Telos

Zephyr

Digital Network Audio Transceiver

The Best Way to Hear from There™

User's Manual

Manual Version: 3.0,

November, 1997

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 the Zephyr 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.

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Test Lines

To aid you in testing and demonstrating your Zephyr Telos Systems has the following test numbers available:

- USA: +216 781.9310, +216 781.9311 (Layer III Dual transmit @ 32 KHz sample rate)
- Germany: (49) 81 61 42 061 Dial this number twice (Layer III Dual transmit @ 32 KHz sample rate)
- Germany: (49) 81 61 42 062 (Layer II Mono @ 32 KHz)

Updates

The operation of the Zephyr is nearly entirely determined by software. A continuous program of improvement is underway. Please be sure to send in the registration form to be notified of new software releases.

Trademarks

Copyright

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.

IMPORTANT NOTE! The Zephyr memory backup is powered by Lithium battery BT-1. The following precautions must be followed when working on the Zephyr motherboard or when this battery is replaced:

1) Do not short the battery terminals (or traces connected to these terminals) together.

2) Lithium batteries contain lithium and may be considered hazardous. Local procedures must be followed when disposing of used batteries. Do not dispose this battery by burning.

3) Do not attempt to open the sealed battery container.

WARNING: To reduce the risk of electrical shock, do not expose this product to rain or moisture. Do not shower with the unit.

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.

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.”

User’s Manual

Telos Zephyr

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.

With the Zephyr, we've 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.

The Zephyr is 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 the 90s!

Steve Church

Conventions Used in this Manual

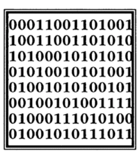
We use the following symbols to draw your attention to particularly important points.

IMPORTANT!

This information is essential to getting the Zephyr to work, or to prevent damage to it. To avoid headaches please read these.

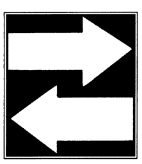
HOT TIP!

This information will probably come in handy at some point. You will probably wish to read these.



ISDN TIP!

Important information for those using ISDN, particularly the Zephyr's built-in ISDN terminal Adapter. Recommended reading for all ISDN users.

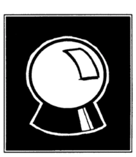


COMPATIBILITY TIP!

Important information about modes between Zephyrs, and particularly for between Zephyrs and other codecs. You should read these.

DEEP TECH NOTE!

Amaze your technical friends with your voluminous knowledge of the detailed technical intricacies of the Zephyr and ISDN and Coding technologies! Not necessary for the ability to operate the Zephyr, but this information might come in handy if you ever need to troubleshoot.



CURIOSITY NOTE!

Amaze your friends with your grasp of ISDN and Coding trivia! Not necessary for the ability to operate the Zephyr.

SECTION 1

QUICK RESULTS!

WHO CAN USE THIS SECTION?

You have just opened the carton of one of the world's most sophisticated pieces of audio transmission equipment. We know how you feel. You don't want to sit around and read a manual. You want to plug- in your new Zephyr and hear what it can do. You want to connect to a distant location and be amazed at how good audio can sound over the telephone. We know because we feel the same way when we get a new, expensive, piece of gear.

As much as we understand your excitement, we ask you to please peruse the following to get the most out of your Zephyr.

This manual is designed to fully instruct you on the capabilities of your Zephyr and to provide you with interesting and useful information on audio coding and digital telephony. (Unlike most other manuals, large sections don't even require that you be sitting in front of the product.) We hope you will take the time to read the manual. We trust you will enjoy it and that it will provide you with new and useful information that you can use to impress others.

If you need to put the unit in service today or have experience with other, similar equipment, this section will allow you to get started quickly with your Zephyr.

" Quick Results" Are for You If You...

- Know how to dial a telephone call.
- Are not intimidated by looking at the front panel of the Zephyr.
- Are willing to erase all of the settings in your Zephyr that may have been programmed before you got it.
- Have an ISDN line. (Before starting this process, US users should contact their telephone company to verify that the line was installed exactly as specified in the ISDN order instructions found in the appendix of this manual.) Sorry, but there are too many variables when connecting via Switched 56 or a dedicated digital service for us to include such a set- up in these "Quick Results." If you are using a non-ISDN setup see the instructions for your CSU/DSU and Section 6 Non ISDN Networks.
- Understand and have before you all of the items listed in the section that follows called "What you need before you start."
- Understand that these "Quick Results" only get you started and will not answer all of your questions or demonstrate all of the features of Zephyr.
- Promise to read the full manual later.

Now, let's get started.

WHAT YOU NEED BEFORE YOU START...

There are two categories of items you need: First, the equipment to connect your Zephyr to audio equipment and to the ISDN circuit with two B channels; second, some information that should have been provided to you by the person who installed your ISDN.

Hardware

- A Zephyr with a built- in terminal adapter. Sorry, but there are too many variables when working with an external terminal adapter for us to include such a set- up in these "Quick Results."

- 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. There is an NT1 built- in to your Zephyr's terminal adapter. Looking at the back of the unit, if there are two telephone- type modular jacks in the back, you have a built- in NT1. Generally, you will find this only in the US and Canada. If there is only one telephone- type connector, you need an external NT1. *Cable required: A standard telephone- type modular cable that fits into the lower jack on the Zephyr.*

        > IMPORTANT!
        >
        > An RJ-11 style 6-position plug can be inserted into the center of an RJ-45 style 8-position jack, if the Telco charged you extra and installed a RJ-45 jack for your ISDN line.

    2. OR- You have an older Zephyr with an external NT1 (or the European model). The NT1 is about the size of a small book, has some telephone- type connectors, and usually has some little indicator lights. *Two cables required: The first is a large, telephone- type modular cable with eight wires that fits into the upper (or only) 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.*

    3. OR- Your telephone company (Telecom) has provided one. Generally, this is the case in Europe and Asia. Cable required: A large, telephone- type modular cable with eight wires that fits into the upper (or only) jack on the Zephyr and a jack on the NT1 that is labeled "Terminal." Your NT1 should have been wired to the ISDN line by your Telco.

- Audio connections. You want to hear something, don't you? Connections are needed for sending audio to the remote location and receiving audio from it. You will need to provide the appropriate cables. For connection to the Zephyr, the two send cables must have male XLR connectors and the two receive cables must have female XLR connectors. The termination on the other side of the cables depends on your equipment. You may choose to use headphones to listen to the receive audio; in this case you will not need receive cables.

Information on your ISDN line

Zephyr needs to know some things about your ISDN line to work properly. Your ISDN circuit consists of two, digital “B channels”. Just like a regular telephone line, these channels have telephone numbers so that you can send and receive calls. The Zephyr, as with most ISDN equipment, refers to these channels are “lines” Unlike regular telephone lines, both lines may have the same telephone number assigned to them.

There are several different types of ISDN service. You must know what type you have. In addition, ISDN circuits in the US and some other countries may also have Service Profile IDentification (SPID) numbers. SPIDs, when used, *must* be programmed into the Zephyr. In rare circumstances, US users may also need Directory Numbers (DNs) if the telephone numbers are not incorporated into the SPIDs.

Here is a summary of what you need to know and a place to write it down. You may want to make a copy of this page and keep it next to your Zephyr.

✓ **ISDN type, check one:**

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

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

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

❒ Other *(May not be supported by Zephyr. Contact Telos Customer Support.)*

✓ **Your ISDN telephone numbers:**

Line 1 ____________________________________

Line 2 ____________________________________

✓ **SPID numbers, if applicable:**

Line 1 ____________________________________

Line 2 ____________________________________

ISDN TIP!

If you have European ISDN or AT&T Point-to-Point, it is acceptable if you only have one number that applies to both lines. If you have another ISDN type and only one number, contact Telos Customer Support.

ISDN TIP!

European users should disregard all references to SPIDs. Euro ISDN *does not* have SPIDs! If your ISDN configuration requires MSNs they may be entered in the MSN/SPID 1 & 2 screen.

✔ **DN numbers (7 digits), if applicable:**

Line 1 ______________________________

Line 2 ______________________________

If your ISDN line is connected to an internal phone system that requires you dial a prefix for an outside line, write it here: ________

SUCCESS, STEP-BY-STEP:

Connecting The Input/Output Audio

Got your audio cables ready? Here is where to connect them:

Send audio is connected on the back panel. Use the two XLR connectors labeled "INPUT, SEND TO NETWORK." Select MICrophone or LINE level using the push-button between the two connectors. *If you are not certain about how to make this connection, refer to Zephyr at a Glance (section 3).*

Receive audio can be connected from the headphone jack on the front panel. Alternately, it can be connected on the back panel. Use the two XLR connectors labeled "OUTPUT, RECEIVE FROM NETWORK." Select - 10 or +4 level using the push- button between the two connectors. *If you are not certain about how to make this connection, refer to Zephyr at a Glance (section 3).*

Connecting to The ISDN Line

Making the physical connection to the ISDN line varies with your NT1.

1. Zephyr's built- in NT1.

    - Using a standard RJ- 11 style telephone- type modular cable, interconnect the *lower* modular jack on the Zephyr and the jack installed by the telephone company. Be careful to insert the cable in the center of each jack, as it is the middle two wires that have the ISDN circuit.

2. User purchased external NT1

    - Using a large RJ- 45 style, telephone- type modular cable with eight wires, interconnect the upper (or only) jack on the Zephyr and a jack on the NT1 that is labeled "Terminal." *This cable must have eight wires and is provided with the Zephyr.*

    - Using a telephone- type modular cable, interconnect the jack on the NT1 that is labeled "Line" and the jack provided by the telephone company.

    - You do not need to provide power to the NT1. The NT1 is powered from the Zephyr. Note: If the NT1 has an external power supply *DO NOT* connect it! If this arrangement does not work contact Telos customer support.

3. Telephone company (Telecom) provided NT1

    - Using a large RJ 45 style, telephone- type modular cable with eight wires, interconnect the upper (or only) jack on the Zephyr and a jack on the NT1 that is labeled "Terminal." *This cable must have eight wires and is provided with the Zephyr.*

IMPORTANT!

In the case of (3), above, the NT1 will be powered by the Telco. If your Zephyr's terminal adapter does not show the Euro telecom approval symbol (looks like 2 hockey sticks in a circle) you should contact Telos Systems customer support before proceeding. This is because the Zephyr's built in supply and the power supply from the NT1 can conflict and cause damage to the Zephyr, NT1, or both.

- Your NT1 should have been wired to the ISDN line by your Telecom.

You do not need to provide power to the NT1. The NT1 is powered from the ISDN line or from the Zephyr.

Powering Up

Time to turn on the Zephyr!

- Connect the provided IEC power cable to the Zephyr. As you face the back, the connection is on the lower right.
- Connect the other end of the IEC cable to your AC mains. The Zephyr's power input is universal, accepting anything from 100 to 240 Vac 50/60Hz.
- Turn the power switch, located just above the power cable connector, to the on position. (I- bar showing.)
- Your Zephyr will start up. After a few moments the front panel LCD display screen will display the status of your Zephyr. On the lower left, the SYNC indicator should be illuminated. If you have a proper audio connection and are feeding an audio signal to the Zephyr, you will see activity on the two SEND meters on the left- side of the front panel. (With some mono transmit modes, there will be no indication on the Channel B send meter.)

Basic Configuration

Your Zephyr should display a screen that looks like this:

```
init   |Xmt:L3 Dual
       |Rcv:L3 Stereo
init   |56kbps 32kHz
       |ISDN
```

or, in the case of a 3 DSP Zephyr, like this:

```
init   |Xmt:L3 Mono
       |Rcv:L3 Stereo
init   |56kbps 32kHz
       |ISDN
```

If it does not, your unit has been previously configured. If it was, and you want to proceed with the Quick Start, you must erase all of the previous settings. *Before you go to the next step, check with the last person who used this Zephyr to be certain that he or she no longer needs these settings.*

To reset the Zephyr to the factory defaults:

1. Press the <HELP> button, followed by a press of the star <*> button. If prompted to do so press the <#> to confirm this action.

2. It may take a few moments for the cold boot to start. Once started, the LCD display will show the initialization screen and then the screen above will appear. (Pre- Layer II versions, without on- line help, require two presses of the <HELP> button before the <*> .) Note: some versions may require you to push the <#> key to confirm that you wish to erase all settings.

We are now ready to configure your Zephyr. *Follow these instructions* exactly for the most reliable path to success!

We will be using the <UTIL> button. This will take you through the Utility menus where most of your configurations are stored.

1. You now have two options:

   - If you have EuroISDN or another service that does not have SPIDs, press the < UTIL > button seven (7) times and *skip to step 7.*

   - If you have SPIDs, press the < UTIL > button five (5) times and continue.

You should see a screen that looks like this:

```
SPID 1 & 2:
[                        ]
>                        <
    <YES> store
```

2. Using the dial pad on the front right panel, enter the SPID of your first line. (Note that there are no hyphens, punctuation, or blank spaces in SPIDs; they are one continuous number.) The <NO - > button can be used to back up (move the cursor to the left) to delete characters for editing. Then press the <YES +> button to store it.

3. Press the <SEL ▼ > button. Using the dial pad, enter the SPID of your second line. Then press the <YES +> button to store it.

4. You now have two options:

   - If you do not require Directory Numbers, **and most users will not**, press the < UTIL> button two (2) times and *go to step 7*.

   - If you have Directory Numbers, press the < UTIL > button once and proceed.

You should see a screen that looks like this:

```
Directory 1 & 2:
[                        ]
>                        <
    <YES> store
```

5. Using the dial pad on the front right panel, enter the 7- digit Directory Number of your first line. (Note that there are no hyphens, punctuation, or blank spaces in Directory Numbers; they are one continuous number.) The <NO - > button can be used to back up (move the cursor to the left) to delete characters for editing Then press the <YES +> button to store it.

6. Press the <SELECT ▼> button. Using the dial pad, enter the 7 digit Directory Number of your second line. Then press the <YES +> button to store it. Now press the < UTIL > once.

You should see a screen that looks like this:

```
Telco      [Natl I-1]
Panic Dial       NO

    <NO> options
```

7. Press the <NO - > button until the type of ISDN you have appears on the screen. Then press the <YES +> button to accept it. Even if the correct ISDN type is shown, press the <YES +> button to restart the ISDN and initialize the line.

The LCD screen will show the Zephyr initializing and, after about 10 seconds, you should see a screen that looks like this:

```
Ready  |Xmt:L3 Dual
       |Rcv:L3 Stereo
Ready  |56kbps 32kHz
       |ISDN
```

In addition, you should observe the following:

- The SYNC lamp will be glowing red on the left side of your Zephyr's front panel.

- If your Zephyr has an internal NT1, the little green light (at the bottom left, of the back of the unit) is illuminated solid, not blinking.

- If you have an external NT1, there are no error lamps illuminated on it.

8. If these conditions are met, you are ready to place your first call. If not, something is not right with your ISDN line, the way you have configured your Zephyr or both. Since these are Quick Results, the tips below are brief and note the most frequent problems encountered.

   - If you have SPIDs and Directory Numbers, check to see whether you have entered them correctly. If you have not, you can erase all of the settings in the Zephyr by pressing the <HELP> button followed by pressing the star (*) button. You must now repeat the entire configuration process.

   - Check all of your ISDN line connections to be certain they are firm and correct. Refer to the instructions above.

   - Your ISDN line may not be activated or may be provisioned incorrectly. Contact your ISDN line provider. US users should confirm that the line was installed exactly as specified in the ISDN ordering instructions found in the appendix of the manual.

   - If you have a Zephyr with an internal NT1, the green status LED (near the jack, on the rear panel) should be on solid. If it is not, you have a very basic line problem – see section 10 (Advanced Problem Solving) of the manual for more advice and guidance.

   - If you have a manual for your external NT1, see if there is an interpretation of the error messages displayed.

At this point if the conditions in item 12 are still not met, contact Telos Customer Support for assistance. When you call, have the information on your ISDN circuit on hand.

Placing Your First Call (to Yourself)

Your first call is to be placed to yourself. You will be using your first ISDN line to connect to your second ISDN line. If this works, you know you are connected to your local telephone company office and your Zephyr is programmed correctly. (This procedure will not work if you only have ordered only one ISDN B channel from the Telco.)

Follow the step- by- step instructions:

1. Press the <DIAL> button once. You will see a screen that looks like this:

```
Dial: ZEPHYR, Line 1
[                        ]

<NO> opts <YES> last
```

Since you are dialing from the first line, use the keypad to enter the phone number of your second line. If you need a code to access an outside line, be sure to enter it. Since this is a local call you should not need a “1” or area code.

2. Press the <DIAL> button again. You should see a series of messages that say “ConnW, ConnP, 0:00.” If successful, the two red LINE lamps just to the left of the LCD display will be illuminated, the receive LOCK lamp will light, and, if you are sending audio down your Zephyr, you will see activity on both the send and receive meters. If connected to your studio audio gear, you should be able to hear your feed on the receive audio outputs or your headphones. Note that if the headphone volume is too low, refer to manual section 4 (Installation/Basic Operation) on how to turn it up.

3. Whether or not you have successfully connected, slowly press the <DROP> button repeatedly until *both* of the LINE lamps have gone out. This may take up to four presses.

4. If you have successfully connected, go on to the next section. If you have not, try again. Try dialing the other line number provided to you; perhaps the installer reversed them. If you fail after a few more attempts, contact Telos Customer Support.

Your Next Call, to Telos

Located less than a mile from the Rock ‘n’ Roll Hall of Fame, there is always something interesting playing on the Telos ISDN test line. Let’s call Cleveland. If you prefer, you can call our test line in Europe that is listed under “Customer Service” in the front of this manual;. Be certain to call a Layer III line.

Follow the step- by- step instructions:

1. Press the <DIAL> button once. Use the keypad to enter the first Telos test line number. Be certain to precede it with any number you may need to dial to access an outside line and/or to place a long distance or international call. The first number is +1 216.781.9310. Press the <DIAL> button again.

2. You should see a series of messages that say “ConnW, ConnP, 0:00.” If successful, the top (1) red LINE lamp just to the left of the LCD display will be illuminated. You may not hear audio or see any meter action until you have connected the second line. If you do not connect, go to step 5.

3. Press the <DIAL> button once. Use the keypad to enter the second Telos test line number. Again, be certain to precede it with any number you may need to dial to access an outside line and/or to place a long distance or international call. The second number is +1 216.781.9311. Press the <DIAL> button again.

You should see a series of messages that say “ConnW, ConnP, 0:00.” If successful, the bottom (2) red LINE lamp just to the left of the LCD display will be illuminated and the receive LOCK lamp will be illuminated. Unless the CD player at Telos has stopped, you should see activity on the receive meters and hear music.

4. Whether or not you have successfully connected, slowly press the <DROP> button repeatedly until *both* of the LINE lamps have gone out. This may take up to four presses. You want to be sure to disconnect as this may be an expensive call.

5. If you have successfully connected, go on to the next section. If you have not, try again. If you do not succeed, here are a few things to try:

    - US users should try calling using another long- distance carrier. To do so, insert the carrier's five- digit access code. The number to dial starts with any prefix needed for you to get an outside line, followed by the long distance access code, followed by the Telos test line number. See section 4 (Installation/Basic Operation: Solving ISDN Problems) of the manual for additional information. Let us say you don't need a prefix (and most people don't) and you want to use MCI. Dial:

        - **Line 1: 10222 1 216.781.9310**

        - **Line 2: 10222 1 216.781.9311**

    - If this works, you should contact your phone company to determine how to resolve the problem with your primary long distance carrier.

    - Wait an hour and try again. The Telos test line may be busy.

    - Call Telos Customer Support for assistance. The terminal adapter in Zephyr has a built- in ISDN analyzer that will provide plain language error messages in many circumstances. Have these messages ready when you call as they will help speed the process of solving your problem.

What's Next?

Now you are ready to call the world. Remember your promise to read the entire manual! If you want to skip around the manual a bit, the chart that follows can tell you where to find answers to the most frequently asked questions.

Further assistance in using your Zephyr can be found in the HELP menu, which may be accessed at any time. When the <HELP> button is pressed, the LCD display provides information on the item that was previously displayed. When HELP information exceeds the available space on the LCD screen, press the < SELECT ▼ > button to show more information. The < SELECT ▲ > button may be used to review.

To return from the menu in which you were working, press the appropriate menu button. Pressing the <HELP> button a second time serves an "escape" function that returns the LCD to Zephyr's main screen.

The HELP menu may also be accessed from the Zephyr main screen. From this location, it provides an explanation of the display information on the main screen and some basic trouble- shooting tips.

Where to find answers to frequently asked questions

Question	Manual Section
How do I order my ISDN line?	ISDN information and orde forms: Appendix
What are the meanings of all the meters and indicators on the front panel?	Zephyr at a Glance: Section 3
Should I use Layer III, Layer II, or G.722?	Audio Coding Principles: Section 7
What other codecs are compatible?	Codec Compatibility Info: Appendix
How should the remote site device be configured when it is not a Zephyr?	Codec Compatibility Info: Appendix
How can I connect the Zephyr to a computer?	Remote Control: Section 9
How do I contact Telos?	Customer Support: Page 2
Connector Pin- outs?	Zephyr at a Glance: Section
Specifications?	Specifications & Warranty: Section 14

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SECTION 2

INTRODUCTION

THE BEST WAY TO HEAR FROM THERE™

Introducing Zephyr

Zephyr makes transmitting CD- quality audio as easy as sending a fax. Integrating the most advanced audio coding and digital telephone technologies in a single unit, Zephyr is the perfect choice for news, sports, music, and production.

On just one ISDN circuit, Zephyr can transmit two- way, 20kHz stereo audio plus ancillary data virtually anywhere in the world. And it can transmit 20kHz mono audio using only one of an ISDN line's two channels. No other telephone- based system provides Zephyr's full bandwidth with equal quality and economy in both stereo and mono.

To code the audio signals for transmission, Zephyr uses ISO/MPEG Layer III. International listening tests and the satisfaction of thousands of users certify that Layer III is best for ISDN. With the vast majority of program material, the coded/decoded audio cannot be distinguished from the source. Telos Zephyr pioneered the use of this increasingly popular mode.

In addition to Layer III, Zephyr also incorporates the ISO/MPEG Layer II and G.722 coding methods. Coding on the transmit and receive paths can be independently selected from the three offered methods to achieve the compatibility and performance appropriate to your application.

The available ISDN terminal adapter built into Zephyr was designed by Telos specifically for broadcast and professional audio applications. An available data port with V.35 and X.21 network interfaces is also offered to connect to external terminal adapters and CSU/DSUs.

The controls are straightforward for easy operation by non- technical users. Zephyr includes full metering, a headphone jack, mic/line inputs, input protection limiting, and the capability to be remotely controlled via computer. Only the Telos Zephyr combines the power of Layer III with a user interface understandable by non- technical users.

Telos sees ISDN as a tremendously empowering technology for new and creative audio programming and production. We are dedicated to making ISDN an easy- to- use, effective tool for broadcast and audio professionals.

For more than a decade, you have depended on Telos for the innovative, value- priced, and reliable equipment for the broadcast- to- telephone interface. With Zephyr, we once again offer you the best sound, the most practical features, and the industry's finest customer support.

Features and Benefits

- Zephyr is specifically designed for use with one ISDN circuit and other low bit- rate transmission paths. Each Zephyr unit serves as both a transmitter and a receiver. It is the premier product in a complete range of ISDN audio products that includes the ZephyrExpress™ portable unit with built- in mixer.

- Full duplex stereo operation with up to 20kHz audio can be transmitted on one ISDN line (2 B channels) or two Switched 56 lines using Layer III coding.

- Mono 20kHz operation can be accomplished on a single ISDN "B" channel or one Switched 56 line. Zephyr offers full broadcast quality using ISO/MPEG Layer III on one 56/64kbps channel. Other coding schemes force you to sacrifice audio quality or use two channels at twice the transmission cost.

- Telos' advanced digital signal processing (DSP) experience harnesses the power of Layer III efficiently and economically. Our sophisticated hardware platform allows future upgrades to be quickly installed, while our front panel user interface is elegantly simple.

- Zephyr is compatible with codecs (digital audio coder/decoders) that use ISO/MPEG Layers II, III and/or G.722. (See separate compatibility list.) With Zephyr, you are in touch with the world.

- The front panel has a clean, uncluttered design for simple operation with full metering, call- duration timer, headphone jack, and straightforward controls.

- The ISDN terminal adapter and NT1(where required) are built- in. Everything required for the ISDN connection is included, and connection to the telephone network is via a single modular cable. Models are available without the terminal adapter for lower cost when used for non- ISDN applications.

- An available V.35/X.21 port allows Zephyr to be connected to one or two Switched 56 lines, satellite links, and other data paths. You select V.35 or X.21 merely by connecting the appropriate cable.

- The ISDN Telephone feature (G.711) allows Zephyr to dial any standard analog telephone line for low- grade voice communications.

- The split channel mode allows individual mono signals to be transmitted to separate sites. This feature is ideal for bilingual programming as the audio on each channel is completely separate.

- Auto- dial sets include the codec section settings (such as bit rates and transmit and receive coding choices) and the numbers of the remote location you wish to dial. Fifty complete auto- dial sets can be rapidly accessed.

- Bi- directional, RS- 232 serial data at 9600bps for communications and control are transmitted simultaneously with the audio. The communications mode is used to transmit data between the two connected locations, while the control mode is used to operate either a local Zephyr using a computer or a remote unit via a modem.

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

- Four end- to- end parallel "contact- closures" can be used to control recorders and other devices. The first input can be used as a "panic dialer" to automatically connect to a frequently called or emergency number.

- A fifth logic output serves as an in- use indicator when an ISDN line is connected and as an alarm when the ISDN connection is terminated. This closure can also be used at either the time of connection or the time of disconnection to start or stop an audio recorder or other device.

- Zephyr's careful analog design helps to ensure superior sound.

- The selectable input audio limiter prevents your audio from distorting when your talent "screams" and the program signal peaks instantaneously. The limiter is very fast and tight and does not assert itself during times of normal audio levels.

- A comprehensive self- diagnostic routine is performed when the unit is turned on. Problems in the unit, or with the ISDN line, have associated error messages that can be used for troubleshooting.

- Modular design allows you to keep your Zephyr up- to- date with the latest upgrades.

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

- Sophisticated Zephyr Communications and Control Software package, included at no charge, offers two significant features.

    1. When the serial port is in the communications mode, two text windows are displayed for bi- directional communication between your two sites.

    2. In the control mode, the software allows you to program all of the Zephyr settings for your unit. Auto- dial sets can be stored from or written to the Zephyrs. Several auto- dial sets can be saved to disk and downloaded when required.

Overview: Zephyr, the Box

Front Panel

Status information is displayed on both the LED array and the LCD screen:

- Complete metering of send and receive audio levels on both channels is provided.
- Limiter activity indicators further assist in setup.
- Send SYNC and receive LOCK indicators show system status.
- Transmission and reception modes and bit rates and other user- selectable features appear on the multifunction LCD screen.

Headphone jack and front panel dialing speaker allow confidence monitoring.

The right- hand keypads offer easy- to- use, yet comprehensive control:

- Dialing functions include manual dialing and one- button auto- dialing of 50 dialing sets.
- Utility menu controls transmit and receive modes, data rate, headphone volume, and other features.

Rear Panel

Hardware connections to Zephyr are straightforward:

- ISDN terminal adapter is built- in. The ISDN line is simply connected to the modular jack.
- Data port offers V.35 and X.21 when used with appropriate cabling.
- Parallel port is used for end- to- end closures and control.
- Serial ports provide for both unit control and bi- directional, end- to- end data at 9600bps.
- Analog audio outputs offer selectable levels.
- Analog audio inputs may operate at microphone or line levels.
- Optional AES/EBU input/output interface module offers sample rate and several clock options. Analog and digital output are simultaneously active.
- Universal switching power supply automatically configures for line voltage and frequency.

Ordering Guide

- 5 DSP models offer two channels of transmit and receive capabilities for mono, dual- mono, stereo, and split- channel operations. Note: 5 DSP model required to transmit ISO/MPEG Layer II 2- line "Mono- 128" mode.

- 3 DSP models transmit ISO/MPEG Layer II/III on only one channel but receive on two channels. This "mono transmit" unit is an excellent choice for single- channel, duplex operations and also serves as an economical two- channel receiver. A 3 DSP unit can be easily upgraded to stereo in the field by installing a single circuit card. Note: 3 DSP units support G.722 send and receive on both channels.

Part #	Model Description
9202	5 DSP (Stereo transmit/receive) with ISDN terminal adapter only
9201	5 DSP (Stereo transmit/receive) with V.35/X.21 only
9200	5 DSP (Stereo transmit/receive) with ISDN terminal adapter & V.35/X.21
9102	3 DSP (Mono transmit/stereo receive) with ISDN terminal adapter only
9101	3 DSP (Mono transmit/stereo receive) with V.35/X.21 only
9100	3 DSP (Mono transmit/stereo receive) with ISDN terminal adapter & V.35/X.21
9161	AES/EBU digital input/output interface module with DB- 9 and XLR connectors

NOTES:

5 DSP Unit required to transmit ISO/MPEG Layer II 2- line "Mono- 128" mode

AES/EBU modules require factory installation and possible extra cost motherboard upgrade

Contact Telos Systems for details on upgrade kits and cables for the data port.

Zephyr Transmission Modes

Modes can be individually selected for the transmit and receive paths.

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

(Require 5 DSP Zephyr unless otherwise noted)

- Layer III Joint- Stereo at 20kHz or 15kHz for maximum fidelity.

- Layer III Independent Stereo at 15kHz for independent audio and surround- sound transmission.

- Layer III Dual- Mono at 15kHz when each channel has unique audio.

- Layer II Joint- Stereo at 20kHz for compatibility.

- Layer II independent stereo at 7.8 or 9.8kHz to preserve stereo image.

- Layer II mono at 128kbps and 20kHz for compatibility.
- Dual- Channel G.722 at 7kHz for lowest delay and/or compatibility *.

  * Available on 3 DSP models

Mono Modes using one ISDN " B" channel

- Layer III at 15kHz for maximum fidelity.
- Layer II at 7.8kHz or 9.8kHz for compatibility.
- G.722 at 7kHz for lowest delay and/or compatibility.

Split-Channel Modes using two ISDN " B" channels

- Individual mono signals are sent to and received from separate sites. Can be accomplished using Layer III and/or G.722.

HOT TIP!

You will want to refer manual section 7 (Audio Coding Principles) and section 8 (Detailed Menu Reference) in order to better understand the above information.

ISDN Telephone Mode

- G.711 is used to call a standard POTS telephone for low- grade voice communications.

Introducing ISDN

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

ISDN Configurations

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

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

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

Other Digital Telephone Services

When used with an external CSU/DSU, Zephyr works flawlessly with Switched 56 and other digital telephone services. Zephyr, using its V.35/X.21 data port (on models so equipped) and an external CSU/DSU, can also transmit high- quality audio over fractional T- 1, DDS (Digital Data System) or any similar service offering synchronous data connections. Services that provide a full time link between two locations are generally called Dedicated Digital Service. The nature of the local service will determine what Zephyr model and external equipment are required.

Switched 56, as the name implies, has only one channel at 56kbps and is often available in US locations where ISDN service has not yet been implemented. *Since they are both dial- up services an ISDN number can call a Switched 56 number and vice versa.*

For point- to- point audio delivery, ISDN has advantages over satellite. ISDN eliminates the inflexibility of reserved satellite time. ISDN is fully two- way, and startup hardware costs are significantly lower. Overall, ISDN has significant advantages for most occasional and point- to- point feeds and offers economical and reliable backup to your satellite system.

While satellite is still a viable choice for full- time, one- way point- to- multipoint transmission, ISDN BRI and PRI can be a more flexible and economical option for moderate- sized networks. With ISDN cost declining in many locations, larger networks may find it a very appealing alternative to satellite distribution.

ISDN Terminal Adapters

“Terminal Adapter” is the term used to describe the part of an ISDN system which performs the dialing function. It may be a separate “modem- like” box, or may be integrated with other functions.

The Zephyr Terminal Adapter

Zephyr has made connection to ISDN easy by incorporating both a codec and a terminal adapter into a single integrated unit. In the past, you needed to connect external terminal adapters to your codecs. Unlike other manufacturers who buy ISDN terminal adapters to build into their codecs, we developed our own. This enables us to enhance the feature set for broadcast users and create a unified, easy- to- use set of controls for both the codec and terminal adapter functions.

The advantages of Zephyr’s terminal adapter:

- Programming is minimal. Settings, such as bit rate, are selected using a single menu choice for both the codec and terminal adapter sections of Zephyr.

- The hardware and the software are designed for worldwide compatibility.

- When being operated in the US and other countries where the telephone network connection is a “U” interface, Telos delivers the Zephyr internal terminal adapter with the Network Termination unit (NT1) built- in. No external devices are needed for connection to the ISDN line. The “S/T” and “U” interfaces loop, allowing the internal NT1 to be used for other ISDN terminal equipment.

- When operated in countries where regulations require that an external NT1 be installed by the telephone service provider, the Zephyr’s terminal adapter has only the required “S/T” interface.

- The ISDN analyzer function provides easy- to- read error messages when a problem occurs on the ISDN line. Connecting a computer to the serial port allows complete analysis of the ISDN line status for Telos or your telephone service provider to troubleshoot.

Questions and Answers About Audio Coding

Why is coding required to transmit audio over ISDN?

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

How can coding be accomplished?

One might think that lossless, redundancy- reducing methods (such as those used for computer hard- disk compression) would be ideal for audio. Unfortunately, there is not enough redundancy in the audio signal for the significant reduction required by ISDN. Other coding schemes, such as Adaptive Delta Pulse Code Modulation, have limited bandwidth and fidelity, as evidenced by 7kHz codecs that use G.722. To develop coding algorithms with sufficient power to achieve the desired reduction, the audio industry has turned to psychoacoustics. Using carefully researched psychoacoustic principles, coding processes have been designed to reflect the way in which human hearing interprets audio information.

How does perceptual coding work?

With perceptual coding, only information that can be perceived by the ear and the brain is retained. It has been discovered that certain audio creates a “mask” that hides other audio. This represents the brain’s own form of data rate reduction. The masking depends on the frequency, the level, and the spectral distribution of both the masker and the masked sounds. These masks occur in both the frequency and time domains. (See the figures.)

Perceptual coding takes advantage of masking by reducing the resolution of signals that fall below the mask.

Why did Telos select ISO/MPEG Layer III perceptual audio coding as the primary method in the Zephyr?

Layer III is the most sophisticated method for the coding of digital audio. When combined with ISDN, our Layer III codec (coder/decoder) makes it possible to transmit broadcast- quality audio with the convenience, global availability, and low cost of the dial- up telephone network.

The final report of ITU Radiocommunication Task Group 10/2 clearly recommends Layer III for ISDN bit rates, and the daily experience of Layer III users further supports those findings. Of course, the best way to know how good Layer III sounds in Zephyr is to hear it for yourself.

It's agreed that Layer III is the perfect choice for ISDN. How come more manufacturers are not using it and why does Telos include Layer II?

Good questions. Implementation of Layer III requires significant computer power and carefully engineered digital signal processing (DSP). Telos Systems has been the world's leader in DSP for the broadcast- to- telephone interface for over a decade. We know the technology, we know the ups and downs of connecting to telephone networks, and we know our customers. This put Telos in a unique position to launch a sophisticated, multifunction ISDN transceiver that supported the use of Layer III along with other coding options.

We have implemented Layer II to offer compatibility with the installed base of users. Layer II is the most commonly used perceptual coding scheme and products became available before Layer III products and many users are understandably unwilling to "retire" their old equipment.

But Layer II is used more widely for satellite program distribution and is an important contender for Digital Audio Broadcast. Why is this?

Layer II is currently the accepted coding scheme for applications with bit rates of 128kbps per audio channel. There is no logic, however, in concluding that because Layer II is preferred for satellite, it should be used for ISDN. Nor is it true that if you use Layer II in one part of your audio chain it is the best choice for all parts of the system.

Some Layer II-only manufacturers have criticisms of Layer III. Can you address them?

Certainly. There have been interviews in several journals with Layer II- only manufacturers who state that Layer III is too complex and too expensive. When one considers Zephyr, it is obvious that these are unsound arguments. Complexity and cost are design hurdles that Telos has overcome to create products that are reasonably priced and easy to use and incorporate the most appropriate technology. There is nothing that prevents the Layer II- only manufacturers from making the same investment in resources and commitment to their customers. They, for whatever reasons, choose not to do so.

Where does Telos stand on compatibility?

Our industry must avoid proprietary schemes. It is unfair to lock customers into old technology in order to maintain compatibility with their existing equipment, rather than allowing them to take advantage of emerging technologies.

It is essential for codec manufacturers to comply with the coding algorithms that have been carefully crafted. Our coding is “laboratory- reference,” strictly adhering to the algorithms of their developers, Fraunhofer for Layer III and the IRT for Layer II. We have strong working relationships with these laboratories and intend to utilize the best and most powerful coding algorithms regardless of their origins.

Do you see cascading multiple codecs as a problem?

Good question. It appears that, in the near future, most broadcast audio signals will be subject to multiple encode/ decode cycles before they reach the listener. Unfortunately, there is no completed research into the effects of cascading using “real- world” broadcast audio chain configurations. Our view is that one should code only where the bits are unavailable or expensive – and then use the most bits and the most powerful coding method. And, proceed with caution, using your ears as well as all available information. In today’s world coding is everywhere, ask questions, many digital systems use this technology..

About Telos Systems

At Telos, our mission is to give you the tools to exploit the full potential of available analog and digital bandwidth. Wherever telecommunications and broadcasting meet, you’ll find us. With that in mind, we’ve compressed a lot of innovation into the decade since introducing the Telos 10 digital hybrid talk show system, the first broadcast product to use DSP.

