono signal, use channel A.

### 12:  How do I use line-level sources?

The rear-panel Line input jacks will accept XLR or 1/4" plugs, balanced or unbalanced. Wiring details appear later in this booklet. `Line sensitivity` can be adjusted in the Utility Screen's `audio` submenu: select `Profess` for +4dBu sources, and `Consumer` for -10dBu sources.

The small gray routing button next to the line 1&2 gain pot lets you turn the line off, select A B to send both inputs as a stereo pair to both codec channels, or A+B to sum both inputs to mono and send it to both channels.

If you want separate gain or routing controls for multiple line-level sources, you can temporarily convert either or both microphone inputs. Use the microphone sensitivity adjustment (page 1-22), and set it for line +1 dB.

> **HOT TIP:**
>
> The Line inputs do not pass through ZephyrExpress' built-in limiter. If you're using a line-level source that might have extreme peaks and want to avoid distortion through the codec, connect it through the microphone inputs with the mic limiter turned on.

## 13: How do I create a monitor mix?

ZephyrExpress has two independent stereo monitor mixers, each with two routing buttons and two related level knobs. Both mixers have headphone outputs on the rear panel. Monitor 1 also has XLR outputs on the rear; Monitor 2 also has a headphone output on the front panel.

One routing button on each monitor mixer is marked SEND: use it to select one or both send channels carrying local audio. Adjust its level using the adjacent knob, and panning (if necessary) with the Monitor XMT selection in the Audio submenu.

The RECEIVE button and knob combinations do the same thing for audio received from the distant studio.

We built two separate monitor mixers in ZephyrExpress for flexibility:

- If you're playing music at the remote site, you can feed just the local signal to Monitor 1 and use it for a public address system. Feed both local and distant signals to Monitor 2, and talent will be able to hear both the local mix and cues coming back from the studio.

- If you're originating voice at the remote site and playing music at the studio, have the studio send monaural music back on one of the receive channels. Route the local signal and the received channel to Monitor 1 for a public address system. You can then use the other receive channel for cues, routed to Monitor 2 only. Or have the studio send stereo music over both channels, and route them both to the public address system.

## 14: How do I test my audio and compression setups?

Use the Monitor 2 Send controls to listen to the signal you'll be sending to the studio, with headphones plugged into the front- or rear-panel Monitor 2 jacks. Make sure the microphone and line routing switches are set correctly. Audio coding works best when average levels hover just below -10 dB on the meters — the red LEDs — and peaks don't light the 0dB LED. If you have difficulty reaching these levels with a reasonable knob setting, or hear distortion at these levels, adjust microphone sensitivity (page 1-22). If

you can reach the right average level but peaks are too loud, make sure the limiter is turned on.

To test the digital compression engine, verify in the Status Screen that both transmit and receive are set to the same coding method (use L3-Stereo as a generic receiver when transmitting with L3-Dual, L3-Stereo, or L3-Joint). If necessary, change the methods using the Codec submenu. Then scroll down the Codec submenu to Loop Mode, and set the mode to Near. After a few seconds, both LOCK LEDs should light. Turn off the Monitor 2 SEND routing switch, and set the Monitor 2 Receive routing switch to A B. Now, any audio you send should be heard in the headphones, after a slight coding delay.

Don't forget to set Loop Mode back to Off before connecting to the studio.

### ISDN Connection

## 15: How do I place a call?

**DIAL**

1. Tap the DIAL button. A screen like this will appear:

```
┌──────── Select Dial Type ────────┐
│                                   │
│ [Manual...]  [Auto...]  [Recent...]│
│                                   │
│     Activate a master setup       │
│ ┌──────────┐  ┌──────────┐        │
│ │  Ready   │  │  Ready   │        │
│ └──────────┘  └──────────┘        │
└───────────────────────────────────┘
```
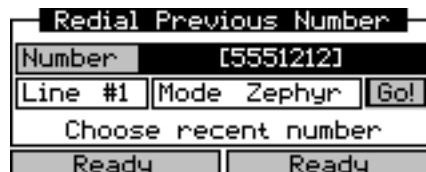
2. Select Manual, Auto, or Recent. Auto lets you select pre-programmed setups; Recent lets you automatically redial numbers you've entered manually.

If you select Auto, a screen like this will appear:

```
┌──────── Dial a Setup ────────┐
│ [Setup]  [FM Mstr Ctrl]      │
│ ┌──────────┐  ┌──────────┐   │
│ │   View   │  │   Go!    │   │
│ └──────────┘  └──────────┘   │
│       Choose setup           │
│ ┌──────────┐  ┌──────────┐   │
│ │  Ready   │  │  Ready   │   │
│ └──────────┘  └──────────┘   │
└──────────────────────────────┘
```

Choose a setup from the top of the screen. When you tap, the Go! box will be selected. Tap again to dial that number. If any menu setups were preprogrammed with that Autodial setup, ZephyrExpress will apply them before placing the call. This may change the coding method, audio setups, or even ISDN setup and system behavior depending on how it was programmed.

If you select Manual or Recent, a screen similar to this will appear:

```
┌──── Redial Previous Number ────┐
│ [Number]     [5551212]         │
│ [Line #1] [Mode Zephyr] [Go!]  │
│     Choose recent number       │
│ ┌──────────┐  ┌──────────┐     │
│ │  Ready   │  │  Ready   │     │
│ └──────────┘  └──────────┘     │
└────────────────────────────────┘
```

Select a recently-called number or enter a number from the keypad. If you're entering manually, don't forget to add any codes or "1" required in your dialing area. If you make a mistake, turn the knob counter-clockwise to back up.

**HOT TIP:**

If your phone system requires an access code to get an outside line, you can enter up to three digits in `ISDN settings: Outside line prefix`. This code will then be automatically applied to every call you make.

If you can't place a call, check the prefix setting.

> 3. After you tap the EDIT knob to confirm the number, the word `Go!` will be highlighted. Tap again to place the call... or select an option:

`Line` selects one of the two separate bearer channels of an ISDN circuit.

`Mode` selects `Zephyr` for audio transfers to another codec, or `Phone` to talk to a standard analog telephone using a microphone and headset plugged into ZephyrExpress.

> The `Phone` feature lets you call the studio (or order pizza) without requiring an analog line and separate phone at the location. It can be very handy for tracing problems, since you can call the studio engineer using a telephone connection while testing various coding method and bitrate combinations on the digital one.

> Some of ZephyrExpress' receive functions use `Line 1` only. If you're expecting to receive a codec call, leave this line available and make your analog calls on `line 2`.

> You don't have to use any special settings for the studio to call you. If an analog call is received on one of the ISDN lines, ZephyrExpress will automatically connect and switch that line to Phone mode for the duration of the call.

**HOT TIP:**

Telephone companies often charge less for voice calls than for data, so you can save money during the broadcast: send the studio high-quality voice on one ISDN channel, but place a standard call on the other channel for cueing.

**HOT TIP:**

The keypad buttons don't generate DTMF ("Touch-tone") tones, even if you're on a standard telephone call.

If you want to access voice mail or remotely-control the transmitter from ZephyrExpress, get an accessory DTMF generator at an electronics store and pick up its signal with a mic.

## 16: What do I do if they can't hear me?

First make sure your ISDN is properly configured.

If your call is received at the studio but they can't hear your audio:

> 1. Can you can see the signal on one or both of the Send meters? If not, check the routing switches and microphone settings.

2.  Are they set to a receive coding method that matches your send method? Try changing Codec settings while they listen.

3.  Try changing the Codec: Bitrate and redialing. Some connections will pass one rate but not the other.

4.  If you're working in stereo, make sure both ISDN lines are connected. If one of them hasn't been dialed or has dropped the connection, they *won't* hear a single channel of audio — instead, they'll hear nothing at all.

If they can hear you but the pitch sounds wrong, the Codec: Sample Rates probably don't agree.

More troubleshooting tips appear later in this manual.

## 17:  How do I send a test tone?

Open the Audio submenu and then one of the microphone's submenus. The last item in each mic submenu lets you turn test tone on or off. Then adjust the routing and level using that microphone's controls.

> **NOTE:**
>
> G.722 relies on the audio signal itself for synchronization, and can have problems with steady test tones. If you're using this coding method, avoid using the tone generator.

## 18:  How do I hang up?

DROP

Press the DROP button. A screen like this will appear:

```
┌─Select Line to Disconnect─┐
│                           │
│  Both     Line 1   Line 2 │
│                           │
│ Press jog to drop line 1&2│
│    00:47       00:47      │
└───────────────────────────┘
```
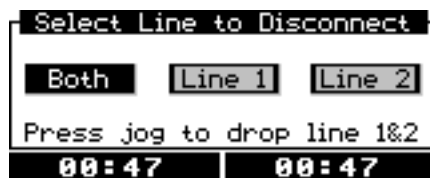
One or both lines will be automatically selected, depending on which are currently connected. Tap the EDIT knob to hang up, or select just one line if that's all you want to disconnect.

Pressing DROP doesn't do anything if neither line is connected.

## 19:  Where do I find answers to other questions?

| Question | Manual Section |
|---|---|
| How do I order my ISDN line? | ISDN information and order forms: Appendix |
| How do I connect ZephyrExpress? | Hardware Details in Section 3; also Audio Details in Section 4; |
| How do the controls work? | Hardware: Section 3 |
| What are the meanings of all the menu items? | Full Menu Reference: Section 7 |
| Should I use Layer III, Layer II, or G.722? | Brief Overview in Section 2; also Coding Details in Section 5 |
| What other codecs are compatible? How should they be configured? | Codec Compatibility Info: Appendix |
| How can I connect the ZephyrExpress to a computer or remote control? | System-Level Functions: Section 6 |
| How do I contact Telos? | Customer Support:  Page 2 |
| Connector Pin-outs? | Hardware Details: Section 3 |
| Specifications? | Specifications & Warranty: Section 14 |

## SECTION 3

# ZEPHYREXPRESS HARDWARE

# About our hardware...

ZephyrExpress, like almost everything else these days, is a software-driven device. This lets us put more features into the system, and means new updates can be installed without opening the case.

But hardware's important too. We put a lot of effort into making a system that would be compact and field-worthy, while looking and feeling like the precision audio equipment it is. The exterior features — all the knobs, buttons, and connectors — are organized in a logical way for intuitive operation. Even the design of the eight protective bumpers was carefully researched (though we must confess: we colored them teal just because we think it's pretty).

The first part of this section explains the panel features. The second details the connector and wiring standards. A separate section about hardware maintenance is at the end of this manual.
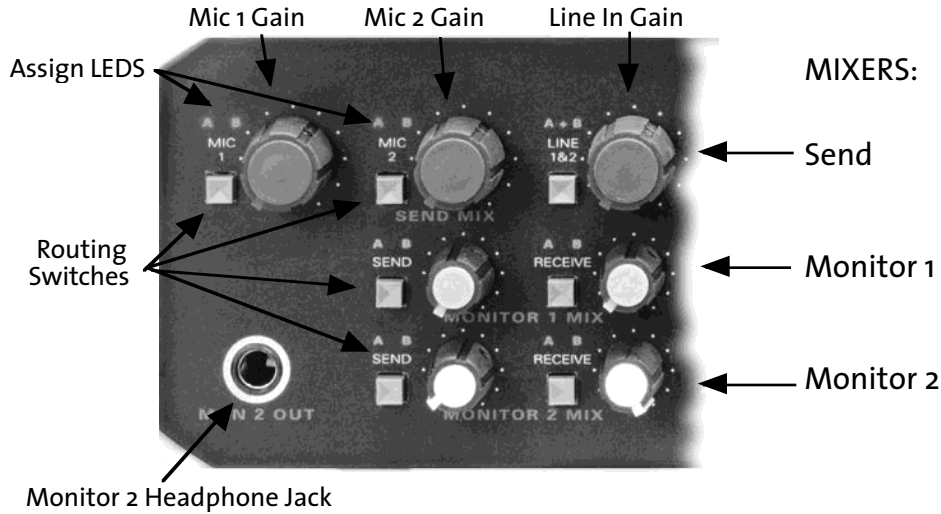
# Front Panel

This overview will help you get oriented:
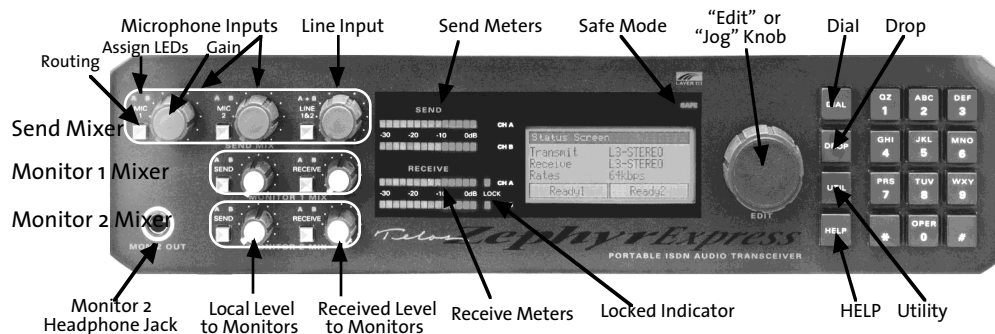
ZephyrExpress has a self-test function for the front panel LEDs: all of the indicators, plus every segment of the VU meters, should glow for a few seconds when you first apply power.

## The Mixer Section



Mic 1 Gain    Mic 2 Gain    Line In Gain

Assign LEDS

MIXERS:

Send

Routing Switches

Monitor 1

Monitor 2

Monitor 2 Headphone Jack

## Send Mixer

### Mic 1 and Mic 2 Routing Switches

Press these buttons to cycle whether each mic's signal will be sent to channel A, channel B, both channels, and off. The letters above each button will light up in green to indicate which channels are selected. This determines which one or two ISDN transmit channels will get the signal from that microphone. If you're using Layer 2 mono or G.722, you must route the signal to A.

> HOT TIP!
>
> The microphone A and B LEDs are also overload indicators. Normally the letters appear in green, and flash red on peaks. If they're turning red frequently, lower that microphone's `Sensitivity` in the `Audio: Microphone` submenus.

> TECHNICAL TIP!
>
> If you've turned `Phantom Power` on in the `Audio: Microphone` submenus, it won't be affected by the routing switches. You can use the buttons to mute the microphones, without worrying about a "thunk" when they come back on.

### Line 1&2 Routing Switch

Press to cycle between A B, A+B, and off. A B treats the two line inputs as a stereo pair, routing them to both ISDN channels. A+B combines the two inputs and then routes them to both channels, so you can use two different mono sources. If you want a mono line source to be routed to only one of the send channels, just plug in a single XLR connector — they have the letters A and B below them, to indicate their channel routing — and use the A B setting.

### Level Controls

The three large knobs, with black caps, control the send level for their respective inputs. MIC 1  and MIC 2 are mono controls; LINE 1&2 is a stereo control.

## Monitor Mixers

### **Send** Routing Switches

There are two of these: one for each of the two stereo monitor circuits. Press to cycle between A, B, A B, and off. This determines which of the two SEND signals — the ones mixed by the large black knobs, and showing on the upper VU meters — will be sent to each monitor. Each of the two send signals is mono; you can adjust how they're panned in the monitors by using the Audio: Monitor1 XMT and Monitor 2 XMT submenus.

### **Receive** Routing Switches

Again, there are two: one of each of the two stereo monitor circuits. Press to cycle between A, B, A B, and off. These determine which of the two signals received via ISDN — the signals displayed on the lower VU meters — will be sent to each monitor. Each received signal is mono; you can adjust how they're panned in these monitors by using the Audio: Monitor1 XMT and Monitor 2 XMT submenus.

### Level Controls

These determine how much of the send and receive signal will be applied to each of the two stereo monitor circuits. Monitor 1's knobs have gray caps, matching the gray rings around the MON 1 headphone and XLR output jacks on the rear panel. Monitor 2's knobs have white caps, matching the white rings around the MON 2 headphone jacks on the front and rear panel.

# Display section

Send Meters   LCD Display   SAFE Indicator



Receive Meters   Locked Indicators
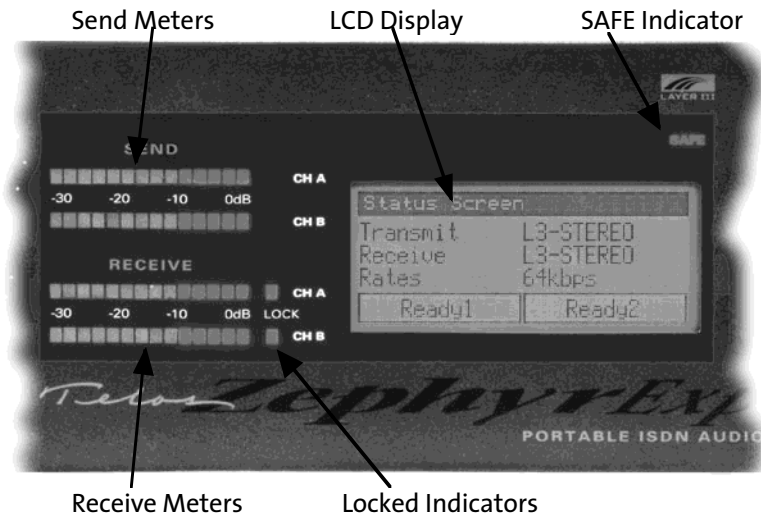
## Send and Receive Meters

The SEND meters indicate the levels being sent to the audio encoders. This is important, because codecs work best when they've got a reasonably high signal to work with. Adjust the mixer so that most of the green LEDs below -10dB are lit during average program material, with the red LEDs flashing occasionally. As with most digital devices, 0dB is the clipping point: there isn't any headroom above it, and this LED shouldn't be lit.

> HOT TIP!
>
> Heavily-compressed material, such as pre-recorded commercials and most pop music, can be set with their average levels in the red area. Just make sure that peaks never light the top LED segment.

The RECEIVE meters show that audio is being received and decoded. They're calibrated to match the send levels at the distant encoder.

> HOT TIP!
>
> ZephyrExpress provides a `Far Loopback` diagnostic mode to test its analog and digital conversion circuitry, and help distant studios test their codec and the line. When this mode is active, the meters are turned off; the middle LED of each meter will light steadily as a reminder that normal operation won't be possible until `Loopback` is turned off.

## Lock LEDs

These light when the receive decoder is locked onto a valid coded signal, and the distant transmitter's coding method matches the local decoder. They'll flash if the receive decoder is set to one Layer 3 mode, such as mono, while the incoming signal is a different Layer 3 mode. If the signal and receiver are set to totally different algorithms, or there's no coded signal coming in, they won't light at all. They'll turn off and received

audio will be muted if the coded audio bitstream is corrupted because of transmission problems.

## LCD Display

This normally displays the Status screen.
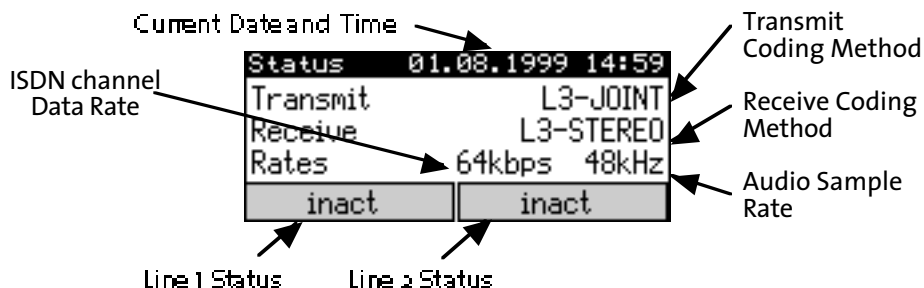


When you press the DIAL, DROP, UTIL, or HELP buttons, the screen changes to show that function. Dial, Drop, and Help are self-explanatory. Util opens ZephyrExpress' Utility menu, where you can change system parameters.

To return to the Status screen from one of the other screens, press the EDIT knob toward the panel for three seconds — it works like a pushbutton, as well as a rotary control. If you're in a Utility submenu, this takes you to the next higher level; keep holding the knob down until the Status screen reappears.

If you're in a hurry and don't want to hold the knob for three seconds, hold it in and turn it counter-clockwise until the Status screen appears.

Screens other than Status time out, and revert to the next higher screen, if you haven't done anything for a preset time. Adjust the timeout value in the System menu.

The LCD's contrast and backlighting are adjustable for various viewing conditions, also by using the System menu. The entire menu system is described in section 7.

## Blank Screen?

If ZephyrExpress' LEDs light when you first apply power but the LCD screen appears dark, or the LCD screen lights but no characters appear, the last user may have left its brightness or contrast at an extreme setting. To reset:

Press the DROP button and hold that button in while turning the EDIT knob clockwise to reset the backlighting. It may take a few turns.

Press the DIAL button and hold that button in while turning the EDIT knob either direction to change the LCD Contrast.

## SAFE indicator

The word SAFE lights in red when SAFE MODE is turned on from the rear panel. In this mode, various menus are locked and cannot be changed. You can set which menus will be blocked by using the Safe mode menu (with safe mode turned off at the time, of course).

The SAFE indicator flashes briefly when changing codec modes.

If SAFE is blinking, it may mean that someone left ZephyrExpress in one of its loopback modes when they last turned the system off. Normal operation will be impossible until you use the Codec menu to turn Loopback: Off. Blinking is normal if you have a European ISDN card and aren't connected to an ISDN circuit.

# Control section



"Edit" or "Jog" Knob    Dial    Drop    Numeric Keypad

HELP    Utility
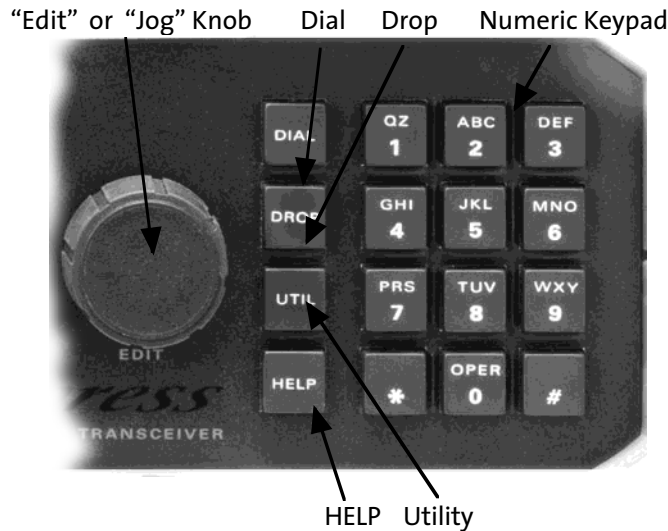
## Edit knob

Turn this knob clockwise or counterclockwise to cycle through selections in a menu, and through fields in the Dial screen. If you're entering text or a number using the keypad, turn counter-clockwise to erase the current character and back up. If you're entering text, turn clockwise to get to the next character.
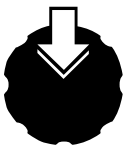
The Edit knob encoder also includes a pressure-sensitive switch. If you press it towards the front panel, it works like a pushbutton.

Tap the knob momentarily to activate a menu item so you can make choices, to confirm a choice, or to exit a text or numeric entry.

Hold the knob in for three seconds to exit the current screen. Hold the knob in and turn it counter-clockwise one click to jump to the top of the menu. Hold it in and turn it a lot of clicks to get to the status screen quickly.

If the status display is showing, anything you do to the knob will bring up a Help screen. That's because you can't use the knob to edit any of the fields in the status display. If Help appears, you may tap the knob again to return to the status screen.

The knob generates a clicking sound when you turn or press it. You can adjust click volume in the System menu.

## DIAL button

Dialing is a screen-driven software function. We've included complete instructions here, in the Hardware section of this manual, because the DIAL and DROP buttons don't make any sense without it.

DIAL · Press DIAL to initiate an ISDN or analog phone call. The display will show the dialing selection screen:

```
┌────── Select Dial Type ──────┐
│                               │
│  Manual...  Auto...  Recent...│
│                               │
│   Press jog to enter number   │
│ ┌──── Ready ───┐┌──── Ready ──┐│
└───────────────────────────────┘
```

Select Manual... if you want to enter a number directly. If you've manually placed a call since turning ZephyrExpress on, the last number entered will already be in the Manual dialing screen; you can use it or replace it with something else.

Select Auto... to choose from a pre-programmed list of places to call. Automatic dialing can be linked to codec and other menu setups, to change ZephyrExpress settings based on where you're calling. You can edit or add to the list by using the Autodial setups menu.
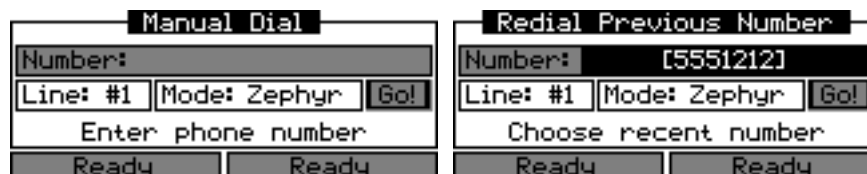
Select Recent... to choose from ZephyrExpress' own list of recent calls it's made. Recent calling keeps track of the line you called on and whether it was a high-quality or standard analog POTS call, but not the menu setups used for that call.

When you select Manual or Recent, the display changes to a Dialing screen. There are separate dialing screens for each of these two modes, but they're virtually identical:

```
┌───── Manual Dial ─────┐     ┌── Redial Previous Number ──┐
│ Number:               │     │ Number:       [5551212]    │
│ Line: #1  Mode: Zephyr  Go! │ Line: #1  Mode: Zephyr  Go!│
│   Enter phone number  │     │   Choose recent number     │
│ ┌─ Ready ─┐┌─ Ready ─┐ │     │ ┌─ Ready ─┐┌─ Ready ─┐     │
└───────────────────────┘     └────────────────────────────┘
```

When you enter the screen, the Number: field will be active. If you're dialing manually, enter a phone number using the keypad. If you're redialing, use EDIT to select a number from the list of recently-called ones.
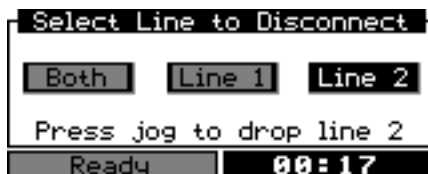
After you press EDIT to confirm, the Go! box will turn black. Press EDIT to place the call, or turn EDIT to highlight either the Line: or the Mode: field. You'll use Mode to place an

analog POTS call, as described in Section 4 (Audio Operations). Once you've changed one of these fields, press EDIT again. Then turn the knob to re-highlight the `Go!` field, and press EDIT to place the call.

## **DROP** button

DROP   Press DROP to hang up an ISDN or analog phone call, regardless of whether you placed it or someone else called your ZephyrExpress. The display will show the drop selection screen:

```
┌──Select Line to Disconnect──┐
│                             │
│  [ Both ]   [ Line 1 ]   Line 2 │
│                             │
│   Press jog to drop line 2  │
│ ┌──────────┐ ┌───────────┐  │
│ │  Ready   │ │  00:17    │  │
└─└──────────┘─└───────────┘──┘
```

If only one line is in use, it will be pre-selected when the screen appears (such as `Line 2`, above). If both lines are in use, select which one(s) to drop. When you press EDIT, the call will be disconnected.

⚠ USER NOTE!

If you press DROP while neither line is in use, ZephyrExpress will beep but nothing else will happen. You can't drop a non-existent call.

## **UTIL** button

UTIL   Press UTIL to open the Utility menu, where you can access various ZephyrExpress settings. This is fully discussed in Section 7.

## **HELP** button

HELP   Press HELP to open ZephyrExpress' context-sensitive HELP system. You'll see various screens, based on what you were doing when you asked for help. Here's a typical one, that you'd see if you pressed HELP while the Utility Screen's `Audio` menu selection is highlighted:

```
Help: Audio settings
Go to the audio settings
submenus to change
microphone & line input
settings, monitor panning
┌─────────────────────────┐
│  Press jog to return  ▼ │
└─────────────────────────┘
```

If the bottom line has a downward triangle ( ▼ ), more text is available on this topic. Turn EDIT clockwise to scroll down.

If the top line has an upward triangle ( **Help: Audio settings        ▲** ), there's more text scrolled off the top of the screen. Turn EDIT counter-clockwise to retrieve it.

When you're finished reading, press EDIT to return to where you were when you pressed the HELP button.

## Numeric Keypad

Used to enter numbers for dialing, and text for setup names. Dialing works like a telephone.

To enter text characters, press the key with the appropriate letter printed on it. Keep tapping it to cycle through the possible choices. For example, tapping the 2-key in a text field will first enter A. Repeated taps will change that to a B, then C, then the lower-case letters, then the number 2. Enter a space by pressing 1 three times. Punctuation characters appear on the * and # keys.

For example:

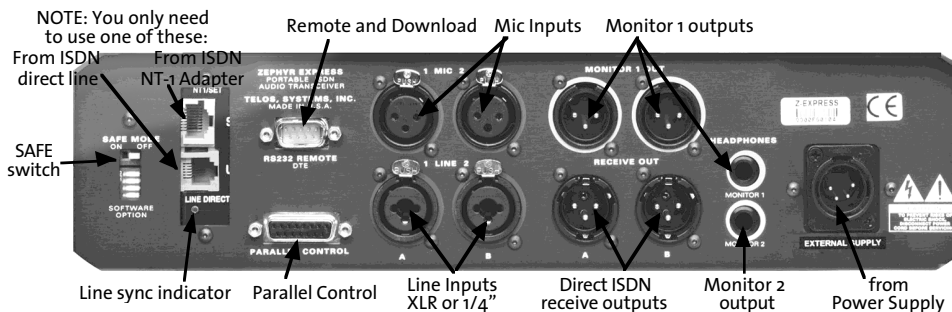| | |
|---|---|
| Press the number 2 once….. | See the character A |
| ... press it again….. | See the character B |
| ... press it again….. | See the character C |
| ... press it again….. | See the character a |
| ... press it again….. | See the character b |
| ... press it again….. | See the character c |
| ... press it again….. | See the numeral 2 |
| ... press it yet again….. | the cycle starts again from A |

To move to the next character in a field, tap any other key or turn the EDIT knob clockwise. To back up, turn the knob counter-clockwise.

---

ISDN TIP!

ISDN doesn't use DTMF "Touch-tones", even for calls to standard analog phones, so the keypad doesn't generate these sounds. If you need those tones to control a remote device, get an accessory tone generator at an electronics store and either pick up its signal with a microphone, or modify the generator for line-level output.
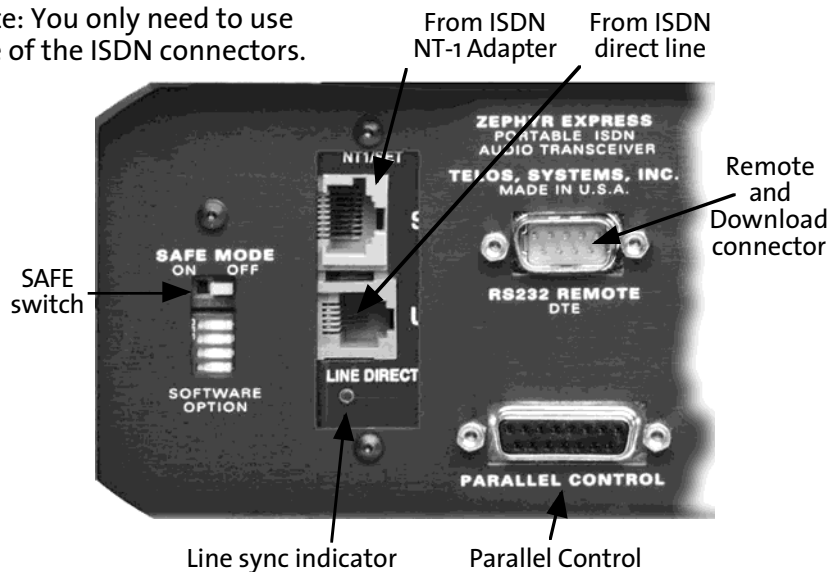
---

# Rear Panel



NOTE: You only need to use one of these:
From ISDN direct line    From ISDN NT-1 Adapter    Remote and Download    Mic Inputs    Monitor 1 outputs

SAFE switch

Line sync indicator    Parallel Control    Line Inputs XLR or 1/4"    Direct ISDN receive outputs    Monitor 2 output    from Power Supply

Note: This picture shows the back of a ZephyrExpress intended for sale in North America. Units shipped elsewhere will have only the upper NT1 phone jack, and no Line Sync indicator.

# Data connection section

Note: You only need to use one of the ISDN connectors.

From ISDN NT-1 Adapter

From ISDN direct line

Remote and Download connector

SAFE switch

Line sync indicator

Parallel Control

## MODE switches

### SAFE MODE

Turn ON to protect menu settings against accidental changes. This switch can be turned off with a small screwdriver blade or ballpoint pen.