We believe the new possibilities offered by digital telephone services and high-performance audio coding are the most exciting things happening in broadcasting technology today. The future is at the doorstep, and our role is to do our best to bring it to you.

As our President says:

> “ The best way to participate in the future is to create it!”
>
> - Steve

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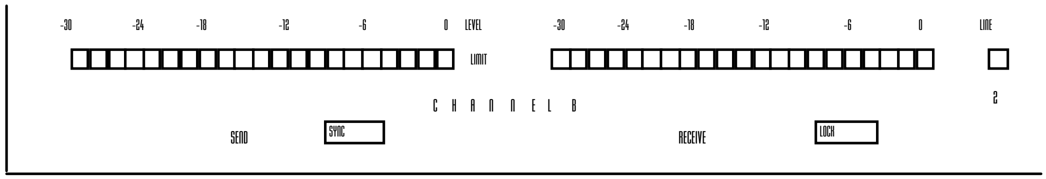
SECTION 3

ZEPHYR AT A GLANCE

FRONT PANEL

Display



Audio Level Meters

The audio level LEDs provide a check on the send and receive audio.

The SEND audio level LED meters aid send level adjustment. Send audio should be adjusted so that the green segments are illuminated, with the red segments flashing occasionally. The 0 dB segment corresponds to the system clip point. However, even this segment may illuminate without audible trouble owing to the efficacy of the internal limiter, if engaged.

The RECEIVE audio level LEDs indicate that audio is being received and decoded. They are calibrated to match the send audio level LEDs at the encoder.

Limit Indicators

The Zephyr has a send audio limiter which serves to keep the usual “digital nasties” from happening when the send program signal grows instantaneously too large. This is a very fast and very “tight” limiter which does not assert itself during times of normal audio level. The idea here is to keep the system out of trouble when, say, a sports announcer, on remote, gets excited.

With a sine tone, the limiter engages at 15 dB above the nominal operating value, but since the limiter activity depends upon the peak- to- average ratio of the program material, it may appear to operate at somewhat less than 15 dB above nominal with some kinds of audio as input.

The LIMIT LEDs should only illuminate occasionally. If they are on consistently, reduce the send level.

Status LEDs

The SYNC LED near the SEND meter indicates “all’s well” with the system. The software must be loaded, up and running, and internal system clocks properly functioning. (This LED may not always be on when you are not using the internal ISDN interface. In this case, it reflects the status of the external interface, as well as the Zephyr’s internal condition. Refer to manual section 6 (Non- ISDN Networks.)

The LOCK LED near the RECEIVE meter illuminates when the receive decoder has locked on to a valid coded signal from a unit who’s transmit mode set to correspond to your receive mode.

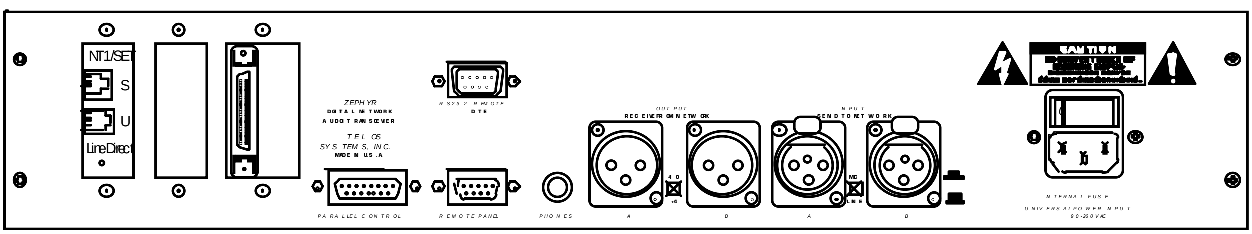
LINE LEDs

Each of these corresponds to an ISDN channel. When illuminated, indicates that the channel is in- use.

LCD and Pushbuttons

The use of the pushbuttons and LCD are described in tutorial fashion in section 4 (Installation & Basic Operation) and in full detail in section 8 (Detailed Menu Reference).

REAR PANEL



Slot Positions

The Zephyr employs a modular architecture, permitting future expansion and flexibility. There are three slots for expansion cards, each of which allow connector access through the rear panel.

These slots accept modules which interface to the transmission path being used.

The optional AES/EBU interface may also occupy one of these slots.

ISDN

Permits connection to ISDN telephone lines. There are two types of interface, one for the USA and another for Europe and other parts of the world. For additional

> IMPORTANT!
>
> As with any piece of electronic gear, it is advisable that precautions be taken to prevent damage caused by power surges. Standard telephone line surge protectors can be used to offer some degree of protection at the U interface jack. It is the user's responsibility to ensure that adequate protection is provided.

information and block diagrams see manual section 4 (Installation & Basic Operation).

The ISDN interface may have an integrated NT1, and in that case it includes both S/T and U connections. The U connection, on the smaller jack, can go directly to a line from a telephone central office. The S/T connection, on the larger 8- pin jack, has two possible functions:

IMPORTANT!

An RJ-11 style 6-position plug can be plugged into the center of an RJ-45 style 8-position jack if your Telco charged you extra and installed an RJ-45 jack for your ISDN line.

- Connect to an external NT1.
- Allow connection to another ISDN device, such as another Zephyr or an ISDN telephone.
- If using the PS- 2 power available on the S interface, the jumper JP- 1 on the ISDN terminal adapter card must be between Pins 1 and 2.

IMPORTANT!

If using an external NT1, or connecting another device, be sure to check power arrangements. The Zephyr provides “PS-2” power on the S/T jack. If your NT1 or other device has an external power supply it is essential that the 2 power leads not be interconnected or *damage to the Zephyr, external equipment, or both, may occur*. Contact Telos Systems customer support for additional information, if needed.

Europe and other

The ISDN interface has only an S/T connection. It must be used with an external NT1, a device usually provided by the telephone authority.

More information can be found in the Installation/Operation and ISDN sections.

This version, identified with the Euro telecom approval mark (looks like 2 hockey sticks in a circle), does not provide power on the S/T interface.

IMPORTANT!

In this case, the NT1 will be powered by the Telco. If your Zephyr’s terminal adapter does not show the Euro telecom approval symbol (looks like 2 hockey sticks in a circle) you should contact Telos Systems customer support before proceeding. This is because the Zephyr’s built in supply and the power supply from the NT1 can conflict and cause damage to the Zephyr, NT1, or both.

ISDN S Interface

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)**

The S interface is a standard modular RJ- 45 style jack and is supplied for special applications and for use outside of North America.

* Not connected on units sold outside of the US and Canada

ISDN U interface

PIN	*FUNCTION*
1	**N/C**
2	**N/C**
3	**Line**
4	**Line**
5	**N/C**
6	**N/C (top pin)**

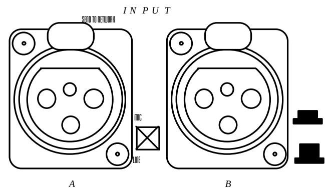
The U interface is a standard modular RJ- 11 style jack and is the usual connection point for North American users. The U interface is not present on units shipped outside of US and Canada.

V.35/X.21

Offers interface to digital bitstreams, primarily for non- ISDN applications. Both V.35 and X.21 equipment can be accommodated with the use of the appropriate cable.

More information can be found in manual section 6 (Non- ISDN Networks.)

Send (Input) Audio



PIN	FUNCTION
1	**Ground**
2	**Audio +**
3	**Audio -**

- Active balanced.
- Switch out = LINE: - 15 to +4 dBu nominal level.
- LINE clip point is +24 dBu.
- Switch in = MIC: - 68 to - 35 dBu level.
- Bridging (approximately 100K•) impedance.

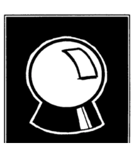
The inputs may be sourced from either line or microphone level signals, depending upon the position of the rear panel switch located near the XLR connectors.

Unbalanced sources may be used by connecting pins 1 & 3 to the source ground while the signal high is connected to pin 2.

Send gain may be adjusted in the <VOL> volume menu. The send level indicated on the LCD is the “nominal” operating level. More information about audio levels can be

HOT TIP!

In MONO transmit modes the audio appearing from the “A” input will be used.

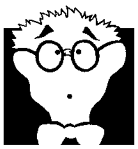


CURIOSITY NOTE!

The Zephyr has the more common pin-outs used for 3 pin XLR inputs/outputs. You can easily remember the correct signals when wiring connectors using the phrase “George Washington Bridge”. Pin 1 = G = Ground, Pin 2 = W = “+” = White (typical color in mic cable, if there is no white there will be a red conductor), and Pin 3 = B = “-” = Black (common color in most audio cables).

found in section 8 (Detailed Menu Reference) and the Appendix.

The limiter engages at 15 dB above the nominal operating value, giving this much "linear" headroom. There is then another 10- 20 dB headroom provided by the limiting action.



DEEP TECH NOTE!

The limiter activity depends upon the peak-to-average ratio of the program material and may appear to operate at less than 15 dB above nominal with some audio as input.

Audio input levels above +4 dBu nominal require an external pad.

It is possible to bypass the send limiter. See manual section 11 (Technical Information) for details on how to do so.

Advantages and reasons for doing so:

- Dynamic range increases by 7dB, this may be desirable, particularly in non-broadcast applications.

- Total harmonic distortion (THD) + noise drops significantly with limiter bypassed, this may be desirable, particularly in non- broadcast applications.

- If another limiter will be used. *Caution: use of non- linear processing such as clipping, multiband compression etc. are not recommended before any perceptual coder.* Contact Telos Systems customer support for a paper delivered at the AES by Frank Foti on this subject.

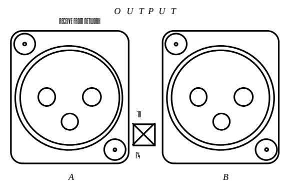
Disadvantages and reasons for not disabling the limiter:

- THD and Dynamic range are quite acceptable with the limiter engaged.

- Loss of send level control; this control is disabled when the limiter is disabled, source must be 4 dBu nominal (see appendix for additional information on levels). External gain stages or pads may be required.

- Tight control of the level before the Zephyr *must* be maintained or digital clipping of the A/D converter could result.

Receive (Output) Audio



PIN	FUNCTION
1	**Ground**
2	**Audio +**
3	**Audio -**

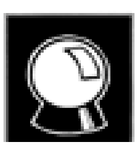
- Active differential.
- Output level: - 10 (switch in) or +4 dBu (switch out), nominal. Rear panel switch selects.
- Impedance: 100• x 2

If a single- ended output is required, connect between ground and either of the output pins. Do not ground the unused pin.

More information about audio levels can be found in the Appendix.

HOT TIP!

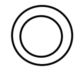
In MONO Receive modes the incoming audio will appear on both “A” and “B” outputs. So you will not need to provide special patching when a MONO feed is sent into your stereo equipment.



CURIOSITY NOTE!

The Zephyr has the more common pin-outs used for 3 pin XLR inputs/outputs. You can easily remember the correct signals when wiring connectors using the phrase “George Washington Bridge”. Pin 1 = G = Ground, Pin 2 = W = “+” = White (typical color in mic cable, if there is no white there will be a red conductor), and Pin 3 = B = “-” = Black (common color in most audio cables).

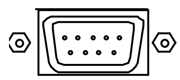
Phones



PHONES

For connection to headphones. Provides a monitor of the received audio signals. The level may be changed in the Volume menu refer to manual section 8 (Detailed Menu Reference). This jack is wired in parallel with the front panel headphone jack.

RS-232



RS232 REMOTE

PIN	*FUNCTION*
2	**Rx (Computer to Zephyr)**
3	**Tx (Zephyr to Computer)**
4	**DTR (Zephyr ready output)**
5	**Ground**

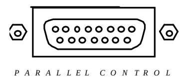
> HOT TIP!
>
> This port is configured for DTE (*Data Terminal Equipment*) as opposed to DCE (*Data Communication Equipment*) operation – meaning that it looks like a computer, not a modem. Therefore, a custom or null-modem cable is required to connect to a computer, while a standard cable is required to interface to a modem. Cables may be easily fabricated or may be obtained from a computer store.

Using a male DB- 9, this is an RS- 232 serial port using the standard PC- style format.

The default line protocol is: 8 bits, 1 Stop Bit, No Parity, 9600 Baud.

A computer plugged into this port can operate the system by remote and can access a number of special diagnostic modes, as described in section 9 (Remote Control). A modem set up for auto- answer allows access to the system from a remote location.

Parallel Port



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

This is a female DB- 15 connector which provides parallel control functions.

INPUTS

All *inputs* are specially treated to accept either a voltage (up to 24 Vdc), or a closure to ground, which may be provided by switches, relays, or logic outputs. The inputs are active low.

See the manual section 8 (Detailed Menu Reference) for details on using the panic dial

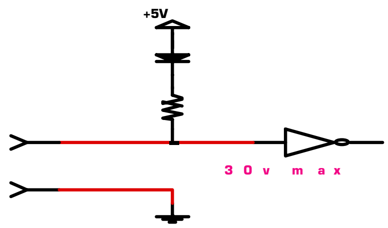
DEEP TECH NOTE!

The Zephyr's "universal" logic input circuit can be used with switch or relay closures, voltage levels up to 24 Vdc, or logic outputs – either "totem-pole" or open-collector.

HOT TIP!

With system firmware versions 2.66 or later inputs 0-3 can each be used as multiple panic dial inputs by setting Panic Dial to "1-4" in the utility menu. Prior to this version there is a single panic dial input using input 0.

feature.



Parallel logic input circuit.

OUTPUTS

All *outputs* are open- collector closures to ground, and are also active low. These will require a pull- up resistor to function with other logic inputs. Some equipment have the pull- ups built into their control inputs – check the device's manual to be sure. If there is no pull- up in the interfaced equipment, you'll have to add one. An appropriate value is 2.2K•.

Current should be limited to 400ma maximum per output with total output restricted to 1 amp (200ma each output if all five will be used).

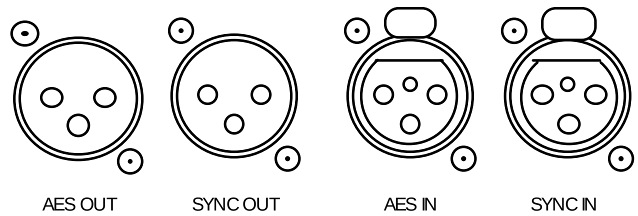
If used with a relay or LED then tie your external power source ground to pin 1 (or use the 5 VDC power supplied on pin 8) and run this power source through your device with a resistor in series to limit maximum current to 400ma.

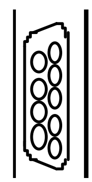
The Status Output goes active low when the condition(s) selected by the Stat Out Utility menu item are satisfied. Options are:

- Rcv lock
- Line 1 active
- Line 2 active
- Lines 1 & 2 active
- Lines 1 or 2 active

For additional information refer to manual section 8 (Detailed Menu Reference).

AES/EBU





AES/EBU DB-9 Pin outs:

PIN	*FUNCTION*
1	**Input +**
2	**Input -**
3	**Input Ground**
4	**SYNC Input +**
5	**SYNC Input -**
6	**SYNC Input Ground**
7	**Output +**
8	**Output -**
9	**Output Ground**

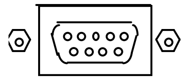
These are optional inputs and outputs for AES/EBU format digital audio signals. The female DB-9 is always present in units with AES/EBU. Optional XLR connections may also be present.

The input is internally sample-rate converted, so may accept sources at any of the usual rates.

The output may be synched and sample-rate converted to either the input AES signal or an independent sync signal presented at the SYNC IN connector. The synchronization is accepted from a standard AES format signal. Only the sync information is stripped; any audio which may be present is ignored. The SYNC OUT is a buffered version of the SYNC IN signal.

More information about the various modes is in section 4 (Installation & Basic Operation) and section 8 (Detailed Menu Reference).

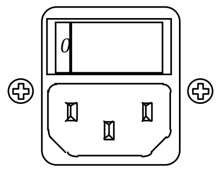
Remote Panel



REMOTE PANEL

Reserved for possible future expansion.

AC Power



INTERNAL FUSE

UNIVERSAL POWER INPUT
90-260VAC

Important Safety Tip: If fuse replacement is required, please note: **For continued protection against fire, replace fuse only with same type and value.** See the DETAILED TECHNICAL INFORMATION section for information and cautions.

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 is the user's responsibility to ensure that adequate protection is provided.

The AC receptacle connects AC to the unit with a standard IEC connector and provides a power on/off switch. The power supply has a universal AC input, accepting a range from 100 to 240 Vac, 50- 60 Hz. A fuse is located inside, on the power supply circuit board.

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SECTION 4

INSTALLATION & BASIC OPERATION

INTRODUCTION

This section offers a guide to the basic operation of the Zephyr in a format which is mostly tutorial in style. As in *Quick Results,* you are led through the steps which result in a successful connection, but additional options are explored, and a more thorough description of common functions and modes is given. We also provide some practical advice on dealing with codec delay and other operational matters.

It is not, however, the ultimate resource. Other sections have more detailed information on many topics:

- *Zephyr at a Glance* (section 3) covers most user- level hardware questions; connector pin- out, levels, and the like.
- While this section covers in detail connecting to the Zephyr's ISDN interface, *ISDN* (section 5) explains what you need to know in order to order the line. The appendix contains order forms to send to your telephone service provider.
- *Non- ISDN Networks* (section 6) is for those who will be using Switched- 56 and other types of non- ISDN data service.
- *Audio Coding Principles* (section 7) helps you decide which method to use for a given application.
- *Detailed Menu Reference* (section 8) has thorough descriptions of each of the menu items and modes. It is the best place to look for operating information and advice, once you understand the basics and are ready to move on. This is the first place to turn to for questions about compatibility and many other advanced issues.
- All of the manual has information which can aid tracking down problems, but *Advanced Problem Solving* (section 10) is specifically dedicated to this cause.

CONNECTING TO ISDN

The Zephyr's internal ISDN interface is installed into the slot closest to the edge. This module may have one or two modular jacks accessible through the rear panel opening, depending upon whether it includes the integral NT1.

In Europe, the Telco always provides the NT1 and the ISDN 'S' interface will be used via the 8 pin RJ- 45 jack, and the ISDN interface is provided only with this connection.

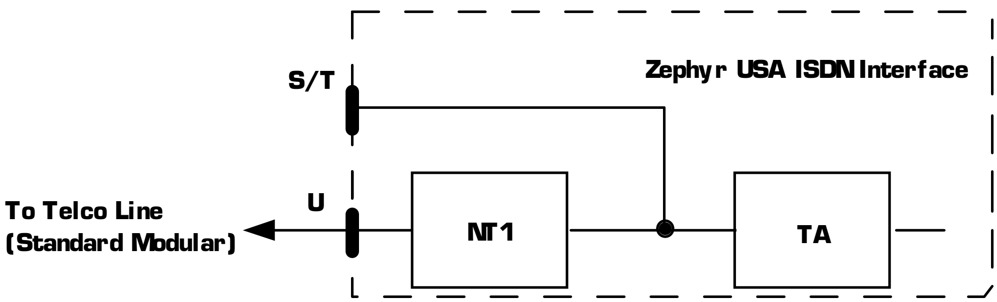
In the USA, a direct connection to the Telco "bare copper" is possible because of the integral NT1.

The U and S/T Interface (USA & Canada)

Connect the ISDN line from the telephone central office directly to the *lower* RJ- 11 style “U interface" modular jack on the rear panel ISDN interface using the standard phone cable provided..

IMPORTANT!

An RJ-11 style 6-position plug can be inserted into the center of an RJ-45 style 8-position jack if the Telco charged you extra and installed a RJ-45 jack for your ISDN line.

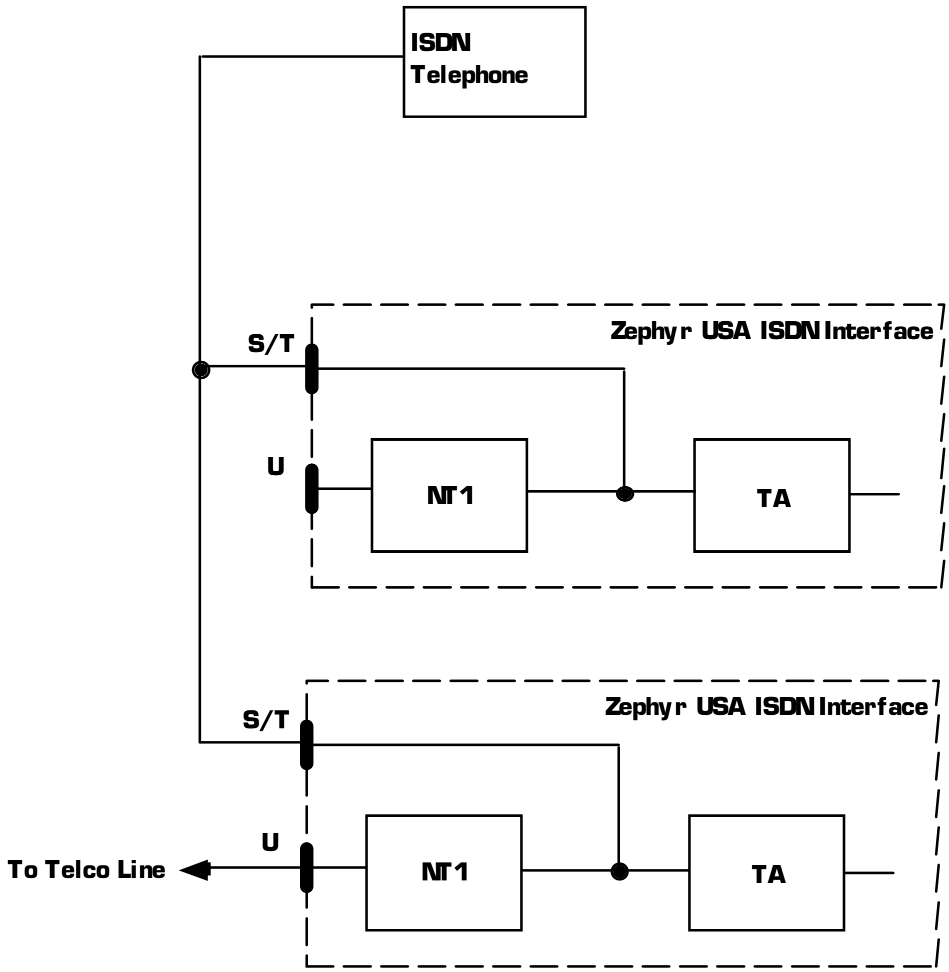


Normal set-up. The U jack connects directly to the Telco line and the S/T jack is unused.

The Zephyr's S/T interface is always active, and may be used to parallel additional ISDN terminal equipment, such as an additional Zephyr or ISDN phone set, to the line. The internal NT1 operates just as if it were an external box in order to permit this to occur (see diagram, below).

U Interface Status LED

The LED indicator of the status of the ISDN connection is on the rear panel near the U jack. If the NT1 is inactive, the LED will remain in the OFF state. Activation in progress is indicated by a rapidly blinking LED (about five times per second). If NT1 can contact the central office, the LED will blink slowly (about once per second). The LED will come on solid when all handshaking is completed and the basic line connection is good.



A set-up taking advantage of the S/T jack. ISDN permits multiple devices to connect to this bus. Here, a second Zephyr and an ISDN phone share the line.

HOT TIP!

We have tested and approved the ISDN telephone sets in the ITT Cortelco family. When configured properly, they may be powered from the Zephyr's internal supply.

IMPORTANT!

If using an external NT1, or connecting another device, be sure to check power arrangements. The Zephyr provides "PS-2" power on the S/T jack. If your NT1 or other device has an external power supply it is essential that the 2 power leads not be interconnected or *damage to the Zephyr, external equipment, or both, may occur*. Contact Telos Systems customer support for additional information, if needed.

It is also possible to use an external NT1, connecting it to the S/T jack. The internal

NT1 will automatically be disabled. The NT1 power may come from the Zephyr, so no additional power supply is needed. Some telephone sets may be powered from the Zephyr.

The S/T cable is has larger plugs than a normal modular phone cable, using 8- pin RJ-45s. Four of the wires are used for the S interface and two are used to convey power from the Zephyr to the NT1.

HOT TIP!

NT1s often have a reset button which may be pressed to re-start the NT1 and re-initialize the Telco connection, in the event of trouble. The Zephyr will sense the reset and will respond by also re-initializing its part of the ISDN connection. However, any connection in progress will be lost when this happens.

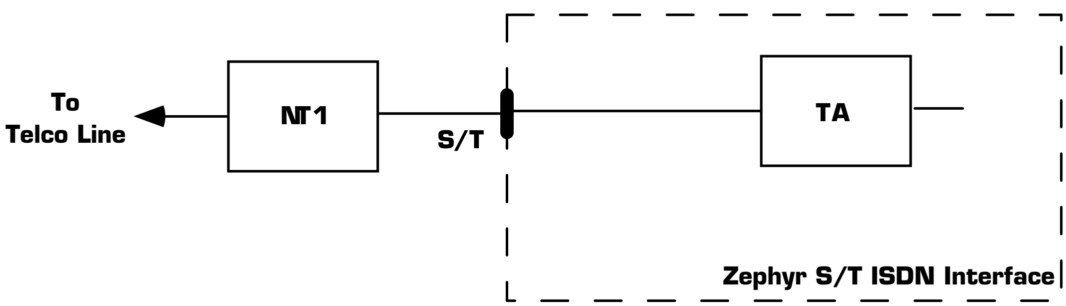
The S/T Only Interface (Europe and Elsewhere)

The Zephyr connects to the NT1 at one of the TERMINAL jacks. Use only an 8-conductor cable.

Unlike the USA version, this ISDN interface does not provide power to the NT1 and it must be provided independently.

IMPORTANT!

In this case, the NT1 will be powered by the Telco. If your Zephyr's terminal adapter does not show the Euro telecom approval symbol (looks like 2 hockey sticks in a circle) you should contact Telos Systems customer support before proceeding. This is because the Zephyr's built in supply and the power supply from the NT1 can conflict and *cause damage to the Zephyr, NT1, or both*.



The usual set-up outside the USA. The S/T jack is connected to the NT1 provided by the telephone authority.

CONNECTING AUDIO

Connect the audio input at the rear panel XLRs and confirm that the level is OK by observing the send meters. If required, adjustment may be made by selecting the Volume menu.

The Zephyr can accept either line or microphone level. There is a rear panel switch located near the input connectors for selection of the level mode.

Connect output cables to the rear panel output XLRs. Outputs may be either - 10 dBm or +4 dBm, selected by a switch near the connectors.

Note that, in these transmit modes:

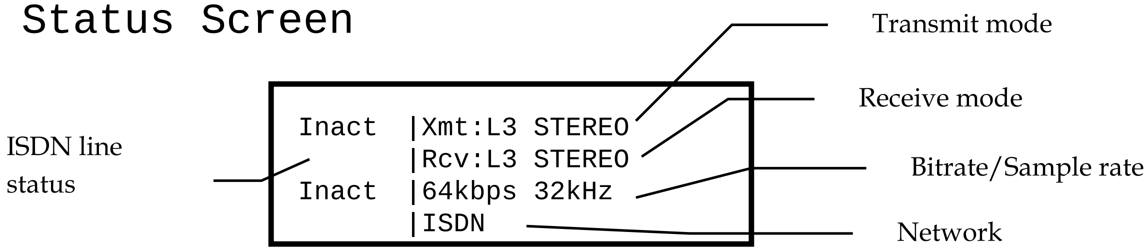
- `L2 MONO`
- `L2 MONO 128`
- `L2 HALF/24`
- `L3 DUAL` (3 DSP “mono” Zephyr versions only)

only the Channel A input is used, and only the corresponding meter indicates anything.

LCD AND PUSHBUTTONS

Here, we offer an introductory, tutorial approach to operating the Zephyr. A detailed description of all menu choices and functions is given in the Detailed Menu Reference section.

Status Screen



This screen appears immediately after power- up and initialization. It is also the system “resting” or “status” screen which appears after some time has passed since any user button press. It offers a summary of the Zephyr’s operating status, and is the Zephyr’s “resting” screen. Presses of the <HELP> button will also return you to this screen.

HOT TIP!

If the LCD has no backlight or has the contrast set so that you can't read it, you should enter the special front-panel diagnostic mode. Press simultaneously and hold the <YES +> and <NO -> buttons while powering-up the unit. You can then use the <SEL> buttons to change the backlight and the <YES +> and <NO -> buttons to adjust the contrast. The Zephyr should then be switched-off and on again to return to normal operation. You can then use the normal menu item for further adjustment (since you can now read it!). If this does not work then the Zephyr's internal RAM settings may have been scrambled. In this case hold the HELP key depressed for 5-10 seconds and then release. The default contrast and brightness values should now be restored.

When in ISDN mode, the words to the left indicate the "Line Status" of the Telco connection. After a call is established, they display the duration in HR:MN format.

The top line shows the currently selected Xmt (Transmit) mode; the second line shows the Rcv (Receive) mode; the third line shows the network bit rate and audio sampling rate; the last line shows the network interface (ISDN or V.35) selected.

Getting Help

For almost every menu item there is help available. It can be accessed by pressing the <HELP> button. The help function is "context- sensitive," meaning that it displays information about the currently selected item.

For instance, pressing <HELP> while the Status screen is displayed displays the following:

```
Line status is
indicated on the
left side of the
LCD...
```

Most help has multiple screens of text. To scroll within them, use the <SEL ^> & <SEL •> keys. The last line of help is always followed by `--` , to let you know that there is no more on this topic.

To return to the Status screen, press the <HELP> button again.

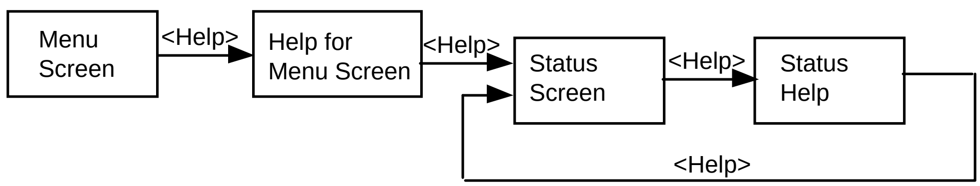
If the help is accessed from a menu item, you return to it by pressing <UTIL> (or <VOL> or <DIAL> or <DROP> or <AUTODIAL> if help was accessed from within one of the latter menus).

To go to the Status screen from a menu item, press <HELP> twice (only once if you are already in a help screen).

Pressing help repeatedly…

- Displays help about the selected menu item
- Displays status screen
- Displays help about the status screen
- Displays status screen etc.

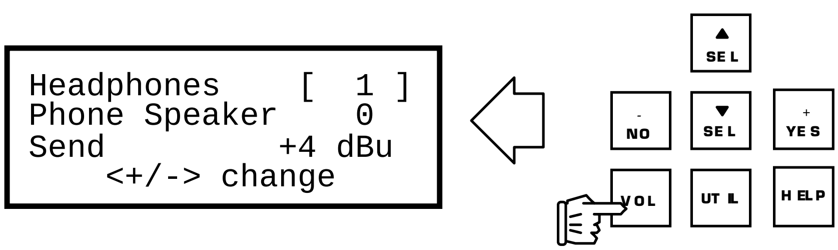
As illustrated below:



Using The Menu System

Menu screens appear on the LCD when either the <VOL>, <UTIL>, <DIAL>, <AUTODIAL>, or <DROP> buttons are pushed.

For instance, the <VOL> button causes the Volume screen to be displayed:



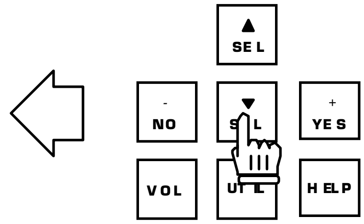
In this menu, you can adjust the headphone volume, the small front panel “phone speaker” volume, and the send audio level.

The settings in this screen are immediately affected by pressing the <NO - > and <YES +> buttons.

When the screen first appears, `Headphones` is selected. This is indicated by the brackets being at either side of the value `1` for headphone level. The <SEL> buttons select the parameter to be changed, and the brackets move to surround the selected value/mode. As you would expect, the movement happens in the direction of the arrow on the button.

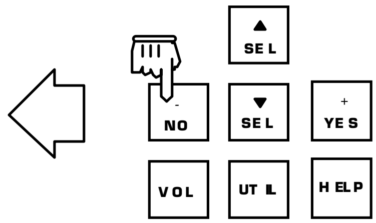
Let’s say we want to adjust the send level. Press the <SEL •> button twice to position the brackets around the send value:

```
Headphones       1
Phone Speaker    0
Send         [ +4 dBu]
    <+/-> change
```



The value is at the default +4 dBu. The <NO- /YES+> buttons are used to change the input gain:

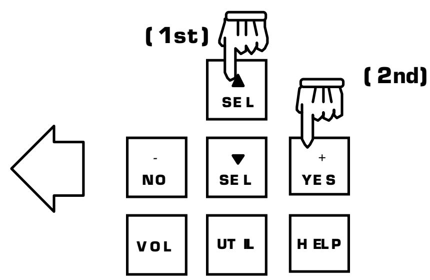
```
Headphones       1
Phone Speaker    0
Send         [ +3 dBu]
    <+/-> change
```



Note the prompt line. It always appears at the bottom of the screen to tell you what is expected.

Now perhaps we want to adjust the headphones volume. First use the <SEL ▲> button to get back to Headphones, and then again the <YES+> and <NO- > buttons to adjust the volume:

```
Headphones    [  2 ]
Phone Speaker    0
Send           +3 dBu
    <+/-> change
```



All of the new values are saved in non- volatile memory so that the settings are preserved even after a power- down.

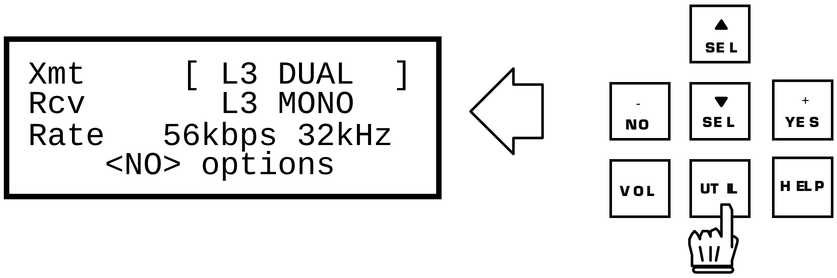
IMPORTANT!

Only the essential menu items will be covered here; all are covered in detail in the section 7 (Detailed Menu Reference).

Introducing The Utility Menu

The Utility menu has more menu items than can fit at once, so it consists of multiple screens, each of which you can get to by either pressing the <UTIL> button multiple times or by scrolling with the <SEL ˆ> and < SEL • > buttons. This menu is for the selection of various operating modes like the coding method, bitrate, ISDN line protocols, and many other things.

The first screen looks like this:

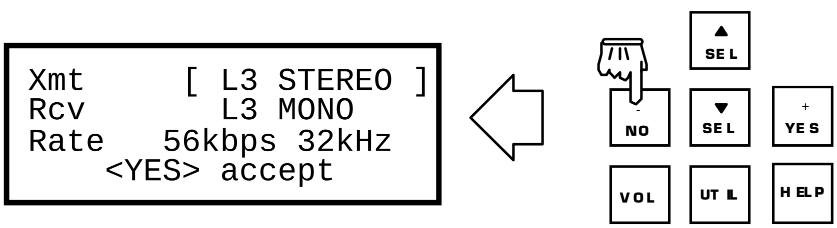


Note that the prompt is different from the one in the Volume screen. Use <NO- > to cycle through the options and <YES+> to confirm and activate your choice. Repeated presses of <NO- > cycle you through all of the available options. But, no action is taken until <YES+> is pressed – we're "just looking."

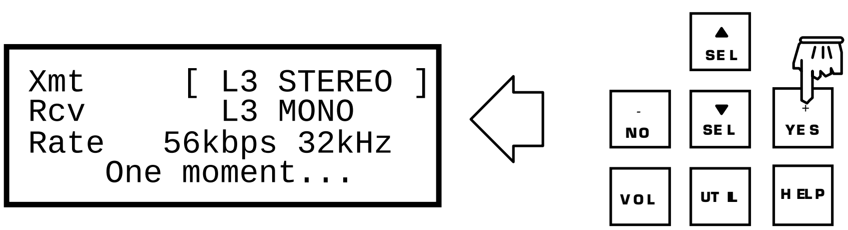
Maybe you think you want to change the mode, but you're not sure what the possibilities are. So, you can press the <NO-> button to see all of the modes for this menu item.

The two- step select- and- then- confirm process also protects you from making changes which could disturb a transmission in progress.

After a <NO- > press, the prompt at the bottom changes to let you know that the displayed mode may now be accepted by pressing the <YES> button:



You decide you want stereo, so the <YES+> button is pressed when `L3 STEREO` mode appears:



Changing the transmit mode takes a little time – about 3 seconds – as the system must re- configure, so you see `One Moment...` on the prompt line at the bottom of the screen, during which the system is not operating and it is not possible to press any other buttons and any audio transmission in progress will be temporarily interrupted.. (Actually, you *can* press the buttons of course, but nothing will happen.)

The Utility menu offers the following items:

Screen 1:	`Xmt, Rcv, Rate`
Screen 2:	`Network, AES In, AES Out`
Screen 3:	`Auto Answer, Loopback, Stat Out`
Screen 4:	`Store Setup, Category`
Screen 5:	`SPIDs`
Screen 6:	`Directory numbers`
Screen 7:	`Telco, Panic Dial, Compatibility Mode`
Screen 8:	`LCD contrast, LCD backlight, Ancil channel`
Screen 9:	`Copyright & version numbers & date`

The first set of items you will need to use are those required to set the Zephyr to conform to the characteristics of your ISDN line...

Setup to the ISDN Line

Network

Confirm that you are set to use the Zephyr's internal ISDN interface. Go to the 2nd Utility screen by pushing <UTIL> twice, where the `Network` menu item should indicate `ISDN` as the mode, as shown below:

```
Network       [ISDN]
AES In  NO (ANALOG)
AES Out  NO CONVERT
    <NO> options
```

If ISDN is not indicated, change to it.

> IMPORTANT!
>
> The other option for this parameter is V.35, for use with an external CSU/DSU or Terminal Adapter.

SPID Number

One or two of these numbers were given to you by the phone company. (If you are using National I- 1 lines, as in this example.) They are entered in the 5th Utility screen. Press the <UTIL> button repeatedly until you see:

```
SPID 1 & 2:
[                  ]
>                  <
    <YES> store
```

> ISDN TIP!
>
> European users should disregard all references to SPIDs. Euro ISDN *does not* have SPIDs! If your ISDN configuration requires MSNs they may be entered in the MSN/SPID 1 & 2 screen.

Enter the first SPID number with the numeric keypad. As soon as you begin to enter a number, the prompt line changes to let you know that you can press <NO> to delete the last character for editing:

```
SPID 1 & 2:
[2                 ]
>                  <
<NO> <- <YES> store
```

When the first SPID is fully entered, press <YES> to store the value, and then move the brackets to the next line, in preparation for entry of the second SPID:

```
SPID 1 & 2:
>21678193100111    <
[                  ]
<NO> <- <YES> store
```

Enter this 2nd SPID, and then press <YES> to store it.

In our example, two SPIDs are required. If you have only one, you press <YES> to store after entering the first, and leave the second blank.

```
SPID 1 & 2:
>21678193100111     <
[21678193110111     ]
<NO> <- <YES> store
```

After both are entered, press <YES> to store them:

```
SPID 1 & 2:
>21678193100111    <
[21678193110111    ]
   Number stored
```

ISDN TIP!

The SPID is not required for PTP and ETS300 Telco modes. More on this in subsequent sections.

ISDN TIP!

Usually the SPID is the area code+phone number+0101. But we have seen some which include a two-digit prefix and a two-digit suffix, and many other variations are possible. See the Appendix for a list of known working SPIDS by Telephone Company.

ISDN TIP!

Despite the last paragraph, you cannot assume anything about the SPID! If the above advice works, consider yourself lucky; if it doesn't work, there is no substitute for getting the correct SPID from your Telco!

Directory Number

Directory Numbers are the 7- digit telephone numbers assigned by the telephone company to this line. There may be one or two. **These will generally not be required**.

Only in the case where at least four digits of the DN are not contained somewhere within the SPID do you have to do anything here. In the vast majority of cases, you can just ignore the Directory screen, and move ahead to the Protocol selection, below.

If the DNs are required, move to the next Utility screen, and enter them using the same method as with the SPIDs. Enter the seven digits of the phone number:

```
Directory 1 & 2:
>2418913           <
[2418914           ]
    <YES> Store
```

Protocol

Finally, we have to tell the Zephyr which protocol the ISDN line is using. You should have gotten this information from the phone company. Press the <UTIL> button until you get to this menu screen:

```
Telco      [  PTP   ]
Panic Dial      NO

    <NO> options
```

The screen indicates the mode PTP. Assume a line protocol of National ISDN- 1 is needed, so you need to change to this value. Press the <NO-> button until `NATL I-1` appears, and then press <YES+> to select it.

> ISDN TIP!
>
> Even if the correct protocol is shown under the Telco setting, you must still tell the Zephyr to reinitialize the ISDN settings by pushing <YES+> . You should also follow this procedure whenever you change your SPIDS.

A restart of the ISDN will be initiated, and the LCD will signal this on the bottom line:

```
Telco      [Natl I-1]
Panic Dial      NO
 Restarting ISDN...
```

After about five seconds, the telephone line will begin to respond and the initialization process is underway. If all goes well, you'll have the following succession of screens, as the line goes through the required start- up phases:

```
inact|Xmt:L3 Stereo
     |Rcv:L3 Mono
inact|56 kbps 32kHz
     |ISDN
```

```
init |Xmt:L3 Stereo
     |Rcv:L3 Mono
init |56 kbps 32kHz
     |ISDN
```

```
wait |Xmt:L3 Stereo
     |Rcv:L3 Mono
init |56 kbps 32kHz
     |ISDN
```

```
Ready|Xmt:L3 Stereo
     |Rcv:L3 Mono
wait |56 kbps 32kHz
     |ISDN
```

```
Ready|Xmt:L3 Stereo
     |Rcv:L3 Mono
Ready|56 kbps 32kHz
     |ISDN
```

These should appear in sequence within about 10 seconds. If you do not get to `Ready`, Ready something is wrong, either with the line or the Zephyr setup.

> ISDN TIP!
>
> Recall that the ‘AT&T PTP’ Telco mode does not require SPIDs, so the line status should go immediately to ‘Ready’ in this case. Refer to manual section 5 (ISDN) for additional information on ISDN protocols.

> ISDN TIP!
>
> If a wrong SPID is attempted – a common problem – the status word indicates this by staying at ‘wait.’.

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?) they may have given you a 2 B-channel AT&T "Custom Multipoint" line with 2 phone numbers and 2 SPIDS. Zephyr does not support this configuration. However, in an emergency, to save a remote, you can usually get things working (temporarily) as follows:

1) Remove both SPIDS and both directory numbers

2) Set Telco to PTP

3) Have another Zephyr call either of your phone numbers twice. They should be able to connect.

4) The line will now function normally for outgoing calls until Zephyr is rebooted or the ISDN connection is reinitialized.

See the subsection Solving ISDN Problems a few pages forward in this section for more

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?) they may have given you a 2 B-channel National ISDN line with only one SPID and one phone number. This may work, however you will need to proceed as follows:

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

troubleshooting advice.

Making A Call

With all the setup preliminaries now behind us, it's time for gratification! Press the <DIAL> button to get underway with a connection:

ISDN TIP!

When you get "Ready Ready", your first call should be to yourself. Dial from the first line to the second, and make sure that you can connect in this most basic condition. This only works if you have ordered both B channels (2B+D) from your Telco, of course.

```
DIAL: ZEPHYR, Line 1
[                  ]

    <NO> options
```

The first LCD line indicates that you'll be dialing in "ZEPHYR" mode on ISDN line 1. (Zephyr mode means that we want a hi- fi call to another codec, rather than one to a regular phone.) These are the defaults. If we wanted to change either the mode or line channel, pressing the <- NO> button would allow a change of mode.

When you begin to enter the number, the LCD prompt line changes to let you know that you can backspace/delete with the <-NO> button for editing:

```
DIAL: ZEPHYR, Line 1
[12                ]

<NO> <- <DIAL> dial
```

You should dial the number just as you would a normal telephone number, with a 1+Area Code for long distance, etc:

```
DIAL: ZEPHYR, Line 1
[12167819310       ]

<NO> <- <DIAL> dial
```

We have dialed 1 216.781.9310, the first number for the Telos Zephyr test line. This line is always available for you to use to confirm that your system is working. (The number for the second test line channel is 1 216.781.9311, which we may optionally call later.)

With the number fully entered, press the <DIAL> button to go, and if all is well, the status will change as the call connection progresses:

```
ConnW|Xmt:L3 Stereo
     |Rcv:L3 Mono
Ready|56 kbps 32kHz
     |ISDN
```

Zephyr is waiting for a line to proceed

```
ConnP|Xmt:L3 Stereo
     |Rcv:L3 Mono
Ready|56 kbps 32kHz
     |ISDN
```

Call is proceeding. If there is a problem Zephyr could stay at this screen for a minute or so. You can abort the call by pressing the <DROP> key.

```
0:00 |Xmt:L3 Stereo
     |Rcv:L3 Mono
Ready|56 kbps 32kHz
     |ISDN
```

The call is now connected and Zephyr is displaying the call duration timer.

This is what we've been waiting for!

You will hear an audible 'bleep,' and the Line 1 LED will come on, indicating the successful connection. (The Volume/Phone Speaker menu item can turn the 'bleep' down, so it is possible you may not hear it.)

Assuming L3 MONO receive mode @ 32 KHz sample rate, with the first line connected, we're ready for audio. If the unit at the other end is correctly transmitting, the LOCK LED will illuminate and the receive meters will show audio.

If we need two audio channels (L3 STEREO mode @ 32 KHz), then the second line will need to be dialed and connected. Press the <DIAL> button again and notice that the Zephyr automatically knows that you now want the 2nd line:

```
DIAL: ZEPHYR, Line 2
[                  ]

    <NO> recall
```

As before, when the number is fully entered, press <DIAL> to go, and the status will cycle:

```
0:01 |Xmt:L3 Stereo
     |Rcv:L3 Mono
ConnW|56 kbps 32kHz
     |ISDN
```

```
0:01 |Xmt:L3 Stereo
     |Rcv:L3 Mono
ConnP|56 kbps 32kHz
     |ISDN
```

```
0:01 |Xmt:L3 Stereo
     |Rcv:L3 Mono
0:00 |56 kbps 32kHz
     |ISDN
```

Status Screen During A Call

When the connection is made, the status screen will appear, and a timer will begin for each line to show the call duration.

Then, after a call is connected, the status word becomes a call timer.

`XX:XX:  Time in HR:MN format.`

The colon blinks every second to indicate that the unit is operating, as a “peace- of-mind” indicator.

```
0:02|Xmt:L3 Stereo
    |Rcv:L3 Mono
0:01|56 kbps 32kHz
    |ISDN
```

Solving ISDN Problems

If you don’t get the `Ready` indications, the SPIDs are the most probable problem. We have had a number of experiences where Telco people have given incorrect SPIDs to users. For additional information refer to manual section 10 (Advanced Troubleshooting), section 5 (ISDN) and section 4 (installation & Basic Operation)

To aid troubleshooting, here is the meaning of each step:

`inact:`	No line connected, or connected to wrong jack on Zephyr
`init:`	Before sending SPID to ISDN network, or after seeing external NT1
`wait:`	SPID has been sent to network
`Ready:`	SPID has been accepted and everything is OK
`ConnW:`	Outgoing call in progress; waiting for line.
`ConnP:`	Outgoing call in progress; call proceeding
`ConnR:`	Outgoing call ringing
`0:00`	Call connected
`Disc:`	Call disconnected

Your second call could be to our test line. It is set for Layer 3 DUAL/MONO transmit. It may be called by a mono or a stereo unit, at either 56 or 64kbps set to L3 Mono Rcv (only one call required) or L3 Stereo Rcv (both calls must be made) at 32kHz sample rate.

The numbers are:

+1 216 781.9310

+1 216 781.9311

If you are able to call locally, but are unable to connect a long- distance call, you may want to try another long distance carrier. Just as with voice lines, you may choose a carrier on a per- call basis by prefixing the number with the 10XXX carrier selection code.

Some carriers and codes that we've tried are:

- ❑ AT&T 10288
- ❑ MCI 10222
- ❑ Sprint 10333

You must dial the full number, including the 1 or 011 + country code following the prefix.

If none of the foregoing helps, see section 5 (ISDN), section 8 (Detailed Menu Reference), and section 10 (Advanced Problem Solving).

Dropping The Connection

> IMPORTANT!
>
> Be sure both lines are dropped, indicated by both Line lights being extinguished. Phone charges can add up quickly on long distance calls!

To release the connection, press the <DROP> button.

```
Drop: [ BOTH LINES ]

   <NO> or <DROP>
```

Zephyr tries to guess whether you want to drop one line or both, depending upon the Xmt and Rcv modes. In this case, the modes are Stereo, and `BOTH LINES` has been selected for you. If this is OK, just press the <DROP> button again to go ahead and do it.

If you wanted to be selective, you could have chosen `LINE 1` or `LINE 2` for the Drop parameter before pressing the <DROP> button the by pressing the <NO ->botton.

Using Auto Dial

Permits recalling stored set- ups and numbers and then connecting according to the stored parameters and numbers. The autodial function will recall codec modes as well as telephone numbers. For instance, you may have someone you connect to in G.722 mono at 56kbps, while others need Layer III stereo at 64kbps.

The third LCD line follows the DISPLAY parameter – you may look at the NAME text, either of the two stored telephone NUMbers, or the system MODE to ensure that you have what you want.

The autodialer will dial one or both lines depending upon whether one or two numbers were stored initially.

After a valid selection is made, press <AUTODIAL> again to connect.

Press the <AUTO DIAL> button:

```
Auto Dial:   [  1 ]
Display       NAME
Telos Test Line
<+/-> or <AUTODIAL>
```

There are 50 possible auto dial set- ups. This example screen indicates that the Telos test line is programmed into the #1 set- up. With the brackets around the Auto Dial number value, press the <-NO> or <YES> button to see other set- ups:

```
Auto Dial:   [  2 ]
Display       NAME
Mr Voice-over
<+/-> or <AUTODIAL>
```

You can also choose an setup using the keypad to type it's 2 digit number. For instance press <0, 1> to see another set- up:

```
Auto Dial:   [  1 ]
Display       NAME
Telos Test Line
<+/-> or <AUTODIAL>
```

The Display parameter on the screen above selects that we see the Name of the set- up on the LCD third line. To see the programmed telephone number for this set- up, the Display value can be changed by moving the brackets Display by using the <SEL> buttons and pressing <YES +> to select `NUM1`:

```
Auto Dial:      1
Display      [NUM1]
12167819310
<+/-> or <AUTODIAL>
```

It’s possible to have two numbers stored (for a two- channel call), and you can see the second by choosing `NUM2` by pressing<YES +> again:

```
Auto Dial:      1
Display      [NUM2]
12167819311
<+/-> or <AUTODIAL>
```

The final value for Display shows the stored mode for this set- up:

```
Auto Dial:      1
Display      [MODE]
T=L3Du R=L3St 56/32
<+/-> or <AUTODIAL>
```

Press the <AUTO DIAL> button a second time to make the connection.

In the single channel case, the screen will change to:

```
ConnP|Xmt:L3 Dual
     |Rcv:L3 Stereo
Ready|56 kbps 32kHz
     |ISDN
```

```
0:00 |Xmt:L3 Dual
     |Rcv:L3 Stereo
Ready|56 kbps 32kHz
     |ISDN
```

And in the two channel case, it will change to:

```
ConnP|Xmt:L3 Dual
     |Rcv:L3 Stereo
ConnP|56 kbps 32kHz
     |ISDN
```

```
0:00 |Xmt:L3 Dual
     |Rcv:L3 Stereo
0:00 |56 kbps 32kHz
     |ISDN
```

Ultimately, the resting status screen will come on, as always, after successful dialing:

```
00:01|Xmt:L3 Dual
     |Rcv:L3 Stereo
00:01|56kbps 32kHz
     |ISDN
```

Note that with a unit properly pre- programmed, it is possible to get a Zephyr on the air with just two presses of the <AUTO DIAL> button; one to get the menu, and the second to go.

Storing Setups

Here we offer only an introduction to storing setups. Step- by- step instructions are included in section 8 (Detailed Menu Reference).

```
Store setup   [  1 ]
Category       NAME
>                  <
    <+/-> change
```

This screen, part of the UTILITY menu, allows you to store setups for later use by the autodial feature. Stored setups include a name for reference, one or two telephone numbers, and all necessary operating mode values.

HOT TIP!

A good idea is to test the setup parameters by actually dialing and confirming that everything works before doing the store setup operation.

Only the setup name, and telephone numbers are required to be entered – all of the mode information (Xmt, Rcv, and rates) is taken from the system state at the time the name text is entered. *So, make certain that the values you desire are in the system before storing from this screen.*

When the brackets surround the `Category` menu item mode, the <+/-> keys select which of the three values is to be entered. The information to be entered is done so on the third line. Therefore, you will be selecting each `Category` and then using the `<SEL>` button to get to the third line, you will enter the information and store it, and then will to back to `Category` and select the next `Category`. When you are happy with each

> HOT TIP!
>
> If you are changing an old setup, and you don't erase a left-over and undesired second number, the system will try to dial it. You must erase any such number you don't need by backspacing to clear it out.

entry, press <YES> to store everything.

The dialing keypad is used in a clever (albeit a bit painful – we only have a few keys to work with here!) way to enter the alphanumeric name text. Each press cycles through all of the characters printed on the button cap. When the desired character is displayed, press the <YES+> button to accept the character and advance to the next position. Example: The '2' button is also used to get the characters 'A' 'B' and 'C'. The '1' button can be used to get 'Q' 'Z' or a space. As usual, <NO- > backs up and deletes for editing. Then return to `Category` to change to `NUM1` or `NUM2` to enter the numbers. The '0' button has special characters; '*' and '#' can also be used.

> HOT TIP!
>
> Some ISDN lines are configured to have the same telephone number for both channels. If you simply put the same number into both setup number fields, it probably won't work. (This is because the Central Office thinks that the Zephyr has erroneously sent the same setup twice.).
>
> However, we have discovered a work-around that seems to get the job done. When you are storing a setup for such a destination, enter the number in both NUM1 and NUM2, but put the symbol "#" after one of them. The Central Office will discard the extra digit and then properly connect both channels.

One or two telephone numbers may be stored. If there is only one, only a single ISDN call is made; entry of two numbers causes both lines to be dialed.

Calls To A Regular Phone (G.711 calls)

One of the great features of the Zephyr with its internal ISDN interface is that you can make voice calls to regular telephones, not just to other codecs. In fact, you can make a mono Zephyr hi- fi call on one channel and a voice call on the other simultaneously.

Press the <DIAL> button:

```
Dial: ZEPHYR, Line 1
[                  ]

    <NO> options
```

Unlike last time, you will now press the <NO> button to ask for the dial options, and a second screen will appear to allow you to choose the call type and ISDN line:

```
Dial options:
Call Type   [ZEPHYR]
ISDN Line      1
    <NO> options
```

In our example, we do not have a coded, hi- fi connection yet going, and we just want to make a call before doing so – maybe to let someone know that a Zephyr connection is on the way. This means that we can use ISDN line 1, the default. But we want to change the call type to `PHONE:`

```
Dial options:
Call Type   [PHONE ]
ISDN Line      1
    <NO> options
```

As usual, we have pressed <NO -> to see the options , and then press <YES +> when `PHONE` is displayed. This bring us back to the main Dial screen. Enter the number to call, and the <DIAL> button to proceed:

```
DIAL: PHONE, Line 1
[15551212         ]

<NO> <- <DIAL> dial
```

The call is made through the ISDN line and the telephone network handles the translation to analog at the other end. Mind you, the Zephyr has no analog line connection!

There is no handset connection, so you must connect a microphone to the analog audio input in order to talk. The usual analog audio output, headphones, or the Zephyr's front

> DEEP TECH NOTE!
>
> The American and European coding method for regular calls is slightly different – the USA uses µLaw while Europe uses Alaw. The appropriate mode is selected by the Zephyr, following the Telco ISDN mode. PTP and NI-1 select µLaw, while ETS300 selects A-Law. Note that there is no problem calling between areas with different methods, as the Telco performs an automatic translation.

panel speaker may be used for listening. (The front panel speaker is located behind the keypad, and its gain is controlled in the Volume menu.)

SETTING AUDIO CODING MODES

Transmit (Xmt) and Receive (Rcv) Modes

This menu is used to set the basic operating parameters.

The transmit and receive modes select the coding method you want to use for the audio transmission. Determining the most appropriate for a given application is covered later, but one critical thing to remember is that the modes at the two ends must be compatible. The transmit mode at this end must match the receive mode at the other end, and vice- versa.

The values given in the example will work with the Telos test line – which is likely to be one of your first calls:

IMPORTANT!

Note that the Rcv modes are a subset of the Xmt modes; in other words, a given Rcv mode will generally be applicable to more than one Xmt mode. For instance, the L3 STEREO Rcv mode is can decode audio from the following Xmt modes: L3 JSTEREO, L3 STEREO, and L3 DUAL.

IMPORTANT!

If you select L3 DUAL in the UTILITY menu and the resting screen then shows L3 MONO, this indicates you have a 3 DSP unit. The same is true if a selected layer 3 mode gives a HARDWARE NOT AVAILABLE message. For information on Zephyr configurations see manual section 2 (Introduction).

```
Xmt       [  L3 DUAL  ]
Rcv          L3 MONO
Rate    56kbps 32kHz
    <NO> options
```

To change any menu item, use the <SEL> keys to select it, the <NO> key to cycle through the modes, and the <YES> key to activate your choice.

Transmit and receive mode changes take about 2 seconds to take effect, as indicated by `One Moment...` on the bottom LCD line.

For information on choosing appropriate coding modes refer to manual sections 8 (Detailed Menu Reference), and 7 (Audio Coding Principles).

Bitrate and Sampling Rate

The first value in the `Rate` menu item mode option is the network bitrate.

It is preferable (for maximum audio quality) to use 64kbps when possible, but it is not always possible owing to the limitations of some Telco digital connection paths. 56kbps works more universally, so it is the value we'll use here:

```
Xmt       L3 STEREO
Rcv       L3 MONO
Rate  [56kbps 32kHz]
    <NO> options
```

The second value, here 32kHz, is the audio sampling rate. It must be set to match the value of the unit at the other end. More information about choosing the appropriate sampling rate is in section 7 (Audio Coding Principles).

> COMPATIBILITY TIP!
>
> On the Zephyr, we refer to the per-line bitrate. In the stereo modes, the total bitrate is two times this number. This can be confusing because some other codecs use the total, rather than the per-line value.

> ISDN TIP!
>
> With the internal ISDN interface, the rate value is used both to set the Zephyr coder/decoder parameters and to tell the ISDN network what rate to use during call setup.
>
> Rate adaptation for the Bit Rate is automatically used on the ISDN channel. Further, when a call is *answered* at the Zephyr, the bit rate value is updated to properly reflect the ISDN network condition.

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 8 (Detailed Menu Reference).

Receive, if used, must be set to the G.722 mode. The Layer II/III decoders are unable to accommodate the independent signals.

> IMPORTANT!
>
> You must make the call to the telephone number for Line 1, as the mono L II/III decoder will only work with a signal coming from the first line.

Zephyr Xmt-To-Rcv Mode Compatibility Table

This table shows which Zephyr Xmt and Rcv 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 L3 STEREO	L3 STEREO	15/20kHz*	Audio output channels correspond to input.
L2 MONO (Channel A)	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	8.6 kHz	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.

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 talent does not hear himself via the studio cue return. Rather, his microphone is mixed locally with a studio feed which has everything *but* the remote audio – thus the "mix- minus" designation. The announcer gets in his headphones a non- delayed version of himself and a slightly delayed version of all of the studio pieces.

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. Problem. 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 could increase the amount when these remotes are in progress.

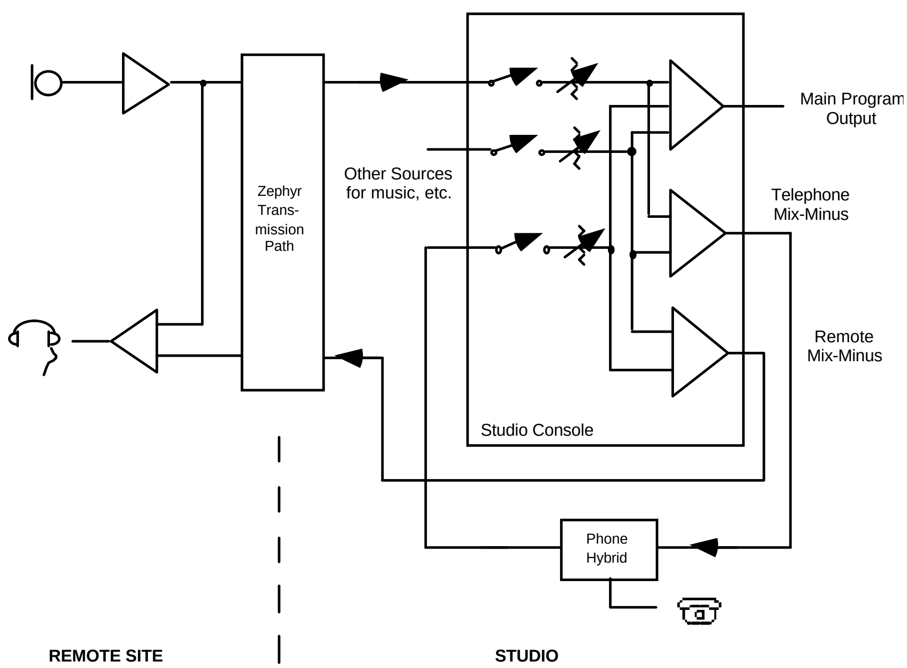


Diagram showing system set-up for remotes with delay in the transmission *path and phones taken at the studio. Note that this is the same as required for satellite links.*

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 which sacrifices fidelity for delay (after all, the callers need not be in high fidelity). This round trip delay issue will also effect your choice of remote- to- studio coding. If the show will be 90% talk with just a small amount of music then 200 msec of delay can be saved by using L3 DUAL Xmt/ L3 MONO Rcv for this path rather than using the slower L3 STEREO mode. Another trick is to use a POTS call (either by Zephyr or using a phone coupler) for the studio- to- remote link which will make the delay in that direction very small. Other intermediate tradeoffs are possible and will be dependent on your format. Talent's thinking time can be significantly reduced by drinking a strong cup of coffee!

For information on the tradeoff between audio quality and delay refer to manual section 7 (Audio Coding Principles).

AES/EBU

This is offered as an option, so it may not be present in your unit. The Zephyr's AES/EBU capability was designed for flexibility. Sample rate conversion is possible on both input and output. An input is provided for an external sync signal.

Two LCD menu items are used to specify operational modes.

The choices for input mode are:

NO (ANALOG)

Use analog input.

S/R CONVERT

Use AES/EBU input and sample rate convert to selected sampling rate (32 or 48kHz, 16 kHz in case of G.722).

SYNC TO NET

Use AES/EBU input but do not sample rate convert. Input signal must be synchronous to the clock on the AES/EBU- OUT (which is locked to the ISDN network clock).

Output mode options are:

NO CONVERT

Do not sample rate convert. Uses sampling rate of received signal.

32 kHz/44.1 kHz/48 kHz

Sample rate convert to the specified frequency.

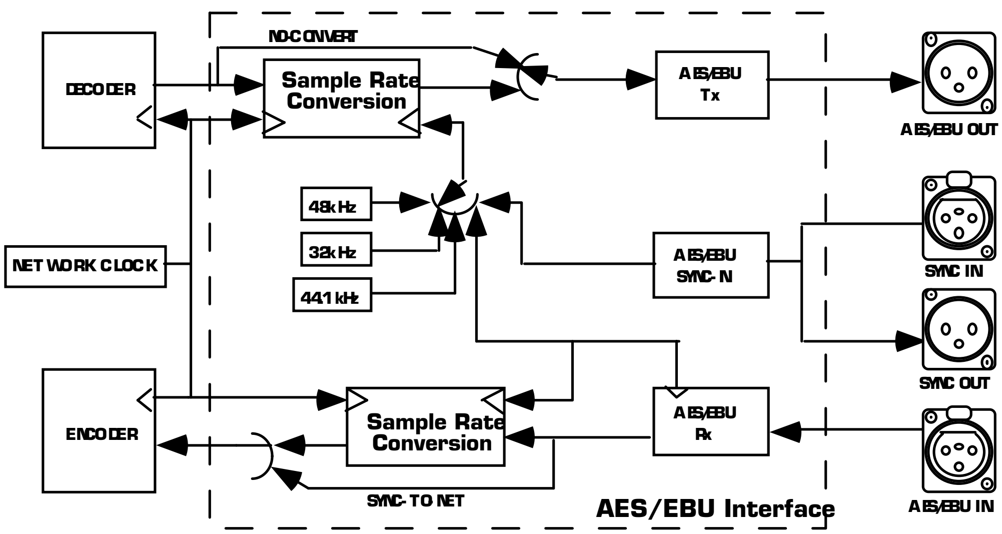
EXTERNAL

Sample rate convert to the sampling rate provided at the SYNC- IN connector of the AES/EBU card.

<u>AES IN</u>

Sample rate convert to the sampling rate retrieved from the AES/EBU input signal.

The output may be synched and sample- rate converted to either the input AES signal or an independent sync signal presented at the SYNC IN connector. The synchronization is accepted from a standard AES format signal. Only the sync information is stripped; any audio which may be present is ignored. The SYNC OUT is a buffered version of the SYNC IN signal.



Signal flow diagram for AES/EBU interface. Sample rate conversion on both input and output permits maximum flexibility.

SECTION 5

ISDN

ISDN BASICS

Background

It is the introduction of digital transmission services from telephone providers which has made the Zephyr possible.

The telephone infrastructure is moving from analog to digital. Telephony made the first significant use of digital audio techniques: 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. While they were intended originally for simple single-channel analog, engineers discovered that the common copper wire pairs 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 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 odd that we have been using modems to convert digital information to analog beeps just to accommodate the mile or two of ancient analog linkage at each end of a thousand- mile long connection.

ISDN is the technology which has evolved to eliminate this analog bottleneck yet still utilize existing copper infrastructure.

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.

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.

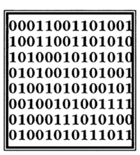
CURIOSITY NOTE!

European Users should disregard all references to SPIDs. Euro ISDN does *not* use SPIDs. In your ISDN configuration the required MSNs may be entered in the MSN/SPIDs 1 & 2 screen.

ISDN TIP!

While the D channel is always present for signaling, 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 and is not used by Zephyr.

From the perspective of telephone network routing, each B channel appears to be a separate line with it’s 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 Zephyr menus and LEDs.



ISDN TIP!

In some cases your ISDN line, while having 2 B channels, will only have only one phone number. This is the case of the AT&T Custom PTP protocol. This does not usually present a problem. This works as if the lines are assigned to a short “hunt group” i.e.; the first incoming call will be assigned to Line 1, and the second incoming call will be assigned to Line 2. Of course you have the option of which line to use on outgoing calls.

DEEP TECH NOTE!

The actual 2 B channels of the BRI are assigned on a per call basis. Therefore, from a theoretically correct viewpoint, Lines 1 and 2 do not fully correspond to B1 and B2.

SPIDS

Service Profile IDentification numbers (SPIDS) are only required with the Zephyr when you are using the National I- 1 or DMS Customer functional ISDN protocols in the USA. This number is given to the user by the phone company and must be entered into the Zephyr 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 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. You should get one SPID for each B channel you ordered.

Upon power- up, connection of the ISDN line, or reboot, the Zephyr and the Telco equipment go through an initialization/identification routine. The Zephyr sends the SPID(s) and, if it is correct, this fact is signaled by the network. Thereafter the SPID is not sent again to the switch.

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**, please

get the spids!

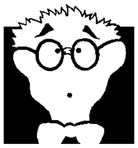
You must have this number, and it must be 100% correct, or the system won't work. Don't let the installer depart without leaving them. You've been warned!

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 is only temporary and the line will fail to initialize next time the Zephyr is booted

ISDN TIP!

If you see WAIT as your line status your SPID is incorrect. In this case check the SPID. For a list of known working SPIDs by telephone company see the Appendix.



DEEP TECH NOTE!

The intention of the SPID is to allow the Telco equipment to automatically support different terminal equipment feature requirements by sensing different SPIDs from each user terminal sharing a BRI circuit. For instance, multibutton phones could retain function assignments when moving from line to line. In this case, the line number would probably not be used as the SPID. Or, to allow a variety of different types of equipment, with different service requirements, to share one ISDN line. None of this matters with our application, but we must enter the SPIDs nevertheless.

CURIOSITY NOTE!

There is hope on the horizon that SPID difficulties will become a thing of the past. For one thing, the Telcos are beginning to standardize on area code+phone number + 0101 for SPIDs on National ISDN lines.

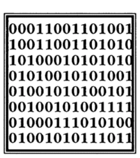
Standards for future versions of National ISDN will include automatic SPID assignment/selection and non-initializing terminals which would be allowed to operate without a SPID, albeit with only basic functionality.

Telos Systems is monitoring these developments carefully.

Directory Numbers (DNs)

Directory Numbers, or DNs for short, are the 7 digit telephone numbers assigned to the ISDN line (as would typically found in the 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 be difficulties related to receiving calls.



ISDN TIP!

However, it is not usually necessary to explicitly enter them, as they are almost always contained within the SPID, and the Zephyr is smart enough to look there first. The only time it is required to enter the DN is in the very rare case that the seven digits of the DN are not included somewhere within the SPID. When DNs are required, only the last seven digits (no area code) need be entered.

Long-Distance Digital Connectivity

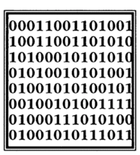
Long- distance connectivity is routinely available in most parts of the USA from the "big- three" carriers: AT&T, Sprint, and MCI. Connectivity between any two given points is somewhat variable. The "dial 1+" default carrier may be chosen at the time you order the line, just as with traditional voice lines. Also, just as with voice lines, you may choose a carrier on a per- call basis by prefixing the number with the 10XXX carrier selection code.

You must dial the full number, including the 1 or 011 + country code following the prefix.

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.

The Zephyr 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 Zephyr depending upon the rate selected by the calling party. Since the Zephyr cannot communicate at both rates simultaneously, it will ignore rate adaptation information on the second incoming call remaining at the rate set for the prior call.

In our experience, the only sure way to know the capacity of a given connection is to try it, first at 64kbps and then at 56kbps if the higher rate fails.



> ISDN TIP!
>
> Not all carriers can handle ISDN connections at all. For the latest information check the section in the Appendix "ISDN, BRI, Zephyr, and You".

> 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.

HOW TO ORDER ISDN

> 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.

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 a Zephyr.

The easiest way to order the line is to use the Faxable form in the Appendix, *ISDN BRI, Zephyr, & You.* There is also a list of contact telephone numbers for most of the regional telephone companies.

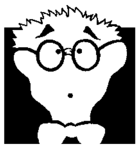
Details, Details

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 are 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 in contrast to PSD, Packet Switched Data, which is possible, but irrelevant to our needs.

Zephyr allows the option of a voice call on either of the channels as well as the coded hi- fi audio. The Zephyr's voice capability exists on both channels, even when your Zephyr is a 3- DSP "mono" version. Thus you can make calls to any normal telephone number on one channel while a program is being transmitted on the other. A limit of one voice call may be made.

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.

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' (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. Zephyr's shipped outside the USA and Canada have the S interface only, as do units made prior to 1996. In the USA and Canada the NT1 is usually provided by the user, however Zephyrs sold in the US and Canada and therefore built- in to the ISDN terminal adapter.

In the USA and Canada, an external NT1 can be powered by the Zephyr.

Refer to manual section 3 (Zephyr at a Glance), and section 4 (Installation & Basic Operation) for additional information on NT1s.

Terminals and Terminal Types

Any equipment connected to an ISDN line is a 'terminal' – whether phone, computer, or Zephyr. 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 Zephyr varies depending upon protocol and is given below.

Zephyr ISDN Compatibility

The Zephyr's internal ISDN interface (sometimes called by it's generic name 'Terminal Adapter') is used to connect to ISDN telephone lines. A simple menu selection easily adapts the Zephyr to the various types of service offered by the range of central office switches installed 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 the Zephyr 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 equipment used on the line and the standards which are followed. While each will work with the Zephyr, the differences need to be taken into account when lines are ordered and the Zephyr 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 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.

The Zephyr supports all of these with the appropriate selection of the "Telco" 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 sometimes preferred because you don't have to trouble with the SPIDs.

Ordering: CO Switches and Protocols

Here are detailed descriptions of what you tell the phone company you want and the corresponding Zephyr settings for each of the protocols. Remember, this information is included in the ISDN Ordering Form included in the Appendix

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 you do not require the Zephyr’s ability to call a regular (POTS) telephone you may specify Capability Package “R”

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

Terminal Type: A

10XXX: 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.

ZEPHYR SETTINGS (FOR YOU TO ENTER)

Set Telco to: Natl I-1

SPIDs: Enter one or two numbers, depending upon number of active B channels

AT&T Point-to-Point (Custom) (USA, Israel, Japan, 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 Zephyr functions. It is the most convenient protocol for Zephyr 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

10XXX: 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.

ZEPHYR SETTINGS (FOR YOU TO ENTER):

Set Telco 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 the Zephyr.

Order 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

10XXX: 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

> ISDN TIP!
>
> Zephyr firmware rev 2.65 or is later recommended for users on the Northern Telecom DMS 100 switch. Consult Telos Systems customer support for upgrade information.

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 Telco to: Natl I-1

SPIDs: Enter one or two numbers, depending upon number of active B channels

Euro-ISDN (Europe, Hong Kong, some others)

The Zephyr 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)

ZEPHYR SETTINGS (FOR YOU TO ENTER)

Set Telco to: ETS300

SPIDs: Not required.

MSNs: If multiple subscriber numbers are used, they should be entered in the MSN/SPIDs screen.

SECTION 6

NON-ISDN NETWORKS

ISDN TIP!

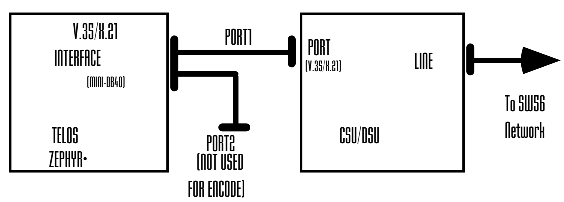
Zephyr firmware rev 2.53 or later required for Euro ISDN. Version 2.69 or later is highly recommended for maximum Euro ISDN compatibility. Consult Telos Systems customer support for upgrade information.

SWITCHED 56

Switched 56 is a transition technology which is being offered primarily in areas where ISDN is not available. As the name suggests, this is a service which permits a single bi-directional digital data stream to be sent at a 56kbps rate. It can use either one or two copper pairs (“2 wire” and “4 wire”, respectively), depending upon the particular Central Office configuration. In most areas, switched 56 and ISDN are “inter- worked” within the Central Office switches, so that subscribers of both services can connect to each other (at a 56kbps rate, of course).