> IMPORTANT!
>
> Don't attempt to move the SAFE MODE switch with a pencil. if the point breaks off, a piece of conductive graphite may fall inside ZephyrExpress where it will probably cause a short.

### Software Options

These switches are reserved for future functions.

## ISDN Connections

In most cases, you'll use only one of these at a time:

### S Interface

Connect to external NT1 Network Terminator, if supplied by your phone company. This is common practice in Europe.

**IMPORTANT!**

The S interface, in US versions, provides power for the NT1. Disconnect any other power supply connected to the NT1. Failure to do so will result in damage to ZephyrExpress, the NT1, or both!

European units — indicated by a single ISDN connection and the Euro Telecom logo shown here — do not provide power for the NT1. You may connect these directly to a powered NT1.

The S interface is a standard modular RJ-45 style jack. Ethernet 10Base-T cables work best for this application, and are available at most computer stores.

| S Interface Pinouts ||
|-----|---------------------------|
| *PIN* | *FUNCTION* |
| 1 | **N/C** |
| 2 | **N/C** |
| 3 | **S Transmit to network +** |
| 4 | **S Receive from network +** |
| 5 | **S Receive from network -** |
| 6 | **S Transmit to network -** |
| 7 | **PS2 Power -** |
| 8 | **PS2 Power + (Top pin)** |

**Pins 7 and 8 are not connected on units sold in North America.**

If you're using a European ZephyrExpress in North America, you'll need to provide your own NT1 and connect its Terminal jack to the S interface.

If you're using a North American ZephyrExpress with an external NT1, you'll need to provide an external power supply for the NT1.

When the U interface (below) is connected directly to the ISDN wiring, you can use the S interface jack to connect another device to the *output* of ZephyrExpress' internal NT1.

The most common use for this would be when ZephyrExpress is using one ISDN channel for audio transmission, and you want to use a separate ISDN telephone or computer terminal adapter on the other channel.

## U Interface

This jack appears only on units shipped within North America. It can be connected directly to the ISDN wiring, to use ZephyrExpress' built-in NT1 terminator. This is common practice in the US.

Only the two center conductors (red, green) of the ISDN jack are used with a U interface. If your phone company provided an 8-conductor RJ-45 jack, you can insert a standard RJ-11 or RJ-14 telephone line cord and it will properly center itself to connect to those conductors.

| U Interface Pinouts | |
|---|---|
| *PIN* | *FUNCTION* |
| 1 | **N/C** |
| 2 | **N/C** |
| 3 | **Line** |
| 4 | **Line** |
| 5 | **N/C** |
| 6 | **N/C (Top pin)** |

### Line Direct LED

This LED indicates the state of ZephyrExpress' internal NT1. It blinks rapidly when the built-in NT1 is first initializing the ISDN line, or if there is no ISDN line connected. It blinks slowly when contact is made with the central office. It will light steadily when all handshaking is complete and the basic line connection is good.

The LED is not supplied on units shipped outside of North America.

## Remote Control jacks

ZephyrExpress has two computer-style jacks on the back panel. A DB-9 is used for serial RS-232 communication with the system, and a DB-15 is used for parallel contact closures and logic-level inputs.

### RS-232

The RS-232 port is used for:

• remote control of ZephyrExpress menus from a modem or personal computer

• downloading new ZephyrExpress software.

Complete instructions and the remote control command language appear in section 6.

The port accepts a male DB-9 and matches the format found on personal computers. The default line protocol is 8 bits, 1 Stop Bit, No Parity, 9600 bps. You can change the baud rate for remote control or downloads from the System menu.

| RS-232 Pinouts | |
|---|---|
| *PIN* | *FUNCTION* |
| 2 | **Rx (Computer to ZephyrExpress)** |
| 3 | **Tx (ZephyrExpress to Computer)** |
| 4 | **DTR (ZephyrExpress ready output)** |
| 5 | **Ground** |

HOT TIP!

This port is configured as DTE, not DCE. This means pins 2 and 3 are wired like a computer instead of as a modem. You can plug in a standard modem, set for auto-answer, to remotely control ZephyrExpress for automated remote broadcasting. If you want to connect directly to a computer, you'll need a null modem (a molded plug/jack with crossed wires, available from computer stores) or will want to cross pins 3 and 4 in the cable.

### Parallel Port

The parallel port functions are not available as of ZephyrExpress software version 1.2.1. If you're using a subsequent update that supports the parallel port, here's how it's wired:

| Parallel Port | | | | |
|---|---|---|---|---|
| *PIN* | *FUNCTION* | | *PIN* | *FUNCTION* |
| 1 | **Ground** | | 9 | **N/C** |
| 2 | **Output 2** | | 10 | **Output 0** |
| 3 | **Status Out** | | 11 | **Output 1** |
| 4 | **Output 3** | | 12 | **N/C** |
| 5 | **N/C** | | 13 | **Input 2** |
| 6 | **Input 3** | | 14 | **Input 1** |
| 7 | **Input 0/Panic Dial** | | 15 | **N/C** |
| 8 | **+5 volts 400 ma max.** | | | |

## Inputs

All parallel port inputs are wired to accept either a voltage (up to +24 DC) or a closure to ground. Ground closures can be through switches or either pulled-up or open-collector logic outputs.
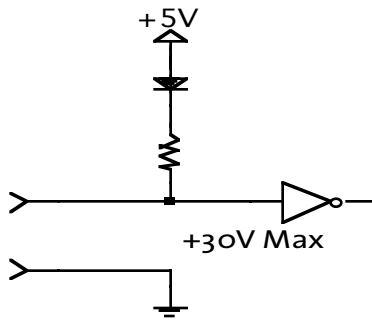
*Parallel logic input circuit.*

## Outputs

The outputs are open-collector closures to ground, active low. They require a pull-up resistor to use with logic-circuit inputs. The pull-up may be built into the external device; if not, use a 2.2kΩ resistor to the port's +5v supply.

Current should be limited to 400mA for each individual output and no more than 1 amp total. This isn't a problem with logic circuits or individual LEDs, but you should check current consumption if using solid-state audible alarms, relays, or incandescent lights.
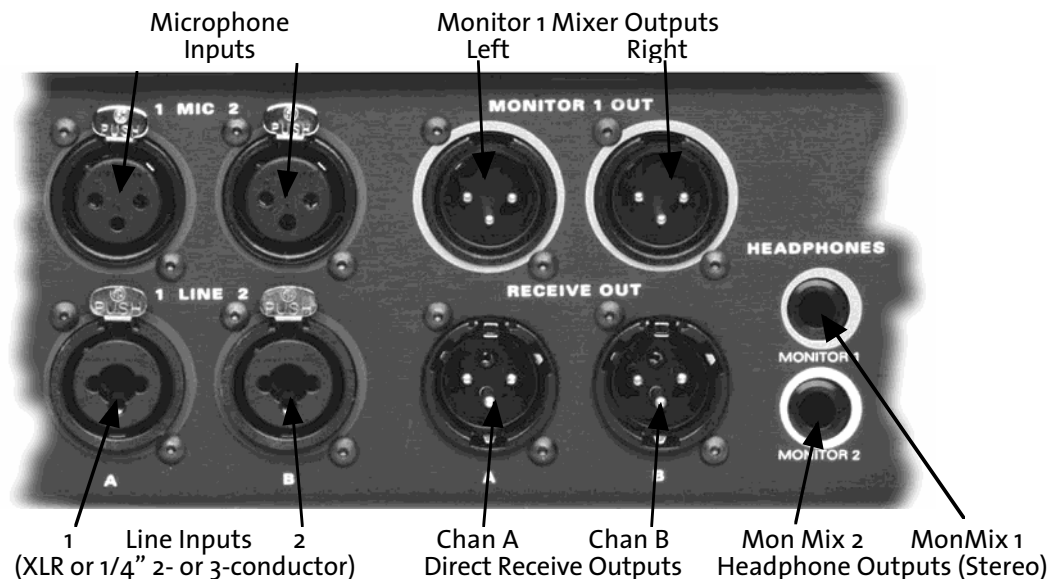
If used with a relay or LED, wire the other side of that component to the port's +5v supply; or wire to an external supply (no more than +24 v) and connect that supply's ground to the port's ground on pin 1.

⚠️ WARNING!

LEDs and low-voltage relays connected to ZephyrExpress' power supply need a current-limiting resistor in series.

# Audio i/o section



Microphone Inputs

Monitor 1 Mixer Outputs
Left    Right

1    Line Inputs    2
(XLR or 1/4" 2- or 3-conductor)

Chan A    Chan B
Direct Receive Outputs

Mon Mix 2    MonMix 1
Headphone Outputs (Stereo)

## About the XLR Connectors

ZephyrExpress XLR connectors are electronically balanced with pin 1 ground, pin 2 hot, and pin 3 return. Line inputs are bridging, high-impedance.

Outputs are active-differential balanced, at a nominal +4dBu; an external resistive pad may be necessary for consumer amplifiers.

| All XLR Pinouts | |
|---|---|
| *PIN* | FUNCTION |
| 1 | **Ground** |
| 2 | **Audio +** |
| 3 | **Audio -** |

• To use the line inputs with unbalanced sources, connect the signal to pin 2 and ground to pins 1 & 3. You may need to raise the inputs' sensitivity in the Audio menu.

• To feed the line outputs to an unbalanced device, connect pin 2 to the signal input and pin 1 to ground. Do not connect pin 3.

HOT TIP!

ZephyrExpress' line outputs are at +4dBu, which may be too hot for the unbalanced inputs on most consumer and semi-pro equipment. You'll probably need an attenuator, which can be as simple as a 5kΩ audio-taper potentiometer:
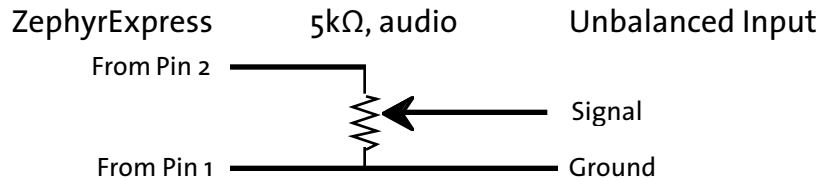


CURIOSITY NOTE!

ZephyrExpress uses the "pin-2-hot" scheme shared by most modern equipment. You can remember the pins with the mnemonic "George Washington Bridge": pin 1 = G = Ground; pin 2 = W = White = "+" in most cables; pin 3 = B = "-" = Black.

## Microphone Inputs

These are high-impedance inputs that'll work with most modern microphones. Some older transformer-coupled mics might show a high-frequency rise when unloaded; if this is a problem bridge pins 2 and 3 with a 680Ω or 270Ω resistor, depending on the mic's nominal impedance.

Using Audio: microphone submenu choices, phantom power can be supplied at either or both inputs, and nominal gain is separately adjustable to –60dBu (for ribbon

microphones), `-45dBu` (most dynamic and condenser mics), or `-30dBu` (very hot condenser mics or dynamics in loud environments).

Either microphone input's sensitivity can also be set to `line`, with a nominal level of +1dBu, for use as an additional line input. When `line` is selected, phantom power is not available.

> **IMPORTANT!**
>
> We don't recommend using unbalanced microphones.
>
> Aside from the fact that their wiring is less immune to noise, using an unbalanced mic while phantom power is turned on can damage the mic, ZephyrExpress, or both.

## Line Inputs

These jacks also accept standard 1/4" phone plugs, either balanced 3-conductor or unbalanced 2-conductor. Level is nominal `-10dBu` or `+4 dBu`, as selected in the `Audio` submenu. Peak input level is +19dBu.

The line input impedance is approximately 100kΩ, so it will bridge almost any source. Some transformer-coupled devices or equalized analog telephone lines may require a 680Ω resistor in parallel to prevent high-frequencies from being emphasized.

| 3-conductor input plug | |
|---|---|
| *CONTACT* | FUNCTION |
| Tip | **Audio +** |
| Ring | **Audio -** |
| Sleeve | **Ground** |

| 2-conductor input plug | |
|---|---|
| *CONTACT* | FUNCTION |
| Tip | **Audio +** |
| Sleeve | **Ground** |

If using an unbalanced source with a 3-conductor plug, wire the ground to both the ring and the sleeve.

## Monitor 1 XLR outputs

These carry the stereo signal from the Monitor 1 mixer, for use with monitor amps and public address systems. Their signal switching and volume is controlled from the front panel, and signal panning is controlled in the `Audio` menu.

## Direct ISDN Receive outputs

These are the direct outputs of the audio decoder, with no level control or panning. Their primary use is to feed a cue-only circuit when both monitor mixers are being used for other purposes, or for when ZephyrExpress is being used as a remote receiver only.

If you want to hear the local microphones through these outputs, they have to be sent back from the distant studio (which will result in coding delays).

### Monitor 1, Monitor 2 Headphone outputs

Stereo jacks with left on the tip, right on the ring, and ground on the sleeve. Monitor 2's front- and rear-panel headphone jacks have separate output buffers, so one can be used to drive low-impedance headphones while the other drives high-impedance ones or an unbalanced monitor amplifier.

| Headphone Jacks | |
| --- | --- |
| *CONTACT* | FUNCTION |
| Tip | **Left** |
| Ring | **Right** |
| Sleeve | **Ground** |

# Power connection



Serial Number

From Power Supply

EXTERNAL SUPPLY
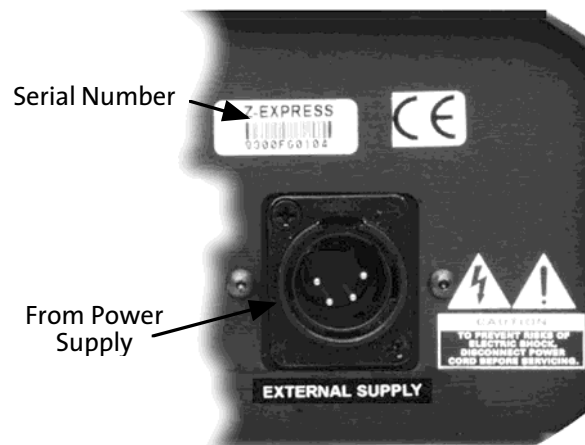
### Power Supply Jack

The 4-pin XLR connector is designed to be used with ZephyrExpress' modular power supply.

> WARNING!
>
> Be sure the supply is turned off at its switch, or disconnected from the AC line, before plugging or unplugging this connector.
>
> Failure to do so can erase ZephyrExpress memory settings and damage the unit.

We don't recommend using any other power supply. If you must use a different power source in an emergency **be sure to turn on or off all voltages simultaneously.**

| Power Input | |
|---|---|
| *PIN* | Voltage |
| 1 | **+5Vdc @ 5A** |
| 2 | **+12Vdc @ 1.5A** |
| 3 | **-12Vdc @ .5A** |
| 4 | **System Ground** |

## Power Supply



A separate modular power supply is provided with ZephyrExpress. Make sure it is not turned on, or check to make sure it isn't connected to an AC line. Then plug its 4-pin XLR connector into the jack on the right side of the rear panel.

The supply is self adjusting between 100 - 250 VAC, 50-60Hz. Power consumption is 150 Watts. The supply is cooled by convection through slots in the case; make sure these aren't blocked during use. It is normal for the case to become warm during use.

The input connector is a standard IEC, used on electronic equipment and most computers.

| Power Supply XLR Output | | |
|---|---|---|
| *PIN* | *Wire Color\** | Voltage |
| 1 | Black | **+5Vdc @ 5A** |
| 2 | Yellow | **+12Vdc @ 1.5A** |
| 3 | White | **-12Vdc @ .5A** |
| 4 | Blue + Red | **System Ground** |

*Color codes correspond to power supply model HES61-30

**IMPORTANT:**

Never connect or disconnect the modular supply while it is powered! Make sure the power supply's green LED is dark before plugging or unplugging its XLR connector.

Connecting or disconnecting the modular supply while powered can erase memory settings or even damage ZephyrExpress.

WARNING!

There are no user-serviceable parts in the power supply. Do not attempt to open it, as hazardous voltages may be present.

WARNING!

We urge you to keep the AC ground conductor connected, and not clip it or use an adapter that defeats the ground. Failure to keep the ground connected can create serious safety hazards. It also stops the internal power-line filter from working properly.

IMPORTANT!

As with any piece of electronic gear, it is advisable that precautions be taken to prevent damage caused by power surges. Standard line surge protectors can be used to offer some degree of protection. It's your responsibility to provide adequate protection.

# AUDIO OPERATIONS

# A console at your fingertips...

ZephyrExpress includes three separate 4-input stereo mixers, designed to make remote broadcasting easier. They incorporate both digital and analog technology, giving you the best of both worlds.

We used sophisticated analog circuits for:

- Wide input range

- Smooth limiting, with the ability to handle sudden peaks without digital artifacts

- Uncompromised audio specs

- Field-worthy reliability

But we control these circuits digitally, for:

- Versatility

- Fast, repeatable setups

- The ability to store presets for different sites

- The ability to lock critical settings against accidental tampering

Audio quality matches the highest standards:

- Frequency response +0, -0.1dB between 20Hz - 20kHz

- Total harmonic distortion < 0.019% @ 10dB, 20Hz - 20kHz

- Dynamic range of +94dB, 20Hz - 20kHz

## Mix flexibility

ZephyrExpress' SEND mixer can control two microphones — each with switchable sensitivity, limiting, and low-cut filters — and either two mono or one stereo line-level devices. The line inputs accommodate professional or consumer-level sources, balanced or unbalanced, on XLR or 1/4" plugs. The microphone inputs can be switched to use as one or two additional balanced line-level inputs, or to use as test oscillators. All of the inputs have flexible routing and muting switches.

ZephyrExpress has two separate MONITOR mixers, for ultimate flexibility when setting up remote broadcasts. You can send local mics down the ISDN line, while mixing them with high-quality music received over ISDN for a house PA system... at the same time monitoring distant studio cues along with the rest of the mix in a producer's headphones. You can even carry a mic out into the audience, while all this is going on, and control its mono feed to the local PA system without affecting the ISDN transmission.

Each monitor mixer has routing, pan, and level controls for the two send channels and for the two receive channels. Stereo panning can be set to switch automatically depending on how a signal is routed. One monitor mixer features XLR balanced outputs

to feed a local PA system or DAT recorder, along with a rear-panel stereo headphone jack. The other monitor mixer has dual stereo headphone outputs to feed producer and talent simultaneously.

If we sound proud of the design of ZephyrExpress' mixers, we are. But if you'd prefer to bypass them entirely, there are also separate direct-codec outputs . You can use these for high-quality stereo monitoring regardless of the mixer settings.

## The Send Mixer



Routing / Overload Indicators

Send Meters

Routing Mic 1 — Level

Routing Mic 2 — Level

Routing Line — Level
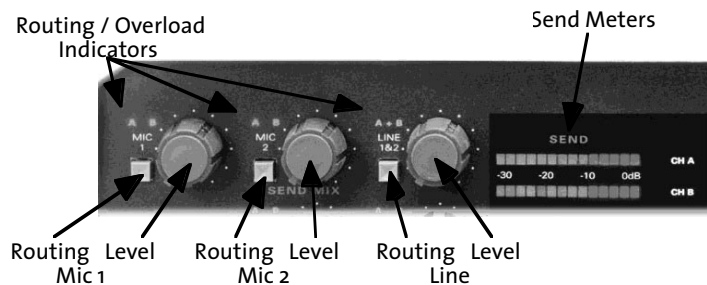
## Microphone Inputs

Each of the two microphone inputs has the same signal flow:



**BOLD FACE: Front Panel Hardware**
NORMAL FACE: Menu-item Software Switches
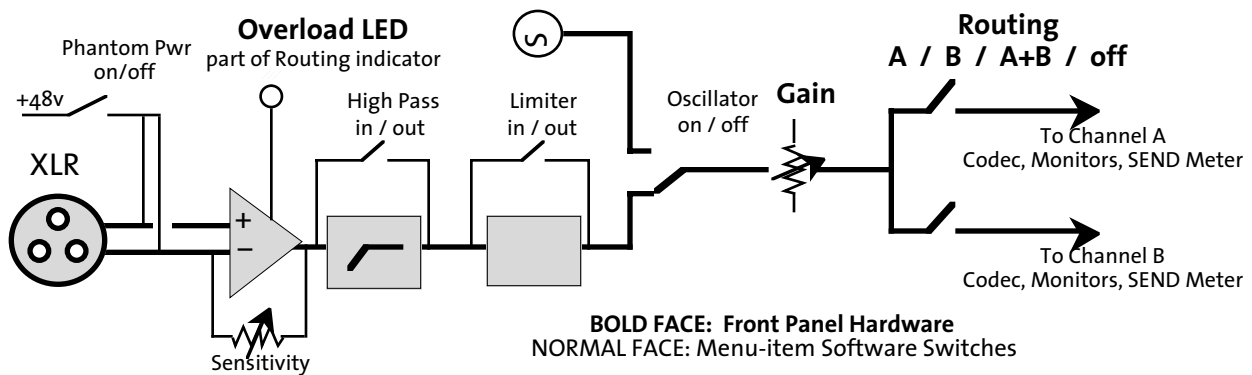
The XLR inputs are differentially balanced, with pin 2 hot. Input impedance is 1kΩ per leg, or 2kΩ balanced, to match most professional microphones. The phantom power supply provides 48vdc at 6 mA per microphone.

PHANTOM POWER...

provides DC to charge a condenser microphone and power its preamplifier on the same 2-conductor balanced shielded cable that's carrying the mic's audio. It does this by applying the voltage equally to both signal wires, returning the voltage through the cable's shield. Since audio is carried differentially in a balanced wiring scheme, the two don't interfere: there is no DC voltage from one signal wire to the other.

It's called "Phantom" because a properly-wired dynamic or ribbon microphone won't see the voltage at all.

BEWARE THE PHANTOM!

While properly-wired balanced microphones can't be damaged by phantom power, improperly-wired ones will cause trouble. Plugging in an unbalanced mic, using a "cough button" that shorts the mic's signal to ground, and some kinds of cable shorts can damage the microphone, ZephyrExpress, or both.

If you're not using a condenser microphone that specifically requires phantom power, leave that setting turned off.

Input parameters are adjustable in separate submenus, one for each microphone, from the `Audio: Microphones` submenu:

`Microphones`

`— Mic 1`

`— Sensitivity`  Selects the nominal level for a full scale reading on the meters (?) with the microphone's level control fully up. Select `high +60dB` for most ribbon mics, `mid +45dB` for most dynamic and condenser mics, `low +30dB` for hot condenser mics or dynamic mics in very loud surroundings. You can also select `line +1dB` to use the input with line-level sources.

`—Phantom Power`  Select `On` or `Off`, but see the warning above. Phantom power is automatically turned off when you select line sensitivity.

`—Low Cut Filter`  Applies a gentle filter to reduce wind noise, popped plosives, and building rumble. It's a good idea to leave this on for all voice applications, and turn it off for music transmissions.

`—Mic 1 test tone`  Substitutes an 860Hz sine wave for the microphone's signal, for testing the ISDN connection and setting nominal volumes through the broadcast chain. The tone's routing and level are adjusted by the front-panel MIC 1 controls.

`—Mic 2`

`—Sensitivity`  See above

`—Phantom Power`  "

`—Low Cut Filter`  "

`—Mic 2 test tone`  "

`—Limiter bypass`  Applies a limiter to both microphones at once to preserve stereo imaging.

ZephyrExpress' limiter engages at 15 dB above the normal operating value, so minor overloads aren't affected. The limiting action itself gives you as much as another 20 dB of headroom.

Why you might want to leave the limiter off:

- Dynamic range increases by about 7 dB, which may be desirable for non-broadcast applications.

- Total harmonic distortion and noise figures are slightly better with the limiteroff. You might want to turn off ZephyrExpress' limiter if another limiter is already being used in the circuit.

Why you might want to leave the limiteron:

- It prevents digital clipping and overloading of the A/D converter.

- Total harmonic distortion and noise figures are quite acceptable with the limiter engaged.

## Mic Routing Switches

The front-panel MIC 1 and MIC 2 routing switches cycle through four settings, as indicated by green LEDs above the buttons:

- A    Mic is routed to the codec via the A send bus: this is the proper setting for almost all mono transmission, and the left channel for stereo.

- B    Mic is routed via the B send bus, for a second mono transmission under `Layer 3-Dual` or the right channel of a stereo signal.

- A B   Mic is routed to both send busses. Use this setting for mono mics when mixed with stereo line-level sources in a stereo transmission, or to send the same signal to two sites under `Layer 3-Dual`.

- (OFF)   LEDs don't light and the microphone is turned off. Switching is instant and pop-free, so you can use this setting as a mic mute. Phantom power is still applied when the routing is off, so you can turn the signal back on without a "thump".

The A and B LEDs are dual color: they'll glow red when a mic preamp is overloaded. Occasional overloads may be acceptable — the analog preamp can handle them gracefully — but if the lights stay red, you should lower the preamp's `sensitivity` in the `Mic 1` or `Mic 2` menu.

## Line Inputs

The two line inputs can be used for a single stereo or one or two mono signals.



**Line A**
XLR/TRS

**Gain**

**Routing
Stereo (A B) /
Mono (A+B) / off**

+
−

To Channel A
Codec, Monitors, SEND Meter

**Line B**
XLR/TRS

+4 / -10  select

+
−

To Channel B
Codec, Monitors, SEND Meter

**BOLD FACE:  Front Panel Hardware**
NORMAL FACE: Menu-item Software Switches

+4 / -10  select
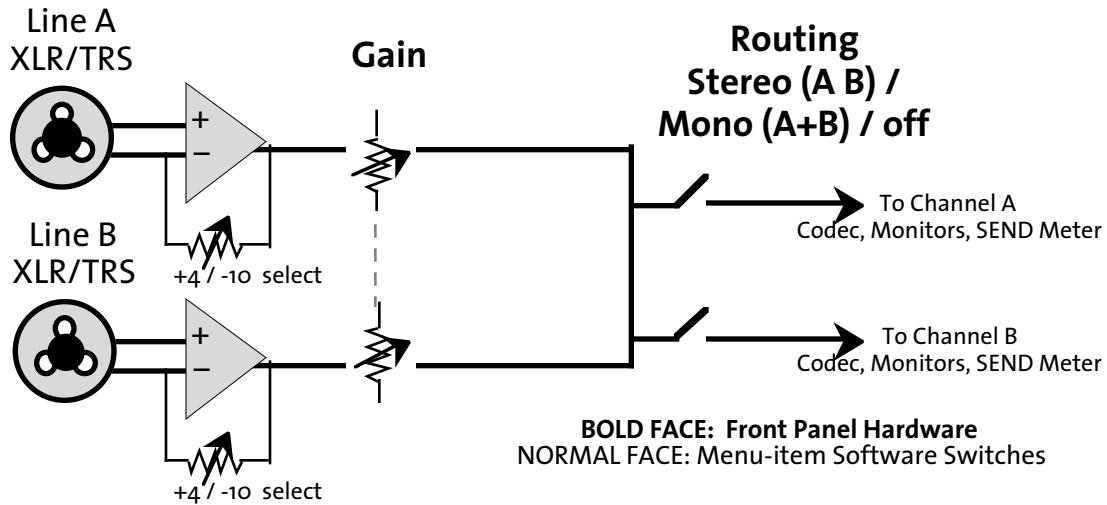
Each input jack accommodates XLR or 1/4" phone plugs, wired balanced (pin 2 or tip hot) or unbalanced (signal on pin 2 or tip; ground to pins 1 and 3 or to ring and sleeve). Standard 2-conductor phone plugs can also be used for unbalanced signals: their sleeves will contact the jack's sleeve and ring connections.

Input impedance is  approximately 100kΩ, to bridge any line-level source. Transformer-coupled devices and equalized analog telephone lines can show a slight high frequency rise when they're not terminated; if this is a problem, put a 680Ω resistor in parallel with the input.

There is one menu adjustment for the line input, Audio: Line sensitivity. Set it to Profess for +4dBu sources, and Consumer for -10dBu sources. The line input can handle up to +19dBu in its Profess setting without clipping.

## Line Routing Switch

The front-panel LINE 1&2 routing switch controls both line inputs simultaneously. It cycles between A B , A+B, and off.

• In A B mode, Line 1 is routed to the channel A send bus and Line 2 is routed to channel B: This is the normal mode for a stereo signal.

• In A+B mode, the two line inputs are summed to mono and then routed to both busses. This lets you use two different mono line-level sources.



HOT TIP!

If you want to use two line inputs with separate gain controls or more flexible routing, convert the mic inputs to line-level.

## Send Level Controls

The three large black knobs control the level of their respective signals to the send bus. They are calibrated in arbitrary gray dots that don't mean much, since the best way to mix is by using your ears and the SEND meters.

## SEND Meters

The two 14-segment SEND level meters have a fast attack and slow decay time, to alert you to any problems in the transmitted level. They monitor the signal as it enters the audio encoder, which works best when it's got reasonably high levels to work with. Adjust the mixer so that most of the green LEDs are lit during average program material, with the red ones flashing occasionally.

Heavily-compressed material, such as pre-recorded commercials and popular dance music, can be set to average in the red area as long as the 0dB segment isn't lit. Zero is the clip point, and there just isn't any headroom above it.

```
              SEND
  ▢▢▢▢▢▢▢▢▢▢▢▢▢▢        CH A
  -30      -20     -10      0dB
  ▢▢▢▢▢▢▢▢▢▢▢▢▢▢        CH B
```
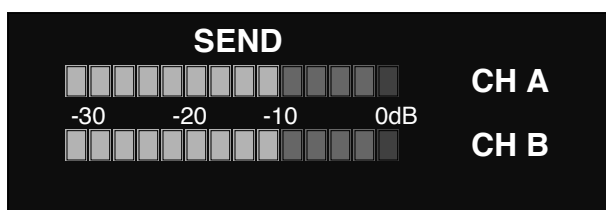
**HOT TIP!**

The meters don't operate on channels that are in standard telephone / G.711 mode. During `Far Loopback` test mode, both meters are off and the center segment is lit as a reminder that normal operation is disabled.

**DIGITAL AUDIO TIP!**

You'll notice that normal operating level is around -10dB and zero is at the top of the scale. This is different from analog VU and deciBel meters you may be used to, but it's standard practice on digital devices.

The reason is that analog equipment has a few dB headroom above the normal "zero" level (and zero itself is arbitrarily matched to a particular voltage, magnetic strength, or modulation). Digital equipment, on the other hand, counts deciBels down from absolute full scale — the condition where all bits are turned on — and there just isn't anything that can be louder that that.

# The Monitor Mixers

MONITOR 1

Send
Routing Level

Receive
Routing Level

Receive Meters



Mon 2
Headphone Jack

Routing Level
Send

Routing Level
Receive

Lock Indicators
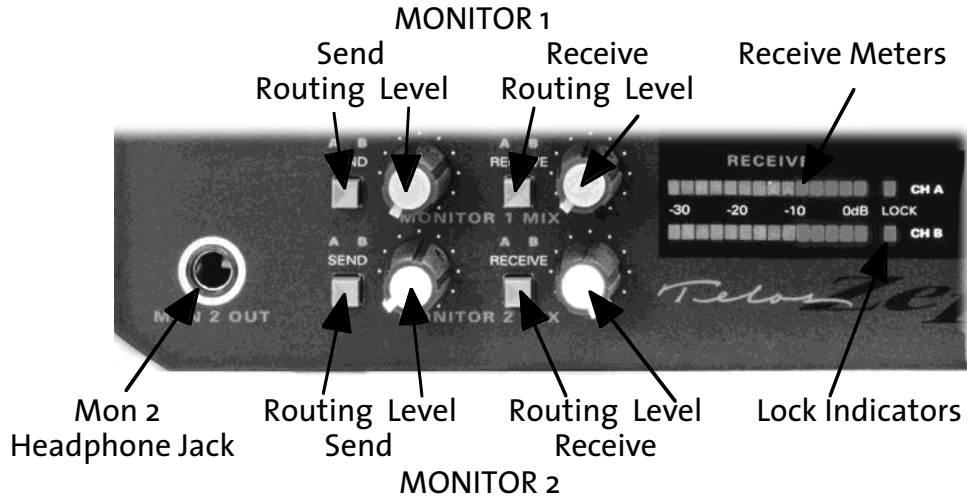
MONITOR 2

There are two independent monitor mixers.

- MONITOR 1 has a stereo headphone jack and two XLR output jacks on the rear panel. Its jacks are marked with gray circles, and its knobs have gray inserts.

- MONITOR 2 has stereo headphone jacks on the front and rear panel. Its jacks are marked with white circles, and its knobs have white inserts.