A switched 56 line requires a *Channel Service Unit / Data Service Unit (CSU/DSU)* to interface the digital line with the Zephyr.

These come in two varieties: two- wire and four- wire. The one you need depends upon the service style offered by your telephone company.



Typical setup for mono configuration. Only one port is used.

In many cases you can purchase this equipment from your Telos Systems dealer. Some manufacturers for these Switched- 56 CSU/DSUs are:

Adtran
901 Explorer Ave

Huntsville, AL 35806

800- 827- 0807

Adtran makes all sort of equipment for the telecom industry. The telcos buy equipment from Adtran. Reputable company.

INC
Integrated Network Corp.

757 Route 202/206

Bridgewater, NJ 08807

+1 908.218.1600

This company offers high- quality and, in our experience, excellent customer support.

MORE NETWORK OPTIONS

Primary Rate ISDN

There is also a higher- capacity service referred to as “Primary Rate ISDN.” Similar to the T1 technology described earlier, this permits subscriber access to up to 23 channels of 64kbps each. A customer may use as much capacity as necessary and be charged accordingly.

Primary Rate ISDN requires a special *Channel Bank* or *Bridge* to break out the individual channels. These are usually modular, permitting each channel (time slot) to be used individually as desired. Each Zephyr would require 1 or 2 channels for bi- directional mono and stereo, respectively.

Digital Data System (DDS)

A DDS is a dedicated, “nailed- up” connection between two specific locations. Also referred to as dedicate digital service.

Use of this service requires a CSU/DSU (*Channel Service Unit/Data Service Unit*) to interconnect the line with the Zephyr.

Other Possibilities

Any transmission channel which can convey 56 or 64kbps synchronous digital signals in real time may be used with the Zephyr. Possibilities include T1, satellite links, spread-spectrum RF transmission systems, and ATM.

When considering a technology such as ATM it is essential that your service has a “Committed Information Rate” of at least 56 or 64Kbps and that it is a “Constant Bit Rate” service

HOT TIP! When ordering lines, such as DDS or T1, which will not be used for interstate access be sure to file an “Interstate Access Charge Exemption Form”. This should result in significant savings in monthly fees.

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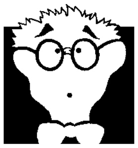
USING THE ZEPHYR WITH NON-ISDN NETWORKS

V.35/X.21 Network Connection

If present, the V.35/X.21 interface is supplied in the rightmost slot, looking from the rear. It permits connection to transmission paths other than ISDN. It may also be used with external ISDN Terminal Adapters in the (rare) case where the available ISDN service is not compatible with the Zephyr’s internal TA.



DEEP TECH NOTE!

The standard uses electrically balanced transmission for both clock and data (but not for the special auxiliary signals, which are identical to RS-232). The transmit voltage level is ± 5.5V, but receivers must handle wider swings.

CURIOSITY NOTE!

You might wonder why the much more common RS-232 is not used. Answer: V.35 is synchronous, meaning that the bit clock is transmitted between the two ends. Self-clocking standards, like RS-232, require overhead start and stop bits, slowing and chopping the bit flow. And, just as with audio, balanced transmission is more reliable in a noisy environment, or in one which has ground potentials at differing levels.

First, A Glossary

Some terms you will need to navigate this section:

V.35. A standard for interconnecting synchronous digital data paths. Like RS- 232, it defines signals and (not officially) connectors and pin- outs so that equipment from various manufacturers may talk with each other. The usual connector is a big boxy AMP type which was chosen by AT&T decades ago. Most terminal equipment sold for the US market supports the V.35 standard.

X.21. A more modern standard, also for interconnecting synchronous digital data paths. This standard specifies the much more reasonable DB- 15 connector. Fortunately, it is possible to design a receiver which accepts both V.35 and X.21 signals. This we have done with the Zephyr, allowing connection to both types just by changing the interconnect cables. The X.21 standard is frequently seen in Europe.

Throughout this manual, we use the universal designation “V.35/X21” to refer to the Zephyr’s digital I/O port protocol.

CSU. ***Channel Service Unit. DSU. Data Service Unit.*** This is what the interface between the Zephyr and digital channel is usually called. It converts the signals from the telephone line to the required V.35 or X.21 standard and provides dialing capability. (Some have keypads and LCD displays for this function; others rely upon external computer control.) The designation CSU is most often used when terminating Switched- 56 lines. Modular CSUs are also used within a ***Channel Bank*** to break out the various paths. Originally the CSU and DSU were separate pieces of equipment. Now they are nearly always one piece of gear variously referred to as a CSU, DSU, CSU/DSU, or a DSU/CSU.

TA. Terminal Adapter. Performs same function as CSU. This terminology is most commonly used with equipment applied to ISDN lines.

Throughout this manual, we use the universal designation “TA/CSU” to refer generally to an external ISDN or Switched- 56 interface device.

CURIOSITY NOTE!

In the old days a CSU (Channel Service Unit) was used to terminate the digital circuit. One or more DSUs (often in the form of plug in cards) were used to provide the interface to the terminal equipment. These days the 2 devices are nearly always combined and are variously referred to as CSUs, DSUs, CSU/DSUs, and DSU/CSUs!

DSP. Digital Signal Processing. The Zephyr uses DSP to perform the coding and decoding functions. Also refers to a *Digital Signal Processor,* the actual chips which perform the processing functions. One or two plug- in DSP card modules may be installed in the unit giving a total of 5 (2 card version) or 3 (1 card version) DSPs.

One-DSP card/Two-DSP card configuration. Refers to Zephyr configurations. The “mono” one- DSP card configuration is that which exists when only one DSP card is installed within the unit (3 DSPs would be present); the “stereo” two- channel configuration requires two DSP cards (for a total of 5 DSPs). The two- DSP card configuration also usually needs to have both digital ports connected to a transmission path. However, the two- channel configuration supports single- channel as a subset. A full discussion of these issues is to be found in section 11 (Technical Information) and a review of available Zephyr configurations appears in section 2 (Introduction).

Cables

The Zephyr connects to digital transmission channels via the ports from

HOT TIP!

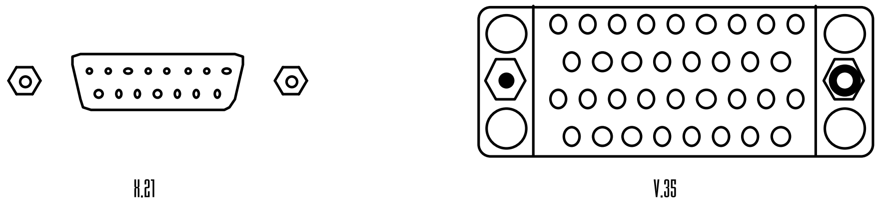
A “mono” one-DSP card configuration can support 2 bi-direction audio channels if the G.722 coding mode is used.

HOT TIP!

A mono one-DSP card unit can *receive* a STEREO signal using both ports.

the V.35/X.21 interface module.

The cables available from Telos, connect to standard terminal adapters or CSUs, which, in turn, connect to the digital network channels. The design of the port electronics permits interface with either V.35 or X.21 equipment – the connector style and pin- out is the only difference.



You can tell whether you have V.35 or X.21 by looking at the connectors. These are the " footprints" of the two types.

Each V.35/X.21 connection conveys both send and receive digital signals.

A single- port cable has only one V.35/X.21 plug, while the two- port cable version has two V.35/X.21 plugs. A two- port cable may be used for single- port applications by simply letting the unused plug go unconnected.

The compact plug at one end of the cable snaps onto the mating connector on the Zephyr with a locking action. To release the plug, the two pushbuttons at either end of the plug housing are pressed simultaneously and the plug is withdrawn.

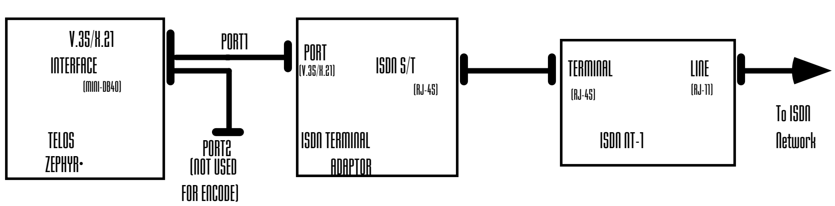
The plugs at the other end are labeled "PORT 1" and "PORT 2" and are connected to the TA/CSU as follows:

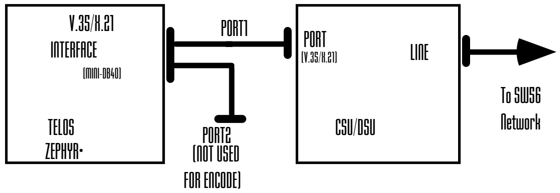
> IMPORTANT!
>
> We had to use a small socket in order to make more room on the rear panel (sorry), so you cannot plug directly into it with off-the-shelf cables. You must have the appropriate cable from Telos to connect!
>
> The V.35/X.21 interface, while a single physical plug, has two electrical ports, of which one or both may be used.

One Port Operation

- For one- port mono operation, only the PORT 1 plug is used. It connects to the TA/CSU V.35/X.21 socket.

- In the `MONO/DUAL` mode, when used for just one mono audio program, only one V.35/X.21 connection is required. Audio channel A corresponds to PORT 1 and audio channel B corresponds to PORT 2.



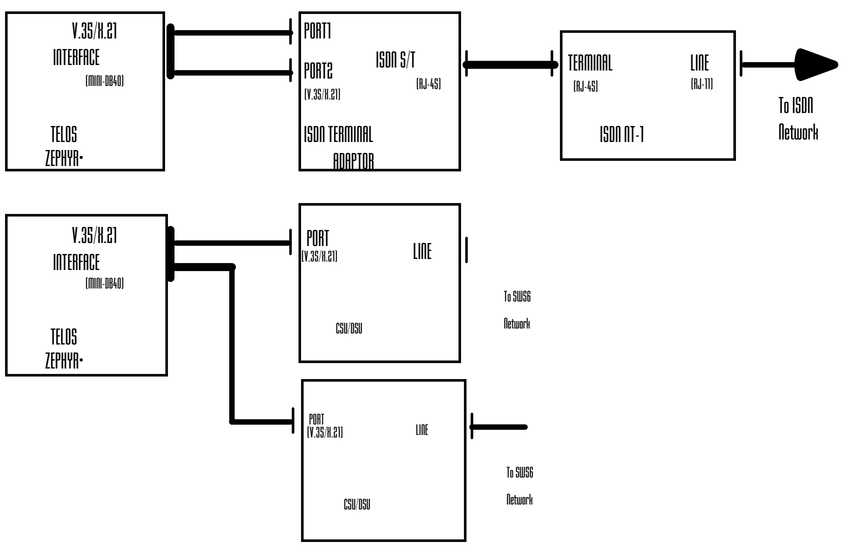


Two Port Operation

- For two- port operation, two digital transmission paths are required, so both the PORT 1 and PORT 2 plugs mate to the TA/CSU digital connections. The TA/CSUs may be separate for each path, or may be a single unit with two data ports. In either case, the system automatically sorts out which channel is which and therefore it is irrelevant which plug goes to which port.

- For two- port operation in the `MONO/DUAL` mode, audio channel A corresponds to PORT 1 and audio channel B corresponds to PORT 2, and both are used.

- For two- port operation where the receive mode is set to L3 MONO only the data received on PORT 1 will be decoded, although this audio will appear at both the A and B receive audio jacks.



> HOT TIP!
>
> When troubleshooting equipment problems or failures keep the following in mind.
>
> •In L3 DUAL Xmt mode channel A audio will be coded and sent out PORT 1 and channel B audio will be coded and sent out PORT 2.
>
> •If the receiving unit is put into L3 MONO Rcv mode it will be receiving from PORT 1 only.
>
> Therefore, by systematically switching cables, TA/CSUs, and lines, you will be able to isolate the problem to a single line or piece of equipment.

Control Leads

There are a number of “control leads” defined in the V.35 and X.21 standards. TA/CSUs will frequently are indicator lights showing the status of these signal which may be useful when troubleshooting problems.

In the direction Zephyr - > TA/CSU, the Zephyr provides DTR *(Data Terminal Ready)* and RTS *(Request to Send)*. Both of these signals are made active when the Zephyr is ready for operation.

In the direction TA/CSU - > Zephyr, only CD *(Carrier Detect)* is implemented. This signal is used to tell the Zephyr which port channel has been connected and activated. It is used for internal selection of the network clock source.

> HOT TIP!
>
> Since the data is generally clocked from the master clock at the Telco anyway, hand shaking is unnecessary. Timing variations between the various handshake signals varies and may cause equipment incompatibilities. If this occurs it is entirely ok to disable handshake signals on the TA/CSU using the appropriate DIP switch or command.

Configuring CSU/DSUs

There will generally be at least two main configuration menus. One for DTE (Data Terminal Equipment) and one for the loop or line.

DTE Options: Set to “Synchronous” and set the bit rate to match the Zephyr and line’s bit rate (usually 56kbps). If necessary, you may disable any handshaking signals by setting them to “forced” or “ignored” since handshaking is not required with synchronous data.

Loop or Line Options: Set to the speed of your line (usually either 56 or 64kbps). Your Telco should be able to give you the correct setting for any other options.

Clock Options: It is essential that the clock option be set to “external” or “from loop”

SECTION 7

AUDIO CODING

OVERVIEW

Introduction to Audio Coding

Without data rate reduction, high quality audio requires a transmission capacity of about 700kbps for each audio channel. At 56kbps, a telephone channel offers a rate about 8% of the Compact Disc's.

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.

For high-fidelity transmission, algorithms with more power are required. These are based on psychoacoustics, where the coding process is adapted to the human perceptual system. 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)* in order to develop a universal standard for encoding moving pictures and associated audio for use with digital storage and transmission media. The standard was finalized in November 1992 with three related algorithms, called Layers, being defined for encoding of audio taking advantage of psychoacoustic effects. While Layer I and II are intended for compression factors of about 4 and 6-8 respectively, with Layer III factors of up to 12.5 can be achieved.

Basic Principles of Perceptual Coding

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

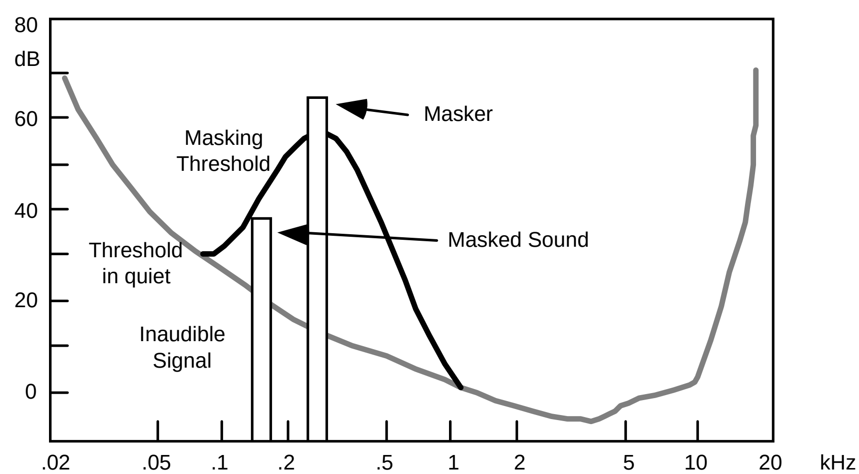
Lossless – which, for audio, translates to noiseless – coding with perfect reconstruction would be an optimum system, since no information would be lost or altered. It might seem that lossless, redundancy-reducing methods (such as PKZIP, Disk Doubler, and others used for computer hard-disk compression) 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. Since any system intended for a telephone channel is required to have a consistent output rate and must accommodate the worst case, no compression is possible 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.

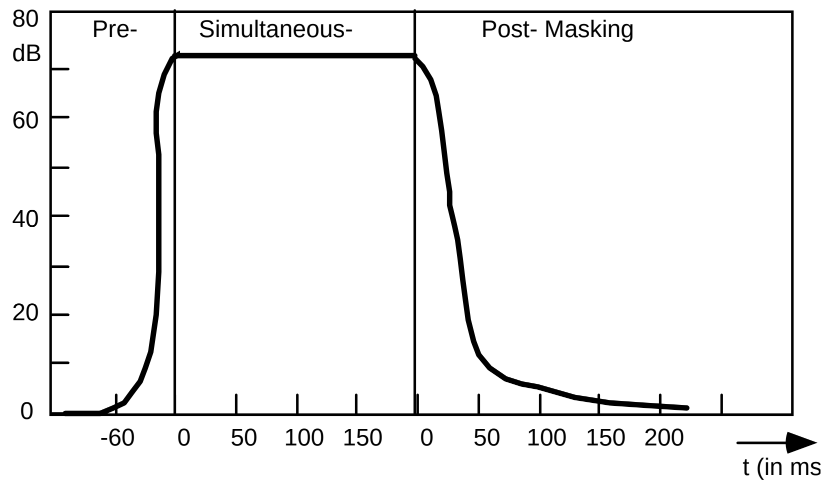
All signals below the upper "threshold of audibility" curve, or Masking Threshold are not audible, so we can drop them out or quantize them crudely with the least number of bits. Any noise which results from crude quantization will not be audible if it occurs below the threshold of masking. The masking 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 III

MPEG Layer III is the most powerful coding method available in the Zephyr, 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.
- CD- quality stereo 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 III Features

Psychoacoustic Masking

The audio in Layer III is divided into 576 frequency bands. First, a polyphase filter bank performs a division into the 32 “main” bands which correspond in frequency to those used by the less complex Layer II. Filters are then used to further subdivide each of the 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 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.

Bit Reservoir Buffering

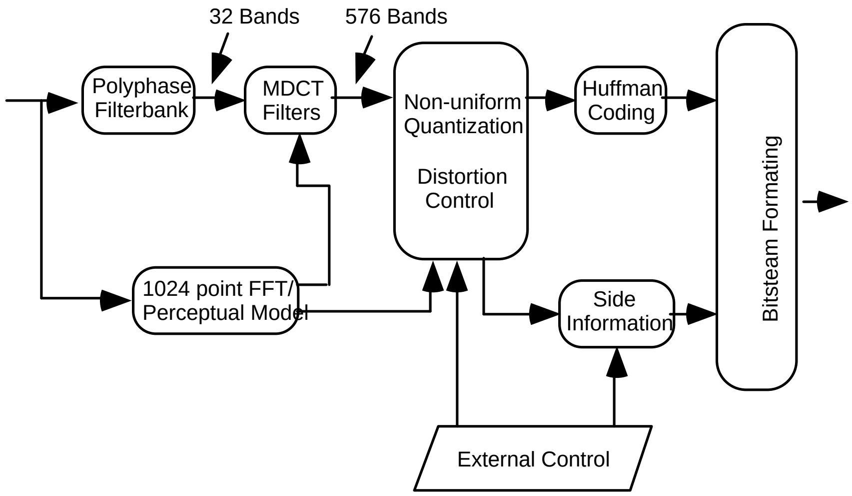
Often, there are some critical parts in a piece of music that cannot be encoded at a given data rate without audible noise. These sequences require a higher data rate to avoid artifacts. Layer III uses a short time “bit reservoir” buffer to address that need. Similar to a savings account, this buffer 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

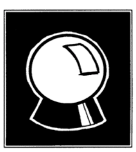
The bit reservoir buffer offers an interesting capability: an effective solution for the inclusion of such ancillary data as text or control signaling. The data is held in a separate buffer and gated onto the output bitstream using the bits allocated for the reservoir buffer when they are not required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

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.



Block diagram of the Layer III coding process.



CURIOSITY NOTE!

Layer III 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 II algorithm called MUSICAM, had the advantage of a lower complexity. LIII came from a merging of both systems: For easier transcoding, LIII adopted the LII filterbank for the first stage, and the bitstream rate was adjusted.

ISO/MPEG LAYER II

MPEG Layer II is the world's most popular perceptual coding method. This is primarily because it is less complex to implement than Layer III, particularly at the encoder. It is the preferred choice for applications where greater than 120 kbps/channel is available, such as satellite links and high- capacity terrestrial paths such as Primary ISDN or T1 channels.

We include it in the Zephyr in order to offer compatibility with the widest variety of codec equipment.

Our implementation is the very highest quality, the critical DSP code having been obtained and licensed directly from the primary inventor of LII, the IRT in Munich.

Joint Stereo

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

Mono-128

Zephyr offers the ability to send mono using Layer II utilizing both b channels (112/128kbps). This mode offer the best fidelity mono 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 the Zephyr 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, about 42 ms or less.

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.

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.

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 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

IMPORTANT!

The locking can be sensitive to audio present on the G.722 path, as it relies upon the properties of the audio bitstream 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, removing, or lowering (to around -12dB) the audio for a few seconds will generally cause lock to occur. 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.

this is inherent in G.722’s statistical framing and is not an implementation problem with the Zephyr.

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 III at 56/64kbps can be cascaded with 2 - 5 passes of Layer II operating at high (112kbps+/channel mono; 192kbps+ Joint Stereo) bitrates with good results.

- Informal tests at the Telos lab with two passes of Zephyr Layer III have proven successful, with listeners noticing no audible degradation, even on “difficult” CDs.

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

- One user has reported that two passes of Zephyr Layer III, 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 – the least "crunching", for instance, the lowest reasonable sample rate and the highest available bit rate – 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 the Zephyr may be used in a mode where a mono program is split over two digital network channels.

- Get the Layer III advantage on low bitrate channels.

The people at Fraunhofer IIS, who developed the Layer III 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 LII And LIII Signal Chains

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

At low bit- rates, Layer III gets more signal- to- mask margin than Layer II. 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 LIII naturally exploits this additional dimension. The down-side is that when many stages of LIII 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 LII 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 LIII be used. The idea is that LIII be used at ISDN/SW56 bit- rates for field pick- up and that LII 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: LIII for 64kbps/channel and LII 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 LIII sound noticeably better than two of LII; a pass of LIII followed by a pass of LII also sounds better than two of LII. And we've had customers who have used a pass, or two, of LIII followed by SEDAT without evident problems.

CHOOSING THE CODING METHOD MOST APPROPRIATE TO YOUR APPLICATION

The following chart describes and compares some of the important characteristics of each method.

Audio Coding Comparison Chart

	Layer III	Layer II	G.722
Algorithm	Perceptual+Huffman	Perceptual	ADPCM
Audio Freq. Response/mono			7 kHz
24 kHz sample rate	**- -**	8.6kHz	- -
32 kHz sample rate	15 kHz	- -	- -
48 kHz sample rate	20 kHz	7.8/9.8 kHz**	- -
Audio Freq. Response/stereo	15/20 kHz*	20 kHz	7 kHz
Delay at HALF/24 kHz	**- -**	990 ms	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"	12:1	6- 8:1***	4:1
Bands	576	32	2
Frequency Resolution (48 kHz)	42 Hz	750 Hz	- -

* 15 kHz at 32 kHz sample rate; 20 kHz at 48 kHz sample rate.

** 7.8 kHz at a 56 kbps network rate; 9.8 kHz at 64 kbps.

*** 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.*

Because the Zephyr includes all three popular coding methods, it is possible to choose the most appropriate for each application.

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 III'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 II requires the next longest delay, and G.722 has minimal delay.

The Zephyr 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 (Installation/Basic Operation- Dealing With Delay) 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 LIII or LII for only the on-air direction.

Dual vs Stereo vs Joint Stereo in Layer III

With one transmission path,

- LIII DUAL provides mono capability.

With two transmission paths,

- LIII DUAL mode is simply two simultaneous mono channels.
- LIII STEREO mode compensates for any delay between the two transmission paths, but keeps the two audio channels completely independent.

> DEEP TECH NOTE!
>
> The longer delay in the STEREO modes results from the mux/demux process. In order to accommodate the possible differential delay between the two transmission channels, the system must buffer the two digital signals. The frames are read out from the buffer according to their numbering, which is generated during the encoding process. This makes for a very reliable method to ensure that the channels are in sync.
>
> The MONO and DUAL/MONO modes do not require buffering and therefore have a lower delay. While it is possible to use the LIII DUAL/MONO mode for stereo transmission, the system is unable to correct for delay differences between the two channels and significant phase problems are likely to occur.

- LIII JSTEREO (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 LIII JSTEREO 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 the Zephyr. This could be used for particularly critical mono material. The identical signal must be fed to both inputs, preferably by splitting the signal just before it enters the Zephyr; the system automatically takes care to properly assign the bits and no special mode is required.

> IMPORTANT!
>
> The Layer III decoder in L3 STEREO mode requires that both channels be connected and operating in order for the decoder to function. Until both channels are present, the decoder will not output anything, even if the Xmt mode at the other end 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.

Mono vs MONO128 vs Dual vs JSTEREO in Layer II

With one transmission path,

- LII MONO provides mono capability.

With two transmission paths,

- LII MONO128 provides the best Layer II quality by combining both transmission paths to achieve a higher bitrate. This mode require a 5 DSP (Stereo) Zephyr configuration and 2 data paths.

- LII DUAL mode is simply two simultaneous mono channels.

- LII JSTEREO mode uses the “intensity coding” method in order to provide maximum quality for stereo program material and compensates for any delay between the two transmission paths.

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 III 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 not transmitted, anyway.

Layer II primarily operates at 48kHz sample rate. However, the L2 HALF/24 Rcv and Xmt modes allow operation at 24kHz when this sample rate is desired, primarily when communicating with “single line” capable Layer II codecs.

Decoder Limitations

- The Layer III decoder in L3 STEREO mode requires that both data channels be connected and operating to function. Until both channels are present the decoder will not output anything, even if the Xmt mode at the other end is not stereo.

- 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 support fully independent operation.

- A typical application which works perfectly within the above constraints is the one used for dual- language sports broadcasts, with separate studio sites. L III Dual Xmt mode is used from the stadium to the separate studio sites, while G.722 is used for the cueing channel from the studios back to the stadium.

Compatibility

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

The Zephyr’s 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.

Manual section 8 (Detailed Menu Reference) has more information about compatibility, and the Appendix contains a section detailing known non- Telos equipment compatibility.

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SECTION 8

DETAILED MENU REFERENCE

COMPATIBILITY TIP!

Some of the features detailed in this manual may not be available when communicating with Zephyrs using firmware versions earlier than version 2.69. Each Zephyr will connect with every other Zephyr, but you may need to modify some of your menu settings. The most frequent compatibility issues are listed below and in other appropriate sections of the manual.

MENU OVERVIEW

Zephyr has six menus; UTILITY, DIAL, AUTO- DIAL, DROP, VOLUME, and HELP. Each menu is accessed from a dedicated button on the front panel. As Zephyr is designed for ease of operation, each menu has, at most, two levels. There are no complex menu-trees to remember.

Telos Systems is continuously updating Zephyr's features. The manual section that follows is for system firmware version 2.69. The system firmware version is the first number displayed on the third line of the LCD when you power on your Zephyr. If your Zephyr's menu differs from the description provided, check to see if you have an earlier or a later version of the firmware. If your firmware version is earlier, contact your dealer or Telos Systems customer support to learn how to obtain an upgrade. If you have a later version, changes in the menu are documented in the on- board HELP menu.

This manual section summarizes the menu items and many of the considerations involved in selecting the operating parameters. More details to help you make menu choices are found in background information in the appropriate manual section.

Menu Summary

UTILITY

The UTILITY menu is used to setup the parameters of the two major sections of the Zephyr; the audio codec and the network interface. Zephyr is a truly integrated transceiver, so you need to make only one adjustment to change a setting in both sections.

HOT TIP!

Many of the menu items require confirmation of your new setting to prevent accidental changes. The prompts on the fourth (bottom) line of the LCD will advise you when confirmation, with another button press, is required. If your Zephyr does not seem to "accept" your changes, failure to confirm the setting may be the cause.

From this menu, you can change the encoding and decoding schemes, transmission bit rate, and codec sampling rate. Selection and configuration of the network interface, as well as storage of auto- dial setups, are also accomplished from the UTILITY menu. Other items configured from this menu include configuration of the STATUS OUTPUT feature and the optional AES/EBU Digital Input/Output Interface Module.

DIAL

As the name implies, this menu is used to dial calls with Zephyr's internal terminal adapter. The number of the receiving party is manually entered and you can select whether you are placing a high- fidelity call via ISDN or a voice- grade call to a Plain Old Telephone Service (POTS) phone.

AUTO-DIAL

This menu is used to place calls to frequently accessed numbers stored as auto-dial setups. While you can verify the recipient's number, codec settings, and transmission rates from the AUTO- DIAL menu, you cannot make changes. All changes to the auto- dial setups are accomplished from the UTILITY menu.

DROP

From this menu, calls placed using the DIAL and AUTO- DIAL menus are terminated.

VOLUME

Audio levels for the front panel headphone jack and telephone monitoring speaker are adjusted from this menu. In addition, the VOLUME menu can be used to adjust the level of the input audio to be sent to the digital network.

HELP

The HELP menu provides brief descriptions of all of the Zephyr menu items, as well as information on other status information displayed on the LCD display.
It can be accessed at any time by simply pressing the <HELP> button on the front panel.

Using the Menus

Accessing a menu is as easy as pushing a button on the front panel. All menus, except the HELP menu, operate in the same manner.

When you initially select a menu, you see a display of up to three menu items on the four- line LCD display. Each of the menu items displays its name on the left and its current setting on the right.

Brackets around a setting indicate that this menu item is currently available for modification. To select a different menu item, use the <SELECT ▲> and <SELECT ▼> buttons. The brackets move up and down to indicate the selected menu item.

To change the settings of any menu item, the <NO (-)> and <YES (+)> button are used. The fourth row of the LCD display describes how to change the settings of the selected menu item using these two buttons.

DETAILED MENU REFERENCE

Utility Menu

Summary of Utility Menu Items

Xmt	Transmit (encoding) mode
Rcv	Receive (decoding) mode
Rate	Rates of transmission and codec sampling
Network	Network connection
AES In	AES/EBU input sampling rate
AES Out	AES/EBU output sampling rate
Auto Answer	Auto Answer activation
Loopback	Loopback activation
Status Out	Status available on logic output
Store Setup	Auto-Dial setup number selection
Category	Category of Auto-Dial setup to be changed
SPID 1 & 2 /MSN 1 & 2	SPID/MSN entry fields when required
Directory 1 & 2	Directory number entry fields (rarely required)
Telco	ISDN protocol selection
Panic Dial	Panic Dial activation and setup selection
Compatibility	Used only for certain subscription services
LCD contrast	LCD contrast
LCD backlight	LCD backlight
Ancill Chan	Ancillary data transmit channel
Version Info	Details currently installed firmware

The UTILITY menu may be scrolled in either a forward or reverse direction using the < SELECT ▲ > and < SELECT ▼ > buttons.

Xmt Transmit (encoding) mode

IMPORTANT!

The following Xmt modes require a 5 DSP (Stereo) Zephyr: L3 STEREO, L3 JSTEREO, L2 MONO-128, L2 DUAL, L2 JSTEREO. A 3 DSP (Mono) Zephyr will display "Hardware Not Available" if these modes are selected. For information on Zephyr configurations see manual section 2 (Introduction). A 3 DSP (Mono) Zephyr can be upgraded to 5 DSP (Stereo); Contact Telos customer support for additional information.

HOT TIP!

A typical application which works perfectly within the constraints given above is the one used for dual-language sports broadcasts, with separate studio sites. L III DUAL mode is used from the stadium to the two separate studio sites, while G.722 is used for the cueing channel from the studios back to the stadiums.

This menu item selects the method used to encode local audio for transmission to the remote site. *The remote unit must be set to a compatible receive mode.*

Zephyr offers the choice of IS0/MPEG Layers III and II, as well as G.722. Selection of transmit mode is accomplished by choosing the item in the UTILITY menu and using the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

Xmt modes may be changed while a call is in progress.

IMPORTANT!

Changing the Xmt, Rcv, or Rate options while connected will cause the Zephyr to cease transmitting and receiving audio as the new code is loaded.

Available modes are:

- **L3 DUAL** Layer III Single or Dual Channel Monaural

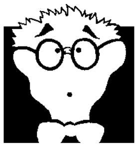
  This selection is used for independent monaural audio channels using Layer III. This mode permits the transmission of one mono signal over a single 56/64kbps channel. In addition, two signals can be sent to a single site or to two different locations over two 56/64kbps transmission channels.

  15kHz audio bandwidth per channel is achieved with the recommended 32kHz sample rate. 20kHz audio bandwidth is achieved per channel with a 48kHz sample rate.

  The two audio channels are "tagged," so that encode channel A appears on the decoder's channel A, and encode channel B appears on the decoder's channel B, regardless of a swap of the transmission paths.

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 each site.



DEEP TECH NOTE!

In the JSTEREO and STEREO modes, the two audio channels are output internally from the coder as a single mixed bitstream and then split for transmission. At the end of the process, there is no correspondence between audio channels and ISDN lines.

Receive, if used, must be set to the G.722 mode. The Layer II/III decoder is unable to accommodate the independent signals.

- **L3 JSTEREO** Layer III Joint Stereo

HOT TIP!

In cases where a return from only one site is required, the return can be made using LIII Mono. The site which requires bi-directional audio must connect to Line 1 for this to work.

This selection is best for most stereo program material.

15kHz stereo audio bandwidth is achieved with a 32kHz sample rate. 20kHz stereo audio bandwidth is achieved with a 48kHz sample rate.

The two audio channels are “tagged,” so that encode channel A appears on the decoder’s channel A, and encode channel B appears on the decoder’s channel B, regardless of a swap of the transmission paths.

HOT TIP!

Some older firmware versions of the Zephyr have only one Layer III stereo mode selection. This mode. labeled STEREO, is actually Joint Stereo. We had thought everyone would want to use this mode for stereo. We were wrong, and later added the ability to choose either of the stereo modes.

- **L3 STEREO** Layer III Discrete Channel Stereo

  This selection is best for stereo signals with a rich stereo image or when signals are to be later used in surround- sound applications. It uses two 56/64kbps transmission channels, such as single ISDN circuit.

  15kHz stereo audio bandwidth is achieved with the recommended 32kHz sample rate. 20kHz stereo audio bandwidth is achieved with a 48kHz sample rate.

  The two audio channels are “tagged,” so that encode channel A appears on the decoder’s channel A, and encode channel B appears on the decoder’s channel B, regardless of a swap of the transmission paths

- **G.722** G.722 Coding

This selection permits independent audio channels using G.722. This mode permits sending one mono signal over a single 56/64kpbs transmission channel. In addition, two signals can be sent to a single site or to two different locations over two 56/64kpbs transmission channels.

Audio bandwidth of 7kHz is achieved. This mode is usually chosen for backwards compatibility or lowest delay on communications paths.

The two audio channels are connected according to the transmission paths. That is, a swapping of the paths results in a swapping of the audio channels.

- **L2 HALF/24** Layer II Single channel special half sample rate (24kHz).

  For compatibility with single line capable codecs when they are used in 24kHz sample rate mode. This Mono mode divides the usual 48kHz in half. Since the master sample rate clock remains at 48kHz this transmit mode can be combined with any receive mode operating at 48kHz.

  Audio bandwidth of 8.6 kHz is achieved.

- **MONO** Layer II Single Channel Monaural

  This selection permits the transmission of one mono signal over a single 56/64kpbs channel.

  Audio bandwidth of 7.8 or 9.8kHz is achieved, depending on transmission bitrate.

- **L2 MONO128** Layer II Broadband Single Channel Monaural

  This selection permits transmission of one mono signal over two 56/64kpbs transmission channels, such as a ISDN circuit. (Total transmission rate of 112/128kbps.)

> COMPATIBILITY TIP!
>
> Zephyr's monaural Layer II implementation prior to version 2.69 does not include the L2 HALF/24 Xmt and Rcv modes. If this mode is required contact Telos Systems customer support for upgrade information.

  20kHz audio bandwidth per channel is achieved.

  This is the best quality Layer II mode.

- **L2 DUAL** Layer II Dual Channel Monaural

  This selection permits independent audio channels using Layer II. This mode sends two mono signals to a single site over two 56/64kpbs transmission channels. Sending the two signals to different sites is not possible.

  Audio bandwidth of 7.8 or 9.8kHz per channel is achieved.

  The two audio channels are "tagged," so that encode channel A appears on the decoder's channel A, and encode channel B appears on the decoder's channel B, regardless of a swap of the transmission paths.

COMPATIBILITY TIP!

The Layer II Broadband Single Channel Monaural mode (L2 Mono128) uses the CDQ method of multiplexing its single signal over the two transmission channels. This method is compatible with other Zephyrs and a majority of Layer II-only codecs. See the Compatibility Section of the Appendix for more details.

- **L2 JSTEREO** Layer II Joint Stereo

  This selection is good for stereo signals using two 56/64kbps transmission channels,

  such as single ISDN circuit.

  20kHz stereo audio bandwidth is achieved.

  The two audio channels are "tagged," so that encode channel A appears on the decoder's channel A, and encode channel B appears on the decoder's channel B,

HOT TIP!

As a 3 DSP (Mono) Zephyr supports all receive modes, it makes an excellent low cost receiver in applications where the return feed is mono, or not required.

IMPORTANT!

Note that the Rcv modes are a subset of the Xmt modes; in other words, a given Rcv mode will generally be applicable to more than one Xmt mode. For instance, the L3 STEREO Rcv mode is can decode audio from the following Xmt modes: L3 JSTEREO, L3 STEREO, and L3 DUAL.

COMPATIBILITY TIP!

Keep in mind that compatibility is also affected by the bitrate/sampling rate menu item.

regardless of a swap of the transmission paths.

Rcv Receive (decoding) mode

This menu item selects the method used to decode the audio received by the local Zephyr. The remote unit *should have a corresponding transmit mode selected.* Many of the receive modes correspond to more than one transmit mode.