Otherwise, the two mixers are identical. The following block diagram applies to both of them.
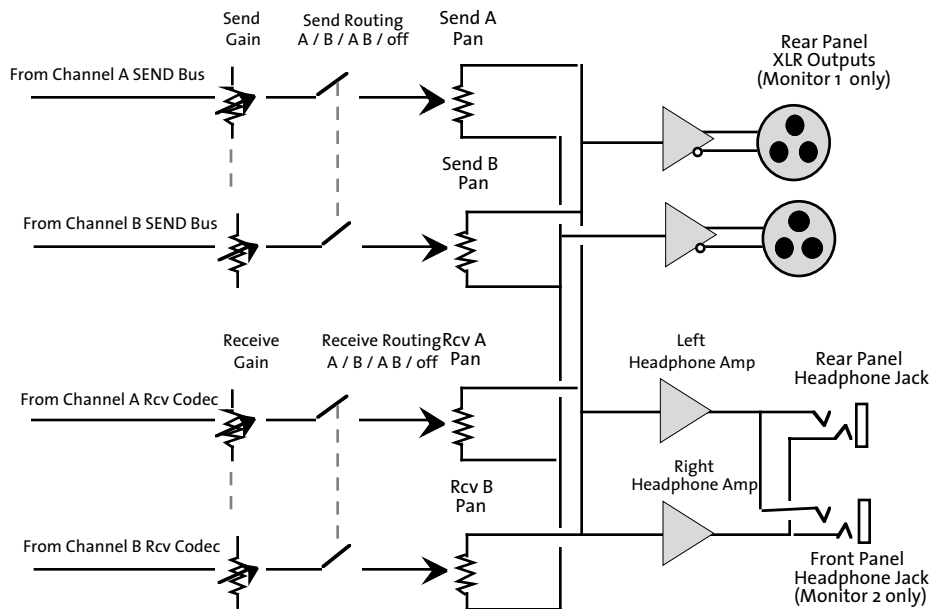


## Monitor Routing Switches

Each of the two Monitor SEND routing switches (MONITOR 1 SEND  MONITOR 2 SEND) works the same way, cycling through four choices:

- A routes the signal from the Channel A send bus to the monitor.

- **B** routes the signal from the Channel B send bus to the monitor.

- **A B** routes both channel send busses to the monitor.

- **(OFF)** disconnects the send signal from the monitor.

The two Monitor RECEIVE routing switches (MONITOR 1 RECEIVE, MONITOR 2 RECEIVE) do the same thing, only with the signal from the A and B Receive Decoders.

But these are *stereo* monitor mixers, and the individual send busses or receive channels may be mono or part of a stereo pair. So we've given you a lot of flexibility in how the signals are panned in the monitor circuits.

## About Monitor Panning

Since ZephyrExpress' mixers are digitally controlled, we were able to build in a lot of versatility.

Each of the four possible sources (A and B send bus, A and B receive codec output) is connected to *four* separate pan controls for each monitor mixer. Each control can be set to any position in the left/right field. (The controls themselves appear in the Audio menu, and are discussed below.) ZephyrExpress will use the one control that matches that source's routing switch:

- If you select **A** on the routing switch, the submenu's A setting will be used.

- If you select **B** on the routing switch, the submenu's B setting will be used.

- If you select **AB**, the mixer will use the submenu's AB setting A position for the A channel and the AB setting B position for the B channel.

The drawing that follows describes what the software's actually doing. Assume that the hidden "Monitor 1 Receive" block is identical to the Send block you can see. By the way, this is all repeated for the MONITOR 2 circuit... making a total of 16 pan pots in ZephyrExpress.
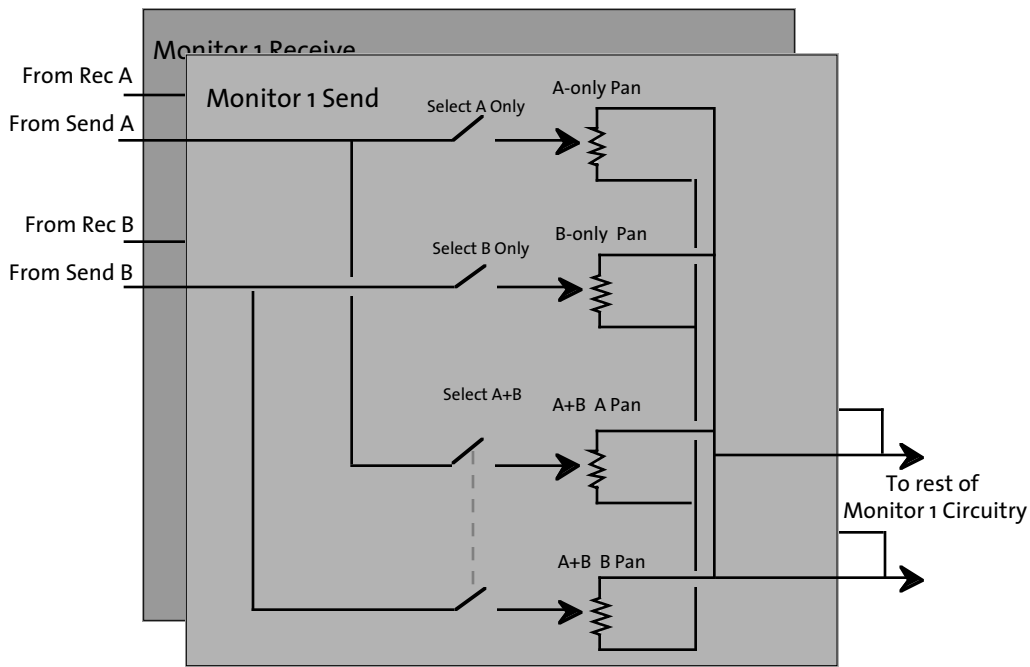
## Adjusting individual monitor panning

Scroll down the `Audio settings` screen and you'll see four submenu entries, one for each of the four monitor outputs:
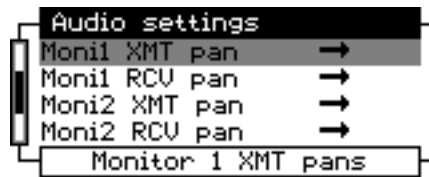


The four submenus have identical structures:
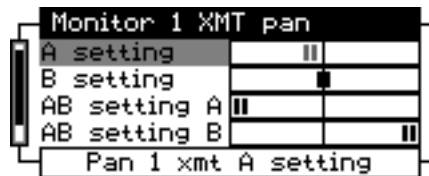


Select which setting you want to adjust by highlighting it and pressing EDIT. The position indicator ( ▮▮ ) will start flashing to indicate it's active. Turn EDIT to move it toward the left or right. If there's signal playing through that monitor, you can hear it pan in your headphones. When you've adjusted any setting's pan the way you want it, press EDIT to confirm.

Panning is continuous and can be completely arbitrary. In the screen shots above, the `A setting` is just slightly to the left of center.

You don't have to pan mono signals to the middle:

- Separating the microphones slightly to match their physical positions may make it easier to reach for the right knob when mixing a live show... even if you're sending their signals in mono to the studio.

- If you need more than two separate monitor mixes, you can use the individual stereo channels as monaural mixes and pan the signals to determine which mix will hear them.

> **HOT TIP!**
>
> The right- and left-most positions of each panning control are a switch that directs all of the signal to that side. This means you'll hear a slight jump when you start to pan from the extreme side to the center, as the opposite side kicks in at a reduced volume.

## Receive Meters
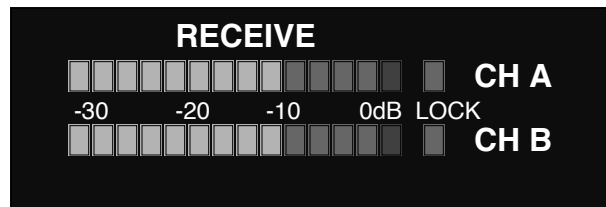


The two 14-segment RECEIVE meters have response times and calibration similar to the SEND meters (page 4-7). They report the digital signal exactly as received from the distant studio: nothing happens in ZephyrExpress that can change the levels between the codec, these meters, and the RECEIVE OUT XLRs on the back panel.

If the meter readings are too low or too high, ask the distant studio to change their send level. If the signal is occasionally lighting the red LEDs but you think you're hearing distortion — particularly on peaks or sibilance — they could be overdriving their codec's limits (many units force you to work "blind", with no send meters). Ask them to lower their console level significantly — as much as 6dB to 12dB might be necessary before you hear any improvement.

These meters aren't active when receiving a G.711 (analog telephone) call, because that technology doesn't use ZephyrExpress' main digital signal processing. The meters are totally disabled in `Far Loopback` mode, and the center LEDs light up in that mode to remind you that normal operation is impossible until you turn `Loopback Off`.

## LOCK lights

The two red LOCK LEDs next to the receive meters will light steadily when ZephyrExpress is receiving a valid signal, the distant transmitter's coding method matches the local decoder, and the sample rates match. They'll flash if both codecs are synchronized and set for Layer 3, but the Layer 3 channel modes (`mono`, `joint stereo`, and so on) don't match.

Normally both LOCK LEDs work together. If the receive mode is G.722, however, they'll operate independently to indicate the status of individual channels.

# Analog (G.711) phone calls

ZephyrExpress can send or receive standard analog (or POTS[1]) calls on either ISDN line, eliminating the need for a separate telephone and analog circuit at the remote site. The call is carried as ISDN data to the telephone company's switching office, where it's converted and treated as a standard connection.
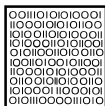
- Station engineers can call the remote site before establishing a high-quality connection, to verify coding methods and bitrates.

- A studio producer can talk to the remote engineer while a mono broadcast is in progress, to relay cues and other information. The call can be routed to the talent's headset, or just to the engineer.

> ISDN TIP!
>
> Since telcos usually charge less for a standard voice connection, using G.711 for talkback and cues can cut the cost of remote broadcasts.

- Late-running talent, stuck in traffic, can call the remote site's ISDN number from their cellphones to alleviate panic.

Only one POTS call is possible at a time.

## Analog call signal routing

The G.711 codec connects to either the Channel A or Channel B send and receive busses, depending on which ISDN line is used for the call. Normal microphones, headsets, routing switches, and level controls are used just as if the call were an ISDN broadcast. Of course, audio quality is limited by the analog phone system.

Incoming POTS audio is also routed to ZephyrExpress' internal speaker, so you can hear that a call is in progress even if you're not wearing headphones. The speaker level is controllable by the System menu's POTS volume setting, and it can be turned off by selecting 0% as the setting.

Since G.711 bypasses ZephyrExpress' main digital signal processing, analog telephone levels are not displayed on the RECEIVE meter.

## Receiving calls

ZephyrExpress automatically switches to G.711 mode when a POTS call comes in on either ISDN line. All the caller has to do is dial one of the lines' directory numbers, with any access or area codes they'd normally use for a phone call.

When a call is received, ZephyrExpress picks up on the first ring. The internal speaker chirps once, and the call timer for that line starts counting. If the POTS volume is turned

---

[1] The acronym stands for Plain Old Telephone Service. We didn't make it up.

up, you'll be able to hear the caller through the internal speaker. Otherwise, you can bring up caller audio through one of the monitor circuits as RECEIVE A or RECEIVE B, depending on which line the call came in on.

> HOT TIP!
>
> ZephyrExpress places most mono codec calls using the Channel A send bus and ISDN Line 1. It's a good idea to tell analog callers to use Line 2 — or give them only that number — so Line 1 remains free.

## Placing calls

**DIAL** To place a POTS call, press the DIAL button. In the screen that appears, select Manual…, Auto…, or Recent… and press EDIT to confirm. If you've selected Manual or Recent, a standard dialing screen will appear:

```
┌──────── Manual Dial ────────┐
│ Number:  5551212            │
│ Line: #1  Mode: [Phone]  Go!│
│      Choose call type       │
│  Ready          Ready       │
└─────────────────────────────┘
```
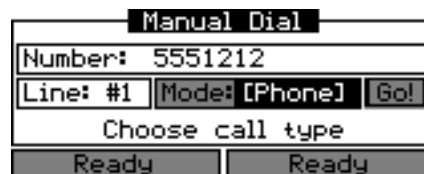
Enter the number, or select it from a list of recent calls, and tap the EDIT knob to confirm. Then turn EDIT one click counter-clockwise to highlight the Mode: field. Tap to confirm, and choose Phone as the mode. Tap again to confirm, and either choose a different Line or highlight Go! and tap to place the call.

When the distant phone picks up, ZephyrExpress will chirp and you'll be connected. The line's call timer will start counting. Remember, they won't hear you unless you've routed a mic to the send bus corresponding to that line and turned up the mic's level.

If the distant line is busy and the POTS volume is turned up, you'll hear a busy signal. DROP the connection and try again later.

## Hanging up

**DROP** Dropping a POTS call is identical to dropping a high-quality one. Press the DROP button. A screen like this will appear:

```
┌─── Select Line to Disconnect ───┐
│  Both    Line 1    Line 2        │
│  Press jog to drop line 2        │
│  Ready          00:17            │
└──────────────────────────────────┘
```
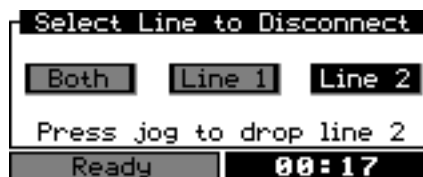
If the POTS call is the only line in use, its line will be pre-selected. Press EDIT and the call will be disconnected.

> **⚠ BE CAREFUL!**
>
> If a POTS call was placed while a high-quality transmission is taking place on the other line, Both will be pre-selected in the Drop screen. Select Line 2 before pressing EDIT, or the high-quality call will also be dropped.

Of course, a POTS connection will also be dropped if the distant phone hangs up.

After the call is completed, ZephyrExpress will disconnect its internal G.711 codec and reconnect that line to the high-quality DSP codec.

# Dealing with Delay

## Mix-Minus

All perceptual coders have too much delay for talent on remote to hear themselves via a round-trip loop. Therefore, a special *mix-minus* arrangement is required – exactly the same as has been used with satellite linked remotes for years.

The principle is this: The remote performers do not hear themselves via the studio cue return. Rather, their microphones are mixed locally with a studio feed which has everything *but* the remote audio – thus the "mix-minus" designation. The announcers hear a non-delayed version of themselves and a slightly delayed version of all of the studio pieces.

That's why ZephyrExpress' monitor circuits have separate routing and level controls for SEND and RECEIVE audio: it's a trivial matter to create a headphone mix that uses the local send signal for the microphones and a music-only signal from the remote codec. The only caveat is that the distant studio must be able to send the music-only signal, without the announcers' microphones mixed in.

The headphone mix can also be sent to a local public address system, for the benefit of a live audience at the remote. If the live audience can tolerate monaural music coming back from the studio, both the send and receive paths can be on a single ISDN line using Layer 3mono. The other ISDN channel could be used for a POTS call from the studio, to let the studio producer talk directly to the remote engineer using ZephyrExpress' second monitor circuit.

## Phones and Remotes

To save money and hassle, callers are usually received at the studio, rather than at the remote site. In this situation, phones need to be fed to the remote talent so that they can hear and respond to callers. And the phone callers need to hear the talent. In many cases, the remotes are sufficiently distant that the station can not be monitored for the caller feed. Even if it could, the profanity delay would be a problem, since the talent needs to hear the phone pre-delay.

The talent hears callers via the return path. As before, this return is fed with mix-minus: a mix of everything on the program bus minus the remote audio.

As for the second half of the equation, the callers hear the talent because the remote feed is added to the telephone mix-minus buss. No problem if you have a set-up which permits selective assignment to the phone mix-minus.

The most common problem with this arrangement is a result of a phone hybrid with too much leakage combined with the system delay. If the hybrid isn't doing a good job of preventing the send audio from leaking to its output, the special remote send mix-minus is corrupted. Remember, if any of the announcer audio from the remote site is returned via the monitor feed, it will be delayed by the digital link, causing an echo effect. The answer is to make sure you have the best possible hybrid with the maximum trans-hybrid loss. If it has variable override (caller ducking), you should increase the amount when these remotes are in progress.
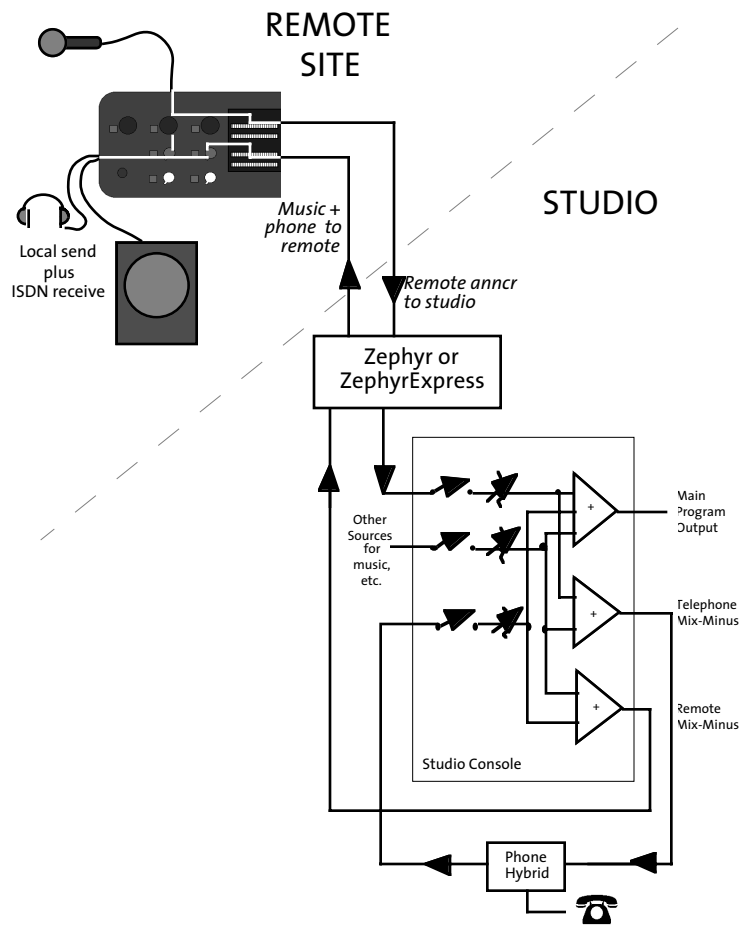
*Diagram showing system set-up for remotes with delay in the transmission path and phones taken at the studio. Note that this is the same as required for satellite links.*

Another issue worth considering is the round trip delay. The apparent on-air response time of the talent to callers' comments will be the sum of {studio-to-remote delay + remote-to-studio delay + talent's thinking time}. For this reason the studio-to-remote path will generally use the G.722 mode, sacrificing high-fidelity for speed: after all, who needs a high-fidelity line to relay a phone call? If G.722 proves too ragged to use for music or commercials being sent from the studio to a remote site's public address system, use a Layer 3mono mode for high-quality audio but send the telephone caller's audio on ZephyrExpress' second line  as a POTS signal.

Other intermediate tradeoffs are possible and will be dependent on your format. You may also be able to significantly speed up the talent's  thinking time by judicious applications of black coffee.

For information on the tradeoff between audio quality and delay, refer to the next section.

## A

analog circuits · 4-2
analog phone calls · 4-12

## C

coding delay · 4-14
coding methods · 4-12

## D

delay · 4-14

## G

G.711 · 4-12
G.722 · 4-12

## H

headphone jacks · 4-8

## L

Layer 3 · 4-5, 4-11
limiter · 4-4, 4-5
line inputs · 4-2, 4-6
LOCK lights · 4-11
low cut · 4-4

## M

meters · 4-4, 4-7, 4-11
microphone inputs · 4-2, 4-3
mixer · 4-2
mix-minus · 4-14
monitor · 4-2
monitor mixer · 4-8, 4-9
monitor output · 4-10

## O

overload indicator · 4-5

## P

panning · 4-9
phantom power · 4-3, 4-4
POTS · 4-12
power supply · 4-3

## R

routing · 4-5, 4-6, 4-8

## S

sensitivity · 4-4, 4-6
stereo pan · 4-9

## T

test tone · 4-4

# SECTION 5

# AUDIO CODING

# OVERVIEW

## Introduction to Audio Coding

Audio takes up a lot of data.

Without data reduction, CD-quality quality audio — 16 bits at 44.1kHz sample rate — requires a transmission capacity of about 705 thousand bits per second (kbps) for each audio channel. But the wires we use for remote broadcasting are on a telephone system designed for voice-grade communications: 8 bits at 7kHz sample rate, or 56kbps per channel. That's less than 8% of what we need.

> CURIOSITY NOTE
>
> You can arrive at these same numbers with nothing more complicated than grade-school math. Just multiply the sample rate by the sample depth: 44,100 samples per second * 16 bits per sample = 705,600 bits per second for CD-quality mono audio.

You can reduce the data requirements by lowering the quality somewhat. 13 bits would yield a respectable 78 dB dynamic range, certainly adequate for home listening. And a 32kHz sample rate — with careful equipment design — will give you flat response to 15kHz, the practical limit for analog FM broadcasting in North America. Unfortunately, that still leaves us with telephone data channels about 86% too small to do the job. Besides, 13 bits is an awkward bit depth for computers to deal with, and the audio it produces isn't clean enough to survive today's transmitter processors.

> CURIOSITY NOTE
>
> Bit depth and sample rate translate easily into audio specifications. Digital audio must have a sample rate of at least twice the desired bandwidth, so 15kHz audio requires (after a safety margin) 32kHz sampling. Each bit of sample depth represents slightly more than 6dB of dynamic range.

The first practical coding methods used a principle called ADPCM, Adaptive Delta Pulse Code Modulation. This takes advantage of the fact that it takes fewer bits to code the difference, or delta, between successive audio samples compared to using the individual values. Further efficiency is had by adaptively varying the difference comparitor according to the nature of the program material. G.722 and APT-X are examples of ADPCM schemes. They achieve around a factor of 4 reduction in bitrate.

G.722 achieves additional efficiency by allocating its bits to match the patterns in the human voice, and it's considered adequate for news and talk programming over ISDN. But for high-fidelity transmission, algorithms with more power are required. These are based on psychoacoustics, where the coding process is adapted to the way we hear sounds. There are several algorithms available, with varying complexity and performance levels.

Some years ago, the international standards group ISO/IEC established the ISO/MPEG *(Moving Pictures Expert Group)*, to develop a universal standard for encoding moving pictures and sound for digital storage and transmission media. The standard was

finalized in November 1992 with three related algorithms, called Layers, defined to take advantage of psychoacoustic effects when coding audio. Layer 1 and 2 are intended for compression factors of about 4:1 and 6 or 8:1 respectively, and these algorithms have become popular in satellite and hard-disk systems. Layer 3 achieves compression up to 12.5:1 — 8% of the original size — making it ideal for ISDN.

# Basic Principles of Perceptual Coding

With perceptual coding, only information that can be perceived by the human auditory system is retained.

Lossless – which, for audio, translates to noiseless – coding with perfect reconstruction would be an optimum system, since no information would be lost or altered. It might seem that lossless, redundancy-reducing methods (such as PKZIP, Stuffit, and others used for computer hard-disk compression) would be applicable to audio. Unfortunately, no constant compression rate is possible due to signal-dependent variations in redundancy: There are highly redundant signals like constant sine tones (where the only information necessary is the frequency, phase, amplitude, and duration of the tone), while other signals, such as those which approach broadband noise, may be completely unpredictable and contain no redundancy at all. Furthermore, looking for redundancy can take time: while a popular song *might* have three choruses with identical audio data that would need to be coded only once, you'd have to store and analyze the entire song in order to find them. Any system intended for a real-time use over telephone channels must have a consistent output rate and be able to accommodate the worst case, so effective audio compression is impossible with redundancy reduction alone.

Fortunately, psychoacoustics permits a clever solution! Effects called "masking" have been discovered in the human auditory system. These masking effects (which merely prove that our brain is also doing the equivalent of coding) have been found to occur in both the frequency and time domains and can be exploited for audio data reduction.

Most important for audio coding are the effects in the frequency domain. Research into perception has revealed that a tone or narrow-band noise at a certain frequency inhibits the audibility of other signals that fall below a threshold curve centered on a masking signal.

The figure below shows two threshold of audibility curves. The lower one is the typical frequency sensitivity of the human ear when presented with a single swept tone. When a single, constant tone is added, the threshold of audibility changes, as shown in the upper curve. The ear's sensitivity to signals near the constant tone is greatly reduced. Tones that were previously audible become "masked" in the presence of "masking tones," in this case, the one at 300 Hz.

All signals below the upper "threshold of audibility" curve, or Masking Threshold are not audible, so we can drop them out or quantize them crudely with the least number of bits. Any noise which results from crude quantization will not be audible if it occurs below the threshold of masking. The masking depends upon the frequency, the level, and the spectral distribution of both the masker and the masked sounds.
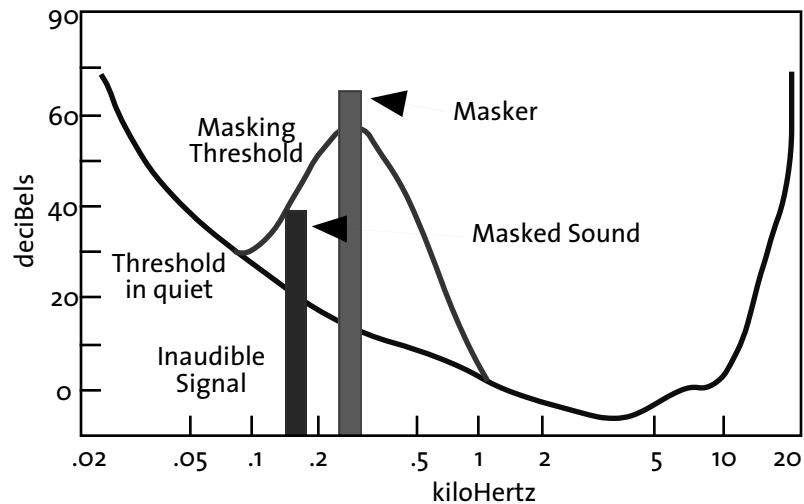
*Masking effects in the frequency domain. A masking signal inhibits audibility of signals adjacent in frequency and below the threshold.*

To benefit from the masking effects, perceptual coders use a filterbank to divide the input audio into multiple bands for analysis and processing. The maximum masked noise level is calculated depending upon the spectral content, and the available bits are allocated so as to keep the quantization noise below the masking threshold at every point in the spectrum.

While coding efficiency increases with more bands and better frequency resolution, the time domain resolution decreases simultaneously owing to an inevitable side-effect of the band filtering process: higher frequency resolution requires a longer time window – which limits the time resolution. Happily, masking works also in the time domain. A short time before and a longer time after a tone is switched on and off, other signals below a threshold amplitude level are not noticeable. Filterbanks with higher frequency resolution naturally exploit the ear's time-masking properties.
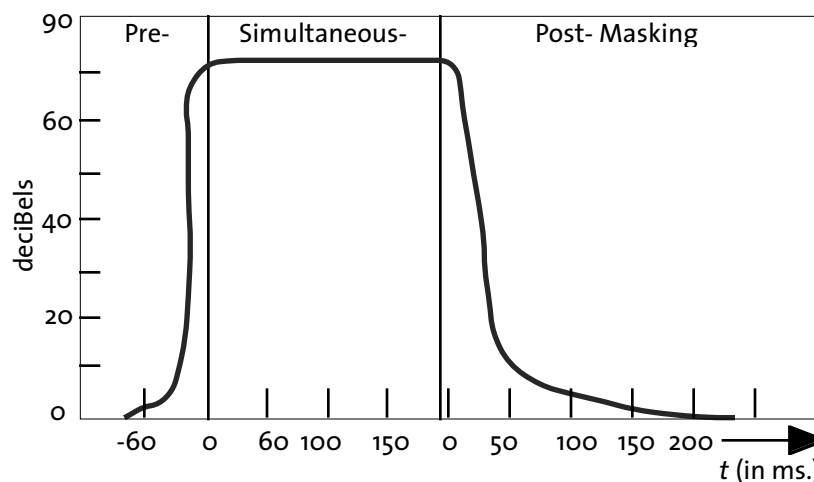


*Masking effects in the time domain. Masking occurs both before and after the masking signal.*

> ⚠️ IMPORTANT!
>
> Due to a perceptual coder's reliance on precise principles of human perception, audio to be coded should not be processed with any non-linear dynamics-processing such as clipping, multi-band compression or limiting. Wideband compression or AGC is acceptable and may be desirable if a consistent level cannot otherwise be achieved.
>
> The same is true to audio which has been decoded after passing through a perceptual coding cycle, but to much lesser degree.
>
> For more information contact Telos Systems customer support for a copy of a paper delivered at the AES by Frank Foti on this topic.

# ISO/MPEG LAYER 3

MPEG Layer 3 is the most powerful coding method available in ZephyrExpress, and we particularly like it because it is perfectly matched to the bitrates available on ISDN BRI lines.

It offers:

- 15 or 20kHz mono or stereo audio bandwidth.

- Full-fidelity mono on a single 56/64kbps channel.

- Near-CD quality stereo[1] on a single ISDN Telco circuit.

- Affordable, transparent, audio transmission for AM/FM radio or television audio.

After extensive testing by broadcasting organizations around the world under the direction of the CCIR, it has been designated as the most powerful of the three audio coding systems standardized in ISO/MPEG IS-11172. It is specifically recommended for 56 and 64 kbps channels.

## Layer 3 Features

### Psychoacoustic Masking

The audio in Layer 3 is divided into 576 frequency bands. First, a polyphase filterbank performs a division into the 32 "main" bands which correspond in frequency to those used by the less complex Layer 2. Filters are then used to further subdivide each of the

---

[1] 20kHz bandwidth, 96dB dynamic range... just like a CD. The encoding process can cause some loss that a trained listener, using carefully-selected program material and good monitors, could detect. But most people won't hear the loss, and there's no accepted way to measure it.

main bands into 18 more. At a 32 kHz sampling rate, the resulting bandwidth is 27.78 Hz – allowing very accurate calculation of the masking threshold values. Sufficient frequency resolution is available to exceed the width of the ear's critical bands (100 Hz below 500 Hz; 20% of the center frequency at higher frequencies) across the audible spectrum, resulting in better hiding of noise than would otherwise be possible.

## Redundancy Reduction

Redundancy reduction is accomplished by a Huffman coding process to take advantage of the statistical properties of the simplified signal output from the psychoacoustic stage. Values that appear more frequently are coded with shorter words, whereas values that appear only rarely are coded with longer words. This results in an overall decrease in the data rate, with no degradation, since it is a lossless reduction scheme.

Notice that this redundancy reduction process is the ideal supplement to psychoacoustic masking. In general, maskers with high tonality have more redundancy but allow less masking, while noise-like signals have low redundancy and high masking effect.
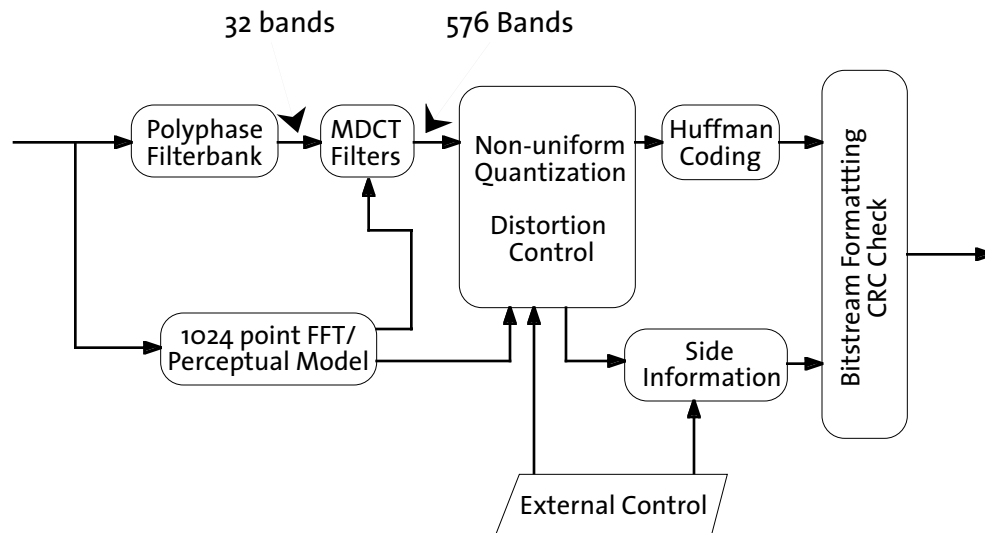


*Block diagram of the Layer III coding process.*

## Bit Reservoir Buffering

Often, there are some critical parts in a piece of music that cannot be encoded at a given data rate without audible noise. These sequences require a higher data rate to avoid artifacts. Layer 3 uses a short time "bit reservoir" buffer to address that need. Similar to a savings account, this buffer is filled in "easy times" with data bits that are not required for the actual frame. If a critical part occurs, the encoder can use the saved bits to code this part with a higher data rate.