Zephyr offers the choice of IS0/MPEG Layers III and II, as well as G.722. Selection of receive mode is accomplished by choosing the item in the UTILITY menu and using

> IMPORTANT!
>
> Changing the Xmt, Rcv, or Rate options will cause the Zephyr to momentarily cease transmitting and receiving audio as the new code is loaded.

the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

Rcv modes may be changed while a call is in progress.

Available modes are:

- **L3 STEREO**

  Receives when the remote transmitter is set to any of the Layer III modes (2 transmission channels required):

  - L3 DUAL
  - L3 STEREO
  - L3 JSTEREO

- **L3 MONO**

  Receives only when the remote transmitter is set to the L3 DUAL mode and when only one channel of audio is required.

  The mono audio appears on both A & B output channels.

- **G.722**

> IMPORTANT!
>
> The input to the L III mono decoder is taken only from ISDN Line 1, or from Port 1 when an external CSU/TA is used. Any signal connected on the second line will be ignored!
>
> More specifically, at the encoder, you must dial the number assigned to the first ISDN B channel at the decoder.

  Receives only when the remote transmitter is set to G.722.

Unlike in the LII/III modes, mono audio appears only on the output which corresponds to the incoming digital channel/line.

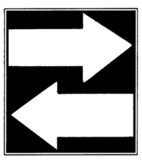
- **L2**

  Receives when the remote transmitter is set to any of the following Layer II modes:

  - L2 MONO
  - L2 MONO128
  - L2 DUAL
  - L2 JSTEREO

When the source encoder is in LII MONO mode, the mono audio appears on both output channels.

- **L2 HALF/24** Layer II Single channel special half sample rate (24kHz).

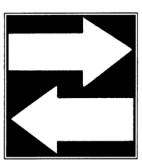


> COMPATIBILITY TIP!
>
> The L2 receive mode uses the CDQ method of multiplexing the two transmission channels to a single bit stream when transmissions are in the L2 MONO128 and L2 JSTEREO modes. This method is compatible with other Zephyrs and a majority of Layer II-only codecs. See the Compatibility Section of the Appendix for more details.

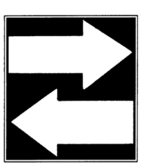
Receives when the remote transmitter is set to the following Layer II mode:

- L2HALF/24

This mode will also receive L2 Mono at 48kHz.



> COMPATIBILITY TIP!
>
> Zephyr's monaural Layer II implementation prior to version 2.69 does not include the L2 HALF/24 Xmt and Rcv modes. If this mode is required contact Telos Systems customer support for upgrade information.



> COMPATIBILITY TIP!
>
> Older Zephyr hardware versions may not support Layer II or the 48kHz sample rate. If this is the case, Zephyr will display "Hardware Not Available" if these modes are selected. Contact Telos Systems customer support for information on upgrades should these modes be required.

Zephyr Xmt-To-Rcv Mode Compatibility Table

This table shows which Zephyr Xmt and Rcv 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 L3 STEREO	L3 STEREO	15/20kHz*	Audio output channels correspond to input.
L2 MONO (Channel A)	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 output 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: 7.8kHz at 56kbps; 9.8kHz at 64kbps.*

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

Rate Transmission bit rate and codec sampling rate

This menu item selects the rate of transmission (of both the codec and network interface) and the codec sampling rate.

> IMPORTANT!
>
> Changing the Xmt, Rcv, or Rate options will cause the Zephyr to cease transmitting and receiving audio as the new code is loaded

> COMPATIBILITY TIP!
>
> It is recommended that you read the descriptions of the menu items that select the receive and transmit modes before changing this rate setting. These three menu items are closely related.

In most operating modes, the Zephyr accepting the data call will automatically adjust its transmission bit rate to that of the Zephyr placing the call. However, the transmission rate information may be lost during the call setup of an international or long- distance call. In addition, the sampling rate must always be set identically on both units and will not automatically adjust. We therefore recommend that the two units be set to identical rate settings.

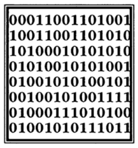
> IMPORTANT!
>
> This mode may be changed while a call is in progress, but the bitrate affects both the coding and the setup of the ISDN connection. The ISDN setup information is passed to the other end and is used by the Zephyr to set the receive bitrate – But this only happens at the beginning of the call.
>
> This means it is possible to get the two ends out of sync, if you change the bitrate after call setup is completed. In that case, one or the other of the Zephyr's bitrate would have to be changed manually to correct this. Therefore we recommend disconnecting and redialing if a bitrate change is made.

Selection of transmit and sampling rates are accomplished by choosing the item in the

> HOT TIP!
>
> On the Zephyr, we refer to the per-line bitrate. In the stereo modes, the total bitrate is two times this number. This can be confusing because some other codecs use the total, rather than the per-line value.

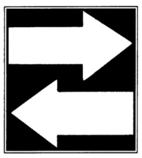
UTILITY menu and using the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection. The transmission rate is indicated on the left and the sampling rate is indicated on the right.

The data transmission rate of both the codec and network interface sections of Zephyr are controlled from this menu item. Zephyr offers transmission rates of 56kbps and 64kbps. 56kbps rates are used for Switched 56 and other transmission modes where “64kbps- clean” data service is not available. These circuits are generally found only in the USA.



ISDN TIP!

USA users whose Zephyr does not lock to the remote Zephyr at 64kbps should drop the call, change the transmission rate of both units, and redial to determine if the reason is a 56kbps limitation of the data service. Users outside the USA who are connecting to USA sites may also find a need for this mode.



COMPATIBILITY TIP!

Older hardware versions of the Zephyr operate at only 32kHz and cannot connect with another Zephyr operating at a 48kHz transmission rate. Telos Customer Support can provide information on upgrading the Zephyr for greater compatibility.

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.

HOT TIP!

Zephyr’s Layer II operates at only 48kHz sampling (although 24kHz operation is possible with the L2 HALF/24 modes). If you select Layer II in the transmit or receive mode, the sampling rate will automatically change to 48kHz if 32kHz is the current selection. The message “Sampling Rate Change” will appear on the fourth line of the LCD when this occurs.

For G.722- only operations, select the appropriate data transmission rate and any sampling rate.

Available modes are:

- 56kbps 32kHz

This mode is used for Layer III- only and Layer III/G.722 operations where “64kbps- clean” data service is not available.

- 64kbps 32kHz

  This mode is used for Layer III- only and Layer III/G.722 operations.

- 56kbps 48kHz

This mode can be used with any transmit/receive combination where “64kbps- clean” data service is not available.

- 64kbps 48kHz

This mode can be used with any transmit/receive combination.

Network Network connection

This menu selects the network interface used by Zephyr to connect to the “outside” digital world. Connection options depend upon the network cards installed in the local Zephyr. While you can have more than one network interface card installed, only one network interface may be used at any time. Changing to the other option while the first is in use, will cause loss of audio.

Selection of network connection is accomplished by choosing the item in the UTILITY menu and using the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

Available network connections are:

- **V.35** V.35/X.21 Network Interface

  Available if the corresponding network interface card is installed. When this mode is selected. Zephyr’s coded digital audio is routed to the V.35/X.21 connector. By using an appropriate interface cable, the data is passed to an external data service unit (CSU/DSU) for transmission via ISDN, Switched 56, Dedicated Digital Service, fractional T1, etc.

- **ISDN** ISDN Terminal Adapter

  Available if the corresponding network interface card is installed. When this mode is selected. Zephyr’s coded digital audio is routed to the internal ISDN terminal adapter. The ISDN terminal adapter is controlled from other menu items in the UTILITY menu, as well as from the DIAL, AUTO- DIAL, and DROP menus.

AES In AES/EBU input sampling rate

This menu is only active when the optional AES/EBU digital input/output interface module is installed. If you use this menu item without the card installed, you will get a message on the fourth (bottom) line of the LCD informing you that you have no AES/EBU card.

Zephyr can accept either an analog or digital audio input. With AES/EBU installed, the

> IMPORTANT!
>
> Before making any adjustment to this setting, it is recommended that you read the descriptions of the menu item that selects the AES/EBU output sampling rate immediately following this entry.

analog and digital audio outputs are simultaneously available.

Zephyr's AES/EBU digital input/output interface module provides maximum flexibility for connection to digital studio equipment. Sample rate conversion is possible on both input and output paths, with support for external sample rates of 32kHz, 44.1kHz, and 48kHz.

The interface may accept clock from an external source and may provide clock when required. On the input side, sample rate conversion is normally used and may accommodate a source clock frequency of 32kHz, 44.1kHz, or 48kHz. A special mode permits bypass of sample rate conversion when it is desirable to synchronize source equipment to the network clock. Zephyr does not have a dedicated clock output, but the internal AES/EBU transmitter (which is locked to Telco clock and output at 32kHz and 48kHz) can be used as a "master" clock source.

Selection of AES/EBU input sampling rate is accomplished by choosing the item in the UTILITY menu and using the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

Available modes are:

- NO (ANALOG)

  When this mode is selected, the AES/EBU input is defeated and the analog audio input is used.

- S/R CONVERT

  When this mode is selected, the AES/EBU input is active. The sample rate is converted to match the sample rate selected in the RATE menu item. Possible selections are 32kHz or 48kHz when using Layer III and 48kHz when using Layer II.

  The analog input is defeated.

- SYNC TO NET

  When this mode is selected, the AES/EBU input is active. The AES/EBU digital audio input must be externally synchronized to the clock provided by the AES/EBU digital audio output signal. Operation is possible at 32kHz or 48kHz when using Layer III or at 48kHz when using Layer II.

  The analog input is defeated.

HOT TIP

Be certain the your audio input is synchronized to the network clock. To accomplish this, you need to take the clock from Zephyr's AES/EBU output and synchronize your equipment to that clock. 44.1kHz is not available in this mode.

HOT TIP!

When communicating with units while set to S/R CONVERT or SYNC TO NET the far-end Zephyr will not get receive lock unless the sending unit has an AES/EBU signal present.

AES Out AES/EBU output sampling rate

This menu is only active when the optional AES/EBU digital input/output interface module is installed. If you use this menu item without the card installed, you will get a message on the fourth (bottom) line of the LCD informing you that you have no AES/EBU card.

Both the analog and digital audio outputs are active in all modes.

IMPORTANT!

Before making any adjustment to this setting, it is recommended that you read the descriptions of the menu item that selects the AES/EBU output sampling rate immediately preceding this entry.

Clock for the AES/EBU digital audio output may be derived from three sources. The standard 32kHz, 44.1kHz, and 48kHz sampling frequencies may be internally generated, supplied from an external source, or derived from the AES/EBU digital audio input signal. When the "native" sample rate is acceptable, (e.g., identical to the sampling rate of Zephyr's codec section) the sample rate converter may be bypassed.

Selection of AES/EBU output sampling rate is accomplished by choosing the item in the UTILITY menu and using the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

Available modes are:

- NO CONVERT

  When this mode is selected, there is no sampling rate conversion of the AES/EBU digital audio output. It will be identical to the main sampling rate used by the Zephyr codec section. This selection, of 32kHz or 48kHz, is made elsewhere in the UTILITY menu.

- 32kHz

  When this mode is selected, the sampling rate of the AES/EBU digital audio output is converted to 32kHz. Derived from the network (Telco) clock.

- 44.1kHz

  When this mode is selected, the sampling rate of the AES/EBU digital audio output is converted to 44.1kHz. Generated by a local crystal oscillator.

- 48kHz

  When this mode is selected, the sampling rate of the AES/EBU digital audio output is converted to 48kHz. Derived from the network (Telco) clock.

- EXTERNAL

  When this mode is selected, the sampling rate of the AES/EBU digital audio output is converted to match the external clock provided at the SYNC- IN connector on the Zephyr's AES/EBU digital input/output interface module.

- AES IN

  When this mode is selected, sampling rate conversion is performed to match the external clock received from the AES/EBU input signal.

Auto Answer   Auto Answer activation

Zephyr's internal terminal adapter is capable of automatically answering incoming ISDN data calls. Calls received from another Zephyr or Zephyr- compatible device will cause your local Zephyr to automatically answer.

Activation of Auto Answer function is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons to change the mode displayed. Confirmation of the change is not required.

The usual operating mode is for this item to be set to YES, as you want the Zephyr to respond to incoming calls by immediately connecting them. When NO is selected and a call rings in, a ringing sound will be heard and a screen appears with the call's originating number (caller ID), if known, and options to permit manual answering of the call.

Loopback   Loopback activation

(This is an advanced feature and may be ignored by most users.)

Activation of loopback function is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons to change the mode displayed. Confirmation of the change is not required. For detailed information see Section 10 (Advanced Problem Solving).

Available modes are:

- Off

- Far: The far loopback mode is "bilateral" or two- way. It loops the local audio directly from input to output. At the same time, the distant end gets it's signal returned via the Zephyr in Far loopback mode.

HOT TIP!

When the Far loopback function is ON, the meters on the Zephyr are inactive.

- Near: Incoming audio gets coded in the usual manner, sent into the ISDN interface card and from there looped back to the decoder where it is decoded. Allows a check of over 90% of Zephyr's functionality with no ISDN line present

> HOT TIP!
>
> Near loopback is only available on units with ISDN interface card installed. If no ISDN interface is installed "hardware not available" will be displayed.

> HOT TIP!
>
> If your friends want to hear the how good layer III sounds, you can use this mode to demonstrate the Zephyr without an ISDN line being present.

NOTE: Compatible Xmt and Rcv modes will be required.

These modes are intended as a simple debugging aid. For additional information refer to manual section 10 (Advanced Problem Solving)

Status Out   Status available on logic output

(This is an advanced feature and may be ignored by most users.)

Zephyr provides a logic output that serves an in- use indicator when the Zephyr is in use and as an alarm when the connection is terminated. This output (appearing as an open-collector closure to ground on the parallel control connector on the back) can be active in the user's choice of states depending on how you use your Zephyr. You may chose to use this closure at the time of connection or disconnection to trigger a signal or to start or stop an audio recorder or other device. The status is indicated when either network interface (V.35 or ISDN) is used.

For pin outs & electrical specifications of this output see manual section 3 (Zephyr at a Glance)

Selection of status of the logic output function is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons to change the mode displayed. Confirmation of the change is not required.

Available modes for activation of the logic output are:

- Rcv lock

  When this mode is selected, the logic output will be active (closed- to- ground) when the receiver is locked. This indicates that a coded digital audio signal compatible with the selected receive mode is being received and decoded by the Zephyr.

- Line 1

  When this mode is selected, the logic output will be active (closed- to- ground) when only Zephyr's first data channel is connected. There is no indication of whether audio is being decoded.

- Line 2

  When this mode is selected, the logic output will be active (closed- to- ground) when only Zephyr's second data channel is connected. There is no indication of whether audio is being decoded.

- Line 1&2

  When this mode is selected, the logic output will be active (closed- to- ground) when both of Zephyr's data channels are connected. There is no indication of whether audio is being decoded.

- Line 1or2

  When this mode is selected, the logic output will be active (closed- to- ground) when either (or both) of Zephyr's data channels is connected. There is no indication of whether audio is being decoded.

Store Setup	Auto-Dial setup entry- Setup selection

NOTE: Unlike other menu items, the Store Setup entry and the Category entries work together to allow entering multiple items. See the step by step directions which follow.

Zephyr allows storage of up to 50 auto- dial set ups. All relevant Zephyr settings are stored, including transmission and sampling rates, send (XMT) and receive (RCV) coding schemes, and the ISDN phone number(s) of the remote location(s) you wish to dial. Auto- dial setups are created in the fourth UTILITY menu. *This menu's name STORE SETUP should serve as reminder that whenever accessing this menu the setup created will include the current setup information from the first utility menu* (i.e. Xmt mode, Rcv mode, and rates). When you want to use a setup, do so from the AUTO- DIAL menu.

This menu item selects the number of the setup you wish to create or modify. The setups are numbered sequentially from one to 50. While the stored setups are primarily used with the internal terminal adapter, it can be used with the V.35/X.21 network interface to change the transmission and sampling rates and send (XMT) and receive (RCV) coding schemes

Selection of the auto- dial setup number is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons to raise or lower the number displayed. Confirmation of the change is not required.

To enter the parameters of the setup, go to the next item in this UTILITY menu, CATEGORY. See step- by- step procedure following.

Category	Category of Auto-Dial setup information to be entered

NOTE: Unlike other menu items, the Store Setup entry and the Category entries work together to allow entering multiple items. See the step by step directions which follow.

There are three parameters of the auto- dial setup that are modified from this menu item; the name of the setup and the two phone numbers at the remote location. As more than one item is entered, we recommend that you read all of the following step- by- step instructions later in this section before entering any data.

IMPORTANT!

The setup you are about to create includes the send (XMT) and receive (RCV) coding schemes and transmission and sampling rates in their present states in the Zephyr UTILITY menu. We strongly recommend that you first return to the top of the UTILITY menu to verify these settings, which are found in the first three UTILITY menu items, prior to entering information in the CATEGORY item.

HOT TIP!

Be certain that the STORE SETUP number selected above the CATEGORY menu item is the auto-dial setup you want to modify. This will avoid accidental erasure or modification of an important setting!

Creating a new setup for use with Auto dial

1. Verify Xmt, Rcv, Bit Rate, and Sample rate are set as desired for the setup about to be created. After all, there's no point in dialing and connecting automatically if your modes are automatically changed to something other than what you require!
2. Press the <UTIL> button until the STORE SETUP screen is seen.
3. Select the STORE SETUP number to be entered or modified using the instructions above.
4. Select the CATEGORY item in the UTILITY menu by using the <SEL> buttons.
5. Use the <YES (+)> and <NO (-)> buttons to select the category you want to modify. In order, these items are:
   - **NAME** Name of the Auto- Dial Setup
   - **NUM1** First Number of the Auto- Dial Setup
   - **NUM2** Second Number of the Auto- Dial Setup
6. After each of the above is selected press the <SEL DOWN> button to move to the entry field.
7. Use the keypad to enter the desired data (see below).
8. Store the desired data with a confirming press of the <YES (+)> button.
9. Press the <SEL UP> button to move back up the CATEGORY menu item.

HOT TIP!

Setups can be stored without any phone numbers. Such setups can be useful to leave the Zephyr in the correct modes to accept an incoming ISDN call or to be in the correct modes when using the V.35/X.21 network interface.

HOT TIP!

Some ISDN lines are configured to have the same telephone number for both channels. If you simply put the same number into both setup number fields, it probably won't work. (This is because the Central Office thinks that the Zephyr has erroneously sent the same setup twice.).

However, we have discovered a work-around that seems to get the job done, in most cases. When you are storing a setup for such a destination, enter the number in both NUM1 and NUM2, but put the symbol "#" after one of them. The Central Office will discard the extra digit and then properly connect both channels

10. Repeat steps four through eight until all three categories have been entered.

Additional considerations when entering CATEGORY information are:

- **NAME** Name of the Auto- Dial Setup

  The dialing keypad is used to enter the alphanumeric name text. Each press cycles through all of the characters printed on the button cap. When the desired character is displayed, press the <YES (+)> button to accept the character and advance to the next position.

  For example, the "2" button is used to get the characters "A," "B," "C," and "2". The "1" button can be used to get "Q," "Z," "1," and a blank space. Special characters may be entered as when needed. "*" and "#" my be entered from their corresponding buttons and minus, colon, and slash ("- ", ":", and "/") may be entered using the "O" button.

  After the character you desire is displayed, you will be prompted to press the <YES (+)> button to accept it. The <NO (-)> button backs up (to the left) and deletes characters for editing.

  Do not forget to confirm and store the name of the setup by pressing the <YES (+)> button a second time after you enter the final character of the name. You will be prompted to do so by the fourth (bottom) line of the LCD screen.

- **NUM1** First Number of the Auto- Dial Setup

- **NUM2** Second Number of the Auto- Dial Setup

  The dialing keypad is used to enter the numbers. The <NO (-)> button backs up (to the left) and deletes characters for editing.

Do not forget to confirm and store the numbers by pressing the <YES (+)> button. You

HOT TIP!

If you are changing an old setup, and you don’t erase a left-over and undesired second number, the system will try to dial it. You must erase any such number you don’t need by backspacing to clear it out.

will be prompted to do so by the fourth (bottom) line of the LCD screen.

Changing the mode information for a given setup

1. From the first UTILITY menu set the XMT, RCV, and RATE information as desired for the edited setup.

2. Press <UTIL> until STORE SETUP is displayed

3. Use the <NO(-)> and <YES(+)> keys to scroll to the desired setup

4. Push the <SEL DOWN> key twice to get to the third line where the name is displayed.

5. Push <YES(+)> key to re- store the setup with the new mode information.

SPID 1 & 2 / (MSN 1 & 2) SPID or MSN entry fields when required

This menu item is only used when Zephyr has the internal ISDN terminal

ISDN TIP!

European users should disregard all references to SPIDs. Euro ISDN *does not* have SPIDs! If you ISDN configuration requires MSNs they may be entered in the MSN/SPID 1 & 2 screen.

ISDN TIP!

If you have PTP (AT&T Point-to-Point) or Euro-ISDN Telco mode, you should not enter anything in the SPID fields.

ISDN TIP!

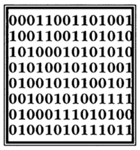
In an emergency, to save a remote, you can sometimes get a line working without the proper SPIDs by dialing into it. This fix is only temporary and the line will fail to initialize next time the Zephyr is booted.

adapter installed. This screen will read “MSN 1 & 2” if the “Telco” menu item is set to “ETS 300”.

In the USA, the most common ISDN service is National ISDN- 1. This service requires that the terminal unit, in this case your Zephyr, provide SPID (Service Profile

IDentification) numbers to the telephone network. These numbers are provided by the telephone company and are entered in this menu item.

Below the SPID 1 & 2 heading are two entry fields. Select the first field (immediately below the heading) by using the < SELECT ▲ > and < SELECT ▼ > buttons. Enter the

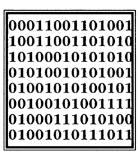


> HOT ISDN TIPS ON SPIDs!
>
> 1. Incorrectly entered SPIDs are the most common problem USA users have placing ISDN calls. Compare the SPIDs you have entered with the SPIDs provided to you on by the phone company to be certain that you have entered them correctly. Do not add anything to your SPIDs.
>
> 2. Note that there are no hyphens or dashes (-) in SPIDs. If your installer has included them in the SPID, ignore them.
>
> 3. For a list of known working SPIDs by Telephone Company see the appendix
>
> 4. If you have any questions about your SPIDs, call your phone company.

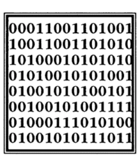
SPID number of your first line. The dialing keypad is used to enter the numbers. The <NO (-)> button backs up (to the left) and deletes characters for editing.

Confirm and store the first SPID number by pressing the <YES (+)> button.
You will be prompted to do so by the fourth (bottom) line of the LCD screen.

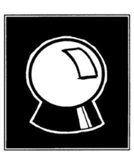
Press the < SELECT ▼ > button to move down to the second SPID entry field. Enter the



> ISDN TIP!
>
> If the ISDN line is already connected, you will have to re-initialize the ISDN by pressing <YES> at the Utility menu's Telco menu item, or by powering the Zephyr down and turning it on again, or by un-plugging and reconnecting the ISDN line.



> ISDN TIP!
>
> If you see WAIT as your line status your SPID is incorrect. In this case check the SPID. For a list of known working SPIDs by telephone company see the Appendix.



> CURIOSITY NOTE!
>
> There is hope on the horizon that SPID difficulties will get better. For one thing, the Telcos are beginning to standardize on area code+phone number + 0101 for SPIDs on National ISDN lines.
>
> Standards for future versions of National ISDN will include automatic SPID assignment/selection and Non-initializing terminals which would be allowed to operate without a SPID, albeit with only basic functionality.
>
> Telos Systems is monitoring these developments carefully.

SPID of your second ISDN channel as above. Don't forget to confirm and store!

Directory 1 & 2 Directory number entry fields when required

IMPORTANT!

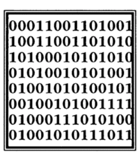
Directory numbers are rarely needed in the USA and never needed with European ISDN. Entering Directory numbers when you do not need them may prevent you from properly receiving ISDN calls.

This menu item is only used when Zephyr has the internal ISDN terminal adapter installed.

`Most users will leave these fields blank.` They are used in the very rare circumstance that the SPIDs do not incorporate the local ISDN phone numbers. If you feel you have such service, contact Telos Customer Support for verification before entering anything in this menu item.

Should Directory numbers be required, they are entered in a manner identical to the entry of SPIDs detailed above. The Directory number is your 7 digit telephone number (no area code) as would be found in the telephone directory.

Telco ISDN protocol selection



ISDN TIP!

In some circumstances, the Zephyr main display may provide the false indication that Zephyr is READY to place and receive ISDN calls when the wrong ISDN Telco type is entered. Please be certain to make the correct selection. If you are not certain of the ISDN type, contact your phone company.

This menu item is only used when Zephyr has the internal terminal adapter installed. It selects the type of ISDN network to which you are connected.

Selection of ISDN Telco type is accomplished by choosing the item in the UTILITY

HOT TIP!

Pressing <YES> at this menu item causes a full reset of the ISDN connection.

If you are trying different SPIDs, for instance, this can be used to send them to the network.

menu and using the <NO (-)> button to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

Available modes are:

- `Natl I-1` National ISDN- 1 Protocol

  This mode is for most users in the USA and for other users of the USA National ISDN protocol. It supports National ISDN- 1, National ISDN- 2, and nearly all other

USA protocols with the exception of AT&T's Point- to- Point and Point- to- Multipoint protocols. (Note that Zephyr does not support the AT&T Custom Point- to- Multipoint ISDN protocol. If you have this protocol, you will need to have your ISDN service modified. Contact Telos Customer Support for details.). This is also the correct selection if you have DMS- 100 custom ISDN.

- `PTP` AT&T Point- to- Point (Custom) Protocol

  This mode is to be used only by USA users who have AT&T Point- to- Point protocol. Other users should not make this selection.

- `ETS300` European ISDN Protocol

> 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?) they may have given you a 2 B-channel National ISDN line with only one SPID and one phone number. This may work, however you will need to proceed as follows:
>
> 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

> 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?) they may have given you a 2 B-channel AT&T "Custom Multipoint" line with 2 phone numbers and 2 SPIDS. Zephyr does not support this configuration. However, in an emergency, to save a remote, you can usually get things working (temporarily) as follows:
>
> 1) Remove both SPIDS and both directory numbers
>
> 2) Set Telco to PTP
>
> 3) Have another Zephyr call either of your phone numbers twice. They should be able to connect.
>
> 4) The line will now function normally for outgoing calls until Zephyr is rebooted or the ISDN connection is reinitialized.

This mode is to be used only with the European ETS- 300 ISDN protocol, known as Euro- ISDN.

Panic Dial   Panic Dial activation and setup selection

(This is an advanced feature and may be ignored by most users.)

This menu item is only used when Zephyr has the internal terminal adapter installed.

Zephyr provides a logic input that can be used with your external equipment to automatically dial and connect to a frequently called or emergency number. This input (appearing on the Parallel Control connector on the back) can alternately be used as one of the four end- to- end logic closures. For pin outs, see manual section 3 (Zephyr at a Glance).

This menu item activates this function and selects the auto- dial setup you want to use for the panic dial function. Available auto- dial setups are those stored in addresses one through 50

When activated, the panic dial routine takes over and over rides any current calls. Current call(s) will be dropped and panic dial will initiate a dialing sequence and attempt to call the designated number(s). The panic dial routine will continue to maintain this connection despite line problems until the panic dial input is no longer asserted.

Configuration of the panic dial mode is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons. When the display reads "NO," the panic dial mode is inactive. When set to any number between one and 50, the panic dial number is the corresponding auto- dial setup. Confirmation of the change is not required.

Alternatively, if this option is set to "1 - 4" each of the 4 logic inputs 0- 3 will trigger setups # 1 - 4. For pin outs see section 3 (Zephyr at a glance).

HOT TIPS!

Users have found the following "Panic Dial Tricks" handy:

1. Mode reset. Program the stored setup(s) to be used without any phone numbers. Name them after the associated modes stored with the setup(s). You can now use push a momentary switch connected to the panic dial input(s) and Zephyr will switch to a designated mode ready for an incoming call.

2. Drop and reset. Same as above. Since panic dial takes precedence over standard calls, you can push this momentary switch and drop a call and return to a default configuration (as programmed into the associated setup) ready for an incoming call.

Some users have done this with a timer that automatically closes the contacts a few times a day. While the potential drawbacks are obvious, this can prevent 48 hour phone calls over the weekend!

3. Quick dialer. Get some push button interlocking switches (5 by 1 pole, single throw, interlocked such that only one can be pressed at a time). Leave one disconnected and connect the other 4 to the 4 logic inputs. Set panic dial to "1-4". You can now call any of the first four autodials by pressing the appropriate button. Pushing the disconnected button would drop the call. When used in conjunction with an LED connected to the Status Out this makes a slick (but limited) remote control interface.

One application of this is to allow the Zephyr to be used in an automatic back-up application. The alarm output of a satellite receiver, for instance, may cause an automatic connection to a Zephyr at the source. A similar idea would be to use ISDN to back-up a broadcast RF Studio-to-Transmitter-Link system. Remember, the panic dial input must remain asserted until you desire the call to be dropped.

Compatibility Mode

(This is an advanced feature and may be ignored by most users.)

Currently effects LII encode mode only.

This feature is used by companies that offer subscription services or allow connection to their facility only to members who are enrolled in a membership program.

Unless you participate in one of these services, this menu item does not effect the operation of your Zephyr. Most Zephyr users can ignore this menu item. Future options may include special modes for connecting to non- standard codecs.

Also has an option to enhance compatibility with the Dialog4 family of codecs.

LCD Contrast LCD contrast adjust

This menu adjusts the LCD contrast. It has a range from one to nine, although settings below five may not be visible in a typically lit room.

Changing the LCD contrast is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons to raise or lower the number displayed. Confirmation of the change is not required.

LCD Backlight LCD backlight adjust

Changing the LCD backlight brightness is accomplished by choosing the item in the UTILITY menu and using the <YES (+)> and <NO (-)> buttons to raise or lower the number displayed. Confirmation of the change is not required.

Ancil Chan Ancillary data transmit channel

(This is an advanced feature and may be ignored by most users.)

The ancillary data stream containing the contact closures and RS- 232 data can be sent out with one or both of the 2 data streams corresponding to Line 1 or Line 2.

Version Info Details currently installed firmware

The final item on the UTILITY menu is a read- only display. The display contains information you will need to provide Telos Customer Support should you call for assistance.

On the third line, three numbers provide complete version numbers of the firmware installed. The first number is the system firmware, the second is the ISDN firmware, and the third is the DSP and XILINX firmware.

The fourth line of the display has a date and code number. These refer to the firmware and do not reflect the date of manufacture of your Zephyr.

Please have this information if you call Telos Systems customer support.

Utility Menu - The Complete Works

Users have asked for a list of the Utility screens to make things easier when walking someone through the menus over the phone. For information on what each *menu item means* see the chart Utility Menu - Summary of Utility Menu Items earlier in this manual.

Menu Items	Options
Xmt	L3 DUAL
	L3 JSTEREO
	L3 STEREO
	G.722
	L2 HALF/24
	L2 MONO
	L2 MONO128
	L2 DUAL
	L2 JSTEREO
	-
Rcv	L3 STEREO
	L3 MONO
	G.722
	L2
	L2 HALF/24
	-
Rate	56Kbps @ 32kHz
	64Kbps @ 32kHz
	56Kbps @ 48kHz
	64kbps @ 48kHz

Network	ISDN
	V.35
	-
AES In	NO (ANALOG)
	S/R CONVERT

	SYNC TO NET
	-
AES Out	NO CONVERT
	32 kHz
	44.1 kHz
	48 kHz
	EXTERNAL
	AES IN

Auto Answer	YES
	NO
	-
Loopback	OFF
	NEAR
	FAR
	-
Status Out	RCV LOCK
	LINE 1
	LINE 2
	LINE 1 & 2
	LINE 1 OR 2

Store Setup	1 ... 50
Category	NAME
	NUM1
	NUM2

SPID 1 & 2 /MSN 1 & 2	SPID/MSN entry fields when required

Directory 1 & 2	Directory number entry fields (rarely required)

Telco	Natl I-1
	ETS300
	PTP
	-
Panic Dial	NO
	1 ... 50
	1-4
Compatibility	Used only for certain subscription services

LCD Contrast	1 ... 9
LCD Backlight	1 ... 9
Ancill Chan	Normal
	None
	Right
	Both
	-
Version Info	2.69/2.00/2.32 <for example>
	-

Dial Menu

This menu is used to manually dial using Zephyr's internal terminal adapter (ISDN Network Interface) and functions only when the terminal adapter is installed and the Utility menu Network option is set to ISDN.. The number of the receiving party is entered and dialing is initiated from this menu. Selecting whether you are placing a high- fidelity call via ISDN or a voice- grade call via Plain Old Telephone Service (POTS) is performed using a sub- menu called DIAL OPTIONS.

Access this menu by pressing the <DIAL> button on the front panel of the Zephyr. The top row of the LCD display indicates the type of call to be placed as either ZEPHYR (high- fidelity audio to another switched data line) or PHONE (to a standard Plain Old Telephone Service (POTS) phone or an ISDN line with Circuit Switched Voice Service.). The display also shows which of the ISDN lines is to be dialed.

A press of the <DIAL> button is ignored, and it is not possible to access this menu, when no ISDN line is connected and ready; also when both lines are already in- use.

At this point, you have three options, which are prompted by the fourth (bottom) row of the LCD display:

1. Press the <NO (-)> button to access the DIAL OPTIONS sub- menu to change the call type or line to be dialed. When the button is pressed, you will see the sub- menu (described below).

2. Press the <YES (+)> button to recall the last number called on the selected line. When the button is pressed, the last number will appear between the brackets on the second row of the LCD display. The <NO (-)> button backs up (to the left) and deletes numbers for editing. Numbers may be added from the keypad.

1. Manually enter the number you want to call. The <NO (-)> button backs up (to the left) and deletes characters for editing.

> HOT TIP!
>
> When using ETS300, you may enter the receiving party's sub-address. To do so, dial the primary number, then the star (*) key, and then the sub-address. This feature is not available with National ISDN or PTP.

After making any changes in the DIAL OPTIONS menu and recalling or entering the phone number, press <DIAL> again to initiate the call. Call progress is shown on the associated line status area of the LCD display. To drop a call, use the DROP menu.

Line status/Call status states are:

`inact:` No line connected

`init:` Before sending SPID to ISDN network, or after seeing external NT1

`wait:` SPID has been sent to network

`Ready:`	SPID has been accepted and everything is OK
`ConnW:`	Outgoing call progress; waiting for line.
`ConnP:`	Outgoing call in progress; call proceeding
`ConnR:`	Outgoing call ringing
`0:00`	Call connected
`Disc:`	Call disconnecting

DIALING OPTIONS SUB-MENU

This sub- menu, accessed from the DIAL menu, is used to change the call type and select the ISDN line (1 or 2) to be dialed. Zephyr defaults to high- fidelity “Zephyr” calls to switched digital lines, such as ISDN, and to initiate calls from line 1 first.

To access the DIALING OPTIONS sub- menu, press the <DIAL> button followed by the <NO (-)> button. If you press any button between pressing <DIAL> and <NO (-)>, you will not be able to access the sub- menu. If this happens, press <HELP> twice to escape and start again.

To select one of the two dialing options in the sub- menu, use the < SELECT ▲ > and < SELECT ▼ > buttons to choose the menu item you want to change. The <NO (-)> button is pressed to change the mode displayed. The <YES (+)> button must be pressed to accept and store your selection.

The dialing options are:

- Call Type

Zephyr can place a high- fidelity call to an ISDN, Switched 56, or other switched digital circuit on one of its channels while placing a standard grade call to a POTS line on the other channel. Refer to the manual section on ordering ISDN service if you want this capability as it requires special ISDN line provisioning.

Available call type options are:

- ZEPHYR

  This mode is selected when the receiving line is an ISDN, Switched 56, or other switched digital circuit. High- fidelity coded digital audio will be transmitted to be decoded by the receiving party. The local Zephyr may, in turn, simultaneously decode the signal from the remote site.

- PHONE

  This mode is selected when the line to be called is an ordinary POTS line. The audio quality is that of a standard telephone call. While the remote party may be monitored from the front panel speaker, audio is sent to the remote site via the standard, rear- panel connections.

- ISDN Line

Zephyr defaults to line 1 if available or line 2 if line one is in use. This menu item selects the ISDN line to be used for the call.

Available options are:

- 1
- 2

Auto-Dial Menu

This menu is used to place calls to frequently accessed numbers stored as auto- dial setups and functions only when the terminal adapter is installed. While you can verify the recipient's number, codec settings, and transmission rates from the AUTO- DIAL menu, you cannot make changes. All changes to the auto- dial setups are accomplished from the UTILITY menu.

Access this menu by pressing the <AUTO- DIAL> button on the front panel of the Zephyr. To place a call:

1. After entering the AUTO- DIAL menu, select the number of the stored auto- dial setup you want to dial using the <YES (+)> and <NO (-)> buttons until the number appears on the top row of the LCD display. You can confirm that you have selected the correct destination, as the name assigned to that site is displayed on the third line of the LCD display.

   Or alternatively, enter the 2 digit number of the desired auto dial using the keypad.