## Ancillary Data

The bit reservoir buffer offers an interesting capability: an effective solution for the inclusion of such ancillary data as text or control signaling. The data is held in a separate buffer and gated onto the output bitstream using the bits allocated for the reservoir buffer when they are not required for audio.

### Layer 3 Joint Stereo

A joint stereo mode permits advantage to be taken from the redundancy in stereo program material. The encoder switches from discrete L/R to a matrixed L+R/L-R mode dynamically, depending upon the program content. The matrixed mode of operation takes of advantage of the usual redundancy of the "center" channel information and therefore significantly improves overall fidelity.

> WARNING!
>
> Joint stereo takes advantage of the way sounds distribute themselves over a normal stereo field. If you're transmitting two signals that don't comprise a normal stereo field — as would be the case if you're transmitting an MS matrixed signal or two totally independent audio channels — it won't save you anything.

> CURIOSITY NOTE!
>
> Layer 3 is the direct successor to the "ASPEC" algorithm, which was based on coding methods by Fraunhofer/University of Erlangen, AT&T, Thomson Consumer Electronics, and CNET (Centre National d'Edudes des Télécommunications).
>
> In the first tests for ISO in 1990, ASPEC showed the best sound quality, with an advantage at lower bit rates. The precursor to the Layer 2 algorithm — originally called MUSICAM[1] — had the advantage of lower complexity and simpler processing. L3 came from a merging of both systems: For easier transcoding, L3 adopted the L2 filterbank for the first stage, and the bitstream rate was adjusted.

## ISO/MPEG LAYER 2

MPEG Layer 2 is the world's most popular perceptual coding method, primarily because it's easier and less expensive to implement — particularly at the encoder — and practical devices using it were available earlier than Layer 3. It's a preferred choice for applications where very large data capacity is available, such as satellite links, high-capacity Primary ISDN or T1 circuits, and hard disk storage systems using Ethernet for signal distribution.

We include it in ZephyrExpress to offer compatibility with the widest variety of codecs.

Our implementation is the highest quality, using DSP code licensed directly from the algorithm's primary inventor: the IRT in Munich.

---

[1] While "Musicam" was the original European developer's name for a particular coding process, it was subsequently registered as a product trademark by a US company... forcing world-wide users to drop the name and use "Layer 2" instead. That company's products and the internationally-accepted algorithm have nothing to do with each other, except that ISO/MPEG Layer 2 is one of the processes supported by the products.

## Layer 2 Joint Stereo

The Layer 2 joint stereo mode uses an "intensity coding" method. This method has high coding power and is quite effective, but hurts stereo separation on some program material. Audio above 3 kHz or so is combined to mono and panned to one of seven positions across the stereo stage.

## Layer 2 Mono-128

ZephyrExpress lets you combine ISDN channels and use the resulting 112/128kbps data stream for a single Layer 2 monaural channel. Since a bigger data path is available, this mode uses the least compression and can give you the highest mono quality for particularly critical applications.

# G.722

This technology pre-dates perceptual coding. It is much simpler than the transform methods, but suffers from poorer audio performance. It has the benefit of low cost and the unique advantage of low delay. It has been around as an international standard the longest and is probably the most widely used system. In our view, this technology is acceptable for mono voice where high fidelity is not necessary. It is good also for cueing and intercom channels.

We have included G.722 in ZephyrExpress because:

- It has been the most popular coding method until recently so there are many of these codecs in use. Because it is a standard, codecs from various manufacturers have a good probability of being able to interwork with one-another. (We've tested with many units and have found no problems so far.)

- G.722 has the lowest delay of all popular coding methods.

This method was invented in the late 70s and adopted as a standard in 1984 by the CCITT, the *Consultive Committee for International Telephony and Telegraphy*, a division of the United Nations. The technique used is Sub-Band ADPCM, Adaptive Delta Pulse Code Modulation, which achieves data reduction by transmitting only the difference between successive samples. G.722 does this in two audio frequency sub-bands: 50-4kHz and 4kHz-7kHz.

> DEEP TECH NOTE!
>
> Only two bits are allocated per sample for audio frequencies above 4 kHz – sufficient for conveying the sibilance in voice signals, but not very good for intricate musical sounds. Also, the "predictor model" used to determine the step size in the adaptive function is designed only for speech. This is why music transmitted via G.722 has a distinct 'fuzzy' quality.

G.722 has a frequency response extending to 7 kHz with fairly poor fidelity. Unless there is no alternative, it should not be used for music.

G.722 uses a procedure called "statistical recovery timing" or "statistical framing" to lock the decoder to the data stream. (We use the procedures specified in ANSI standard T1.306-1989.) This process usually happens instantaneously, but can take up to 30 seconds.

Other strange effects may be observed. Tones and noises may be present before locking occurs, and some continuous audio tones may cause momentary unlocking. Please note this is inherent in G.722's statistical framing and is not an implementation problem with ZephyrExpress.

IMPORTANT!

The locking can be sensitive to audio present on the G.722 path, as it relies on properties of the audio itself. Some audio material and tones can prevent lock from ever happening. Silence is the most reliable signal for locking, and undistorted voice is usually OK. The most common problems are with sine tones and distorted voice or music signals, in which case turning off the audio signal — or lowering it 12dB or so — for a few seconds will generally let the system relock. In very rare cases, it may be necessary to disconnect and redial.

HOT TIP!

Another characteristic of G.722's statistical framing is that the decoder may remain locked despite serious corruption of the data, although audio fidelity will degrade. In cases where you are unable to get a good connection you may find this characteristic desirable, however you will find that the fidelity will be substantially less under these circumstances.

## CASCADING

This section is preliminary, as coder cascading is an active field of investigation among algorithm designers, standards organizations, and users. Telos urges you to be wary and to let your own ears be the final judge until better information becomes available.

Some of what we do know:

- Some recent CCIR tests have demonstrated that one pass of Layer 3 at 56/64kbps can be cascaded with 2 - 5 passes of Layer 2 operating at high (112kbps+/channel mono; 192kbps+ Joint Stereo) bitrates with good results.

- Informal tests at the Telos lab with two passes of ZephyrExpress Layer 3 have proven successful, with listeners noticing no audible degradation, even on "difficult" CDs.

- Tests with APT-X followed by one or two passes of Layer 3 or one or two passes of Layer III followed by APT-X proved to be quite acceptable.

- One user has reported that two passes of ZephyrExpress Layer 3, followed by one pass of SEDAT, is OK. (Stereo program mode.)

- The goal is to get as much "coding headroom" as possible at each stage. This is achieved when you:

  1. Use the most possible bits at each stage, with the least possible compression (for example, by lowering the sample rate, and using 64kbps rather than 56kbps connection), and/or

  2. Use the more powerful coding method of those available at each stage.

At the moment, we offer the following advice:

- Use coders only where necessary. Consider the alternatives at each stage. With the cost of hard disk capacity falling, is it really necessary to crunch at that point?

- Use the maximum bitrate you can afford at each stage. Hard disk recorders and other studio systems often have an option to adjust this. For very critical work, remember that ZephyrExpress may be used in a mode where a mono program is split over two digital network channels.

- Get the Layer 3 advantage on low bitrate channels.

The people at Fraunhofer IIS, who developed the Layer 3 algorithm, have introduced a perceptual coding analyzer. This device has the potential of making objective measurements a reality. We'll be hearing more about this.

## Mixed MPEG Layer 2 And Layer 3 Signal Chains

What about the case where you will be using Layers 2 *and* 3 in a signal chain? It turns out that the two methods are nicely complimentary.

At low bit-rates, Layer 3 gets more signal-to-mask margin than Layer 2. This is why it performs better in the low bit-rate tests. It accomplishes this by using a filter bank with more bands, 576 vs 32. One effect of this is "time spread." (More frequency resolution requires a longer time window. It's a law of physics thing...) For a small number of passes (one or two), this is good, as the ear has masking in the time domain as well as the frequency domain, and Layer 3 naturally exploits this additional dimension. The downside is that when many stages of Layer 3 are used at low bit-rates, the time spread can become audible (softening of transients and pre-echoes, mostly), and this is a bad thing. While Layer 2 does not have this problem, it has another. Because it is closer to the edge for s/n, multiple generations result in unmasking (noise and grit, mostly).

But the ISO/MPEG people do not propose that a bunch of passes of Layer 3 be used. The idea is that Layer 3 be used at ISDN/SW56 bit-rates for field pick-up and that Layer 2 be used at higher bit-rates in other parts of the signal path.

This is why the ISO group decided to recommend the Layers as they did: Layer 3 for 64kbps/channel and Layer 2 for equal to or greater than 128kbps/mono channel.

Our own experiments with codec cascading confirm that this is the right approach: the two coding methods seem to complement each other. Two passes of Layer 3 sound noticeably better than two of Layer 2; a pass of Layer 3 followed by a pass of Layer 2 also sounds better than two of Layer 2. And we've had customers who have used a pass, or two, of Layer 3 followed by SEDAT without evident problems.

# CHOOSING THE CODING METHOD FOR YOUR APPLICATION

This chart describes and compares some of the important characteristics of each method. Because ZephyrExpress includes all three popular coding methods, it's possible to choose the most appropriate one for each application.

## Audio Coding Comparison Chart

|  | __Layer 3__ | __Layer 2__ | __G.722__ |
|---|---|---|---|
| **Algorithm** | Perceptual+Huffman | Perceptual | ADPCM |
| **Audio Freq. Response/mono** |  |  | 7 kHz |
| 24 kHz sample rate, 64 kbps line | 15 kHz | 8.6 kHz* |  |
| 32 kHz sample rate, 56 kbps line | 15 kHz | 7.8 kHz* |  |
| 48 kHz sample rate, 64 kbps line | 20 kHz | 9.8 kHz* |  |
| 48 kHz s/r, 128 kbps line** |  | 20 kHz |  |
| **Audio Freq. Response/stereo** |  |  | 7 kHZ |
| at 32 kHz sample rate | 15 kHz |  |  |
| at 48 kHz sample rate | 20 kHz | 20 kHz |  |
| **Delay at HALF/24 kHz** |  |  | 45 ms |
| **Delay  at 32 kHz/mono** | 275 ms |  | 45 ms- |
| **Delay  at 32 kHz/dual mono** | 275 ms |  | 45 ms- |
| **Delay at 48 kHz/mono** | 225 ms | 160 ms | 45 ms- |
| **Delay at 48 kHz/dual mono** | 225 ms | 160 ms | 45 ms- |
| **Delay at 48 kHz/mono-128** |  | 220 ms |  |
| **Delay at 32 kHz/stereo/jstereo** | 390 ms |  | 45 ms- |
| **Delay at 48 kHz/stereo/jstereo** | 270 ms | 220 ms | 45 ms- |
| **Joint Stereo** | MS Matrix | Intensity Coding |  |
| **ISO Target Bit Rate** | 64kbps/channel | 128kbps/channel | N/A |
| **Coding Power as In/Out Ratio** | 12:1 | 6-8:1*** | 4:1 |
| **Frequency Bands** | 576 | 32 | 2 |
| **Frequency Resolution (48 kHz)** | 42 Hz | 750 Hz | -- |

* Layer 2 was not designed for the low bitrates of a single ISDN channel, so the response is deliberately limited in some Layer 2 conditions to reduce unpleasant high-frequency artifacts.

** Requires two 64kbps ISDN channels.

*** 12:1 in Intensity Joint Stereo mode

-G.722 delay will vary. Spec is 35 ms +/- 10 msec for 45 ms maximum

Delay times may vary depending upon ISDN line delay and other factors.

Frequency response is given for swept sine test; response with program material may vary *owing to the dynamic nature of the coding process.*

## Delay vs. Quality

Looking at the chart, one thing that should be apparent is that there is a trade-off between delay and audio performance. Layer 3's excellent audio performance requires a significant delay. This is because some of its power comes from the ability to analyze the audio over a relatively long period, and because the audio must traverse four DSPs in the encoder. Layer 2 requires the next longest delay, and G.722 has minimal delay.

ZephyrExpress permits the coding mode for the send and receive paths to be independently chosen, so the choice may be optimized for the specific requirements of each direction.

It is generally agreed that delays of over 10 ms make live monitoring difficult. When modes other than G.722 are used and live transmission of remote programs is required, operational methods like those routinely used with satellite links are a necessity. The manual section 4 (Audio) has more information on this topic.

> **HOT TIP!**
>
> The "round-trip" delay in a typical remote broadcast may be reduced by using the G.722 algorithm for the return cueing path and Layer 3 or Layer 2 for only the on-air direction.

## Dual vs Stereo vs Joint Stereo in Layer 3

**With one transmission path,**

- `L3 DUAL` provides mono capability.

**With two transmission paths,**

- `L3 DUAL` mode is simply two simultaneous mono channels.

- `L3 STEREO` mode compensates for any delay between the two transmission paths, but keeps the two audio channels completely independent.

- `L3 JOINT` (Joint Stereo) mode takes advantage of the redundancy which usually is present in stereo program material. The encoder switches from discrete L/R to matrixed L+R/L-R dynamically. When in L+R/L-R mode bits are allocated dynamically to these two bit-streams allowing maximum advantage be taken advantage of redundancy between the two channels.

> **HOT TIP!**
>
> In L3 Joint Stereo mode, when the program is completely mono, the full bitrate of the two transmission channels are combined, resulting in the best quality mono signal with ZephyrExpress. This could be used for particularly critical mono material. The identical signal must be fed to both send busses by setting the routing switch to AB for microphones inputs, or A+B for line inputs.

IMPORTANT!

The Layer 3 decoder in `L3 STEREO` mode requires that both ISDN lines be connected and operating in order for the decoder to function. Until both lines are present, the decoder will not output anything, even if the transmission mode at the distant studio is not stereo. Any data drop outs would cause interruption in both audio channels.

If your application requires the ability for the two channels to come and go independently, such as when they are from independent sites, you must use G.722 That is the only decode mode which supports fully independent operation.

## Dual Site Operation

It is possible to use the dual mono transmit mode to send to two receivers at different sites. Set the transmit mode to `L3-DUAL` and dial Line 1 at each site.

This topic is covered more thoroughly in section 7 (Menu Reference). Receive, if used, must be set to the G.722 mode. The Layer II/III decoder is unable to accommodate the independent signals.



IMPORTANT!

You must make the call to the telephone number for Line 1. The mono Layer 2/3 decoder will work only with signals coming in on the first line.

## Layer 2: Mono64kpbs vs MONO128; Dual vs JSTEREO

**With one transmission path,**

• The only usable L2 setting is monaural, at 64kbps (`L2 MONO64`).
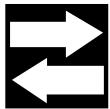


COMPATIBILITY TIP!

Layer 2 Mono64 is called simply Layer 2 Mono on Zephyr.

**With two transmission paths,**

• `L2 MONO128` provides the best Layer 2 quality by combining both transmission paths to achieve a higher bitrate. This mode requires 2 data paths, or both ISDN lines.

• `L2 STEREO` mode is simply two simultaneous mono channels, with compensated delay to preserve phase. Fidelity is identical to `L2 MONO64`, since the algorithm can't take advantage of redundancy between the channels.
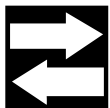


COMPATIBILITY TIP!

Layer 2 Stereo is called Layer 2 Dual on some systems.

- `L2 JOINT` (Joint Stereo) mode uses the "intensity coding" method in order to provide maximum quality for stereo program material. Delay is also compensated to preserve phase.

## Sampling Rate

This option sets the sample rate for the transmitted and received coded audio. Input/output is independent and is set in the AES menus. The Layer 3 mode may be operated at either 48 kHz or 32 kHz sample rate.

48 kHz offers lower delay and 20 kHz audio bandwidth. However, the 32 kHz rate is generally preferred for broadcast applications because no bits are wasted on frequencies above 15 kHz – which are usually not transmitted in analog broadcasting, anyway.

Layer 2 primarily operates at 48kHz sample rate. However, the `L2 HALF/24` modes allow operation at 24kHz when this sample rate is desired, primarily when communicating with "single line" capable Layer 2 codecs.

# Compatibility

## Between Telos Codecs

The table below shows which Zephyr or ZephyrExpress transmit and receive modes may operate with each other, the resulting audio bandwidth for each, and information which describes what happens to the audio channels when they are output from the decoder.

| Xmt Mode | Rcv Mode | Audio Resp. | Notes: |
|---|---|---|---|
| L3 DUAL (one Line) (Channel A) | L3 MONO (Line 1) | 15/20kHz* | Audio appears on both outputs. |
| L3 DUAL (two Lines to two sites) | L3 MONO (Line 1) | 15/20kHz* | Audio appears on both outputs at both sites. |
| L3 DUAL (two Lines to one site) | L3 STEREO | 15/20kHz* | Audio output channels correspond to input. |
| L3 JSTEREO or STEREO | L3 STEREO | 15/20kHz* | Audio output channels correspond to input. |
| L2 MONO64 | L2 | 7.8/9.8kHz** | Audio appears on both outputs. |
| L2 MONO128 (Channel A) | L2 | 20kHz | Audio appears on both outputs. |
| L2 JSTEREO | L2 | 20kHz | Audio output channels correspond to input. |
| L2 DUAL | L2 | 7.8/9.8kHz** | Audio output channels correspond to input. |
| L2 HALF/24 | L2 HALF/24 | 8.6kHz | Audio appears on both outputs. |
| G.722 | G.722 | 7kHz | Audio output corresponds to Line connection. |

\* Depending upon sampling rate: 15kHz at 32 kHz; 20kHz at 48 kHz.
\*\* Depending upon bitrate: 8kHz at 56kbps; 11kHz at 64kbps.

As well, both transmission bitrate and audio sampling rate must correspond in order for operation to occur.

## With Non-Telos Equipment

Layer 2 offers compatibility with the widest variety of non-Telos equipment. ZephyrExpress bitstream is in standard ISO format and may be used to communicate with any codec supporting this standard. In modes which require two ISDN channels, the channel-splitting (IMUX) method becomes an issue. ZephyrExpress supports the CDQ splitting scheme, so it may be used with the codecs from a variety of vendors which support this mode.

While Layer 3 is a more complex algorithm to implement and so took longer to be supported, some non-Telos boxes can now communicate with Layer 3. Our implementation follows the strict ISO/MPEG standard.

ZephyrExpress' G.722 mode offers compatibility with almost all codecs which use this coding method, and which do not use the rare H.221 framing scheme.

Section 7 (Menu Reference) includes more information that may be helpful in picking a standard for compatibility. The Appendix contains a section detailing known problems with non-Telos equipment.

# MENU REFERENCE

# The Menu System

ZephyrExpress' operating parameters are set through a series of nested menus, organized by function. Many menu items are "live" — changing the value makes an immediate change in ZephyrExpress' behavior — and every item is supported by specific, unique online HELP. To find out what a item menu does (without reading this section), highlight it and press the HELP button.

HOT TIP!

You can pre-program ZephyrExpress to simplify operation, so a field engineer never has to open any menus — even if the unit is used at multiple sites, with different codec and ISDN requirements. See "Master Setups", page 6-7.

You can lock any or all menus so they can't be accidentally changed in the field. See "Safe Mode", page 6-19.

IMPORTANT NOTE!

If a menu item is locked, you'll see the message `This menu is disabled in SAFE mode for your own security!` when you try to change it. Don't be offended.

Instead, take a small screwdriver or ball-point pen[1] and flip the SAFE switch on the rear panel. The red SAFE light on the front panel will go out, and you'll be able to change the menu settings.

## A typical menu

Takes you to next higher menu

Scrollbar

Menu Name

Higlighted item (gray)

This item is a submenu

This item is a setting

Description of highlighted item

Menus are navigated by turning the EDIT knob to scroll up and down, highlighting each item in turn. Most menus have more than four choices, so there may be additional lines scrolled off the top or bottom. The Scrollbar on the side of the screen moves up and down like an elevator car, to show you how far you've scrolled through the available lines.

When you highlight an item, it turns gray.

---

[1] Don't use a pencil. The graphite tip can break off and cause shorts inside ZephyrExpress. This happened to one of our early field units, and it wasn't pretty.

- If the item is all text or numbers (such as `Line sens   Profess`), it directly changes this operating parameter. Select it by pressing EDIT, and you can change its setting..

- If the item has an arrow icon (such as `Microphones ➡` ), it opens up a submenu when you press EDIT. Here you'll find additional choices related to the item (in this case, you can set each microphone's sensitivity, phantom power, and so on).

- The top item in any menu is an bent arrow, facing up (`••➜`). When you select this item, it takes you back to the menu immediately above.

- The second item in most menus is a Setting (`Default` in the picture). Selecting this item changes all every parameter below it to a preset value. Settings are programmed at the factory or by the user. This feature is described fully on page 6-6.

Once you've highlighted the item you want to change, tap EDIT in toward the panel. The item will turn black to indicate it's selected.

```
┌─────────────────────────┐
│ Audio settings          │   Selected item (black)
│ ••➜                     │
│     [Default*]          ▶
│ Microphones       ➡     │
│ Line sens     Profess   │   Brackets indicate this
│ ─ Activate audio setup ─│   setting is currently active
└─────────────────────────┘
```
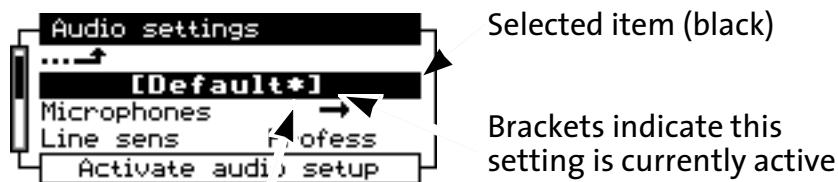
Asterisk indicates factory preset

If you turn EDIT when an item is selected, you can adjust the item's setting. In the picture above, a Setup is selected: turning EDIT brings up a list of other Audio setups. If you had selected `Line Sens`, turning EDIT would  it would let you choose `Profess` or `Consumer` sensitivity.

- While you're going through the possible settings for any item, brackets (such as `[Default*]` for the audio setup) indicate the currently active value.

- Possible settings may be on either side of the current one. Turn EDIT both ways to see them all.

- If the name of a preset has an asterisk, it was programmed at the factory.

When you've found the right setting for a selected item, press EDIT to confirm. This changes that item to match the setting.

## Navigation Shortcuts

- From any menu, press and hold EDIT in toward the panel for more than three seconds, and you'll jump to the parent menu. Continue holding EDIT in, and you'll keep jumping to higher levels until you reach the Status screen.

- Hold EDIT in while you turn it one click clockwise, to jump to the bottom item on any menu. In most high-level menus, this is the `setups` item.

- Hold EDIT in and turn it one click counter-clockwise, to jump to the top item on any menu. In menus, this is ⋯⬆ , taking you to the next higher menu or the Utility screen. In the Utility screen, this takes you back to the Status screen.

> **HOT TIP!**
>
> The shortcut above means you can hold EDIT in and spin it *many* clicks counter-clockwise to return to the Status screen... no matter how deep you are in the menu system. This is often the fastest way to escape from menus.

## Getting HELP

ZephyrExpress' context-sensitive HELP system is always aware of which menu item you've highlighted, and will report complete details on how that item works and its possible settings.

For example, if you highlight the `Codec: Compatibility` item...

```
┌─────────────────────────┐
│ Codec Settings          │
│ Bitrate        64kbps   │
│ Sample Rate    32kHz    │
│ Compatibility  Zephyr   │
│ Ancill Data    Normal   │
│   Codec compatibility   │
└─────────────────────────┘
```
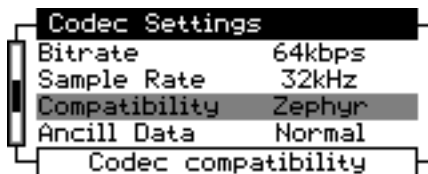
...and then press HELP , you'll see a description of how to choose the right Compatibility setting:

```
┌─────────────────────────┐
│ Help: Compatibility  ▲  │
│ Zephyr Use this mode for│
│ connections with other  │
│ Zephyr units.           │
│ Dialog4 This will enable│
│   Press jog to return ▼ │
└─────────────────────────┘
```
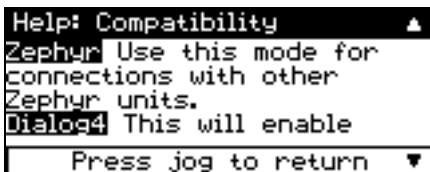
Most Help screens are larger than four lines. A white arrow in the top line (▲) means you can scroll up. A black arrow in the bottom line (▼) means you can scroll down.

> **HOT TIP!**
>
> When the Status screen is showing, doing anything at all to the EDIT knob — pressing it, or turning it in either direction — will display general help.

When you're finished looking at Help, press EDIT toward the panel to return to where you were.

> **WE'RE NOT FORTUNE TELLERS...**
>
> Telos Systems is continuously updating and improving ZephyrExpress' features, so this manual —based on software 1.2.1 — might not accurately reflect every-thing you see on your screen.
>
> Even if your manual is older than your software, items that are described will work according to the manual. And you can always press HELP while highlight-ing anything that was invented after we wrote this book.

# Menu Overview

Utility Menu

**Codec**
Transmit
Receive
Bitrate
Sample Rate
Compatibility
Ancill Data
Status Out
Loop Mode
Contacts Out
Codec Setups

**ISDN**
Switch type
SPID #1
SPID #2
DN #1
DN #2
MSN #1
MSN #2
Outside Line
Panic Dial

**System**
LCD Contrast
LCD Backlight
Click Volume
POTS Volume
Menu Timeout
Baud Rate
Set Time
Software
About...

**Audio**

**Microphones**

**Mic 1**
Sensitivity
Phantom Power
Low Cut Filter
Test Tone

**Mic 1**
Sensitivity
Phantom Power
Low Cut Filter
Test Tone

Limiter Bypass

Line Sensitivity

**Safe Mode**
Codec
Audio
ISDN
System
Channel Assign
Manual Dial
Autodial

**Mon 1 Xmt Pan**
A Setting
B Setting
AB setting A
AB setting B

**Mon 1 Rcv Pan**
A Setting
B Setting
AB setting A
AB setting B

**Mon 2 Xmt Pan**
A Setting
B Setting
AB setting A
AB setting B

**Mon 2 Rcv Pan**
A Setting
B Setting
AB setting A
AB setting B

Master

Autodial

# Using Setups



```
Codec Settings
...➜
        L3 ■ono*              ◀———  Codec Setup
Transmit        L3-DUAL◀            Items
Receive         L3-MONO◀
   Activate codec setup
```

The first text item in any menu is an automatic `Setup` function, which applies pre-programmed values to all the items in that menu. Setups can be preset at the factory or created by the user. Factory-built setups are indicated with an asterisk at the end of their names. Custom setups are saved through a menu item at the bottom of each menu.

In the picture above, `L3 mono*` is a factory-supplied setup. Like any other codec setup, it changes all of the items in the Codec menu — not just `Transmit` and `Receive`, but also the `Bitrate`, `Sample rate`, and other items you'd see if you scrolled down.

> **HOT TIP!**
>
> Custom setups are a great way to change all the items in a menu quickly as you use ZephyrExpress for different kinds of remotes. If you create a custom ISDN setup for each venue where ZephyrExpress is used, you won't have to worry about SPIDs or other critical numbers being mis-entered in the field.

## Applying a Setup

To apply a setup, highlight the setup name at the top of a menu and press EDIT to select it. Then turn EDIT to scroll through the possible setups for that menu. The currently-applied setup will have brackets around its name. Be sure to turn EDIT in both directions: there may be setups on both sides of the current one.

When you find the desired setup, press EDIT to confirm. You'll see a confirmation screen with the message `Do you really want to activate this setup?` Turn EDIT to highlight `Yes`, and press it to confirm.

You can still change any menu item's setting manually, even after applying a setup.

## Creating or Changing a Setup

> **HOT TIP!**
>
> You can use Master Setups — described on page 7 — to change multiple Setups simultaneously.

The last item in any menu is the `Setups` submenu. Before selecting it, make sure the menu has all the items set the way you'd like them stored for the new or modified setup. Then select `Setups`. You'll see a screen like this:

```
┌─────────────────────────────────┐
│    Edit codec setups            │
│  ...⬆                           │
│ Action [Create new] Go!         │
│ Name                            │
│      Choose edit action         │
└─────────────────────────────────┘
```

Select the `Action` field and choose from `Create new`, `Update`, `Activate`, `View`, or `Delete`. Then confirm.
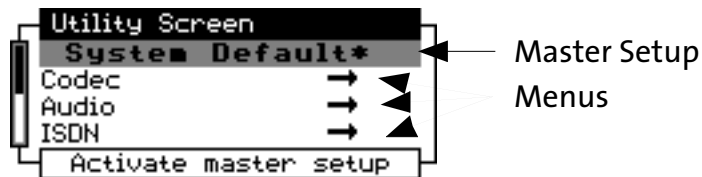
If you're creating a new setup, you'll then be able to enter a name using the keypad. After you confirm the name, the `Go!` box will be selected. Press EDIT again to confirm, and all the current settings for this menu will be stored with that name.

If you're updating an existing setup, the Name field will display the existing setups for this menu. Select one and the `Go!` box will be selected. Press EDIT again, and — after a confirmation screen — all the values for this setup will be changed to match the current menu settings. You can't update a setup that was programmed at the factor (indicated by an asterisk in its name).

If you're Deleting a setup, select it the same way. When you confirm `Go!`, you'll see the message `Do you really want to delete this setup?` Select `Yes` and confirm. You can't delete a setup that's used in a Master setup (below); delete or change that Master setup first. You also can't delete a setup that was preset at the factory.

You can also use `Activate` to apply a setup from this screen, but it's faster to use the `setup` item at the top of the menu.

## Master Setups

```
┌─────────────────────────────────┐
│ Utility Screen                  │
│    System Default*         ◀────── Master Setup
│ Codec          ➡ ◣              │
│ Audio          ➡ ◣              │   Menus
│ ISDN           ➡ ◣              │
│    Activate master setup        │
└─────────────────────────────────┘
```

The top item in the Utility screen is the Master setup It works like individual menu setups, but can change all the menus at once. You can use factory-supplied Master setups or create your own.

┌─────────────────────────────────────────────────────────────────┐
│ HOT TIP!                                                          │
│                                                                   │
│ Master setups don't have to change every menu, so you can create  │
│ your own for particular kinds of remotes — say, `Talkshow` or     │
│ `News Event` or `Concert` — that modify just the `Audio` and      │
│ `Codec` menus without affecting the `ISDN` setups for a venue.    │
└─────────────────────────────────────────────────────────────────┘

## Applying a Master Setup

Highlight the top line of the Utility screen and press EDIT to select it. Then turn EDIT to scroll through the possible Master setups. The currently-applied setup will have brackets around its name. Be sure to turn EDIT in both directions: there may be setups on both sides of the current one.

When you find the desired setup, press EDIT to confirm. You'll see a confirmation screen with the message `Do you really want to activate this setup?` Turn EDIT to highlight `Yes`, and press it to confirm.

You can still change individual menu items setting manually, even after applying a Master setup.

> HOT TIP!
>
> Auto-Dial Setups (page 10) can include a Master Setup. Then, when you select that Auto-Dial number the Maser Setup will be applied automatically.

# Creating or Changing a Master Setup

The next-to-last item in the Utility screen is the `Master Setups` menu. Master setups can be created automatically, capturing the current setups of every ZephyrExpress menu, or they can be created by manually selecting just the menus you want this Master setup to control.

If you're intending to auto-create a Master setup, make sure all of ZephyrExpress' menus are set the way you'd like them before selecting the Master Setups menu.

When you select `Master Setups`, you'll see a screen like this:

```
┌─────────────────────────────┐
│     Edit master setup       │
│  ┌──┐                       │
│  │...↵│                      │
│  ├────────────────────────┐ │
│  │Action [Auto crea...]│Go!│ │
│  ├────────────────────────┘ │
│  │Name                     │ │
│  ├─────────────────────────┤ │
│  │   Choose edit action    │ │
│  └─────────────────────────┘ │
└─────────────────────────────┘
```

Select the `Action` field and choose from `Create new`, `Auto create`[1], `Update`, `Activate`, `View`, or `Delete`. Then confirm.

## Auto-creating a new Master Setup

The easiest way to create a Master Setup is to let ZephyrExpress do it for you. If you confirmed `Action: Auto create`, you'll then be able to enter a name for this new Master Setup. When you then select the `Go!` box and confirm, ZephyrExpress will scan each menu. If a menu setup is currently active, it will note which one; if you've manually entered settings for that menu, it will create a new menu setup with the same name as the new Master Setup. Then it will store all of the individual menu setups as the new Master.

> NOTE!
>
> Auto-created Master setups change every ZephyrExpress setting when they're applied. You can't Auto-create a Master that will just affect a few menus and leave others unchanged. You *can*, however, update previously auto-created Master setups to ignore particular menus.

---

[1] May be appear with brackets as `[Auto crea...]`, if it's the last way you used this screen.

## Manually creating a Master Setup

You can also create Master Setups by picking and choosing individual setups, letting the system auto-create setups only for specfic menus, and telling the system not to change other menus when the Master Setup is applied.

After you select `Create New` and enter a setup name, the screen will scroll down to show a list of ZephyrExpress menus, such as `Codec` in the screen shot below:

```
┌──────────────────────────────┐
│     Edit master setup         │
│ ┌──────────────────────┬────┐ │
│ │Action  Create new    │Go! │ │
│ ├──────────────────────┴────┤ │
│ │Name  CITY HALL            │ │
│ ├───────────┬───────────────┤ │
│ │Codec      │   (custom)    │ │
│ ├───────────┴───────────────┤ │
│ │   Choose codec setup      │ │
│ └───────────────────────────┘ │
└──────────────────────────────┘
```

Choose a menu setup to be stored with this Master setup. You can select from

•    Any existing setup for that individual menu;

•    or `(custom)`, which will create a new menu setup with the same name as the Master setup;

•    or `(none)`, which will tell the Master setup to leave that menu's settings unchanged.

After you've selected what this Master setup should do for each menu, select the `Go!` box. You'll be shown a confirmation screen; select `Yes` to store the setup.

## Updating an existing Master Setup

After you select an existing Master setup in the `Name` field, you be given a scrolling list of menus identical to the one used when manually creating a Master setup. Follow the instructions above to select an existing menu setup, a custom one, or none.

You can't update a Master Setup that was preset by Telos ( indicated with an asterisk in its name).

## Deleting a Master Setup

After you select an existing Master setup in the `Name` field, select the `Go!` box. Then select `Yes` in the confirmation screen to delete the Master setup.

Deleting a Master Setup does not delete individual menu setups, even if they were auto-created when you created the Master.

You can't delete a Master Setup that was preset by Telos ( indicated with an asterisk in its name). We're very retentive about these things.

# Autodial Setups

The last item in the Utility screen lets you pre-program numbers for auto-dialing (see the dialing instructions in Section 3, Hardware). When you select it, you'll see a screen like this:

```
┌──────────────────────────────────┐
│      Edit autodial setups         │
├─┬────────────────────────────────┤
│ │ ...──┘                          │
│ ├────────────────────────┬──────┐│
│ │Action  Create new      │ Go!  ││
│ ├────────────────────────┴──────┤│
│ │Name  MAIN STUDIO              ││
├─┴────────────────────────────────┤
│         Enter setup name          │
└──────────────────────────────────┘
```

In the `Action` field, you can select `Create new`, `Update`, `View`, and `Delete`.

## Creating a new Autodial

If you select `Create New`, you'll be able to enter a `Name` for the setup  This name will appear in the scrolling list when a user selects `Autodial` from the DIAL button.

After you confirm the name, you'll see scrolling fields where you can enter telephone `Number1` and `Number2` for the two lines, a `Call type` for each line (either `Zephyr` or analog `Phone`), and a `Master` setup to be invoked when this autodial entry is used. You may enter only one phone number if you prefer. You may set `Master` can be set to `(none)` if you don't want to  automatically invoke a Master setup when the Autodial is used.

After you confirm the `Master`, the `Go!` box will be selected. Press EDIT to store the setup.

## Editing or deleting an Autodial

If you selected `Update` as the `Action`, the `Name` field will display a scrolling list of existing Autodial setups. Select one, and then change data as necessary in the scrolling fields below it. Then select the `Go!` box and confirm.

If you selected `Delete` as the `Action`, the `Name` field will also display a scrolling list. When you select one, the `Go!` box will be selected. Press Edit to confirm, and select `Yes` in the confirmation screen that appears. You can't delete a factory-preset Autodial: these are Telos' ISDN test lines, and we want them to be available for every user.

# Menu Details

## Codec menu

The first item in this menu is the Codec Setup. See page 6-6.

— Transmit      Selects the transmit coding algorithm[1]:

                     L3-DUAL

                     L3-STEREO

                     L3-JOINT

                     G.722

                     L2-MONO64

                     L2-MONO128

                     L2-DUAL (may appear as L2-STEREO)

                     L2-JOINT

                     L2-HALF/24

— Receive      Selects the receive coding algorithm[2]:

                     L3-STEREO

                     L3-MONO

                     G.722

                     L2

                     L2-HALF/24

— Bitrate      56kbps or 64kpbs per ISDN channel[3]. Both lines will use the same rate. In most modes, ZephyrExpress will automatically adapt to the incoming bitrate when accepting a data call. The bitrate can be changed while a call is in progress, but audio will be lost while the system re-synchronizes. If the bitrate is manually changed while a call is in progress, ZephyrExpress can't auto-adapt and audio will be lost until it's changed back.

---

[1] The distant receiver must be compatible. You can use the DUAL mono or G.722 settings to send to two sites at once. Details on choosing an appropriate algorithm appear in Section 5, Coding.

[2] The distant transmitter must compatible. ZephyrExpress can use different receive and transmit algorithms at the same time.

[3] We refer to a per-line bitrate, even for two-line calls. Some other manufacturers refer to the total bitrate of 112kbps or 128kpbs for two-line calls. It means the same thing.

**ISDN TIP!**

Some long-distance connections don't support 64kbps. If ZephyrExpress won't lock under these circumstances, both you and the distant site should change to 56kbps.

— `Sample Rate`  `32kHz` or `48kHz`. Codec sampling rates can be selected for Layer III (32kHz or 48kHz) and Layer II (48kHz only). Layer II can also operate at 24 kHz using the special L2 HALF/24 transmit and receive modes. Layer III provides 15kHz audio at 32kHz sampling and 20kHz audio at 48kHz sampling. 32kHz sampling is recommended for Layer III because it results in fewer coding artifacts. You are in the best position to judge if the wider audio response of 48kHz sampling is appropriate for your application and you should feel free to experiment.

The sampling rate for G.722 (16kHz) is automatically configured by Zephyr.
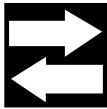
**COMPATIBILITY TIP!**

If incoming audio sounds pitch-shifted, ZephyrExpress' sample rate doesn't agree with the distant studio's. Change it to match... or enjoy the effect.

— `Compatibility`  `Zephyr` — Use this mode with most other modern codecs
`Dialog4` — This mode enables the a frame slipping/repetition feature to avoid occasional dropouts when receiving from most Dialog4 codecs
`Slimline` — This also enables the frame slip feature, and inverts the Layer 3 bitstream data to match Dialog4 Slimline codecs

— `Ancill Data`  For future software options

— `Status Out`  Selects the condition which activates pin 3 of the parallel port[2]:
`DecLock` When decoder is locked
`Line 1` When ISDN Line 1 connects
`Line 2` When ISDN Line 2 connects
`Line 1&2` When *both* Lines 1 and 2 are connected
`Line 1or2` When *either* Line 1 or 2 connects

---

[1] Details on this feature are in Section 6, System Functions. Connection details are in Section 3, Hardware.

[2] See previous footnote.

—Loop Mode          Used for testing (see Troubleshooting, section 9)
                    Off (Normal operation is possible)
                    Far (Incoming ISDN immediately routed back to ISDN line)
                    Near (Local encoded audio immediately routed to local decoder)

⚠ **IMPORTANT NOTE!**

Normal operation is impossible unless Loop Mode is Off. When this mode is on, [Loop Mode] is displayed in the upper right of the Status Screen.

The level meters are disabled in Loop Mode: Far, and the middle LED of each is lit as a reminder.

—Contacts Out       Lets you program the parallel contacts for testing[1]
                    Normal (Contacts reflect incoming Ancillary Data)
                    0000, 0001, 0010, 0011...
                    ...1111 (Any 4-bit pattern can be forced to parallel contacts when you press EDIT)
—Codec setups       See Setups, page 6-6.

## Audio **menu**

The first item in this menu is the Audio Setup. See page 6-6.

—Microphones        Opens the Microphones submenu

  — Mic 1          Opens the Microphone 1 submenu

    —Sensitivity        Selects the nominal level for a full scale reading on the meters with the microphone's level control fully up.
                    High +60dB for most ribbon mics
                    Mid +45dB for most dynamic and condenser mics
                    Low +30dB for hot condenser mics or dynamic mics in very loud surroundings.
                    Line +1dB to use the input with line-level sources.

👊 **HOT TIP!**

The MIC 1 routing LEDs also function as preamp clipping indicators. If they're lighting up in red, lower the sensitivity.

[1] See two footnotes ago.

—`Phantom Power` Applies +48vdc in a phantom configuration[1].

        `On`

        `Off` Phantom power is automatically turned off when you select `Sensitivity: line`.

> ⚠️ **WARNING!**
>
> If you apply phantom power to an unbalanced microphone, or to a mic with shorted wiring or cables, you may damage the microphone and ZephyrExpress. Don't use phantom power unless you're using a condenser microphone that you're sure is wired for it.

—`Low Cut Filter` Applies a gentle filter[2] to reduce wind noise, popped plosives, and building rumble. It's a good idea to leave this on for all voice applications, and turn it off for music transmissions.

—`Mic 1 test tone` Substitutes an 860Hz sine wave for the microphone's signal, for testing the ISDN connection and setting nominal volumes through the broadcast chain. The tone's routing and level are adjusted by the front-panel MIC 1 controls.

— Mic 2

  —`Sensitivity` See Mic 1 submenu, above.

  —`Phantom Power`     "

  —`Low Cut Filter`     "

  —`Mic 2 test tone`     "

—`Limiter bypass` The limiter is applied to both microphones at once to preserve stereo imaging.

— Line Sens     Both line inputs are switched simultaneously.

    `Professional` (+4dBu nominal)

    `Consumer` (-10dBu nominal).

> 🔥 **HOT TIP!**
>
> The line inputs remain electrically balanced even when `Consumer` is selected. You can use this to support long cable lengths (with a transformer at the other end), or for phase inversion. For unbalanced wiring details, see Section 5 Hardware.

---

[1] See Section 4, Audio.

[2] -3.75 dB @ 100 Hz; -10 dB @ 40 Hz.

—Mon1 XMT pan    Opens the Monitor 1 Send panning submenu.

    —A setting    Sets how the Send A bus signal is panned when Monitor 1's A-only
                      routing is used.

    —B setting    Sets how the Send B bus signal is panned when Monitor 1's B-only
                      routing is used.

    —AB setting A       Sets how the Send A bus signal is panned when Monitor 1's
                      AB routing is used.

    — AB setting B       Sets how the Send B bus signal is panned when Monitor 1's
                      AB routing is used.

> **HOT TIP!**
>
> ZephyrExpress' monitor panning is a sophisticated feature designed to support complex routing and mix-minus configurations in the field. Read about it in Section 4, Audio.

—Mon1 RCV Pan    Opens the Monitor 1 Receive panning submenu. See above.

—Mon2 XMT Pan    Opens the Monitor 2 Send panning submenu. Ditto.

—Mon2 RCV Pan    Opens the Monitor 2 Send panning submenu. And again.

—Audio Setups    See Setups, page 6-6

## ISDN menu

The first item in this menu is the ISDN Setup. See page 6-6.

—Switch Type    Selects the ISDN switch protocol, and must match information given
                to you by the telephone company[1].
                Ntl.I-1     National ISDN -1, the most common switch in the US.
                PTP         AT&T Point-to-Point; also for DMS-100 Custom ISDN.
                ETS300      European ISDN

---

[1] ZephyrExpress will not be able to communicate with an ISDN line if this information is wrong. See section 7, ISDN, for complete details.

ISDN TIP!

When you confirm a change of `Switch Type`, ZephyrExpress tries to resynchronize the line.

If you're sure you've entered the correct type but the Status screen doesn't show `Ready Ready`, try changing the type to something else and confirming. Then change it back and confirm again: this often fixes a balky line.

— `SPID#1`  Service Profile Identification Number. Required only by Ntl.I-1.

— `SPID#2`  "

— `DN#1`  Directory Number (7 digits).

— `DN#2`  " Some circuits will use the same number for both 1 and 2.

— `MSN#1`  Multiple Subscriber Number. Used with Euro ISDN to control which number will ring this line.

— `MSN#2`  "

ISDN TIP!

Only one category of numbers —  SPID, DN, or MSN — should generally be used. The others should remain blank.

ISDN TIP!

These numbers usually don't have punctuation. If the installer has given you numbers with hyphens or parentheses, enter the numerals only.

— `Outside line prefix`  Number to be automatically applied before dialing. Used when an internal phone system requires an access code — such as 9 — to reach outside numbers.

HOT TIP!

You can also use the `Outside line prefix` to pre-dial a "1010-xxx" code for an alternative long distance carrier. But if you won't be able to dial local numbers until you remove this prefix.

| | |
|---|---|
| — `Panic dial` | Lets you choose a dialing setup to be activated when pin 7 of the parallel remote control is pulled to ground. Pin 7 must be held to ground until you want to drop the call. |

> **HOT TIP!**
>
> It's not just for panics. You can use the Panic Dial feature for automatic backup of a satellite or alternative ISDN system by connecting it to that system's Lock output. Or connect it to a timer for automatic activation at repeating unsupervised events (such as daily press conferences or meetings), or to an SPST switch for one-touch access to the studio.

| | |
|---|---|
| — `ISDN Setups` | See Setups, page 6-6 |

## System **menu**

The first item in this menu is the System Setup. See page 6-6.

| | |
|---|---|
| — `Contrast` | Adjusts the LCD display's contrast for different viewing angles |
| — `Backlight` | Adjusts the LCD display's backlighting |

> **ADJUSTING A BLANK SCREEN**
>
> If someone has left the `Contrast` or `Backlight` at a position where the screen is invisible, you'll have a hard time finding these menu items to reset them.
>
> To reset contrast from any screen, press DIAL and hold that button in while you turn EDIT.
>
> To reset backlighting from any screen, press DROP and hold it in while you turn EDIT.

| | |
|---|---|
| — `Click Volume` | Adjusts the volume of all of ZephyrExpress' button and menu response sounds. Can be set to 0 for quiet operation. |
| — `POTS volume` | Adjusts the volume of the internal speaker when ZephyrExpress is connected to an analog call. Can be used as an alert for incoming calls[1], or set to 0 for quiet operation. |
| — `Menu timeout` | Sets the length of inactivity before ZephyrExpress reverts to the next higher-level menu. The timer starts again at the higher menu, and will keep reverting until it reaches the Status screen. |

---

[1] You might need the alert because ZephyrExpress picks up on the first ring... so there won't be a long ringing sound.

```
15 seconds
30 seconds
1 minute
5 minutes
30 minutes
Never
```

— Baud rate    Sets the rate in bits per second[1] for RS-232 remote control. The serial format is always 8 bits, no parity, one stop bit (8N1). See section 6, System Level functions.

— System Setups  See Setups, page 6-6

— Set time    ZephyrExpress' clock will keep track of the time and date in US or European styles, depending on how you enter it. To set the clock, select this item and enter the current date and time according to the instructions below. Hours are in military (24-hour) format. You may enter the year as two digits if you prefer; the Y2K rollover is hardly relevant to ZephyrExpress. You may omit entering the seconds if you prefer.

**To set the clock for US-style dates**, enter the date and time using this format. The pound sign — # — is used as the separator.

> `mm#dd#yyyy#hh#mm#ss`    For example, 1:00 PM on December 25, 1999, would be entered as `12#25#1999#13#00#00`, or — if you shorten the year and omit the seconds — as `12#25#99#13#00`.

**To set the clock for European-style dates**, enter the date and time using this format. The star sign — ✳ — is used as the separator:

> `dd✳mm✳yyyy✳hh✳mm✳ss`    For example, 11:30 AM on January 5, 2000, would be entered as `05✳01✳2000✳11✳30✳00`, or — if you shorten the year and omit the seconds — as `05✳01✳00✳11✳30`.

— Software    The first four choices of this item are used when installing new software versions. See section 6, System Functions, or the instructions that are shipped with the software.

This item also includes two choices for resetting ZephyrExpress in case of system problems:

`Reboot` — This reloads the software without restarting the hardware ("warm boot").

`Hart Reset` — This restarts the hardware and reloads the software ("cold boot").

---

[1] Most people confuse Baud rate, used in low-speed modems, with bits per second. But they're not the same thing. Bits per second is correct in this application — goodness, you'd expect *us* to know that — but we couldn't fit it on the LCD.

When you select either choice, you'll see a confirmation screen. Select `Yes` to reset ZephyrExpress.

— `About...`   When you select this item, you'll see a screen like this:

```
About ZephyrExpress
ZephyrExpress
V1.2.1/2.36 (main)
16.07.1998 12:39

      Press jog to return   ▼
```

`V1.2.1/2.36` is the version number. `16.07.1998 12:39` reflects when the software was compiled.

If your ZephyrExpress' `About` screen shows a lower version number, you have old software. If yours shows a significantly higher version number, this manual is probably out-of-date. In either case, contact Telos customer support.

## `Safe mode` **menu**

The first item in this menu is the Safe Mode Setup. See page 6-6.

The other items in this menu can be set to `Block` or `Allow`, to determine whether a user can change the settings for that item when SAFE MODE is turned on from the back panel. See section 3, Hardware, and the front of this section for details.

| | |
|---|---|
| — `Codec` | Locks the Codec menu |
| — `Audio` | Locks the Audio menu |
| — `ISDN` | Locks the ISDN menu |
| — `System` | Locks the System menu |
| — `Channel assign` | Locks the front-panel MIC, LINE, and MONITOR routing buttons |
| — `Manual dial` | Prevents the user from manually entering a phone number to dial. |
| — `Autodial` | Prevents the user from dialing a preprogrammed number. This is most useful when using a contact closure or logic level to initiate a call through the Panic Dial feature. |
| — `Safe Setups` | See Setups, page 6-6 |

HOT TIP!

Locked menus are still viewable from the menu system, even when SAFE MODE is on. They just can't be changed. This is so a remote user can report the settings of a unit without changing the lock mode.

## A

access code · 6-16

arrow icon · 6-3

autodial · 6-10

## D

Dialog4 · 6-12

## G

G.722 · 6-11

## H

help button · 6-2

Help system · 6-4

## K

keypad · 6-7

## L

line inputs · 6-14

lock · 6-2

Loop Mode · 6-13

## M

menu navigating · 6-3

menu overview · 6-5

menu system · 6-2

menu, detailed description · 6-11

## S

SAFE · 6-2

safe mode · 6-19

Settings · 6-3

setup · 6-6, 6-7, 6-9, 6-10

setup, auto-create · 6-8

Slimline · 6-12

software updates · 6-4

# ISDN

# ISDN BASICS

## Background

Like just about every other aspect of our high-tech world, telephones have been making a transition from analog to digital. Digital telephone transmission services — originally intended just to make basic voice calls more cost-effective — have also made the high-quality transmissions of ZephyrExpress possible.

The first significant users of digital audio techniques were the phone companies: In the mid 60's a digital transmission method called "T-carrier" began to be widely deployed to expand the voice-channel carrying capacity of existing copper wires. Engineers discovered that these wires, originally installed for simple single-channel analog, were capable of much higher bandwidth than the 3.4 kHz required for speech. Indeed, it was determined that two of these pairs could be made to relay 24 voice conversations – if they were digitized and appropriately multiplexed. Thus was born the basic technology used for digital telephony today.

The standards that were developed then continue to define the digital telephone network: an 8 kHz sampling rate (resulting from the desired 4 kHz Nyquist frequency to accommodate a 3.4 kHz audio bandwidth, with guard band) with 8 bits of amplitude resolution (instantaneously companded to provide performance roughly the same as a 13 bit linear system producing 78 dB dynamic range for speech signals). Thus the basic voice channel bit rate was established to be 64kbps.  (8kbyte/sec x 8bits = 64kbps.)

These early applications of digital technology were invented by the telephone industry for its own benefit. The fact that they were digital was neither obvious nor important to customers. However, telephone engineers learned to appreciate digital audio for the same reason we in the pro audio community have: immunity to noise and other quality impairments, ease and flexibility of routing and multiplexing, and lower cost due to compatibility with the electronics and media invented for the rapidly advancing computer industry.

Nearly all long-distance calls are now connected from city-to-city using digital paths on fiber cables and most switching and routing is performed by digital machines.

With the digital nature of the modern telephone network is hidden from subscribers, voice and signaling has been delivered just as they have been since the era of wooden phone sets and mechanical bells. In the age of digital communication, this "last mile" bottleneck had become increasingly frustrating for those who have need to send digital information through Ms. Bell's wires.

With most of the network now digital, it is clearly ironic that we have been using modems to convert a computer's digital information to analog beeps... just to accommodate the mile or two of ancient analog linkage at each end of a thousand-mile connection.

ISDN is the technology which has evolved to eliminate this analog bottleneck yet still utilize existing copper infrastructure.

**ISDN TIP!**

While ISDN uses existing copper wires, its high frequencies mean the signal isn't as robust as conventional analog over those wires. How much cable there is from the telco switch or repeater is critical: in most cases, ISDN won't work if the site is more than three miles from the switch.

Your local telephone company must determine if a particular site qualifies on a case-by-case basis.

**HOT TIP!**

Since the total cable distance is critical, the additional wiring you add between the telco interface (often in a "wire room" in the basement) and ZephyrExpress can disable a marginal circuit... particularly in the case of large office buildings, arenas, and concert halls. As a rule of thumb, a few hundred feet is usually benign.

If in doubt, use Category 5 unshielded twisted-pair wire between the telco interface and the NT1 (or built-in NT1 in ZephyrExpress). While ordinary phone wires may work between an external NT1 and an S interface, 10Base-T Ethernet cables are better at rejecting noise.

As a last resort, move ZephyrExpress closer to the telco interface and pipe the audio instead.

## The Basic Rate Interface (BRI)

On one ISDN BRI circuit, there are three simultaneous channels: Two 64kbps "bearer" channels for the transmission of user information and one 16kbps signaling channel for call set-up and status communication. This is Basic Rate Interface (BRI), 2B+D service. It can be implemented over most of the millions of standard copper two-wire phone circuits already in service.

**CURIOSITY NOTE!**

While the D channel is always present for signaling, it theoretically can also carry users' packet data connections. ISDN lines where the D channel is restricted to signaling only are *sometimes* referred to as 2B+0D rather than 2B+D service. Packet service over the D channel is not widely available.

ISDN BRIs are perfectly matched to Zephyr's transmission capabilities. One channel provides FM quality mono, while the two channels can carry near-CD quality stereo.

From the perspective of telephone network routing, each channel appears to be a separate line with its own number and independent dial-out capabilities. Since each has to be dialed or answered separately, they appear to be "lines" to users also. To reduce confusion (hopefully) for non-technical users, we refer to a B channel as a "line" on the ZephyrExpress menus and LEDs.

## SPIDS

Service Profile Identification numbers (SPIDS) are only required with ZephyrExpress when you are using the National I-1 ISDN or DMS Custom functional ISDN protocols in the USA. This number is given to the user by the phone company and must be entered into ZephyrExpress for the connection to function. SPIDs usually consist of the phone number plus a few prefix or suffix digits.... but they don't have to. There is frequent confusion between telephone numbers and SPIDS, even among Telco personnel. While the SPID frequently includes the corresponding phone number, this is not necessarily the case.

If you are using the National I-1 or the DMS Custom functional protocol, your Telco service representative must give you one or two SPID numbers.

Upon power-up, connection of the ISDN line, reboot, or protocol change, ZephyrExpress and the phone company switch go through an initialization and identification routine. ZephyrExpress sends the SPID(s), and if the switch thinks they're correct communication is possible. Thereafter the SPID is not sent again to the switch.

The two line status indicators on the bottom of the status screen will show you how the initialization process is going. Check the screen after ISDN has been connected for thirty seconds orso:

| | |
|---|---|
| `Inact` `Inact` | means no ISDN connection |
| `Init` `Init` | means the `ISDN: Switch type` doesn't match the phone company's setup or SPIDs haven't been entered.<br><br>`Init Init` can also appear when a European ISDN connection has been idle for a while. This is normal, and Ready will appear when you start using the connection again. |
| `Wait` `Init` | means incorrect SPID1 |
| `Ready` `Init` | means only one working ISDN channel |

`Ready   Wait` means incorrect SPID2

`Ready   Ready` This is what you want to see. ISDN is working.

Realtors remind buyers that the three most important factors of real-estate success are Location, Location, and Location! We want to remind you that the three most important factors to ISDN success are:

## Get the SPIDs,
## Get the SPIDs,
# GET THE SPIDS!

You must have these numbers, and they must be 100% correct, or the system won't work. Don't let the installer depart without giving them to you. You've been warned!

> ISDN  TIP!
>
> In an emergency, to save a remote, you can *sometimes* get a line  working without the proper SPIDs by dialing into it. This fix — if it works — is only temporary and the line will fail to initialize next time the Zephyr is booted.

> ISDN  TIP!
>
> If you have trouble getting to `Ready Ready` and you're sure the SPIDs are correct, try unplugging the ISDN line for a minute or two, and then replugging. This shakes things up at the telco, and might fix the problem.

> ISDN  TIP!
>
> If either of the line status indicators on the bottom of the status screen shows `Wait`, you've entered an incorrect SPID. Check to make sure you've entered exactly what the telco gave you. We've also seen cases where the telco gave customers an incorrect  SPID.
>
> If you're sure you've entered it correctly and it still doesn't work, see the Trouble-shooting section for hints on what to say[1] to the phone company.
>
> There's a list of known working SPIDs by telephone company in the Appendix.

---

[1] What you should say in technical terms. What you tell them in scatological terms is your own decision.

## Directory Numbers (DNs)

Directory Numbers, or DNs for short, are the 7 digit telephone numbers assigned to the ISDN line (or what you'd typically find in a telephone *directory*). You may be assigned one or two, depending upon the line configuration. In the case where you have two active ISDN B channels, you will usually have two DNs (but not always). However, the "physical" channels are independent from the "logical" numbers. A call coming in on the second number will be assigned the first physical B channel, if it is not already occupied. Therefore, there must be some way for the Zephyr to sort out which call goes to which channel/line. The DN is used for this function.
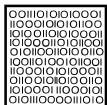
When a call rings in, it contains set-up information which includes the DN that was dialed by the originating caller. The last seven digits are matched with the DNs programmed into the Zephyr and the proper assignment is made. Therefore, problems with Directory Numbers will virtually always result in trouble receiving calls.

## Long-Distance Digital Connectivity

Long-distance connectivity via ISDN is routinely available in most parts of the USA from the "big-three" carriers: AT&T, Sprint, and MCI. Connectivity between any two given points is somewhat variable. The default long-distance carrier may be chosen at the time you order the line, just as with traditional voice lines. Also, just as endless television

commercials remind us for voice calls, you can choose an ISDN long-distance company on a per-call basis by prefixing the number with the "1010-xxx" carrier code.

You must dial the full number, including the 1 or 011 + country code, following the 1010-xxx numbers.

---

ISDN TIP!

If you want to temporarily switch carriers, try prefixing the call with these numbers:

    1010-222 for MCI            1010-288 for AT&T            1010-333 for Sprint

AT&T can be unfriendly to ISDN, and at times will block 1010 access for that service. You may have to contact them before dialing their access code.

Some long-distance carriers — not the big three — can't handle ISDN at all. For the latest information check the section in the Appendix "ISDN, BRI, ZephyrExpress, and You".

---

Some long-distance connections are limited to 56kbps/channel. This limitation is becoming more rare. There is no certain way to know in advance. In addition, some carriers may work at 64Kbps and not 56Kbps. So, you may need to try both or temporarily switch carriers.

---

DEEP TECH NOTE!

This limitation arises from a quirk of the older telephone infrastructure. The channel banks that have been widely employed in the long-distance network have a native 64kbps capability, but "rob" the low order PCM bit on every sixth frame in order to convey supervision information (on-hook/off-hook and dial pulses).

As new telephone plant is built for clear 64kbps transmission and a technology called "Signaling System 7" is deployed to allow the various elements of the phone network to communicate without using the robbed bits, this limitation will disappear. The European network has universal full 64kbps capability.

---

ZephyrExpress provides standard rate adaptation (officially known as 'CCITT V.110') from 56 to 64 kbps and vice versa when required. Bitrate adaptation happens automatically within the system depending upon the rate selected by the calling party. Since ZephyrExpress can't communicate at two rates simultaneously, it ignores rate adaptation information on any second incoming call.

# HOW TO ORDER ISDN

## Dealing with The Phone Company

As is often the case when we broadcasters interface with the phone people, the lines of communication on ISDN can get a little tangled. Face it: we are not the usual customer.

The first order of business is to find someone who knows what ISDN is. While your usual account agent will be the normal entry point, you may be talking to a number of phone people before you find one who understands your needs. Some of the regional

Bell companies offer a single point of contact number for switched digital services. Some Telcos use "Resellers" or "Agents". If so, you should be sure to ask what experience the particular agent has with ordering lines for high fidelity audio codecs. If in doubt, go direct to the Telco. *And always order your line in writing, using the ISDN order forms in the Appendix of this manual.*

This section is intended to be used as a reference. It is probably not necessary to understand everything in order to get an ISDN line for ZephyrExpress.

The easiest way to order the line is to use the Faxable form in the Appendix, *ISDN BRI, ZephyrExpress, & You.* There is also a list of contact telephone numbers for most of the regional telephone companies.

## Details, Details

In order to communicate accurately what it is you need, you might want to learn about the nature of the ISDN service and the vocabulary used to describe it. As with anything, for best results, it helps to know what you're talking about. We already have a good start, but there is more to learn.

### CSD and CSV

Recall that each ISDN BRI has two possible B channels. It is possible to order a line with one or both of the B channels enabled – and each may be enabled for voice and/or data use. Phone terminology for the class of service is CSV for Circuit Switched Voice and CSD for Circuit Switched Data.

CSV is for calls to standard voice phone service and allows ISDN to interwork with analog phone lines and phones. *CSD is required for Zephyr connections.* Even though you may be sending voice, the codec bitstream output looks like computer data to the phone network. Alternate CSD/CSV means both are supported.



DEEP TECH NOTE

Both CSD and CSV are different from PSD — Packet Switched Data (which is irrelevant to ZephyrExpress).

ZephyrExpress allows voice calls on either of the channels as well as the coded hi-fi audio. Thus you can make calls to any normal telephone number on one channel while a program is being transmitted on the other.

Therefore, you may want to order CSV as well as CSD on one or both B channels. To get a line with one B channel to be used with either hi-fi or speech, you would request an ISDN BRI 1B+D line with alternate CSV/CSD capability; for both B channels, you would order an ISDN BRI 2B+D line with alternate CSV/CSD on both channels; if you don't need voice possibility on the channels, you want 2B+D with only CSD enabled.



HOT TIP!

While the D channel is always present for signaling purposes, it can, in theory, be used for packet data connections. ISDN lines where the D channel is used for signaling only are *sometimes* referred to as 2B+0D rather than 2B+D service. D channel packet service is not widely available.

### NT1s

The ISDN standard specifies two reference points, the 'U' and the 'S' interfaces.

The U is the single-pair bare copper from the Telco central office. A device called a 'Network Termination, Type 1' (NT1) converts this to the two-pair S interface.

In Europe and Asia the NT1 is always provided by the phone company, and only the S interface may be on user equipment. ZephyrExpress units shipped outside the USA and Canada have the S interface only. In North America, the NT1 is usually provided by the user: ZephyrExpress units sold within that area include an NT1 built into the terminal adapter.

North American ZephyrExpress terminal adapters provide power for an external NT1, if one is connected. The practice in the rest of the world is for the NT1 to have its own power supply, and ZephyrExpress units sold outside North America don't provide NT1 power.

> **WARNING!**
>
> Using both an external NT1 power supply and the supply in a North American Zephyr-Express at the same time will damage the NT1... and probably the ZephyrExpress as well.
>
> If you have even the slightest belief that this might be possible in your installation, see the detailed warnings in Section 1 (Quick Results) or Section 3 (Hardware).

### Terminals and Terminal Types

Any equipment connected to an ISDN line is a 'terminal' – whether phone, computer, or ZephyrExpress. Point-to-point lines support one terminal, while multipoint lines can have up to eight in some applications.