2. Press the <AUTO- DIAL> button. Call progress is now displayed.

> HOT TIP!
>
> Some ISDN lines are configured to have the same telephone number for both channels. If this is the case your auto dial must include the same number twice. In this case one of the numbers will need to have a “#” added to it so that the Telco switch will not reject the call. For additional details see instructions for Utility Menu - Store Setup, above

> HOT TIP!
>
> Setups can be created and stored using the UTILITY menu without phone numbers. Such setups can be useful to leave the Zephyr in the correct modes to accept an incoming ISDN call or to be in the correct modes when using the V.35/X.21 network interface. Use the two steps above to leave your Zephyr in such a state. In this case Zephyr will display “Changing Modes” on the bottom of the screen.

CURIOSITY NOTE!

The time to connect when using the Auto Dial function will appear to vary. The reason for this phenomenon is that Zephyr first must set the Xmt, Rcv, and Rates information and this can take up to 10 seconds. If this information is already set the same as the stored setup, then Zephyr immediately dials the call.

The AUTO- DIAL menu allow you to confirm all parameters of an auto- dial setup by using the Display menu item. To access the Display menu item, use the < SELECT ▼ > button. The parameter displayed, on the third row of the LCD display, is changed by pressing the <YES (+)> and <NO (-)> buttons. Parameters that may be displayed are:

- Display Options
  - NAME

  This displays the name of the auto- dial setup currently selected.

  - NUM1

  This displays the first of the recipient's numbers used by the auto- dial setup currently selected.

  - NUM2

  This displays the second of the recipient's number used by the auto- dial setup currently selected.

  - MODE

  This displays the transmit encode scheme, the receive decode scheme, the transmission rate, and the sampling rate used by the auto- dial setup currently selected.

Drop Menu

CURIOSITY NOTE!

The status of the Drop options will vary depending on the current Xmt and Rcv modes. If both Xmt and Rcv are set to a STEREO mode, it will display BOTH. Otherwise it will display LINE 1 if both lines are in use.

From this menu, calls placed using the DIAL and AUTO- DIAL menus are terminated and functions only when the terminal adapter is installed.

Access this menu by pressing the <DROP> button on the front panel of the Zephyr. You may choose to drop one or both lines of the call. The top row of the LCD display shows the line or lines about to be dropped. To proceed as indicated, press the <DROP> button again. To change the line(s) to be dropped, press the <NO (-)> button until the desired line(s) is displayed and then press the <DROP> button.

A <DROP> button press is ignored, and it is not possible to access this menu when no lines are active.

Line choices in the DROP menu are:

- BOTH

  When selected and confirmed by pressing the <DROP> button, both ISDN lines are dropped.

- LINE 1

  When selected and confirmed by pressing the <DROP> button, the first ISDN line is dropped. The second line is unaffected. If you are transmitting or receiving coded audio that requires two channels (e.g., L3 JSTEREO) audio transmission/reception will cease.

- LINE 2

  When selected and confirmed by pressing the <DROP> button, the second ISDN line is dropped. The first line is unaffected. If you are transmitting or receiving coded audio that requires two channels (e.g., L3 JSTEREO) audio transmission/reception will cease.

Volume Menu

Summary of Volume Menu Items

Headphones	Volume for the Phones jacks
Phone Speaker	Volume for the small front-panel loudspeaker
Send	Send audio level

Volume Menu Item Details

Headphones

This menu item adjusts the volume at the headphone jack on Zephyr. It has range from one to 10.

Changing the headphone volume is accomplished by choosing the item in the VOLUME menu and using the <YES (+)> and <NO (-)> buttons to raise or lower the number displayed. Confirmation of the change is not required.

Phone Speaker

This menu item adjusts the volume of the phone speaker, located on the front panel behind the dialing keypad. This speaker provides a tone when ISDN channels are connected or dropped. Additionally, it can be used to monitor calls with POTS phones, although you must use the rear panel input to send audio down the telephone line. It also provides a ringing- in sound when auto- answer is off.

The phone speaker has a range from zero to 9. A setting of zero turns the speaker off. Turning off the phone speaker also disables the call status sounds which indicate calls have been rejected, completed, etc.

Changing the phone speaker volume is accomplished by choosing the item in the VOLUME menu and using the <YES (+)> and <NO (-)> buttons to raise or lower the number displayed. Confirmation of the change is not required.

Send

This menu item adjusts the nominal input level of the audio sent from the local Zephyr to the remotely connected unit. Level is first nominally set to microphone (MIC) or line level using the button on the rear panel between the analog send audio connectors.

The SEND menu item controls a 20dBu attenuator and displays a range from +4dBu (most attenuation) to - 15dBu (least attenuation).

HOT TIP!

This menu item displays the nominal input level. As you lower the value, the input level is raised. This is demonstrated by the activity of the SEND meters. See the Appendix for additional information on levels.

Changing the send audio input level is accomplished by choosing the item in the VOLUME menu and using the <YES (+)> and <NO (-)> buttons to raise or lower the number displayed. Confirmation of the change is not required.

Help Menu

The HELP menu may be accessed at any time, from the UTILITY, DIAL, AUTO- DIAL, DROP, and VOLUME menus. When the <HELP> button is pressed, the LCD display provides information on the item that was previously displayed. When HELP information exceeds the available space on the LCD screen, press the < SELECT ▼ > button to show more information. The < SELECT ▲ > button may be used to review.

To return from the menu in which you were working, press the appropriate menu button. Pressing the <HELP> button a second time serves an “escape” function that returns the LCD to Zephyr’s main screen.

The HELP menu may also be accessed from the Zephyr main screen. From this location, it provides an explanation of the display information on the main screen and some basic trouble- shooting tips.

The HELP menu has three special functions. These are activated by a button from the keypad while the help screen for the status screen is displayed. The special functions are:

Warm Boot

A warm boot re- initializes the codec and network interface sections of the Zephyr. This performs the same function as powering the unit down and up again. If you connect additional equipment to your Zephyr, modify the ISDN protocol, change the SPIDs, or perform a similar activity, a warm boot may be necessary.

To warm boot Zephyr, press the hash/pound (#) button after accessing the “Line Status” help screen. It may take a few moments for the warm boot to start. Once started, the will LCD display will show the initialization screen.

Cold Boot

A cold boot re- initializes the codec and network interface sections of the Zephyr. This also *erases all of your battery- backed- RAM settings*, including SPIDs, ISDN protocol, and auto- dial setups, and restores the factory defaults. A cold boot is a drastic measure and should only be performed if absolutely necessary.

> IMPORTANT!
>
> A cold boot erases all of your battery-backed-RAM settings, including SPIDs, ISDN protocol, and auto-dial setups, and restores the factory defaults. It can sometimes solve unexplained transient problems, so it may be worth attempting. However, a cold boot is a drastic measure and should only be performed if absolutely necessary. There, we said it twice!

To cold boot Zephyr, press the star (*) button after accessing the “Line Status” help screen. You will be warned of the action about to take place. If you wish to proceed with the cold boot push the hash/pound (#) button. It may take a few moments for the cold boot to start. Once started, you will LCD display will show the initialization screen.

After a cold boot, you will need to completely reconfigure your Zephyr.
Yes, everything.

RS-232 Initialization

This special function sends AT commands to initialize a modem connected to the RS-232 remote port.

To perform this function, press the number below on the keypad after accessing the “Line Status” help screen. The Zephyr display does not change.

Available options are:

Key	Function
1	Send modem initialization string
*	Immediate coldboot (Erase all parameters to factory defaults).
#	Immediate warmboot.

Special Start-up Functions

(These are advanced features and are not needed by most users. They are listed for reference only, but not fully documented.)

Some special functions are enabled by button presses during power- on start- up. These must be pressed while “Initializing...” is displayed on the last LCD line.

Key	Function
1	Serial port to 2400 bps, command mode, auto transparent mode off.
2	Serial port to 9600 bps, command mode, auto transparent mode off.
3	Serial port to 19200 bps, command mode, auto transparent mode off.
4	Serial port to 2400 bps, transparent mode (Ancillary data mode)
5	Serial port to 9600 bps, transparent mode (Ancillary data mode)
6	Serial port to 19200 bps, transparent mode (Ancillary data mode)
8	Special CRC mode.
9	Activate NRW mode.

SECTION 9

REMOTE CONTROL

CONNECTING TO A COMPUTER

The RS-232 serial port can be used for computer control of the system. Full status and control are available via this port, including ISDN dialing, storage of set-ups, etc.

The communication protocol is simple ASCII text, so it may be used easily with any computer and terminal emulation software. Some user's are using a palm top computer with macros to control their Zephyrs in the field

Some "terminal emulation" software possibilities are:

- `For the PC`: Crosstalk, Procomm, MS Windows built-in Terminal or HyperTerminal, etc.
- `For the Macintosh`: Z-Term, White Knight, etc.

The Zephyr may be connected locally to a personal computer, or to a modem for control from a remote site. There will be a difference in the required cabling, as described in the Installation section. The Zephyr serial port is configured as DTE (Data Terminal Equipment), which means it looks like the connection on a computer, rather than on a modem.

- With local computer operation, a null modem cable is used.
- With a modem connection, a standard (non-null) cable is required.
- Or a custom Zephyr-to-computer cable may be fabricated, as follows:

Zephyr Female DB-9 Pin	Signal Direction	Computer Female DB-9 Pin
5		5
2	<--	3
3	-->	2

Remote Control: Step-by-Step

(See later in this section for information on using the Windows ZephyrControl software)

Physical connection accomplished, the remaining steps are:

- `Step 1`. Start your terminal program, set the correct COM port and select 8N1 (8 bits, no parity, 1 stop bit), 9600 baud, full- duplex for the starting communication parameters.

- **`Step 2`**. Switch on your Zephyr.

Assuming a functioning RS- 232 link, some basic information will be presented during the power- up initialization sequence:

```
Hello world !!

Copyright 1993,94,95,96 TLS Corp. All rights reserved.

V2.68/2.00/2.25, June 15 1996, #120

Warm boot - please wait.....

System is up.
```

If you do not get any text, the problem is with either:

  - The cable.

  - The terminal program basic functionality.

  - Terminal program baud rate settings. If you have the latter problem, you will probably see some “garbage” text. This could happen if the Zephyr was set to a different baud rate and left in that state. In that case, move ahead to the next step.

  - COM port selection (at the computer).

  - Maybe the Zephyr is in “transparent mode.” Send “+++” or ### to return to command mode.

- `Step 3.` Press <RETURN> on your computer. The Zephyr echoes the data sent to it, so if you see the cursor moving down the screen or if you see the command prompt “>>“ you are communicating with the Zephyr; go on to Step 4.

  If the cursor doesn't move, meaning you are not getting echo, or you see garbage characters, the Zephyr is probably not set to 9600 baud. Try setting your terminal software to the other bitrates given in the **baud** command below and then type a few random characters. You will know you’ve found the correct value when the echo corresponds to your typing.

- `Step 4.` After a <RETURN> there will be a prompt, inviting a command to be entered: >>

SYNTAX USED IN THIS SECTION

In the following section we give a summary of the available serial port commands. Command descriptions used in the manual and using the "?" command use the following context. All commands must be followed by <enter|return> before any action will be taken with the exception of <+++> and <###>.

<table>
<tr><th>Symbol</th><th>Description</th></tr>
<tr><td>< ></td><td>Enter one of the arguments contained within</td></tr>
<tr><td>|</td><td>Or (choose one of the options)</td></tr>
<tr><td>-</td><td>Used in place of an optional entry; entry or - must be entered</td></tr>
<tr><td>[ ]</td><td>Optional item</td></tr>
</table>

COMMANDS AND FUNCTIONS

Log in. If there has been no password yet set – which will be the case the first time you access the system – type (user input in **`bold`**):

>>**`login <return>`**

`Welcome to ZEPHYR control system`

If you specified the wrong password, you will not see the Welcome line and the Zephyr will not accept any commands other than the login command.

After you finished your remote session type **`bye <return>`** to disconnect from the Zephyr.

DEEP TECH NOTE!

The purpose of the login procedure is to allow the Zephyr to be connected continuously to a modem and phone line without fear of "hacking."

Assuming a successful login, we're ready to enter commands. A command is entered by typing it and any required arguments, followed by a return.

You may get help on a given command by typing **`?`** followed by the command, or a listing of all commands:

HOT TIP!

A list current to the latest version may be had by entering a ? * command.

Not all of them may be accepted under all conditions, i.e. if the Zephyr is not equipped with an ISDN card, ISDN commands will not be available.

Examples:

`? volume` displays help about the volume command.

```
>>? volume
Command  : volume
Arguments: [H <1..10>] | [S <0..9>]
Task     : Set volume for headphones (H) or phone speaker (S)
```

`?` print a complete list of latest available commands.

```
>>?
? acc aesin aesout atdt autoboot baud bitrate bye cc ccmask
coldboot crc conn direc disc level
log login loop network panic passwd reset rxmode setup spids stat
statout time telco txmode
ver volume
Type '/' to repeat, '.' to edit last command
Type '? <command>' to get specific help
Type '? *' to get detailed list of commands
```

Catalog of Commands And Functions

The following is a alphabetical list of available commands and a description for each.

The indicated default values refer to the values after a cold boot.

? [topic|*]

Get general help or help about specific command or detailed command list.

acc <1+|1-|2+|2->

Accept (+) or reject (-) incoming call on specified ISDN line.

If the Zephyr is not in autoanswer mode an incoming call will be indicated on the screen and with a ringing sound. The user can then accept or reject the call coming in on a specific line.

Example:

`acc 1+` accepts incoming call on line 1.

`acc 2-` rejects incoming call on line 2

aes

Returns status of AES/EBU chips

`aesin <analog|conv|sync>`

Select AES/EBU input mode: use analog input (no AES/EBU), perform sampling rate conversion, assume AES/EBU input signal is sync to the CLOCK- OUT signal.

Default: analog

`aesout <normal|32|44|48|ext|aesin>`

Select AES/EBU output mode: do not convert, convert to 32, 44.1, 48 kHz sampling rate, convert to clock provided at CLOCK- IN pin, convert to clock retrieved from AES/EBU input.

Default: normal

`ancill <+|-|L|R|LR>`

Selection which with which audio channel (A/L or B/R) ancillary data will be sent when Xmt mode is set to dual. Parameters are: +- ancillary data will be sent to site receiving A channel (left) audio; - - disables ancillary data transmission, may be desirable in rare cases for maximum compatibility with other Layer III codecs; L- same as +; R- ancillary data will be sent to site receiving B channel (right) audio; LR- ancillary data will be sent to both sites. (May not be functional until rev 2.69)

Default is +

`answer <A+|A->`

Switch ISDN/phone auto- answer on/off.

`atdt [auto|no|E|[c]]`

Set RS232 to transparent mode (use '**+++**' or to quit).

Default: off

Before entering **+++** no data transmission must occur during a guard time of 1 sec.

In transparent mode, all RS232 data will be transmitted to the remote Zephyr by using MPEG audio's ancillary data feature. Ancillary data received from the remote Zephyr will be output to the RS232 interface.

Note that you cannot use Zephyr control commands while you are in transparent mode.

Note also that data mode does not work with G.722 or LII transmission paths.

The following parameters modify it's operation as follows:

AUTO (Default) allows automatic switching from local command mode to transparent mode when ancillary data is received

NO Disable automatic mode detection (see AUTO, above)

E Displays "escape" character to be used locally to get back into local command mode

E[c] Changes "escape" character to "**c**". Thereafter "**ccc**" will get back to local command mode `autoboot <+|->`

Switch on/off automatic boot on fatal errors.

Default: on

Under some very rare conditions the Zephyr will stop working and print out an error message. If autoboot is on the Zephyr will perform a warm boot after a few seconds to reenter operational status. While the error message is displayed you can press the # button at the front panel to cause a warm boot or the * button to cause a cold boot.

`baud <300,600,1200,2400,4800,9600,19200> [1|2]`

Set RS232 baudrate and stopbits.

Default: 9600 baud

Note that you will have to adjust the baudrate at your terminal program after changing the Zephyr's baudrate.

`bitrate <56/32|64/32|56/48|64/48>`

Select line bitrate/sample rate. (In kbps/kHz.)

Default: 56kbps/32kHz.

`bye`

Logout. Except **`login`**, no commands will be available.

`cc [<0000..1111>]`

Set contact closure output value (4 bits).

This command followed by a binary value sets the local Zephyr's parallel outputs to the value. If connected to a remote Zephyr, the values from that unit will override this value. Since an update is sent every 5 seconds, the entered value will not remain longer than this period.

`ccmask <0000..1111>`

Set contact closure input XOR mask (4 bits). This is used to flip the polarity of inputs to match what is available from the outside world. A value of 1 means the input is inverted from usual. When an input is not connected, this command may be used to simulate the input signal.

Default: 0000

`coldboot`

Initiate cold boot sequence (configuration will be lost).

During a coldboot ALL system parameters will be cleared. This includes setups, type of ISDN, SPIDs, directory numbers etc.

`compat <zephyr|dialog4|slimline>`

Select compatibility mode. Default is Zephyr. Used for enabling special modes for use with non- Zephyr codecs.

`conn <1|2> <number> or conn <1..50>`

Establish ISDN connection on line 1 or 2 or call setup.

Example:

`conn 1 2167819310` call a given phone number.

`conn 17` activates the setup #17. Transmit mode, receive mode and bitrate will be set according to the setup. If there are phone numbers stored in the setup they will then be dialed.

`crc <ISO|encrypt>`

Select bitstream CRC type.

Default: ISO

`direc <phone1|-> <phone2|->`

Enter or delete ISDN directory (phone) numbers.

Default: none

The directory number is the phone number of your ISDN lines without area code. You only need to enter the directory numbers if the SPIDs do not include the phone numbers.

`disc <1|2| 1 2>`

Disconnect ISDN on line 1, 2, or both.

`level <-15 .. +4>`

Select analog input level (dBu).

Default: +4 dBu

`lock [+][COLD|SETUP|AUTO|SPIDS|DIAL]`

When given from the serial port these commands can disable certain front panel functions. Since coldbooting would restore these function it is disabled when any lock mode is active. Use + to add a lock mode. When entered alone all lock modes are canceled.

Parameters are: COLD- locks out front panel cold boot capability; SETUP- locks out front panel access to store setups utility menu. (Stored setups cannot be added or changed, however autodial still works); AUTO- disables auto dial function; SPIDS- SPIDs and Telco utility menus no longer work from front panel; DIAL- disables manual dial function using dial button.

NOTE: The corresponding serial port commands will still function, only manual functionality is disabled.

`log [[+] SYS ISDN IMON SEXT TIME ITIM]`

Info logging: + = add specified log mode

SYS=general system info, ISDN=general ISDN info, IMON=ISDN monitor, SEXT=extended system info, TIME=time every 15 sec. ITIM=print time of ISDN events.

Multiple log modes can be specified separated by spaces. Type log without any parameters to switch info logging off.

`login [password]`

Login. Omit password if none is set.

`loop <off|far|near>`

Loop mode far, near, or off.

Default: off

In loop mode, incoming analog audio data will be fed to the analog output without intermediate coding. Incoming digital data (either ISDN or V.35) will be fed to the corresponding digital output.

`network <ISDN|V35>`

Select network: ISDN with or without autoanswer or V.35.

Default: ISDN if ISDN card is installed

Example:

`network isdn` selects ISDN as the network. Incoming calls will automatically be accepted.

`panic <0..50|1-4>`

Select panic dial setup, 0=off.

Default: off

Parallel contact closure 1 can be configured as 'panic input'. If panic dial is activated (different from 0) and this contact gets active (grounded) then the selected setup will be dialed – provided there is a setup stored. 1- 4 allows 4 parallel inputs to trigger auto dials 1- 4.

As soon as the input becomes inactive (open) the line(s) will be disconnected.

Note that the value of contact closure will be transmitted to the remote Zephyr even if panic dial is activated.

`passwd [oldpasswd] [newpasswd]`

Set or change or delete password.

If there is no password set:

`passwd newpassword` set a new password.

If there already is a password set:

passwd oldpassword newpassword changes to a new password.

passwd oldpassword * deletes the current password.

Default: no password

reset

Reset Zephyr (warm boot).

A warm boot does *not* cause deletion of important system information (type of ISDN, SPIDs, setups etc.).

rxmode <L3MO|L3ST|G722|L2>

Select receive mode (L3- Mono, L3- Stereo, G.722, Layer 2).

Default: L3ST

setup [<1..50>[<-|name <num1|-><num2|-><XmtMode><RcvMode><Rate>]]

Display or define setup. Parameters are: setup number, setup name, two phone numbers, transmit mode, receive mode, bitrate. See 'txmode', 'rxmode' and 'bitrate' command for parameters. Enter setup alone to get a list of setups.

IMPORTANT!

The "name" must not contain any spaces because space is the delimiter to the following phone number. To get spaces, use underscores instead.

HOT TIP!

Setups do not need to have phone numbers associated with them. A setup without phone numbers can be activated and it will bring the Zephyr to the selected transmit and receive mode and bitrate.

Default: all setups are empty

spids <spid1|-> <spid2|->

Enter or delete ISDN SPID numbers.

Default: none

Example:

spids -- will delete all stored SPIDs.

stat

Get system information.

This command prints a list of the current settings:

```
>>stat
DSP cards...........: 2
Transmitter mode.....: Stereo
Receiver mode........: Stereo
Line bitrate.........: 64 kbps
Network..............: ISDN
Switch type..........: Natl I-1
SPID 1 & 2...........: 21678193100111 21678193110111
Directory numbers....:
Auto answer mode.....: ON
Loopback mode........: OFF
Ancill. data from....: Port 1
Analog input level...: +0 dBu
Last boot mode.......: warm
Autoboot mode........: ON
Panic dial...........: None
Info logging.........: System ISDN extendedISDN
Line status 1 & 2....: Ready Ready
Decoder status.......: unsynced
Encoder status.......: synced
```

statout <rcv|1|2|1&2|1or2>

Select activation mode for the parallel port STATUS output bit: on receiver lock, on line 1 connected, line 2 connected, both or either line connected.

When followed by a lock mode parameter, this command changes how the STATUS OUT parallel output is determined, as follows:

rcv	Output active when decoder is locked.
1	Output active when ISDN line 1 is connected.
2	Output active when ISDN line 2 is connected.
1or2	Output active when either ISDN is connected.
1&2	Output active when both ISDN lines are connected.

Default: rcv

`telco <NI1|PTP|ETS300>`

Select ISDN switch type.

Default: NI1

`time [month/day/year-hour:minute[:second]]`

HOT TIP!

The Zephyr software automatically detects a DALLAS "RAMified timekeeper" in socket U3 of the European ISDN card (there is no U3 on the U-card). If the chip is present Zephyr's system time is updated accordingly during startup. When the time is changed using the "time" command, the time in the Dallas chip is also adjusted.

Get or set system time.

The time will start from 1/1/95- 00:00:00.000 after a warm or cold boot (exceptions see below). Once the time is set using the "time" command it will be displayed correctly until the Zephyr is booted.

This is especially useful along with the "log time" command in order to find out when certain things happen (i.e. failure of ISDN lines).

`txmode <L3DU|L3ST|L3JO|G722|L2MO|L2M128|L2DU|L2JO>`

Select transmit mode (Layer3: dual, stereo, joint stereo, G.722, Layer2: mono, mono 128/112kbps, dual, joint stereo)

Default: L3DU

`V35cd <1|2|12|->`

Force carrier detect true on V.35 port 1, port 2, or both. Use "- " to restore normal operation. Not stored through boot cycle.

`ver`

Display version information.

`volume [H <1..10>] | [S <0..9>]`

Set volume for headphones (H) or phone speaker (S).

Default: phone speaker 0, headphones 1

For headphones the volume range is 1...10; for the built in panel speaker it is 0...9. The speaker value determines the volume of ring and disconnect audible indications from the front panel's small loudspeaker.

The Keypad Feature

A remote Zephyr may communicate keys pressed on the dialing keypad after a connection has been made. At the remote end, the data is “invisibly” inserted into the ancillary datastream; at the local end, the data is extracted and forwarded to the serial output.

This feature is intended to control equipment such as a hard disk audio playback system. It would be possible to select specific cuts for listening at the remote end. For instance, a weather service could store forecasts for a number of markets, and subscribers could select the proper one.

The keypad info is output as: `<CR (0x0D)>KEYx<CR>` where `x` is 0- 9, * or #.

The <CR> makes sure that the KEYx message is always the only thing on a line.

Keypad info is sent along with the bitstream in Layer III mode only.

Keypad info is only output on the receiving RS232 port if it is not in on- line mode.

In some screens (SPIDS, Directory numbers, Dial screen) the number keys have a distinct meaning even when already connected. To avoid confusion: The keypad feature is only available when the STATUS screen is displayed.

Modem Initialization

One use of the serial port is to control a remote Zephyr via a separate (analog) phone line. In that case, a modem is directly connected to the serial port, and it may be necessary to initialize it (as a computer would, were it connected to one).

The Zephyr can do this by sending standard Hayes AT command to the modem in order to command it into auto answer mode:

- During bootup, no modem init strings are sent.
- To perform initialization, get to the help screen of the status screen, then press '1'. This will send the init strings.

Example Call Set-Up/Disconnect Sequence

Here is a simple example of a two- channel call being connected and then disconnected. (In this example, the log command is set to display only essential information.)

`conn 1 12167819310`

`>>Line 1: connected`

`conn 2 12167819311`

`>>Line 2: connected`

`disc 1 2`

```
>>Line 1: released

>>Line 2: released
```

Using the Zephyr’s Built-In “ ISDN Analyzer”

ISDN analyzers are generally expensive and elaborate pieces of test gear which enable you to see the transactions which are taking place between the terminal equipment and the telco switch (ours cost $70K!). These are often invaluable for troubleshooting subtle ISDN incompatibility problems.

We’ve built a free analyzer into the Zephyr.

Various levels of detail are possible, according to the chosen parameter(s) of the log command.

Entering **log MON** activates this function in a basic mode. A display of ISDN transactions will be available, as in this example call setup:

```
>>conn 1 7819310


ISDN-> Line 1 offhook

Calling number 7819310 on line 1 (B1)...


>>ISDN-> Null(0)

ISDN-> Line 1: Null(0) => Call-Init(1)

ISDN-> DLE: >> dl_data_req SAPI=0 TEI=121 CR=O,0x4

PD=08 CR=O,4 Type=05 {SETUP}

ISDN-> | IE (0:04) BearerCap

ISDN-> | 02 88 90 [3]

ISDN-> | IE (0:18) ChannelID

ISDN-> | 01 89 [2]

ISDN-> | IE (0:2C) Keypad

ISDN-> | 07 37 38 31 39 33 31 30 [8]

ISDN-> DLE: << ph_data_ind

ISDN-> | SAPI=0, C/R=0, TEI=121 <RR> N(R)=7, P/F=0

ISDN-> DLE: << ph_data_ind

ISDN-> | SAPI=0, C/R=1, TEI=121 <INFO> N(S)=8, N(R)=7, P/F=0

ISDN-> |  PD=08 CR=D,4 Type=02 {CALLPROC}

ISDN-> | IE (0:18) ChannelID

ISDN-> | 01 89 [2]
```

```
ISDN-> | IE (0:95) Shift-5
ISDN-> | IE (5:2A) ?
ISDN-> | 16 80 88 07 37 38 31 39 33 31 30 80 01 0D 80 01
ISDN-> | 14 80 01 14 80 01 [23]

ISDN-> Line 1: Call-Init(1) => Outgoing-Call-Proceeding(3)
ISDN-> Line 1 proceeding
ISDN-> DLE: << ph_data_ind
ISDN-> | SAPI=0, C/R=1, TEI=121 <INFO> N(S)=9, N(R)=7, P/F=0
ISDN-> |  PD=08 CR=D,4 Type=01 {ALERT}
ISDN-> | IE (0:34) Signal
ISDN-> | 01 01 [2]
ISDN-> Line 1: Outgoing-Call-Proceeding(3) => Call-
Delivered(4)
ISDN-> Line 1 ringing
ISDN-> DLE: << ph_data_ind
ISDN-> | SAPI=0, C/R=1, TEI=121 <INFO> N(S)=A, N(R)=7, P/F=0
ISDN-> |  PD=08 CR=D,4 Type=07 {CONN}
ISDN-> | IE (0:34) Signal
ISDN-> | 01 3F [2]
ISDN-> Line 1: Call-Delivered(4) => Active(10)
Line 1 connected
```

A full discussion of the meaning of these messages is beyond the scope of this manual. Those who are interested may request from us a pointer to the appropriate documents (from Bellcore and others).

In rare instances, Telos customer support may ask for this data for troubleshooting difficult ISDN problems. In some cases, this data would be collected into a PC disk file; in other cases, we may ask you to connect a modem so we can monitor "live."

Ancillary Data

Usually, the serial interface is used to control the Zephyr, i.e. to change transmission modes or request status information.

But the serial interface can be switched to transparent or ancillary data mode by using the **atdt** command. While the transparent mode is active, all data sent to the Zephyr's serial input is inserted into the coded audio signal and – provided there is a ISDN or V.35 connection – transmitted to the receiving Zephyr. The latter will extract ancillary data information from the coded audio signal and output it on it's own serial interface if it is in transparent mode.

- The ancillary data transmission is fully bi- directional and does not affect any ongoing audio transmission. The necessary bitrate to transmit ancillary data, however, is taken from the overall available line bitrate.
- Ancillary data that is fed into the Zephyr without having a connection will be lost.
- The ancillary data channel is an on- demand service, which means that the audio bitrate is only reduced if necessary. However, data can be transmitted continuously at the currently selected serial baud rate.
- Ancillary data is only available in Layer 3 mode.
- In Dual/Mono mode ancillary data will be transmitted in the left channel only unless commanded otherwise.

How to Control Ancillary Data Transmission

By default, a receiver will automatically switch to transparent mode if it receives ancillary data from the other end. It will print CONNECT to the serial port and then start outputting received ancillary data.

- To switch off transparent mode and to gain access to Zephyr control commands again, enter **+++** or **###**. (Do not transmit any data within a guard time of 1 second before giving this command.)
- After switching off the transparent mode, the Zephyr will NOT switch back to transparent mode even if it encounters incoming ancillary data ("automatic transparent mode activation" is deactivated). This prevents a local unit from going immediately back to transparent mode when the other end is still trying to send ancillary data.

Ancillary Data Commands

atdt Immediately switches to transparent mode.

atdt w Enables automatic transparent mode activation on incoming ancillary data.

+++ or **###** Switch off transparent mode and automatic transparent mode activation and return to Zephyr command interpreter.

Note that you will have to use the **###** command if you have a modem connection to a remote Zephyr. This is because the **+++** command would activate the modem's command interpreter rather than the remote Zephyr's.

Startup Commands

Startup commands are a simple way to perform basic RS232 configuration during the Zephyr boot sequence. This may be necessary when no PC is available, as when a radio transmitter remote control is to be connected directly to the serial port.

While the display shows the `Initializing...` message, you can press one of the following keys:

1. Sets the RS232 interface to 2400 baud. Disables transparent data transmission and automatic transparent mode activation.

2. Sets the RS232 interface to 9600 baud. Disables transparent data transmission and automatic transparent mode activation.

3. Sets RS232 interface to 19200. Disables transparent data transmission and automatic transparent mode activation.

4. Sets the RS232 interface to 2400 baud. Enables transparent data transmission (Ancillary data mode).

5. Sets the RS232 interface to 9600 baud. Enables transparent data transmission (Ancillary data mode).

6. Sets the RS232 interface to 19200 baud. Enables transparent data transmission (Ancillary data mode).

7. Not used

8. Enable special CRC mode.

9. Special NRW mode.

MS WINDOWS ZephyrControl SOFTWARE

We have developed a software package for control of the Zephyr under Microsoft Windows. It is included on the diskette which comes with your unit.

The application offers a user- friendly method for:

- Managing ISDN settings
- Entering and viewing set- ups.
- Using the end- to- end ancillary data (text) feature.
- Establishing and disconnecting calls.

Hardware Connection

Use a standard serial cable, as described elsewhere in the manual, to connect the Zephyr and the PC’s serial port. Note that the Zephyr is DTE, meaning it looks like a computer, rather than a modem.

A good procedure is to use the MS Windows terminal application in order to test the hardware connection before starting the ZephyrControl software.

Loading and Running the Application

In order to install the Zephyr Control software you need the following system configuration:

- ″IBM PC or compatible computer
- Windows 95, Windows 3.1 or Windows 3.11 in enhanced 386 mode
- A free serial port (COM 1 or COM 2)

To install ZephyrControl run the Windows SETUP program from the installation disk. Then start ZephyrControl by clicking on the icon.

Pressing F1 will get you further help about connecting the Zephyr to your PC as well as program options and commands.

Custom Adaptations

The software is designed to be used either "as- is," or to be modified by you.
To that end, we are willing to share with you the source code. Visual Basic is a language which should be accessible to many PC- savvy users. Contact Telos Systems customer support if you wish a copy of the source code.

PARALLEL PORT AND CLOSURES

For additional information on using the parallel port see manual section 8 (Detailed menu reference) and section 3 (Zephyr at a glance).

Parallel outputs may be affected either by the corresponding input at the remote Zephyr, or by commands on the serial port.

Normally, the closures are simply passed- through from one end to the other, but the following commands may be used influence the process:

`cc [<0000..1111>]`

Set contact closure output value (4 bits) or select LOCK mode.

This command followed by a binary value sets the local Zephyr's parallel outputs to the value. If connected to a remote Zephyr, the values from that unit will override this value. Since an update is sent every 5 seconds, the entered value will not remain longer than this period.

`ccmask <0000..1111>`

Set contact closure input XOR mask (4 bits). This is used to flip the polarity of inputs to match what is available from the outside world. A value of 1 means the input is inverted from usual. When an input is not connected, this command may be used to simulate the input signal.

`statout <rcv|1|2|1&2|1or2>`

Select activation mode for the parallel port STATUS output bit: on receiver lock, on line 1 connected, line 2 connected, both or either line connected.

When followed by a lock mode parameter, this command changes how the STATUS OUT parallel output is determined, as follows:

`rcv`	Output active when decoder is locked.
`1`	Output active when ISDN line 1 is connected.
`2`	Output active when ISDN line 2 is connected.
`1or2`	Output active when either ISDN is connected.
`1&2`	Output active when both ISDN lines are connected.

CC info is transmitted immediately if the input changes; otherwise it is sent every five seconds.

> HOT TIP!
>
> Contact closures are only passed in a transmission path set to Layer III coding mode.

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SECTION 10

ADVANCED PROBLEM SOLVING

GENERAL

Thinking About Problem Solving

Despite best intentions, something could always go wrong. Sometimes troubleshooting a 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 (repeat steps three through five, if necessary);

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, your’s or the calling party’s long distance carrier , and the data circuit and equipment at the other end.

- **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 the Zephyr may present when the problem occurs. Enlist the help of the users to keep track of these messages and give you this information. Keep detailed notes.

- **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. Change the ISDN connection to a known good line; call a known good Zephyr 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. Twenty 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 does not mean that it's OK now.

At 3:00 AM when the problems seem intractable and you're ready to give up engineering, remember that the system has worked and will work. Never panic – you are smarter than it is.

Diagnostic Aids

Fortunately, your Zephyr has some features which are designed to aid your troubleshooting effort.

SEND SYNC status LED

In ISDN units, the SYNC LED is illuminated whenever power is applied. It also indicates that the Zephyr has basic functionality. If not the Zephyr has not completed it's boot cycle, or major clock problems exist within the unit.

In non- ISDN units, the SYNC LED indicates that a clock is present at the V.35/X.21 digital port #1 and at the correct frequency. Since this clock is generated by the CSU/TA, it is the best indication that the CSU/TA is connected and has basic functionality.

RECEIVE status LED

The LOCK LED illuminates to indicate that the decoder is receiving valid data. If this is flashing, or off, there is either the wrong Rcv mode, Sample Rate, Bit Rate, set. Or there is bad data being received.

NT1 status LED(s)

Zephyrs with built- in NT1's have an LED on the ISDN interface card on the back of the unit. Normal status is indicated by the LED being lit. Rapid flashing indicates loss of ISDN line at the lowest (physical) level, or use of the incorrect jack on the Zephyr ISDN interface card. Slow flashing of this LED may indicate problems with the Zephyr.

External NT1's have one or more indicators which can help differentiate line vs Zephyr problems.

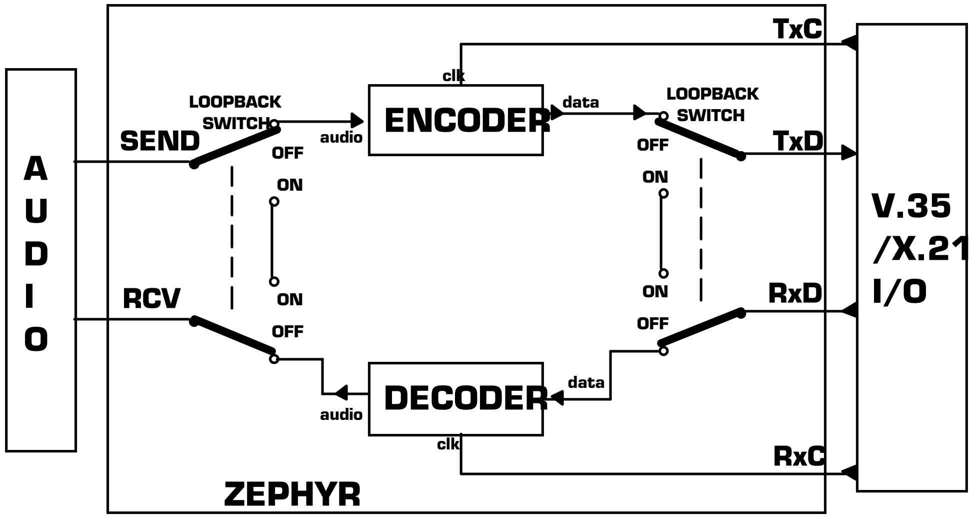
> IMPORTANT!
>
> The Zephyr's meters are off during FAR loopback.