'Terminal Type' is a parameter sometimes requested by the phone people. The appropriate value for the ZephyrExpress varies depending upon protocol and is given below.

## ZephyrExpress ISDN Compatibility

The ZephyrExpress internal ISDN interface (sometimes called by its generic name 'Terminal Adapter') is used to connect to ISDN telephone lines. Use the `ISDN: Switch type` menu selection to adapt ZephyrExpress to the various types of services offered by telephone companies in most parts of the world. No EPROM changes are required.

## Protocols

In a perfect world, all ISDN terminal equipment would work with all ISDN lines, without regard for such arcana as 5ESS, DMS100, CSV/CSD, SPIDs, etc. Unfortunately, the ISDN "standard" has been in evolution for the past years and has only recently begun to settle down. And, sadly, there will remain different standards for the USA and Europe.

The Telco network and ZephyrExpress communicate via a 'protocol' – the language the user equipment and the telephone network use to converse (on the D channel) for setting up calls and the like. This is where there are differences depending upon the central office switch used on that particular line and the standards which are followed. While each

will work with ZephyrExpress, the differences need to be taken into account when lines are ordered and the equipment set up.

In the USA, telephone companies use either AT&T 5ESS, Northern Telecom DMS100, or Siemens EWSD switches. Each of these can support the National ISDN 1 (NI-1) protocol standard, which has been specified by Bellcore, the technical lab jointly owned by the phone companies. However, both AT&T and Northern Telecom had "custom" versions of ISDN which pre-date the NI-1 standard and some switches have not been upgraded to the new format.

In Europe, the common standard is Euro-ISDN, following the ETS300 documents. It is an (apparently successful!) attempt for all of the European telephone networks to use a single, compatible protocol. The Telco authorities in most countries have adopted it already, with most of the rest planning to do so.

CURIOSITY NOTE!

There are also newer NI-2 and NI-97 standards, but they are designed to be compatible with NI-1 for all of the basic functions.

ZephyrExpress supports all of these with the appropriate selection of the `ISDN: Switch type` menu, as follows:

- `PTP` AT&T Point-to-Point Custom.

- `Natl I-1` NI-1 from all switches; Northern Telecom "Functional" custom.

- `ETS300` "Euro-ISDN", ETS300 pan-European protocol.

In the USA, if you have a choice, the AT&T custom PTP protocol is often preferred because you don't have to trouble with the SPIDs.

# Ordering: Central Office Switches and Protocols[1]

Here are detailed descriptions of what you tell the phone company you want and the corresponding ZephyrExpress settings for each of the protocols.

## National ISDN-1 (USA and Canada)

Available on AT&T 5ESS, Northern Telecom DMS100, and Siemens EWSD switches which have newer generation software.

If the Telco uses IOC Capability Packages, specify Capability Package "S".

If they do not use IOCs, use the information that follows:

**CO VALUES (TO TELL THE PHONE COMPANY)**

Line Type: National ISDN-1

Bearer Service: CSD and/or CSV as desired (see above)

TEI: One dynamic per channel

---

[1] This information is also included in our handy ISDN Order Form, in the appendix.

Terminal Type: A

1010XXX: Yes

*Turn off* features such as: packet mode data, multiline hunt, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared directory numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, call waiting, etc.

Get from Telco: One or two SPID numbers, depending upon number of active B channels; one or two directory numbers.

**ZEPHYREXPRESS SETTINGS (FOR YOU TO ENTER)**

Set `Switch type` to: `Natl I-1`

`SPIDs`: Enter one or two numbers, depending upon number of active B channels

# AT&T Point-to-Point (Custom) (USA, Japan, Israel, some others)

Available on AT&T CO switches version 5E4.2 and above. This is the most basic possible configuration of ISDN, but supports all ZephyrExpress functions. It is the most convenient protocol for ZephyrExpress set-up because no SPIDs are required.

**CO VALUES (TO TELL THE PHONE COMPANY):**

Line Type (DSL class): Point-to-Point (PTP)

B1 Service: On Demand (DMD)

B2 Service: On Demand (DMD)

Maximum B Channels (MaxChan): 1 or 2

CSV Channels: Any

Number of CSV calls: 1

CSD Channels: Any

Number of CSD calls: 1 or 2

Terminal Type: A

Number Display: No

Call Appearance Pref: Idle

1010XXX: Yes

*Turn off* features such as: packet mode data, multiline hunt, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared directory numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, call waiting, etc.

**ZEPHYREXPRESS SETTINGS (FOR YOU TO ENTER):**

Set `Switch type` to: `PTP`

`SPIDs`: Not required

## AT&T Point-to-Multipoint (Custom)

Available on AT&T 5ESS CO switches version 5E6 and above. This is becoming essentially obsolete as NI-1 has the same capabilities.

*It is not supported by ZephyrExpress.*

Order AT&T Point-to-Point (Custom) or National ISDN-1 instead.

## Northern Telecom DMS100 'Functional' (Custom, PVC1) (USA, Canada, some others)

Available on Northern Telecom DMS100 switches BCS 31 and above.

**CO VALUES (TO TELL THE PHONE COMPANY)**

Line Type: Basic Rate, Functional

EKTS: No

Call Appearance Handling: No

Non-Initializing Terminal: No

Circuit Switched Service: Yes

Packet Switched Service: No

TEI: Dynamic

1010XXX: Yes

Bearer Service: CSD and/or CSV as desired (see above)

*Turn off* features such as: packet mode data, multiline hunt, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared directory numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, call waiting, etc.

Get: One or two SPID numbers, depending upon number of active B channels

**ZEPHYR SETTINGS (FOR YOU TO ENTER)**

Set `Switch type` to: `Natl I-1`

`SPIDs`: Enter one or two numbers, depending upon number of active B channels

# Euro-ISDN  (Europe, Hong Kong, some others)

ZephyrExpress works with any ISDN line which conforms to the Euro-ISDN, ETS300 standard. Fortunately, this protocol is standardized and there are no further details to worry about.

Bearer Service: CSD and/or CSV as desired (see above)

**ZEPHYREXPRESS SETTINGS (FOR YOU TO ENTER)**

Set `Switch type` to: `ETS300`

`SPIDs`: Not required.

`MSNs`:  Two Multiple Subscriber Numbers may be provided by the telco, but are not required under normal circumstances.



ISDN  TIP!

Sometimes a European telephone company provides more than two telephone numbers for a single ISDN circuit. MSNs are a way of determining which numbers will be answered.

If you enter MSNs in ZephyrExpress, it will answer only those two numbers. Otherwise, ZephyrExpress will answer calls on any of the circuit's directory numbers.

# SECTION 8

# SYSTEM FUNCTIONS

## Introduction

This section covers system-level functions that have aren't part of normal audio or ISDN operations: updating the software, and using the serial and parallel ports.

One other system function — setting the internal clock — is so simple that it's covered in a few paragraphs as `System` menu: `Set time` in section 6.

Actually, all of the system functions in ZephyrExpress are designed to be transparent to the user. Unless you want to learn all its inner details and capabilities and intend to make your ZephyrExpress sit up and do tricks, you can probably skip this section.

# Software Updates

This manual is being written for the first public software release, and there haven't been any updates yet. The procedure for distributing new software — whether it'll be sent on eproms, floppy disk or distributed over the Internet for download from a computer, or even transmitted directly to ZephyrExpress via an ISDN call — is still undetermined.

HOT TIP!

New software and upgrade procedures will always be announced at our web site, www.zephyr.com.

The `System: Software` menu item includes four choices for using new software once it's been downloaded into the unit:

- `Try new`     Reboots ZephyrExpress and runs the new software for this startup only. The older software remains the default, and will be active the next time you start up. But the new software also stays in memory, and you can `Try new` again.

- `Revert`      Reboots ZephyrExpress with the default software. This is the older version until you specifically replace it by using the `Set default` command. The newer version remains in memory, and you can `Try new` again.

- `Set default`       Reboots the unit using the currently-running software and make it the default. If you are currently running the new software under of a `Try new` command, it will become the default and the older software will be erased. However, if you run `Set default` *immediately* after downloading new software, the newer software isn't running yet and won't be made the default.

- `No change`   Escapes from the menu item, doing nothing.

   TO INSTALL NEW SOFTWARE AS THE DEFAULT:

   1.  Download the software into ZephyrExpress

   2.  Select `System Software: Try new`. This restarts ZephyrExpress with the new software. Try it and make sure you're happy with the newer version.

3. Select `System Software: Set default`. This restarts ZephyrExpress. Since the newer version was previously running (because of step 2), it becomes the default. The older version is erased from system memory.

> ⚠ **WARNING!**
>
> `Try new`, Revert, and `Set default` all reboot the system. This drops any current calls, and causes ZephyrExpress to resynchronize to the ISDN line.

# ZephyrExpress data ports

## The Serial Port

ZephyrExpress' serial port lets you control ZephyrExpress' from a modem or personal computer. Any parameter that can be set using a menu can be queried and set via remote control, and you can remotely instruct ZephyrExpress to place a call on one or both ISDN lines. There is no provision for remotely controlling the front-panel routing switches or volume knobs.

You can connect a palmtop or portable computer to ZephyrExpress to quickly set parameters in the field. You can connect ZephyrExpress to a modem or terminal adapter on a separate telephone line for remote control of broadcasts or emergency studio-transmitter links.

The serial port accepts a male DB-9 and matches the pinouts on most personal computers. The default line protocol is 8 bits, 1 Stop Bit, No Parity, 9600 bps. You can change the speed using `System: Baud Rate` or via a remote command.

> **CURIOSITY NOTE!**
>
> Baud refers to the speed of an analog data stream in transitions per second, so the term doesn't really describe what comes out ZephyrExpress' serial port. Modems for analog telephone lines use phase techniques to pack more than one bit per transition, so — except for very slow ones — they should also be described in bits per second rather than baud. But people generally use the terms interchangeably.

| RS-232 Pinouts | |
| --- | --- |
| *PIN* | *FUNCTION* |
| 2 | **Rx (Computer to ZephyrExpress)** |
| 3 | **Tx (ZephyrExpress to Computer)** |
| 4 | **DTR (ZephyrExpress ready output)** |
| 5 | **Ground** |

| ZephyrExpress | signal | Computer |
|---|---|---|
| pin 5 | ground | pin 5 |
| pin 2 | <<< data | pin 3 |
| pin 3 | data >>> | pin 2 |

## Communicating

ZephyrExpress speaks standard ASCII text at the selected baud rate.

When you first turn on ZephyrExpress' power, it sends the message:

```
Hello world!!
ZephyrExpress V1.2.1/2.36, 16.07.1998 12:39:42
Copyright 1997 TLS Corp. All rights reserved.
Current time is 30.03.1999 15:46:10
```

The `V1.2.1/2.36, 16.07.1998 12:39:42` refers to the software version and when it was compiled, not the current date and time.
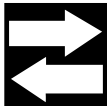
After a few seconds, you'll see the ZephyrExpress system prompt:

```
>>
```

If you do not get any text, the problem is probably:

• The cable.

• The terminal program itself. Does it work in other communications setups?

- The terminal program's baud rate settings. If you have the latter problem, you will probably see some "garbage" text. Verify the setting in the `System: baud rate` item (default is 9600 baud), or try different rates at your computer.

- COM port selection at the computer.

ZephyrExpress echoes any data sent to it, to verify proper operation: If a connection is properly made, anything you type will be returned to you. Try typing something:

>>**Hello,° ZephyrExpress!**

You should see the letters on your computer's screen as you type them.

If you see double letters as you type,

>>**HHeelllllloo,,° ZZeepphhyyrrEExxpprreessss!!**

be sure to turn off *Local Echo* on your computer's terminal program.

ZephyrExpress commands are terminated by a Return character (ASCII CR, Decimal 13, Hex 0D... or just hit the *Enter* or *Return* key on your computer). When you send a Return, the system tries to analyze what you've sent and respond to it:

```
ERROR: Unknown command: hello,
>>
```

If ZephyrExpress doesn't understand what you've sent, it'll let you know. It will also respond `Unknown command` if it doesn't think you don't have a high enough security level to use a specific command.

## Security Levels

ZephyrExpress has two Security Levels, *User* and *Expert*. Each supports one password. You must log in at one of these levels, using its password, before you can remotely control the system. When you're finished with a session you can log out, disabling remote contro until the next login.

DEEP TECH NOTE

We included the log in/out procedure so you can leave ZephyrExpress connected to a modem without fear of a hacker changing its settings.

HOT TIP!

The only difference between levels is that Expert has access to a few additional remote diagnostic tools. You can use the two levels instead to support two different passwords.

- The default password for User-level control is `user`.

- The default password for Expert-level control is `expert`.

You can change the password to something less obvious, using the command detailed later in this section. Passwords are case-sensitive: `user` and `expert` are *not* the same as `User` or `ExPeRt`.

To log in, wait for the prompt. Then type `login` followed by the password for the desired level and press Enter:

```
>>login  user  ¶
```

If it recognizes the password, ZephyrExpress will respond with:

```
Welcome to ZephyrExpress control system user level.
Press ? for help.
>>
```

> ⚠ **WARNING!**
>
> If ZephyrExpress does not respond to **user** or **expert** when you try to log in, somebody changed your password. You can't change it back without knowing the new one. Even a cold boot — which restores all other factory defaults — won't restore the default password[1].

When you're finished with a session at either security level, type **logout**¶ or **bye**¶. This resets ZephyrExpress' communication, and no commands will be accepted until the next login.

## Command Help

As implied by ZephyrExpress' response above, `?` is a specific command. When you send a question mark followed by Return, the system responds:

```
Available commands:
? baud bye conn country disc fan help lcdcont (... and so on)
Type '/' or '-' to repeat, '.' to edit last command
Type '? <command>' to get specific help
Type '? *' to get detailed list of commands
>>
```

The first line is a list of every command available at the current security level.

To get information on a specific command, type a question mark followed by the command name and then Return. For example, if you send:

```
>>?  baud¶
```

ZephyrExpress will reply:

```
baud
[<#300|#600|#1200|#2400|#4800|#9600|#19200|#38400|#57600>]
---
Change the serial port speed immediately or display current
speed.
>>
```

---

[1] That's why they call it "security". There is a difficult and time-consuming procedure that we can sometimes use to extract a lost password, but we're not promising anything.

The reply will always take the same format:

- The first line is the name of the command ZephyrExpress is describing.

- The second line is the *argument* — a word or number that modifies the command.

  Anything within brackets `[` `]` is optional. Some commands have the entire argument within brackets; if you send this command without an argument, it will report the present value. So if you just sent **baud** and Return, ZephyrExpress would send back the current communication rate.

  Anything between `<` `>` is a single argument. A bar | means *or*, as in "choose this argument or the other". So you could send **baud #4800**, *or* **baud #9600**, *or* **baud #19200** and Return to set ZephyrExpress to one of those rates.

- The text after the `---` separator describes how the command works.

- If an argument allows a hyphen, you can use it keep the existing value for that parameter.

- For most commands, you can enter a hyphen (the line should consist of just *–f*) to repeat the last command with all the same arguments.

If you send `?` `*¶`, ZephyrExpress will describe all the commands available at the current security level. Since this is a lot of text sent in a continuous burst, be sure to have your terminal program's *logging* or *capture to file* function turned on.


## Command Language Details

This chart shows the commands available at the User security level.

| Command | Argument | Notes |
|---|---|---|
| **?** | **none** | Display general help instructions |
| | **[topic \|** | Get help about topic. |
| | **\*]** | Display all topics with their help. |
| | *Example* **?** **baud** ¶ | Returns instructions on using the Baud command |
| **baud** | **none** | Display current port speed. |
| | **[<#300\|#600\|#1200\|#2400\| #4800\|#9600\|#19200\| #38400\|#57600>]** | Immediately change to the serial port speed indicated. |
| | *Example* **baud** **#19200** ¶ | Sets serial port to 19.2kbps |

> **⚠ WARNING!**
>
> **baud** changes the port speed as soon as you send Return. You must immediately change your terminal program's speed to match, or communication can't continue.

| | | |
|---|---|---|
| **bye** | **none** | Log out from ZephyrExpress control system. |
| | *Example* **bye** ¶ | No commands will be accepted until the next logon. |
| **conn** | **<1\|2>  <number>** | Using line 1 or 2, place a call to number. |
| | *Example* **conn 1 2417225** ¶ | Place an ISDN call to 241-7225 on line 1. Hypens and parentheses in the phone number are optional. |
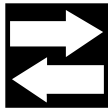
> **⇄ COMPATIBILITY NOTE**
>
> You can command ZephyrExpress to place a high-quality ISDN audio call on either line, but you can't use the remote control to place an analog POTS call.

| | | |
|---|---|---|
| **country** | **none** | Display whether front-panel time and date readout is in USA or German format. |
| | **[<usa\|germany>]** | Set front-panel time/date readout to the specified format. |
| | *Example* **country usa** ¶ | Display will read Month / Day / Year |
| **disc** | **<1\|2>** | Disconnect specified ISDN line. |
| | *Example* **disc 1** ¶ | Hang up call on line 1 |
| **egg** | *heh heh...* | Easter egg. We won't tell you the argument. |
| **help** | **none** | Display general help instructions |
| | **[topic \|** | Get help about topic. |
| |  **\*]** | Display all topics with their help. |
| | *Example* **help lcdback** ¶ | ZephyrExpress will return instructions about the lcdback command. |
| **lcdback** | **none** | Display current LCD backlighting level. |
| | **[<20...79>]** | Set backlighting to value indicated. |
| | *Example* **lcdback 79** ¶ | Backlight will be at its brightest. |
| **lcdcont** | **none** | Display current LCD contrast level. |
| | **[<0...79>]** | Set contrast to value indicated. |

DEEP TECH NOTE!

Remote control values for LCD contrast and backlighting are only vaguely related to the 0-100% settings in the System menu.

| Command | Arguments | Description |
|---|---|---|
| `level` | `none` | Display current encoder input & decoder output level, for testing the meter calibration. |
| `log` | `[<GEN> <SYS> <ISDN> <IMON> <DEB> <off>]` | Used for remote debugging. During a support call, we may ask you to send this command and tell us what the result is. |
| `login` | `[<?> \|` | Display current security level. |
| | `<password>]` | Unlock at the security level associated with this password. |
| | *Example* `login user ¶` | Opens communication at User level. |
| `logout` | `none` | Log out from ZephyrExpress control system. |
| `metertest` | `none` | Resets front panel level meters to normal operation after a test. |
| | `[<0...9 0...9 0...9 0...9>]` | Turns on the four level meter LEDs up to the number indicated, for lamp testing. |
| | | If you're going to specify an argument, you must supply numbers for all four meters: `metertest 9 9 ¶` would be rejected. |
| | *Example* `metertest 9 9 9 9 ¶` | Lights up all four metersas if they were indicating a signal at -10dB. |
| `mic` | `<1\|2\|both> [<phant\|highf> <on\|off>] \| [<hisens\|medsens\|losens\| linesense>]` | Switches microphone phantom power or highpass filter on or off, or sets mic sensitivity. |
| | *Example* `mic 2 highf on ¶` | Turns on the highpass filter for mic 2. |
| `mop` | `none` | Displays the current codec mode (using the codes below). |
| | `[<l3du\|l3st\|l3jo\|g722\| l2mo64\|l2mo128\|l2du\| l2jo\|->]` | Sets the transmit mode to L3-dual *or* L3-stereo *or* L3-joint *or* G.722 *or* L2-mono64 *or* L2-mono128 *or* L2 dual *or* L2-joint. If the parameter is sent as "–", the current transmit mode is kept. |
| | `[<l3st\|l3mo\|g722\|l2\|->]` | Sets the receive mode to L3-stereo *or* L3-mono *or* G.722 *or* L2. If the parameter is sent as "–", the current receive mode is |

|  |  | kept. |
|---|---|---|
|  | **[<br56\|br64\|->]** | Sets the ISDN bitrate to 56kbps *or* 64kbps, *or* keeps the current value. |
|  | **[<sr32\|sr48\|->]** | Sets the audio sample rate to 32kHz *or* 48kHz, *or* keeps the current value. |
|  |  | You must specify all four arguments. |
|  | *Example* **mop - l3mo br64 sr32 ¶** | Set the receive mode to L3-mono, the bitrate to 64kbps, and the sample rate to 32kHz. Leave the transmit mode alone. |
| **passwo** | **<current-pw>    [<new-pw>]** | Change password for current security level. If you don't specify a new password, this level will accept all logins without a password. |
|  | *Example* **passwo user steve ¶** | If the current password is "user", this will change it to "steve". |
|  | *Response* New password for user level:steve<br>>> | ZephyrExpress confirms the new password. |

> ⚠ **WARNING!**
>
> Be sure you can remember the new password. Once you change it, there is no way to restore the original[1]. Check the response to make sure you typed the new password correctly.
>
> Passwords are case-sensitive.

| **rtc** |  | Read current time from ZephyrExpress' real-time clock. Time will be in the format:<br>`30.03.1999 19:02:12, 15331 msec, 15.330856 sec,`<br>where the first string is the date, the second is the time of day, and the third and fourth indicate the total time since last startup. |
|---|---|---|

| **telco** | **none** | Display ISDN switch type. |
|---|---|---|
|  | **[<ETS300\|PTP\|NI1>]** | Select switch type. This will restart the ISDN engine. |
|  | *Example* **telco ptp ¶** | Sets the switch type to Point-to-Point. |
| **version** | **none** | Displays the current software versions. |
|  | `System software : 1.2.1`<br>`Loadset software: 2.36` |  |

---

[1] Well, there possibly *is*... but it's messy and will take up a lot of your time and ours.

In addition, Expert level includes these commands:

| | | |
|---|---|---|
| **numbase** | **<dec\|hex\|bin\|none>** | Set default number base to either decimal, hexadecimal, binary or none. If none is selected all numbers must be preceded by explicit specifiers. |
| | | The default base can always be overridden by explicit specifiers: |

> **%** for binary
> **$** for hex
> **#** for decimal numbers.

---

**DEEP TECH NOTE!**

Numbase is provided for users who want to write their own control software. It affects only the remote control function: the displays remain in decimal at all times.

---

| | | |
|---|---|---|
| **oscill** | **<on\|off>** | Switch on/off test tone oscillator on both mic inputs. |
| **parport** | **none** | Display parallel contact closure status. |
| | **[<val0..valF>   \|** | Sets the four parallel contacts to match the bit pattern and ignore incoming ancillary data. |
| | **<norm>]** | Restores normal operation. |
| | *example* **parport  val3  ¶** | Sets contacts to off-on-on-on. |
| **reset** | **[cold\|res]** | Reset the system by restarting the code without physical reset. **cold** will wipe out all user-defined system parameters and setups. **res** will maintain user settings. |
| | | We may ask you to enter some other arguments during a support call. |
| **rtcset** | **<date  time>** | Set date and/or time of RTC (real time clock). |
| **switch** | | Return information about internal and backpanel switches. |

# The Parallel Port

ZephyrExpress' parallel port has five contact-closure outputs and four logic or contact-closure inputs. Voltages and connection details are in Section 3, Hardware.

| Parallel Port | | | |
|---|---|---|---|
| *PIN* | *FUNCTION* | *PIN* | *FUNCTION* |
| 1 | Ground | 9 | N/C |
| 2 | Output 2 | 10 | Output 0 |
| 3 | Status Out | 11 | Output 1 |
| 4 | Output 3 | 12 | N/C |
| 5 | N/C | 13 | Input 2 |
| 6 | Input 3 | 14 | Input 1 |
| 7 | Input 0/Panic Dial | 15 | N/C |
| 8 | +5 volts 400 ma max. | | |

One of the five outputs, pin 3, is used to report successful connections. Use the `Codec: Status Out` menu item to set this pin to go active low when the receive decoder is locked, or when either or both lines are connected.

One of the four inputs, pin 7, can be used to initiate "Panic Dial" and call a particular number when it's pulled low. Use the `ISDN: Panic dial` menu item to specify an autodial setup, which can also include a master setup that configures the codec or other menus. If no Panic Dial setup is specified, the input can be used for ancillary data.

> **ISDN TIP!**
>
> Panic Dial calls stay connected only while pin 7 is low. When pin 7 is lifted from ground, the call is dropped.

## C

codec mode · 8-9, 8-9

## D

DB-15 · 8-12
DB-9 · 8-3
diagnostic · 8-5

## G

G.722 · 8-9, 8-9

## L

locked · 8-12

## M

meters · 8-9

## O

oscillator · 8-11

## P

parallel port · 8-12
password · 8-6, 8-9
phantom power · 8-9
POTS · 8-8

## R

remote control · 8-3, 8-8
remote control language · 8-4
RS-232 · 8-3

## S

security levels · 8-5
serial port · 8-3
setup · 8-12
software updates · 8-2
syntax · 8-4

## SECTION 9

# TROUBLESHOOTING

# First Steps

You don't want to be reading this.

Chances are, you reached for these pages because something isn't working the way you expect it to.

Cheer up. Most ZephyrExpress and ISDN problems are easily resolved, either by yourself or with the help of the telephone company. Unless you see blue smoke pouring out of the unit, these pages are probably all you'll need to get running again. And if they aren't — or if you *do* see blue smoke — you can always contact Telos Customer Service at

- +1 216.241.7225 in the US (fax +1 261.241.4103)

- +49 81 61 42 467 in Germany (fax +49 81 61 42 402)

- support@zephyr.com

- support@zephyr.de

- http://www.zephyr.com

But try these pages first. It's faster.

> HOT TIP!
>
> It may be that something isn't working the way you expect it to, because it's not supposed to work that way. ISDN and audio coding is not as intuitive as traditional analog audio. If you're not familiar with both of these new technologies, read the first section of this manual before assuming something's wrong.

> ISDN TIP!
>
> If ISDN won't initialize — if you don't see the word Ready twice on the screen — and you're absolutely sure the line is good and ZephyrExpress is programmed properly, try these tricks. One of them will usually work.
>
> - Unplug the telephone line from the back of ZephyrExpress, wait a minute or two, and plug it in again.
>
> - Change the ISDN: Switch type menu setting to the wrong kind of switch and press EDIT to confirm. Wait a few seconds, change it back to the right kind, and press EDIT again.
>
> - Turn ZephyrExpress off, wait a few seconds, and turn it back on again. **DO NOT DO THIS BY DISCONNECTING THE XLR POWER CONNECTOR ON THE BACK PANEL!** This can damage the unit, or erase your settings. Use the power supply's switch, or disconnect it from the power line.

> **ISDN TIP!**
>
> You can check ISDN continuity with an ordinary telephone or pair of 600 ohm headphones. Disconnect ZephyrExpress, and briefly place the phone (or phones) across the line. You should hear a rhythmic clicking — about once per second — or a loud, continuous white noise. If you hear a dial tone, it's not an ISDN line. If you don't hear anything, the line is dead.

> **ISDN TIP!**
>
> Are you sure you're plugged into the right port?
>
> For most US installations, the line should be plugged into the *lower*, 4-conductor jack on the back of ZephyrExpress — even if the phone company supplied an 8-conductor RJ-45 jack with the line. Use a standard telephone line cord: when you plug it into the RJ-45, it'll automatically center itself and locate to the right conductors.

# GENERAL

## Thinking About Problem Solving

Despite best intentions, something could always go wrong. Sometimes troubleshootinga balky set-up can make even the toughest engineer a Maalox and Rogaine addict.

You can't fix any system without the right world view; a zeitgeist of suspicion tempered by trust in the laws of physics, curiosity dulled only by the determination to stay focused on a single problem, and a zealot's regard for the scientific method. Perhaps these are successful characteristics of all who pursue the truth.

In a world where we are surrounded by complexity, where we deal daily with equipment and systems only half-understood, it seems wise to follow understanding by an iterative loop of focus, hypothesis, and experiment.

The notions here apply whether you are solving problems at the system level or at the component level. At the system level, the actions you might take would be very different – checking cables, trying different menu settings – but the thinking is the same.

Too many times, we fall in love with our suppositions. We are quick to overtly or subconsciously assume the problem being chased is due to lousy design, the stupid phone company, or the manager's latest memo.

Armed with a healthy skeptical attitude, the basic philosophy of troubleshooting any system is to follow these steps:

1. Observe the behavior to find the apparent problem;

2. Observe collateral behavior to gain as much information as possible about the problem;

3. Round up the usual suspects;

4. Generate a hypothesis;

5. Generate an experiment to test the hypothesis;

6. Fix the problem;

Then, repeat, if necessary, to attack additional problems.

Let's cover each step of the troubleshooting sequence in detail.

- **Step 1.** Observe the behavior to find the apparent bug. In other words, determine the bug's symptoms. Remember always that many problems are subtle and exhibit themselves via a confusing set of symptoms. Remember, depending on the nature of the problem, the system you are troubleshooting may include your data circuits, yours or the calling party's long distance carrier, and the data circuit and equipment at the other end.

> It's never just "The ISDN isn't working"...
> but *how* isn't it working?

- **Step 2.** Observe collateral behavior to gain as much information as possible about the problem. Does the LCD's problem correlate to an LED flashing? Try to avoid studying a problem in isolation, but at the same time be wary of trying to fix too many at the same time. No one is smart enough to deal with multiple problems all at once – unless they are all manifestations of something more fundamental. Always note any messages ZephyrExpress may present when the problem occurs. Enlist the help of the users to keep track of these messages and give you this information.

- **Step 3.** Round up the usual suspects. At the system level, always suspect the menu set-up, the cables, the phone company line setup, the punch-blocks, etc. At the component level, lots of computer problems stem from the same few sources. Never, never, never forget to check Vcc.

- **Step 4.** Generate a hypothesis. Before changing things, formulate a hypothesis about the cause of the problem. You probably don't have the information to do this without gathering more data.

  Sometimes you'll have no clue what the problem might be. Sometimes, when the pangs of desperation set in, it's worthwhile to try anything practically at random. You might find a bad plug, an unconnected line, or something unexpected. Look around, but be always on the prowl for a working hypothesis.

- **Step 5.** Generate an experiment to test the hypothesis. Plug this ZephyrExpress into a known good line; call a known good Zephyr or ZephyrExpress at the other end; if long-distance doesn't work, try a local call.

  NOTE: You should plan your tests to eliminate 50% of the possible problems in one test, if possible. Just keep careful track so you know what you have eliminated.

- **Step 6.** Fix the problem.

### Other Ideas

Constantly apply sanity checks. Almost thirty years ago, the Firesign Theater put out an album called "Everything You Know is Wrong". Use that as your guiding philosophy in troubleshooting a Zephyr set-up. For example, just because you checked the Telco line last night and it was fine doesn't mean it's OK now.

At 3:00 AM when the problems seem intractable and you're ready to give up engineering as a career, remember that the system has worked and will work again. Relax: you're smarter than the ISDN is, and you'll figure it out.

## Diagnostic Aids

Fortunately, ZephyrExpress has some features designed to aid your troubleshooting effort.

## Look into the Lights

You can learn a lot about an error condition by observing ZephyrExpress' panel lights:

**Front-panel LEDs**

All of ZephyrExpress' LEDs should light for a few seconds on power-up or reboot. The MIC1 and MIC2 routing switches' A and B LEDs will glow red; the other routing LEDs glow green, the meters are fully on, and the LOCK and SAFE LEDs glow red. It's a veritable Christmas Tree. If you don't see these lights, chances are very good you've got a power supply problem.

The LOCK LEDs light when the decoder is receiving valid data. If they're off, chances are the Receive mode doesn't match the distant unit's send mode. If it's flashing, both units are probably in Layer 3 but in different channel or stereo modes. In either case, the cause might also be bad data due to equipment or ISDN problems.

The SAFE LED should either be steadily on or off, depending on how the SAFE MODE switch on the rear panel is set. If SAFE is blinking, it either means:

*   Someone left ZephyrExpress in one of its loopback modes when they last turned the system off. Use the `Codec` menu to turn `Loopback: Off`.

*   You have a European ISDN card (no NT1), and the ISDN circuit is disconnected or dead.

*   There is a hardware problem preventing ZephyrExpress' codec from synchronizing with the telco's clock.

Normal operation is impossible until SAFE stops blinking.

**LCD**

The LCD display should show the word `Initializing...` for a few seconds, and then the Status Screen should appear. The upper right corner of the Status Screen should say `Loading...` for a few seconds, and then display the date and time[1]. If things don't happen

---

[1] Or at least *some* date and time. We don't consider a misset clock to be a major problem.

this way, you've probably got a software problem... but there's a possibility something has gone wrong with the hardware.

If the LEDs light up but you can't see any text on the LCD display, the most probable cause is that someone left the backlighting or contrast at an inappropriate setting.