LOOPBACK modes

- FAR Loopback

  The Zephyr's FAR loopback mode is "bilateral" or two- way. It loops the local audio directly from input to output. At the same time, the distant end gets it's signal returned. The screen for selecting the loopback mode is accessed by pressing the UTIL button twice.

The following block diagram shows what happens schematically when loopback mode is in effect.



This diagram shows the operation of the loopback function. Note that both sides are affected. The local audio is looped, as well as the network signal. (This diagram is labeled for V.35/X.21 I/O, but applies equally to systems using the internal ISDN TA.)

- NEAR Loopback

  Incoming audio gets coded in the usual manner, sent into the ISDN interface card and from there looped back to the decoder where it is decoded. Allows a check of over 90% of Zephyr's functionality with no ISDN line present.

NOTE: Compatible Xmt and Rcv modes will be required.

These two loopback modes are powerful troubleshooting tools:

- NEAR loopback can eliminate substantial portions of the Zephyr's circuitry by testing to see if the coder and decoder are working. Be sure to test in the relevant

> HOT TIP!
>
> Near loopback is only available on units with ISDN interface card installed . If no ISDN interface is installed "hardware not available" will be displayed.

  Xmt and Rcv modes to your situation

- Assuming only one problem, the FAR looback can effectively determine whether the problems experienced are due to the communications channel(s) or one of the Zephyrs.

  - While the problem is occurring put first one Zephyr, and then the second Zephyr into far loopback (after restoring the first to normal status). Be sure to use compatible Xmt and Rcv modes.

  - If either unit can lock to itself consistently looping through the other unit, then the communications path(s) cannot be at fault. The unit which can lock to itself is ok.

    - Within this context the symptoms will now give insight as to whether the problem is with the encoder or decoder.

- If neither unit can lock to itself then failure of one (or both) communications path(s) is likely.

Zephyr Audio Level LEDs

To determine if audio is present, and at what level.

- It is possible for the meter to fail when the relevant section is otherwise working. Metering is done in the digital domain.

CSU/TA Loopback Modes

(Applies to external CSU/TA's only.) Most CSU/TA's have a "Local Terminal" or "DTE" loopback mode which serves to "give you your signal back." Thus, if your send audio can be received when CSU/TA loopback is engaged, the Zephyr and interconnect cable are functioning, and the problem is likely either in the telephone network, the CSU/TA, or the Zephyr at the other end.

Most CSU/TAs also have modes to loop the distant signal back to the sender, as well.

Use similarly to the Zephyr's loopback modes as indicated above.

Condition: Front Panel Display Problems

If the LCD is dark or unreadable, the special front panel diagnostic and set- up mode may be useful.

To enter this mode, press the <YES + > and <NO - > buttons simultaneously while powering- up the Zephyr. All LEDs should illuminate and the LCD displays the famed "Kevin Icon." The <SEL> buttons adjust the backlight level, while the <YES+> and <NO-> buttons control the LCD contrast. The unit must then be switched- off and on again to resume normal operation.

If this does not work then the Zephyr's internal ROM settings may have been scrambled. In this case hold the HELP button depressed for 5 to 10 seconds and release. The default contrast and brightness values have now been set.

If neither of the above works try the COLD BOOT command.

Condition: No Basic Functionality

Does the Zephyr have power?

In ISDN units, the SYNC LED is illuminated whenever power is applied. Is the switch on? Is the cord plugged- in? Does the outlet have power? Is the Zephyr fuse blown?

If replacement is required, please note: **For continued protection against fire, replace fuse only with same type and value.** See the section Gaining Access for information and important cautions.

> IMPORTANT!
>
> The Zephyr's power fuse is located inside the box, on the power supply module. To change it, you will have to remove the top cover. Be sure to remove AC power before opening the unit.

Does the Zephyr have basic functionality?

Upon initial power- up, the LCD shows the software version and the lowest line indicates `INITIALIZING...` After about 5 seconds, the screen changes to the resting, status screen.

During the initializing period, a self- test is performed. If there is a problem, this is reported on the LCD (it will take a few moments for it to scroll across the screen) and progress is halted. The entire 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 OK, connected and receiving valid data. If not illuminated, try cycling through receive modes to see if you can "find" the mode at the other end.

What happens when you call yourself?

Try calling yourself, from one ISDN line to the other (assuming you have both available). Use a 56 kbps rate first, as this is the most universal. Set Rcv mode to L3 Stereo and Xmt to L3 Dual or L3 Stereo) Audio should work in both directions. This confirms that the Zephyr is OK, and that the problem lies somewhere downstream, probably in the Telco's connection facilities either between its own central office switching equipment or to the long- distance carrier you are trying to use.

What happens when you try a NEAR loopback?

If successful, this eliminates over 90% of the Zephyr circuitry as the culprit. Remaining suspects would be primarily the ISDN card, or the V.35 card (if relevant to your problem). Remember, Xmt and Rcv 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.

What happens if you try a different long distance carrier for the call? Try the call at 56 vs 64 (or vice versa)? Try reversing the direction of the call? 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. The above will force you through the network via a different path.

If local calls are ok, but long distance calls fail with all three major long distance carriers, then it is likely there is a tandem trunk problem with one of the local telcos involved in the connection.

Condition: No Audio In Both Directions

Does the Zephyr's loopback isolate the problem?

See above for tips on using the loopback modes.

On the other hand, if you can't receive distant audio, but get a delayed version of your own transmitted signal, it's possible that one of the Zephyrs was left in the loopback mode. We've also run across situations were one of the NT1's or a pieces of the telco's gears was left in loopback mode.

Are the Zephyr 's mode set properly?

The Zephyr's XMT Mode and RCV Mode must be set to the proper conditions at each end in order for the system to work.

Sometimes cycling between modes can clear a decode problem.

Is audio properly connected and configured?

Are the Zephyr's front panel send meters indicating signal present at a correct level? If not, check audio connections and cables. Are the rear panel audio switches set to the proper position?

Condition: Audio In One Direction Only

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.* The following will force you through the network via a different path.

What happens if you try a different long distance carrier for the call?

Try the call at 56 vs 64 (or vice versa)?

Try reversing the direction of the call?

Try placing a local call as a long distance call?

Are audio signals and connections OK?

Goes without saying – its got to be plugged- in to work. Try reversing A and B audio channels

Do you have a good telephone network path in both directions?

Isolate the problem area by using the loopback techniques described above.

Condition: Audio Distorted

Is the rear-panel MIC/LINE switch in the proper position?

Applying a line- level signal to a system configured for microphone levels will make for some pretty awful distortion.

Send gain set OK?

Check the SEND meter for proper indication. Confirm that the LIMIT indicator is flashing not at all, or only occasionally. Input/send gain is set on the Volume menu screen, and may be adjusted to work with levels from - 20 to +4 dBu.

Output switch in proper position?

The - 10/+4 switch near the output connectors should match to the equipment which accepts the Zephyr feed.

“ Fatal Error” Messages

The Zephyr software has a system for self- monitoring. When a major problem is detected, a message is scrolled on the bottom line of the LCD screen. The message gives general information about the problem in a text phrase, and specific information in an error number. (These are usually four- digit numbers in the format 3XXX.” Please note the value and description (it will take a moment for the information to scroll across the screen) in order to assist Telos customer support.

ISDN-RELATED

ISDN TIP!

European users should disregard all references to SPIDs. Euro ISDN DOES NOT have SPIDs! If you ISDN configuration requires MSNs they may be entered in the MSN/SPID 1 & 2 screen.

Condition: Can’t Get ISDN “ Ready” Indication

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 Natl 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 other forms as well.

- 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?) they may have given you a 2 B-channel AT&T "Custom Point to Multipoint" line with 2 phone numbers and 2 SPIDS. Zephyr does not support this configuration. However, in an emergency, to save a remote, you can usually get things working (temporarily) as follows:

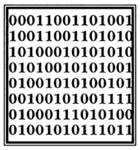
1. Remove both SPIDS and both directory numbers
2. Set Telco to PTP
3. Have another Zephyr call either of your phone numbers twice. They should be able to connect.
4. The line will now function normally for outgoing calls until Zephyr is rebooted or the ISDN connection is reinitialized.

If required, carefully verify that you have the correct SPID from the Telco, and that you

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?) they may have given you a 2 B-channel National ISDN line with only one SPID and one phone number. This may work, however you will need to proceed as follows:

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



ISDN TIP!

In an emergency, to save a remote, you can sometimes get a line working without the proper SPIDs or line configuration by dialing into it. This fix is only temporary and the line will fail to initialize next time the Zephyr is booted.

HOT TIP!

If you suspect the SPIDs given you are wrong, or the basic line provisioning (configuration) is incorrect call the Telco and ask them to verify the SPIDS and provisioning *from the switch*. They'll probably need to have someone call you back. Verify this person is *logged into the switch.* Get their fax number and fax them the Zephyr's ISDN information from the Appendix of this manual.

Sometimes the information in the Business Office computer is wrong. By verifying what is programmed into the switch itself you can save time.

have properly entered it into the Zephyr. It must be exactly correct in order for anything to work! We've seen a few cases where it was given incorrectly 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.

We've included some useful diagnostic aids in the Zephyr to help you determine where a problem is originating. The key are the ISDN status words on the left side of the LCD basic status screen:

- `inact/inact` means no ISDN connection
- `wait/init` means incorrect SPID1.
- `init/init` means wrong protocol set up at CO, or Zephyr is not matched to it
- `Ready/init` means only one working ISDN channel
- Ready/wait means incorrect SPID2
- `Ready/Ready` means OK on both ISDN channels

IMPORTANT!

One caveat regarding the `Ready/Ready` indication: If you have this indication with the Zephyr Telco mode set to PTP and you cannot dial, it is likely that your line is Natl ISDN 1. (The PTP protocol does not require SPIDs, so the Zephyr does not send or check them; it just assumes the line is OK in this regard, so it gives the Ready indication, though the line is waiting for the SPIDs. In this case calls would immediately fail.

Can you place a "pots" call?

From the call options screen change the call type from "Zephyr" to "phone" and try a call. Try this for both "line 1" and "line 2". If you can place pots calls but not data calls then the telco must not have given you CSD (Circuit switched data) capability which is essential for the Zephyr to work

Is your long-distance carrier OK?

If you can successfully call locally, but long- distance calls don't go through, the Zephyr usually displays `Far end disconnect`, `No route available, SW 56 Disconnect, 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:

- ❑ AT&T 10288
- ❑ MCI 10222
- ❑ Sprint 10333

You must dial the full number, including the 1 or 011 + country code following the prefix.

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) as that will affect how it gets routed through the network.

What does the ISDN "Cause" phrase say?

This phrase, which appears on LCD line 2 after dialing, comes from directly from the telephone company equipment, and can be valuable for troubleshooting. Generally these phrases are self- explanatory. Getting them 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 LED indicator of the status of the ISDN connection is on the rear panel near the U jack. If the NT1 is inactive, the LED will remain in the OFF state. Activation in progress is indicated by a rapidly blinking LED (about 5 Hz). If NT1 can contact the central office, the LED will blink slowly (about 1 Hz). The LED will come on solid when all handshaking is completed and the basic line connection is good.

If the LED is off, or blinking slowly, the Zephyr could be the cause. Disconnecting from the ISDN line should cause the Zephyr to return to the rapidly blinking state. Rapid blinking is usually a sign of ISDN problems, although a Zephyr problem is also possible.

Are you using an external NT1?

If so, check the LEDs on the unit.

The Tone Commander NT1 usually provided by Telos has four LEDs, as follows:

- `Power.` Should be lit. Normally, power for the NT1 is provided by the Zephyr over the 8- pin S interface cable. Make sure this cable is installed to the proper connectors. If this blinks then NT1 power is below the required level.
- `Active.` Lights green when all is well.
- `Terminal Error`. Lights red to indicate a problem on the terminal (Zephyr) side. Usually means a hardware problem with the Zephyr, but could be a cable or NT1 problem.
- `Line Error`. Lights red to indicate a problem with the Telco line. This means a very basic kind of problem – usually no physical connection.
  - Yes, you *can* use a analog phone or butt set to do a basic check:
    - You will hear a clicking" sound when a line is connected. If you rapidly go off and on hook you will hear a burst of white noise ("rushing" sound) for a few seconds. If so, you are probably physically connected and the problem is the configuration or NT1.
- Combinations of lights and blinking allow a fine degree of explanation, see the NT1 manual 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 light(s) are acting normally.

The most important tool is the Zephyr's integrated ISDN analyzer. This enables you to look at the transactions between the Telco line and the Zephyr, so that a judgment can be made as to whether the problem lies within the Zephyr or with the telephone service.

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 attention to a line problem. It's less hard to deny evidence provided by their own gear.

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.

With External CSU/TAs

Is the CSU/terminal adapter connected and configured properly?

The SYNC LED near the SEND label on the front panel indicates that a clock is present at the V.35/X.21 digital port #1 and at the correct frequency. Since this clock is generated by the CSU/TA, it is the best indication that the CSU/TA is connected and has basic functionality. If this LED is not lit:

- Check the cable. Is it properly seated into all connectors? Is it in good condition? All pins straight and in good order (Check for loose pins)?
- Do the CSU/TA bitrates and the Zephyr's match? Both must be set to either 56 or 64kbps. On the Zephyr, the bitrate setting is accessed by pressing the UTIL button.
- Is the CSU/TA working? Does it have power? Does it's LCD show normal text? What do the LED status indicators show? Does it respond when a button is pushed? Is it properly configured for the line it is connected to? Note that some CSUs will not provide clock until dialed up. The adapter manual should be consulted for information about these matters, as adapters are too varied to offer detailed advice here!
- Sometimes handshake timing can cause problems. If so, you should "force on" or set to "ignore" handshaking options (CTS, RTS, CD, DTR, and DSR) on the CSU/TA.
- Do you have the proper cable? STEREO and DUAL/MONO operation require that a two- headed "Y" cable be used to connect the Zephyr to two CSU/TA ports.

Does the CSU/TA's loopback isolate the problem?

Most CSU/TA's have a "Local Terminal" or "DTE" loopback mode. Place it in this mode. (Read its manual to determine how.)

This CSU/TA loopback mode serves to "give you your signal back." Thus, if your send audio can be received when CSU/TA loopback is engaged (with Xmt and Rcv set to be compatible), the Zephyr is functioning, and the problem is likely either in the telephone network or the Zephyr at the other end.

Most CSU/TAs can also loop the distant signal back to the sender, as well.

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 dest./ 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 the prescribed period of time allocated (before timer T303 or T310 has expired).

`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 the prescribed period of time (before timer T301 has expired).

`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, cause no: 1, unallocated (unassigned) number shall be used.

`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 off- line.

`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 a “9” or “8” for an outside line, if required. 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)

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SECTION 11

DETAILED TECHNICAL INFORMATION

PHILOSOPHY

In the past few years, the nature of broadcast engineering has changed considerably. At many stations, the engineering staff has been reduced in size and new responsibilities have been added. At the same time, equipment has gotten more complicated and specialized. Thus, many practitioners of the broadcast electronic arts are forced to become “systems” engineers, emphasizing equipment application rather than component- level troubleshooting.

This is probably a positive development, since it really would be impossible for a station engineer to fully understand the internal nuances of all the wonderful new high- tech stuff that is now available to improve station operations! Also, as equipment becomes more sophisticated and specialized, stocking spare parts for every eventuality has become difficult.

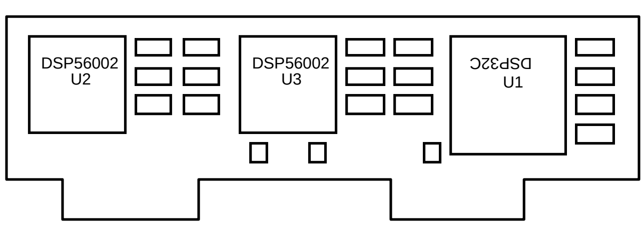
Thus, we at Telos don’t really expect that much component- level troubleshooting will occur. To support you when you need help, we keep spare boards available for fast overnight shipping. In most cases, we will swap boards with you at no cost. We do not charge for most repairs.

However, despite the comments above, we do provide schematics and component level troubleshooting information in case you have the need or desire to tackle a repair (or modification) yourself. Another reason we provide the information is to satisfy your curiosity. If you are like us, you probably just have to know what’s happening inside that fancy box.

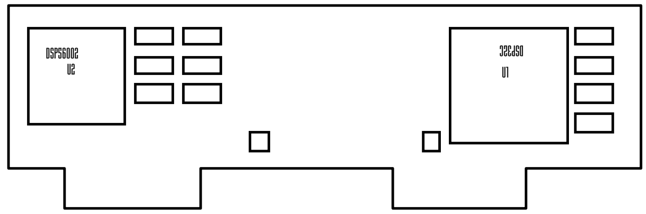
DSP CARDS, CONFIGURATION, MODES

This section contains detailed information on the DSP card configurations and resulting operating mode capability. It describes the usage of the various DSP chips along with the digital audio signal flow. It is a good starting point to acquire understanding the design of the unit.

The Zephyr may have one or two DSP cards installed. The first, leftmost socket position must always have a DSP card, and this card must be the “primary” type – all three DSP chips must be present on the Printed Circuit Board. The rightmost socket position may have a “secondary” DSP card installed – only two DSP chips are present on this module.



A “ primary” (Mono) DSP card. It has all three DSP chips present, and must go into the leftmost socket within the Zephyr.



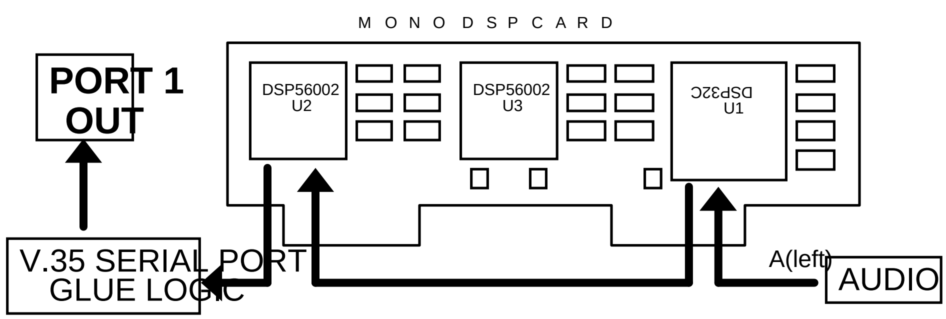
A " secondary" (Stereo) DSP card. It has only two DSP chips present, and goes into the rightmost socket within the Zephyr. It is optional.

The number of DSP cards determines the available operation modes. For consistency, from here forward, the DSP card situation will determine the configuration. The configuration will in turn determine which modes are available. The specific mode in effect is a user selection made via the front panel pushbuttons and LCD.

When reference is made to ports, we are referring to the digital bitstream I/Os, of which two are possible.

Transmit Modes

One DSP Card (3 DSP) Configuration

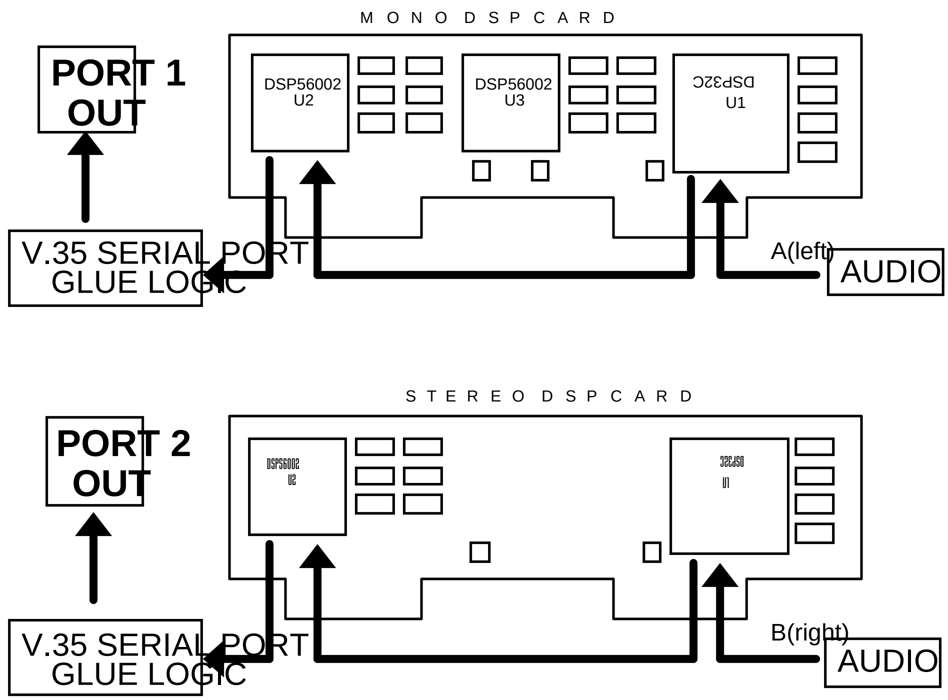


Xmt Mode: L3 MONO; L2 MONO

Only Line 1/Port 1 is used for transmit. However, Line 2/Port 2 may be used for a second receive channel.

G.722 uses only U2 on this card.

Two DSP Card (5 DSP) Configuration

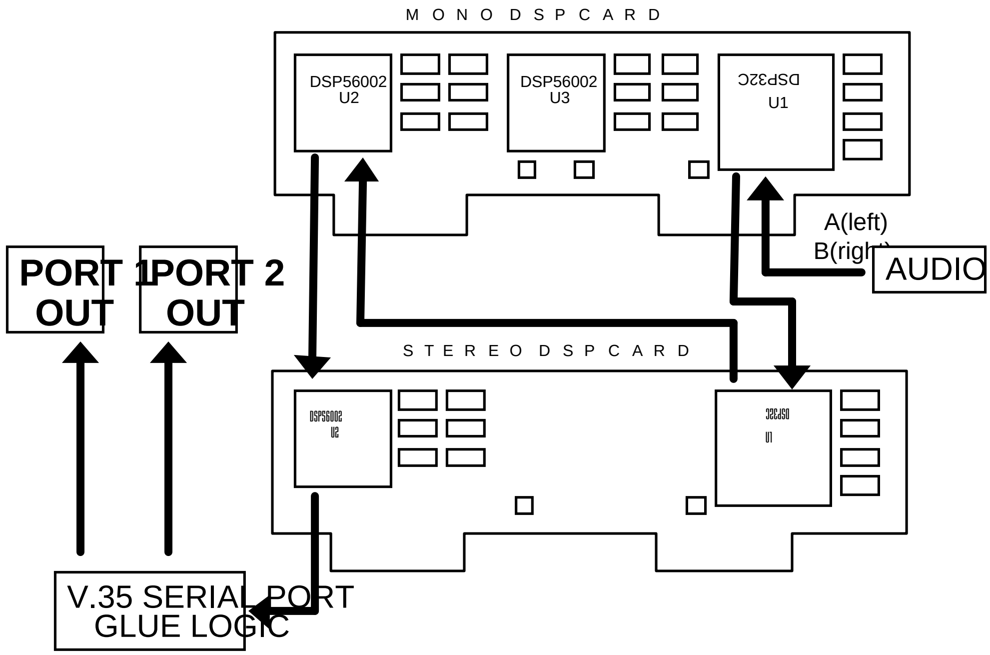


Xmt Mode: L3 DUAL

This mode generates two independently coded bitstreams. It effectively causes the running of two copies of the mono encoder software simultaneously.

(V.35 only.) The system clock is taken from either Port 1 or Port 2. The first port with an active Carrier Detect signal is used, with a default to Port 1 when there is no CD active on either port.

Two DSP Card (5 DSP) Configuration



Xmt Modes: L3 STEREO; L3 JSTEREO

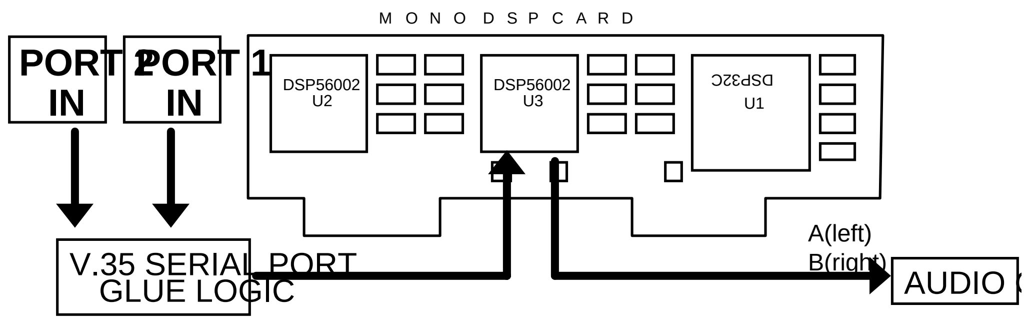
In the above modes, both ports are used with an internal splitter to transmit a single 112/128kbps stream. The last DSP and glue logic split this bitstream into two 56/64kbps channels. Unlike the `DUAL` mode, there is no relation between the audio input channels (A and B) and output ports (1 and 2). Both bitstreams must be sent to the same decoder for proper operation. However, it is irrelevant which ports go to which stream, as the decoder software is able to automatically determine which is which in order to properly reassemble them.

The system clock is taken from either Port 1 or Port 2. The first port with an active Carrier Detect signal is used, with a default to Port 1 when there is no CD active on either port.

Xmt Modes L2 MONO128, L2 DUAL, L2 JSTEREO use both cards, but only U1 on each.

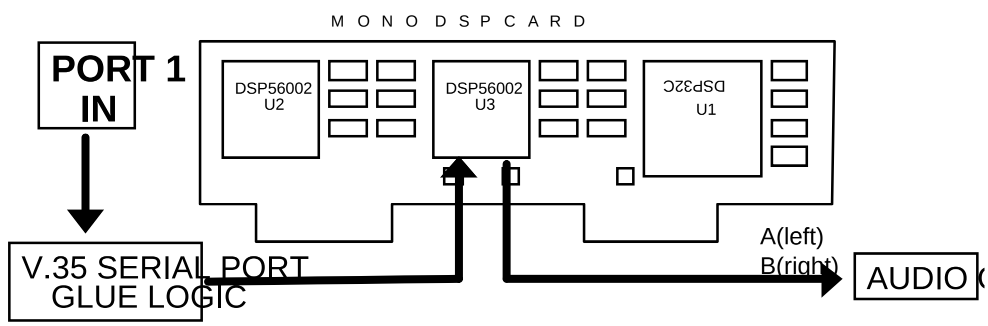
Receive Modes

One (or two) DSP Card Configuration



RCV Mode: L3 STEREO; L2; G.722

One (or two) DSP Card Configuration



RCV Mode: L3 MONO; G.722

Same as above, but with only one input stream.

GAINING ACCESS

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 allservicing to qualified service personnel.

CAUTION: As with most switching power supplies the Zephyr's power supply has lethal voltages, even on parts which might look safe at casual glance. *DO NOT TOUCH ANY PORTION OF THE POWER SUPPLY without removing power cord first.*

Removal of the top plate is the first step to gaining access for service. Remove the ten Philips head screws.

Motherboard Removal

REALLY IMPORTANT NOTE!!

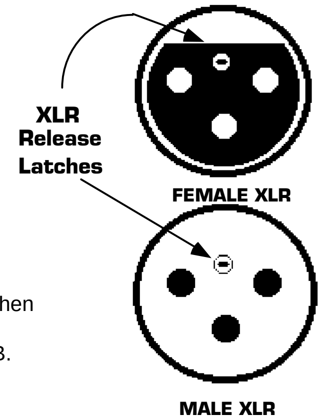
In the case where it is necessary to remove the motherboard, remove the two screws which mount each of the cards to the rear panel, remove the retainer screws on the lower DB connectors, remove the *4* screws which hold the board down (There is one hiding at the rear end of P2!), and then release the XLR connectors according to the instructions below. Disconnect all cables, and away you go.

XLR Connectors

The XLR connectors have retaining screws which have to be turned in order to be released.

XLR CONNECTOR RELEASE:

Insert a small screwdriver into the holes in the connectors, shown at right. Turn the screwdriver about one eighth of a turn counterclockwise to release the connectors. A small screwdriver such as the Xcelite R3322 or R3324 mayneed to be filed down some to fit the slots. Remember to re-tighten the XLR latches when replacing the PCB to ensure correct support for the XLR connectors on the PCB.



Installation/Removal of DSP cards

The DSP card modules are anchored with two tie- wraps per. If it is required to remove the DSPs, you’ll have to cut the ty- wraps and replace them later. Of course, this may not be needed when the unit will be stationary.

Important Note About Disconnecting the Front Panel...

IMPORTANT!

If the front panel is disconnected from the motherboard, then re-connected, the 87C51 will no longer be in “stand-by” mode and will attempt to draw normal operating current–this can completely drain the motherboard’s battery in a matter of days! The proper sequence for re-connecting the front panel includes powering up the Zephyr immediately after making the front panel-motherboard connection, then powering down. This is the only way to put the 87C51 into the proper “stand-by” mode!

COMPONENT-LEVEL CIRCUIT DESCRIPTION and TROUBLESHOOTING

General

Desoldering

While we socket the ICs that have the greatest potential for failure, many of the Zephyr's ICs are soldered in. That's because most of the time the socket is more likely to cause trouble than the IC. This is of no consolation when one of the soldered ICs appears to have failed. When you need to replace a soldered in chip, the right tool is essential. We use a vacuum desoldering system made by Pace (the MBT- 100) and highly recommend it. Cost is about $450 - worth it if you do much PC board troubleshooting work. The only other real alternative is to clip the leads from the top and remove the solder from the holes with solder- wick. We've not had much luck with the non- heated, manual vacuum desoldering devices like the ones from Radio Shack. We do not recommend that newly- soldered connections be defluxed.

> IMPORTANT NOTE!
>
> Zephyr memory backup is powered by Lithium battery BT-1. The following precautions must be followed when working on the Zephyr motherboard or when this battery is replaced:
>
> 1) Do not short the battery terminals (or traces connected to these terminals) together.
>
> 2) Lithium batteries contain lithium and may be considered hazardous. Local procedures must be followed when disposing of used batteries. Do not dispose this battery by burning.
>
> 3) Do not attempt to open the sealed battery container.

Active Low Notation

Whenever a slash (/) or a asterisk (*) is used after a signal designation in the text or on the schematics, an active low is signified.

Circuit Description: Rev. F and Below

The revision level is screened on the motherboard on the edge closest to the front panel, near the battery.

Motherboard Rev A-E: Audio Input

The signal enters the input buffer in balanced form. This is the common "instrumentation amp" balanced input topology. The audio is extensively RF filtered by the combination of ferrite beads, shunt capacitors, and special PI- filter components. The stage gain may be set for either microphone or line level by changing the feedback resistance using SW1A, B. Input audio is passed either to the send limiter, or directly to the A/D converter, depending upon JP1,2.

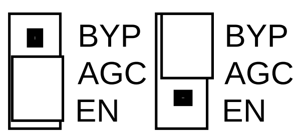
Motherboard Rev F: Audio Input

The revision F motherboard has some significant improvements made to the Audio Input section. Audio enters the Zephyr via XLR connectors and is RF filtered. The stage gain may be set for either microphone or line level by changing feedback resistance using SW1A, B. The signal is then sent to the new “B audio” plug- in board through JA1. The input stage consists of a very low frequency “servo” to remove DC offset, and then to the send limiter. Balanced audio is received at the A/D converter, U16, and the digital output leaves the “B- audio board” through JA2 to be sent to the motherboard and audio encoder. U- is used to multiplex the analog audio source data with the AES/EBU input data based on the front panel setting.

Motherboard Rev A-E: Send Limiter

The Zephyr has an send audio limiter which serves to keep the usual “digital nasties” from happening when the send program signal grows instantaneously too large. This is a very fast and very “tight” limiter which does not assert itself during times of normal audio level.

It is possible to bypass the send limiter. On the motherboard inside the Zephyr, near U34 and U37 are small jumpers. They can be set as follows:



Limiter Enabled. Limiter Bypassed.

As this is intended as an “extreme audiophile” mode, *we bypass the send level Voltage Controlled Amplifier as well, so the front panel send level control does not have any effect.* Note also that the LIMIT LEDs on the front panel continue to function and are useful as an indication that the A/D converter may be clipping the audio.

The limiter section has a dual function: in addition to the limiting, it uses the SSM2120, U23 as a VCA (Voltage Controlled Amplifier) to implement the system send input gain control.

Diodes D6,3 and D5,4 are clipping diodes, which prevent the A/D converter from ever getting a signal which could overload and clip offensively. The op- amp U40 makes plus and minus voltages of equal amplitude and low- impedance to bias the diodes at the proper point to protect the A/D without sacrificing any headroom.

Motherboard Rev F: Send Limiter

The Zephyr has a send audio limiter which serves to keep the usual "digital nasties" from happening when the send program signal grows instantaneously too large. This is a very fast and a very "tight" limiter which does not assert itself during times of normal audio level. It is possible to bypass the send limiter. The blue "AGC" on the "B- audio board" can be set to "IN" or "OUT". "In" places the limiter in the audio path. "Out" bypasses the entire limiter section, and the servo output is routed directly to the A/D converter.

The limiter section has dual functions; in addition to limiting, it uses the SSM2120 (U-23) as a VCA (voltage controlled amplifier) to implement the system "send" input control. When the limiter is bypassed the VCA is bypassed as well and the front panel volume- send option will no longer control input level.

Diode arrays DA1 and DA2 are clipping diodes, which prevent the A/D from ever getting a signal which could overload it and clip offensively.

Motherboard Rev A-E: Analog-to-Digital Converter

The A/D is a single chip, stereo, 16- bit sigma- delta type converter. The A/D oversamples the audio at 4.096 MHz. It applies noise shaping, and then filters and decimates to a 32 kHz sampling rate. The samples are output serially and transmitted to the XILINX chip, and ultimately to the first DSP chip in the send audio chain.

The left and right analog inputs are applied to pins 2 and 27, respectively. Pin 23 is a 4.096 MHz clock input. The divides this clock by two and outputs a 2.048 MHz clock on pin 20. The 2.048 MHz clock is divided in the XILINX part to 32 kHz which is applied to pin 14. Pin 16 is the data output. Both the left and right samples are to be found here, synchronized to the 2.048 MHz bit clock. The left sample starts on the rising edge of the clock.

Motherboard Rev F: Analog-to-Digital Converter

The Crystal Semiconductor 5390 A/D converter is a single- chip, balanced, stereo, sigma- delta converter with 20 bit resolution. It can operate a multiple sample rates including 16, 32, and 48 ksps as used in the Zephyr. The A- weighted SNR of the converter is 110dB. Pin 15 is data output.

Motherboard Rev A-E: Digital-to-Analog Converter

The D/A is a single chip, stereo 18- bit D/A converter with an integral low- pass reconstruction filter. Left and right channel audio exits the chip on pins 2 and 26, respectively. The audio is fed to the chip in digital form on pin 18, with pin 19 being the associated bit clock. Pin 20 is the "left/right" clock which is used to synchronize the serial audio words. It is derived from the A/D's pin 14 L/R clock output.

Motherboard Rev F: Digital-to-Analog Converter

Decoded audio is sent to the “B- audio board” in digital form through JA2. The bitstream then goes to the Crystal Semiconductor 4328 D/A converter. The D/A is a single- chip, stereo 18- bit D/A with an integral low- pass reconstruction filter. Left and right channel audio exits the chip on pins 2 and 26 respectively. The digital audio is fed to the chip on pin 18, with pin 19 being the associated bit clock. It is derived from the A/D’s pin 14 L/R clock output.

Motherboard Rev A-E: Audio Output

The analog audio signals output from the D/A pass analog switches U26C, U26D on their way to the balanced output section. This is a shunt type switch so that it introduces no distortion. When the switch is open, the audio passes; when the switch closes, audio is reduced by around 40 dB owing to the voltage divider effect of the switches and R71,72. The low- level “leakage” of audio when the switch is on is intentional, as it allows the user to hear program at a low level when the phone audio CODEC, U21 is allowed to pass it’s telephone audio to the system output. On the other hand, the analog switches U26A,B, when off provide excellent isolation to prevent the CODEC’s output from worsening the system noise. Note that the switches in the phone CODEC path are configured series.

The output stage uses the ubiquitous NE5532 op- amps to create and present a differential output to the XLR connectors. The output audio is passed through L1- L4, which are PI- filters designed to prevent RF from coming into the Zephyr, and digital clock noise from exiting. For highest quality, AC coupling happens only once in the output path – at C35,36 – and these are electrolytic types as currently favored by audio “tweeks.”

A very high- quality headphone amplifier is made from two LM6321 buffers and the associated op- amp. U22 is a digitally adjusted potentiometer. Under control of the CPU, it permits headphone gain adjustment. The digi- pot is placed in the op-amp’s shunt feedback path, so as not to reduce headroom – the pots operate from ±5Vdc power rails, while the audio path is only limited by the ±15 Vdc rails.

Motherboard Rev F: Audio Output

The analog audio signals output from the D/A pass analog switched U26C and U26D (on the motherboard) on their way to the balanced output section. This is a shunt type switch so that It introduces no distortion. When the switch is open the audio passes; when the switch closes the audio is reduced by approximately 40 dB owing to the voltage divider of the switch and R75 and R90. The low- level “leakage” of audio when the switch is on is intentional as it allows the users to hear program audio at a low level when the phone audio CODEC U21 is allowed to pass it’s telephone audio to the system output. On the other hand, when analog switches U26A and U26B are off excellent isolation takes place thereby preventing the phone Codec’s output from worsening system noise. Note that the switches in the phone CODEC’s path are configured in series. The output of the phone CODEC is only available at the analog outputs.

The output stage uses the ubiquitous NE5532 op- amps to create, and present to the XLR output connectors, a differential output. The output is passed through L1- L4 which are pi- filters designed to prevent RF from coming into the Zephyr, and digital clock noise from exiting. For highest fidelity AC coupling only happens once in the output path- at C35 and C36- and these are electrolytic types as currently favored by audio "tweeks".