**DROP** Press the DROP button and hold that button in while turning the EDIT knob clockwise to reset the backlighting. It may take a few turns.

**DIAL** Press the DIAL button and hold that button in while turning the EDIT knob either direction to change the LCD Contrast.

### NT1 status LED

North American ZephyrExpress units, with built-in NT1s, have a green LED on the back of the unit next to the LINE DIRECT jack. It should be steadily lit.

Rapid flashing means there's no ISDN continuity. Check the line. Be sure it's plugged into the lower ZephyrExpress jack.

Slow flashing is normal when the connection is being established. If the LED continues to flash slowly, check the SPIDs and switch type. And don't automatically assume the numbers that the telco installer gave you are correct.

## LOOPBACK modes

You can check ZephyrExpress' audio circuits by looping the send bus directly into the receive bus, check its coding operation by looping the transmit encoder directly into its receive encoder, or check the entire unit by dialing into another Zephyr or ZephyrExpress that's set to loop an incoming signal directly back.

### Far Loopback for local audio testing

Set Codec: Loop mode to Far to connect ZephyrExpress' send mixer directly to the receive bus. The loopbackcircuit looks like the drawing on the next page.
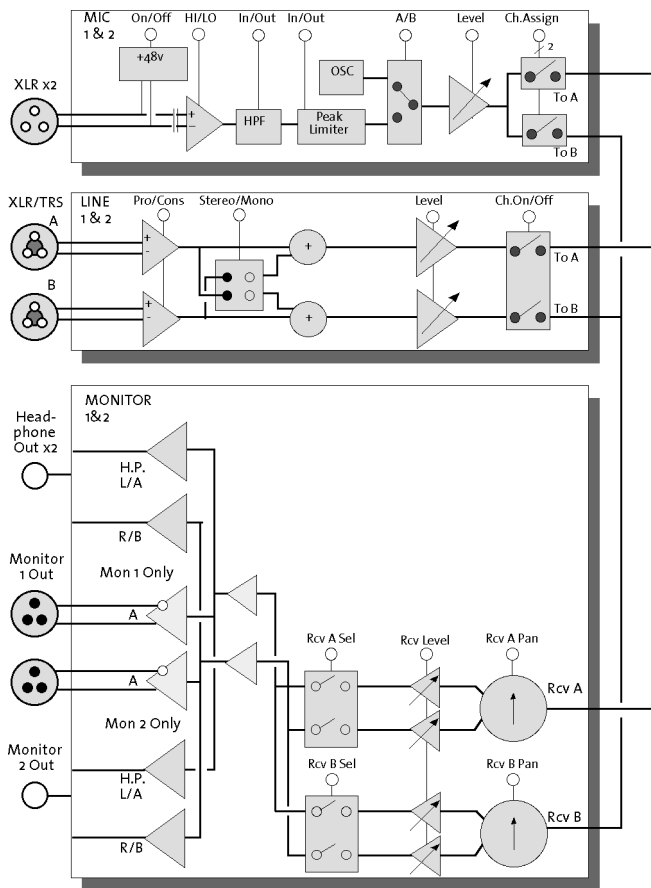
Using Far Loopback and the block diagram, you should be able to trace an audio signal through ZephyrExpress.

When using Far Loopback for local audio testing, the ISDN line doesn't need to be active and the transmit and receive coding methods don't have to match.

⚠ IMPORTANT!

ZephyrExpress' volume meters are driven by the audio codec. For this reason, they're disconnected during Far Loopback mode.

*Block diagram of the audio circuitry during Far Loopback*

In Far Loopback, you should be able to:

1. Route the mic to one or both send circuits;

2. Shout into the mic, and see the A or B routing LED turn red to indicate overload[1];

3. Turn off the Monitor 2 SEND switch, set its RECEIVE switch to AB, raise the level, and hear the microphone in a headphone plugged into the front panel.

If these tests fail, and you're sure the Audio menu settings are correct, the problem is either with the mic and headset and their wiring or with ZephyrExpress' internal audio circuits. You may be able to broadcast using the alternate mic input or the line input, until you can get it fixed. You may be able to receive signals using the direct RECEIVE output on the rear panel.

> ISDN TIP!
>
> If you're in Far Loopback and an ISDN line is connected and properly configured, ZephyrExpress will answer any incoming calls and send the data directly back to the caller. This is handy for testing the data path and the distant studio's codec, as explained on page 9-9.

---

[1] If it doesn't turn red, raise the Sensitivity in that mic's submenu.

### Near Loopback for codec testing

When the `Codec: Loopback` menu item is set to `Near`, incoming audio gets coded in the usual manner, sent into the ISDN interface but then looped directly back to the decoder. This checks over 90% of ZephyrExpress' functionality with no ISDN line present.
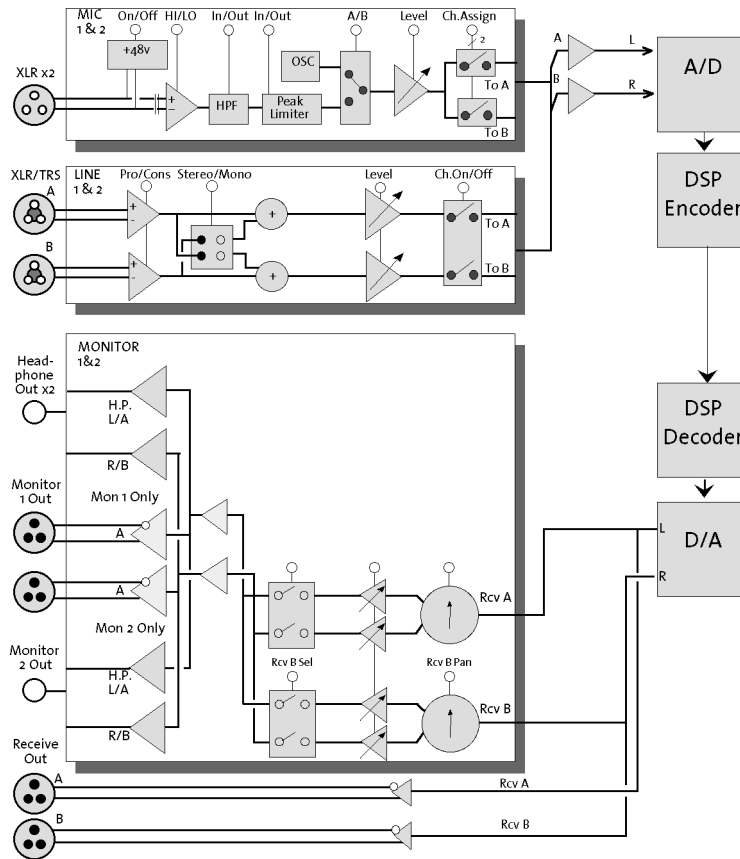


**IMPORTANT NOTE!**

If you are using the European ISDN TA and aren't connected to an active ISDN line, you will hear dropouts in the audio signal during Near Loopback. This is not a problem with the equipment.
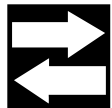
**COMPATIBILITY NOTE!**

Near Loopback requires both the transmit and receive coding methods to be compatible.

**CURIOSITY NOTE!**

Near Loopback is great for trying ZephyrExpress' audio coding algorithms without an ISDN line, and getting used to what the different settings sound like. It's also great for impressing people with the quality that ISDN can yield over a good system.

When Near Loopback is engaged and the coding methods match, the LOCK LEDs should light. Audio can be routed to the send bus, and monitored through the RECEIVE routing switches. Normal coding delays will be present.

In Near Loopback, you should be able to:

1. Route the mic to one or both send circuits;

2. Shout into the mic, and see the A or B routing LED turn red to indicate overload;

3. Raise the mic's level and see it on the send meters;

4. See the LOCK LEDs lit, and the microphone's signal slightly delayed on the receive meters;

5. Turn off the Monitor 2 SEND switch, set its RECEIVE switch to AB, raise the level, and hear the microphone delayed in a headphone plugged into the front panel.

If these tests fail but the Far Loopback tests pass, and you're sure the `Codec` menu settings are correct, the problem is with ZephyrExpress' encoder or decoder circuit. Try testing with a different set of coding methods.

**Far Loopback from the distant studio**

If the Status Screen is showing `Ready Ready`, and there's an audio problem, but Zephyr-Express passes the local Near and Far Loopback tests described above, set the `Codec: Loop mode` item to `Off`. Then have the distant studio set *their* Loopback mode to Far. Set your ZephyrExpress so that both the transmit and receive methods match, and call the distant studio. Their Far Loopback should route your signal directly back to you.

You should hear yourself, after a coding delay. If you do, you've verified that ZephyrExpress and the line are working correctly: the problem is with the distant studio's codec. If you don't, try changing the `Codec: bitrate`.

> **NOTE!**
>
> Don't forget to set `Codec: Loop mode` back to `Off` before attempting normal operation.

**Loopback Summary**

These two loopbackmodes are powerful troubleshooting tools:

• NEAR loopback can eliminate substantial portions of the ZephyrExpress circuitry by testing to see if the coder and decoder are working. Be sure to test in the relevant transmit and receive modes for your situation.

• Assuming only one problem, FAR loopback can effectively determine whether it's due to the communications channel(s) or one of the codecs.

   • While the problem is occurring, first put your ZephyrExpress into Far Loopback while the distant studio stays in normal mode. Then restore your ZephyrExpress to normal and have them use their far loopback mode.

- If either unit can lock to itself, consistently, while looping through the other unit's Far Loopback, then the ISDN connection is good.

- If neither unit can lock to itself then the ISDN connection is failing somewhere. This can be because some part of the connection doesn't support the bitrate you've chosen. It can also happen in the unlikely event that both codecs are broken.

Using two Far Loopback tests, to check units at both ends, will often tell you exactly where a problem lies. For example: If they can't hear you during normal operation, but you can hear yourself when they're in Far Loopback, you know both your transmit coder and receive decoder are working and the line is good. Have them test their codec with your ZephyrExpress in Far Loopback. If they can't hear themselves either, you know it's their receive decoder. If they *can* hear themselves, the problem is an incompatibility between codecs: try a different mode.

## Some Error Conditions

### No functionality; the box is totally or partly dead.

#### *Does ZephyrExpress have power?*

Does the outlet have power? Is the power module's fuse blown? The green LED on the power module should be lit. If it is, then the LCD display on ZephyrExpress should be lit[1]. You can test the power supply for the voltages as listed in section 3, Hardware. Since switching supplies must be tested under load, you'll have to dismantle the its XLR connector, and leave the insert plugged into ZephyrExpress, while checking.

> ⚠ IMPORTANT NOTE!
>
> Do not connect or disconnect the 4-pin XLR power connector on the rear of ZephyrExpress while the power is on. If you're probing voltages with the connector disassembled, make sure the XLR insert stays in place the whole time that power is applied.
>
> Disconnecting the power supply while it's powered can damage ZephyrExpress or erase its memory.

#### *Does ZephyrExpress have basic functionality?*

Upon initial power-up, the LCD display should show the word `Initializing...` for a few seconds, and then the Status Screen should appear. The upper right corner of the Status Screen should say `Loading...` for a few seconds, and then display the date and time.

---

[1] If the LCD doesn't light, check that the backlight and contrast are correctly set. See page 9-6.

During the initializing period, a self-test is performed. If there is a problem, this is reported on the LCD and progress is halted. The message should be noted for discussion with Telos customer support.

If the unit does not reach the INITIALIZING stage, suspect a problem with the system processor, system clocks, or EPROMS

## Condition: ISDN Connecting OK, But No Audio

### What is the state of the LOCK LED?

Check the LOCK LED. On means you're connected and receiving valid data. If it's not lit up, try cycling through receive modes to see if you can "find" the mode at the other end. If it's flashing, you're both in Layer 3 but the stereo mode doesn't match.

### What happens when you call yourself?

Try calling yourself from one ISDN line to the other. Use a 56 kbps rate first, as this is the most universal. Set Codec: Transmit to L3-Dual and Codec: Receive to L3-Mono. Audio should work in both directions. This confirms that ZephyrExpress is OK, and that the problem lies somewhere downstream: either the distant codec is having trouble, or the telco's switching equipment or long distance carrier is having problems.



ISDN TIP!

Try both directions. Sometimes a connection won't carry data when site A calls site B, but works perfectly when site B calls site A.

### What happens if you try a different long distance carrier for the call? If the call doesn't go through at 64kbps, try 56kbps... or vice versa. Try placing a local call as a long distance call.

Since most of the network uses separate transmit and receive paths, it is entirely possible to have a path in one direction but not the other. This is particularly true with long-distance calls, where one end of the connection might be using a different carrier... or not be authorized for long distance at all! The above steps will force you through the network via a different path.

### What happens when you try a NEAR loopback?

If successful, this eliminates over 90% of the ZephyrExpress circuitry as the culprit. Remaining suspects would be primarily the ISDN card. Remember, transmit and receive modes must be compatible.

### Are the coding modes set properly?

The transmit and receive modes and the bitrate must be set to match the unit at the other end.

## Condition: No Audio In Both Directions

*Does ZephyrExpress' loopback isolate the problem?*

See above for tips on using the loopback modes.

*Is the ZephyrExpress mode set properly?*

Your `Codec: transmit` mode must match their receive mode. Their transmit mode must match your `Codec: receive` mode.

Sometimes cycling between modes can clear a decode problem.

*Is audio properly connected and configured?*

Do the loopback tests described earlier. Make sure the `audio` menu settings are correct. Try different microphones and cables.

> HOT TIP!
>
> Use ZephyrExpress' POTS facility to place an analog call over the ISDN line to a standard phone at the distant studio, so you can compare notes on the codecs' settings.

## Condition: Audio In One Direction Only

You've already checked that the codec modes at both ends are set properly, right?

This problem can be either analog or digital in nature. Unlike analog telephone lines, the two directions of a digital line are mostly independent. *It is possible to have a telephone channel failure in one direction only.* Try these steps to force the network to use a different path.

1. What happens if you try a different long distance carrier for the call?

2. Try the call at 56kbps vs 64kbps (or vice versa)?

3. Try reversing the direction of the call?

4. Try placing a local call as a long distance call?

*Are audio signals and connections OK?*

Try changing the routing switch settings, using a different ZephyrExpress input or output, using the other channel, or placing an L3-dual call on the other line.

## Condition: Audio Distorted

*Are the* `Audio: Microphone Sensitivity` *settings correct? Is the microphone wired properly for a balanced input?*

If the `MIC1` or `MIC2` routing LEDs light up red, the sensitivity is too high and you're sure to get distortion.

Some mics — and many direct boxes and press feeds — will choke if phantom power is turned on when they don't expect it.

### Send gain set OK?

With `Audio: Limiter` turned off, check the SEND meters for proper indication. The top two LEDs shouldn't be lighting at all.

### Do the sample rates match?

If the rates don't match — if one codec is set to 48kHz and the other to 32kHz — the signals will be transposed about half an octave and drop out every few seconds. This may be desirable if you're trying for a Mickey Mouse or Darth Vader voice, but is not considered a normal operating condition.

## Condition: Major Echo Action

If you can't receive distant audio, but get a delayed version of your own transmitted signal, it's probable that either your ZephyrExpress or the distant unit is in Loopback mode. Use the `Codec: Loopback` menu item to turn it off, and make sure they perform an equivalent step on their unit. It's also possible that a telco or bridge service left some of their gear in a loopback mode; try the steps for Audio One Direction, above.

If you can receive distant audio but also get a delayed version of your own transmitted signal, they've probably formed a feedback path through their console. Tell them to remove their codec's output from the console bus that's feeding their codec's input.

## Error Messages

The ZephyrExpress software has a system for self-monitoring. When a major problem is detected, a message appears on the LCD screen. The message gives general information about the problem in a text phrase, and specific information in an error number Please note the exact wording of the message and its number, and contact Telos customer support.

# ISDN Problems

## Condition: Can't Get ISDN `Ready  Ready` Indication

### Interpreting the Status indicators

The ISDN status boxes on the bottom of the Status Screen can tell you a lot about how the connection is working:

| `Inact` | `Inact` | means a bad ISDN connection or no ISDN service at all. This can also happen because you've used the wrong jack on the wall or on ZephyrExpress. |

| `Wait` | `Init` | means incorrect SPID1 |

| `Init` | `Init` | means the `telco type` doesn't match the phone company's |

setup or SPIDs haven't been entered.

`Init Init` can also appear when a European ISDN connection has been idle for a while. This is normal, and `Ready` will appear when you start using the connection again.

| `Ready    Init` | means only one working ISDN channel |

| `Ready    Wait` | means incorrect SPID2 |

| `Ready    Ready` | This is what you want to see. ISDN is working. |

ISDN TIP!

If you see `Ready Ready` when the switch type is set to PTP, and you cannot dial, it's likely that your line is really National ISDN-1[1].

**What kind of ISDN do you have, and is the Zephyr set to match it?**

The telephone company should have given you the ISDN protocol when the line was ordered. If you find yourself in the field without this information, it is usually possible to piece together what you need.

- If you are given SPIDS, you probably have National ISDN-1. One SPID means one channel was activated, rather than the usual two. You won't be able to make a stereo connection.

- No SPIDs and only one phone number generally means AT&T PTP. However:

- Sometimes the telco will give you a SPID even when you have PTP. If so, disregard it.

SPIDS beginning with 01 mean you are on the AT&T 5ESS switch. A SPID of 01YYYZZZZ0 generally means that you have PTP. If you have 2 SPIDS with this format suspect that you may have Custom Point-to-Multipoint (PMP) which is not supported by the Zephyr (but see below). 2 SPIDS which look like 01YYYZZZZ000 means you probably have Natl ISDN 1. National ISDN SPIDS can take many other forms as well.

The above generalizations aren't law. Technically, a phone company can assign any string of digits — your phone number, *their* phone number, or even their birthdays — as a SPID.

- In Europe, you have either NET3, EuroISDN or an older protocol, such as ITR6 in Germany or VN2 in France. The Zephyr only works with EuroISDN although it may work with NET3.
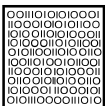
ISDN TIP!

If your Telco did not properly follow the ordering information you sent them from the appendix of this manual (you did use that information, didn't you?)

---

[1] The PTP protocol does not require SPIDs, so ZephyrExpress does not send or check them but assumes the line is okay. Meanwhile, the telco is still waiting for those darn SPIDs.

they may have given you a 2 B-channel AT&T "Custom Point to Multipoint" line with 2 phone numbers and 2 SPIDS. ZephyrExpress does not support this configuration.

If you have Custom Point to Multipoint, you can often get things working temporarily by following these steps:

1. Remove both SPIDS and both directory numbers

2. Set `Switch Type` to `PTP`

3. Have the distant codec call either of your phone numbers twice: that is, they should dial one of your ISDN numbers, drop the line, and then dial it again. They should be able to connect.

4. The line will probably now function normally for outgoing calls until ZephyrExpress is rebooted or the ISDN connection is reinitialized.

ISDN TIP!

If your Telco did not properly follow the ordering information you sent them from the appendix of this manual, they may have given you a 2 B-channel National ISDN-1 line with only one SPID and one phone number.

If you have only one SPID and one phone number on a Natl I-1 line, you can often get things working by following these steps:

1. Enter the SPID they gave you as `SPID#1`

2. Enter your 7-digit directory number twice as both Directory 1 and Directory 2

A REALLY IMPORTANT ISDN TIP!

Carefully verify that you have the correct SPID from the Telco, and that you have properly entered it into the Zephyr.

SPIDs must exactly, perfectly correct for ISDN to work! We've seen a few cases where numbers were transposed by Telco personnel, so it might be a good idea to re-check with them if things are pointing in this direction.

For a list of known working SPIDS by Telephone Company, see the Appendix.

Sometimes the information in the Business Office computer is wrong. By verifying what is programmed into the switch itself you can save time.

### *Can you place a "pots" call?*

From the dialing screen, change the `Mode` from `Zephyr` to `Phone` and try placing a call. Try it on both outgoing lines. If you can place ordinary analog calls but not data calls, then the phone company probably didn't give you CSD (Circuit switched data) capability. The line won't work for data, and won't carry high-quality audio calls.

The faxable order form in the appendix specifies CSD. Make them give it to you: it's the reason you specified ISDN.

### *Is your long-distance carrier OK?*

If you can successfully call locally, but long-distance calls don't go through, ZephyrExpress usually displays `Far end disconnect`, `No route available`, or `Incompatible Bear Cap` as the ISDN "Cause" phrase.

You may try another by dialing the appropriate 10XXX prefix.

Some carriers and codes that we've tried are:

*   MCI        10222

*   AT&T       10288

*   Sprint     10333

You must dial the full number, including the 1 or 011 + country code following the prefix. You may program the 10xxx prefix into the `ISDN: Outside Line Prefix` field, but you must remove the prefix before attempting local calls.

In our experience, other carriers are unlikely to be able to handle ISDN data calls. Let us know if you find otherwise.

You should also try the call at different bit rates (56/64 Kbps): that can affect how the call gets routed through the network.

### *What does the ISDN "Cause" phrase say?*

```
Line 1 disconnect: No
route to destination

        OK
```

This phrase or something similar will appear on the LCD after dialing. It comes directly from the telephone company equipment and can be valuable for troubleshooting. Generally these phrases are self-explanatory.

Getting a Cause phrase means you are at least talking to the Central Office equipment. There is a full description of the phrases and their meanings later in this section.

### *Are you using the Zephyr's internal NT1?*

The green LINE DIRECT LED is on the rear panel, near the U jack of North American ZephyrExpress units. If there's no ISDN line connected, or the line is dead, the light will blink rapidly (about 5 Hz). If NT1 can contact the central office, the light will blink slowly (about 1 Hz). The LED comes on steadily when all handshaking is completed and the basic line connection is good.

If the LED is off, or blinking slowly, there could be a problem with ZephyrExpress. Sometimes disconnecting the ISDN line and rebooting the system will fix things: when power is reapplied, the light should blink rapidly until you re-connect the line..

### *Are you using an external NT1?*

If so, check the LEDs on the unit. The following indicators are typical, but not all NT1s will work this way.

- **Power.** Should be lit. In North America, power for the NT1 is normally provided by the terminal device; in this case, ZephyrExpress. Units sold in North America have both S and U interface jacks (and an internal NT1), and provide power over the 8-pin S interface cable. Elsewhere, power is normally provided by a separate NT1 power supply. Units sold outside North America have just an S interface jack, and do not provide power.

> ⚠️ WARNING!
>
> Do not power the NT1 from both ZephyrExpress and an external supply. If you're using a North American ZephyrExpress and an external NT1, disconnect any other power supply.

- A blinking Power light usually indicates insufficient voltage.

- **Active.** Lights green when all is well.

- **Terminal Error**. Lights red to indicate a problem on the terminal (ZephyrExpress) side. Could be a problem with the cable or with ZephyrExpress itself.

- **Line Error**. Lights red to indicate a problem with the Telco line. This means a very basic kind of problem; usually no physical connection.

> HOT TIP!
>
> You can use an analog phone, high-impedance headphones, or a telephone "butt" test set to do a basic check of ISDN lines:
>
> You should hear a clicking sound or white noise. If you don't, the line is dead.

- Combinations of lights and blinking allow a fine degree of explanation, see the NT1manual for complete information.

## Intermittent ISDN Problems

These are among the most difficult kinds of problems to solve. Rest assured, however, that we've yet to lose one! This section addresses problems where the NT1 indicates that the line is good.

The most important tool is ZephyrExpress' integrated ISDN analyzer. This lets you look at the transactions between the Telco line and ZephyrExpress, to help you figure out where the problem lies.

The usual cause is a Telco problem, like mis-programming or an intermittent connection.

If you are convinced that the problem lies with the ISDN line, you may be able to convince the Telco to have an engineer look at the line with an analyzer. This works much like the Zephyr's internal analyzer, but with more detail, and in a format which is more readily understandable by telephone people. This procedure may sometimes be the only way to get a phone company to pay attention: it's hard to deny evidence provided by your own test set.

Problems which happen at a certain time of day, or a certain day of the week could be related to diagnostic routines the Telco may run periodically.

In rare cases, there could be a software bug in the Zephyr which reacts badly to something coming from the ISDN line. Telos customer support will communicate the problem to our engineering guys for resolution once it has been verified.

## ISDN Cause Phrases/Values

When there is an ISDN problem in the network, a phrase appears on the Zephyr's LCD (or using the built-in protocol analyzer) which communicates information about the cause of the trouble. These "Cause Values" are numbers generated by the network, which the Zephyr translates to the associated phrases. When possible, we use those suggested by the Bellcore standard.

Note that in some cases there may be more than one meaning. This can frequently be evaluated by whether the message has been received by the calling party or the called party.

**Cause No. 1 - Check number, redial**

This cause indicates that the called party cannot be reached because, although the called party number is in a valid format, it is not currently allocated(assigned).

**Cause No. 2 - No route to network**

This cause indicates that the equipment sending this cause has received a request to route the call through a particular transit network which it does not recognize. The equipment sending this cause does not recognize the transit network either because the transit network does not exist or because that particular transit network, while it does exist, does not serve the equipment which is sending this cause.

**Cause No. 3 - No route to destination. Prefix 1 dialed in error**

This cause indicates that the called party cannot be reached because the network through which the call has been routed does not serve the destination desired. This cause is supported on a network-dependent basis.

or

A 1 was dialed when not required. Redial without the 1.

**Cause No. 4 - No prefix 1**

The prefix 1 is required for this number.

**Cause No. 6 - Channel unacceptable**

Channel on called unit is not available.

**Cause No. 8 - Call is proceeding**

Call in process, please stand by.

**Cause No. 14 - Excess digits received, call is proceeding**

More digits were dialed than expected. Called number has been truncated to the expected number.

**Cause No. 16 - Far end disconnect / Normal call clearing**

This cause indicates that the call is being cleared because one of the users involved in the call or the switch has requested that the call be cleared. Under normal situations, the source of this cause is not the network.

**Cause No. 17 - Busy, try again later**

This cause is used to indicate that the called party is unable to accept another call because the user busy condition has been encountered. This cause value may be generated by the

called user or by the network. In the case of user determined user busy it is noted that the user equipment is compatible with the call.

### Cause No. 18 - No far end response

This cause is used when a called party does not respond to a call establishment message with either an alerting or connect indication within a prescribed period of time.

### Cause No. 19 - No answer

This cause is used when the called party has been alerted (has provided an alerting indication) but does not respond with a connect indication within a prescribed period.

### Cause No. 21 - Call rejected

This cause indicates that the equipment sending this cause does not wish to accept this call, although it could have accepted the call because the equipment sending this cause is neither busy nor incompatible.

This cause may also be generated by the network, indicating that the call was cleared due to a supplementary service constraint.

### Cause No. 22 - Number changed

This cause is returned to a calling party when the called party number indicated by the calling party is no longer assigned, The new called party number may optionally be included in the diagnostic field. If a network does not support this cause, an unassigned number message (cause number 1) may appear instead.

### Cause No. 26 - Non-selected user clearing

This cause indicates the user has not been awarded the incoming call.

### Cause No. 27 - Dest. out of order

This cause indicates that the destination indicated by the user cannot be reached because the interface to the destination is not functioning correctly. The term "not functioning correctly" indicates that a signal message was unable to be delivered to the remote party; e.g. a physical layer or data link layer failure at the remote party, or user equipment offline.

### Cause No. 28 - Incorrect number (Invalid number format, address incomplete)

This cause indicates that the called party cannot be reached because the called party number is not in a valid format or is not complete.

### Cause No. 29 - Facility rejected/Special intercept announcement: undefined code

This cause indicates that the user cannot use this feature

or

This cause value indicates that a user in a special business group (i.e. Centrex) dialed an undefined code

**Cause No. 30 - Result of a STATus ENQuiry/Special Intercept announcment:# unassigned**

This cause is included in the STATus message the user sends to the switch when the reason for generating this message was a prior receipt of a STATus ENQuiry message

or

This value indicates that a user from outside a special business group (i.e. Centrex) has dialed a number associated with the business group which is unassigned.

**Cause No. 31 - Network disconnect/Special Intercept Announc.:Call blocked**

This cause is used to report a normal event only when no other cause in the normal class applies.

or

This cause is used to indicate that a user in a special business group (i.e. Centrex) has violated an access restriction feature.

**Cause No. 34 - No circuit available**

This cause indicates that there is no appropriate circuit/channel presently available to handle the call. May be due to switch congestion as well as trunk congestion.

**Cause No. 38 - Net out of order**

This cause indicates that the network is not functioning correctly and that the condition is likely to last a relatively long period of time e.g. immediately re-attempting the call is not likely to be successful.

**Cause No. 41 - Net problem, redial**

This cause indicates that the network is not functioning correctly and that the condition is not likely to last a long period of time; e.g. the user may wish to try another call attempt almost immediately. May also indicate a data link layer malfunction locally or at the remote network interface or that a call was cleared due to protocol error(s) at the remote network interface.

**Cause No. 42 - Net busy, redial**

This cause indicates that the switching equipment generating this cause is experiencing a period of high traffic.

**Cause No. 43 -  Access information discarded**

This cause indicates that the network was unable to deliver user information (i.e. subaddress) to the remote user as requested.

**Cause No. 44 - No channel available**

This cause is returned when the circuit or channel indicated by the requesting entity cannot be provided by the other side of the interface.

**Cause No. 47 - Resource unavailable**

This cause is used to report a resource unavailable event only when no other cause in the resource unavailable class applies.

**Cause No. 50 - Requested facility not subscribed**

This cause is used to report that the cannot use this feature because s/he has not subscribed to it.

**Cause No. 51 - Bearer capability incompatible with service request**

This cause indicates a user request for action was rejected because the action was incompatible with the capability of the call.

**Cause No. 53 - Service operation violated**

This cause indicates the caller has violated the service operation

**Cause No. 57 - Data/voice not auth./Bearer capability not authorized**

This cause indicates that the user has requested a bearer capability which is implemented by the equipment which generated this cause but the user is not authorized to use it. This is a common problem caused by wrong Telco provisioning of the line at the time of installation.

**Cause No. 65 - Incompatible bearcap**

This cause indicates that the equipment sending this cause does not support the bearer capability requested.

**Cause No. 69 - Request facility not implemented**

This cause indicates that the network (or node) does not support the request bearer capability

**Cause No. 70 - Restricted only**

This cause indicates that the calling party has requested an unrestricted bearer service but the equipment sending this cause only supports the restricted version of the requested bearer capability.

**Cause No. 81 - Invalid call reference value**

This cause indicates the equipment sending this cause received a message with a call reference that is not in use on the user interface.

**Cause No. 88 - Incompatible dest.**

This cause indicates that the equipment sending this cause has received a request to establish a call which has low layer compatibility, high layer compatibility or other compatibility attributes (e.g. data rate, DN subaddress) which cannot be accommodated.

**Cause No. 96 - Info missing**

This cause indicates that the equipment sending this cause has received a message which is missing an information element which must be present in the message before that message can be processed.

**Cause No. 97 - Message type nonexistent or not implemented**

This cause indicates the equipment sending this cause received a message with a message type it does not recognize because the message is undefined, or it is defined but not implemented by the equipment sending this cause.

**Cause No. 99 - Information element non existent or not implemented**

This cause indicates the equipment sending this cause received a message that includes an information element identifier not recognized because the information element identifier is undefined, or it is defined but not implemented by the equipment sending this cause.

**Cause No. 100 - Invalid information element contents**

This cause indicates the equipment sending this cause received an information element that it has implemented; however one or more fields of the information element are coded in a way that has not been implemented by the equipment.

**Cause No. 101 - Message not compatible with call state/Protocol error threshold**

This cause indicates the equipment sending this cause received a message that procedure indicate is not permissible at this time.

or

This cause indicates that the switch sending this cause is clearing the call because a threshold is being exceeded for multiple protocol errors during an active call.

**Cause No. 102 - Timeout disconnect**

This cause indicates that a procedure has been initiated by the expiry of a timer in association with error handling procedures.

**Cause No. 111 - Protocol Error, Unspecified**

Among other things, this cause can be displayed if you failed to dial the appropriate code for an outside line from an internal system (usually the digit 7, 8, or 9). Also may be returned if you have some types of restrictions as to the number of calls, etc.

**Cause No. 118 -**

**Cause No. 127 - SW56 disconnect/Internetworking, unspecified**

This cause indicates that an interworking call (usually a call to SW56 service) has ended. May also be seen in the case of a non-specific rejection by your long distance carrier (try again at a different rate).

# SCHEMATICS
# and DATA SHEETS

# SECTION 11

# APPENDIX

# ABOUT AUDIO LEVELS

## "Nominal" Levels

The following will help to understand what we mean by *nominal*: A mixing console has its analog VU meter calibrated so that a test tone set to the 0 dB red-green junction outputs +4 dBu to ZephyrExpress. The mixer is adjusted so that the VU meter looks "normal" on a music or voice program. A nominal +4 dBu is being sent to ZephyrExpress.

## Level, Gain, dBu and dBm

We use dBu, rather than dBm, when describing both input and output levels. dBu is decibels referred to a voltage of 0.7746 volts; unlike dBm, it does not imply any value of circuit impedance or power. ZephyrExpress, as with most modern broadcasting and pro audio equipment, has an output impedance much lower than the input impedance. It operates on a voltage transfer basis and therefore, the dBm, as a power unit, is not appropriate – as the proper unit for voltage based systems is dBu.

dBm is decibels referred to a power level of one milliwatt across 600Ω. The 0 dBu value of 0.7746 volts is the voltage across a 600Ω resistor when exactly one milliwatt is being dissipated in the resistor, which means that under the condition that the measurement is in a 600Ω circuit, dBu and dBm are numerically equivalent.

Which brings us to our concern here: How does ZephyrExpress get along with other equipment?

ZephyrExpress' line input can be set to a nominal +4dBu for professional equipment, or to a nominal -10dBu for consumer equipment. But the gain can be fine-trimmed by using the front-panel volume controls. The only time input impedance matching becomes important is with older transformer-coupled devices or passively-equalized analog phone lines. In this case, a 680Ω resistor might be necessary across the input to prevent frequency-response problems.

However, ZephyrExpress' line outputs are designed to feed modern, high-impedance +4dBu devices. This is particularly important with the direct RECEIVE outputs, which have no gain adjustment.

The outputs themselves have a 200Ω source impedance. When connected to a high impedance input, levels should match as expected. However, if the output is terminated in 600Ω, a drop of 2.5 dB will be observed. This is due to the voltage divider effect of the internal resistance being loaded by the external 600Ω.

Incidentally, +4 dBu is 1.23 VAC, and this value can be read on a Digital Voltmeter (such as the common Fluke) to confirm that this level is what you have.
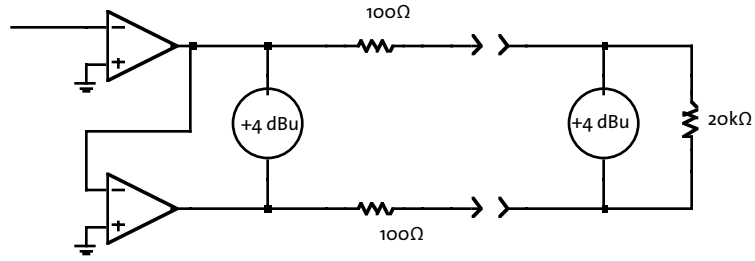
*Here, the output is loaded with a high impedance, as would be the case with modern equipment. ZephyrExpress' voltage level is conveyed to the load with no noticeable attenuation.*
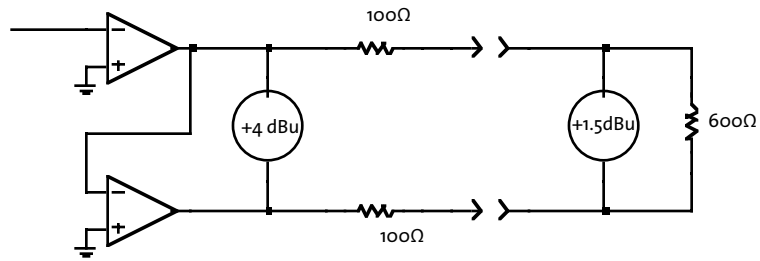


*When a 600 Ω load is used, the voltage divides, with 2.5 dB being lost in Zephyr-Express' internal source resistance. Note that, in this case, the meter across the 600 Ω resistor can be said to be indicating dBu or dBm.*

# CODEC COMPATIBILITY INFORMATION

We believe strongly in the benefits of compatibility. That is why we include a variety of operational modes designed to accommodate the largest number of non-Telos codecs.

We are constantly testing compatibility in our own labs, as well as accumulating information from others regarding using ZephyrExpress with other codecs. For the latest compatibility information please check our web site, http://www.Zephyr.com, or contact Telos customer support.

This is an area constantly in flux, so please contact us with any specific needs not covered below. We also appreciate your contacting us with any information which could be of use to others.

|  | CCS<br>CDQ2000 | CCS<br>CDQ1000 | CCS<br>CDQPrima | CCS<br>Roadrunner | CCS<br>Micro 56<br>Micro 64 | — COMREX —<br>DX200 | DXP/DXR<br>NEXUS | INTRAPLEX<br>4400 |
|---|---|---|---|---|---|---|---|---|
| **LAYER 3** | N/S | N/S | YES | YES | N/S | N/S | N/S | N/S |
| **LAYER 2** | | | | | | | | |
| J-STEREO | YES | N/S | YES | N/S | N/S | YES | N/S | NO |
| DUAL | YES | N/S | YES | N/S | N/S | YES | N/S | N/S |
| MONO | YES | YES | YES | YES | N/S | YES | N/S | N/S |
| 128 MONO | YES | N/S | YES | YES | N/S | YES | N/S | N/S |
| **G.722** | N/S | YES | YES | YES | YES | YES | YES | N/S |

**GENERAL NOTES:**

- YES = Compatible with ZephyrExpress.

- N/S=Not supported by other codec.

- ZephyrExpress supports ISO/MPEG Layer II at 48kHz sampling only.

- ZephyrExpress supports "CCS CDQ protocol compatible" channel splitting for ISO/MPEG Layer II stereo and mono 128 modes.

- ZephyrExpress G/722 uses statistical sync. G.722 sample rate is fixed at 16 kHz regardless of menu setting.

**CCS NOTES:**

- CDQ 1000: Switch off H.221 (DIP switch 7 , on back, in down position; to "AUTO" mode) on CDQ to transmit from CDQ to ZephyrExpress. Terminal adapter must have "BONDING" turned off and be set to 64/56Kbps. Switch on ISO/MPEG bit stream (DIP switches 6 and 8 on back, to up position).

- CDQ 2000: Switch to "Decoder independent from encoder" (Decoder DIP switch #6 in up position; "INDEPENDENT"). Terminal adapter must have "BONDING" turned off and be set to 64/56Kbps.

- Prima: must be set to front panel option "DECODER/GENERAL/INDEP=YES" OR Serial port command "DIN YES". To transmit G.722 sample rate must be set to 16 kHz. Only Primas with serial numbers beginning with 2 support Layer 3.

**COMREX NOTES:**

- DX200 must have ERROR PROTECTION set to "ON" and IMUX set to "TELOS/CCS" or "CCS". Terminal adapter must have "BONDING" turned off and be set to 64/56Kbps.

- Nexus must have "H.221" off.

|  | AEQ | — DIALOG 4 — |  | GLEN | NAGRA | RE | PHILLIPS | YouCom |
|  | ACD3001 | MUSICTAXI | REPORTER | **SOUND** | ARES-C | 660/661 | Baby Blue | ReporterSet |
|---|---|---|---|---|---|---|---|---|
| **LAYER 3** | N/S | LIMITED | MONO | N/S | N/S | N/S | N/S | N/S |
| **LAYER 2** | | | | | | | | |
| J-STEREO | N/S | NO | NO | N/S | NO | see note | N/S | N/S |
| DUAL | N/S | NO | NO | N/S | NO | see note | N/S | N/S |
| MONO | YES | NO | NO | N/S | YES | YES | YES | YES |
| 128 MONO | N/S | NO | NO | N/S | NO | NO | N/S | NO |
| G.722 | YES | YES | YES | YES | YES | YES | YES | YES |

**DIALOG4 NOTE:**

- ZephyrExpress can transmit to DIALOG4 in the modes indicated. For DIALOG4 to transmit to ZephyrExpress, compatibility mode must be set correctly. MUSICTAXI must have "Configuration/ISDN accept mode" set to ALL, and "Configuration/ISDN mode" set to RI, DNRI, or DNR depending on firmware version.

**NAGRA NOTE:**

- Nagra "SYNC" mode must be set to "SRT".

**RE NOTE:**

- Modes which require greater than 64kpbs will not work. Terminal adapter must have BONDING turned off and be set to 64/56 kbps. Zephyr can receive Layer 2 JStereo, Stereo, and Dual from RE when transmitted at 64kbps.

## Comprehensive ZephyrExpress Compatibility List

| | | |
|---|---|---|
| **yes!** | : | compatible, tested by Telos R&D |
| **no!** | : | incompatible, tested by Telos R&D |
| **yes+** | : | compatible, tested by someone else |
| **no-** | : | incompatible, tested by someone else |
| **yes?** | : | compatible, probably (from specs) |
| **no?** | : | incompatible, probably (from specs) |
| **?** | : | unknown |
| **-** | : | this mode is not supported by the product |

- **AETA**

  Incompatible... because of proprietary L-II implementation?

- **AEQ**

  L3-Mono    :  -
  L3-Stereo   :  -
  L2-Mono@48 :   yes!

  L2-Stereo    :  ?
  L2-M128     :  ?
  G.722        :  ?

- **CCS CDQ1000**

  L3-Mono    :  -
  L3-Stereo   :  -
  L2-Mono@48 :   yes!

L2-Mono@24 : must be set to ISO bitstream.

L2-Stereo : -

L2-M128 : -

G.722 : yes! ZephyrExpress can always receive. CDQ must have H.221 switched off in order to receive from ZephyrExpress

- **CCS CDQ2000**

    CDQ2000: Decoder DIP switch 6 up (decoder independent from encoder).

    L3-Mono : -

    L3-Stereo : -

    L2-Mono@48 : yes!

    L2-Stereo : yes!

    L2-M128 : yes!

    G.722 : -

- **CCS CDQ2001**

    See CDQ2000. 48kHz sample rate.

- **CCS Prima**

    Prima: Front panel option decoder/general/indep=yes

    Or serial port command "din yes". Sample rate set to 48kHz

    L3-Mono :
    (some units have layer III and are compatible)

    L3-Stereo :
    (some units have layer III and are compatible)

    L2-Mono@48 : yes!

    L2-Mono@24 : yes!

    L2-Stereo : yes!

    L2-M128 : yes!

    G.722 : yes!

- **Comrex Layer-II DX-200**

    L3-Mono : -

    L3-Stereo : -

    L2-Mono@48 : yes!

    L2-Stereo : yes! Must be switched to "Telos /CCS compatibility" mode for stereo.

    L2-M128 : yes!

    G.722 : Yes

- **Comrex G.722 DXP and DXR**

    G.722 : yes! Standard G.722 only

- **Dialog4 MusicTaxi**

    ZephyrExpress `Codec: Compatibility` must be set to `Dialog4`.

    MusicTaxi "Configuration /ISDN accept mode" must be set to ALL and "Configuration/ISDN mode" to RI.

    L3-Mono : yes+. There will be dropouts in the MusicTaxi to ZephyrExpress path because of D4's asynchronous mode.

    L3-Stereo : yes+, but see above

    L2-Mono@48 : ?

    L2-Stereo : ?

    L2-M128 : ?

    G.722 : yes?

- **Dialog4 ReporterSet**

    L3-Mono : ?

    L3-Stereo : ?

    L2-Mono@48 : ?

    L2-Stereo : ?

    L2-M128 : ?

    G.722 : ?

- **Dialog4 Slimline**

    ZephyrExpress `Codec: Compatibility` must be set to `Slimline`.

    Slimline "Configuration /ISDN accept mode" must be set to ALL and "Configuration/ISDN mode" to RI.

    L3-Mono : yes+. There will be dropouts in the MusicTaxi to ZephyrExpress path because of D4's asynchronous mode.

    L3-Stereo : yes+, but see above

    L2-Mono@48 : ?

    L2-Stereo : ?

    L2-M128 : ?

    G.722 : yes?

- **EELA**

    L3-Mono : -

    L3-Stereo : -

    L2-Mono@48 : yes?

    L2-Stereo : no?

    L2-M128 : no?

    G.722 : yes?

- **Intraplex**

    L3-Mono : -

    L3-Stereo : -

    L2-Mono@48 : yes?

L2-Stereo   : ?
L2-M128    : ?
G.722      : ?

- **PKI G.722 phone**

  G.722      : no? Because of
  H.221 and SETUP indicator.

- **PKI Magic**

  Completely incompatible because of
  J.52?
  L3-Mono    : ?
  L3-Stereo   : ?
  L2-Mono@48 : ?
  L2-Stereo   : ?
  L2-M128    : ?
  G.722      : ?

- **Philips MPR LIIBlue**

  TrueBlue:    : Must be switched to
  "CCS compatibility" mode for
  modes that use both B-channels
  (stereo, mono128)
  L3-Mono    : -
  L3-Stereo   : -
  L2-Mono@48 : yes?
  L2-Stereo   : yes?
  L2-M128    : yes?
  G.722      : yes?

- **Philips MPR Baby Blue**

  Baby Blue   : Must be switched to
  "CCS compatibility" mode for
  modes that use both B-channels
  (stereo, mono128)
  L3-Mono    : -

L3-Stereo   : -
L2-Mono@48 : yes!
L2-Stereo   : -
L2-M128    : -
G.722      : yes!

- **RE**

  L3-Mono    : -
  L3-Stereo   : -
  L2-Mono    : yes!
  L2-Stereo   : no. Wrong channel
  splitting.
  L2-M128    : no. See stereo.
  G.722      : ?

- **Telos Zephyr** (rack-mount)

  L3-Mono    : Of course, yes!
  L3-Stereo   : Some rack-mount
  Zephyrs are mono-only.
  L2-Mono    : yes!
  L2-Stereo   : Some rack-mount
  Zephyrs are mono-only.
  L2-M128    : Mono-only Zephyrs
  do not support this mode.
  G.722      : yes!

- **You/Com**

  L3-Mono    : -
  L3-Stereo   : -
  L2-Mono@48 : yes+
  L2-Mono@24 : yes+
  L2-Stereo   : yes+
  L2-M128    : yes?
  G.722      : yes?

# FINDING PUBLIC ISDN SITES

When you need to find a ZephyrExpress, Zephyr, or compatible codec to connect with in another city, there are a number of resources to which you can turn. There are many hundreds of sites around the country – and the world – which offer a wide variety of services.

Some are music recording studios; some are broadcast stations; some are recording studios specializing in broadcast projects; and many are home studios belonging to announcers or producers. Some come with award-winning engineering or vocal talent, while others are offered "bare." Chances are, you'll find what you are looking for somewhere below...

- **Telos World Wide Web site**

  http://www.zephyr.com will connect you to a variety of information about Zephyr-Express and ISDN, including site pointers.

- **Audiobahn**

  A list of broadcasters and sound studios using codecs, maintained by Jay Rose's Digital Playroom, a broadcast promo studio in Boston.

  These sites have submitted their names so other professionals can call them for newsfeeds, help with remotes, audio transfers, and so on. All services are optional, and fees are negotiated by the parties involved. The list is maintained as a public service to help you find each other... after that, it's up to you.

  Audiobahn is updated approximately once a month. Submission information is on the list, and has to follow a precise format. Comments may be sent by e-mail to: jay@dplay.com.

  - **WWW:**
    http://www.dplay.com

  - **Paper:**
    Telos usually has printouts of the current list. Comrex keeps the list in a fax-on-demand system, for use by their customers.

- **Digifon List**
  Dave Immer's Digifon publishes a list of codec users around the world, heavily geared toward recording studios. Most (those using equipment which supports MPEG Layers II or III) are able to work with ZephyrExpress. Those that cannot (mostly APT-X, but some Dolby) are able to do so via various bridge services, including one offered by Digifon itself. The list is available on the World Wide Web. Send e-mail comments to immer@digifon.com.

  - **WWW:**
    http://www.digifon.com

  - **Phone**
    203-254-0869
    203-256-5723 (Fax)

- **EdNet**
    San Francisco, CA
    415.274.8800

  Service to recording studios and broadcasters. Operates about two hundred sites, mostly at recording studios. They handle billing and booking, etc. Provides "translation" service. Full-service Zephyr and ZephyrExpress dealer.

# List of Known Working SPIDs by Telephone Company

Your SPIDs may be different!  The phone company can assign any darned thing they want to as a SPID... but they *usually* adhere to these standards.

Note, for each line there is only one configuration which will work.  Your SPID is distinct from your telephone number and does not necessarily contain your area code or telephone number (although this is generally the case).

A standardized SPID format for national ISDN of XXXYYYZZZZ0101 has been phased in and is worth trying.

Incorrect SPIDS will be indicated by a line status of "Wait, Init" on ZephyrExpress.

Often the Telco will leave the last "00" or "01" off the end of your SPID when giving it to you.

For the latest info, check http://www.zephyr.com. Additional SPIDS can be found at INTEL's SPID Page (http://support.intel.com/enduser_reseller/isdn/spid-tip.htm).

XXX=Area code          YYY=exchange          ZZZZ=phone number

## Ameritech

XXXYYYZZZZ0101
 — or —
XXXYYYZZZZ0111
 — or —
01YYYZZZZ011
 — or —
XXXYYYZZZZ01

## Bell Atlantic

01YYYZZZZ00
 — or —
01YYYZZZZ000
 — or —
01XXXYYYZZZZ000
 — or —
XXXYYYZZZZ100
 — or —
XXXYYYZZZZ0101
 — or for former Nynex sites —
XXXYYYZZZZ0000

## Bell South

XXXYYYZZZZ0101
 — or —
XXXYYYZZZZ0100
 — or —
XXXYYYZZZZ01
 — or —
01YYYZZZZ0
 — or —
01YYYZZZZ000

## Cincinatti Bell

01YYYZZZZ000
 — or —
SPID1 00YYYZZZZ01

SPID2 00YYYZZZZ02
 — or —
00YYYZZZZ01

## Fort Mills Telephone

XXXYYYZZZZ

### GTE

XXXYYYZZZZ0101

— or —

01YYYZZZZ000

— or —

XXXYYYZZZZ00

— or —

SPID1 XXXYYYZZZZ01
SPID2 XXXYYYZZZZ02

### Northern Pittsburgh Telephone

XXXYYYZZZZ000

### Nynex

See BellAtlantic

### Pacific Bell

XXXYYYZZZZ0101

— or —

SPID1 XXXYYYZZZZ01

SPID2 XXXYYYZZZZ02

— or —

01YYYZZZZ000

— or —

SPID1 XXXYYYZZZZ1

### SNET (Southern New England Telephone)

01YYYZZZZ00

### Southwestern Bell

XXXYYYZZZZ0101

— or —

01YYYZZZZ000

— or —

01YYYZZZZ00

— or —

XXXYYYZZZZ01

### Sprint/Centel

XXXYYYZZZZ100

— or —

XXXYYYZZZZ1

— or —

SPID1  XXXYYYZZZZ000
SPID2 XXXYYYZZZZ100

### US West

XXXYYYZZZZ1111

— or —

01YYYZZZZ00

— or —

01YYYZZZZ000

---

# ISDN BRI, ZEPHYR, AND YOU

This section applies to users in North America and is intended to help you when you order your ISDN line. We assume you have read the description of ISDN in section 8, and are familiar with ISDN's basic concepts. If you haven't read that section, do it now. It'll help you understand the following information better.

ISDN (Integrated Services Digital Network) is the information superhighway at your front door. Because ISDN has only been recently introduced in some parts of the country, there is a great deal of confusion about what it is and how to get it. If you know who to call and provide them with complete information, the ISDN ordering process can be simple and straightforward. This guide takes you through the process step-by-step. Telos Customer Support is available by phone or fax to answer any further questions you may have.

From the perspective of the telephone network, each channel appears to be a separate line with its own number and independent dial-out capabilities. Since each has to be dialed or answered separately, they appear to be "lines" to users also. We refer to a B channel as a "line" on ZephyrExpress menus.

ZephyrExpress' internal interface (sometimes called by its generic name "Terminal Adapter") is used to connect to digital ISDN telephone lines. It easily adapts to the various types of service offered by the range of Central Office (CO) switches installed by telephone companies in the USA and Canada.

# Ordering ISDN

## Dealing with the Phone Company

As is often the case when broadcasters interface with phone people, the lines of communication on ISDN can get a little tangled.

The first order of business is to find someone who knows what ISDN is. While your usual account agent will be the normal entry point, you will probably be talking to a number of phone people before you find one who understands your needs.

Some of the regional Bell companies offer a single point of contact number for switched digital services. Note that in many cases the phone company will need to do a "loop qualification" (line loss test) from your site before they can verify that ISDN will be available. Some telephone companies use "resellers". If this is the case you should inquire what experience this reseller has with installing ISDN for audio codecs.

Here are some contact numbers we use – if you have particularly positive or negative experiences with these offices, please let us know.

| COMPANY | TELEPHONE NUMBER | WORLDWIDE WEB |
|---|---|---|
| Ameritech | 800-TEAMDATA (800-832-6328) | http://www.ameritech.com |
| Bell Atlantic | Business 800-570-ISDN (800-570-4736)<br>Residential 800-204-7332 | http://www.ba.com |
| Bell Atlantic areas that were formerly Nynex, call your account rep or: | | |
| | 800-GET-ISDN (800-428-4736) | http://www.bell-atl.com |
| Bell South | 800-428-ISDN (800-428-4736) | http://www.bell.bellsouth.com |
| Cincinnati Bell | 513-566-DATA (513-566-3282) | |
| GTE | 800-GTE-4WCN (800-483-4926) | http://www.gte.com |
| Natco | 800-775-6682 ext 288 | |
| Nevada Bell | Small Business 702-333-4811;<br>large business 702-688-7100 | |
| Nynex (see Bell Atlantic) | | |

Pacific Bell          800-4PB-ISDN (800-472-4736)          http://www.pacbell.com
                      For questions or assistance 403-944-8130

SNET                  800-420-4736                         http://www.ntplx.net

Stentor (Canada)   Fax server 800-578-4736, document #200 has list of local ordering numbers.
                   403-944-8130 (questions/assistance).

Southwestern Bell  800SWB-ISDN (800-792-4736)          http://www.sbc.com

US West               Fax server 800-728-4949             http://www.uswest.com

                      Small business 800-246-5226

                      For questions or assistance 206-447-4029

*You may need to call this number more than once to find someone who can arrange for fast installation for remotes. Be sure to explain that you are a broadcaster.

## Details, Details

In order to communicate accurately what it is you need, we think you should be familiar with the vocabulary used to describe ISDN. As with anything, for best results, it helps to know what you are talking about. For an in-depth glossary of terminology see manual appendix or visit Telos System's web site at http://www.zephyr.com

### IOC Capability Packages

More and more telcos are using ISDNOrdering Code (IOC) capability packages for ISDN ordering. If your telephone company uses these you need only tell them you need IOC package "S".

### Protocols

In a perfect world, all ISDN terminal equipment would work with all ISDN lines, without regard for such arcana as 5ESS, DMS100, CSV/CSD, SPIDs, etc. Unfortunately, the ISDN "standard" has been evolving for the past years and has only recently begun to settle down.

At their central offices, the telephone companies use either AT&T 5ESS, Northern Telecom DMS100, or Siemens EWSD switching equipment. While each will work with ZephyrExpress, there are some differences which need to be taken into account when lines are ordered and used. Each has a "protocol" – the language the user equipment and the telephone network use to converse (on the D channel) for setting up calls and the like.

There is a standard protocol which all switches may provide, called National ISDN 1 (NI-1). This protocol was standardized and specified by Bellcore, the technical lab jointly owned by the phone companies. However, both AT&T and Northern Telecom had "custom" versions of ISDN which predated the NI-1 standard and some switches have not been upgraded to the new format.

There is also a newer NI-2 and NI-97 standards, but they are designed to be compatible with NI-1 for all of the basic functions.

### SPIDs

Service Profile Identification (SPID) numbers are required in all but one the AT&T proto-cols. This number is given to the user by the phone company and *must be entered correctly* into ZephyrExpress in order for the connection to function. SPIDs usually consist of the phone number plus a few prefix or suffix digits. There is frequent confusion between phone numbers and SPIDs, even among telco personnel. While the SPID frequently includes the corresponding phone number, this is not necessarily the case.

The intention of the SPID is to allow the Telco equipment to automatically adapt to various user requirements by sensing different SPIDs from each type or configuration of user terminal. None of this matters with our application, but we must enter the SPIDs nevertheless. BellSouth has proposed, and most of the other telephone companies have agreed to implement, a standardized SPID for new National ISDN installations. They announced this would be phased in during 1996. As of this writing, it has generally been implemented by most telcos. The standard is area code+phone number+0101 (XXXYYYZZZZ0101).

Unless you are using the AT&T PTP protocol, your Telco service representative should give you one SPID for each B channel you order. *Don't let the phone company installer leave without providing you with the phone numbers of your B channels (called Directory Numbers, or DNs) and your SPIDs!*

### CSD and CSV

Recall that each ISDN BRI has two possible B channels. It is possible to order a line with one or both of the B channels enabled – and each may be enabled for voice and/or data use. Phone terminology for the class of service is CSV for Circuit Switched Voice and CSD for Circuit Switched Data. (In contrast to PSD, Packet Switched Data, which is possible but irrelevant to our needs.)

CSD is required for ZephyrExpress connections. Even though you may be sending voice, the codec bit-stream output looks like computer data to the phone network.

CSV is for interworking with standard voice phone service and allows ISDN to call to analog phone lines and phones. ZephyrExpress allows outgoing voice calls on either of the channels if you have CSV. ZephyrExpress' voice capability exists on both channels. Thus you can make calls to any normal telephone number on one channel while a program is being transmitted on the other. This is a feature you may want to use.

You will be ordering an ISDN BRI 2B+D line with CSD and CSV (sometimes called alternate CSD/CSV) available on any B channel.

### 56/64kbps

All ISDN BRIs have a 64 kbps per channel capacity and almost all local calls operate at this rate. Some long distance connections, however, operate at only 56 kbps.

ZephyrExpress provides standard rate adaptation (officially known as CCITT V.110) from 56 to 64 kbps when required. Bit rate adaptation happens automatically within Zephyr-Express depending upon the rate of an incoming call.

In our experience, the only sure way to know the capacity of a given connection is to try it, first at 64 kbps and then at 56 kbps if the higher rate fails.

If you find that you have the 56 kbps limit on your line, you might want to request the Telco to upgrade the "RBOC-to-IXC tandem" circuit.

### NT1s

The ISDN standard specifies two reference points, the "U" and the "S" interfaces. The U is the single-pair bare copper from the Telco CO. A device called a "Network Termination, Type 1" converts this to the two-pair S interface.

In Europe and Asia the NT1 is always provided by the phone company, and only the S interface may be on user equipment. ZephyrExpress units shipped outside of the USA and Canada have the "S" interface only. In the USA and Canada the NT1 is provided by the user and is therefore built-in to the ISDN terminal adapter.

### Terminals and Terminal Types

Any equipment connected to an ISDN line is a 'terminal' – whether phone, computer, or ZephyrExpress. Point-to-point lines support one terminal, while multipoint lines can have up to eight in some applications.

"Terminal Type" is a parameter sometimes requested by the phone people. The appropriate value for ZephyrExpress varies depending upon protocol and is included on the order forms.

### Long Distance Carriers

Not all long distance carriers can make reliable long distance connections for ISDN. We have found that the following work best (listed from better to worse): MCI, AT&T, Sprint. If you have difficulties with long distance calls you should contact Telos customer support for troubleshooting tips.

For unknown reasons, the long distance companies try to charge extra if you tell them you are using an ISDN line. We therefore advise that you order your long distance through your local telephone company. Tell them to "PIC" you to 288 for AT&T, 222 for MCI, or 333 for Sprint. If the local telephone company insists that you contact the long distance company directly, we advise that you call and just give them your phone numbers. Don't mention what type of line it is.

Since we advise you not mention that the line is an ISDN line, it will be best to call the number listed in your Yellow Pages under "Telephone Long Distance Companies" to order. If you are prepared to admit that you are using ISDN (**and prepared to pay a 25 to 50% surcharge**) you can contact the "Switched Data Services" ordering numbers:

| | | |
|---|---|---|
| AT&T | 800-222-0400 Business | 800-222-0300 Residential |
| MCI Worldcom | 800-727-5555 or 800-888-0800 Business | 800-444-3333 Residential |
| Sprint | 800-326-1015 | |

# The Faxable ISDN Order Form (for use in US only)

The following three pages can be copied, filled out, and faxed to the phone company to order an ISDN line. They should give the phone company all the information they need. *The majority of installations, if ordered in writing, with this information, go smoothly.* If you do experience problems Telos Systems customer support is available to assist you. You may wish to also consult manual section 10 (Advanced Problem Solving) for additional troubleshooting information.

Complete the top portion of the form and send all three pages to the phone company. Keep the form handy and show it to the installer who puts in your line, and ask the installer to verify that the line is configured as ordered.

# **FAXABLE ISDN BRI LINE ORDER FORM** Page 1 of 3

**To:**_____          Telephone Company

**Attention:**_____

**From:**                                                **Location for line:**

Company:_____          Company:_____

Address:_____          Address:_____

City/State/ZIP:_____          City/State/ZIP:_____

Phone:_____          Phone:_____

Contact:_____          Contact:_____


Long distance carrier:_____

Date needed:_____

Termination Date (if known:_____

We request an ISDN Basic Rate Interface (BRI) line for use with a ZephyrExpress hi-fidelity audio codec. This device is used to transmit audio using digital telephone services. It *requires* Circuit Switched Data (CSD). ZephyrExpress transmits audio in real time and therefore needs reliable constant bit rate (CBR) service. It can also make standard POTS voice calls using Circuit Switched Voice (CSV) service. Please advise us if there is a cost penalty for having both CSD and CSV. 10XXX long distance option must be set to "Yes". Clear channel 64kbps access should be provided. If only 56kbps is available, this is acceptable, but please notify us.

If you use IOC Capability Packages, please use Capability Package "S". If you do not use IOCs, use the information on the pages that follow.

ZephyrExpress has an integral BRI terminal adapter which supports these protocols:

- AT&T 5ESS: Custom Point-to-Point (5E4.2 or later), National ISDN-1

- Northern Telecom: Functional (PVC1), National ISDN-1 (PVC2)

- Siemens EWSD: National ISDN-1

We can use any of the protocols given above. Please let us know which protocol you will provide and the switch type. We will provide the NT1 and need a U interface with 2B1Q line coding on an a *standard, six-pin/4-conductor RJ-11-style modular jack*, of which only the center two conductors will be used.

You may call the manufacturer of ZephyrExpress, Telos Systems, at +1 216.241.7225 for any additional required information. Ask for "ISDN Customer Support."

## PROTOCOL: National ISDN-1

From AT&T 5Ess, Northern Telecom DMS100, and Siemens EWSD.

If you use IOC Capability Packages, please use Capability Package "S"

**CO Values:**

Line Type:  National ISDN-1

Bearer Service: CSD/CSV on both channels

TEI: One dynamic per number

10XXX Long Distance Prefix: Yes

***Turn off*** features such as: packet mode data, multiline hunt, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, etc.

**Note for EWSD running NI-97:** Switch must be programmed CLID=DN

Give us:

1) Two SPID numbers, depending upon active number of B channels

2) Two Directory Numbers

## PROTOCOL: AT&T Point-to-Point (Custom)

Software version 5E4.2 and above

**CO Values:**

Line Type (DSL class): Point-to-Point (PTP)

B1 Service: On Demand (DMD)

B2 Service: On Demand (DMD)

Maximum B Channels (MaxChan): 2

CSV Channels: Any

Number of CSV calls: 1

CSD Channels: Any

Number of CSD calls: 2

Terminal Type: A

Number Display: No

Call Appearance Pref:Idle

10XXX Long Distance Prefix: Yes

***Turn off*** features such as: packet mode data, multiline hunt, multiple call appearances, Electronic Key Telephone Sets (EKTS), shared directory numbers, accept special type of number, intercom groups, network resource selector (modem pools), message waiting, hunting, interLata competition, etc.

**Give us:**

1) One Directory Number

## PROTOCOL: AT&T Point-to-Multipoint (Custom)

**Not supported by ZephyrExpress**

Please provide National ISDN 1 or AT&T Point-to-Point (Custom)

## PROTOCOL: Northern Telecom DMS100 'Functional' (Custom, PVC1)

Northern Telecom DMS100 switches BCS 31 and above

**CO Values**

Line Type: Basic Rate, Functional

EKTS: No

Call Appearance Handling: No

Non-Initializing Terminal: No

Circuit Switched Service: Yes

Packet Switched Service: No

TEI: Dynamic

Bearer Service: CSD/CSV on both channels

10XXX Long Distance Prefix: Yes

**Give us:**

1) Two SPID numbers

2) Two Directory Numbers