A very high- quality headphone amplifier is made from two LM6321 buffers and the associated op- amp. U22 is a digitally adjusted potentiometer (digipot) which under control of the CPU allows headphone gain adjustment. The digipot is placed in the op-amp's shunt feedback path, so as not to reduce headroom- the pots operate from the ± 5 Vdc power supply rails whereas the audio path is limited only by the ± 12 Vdc rails.

Motherboard A-F: Phone CODEC

The phone CODEC, U21 is used when standard voice- grade calls are being made via the internal ISDN terminal adapter. This chip has both A/D and D/A as well as input and output low- pass filters. Send audio to the CODEC must originate from the analog inputs. A master clock at 2.048 MHz is applied at pin 27. A 2.028 MHz clock is also applied to pins 2 and 19 as the receive and transmit bit clocks, respectively. The audio, in digital form, enters at pin 15 and leaves at pin 18. The signals at these points are the "PCM highway" format often used in digital telephone systems. These have 32 time slots, each of eight bits audio. The frame sync applied to pins 3 and 16 mark the beginning of the 32 slots. In this application, audio is to be found in the third and fourth slots – ISDN channel B1 in slot 2, and channel B2 in slot 3.

Control of the CODEC is obtained by the CPU signals which are applied to pins 23 - 26. This is a serial control signal which is used for muting, slot selection, etc.

U27 is a specialized chip which provides additional filtering and buffering. It creates a balanced output for the front panels, which get the phone audio for presentation at the internal loudspeaker. This line is RF filtered by L9,10.

Motherboard A-F: PLL

The system requires a very high- quality PLL *(Phase Locked Loop)* in order to ensure that no significant jitter is passed from the network to the A/D and D/A converters. (In digital audio systems, clock jitter translates to audio noise.) U42 is a precision voltage controlled oscillator, which is buffered by U45 and sent to the XILINX for distribution to the rest of the system. The two signals PHASEC- VCO and PHASEC- REF are compared in U41 and a DC control voltage is generated to set the VCO frequency at pin 10. PHASEC- VCO is derived from the VCO oscillator, divided within the XILINX. PHASEC-REF is a divided- down version of the network clock. The PLL time- constant is set by R80 and C50.

U41 pin 1 is an output which is used to generate a locked/unlocked signal sent to the CPU. This is applied to a "digital time- constant" within the CPU software to drive the front panel SYNC LED (located near the SEND meter).

A small three- terminal regulator provides clean DC at 7.6V to U41.

Proper operation can be confirmed three ways:

- Pin 10 should be a constant DC voltage (with just a little ripple, maybe) at some value between 1.5 and 6 Volts. A voltage at or near either ground or the 7.6 V positive rail is indication of trouble.

- Pin 3 and 14 should have clock signals which are the same frequency and in- phase, locked to each other.

- Pin 1 should be high.

Motherboard A-F: System Clock

System clock for the CPU and DSPs is generated by U19, a packaged oscillator, operating at 50 MHz. U20 divides this to the various frequencies required. Sections of U18 buffer the 50 MHz signal for driving the DSP chips.

Motherboard A-F: Xilinx

The two Xilinx chips, LCA1 and LCA2, are general purpose logic chips which are loaded with software from the CPU in order to determine their operation characteristics. Software is downloaded serially via XLNX- DIN (DataIN) and XLNX-CCLK (bit CLK).

Once configured these chips make all the serial "pipeline" connections among the DSPs and provide the switching and routing function to the selected digital interface (V.35/ISDN/etc.), as described in Section 5.2. Both local and network reference clocks are also generated within the Xilinx chips.

Since many of the DSP connections to the Xilinx chips are also software- configurable, and since these connections depend on the specific operating mode, Xilinx pins may sometimes be inputs but at other times outputs. This is different from bi- directional pins because the hardware surrounding each Xilinx pin is set by software to be either one way or the other *on a per- configuration basis*.

For troubleshooting purposes the most informative signal to look at is the DONE/*PROG signal on pin 55 of each LCA (on the right- hand side of the chip as you look at the part number, second pin from the bottom). This pin should be low during the software download period (at powerup and during some mode changes), and will go high as the chip becomes operational. If there is a problem with the download process or the chip itself, this pin will remain low – and the chip will remain inactive.

Motherboard A-F: Microprocessor and Peripherals

The CPU and peripherals are a fairly straightforward, textbook Intel 801C88 processing system. U1 is the CPU, U3,46 provided data/address bus de- muxing, U2 is a buffer for the data bus. U17 buffers some of the other signals.

The UART, U9 and PIO, U10 are standard peripherals, using the 801C88's internal chip selects for access. The UART is dual, and handles both the RS- 232 port and the front panel RS- 485 connection via suitable drivers, U13/16 and U14, respectively. Some of the UART's pins can be configured as parallel inputs or outputs and are used for various I/Os. The 8255 is for additional parallel inputs and outputs. The digi- pots, CODEC, Xilinx chips, and analog switches are controlled by some outputs from this chip, while inputs are taken from the LIMIT and PLL circuits for subsequent processing and display by the CPU.

U4 is the main system program store EPROM, and U8 is the main system RAM, with a capacity of 32 K bytes. Chip select for these is generated directly by the 801C88. U5 - 7 are EPROMS which are used for storage of software for the DSPs, and the Xilinx chips. As there is not enough address space available from the CPU for all that is required, we use two bank select lines, BANKSW0 and BANKSW1 to map the 512 K bytes from the EPROMs into 128 K byte slots available. These line normally have activity only during XILINX and/or DSP downloads.

U12 provides further segmentation of the processor's *PCS1 line for application to the DSPs. It also generates a periodical DOGSTRB signal for U11, the watchdog chip.

The watchdog resets the CPU in the event that the DOGSTRB signal stops. (The DOGSTRB is generated by software when it is properly operating. When it "hangs," the strobe signal is no longer produced, and the dog goes into action.) U11 provides another function: It connects the battery to the RAM when system power goes away. This is how the RAM is able to retain data when the unit is powered- down. The chip select for the RAM is also routed through this chip, in on pin 13, out on pin 12, for protection – when power looks to be unreliable, the chip select is gated off, preventing any random memory- destroying writes.

Don't remove the processor from its socket without the proper removal tool, as this may damage the socket and the MPU itself. (Use PLCC extraction tool from AMP, Cat. No. 821566- 1, or the lower cost No. 822154- 1. Available from Digi- Key at 1.800.344.4539.)

Motherboard A-F: Power Supply

Two three- terminal regulators, VR1,2, provide clean power to the A/D and D/A chips. Input voltage is ±12Vdc and output is ±5Vdc.

Motherboard A-F: Parallel I/O

U16 accepts the parallel inputs from the "outside world," and presents them as proper logic signals to the UART parallel input pins. This chip can take inputs up to 30 Vdc. The resistor and diode arrangement on the input allow use with voltages or closures from the driving equipment.

The ULN- 2003A, U15 is a multiple open- collector darlington driver. It converts the logic signals from the UART's parallel outputs to open- collector's to ground for the outside world. These have high current drive capability – 500 ma per package.

DSP Cards

The DSP cards use surface- mount technology in order to achieve high density. Therefore, field servicing of these cards is not supported.

IMPORTANT WARNING!

If the front panel is disconnected from the motherboard, then re-connected, the 87C51 will no longer be in “stand-by” mode and will attempt to draw normal operating current–this can completely drain the motherboard’s battery in a matter of days! The proper sequence for re-connecting the front panel includes powering up the Zephyr immediately after making the front panel-motherboard connection, then powering down. This is the only way to put the 87C51 into the proper “stand-by” mode!

Front Panel: Audio

Balanced audio arrives at AUDIO+/AUDIO- and is converted to single- ended by U16A. One half of the dual digi- pot, U12 controls the gain of this signal. The other half of this pot gets a signal from the 87C51 processor via U4 and an analog low- pass smoothing network consisting of R22,23, C4,6. This is the source of button click audio. The two signals are summed in U16B and amplified by U17. The loudspeaker is located behind the dial buttons, which have enough “slop” in their panel holes to permit the audio to pass without tooooo much resistance.

Front Panel: Processor and Digital

The front panel communicates with the motherboard via an RS- 485 balanced serial data link. The 87C51 has an internal serial UART subsection. Pushbuttons and LEDs are interfaced in multiplexed form and converted to/from appropriate serial commands. U8 - 11 are the LED column drivers, while U15 is the row driver. The LCD contrast and backlight are controlled by the 87C51 through a digi- pot.

The 87C51 is monitored by a watchdog timer in U13. In the event of software failure (perhaps due to static electricity, or bad hardware) U13 will not have “activity” on pin 11- the front panel’s equivalent to the motherboard’s DOGSTRB signal. U13 will then attempt to reset the 87C51 via pin 12.

U13 is also the battery backup controller for the 87C51. At the onset of powering down the Zephyr U13 forces the 87C51 into a low- current “stand- by” mode. In this mode the whole front panel only draws a few microamps, and can safely be driven by the motherboard’s lithium battery. Once the main 5- volt power falls below 3 volts, U13 switches to battery power, maintaining the memory in the 87C51.

The lower four output bits of U3 contain the master scan counter which controls the anode/column half of the LED matrix, the one- of- eight input to the keyboard matrix, and the serial clock to the digi- pots. As this counter cycles through its

possible states the cathode/row half of the LED matrix is set appropriately, the four outputs from the keyboard matrix are read in, and the data (if any) is clocked out to the digi- pots. If there is a problem with the scan counter, all three functions will be lost – the LEDs, the keyboard, and control over the digi- pots.

ISDN S&U Interface Card (USA only)

Power Supply

The 48 volt power supply circuit (distinguished by U5, the Maxim MAX641) is used to power either an external NT1 (if used) or an ISDN telephone which can be plugged into the Zephyr.

The PS- 2 power available on Pins 7 and 8 of the S connector must be enabled on S&U interface cards Rev B and later. To do so:

- Find the jumper block labeled JP- 1 near the top of the board, on the component side. If the jumper is on pins 2&3 PS- 2 power is off (default). Moving the jumper to pins 1&2 will enable PS- 2 power. IN ANY CASE:

> IMPORTANT!
>
> If using an external NT1, or connecting another device, be sure to check power arrangements. The Zephyr provides “PS-2” power on the S/T jack. If your NT1 or other device has an external power supply it is essential that the 2 power leads not be interconnected or *damage to the Zephyr, external equipment, or both, may occur*. Contact Telos Systems customer support for additional information, if needed

S-Interface

Connector J1, the 8 pin modular jack (labeled Phone/NT1), is used to connect an external NT1 or an ISDN phone to the Zephyr™. Transformers T1, T2, T4, and T5 are used to isolate outside telephone lines from the Zephyr circuitry, as well as step up/down the voltages of analog signals. The Mitel MT8931B (U2) is used as the S-Interface which defines the Zephyr as a Terminal Device (TA). It transmits the coded audio data to the NT1 (network termination), whether internal or external. If an ISDN phone is used, the phone and the Mitel chip share the same communication lines via Time Division Multiplexing, the decoding of which is taken care of by the internal NT1.

U-Interface (NT1)

Connector J2, the 6 pin modular jack (labeled Line/Direct), is used to connect the internal NT1 on the card to the Central Office (Telephone Company). Transformer T3, the Valor PT4084, is used to isolate the Zephyr from the telephone line. The entire NT1 function is provided by the AT&T T7256 chip. 4 wires form the S/T interface of the NT1, and another 2 form the U- Interface.

The U Card must be plugged into slot P1 (furthest right) on the Zephyr motherboard, and system software revision 2.56 or later must by used (EPROM U4). The card can either be used alone as a complete NT1 and TA (most common) or with an external NT1 (e.g. Tone Commander) in the same way as the older ISDN card was used.

Power Supply Detail

The power supply circuit should provide a minimum of 48 volts (50- 53 V typ.) to pin 8 of J1, the S- Interface connector. It should maintain this minimum voltage even under loaded conditions (phone or NT1 attached). Diode D18 is used to prevent a surge of current INTO the Zephyr should a self powered NT1 or ISDN phone be plugged in. The cathode of D18 should be toward the connector.

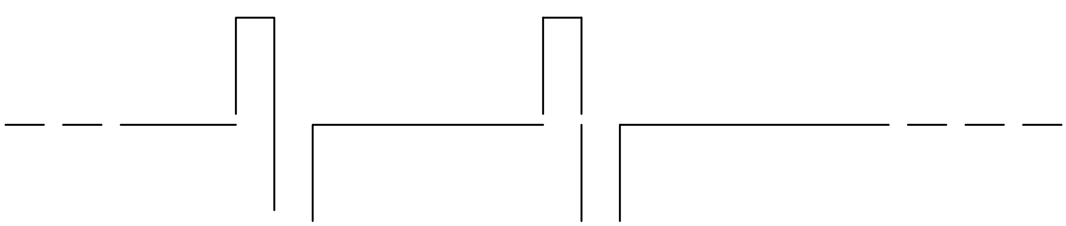
The Maxim chip (U5) monitors the output voltage and adjusts circuit parameters to keep it within range. The control (or "gate") voltage to Q1 is provided by pin 6 of U5 and turns on the FET causing current to flow through the inductor L1. The gate voltage is then removed and the inductor will try to maintain the current flow, charging up C1 with a burst of current. The voltage on C1 is kept from leaking back into the charging circuit by the zener diode D1. The gate control signal switches at a frequency of approximately 50 kHz. Some causes of improper output voltage are:

- Leaky Capacitor C1
- Bad Maxim chip U5 (gate signal will be missing or low in amplitude).
- Blown or leaky Q1 (low or missing current flow through L1 with the gate signal present).
- Improper values for R1 and R2
- Diode D18 backwards or missing

S-Interface Detail

The Mitel MT8931 (U2) is responsible for changing the digital coded audio signal to the ISDN S/T signaling protocol, as well as performing some of the ISDN handshaking with the central office (sending SPIDS, receiving TEI's (Terminal Endpoint Identifier), etc.). The S- Interface is characterized by a 4- wire connection (polarized receive and transmit pairs).

The Mitel chip transmits on pins 3 and 6 of the connector J1. With no NT1 connected, and the U- Interface unplugged, the "Info 0" waveform (see Fig. 1) should be present on each of these pins.



"INFO 0" Waveform

If this signal is not present reboot the ISDN software from the front panel. If the signal still does not appear probe pin 26 of the Mitel chip. If no signal is present, replace the

Mitel chip. If the INFO 0 signal *is* present at the chip, possibilities include cold solder joints, shorted diodes (D2, D3, D4, D5), or a broken wire inside transformer T1.

The Mitel chip uses crystal Y1 for internal timing. A 4 MHz signal should be present on pins 7 and 8 of U2. If this signal is not present, replace Y1.

The other half of the S/T Interface is the NT1 chip (AT&T T7256). The S/T Bus signals from the Mitel chip pass through T4 and T5 to the 4 wire connection on the AT&T chip. Pins 23 and 24 (TNR and TPR) make up the NT1 transmit pair, and pins 26 and 27 (RNR and RPR) are the receiver. INFO 0 should appear at the receive pins with no cables attached.

The diode bridge networks are used for input protection, and the resistors R20, R21, R22, R23, R24, R25, R26 and R27 are used for signal conditioning on the analog S/T signals.

U-Interface Detail

The U- Interface is a 2- wire (ring and tip - to be consistent with analog terminology) signaling protocol proscribed by the ISDN standards. The U is the last interface that a subscriber must deal with between the terminal equipment (Zephyr, telephone, etc.) and the central office. An NT1 is used to change from the 2- wire standard to the 4- wire S/T Bus that all terminal equipment uses. The new ISDN card provides the NT1 internal to the Zephyr, allowing the user to plug the 2- wire U- Interface directly into the Zephyr. The entire U- Interface (and NT1) portion of the card is maintained by the AT&T 7256 chip (U4).

The U- Interface wires come into the card on pins 3 and 4 of the 6 pin modular jack (J2) labeled Line/Direct. D17 (a Teccor® Sidactor™) is used for input overvoltage protection (i.e. lightning, etc.). T3, a Valor PT4084, isolates the line from the circuitry, and provides some signal conditioning. C18 and C19 are filter caps to remove high frequency noise from the analog signals. Pins 31 and 36 (HN and HP) are the inputs for the U- Interface, and pins 32 and 35 are the transmitter outputs. The receiver and transmitter share the same copper pair for signaling.

The AT&T chip runs off the 15.36 MHz crystal, Y2. This oscillator signal must be present for the onboard logic and PLL to function. The chip will lock to the network clock rate for signal synchronization.

U7 (LH1465AB) is used to extract certain “maintenance” pulse from the ISDN bitstream. The pulses are transmitted to the AT&T chip through the opto- isolator, U6 (6N139). The NT1 chip will use these maintenance pulse in communication with the central office during execution of special functions (e.g. loopback).

Indicators and Information:

There are two indicators of the status of the ISDN connection: the front panel, and the rear panel LED. The LED will blink at different rates to indicate status of the NT1 and its connection to the central office. If the NT1 is completely dead or otherwise inactive the LED will remain in the OFF state. If the AT&T chip is alive, it will try to activate a connection with the central office. This is indicated by a rapidly blinking LED (about 5 Hz). If the U- Interface cable is properly attached and the NT1 chip can contact the central office, the LED will blink slowly (about 1 Hz). The LED will come on solid when the NT1 chip makes a successful connection with the Mitel S Chip, and the appropriate handshaking is completed.

The front panel will indicate the status of the connection as well. If no cables are attached, or the card is completely unable to activate a connection with an NT1 (internal or external) the ISDN message will remain at INACT indefinitely. Once the Mitel chip makes contact with an NT1, the ISDN message will read INIT.

This message will stay on the screen until all appropriate handshaking and protocol exchange is complete with the central office (i.e. the C.O. recognizes the Zephyr as a piece of terminal equipment with the appropriate software and then assigns a network address). When this stage is finished, and the Zephyr is ready to make (or answer) a call, the READY message will be displayed. At this point, it is probable that the ISDN card is working properly, and the correct ISDN software is running in the Zephyr.

ISDN S Only Interface Card (Non-USA)

This card is required in parts of the world where the telephone authority provides the NT1 interface unit.

It's circuitry is a sub- set of the S/U version, as there is no power supply or NT1 capability.

V.35/X.21 Interface Module

The V.35/X.21 interface card has two independent sections for buffering input and output of the balanced digital I/O streams. Additionally, a UART offers the possibility to operate terminal adapters which require RS- 232 serial control.

U2,3 are RS232 driver and receiver chips. They have internal "charge- pump" circuitry to generate the required plus and minus voltages from the 5 Vdc input as called- for in the RS- 232 specification (the spec. says between 3 and 12 Volts – these chips generate 9 Vdc).

U6,7 are balanced drivers for the V.35/X.21. U4,5 are quadruple balanced receivers.

The jumper J1 selects which interrupt line is pulsed from the UART when it needs to signal to the CPU. **This should always be in the INT1 position.**

AES/EBU Interface Card

The AES/EBU card is a plug in expansion for Zephyrs with motherboards of Rev. F or later. The card allows the Zephyr to use the standard AES/EBU digital audio interface (provided on many pieces of studio equipment - DAT and CD players are two examples). The card accepts digital input and provides digital audio output, both of which can be synched to a variety of sources. Sampling rates of 32kHz, 44.1kHz, and 48kHz are available on both the input and output sides. The physical interface is provided via the DB- 9 connector on the card edge as well as the second set of XLR connectors on the Zephyr back panel. AES/EBU card functionality is accessed through the LCD user interface, and through the RS232 serial interface.

There are four major sections of the AES/EBU card:

1. The AES/EBU interface circuitry: U9 and U5 (CS8411) are AES/EBU receivers which translate AES/EBU signals into the more useful three signal digital interface (clock, data, frame- sync). U4 (CS8401) is the AES/EBU transmitter chip which generates the AES/EBU compatible signal from the 3- wire interface. Both the receivers and the transmitter are separated from the outside world by transformers (T1- T4).

2. The Sample Rate Conversion circuitry: U1, U2, U3, and U6 (AD1890/1) convert the sample rates of digital audio signals from the AES/EBU signals to the those used internally by the Zephyr, and vice- versa. The chips can be placed in the audio path or removed as needed through the control circuitry.

3. The Control Circuitry: U7 (PAL26V12) is used to allow the Zephyr to address the Xilinx chip (LCA1) and the AES/EBU chips. It generates the appropriate chip select signals for each chip according to where it is located in Zephyr memory. LCA1 is responsible for routing the sample rate converter chips into and out of the digital audio chain. This is accomplished by a series of switches inside the chip which are addressed individually by the Zephyr microprocessor. In addition, the 11.28MHz crystal is routed through the Xilinx chip to provide the 44.1kHz sampling rate not normally supported by the Zephyr.

Connector Descriptions

1. AES- IN: This is the XLR connector for the AES/EBU input from a digital audio source. The Zephyr can automatically adjust the sample of the incoming signal to match the internal Zephyr clocks, or this function can be bypassed if the AES/EBU signal is already synchronized to the digital network.

2. AES- OUT: This connector is used to provide the AES/EBU output signal from the Zephyr receive path. The audio is identical to the Zephyr analog output, but the sample rate is programmable.

3. SYNC- IN: This is used to provide an external AES/EBU signal which the Zephyr can then be synchronized to. Primarily for studios with a master clock, this connector accepts any AES/EBU signal, and uses only the clock.

4. SYNC- OUT: An AES/EBU output that is simply the SYNC- IN routed back to the outside world. This is useful in daisy- chaining several AES/EBU compatible devices with one SYNC signal.

Power Supply

The power supply has a universal AC input, accepting a continuous input range of 100- 240VAC, 50/60Hz. Its outputs provides three separate voltages: a +5V supply which powers all the digital circuits, a ±12V output supplies basic audio circuits and is further regulated on the motherboard to ±5V for critical audio functions like the A/D and D/A sections. A separate small linear regulator supplies the critical PLL power.

FIRMWARE EPROMS

The operation of the Zephyr nearly entirely determined by the software contained in the four EPROMs, U4- 7.

U4 is primarily responsible for the control and ISDN functions, while the others determine the audio characteristics.

How to get version Info (without opening box)

The final item on the UTILITY menu is a read- only display. The display contains information you will need to provide Telos Customer Support should you call for assistance. Note: firmware versions prior to Rev 2.50 do not have this screen. In that case the version number can be observed during boot up.

```
TELOS Systems
ISO/MPEG codec
V2.68/2.00/2.25
July 26 1996, #143
```

Firmware version is on this line.

On the third line, three numbers provide complete version numbers of the firmware installed. The first number is the system firmware, the second is the ISDN firmware, and the third is the DSP and XILINX firmware.

The fourth line of the display has a date and code number. These refer to the firmware and do not reflect the date of manufacture of your Zephyr.

EPROM Replacement

From time- to- time, Telos releases new software in order to provide additional capabilities or to correct bugs.

The EPROMs are socketed to permit field upgrade. To change these chips, the ISDN card must be removed – a simple process of removing the rear panel retaining hex screws and lifting the PCB out. Then, the EPROMs can be pulled from the sockets by using a “Chip puller”.

When the new chips are inserted, take care to ascertain that all pins properly enter the socket holes, and that the EPROM is oriented with the notched end to the left (don’t use the labels as a guide). For best results use an appropriate insertion tool. We see problems sometimes with intermittent units; upon investigation, we discover that the pins are folded under the chip – making contact most of the time, but not very reliably, of course!

SECTION 12

SCHEMATICS

SCHEMATICS

1. Zephyr Motherboard: CPU
2. Zephyr Motherboard: Main- Rev A- E
3. Zephyr Motherboard:Main- Rev F
4. Zephyr Motherboard: Audio- Rev A- E
5. Zephyr Motherboard:Audio- Rev F
6. Zephyr analog I/O B- Audio Board (top level) (Rev F, only)
7. Zephyr analog I/O B- Audio Board A to D (Rev F, only)
8. Zephyr analog I/O B- Audio Board D to A (Rev F, only)
9. Zephyr Front Panel
10. ISDN Interface Card
11. AES/EBU Card
12. AES/EBU Rear- Panel PCB
13. V.35/X.21 Interface Card
14. V.35/X.21 Cables
15. V.35 Connector and Signal Designation

SECTION 13

MANUFACTURER'S DATA SHEETS

MANUFACTURER’S DATA SHEETS

These are the parts whose sheets we find we most refer to, and are chosen to offer the maximum insight and aid when troubleshooting is required.

1. Computer Products NFS40- 7628 Power Supply
2. Crystal Semiconductor CS5338 A/D converter (Rev A- E)
3. Crystal Semiconductor CS5390 A/D Converter (Rev F)
4. Crystal Semiconductor CS4328 D/A converter
5. National DS16F95 Balanced Driver
6. MAX693 Watchdog
7. CD4046BC PLL
8. Dallas DS1267 Digital Pot
9. AT&T T7256 Single chip NT1 Transceiver

SECTION 14

SPECIFICATIONS AND WARRANTY

SPECIFICATIONS (The following specifications are for motherboard revision F)

General

Full duplex, high- fidelity codec using ISO/MPEG Layer III, ISO/MPEG Layer II, and G.722, fully compliant with international standards. Optional integrated ISDN and V.35 interfaces available.

AC Power: 100- 240 volts, 50/60Hz Approx. 150 Watts peak

Dimensions (inches): 17 1/8 wide x 12 1/2 deep x 3 1/2 high; Standard 2 RU x 19” front panel

Shipping Weight: 17Lbs

ISDN Connectivity

Compatibility with National ISDN- 1, AT&T 5ESS Point- to- Point lines; DMS100 Custom; EuroISDN- 2 (ETS300) protocols. Selected by front panel control with no EPROM change required to accommodate the various protocols.

USA and Canada

- Integral NT1 for direct connection to ISDN line via the U interface with an accessible S bus jack for special applications.
- Provides PS- 2 power (48 VDC, minimum) on S interface pins 7&8

Other Countries: S interface only.

Non-ISDN Network Interface

Universal connection for V.35 or X.21 standard. Mini Amp- SCSI type connector interfaces to CSU via adapter cable.

Frequency Response

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

- ISO/MPEG Layer III, all modes:

  20- 14,900Hz at 32kHz Fs.

  20- 20kHz at 48kHz Fs.

- ISO/MPEG Layer II, mono:

  20- 7.8kHz at 56kbps network rate @ 48kHz Fs.

  20- 9.8kHz at 64kbps network rate @ 48kHz Fs.

  20- 20kHz at 112 or 128 kbps network rate @ 48kHz Fs.

  20- 8.6kHz @24kHz Fs

- ISO/MPEG Layer II, joint stereo:

  20- 20kHz @ 48kHz Fs.

- G.722

  25- 7 kHz (+1/- 3dB).

THD+N (Layer II and Layer III end to end)

0.009% (Limiter out).

0.02% (Limiter in).

Dynamic Range (A weighting, limiter defeated, Layer II and Layer III end to end)

94 dB

Send Input

Active balanced, with RF protection.

LINE: - 15 to +4 dBu nominal level. Clip point: +24 dBu.

MIC: - 68 to - 35 dBu level.

Bridging, approximately 100K• x 2 impedance.

XLR female, pin 2 high.

Limiter

Internal limiter on send audio. Prevents clipping in A/D converter.

Receive Output

Active differential.

Level: Rear panel switch- selectable for - 10 or +4 dBu, nominal.

Impedance: 100• x 2.

XLR male, pin 2 high.

Line Bitrates

56 or 64 kbps per channel, front panel selectable.

Multiplex/Demultiplex

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

- Layer III: FHG/Telos protocol.

- Layer II: CCS CDQ protocol compatible.

ISDN Voice Telephone Mode

G.711 standard, uLaw or A- Law. 300- 3,400Hz.

Serial Control and Ancillary Data

RS- 232.

- Asynchronous 8 bits, no parity, 1 stop bit. 300- 19,200 baud.
- Ancillary data on LIII only; maximum 9,600 baud.

Control Ports (end- to- end contact closure emulation)

- 4 inputs and 4 outputs
- Additional status output

Inputs: Open collector closure to ground

Outputs: Sink up to 400ma current to ground

Optional AES/EBU Digital Input/Output Interface

Sample rates supported: 32, 44.1, and 48kHz.

Rate conversion: Input and output independently selectable. Can be bypassed.

Input clock: AES input, network clock, external.

Output clock: AES input, internal clock generator, external.

Digital Converter Resolution

Send Input (A/D)

- L III and L II 20 bits
- G.722 16 bits

Receive Output (D/A)

- L III and L II 18 bits
- G.722 16 bits

ZEPHYR LIMITED WARRANTY

This Warranty covers “the Products,” which are defined as the various audio equipment, parts, software and accessories manufactured, sold and/or distributed by TLS Corp., d/b/a Telos Systems (hereinafter “Telos Systems”).

With the exception of software- only items, the Products are warranted to be free from defects in material and workmanship for a period of one year from the date of receipt by the end- user. Software- only items are warranted to be free from defects in material and workmanship for a period of 90 days from the date of receipt by the end-user.

This warranty is void if the Product is subject to Acts of God, including (without limitation) lightning; improper installation or misuse, including (without limitation) the failure to use telephone and power line surge protection devices; accident; neglect or damage.

EXCEPT FOR THE ABOVE-STATED WARRANTY, TELOS SYSTEMS MAKES NO WARRANTIES, EXPRESS OR IMPLIED (INCLUDING IMPLIED WARRANTIES OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE).

In no event will Telos Systems, its employees, agents or authorized dealers be liable for incidental or consequential damages, or for loss, damage, or expense directly or indirectly arising from the use of any Product or the inability to use any Product either separately or in combination with other equipment or materials, or from any other cause.

In order to invoke this Warranty, notice of a warranty claim must be received by Telos Systems within the above- stated warranty period and warranty coverage must be authorized by Telos Systems. If Telos Systems authorizes the performance of warranty service, the defective Product must be delivered, shipping prepaid, to: Telos Systems, 2101 Superior Avenue, Cleveland, Ohio 44114.

Telos Systems at its option will either repair or replace the Product and such action shall be the full extent of Telos Systems’ obligation under this Warranty. After the Product is repaired or replaced, Telos Systems will return it to the party that sent the Product and Telos Systems will pay for the cost of shipping.

Telos Systems’ authorized dealers are not authorized to assume for Telos Systems any additional obligations or liabilities in connection with the dealers’ sale of the Products.

Telos products are to be used with registered protective interface devices which satisfy regulatory requirements in their country of use.

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SECTION 15

APPENDICES

ABOUT AUDIO LEVELS

“ Nominal” Levels

The following will help to understand what we mean by *nominal*: A mixing console has it’s analog VU meter calibrated so that a test tone set to the 0 dB red- green junction outputs +4 dBu to the Zephyr. 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 the Zephyr.

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. The Zephyr, 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 600•. The 0 dBu value of 0.7746 volts is the voltage across a 600• resistor when exactly one milliwatt is being dissipated in the resistor, which means that under the condition that the measurement is in a 600• circuit, dBu and dBm are numerically equivalent.

Which brings us to our concern here: How does the Zephyr behave when interfaced with various types of equipment?

Unity gain is obtained when:

- The Send level is set to +4 dBu, as indicated on the LCD.
- The rear panel - 10/+4 output switch is set to +4.
- The load impedance is high.

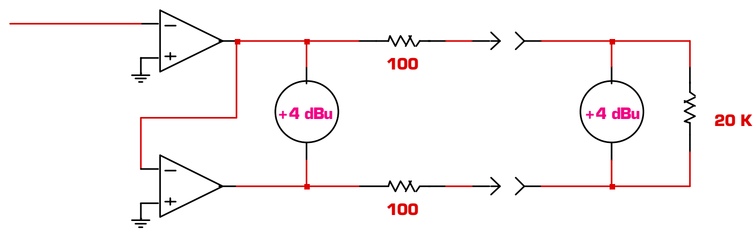
HOT TIP!

In this condition, the system should be used with nominal +4 dBu program signals for proper headroom and signal-to-noise performance.

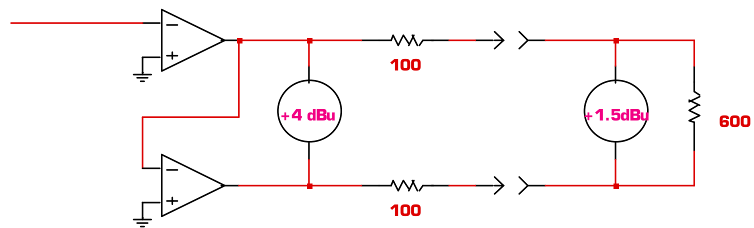
At the Zephyr send input, modern source equipment with metering designed for a high impedance load will behave as expected – the Zephyr’s input meter will correspond to the source’s output meter. Older equipment which has a 600• source impedance and output metering calibrated for a 600• load will appear to send 6 dB more level. This can be corrected by adding an external 600• terminating resistor, or by simply adjusting the send level, either at the Zephyr or at the source equipment.

At the Zephyr receive output, there is a 200• source impedance. When connected to a high impedance input, there will be the expected level. However, if the output is terminated in 600•, 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 600•.

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. The Zephyr's voltage level is conveyed to the load with no noticeable attenuation.



When a 600 • load is used, the voltage divides, with 2.5 dB being lost in the Zephyr's internal source resistance. Note that, in this case, the meter across the 600 • 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 the Zephyr with other codecs.

The lists below, compiled in *January 1997,* offers the most accurate information we have as of this date. 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 CDQ2000	CCS CDQ1000	CCS CDQPrima	CCS Micro 56 Micro 64	COMREX DX200	COMREX DXP/DXR NEXUS	INTRAPLEX 4400
LAYER 3	N/S	N/S	*	N/S	N/S	N/S	N/S
LAYER 2							
J-STEREO	YES	N/S	YES	N/S	YES	N/S	N O
DUAL	YES	N/S	YES	N/S	YES	N/S	N/S
MONO	YES	YES	YES	N/S	YES	N/S	N/S
128 MONO	YES	N/S	YES	N/S	YES	N/S	N/S
G.722	N/S	YES	YES	YES	YES	YES	N/S

GENERAL NOTES:

•YES = Compatible. Zephyr Layer II operation requires Layer III+II firmware and hardware.

•N/S=Not supported by other codec.

•Zephyr supports ISO/MPEG Layer II at 48kHz sampling only.

•Zephyr supports “CCS CDQ protocol compatible” channel splitting for ISO/MPEG Layer II stereo and mono 128 modes.

•Zephyr models 9200,9201, or 9202 required for stereo and mono-128 modes

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 Zephyr. Terminal adapter must have “BONDING” turned off and be set to 64/56Kbps.

•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”. * Newer Primas also support layer 3 and are reportedly compatible.

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 ACD3001	DIALOG4 MUSICTAXI	DIALOG4 REPORTERSet	GLEN SOUND	NAGRA ARES-C	RE 660/661
LAYER 3	N/S	LIMITED	MONO	N/S	N/S	N/S
LAYER 2						
J-STEREO	N/S	NO	NO	N/S	NO	NO
DUAL	N/S	NO	NO	N/S	NO	NO
MONO	YES	NO	NO	YES	YES	YES
128 MONO	N/S	NO	NO	N/S	NO	NO
G.722	YES	YES	YES	YES	YES	YES

GENERAL NOTES:

•YES = Compatible. Zephyr Layer II operation requires Layer III+II firmware and hardware.

•N/S=Not supported by other codec.

•Zephyr supports ISO/MPEG Layer II at 48kHz sampling only.

•Zephyr supports "CCS CDQ protocol compatible" channel splitting for ISO/MPEG Layer II stereo and mono 128 modes.

•Zephyr models 9200,9201, or 9202 required for stereo and mono-128 modes

DIALOG4 NOTES:

•Zephyr can communicate with MUSICTAXI in the modes indicated. Dropouts will occur in the MUSICTAXI-to-Zephyr path; the Zephyr-to-MUSICTAXI path will operate normally. MUSICTAXI must be set to "Configuration/ISDN accept mode" to "ALL" AND "Configuration/ISDN mode" to "RI", "DNRI", or "DNR" depending on firmware version. Zephyr firmware version 2.4//2.14 recommended.

NAGRA NOTE:

•Nagra "SYNC" mode must be set to "SRT".

Comprehensive Zephyr 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
-	:	not supported by product

- **AETA**

  Incompatible because of proprietary L- II implementation?

- **AEQ**

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : yes!

  L2- Stereo : ?

  L2- M128 : ?

  G.722 : ?

- **CCS CDQ1000**

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : yes!

  L2- Stereo : -

  L2- M128 : -

  G.722 : yes! Zephyr can always receive. CDQ must have H.221 switched off in order to receive from Zephyr

- **CCS CDQ2000**

  CDQ2000: Decoder DIP switch 6 up (decoder independent

  from encoder).

  L3- Mono : -

  L3- Stereo : -

L2- Mono : yes!

L2- Stereo : yes!

L2- M128 : yes!

G.722 : -

- **CCS CDQ2001**

  See CDQ2000. 48kHz sample rate.

- **CCS Prima**

  Prima:

  Frontpanel 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 reportedly compatible)

  L3- Stereo : ? (some units have layer III and are reportedly compatible)

  L2- Mono : yes!

  L2- Stereo : yes!

  L2- M128 : yes!

  G.722 : yes!

- **Comrex Layer- II DX- 200**

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : yes!

  L2- Stereo : yes! Comrex: 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**

  MusicTaxi:

  To accept Zephyr calls set "Configuration/ISDN accept mode"

  to ALL and "Configuration/ISDN mode" to RI.

  L3- Mono : yes+, but dropouts MusicTaxi - > Zephyr because of D4's asynchronous mode.

L3- Stereo : yes+, but see above

L2- Mono : ?

L2- Stereo : ?

L2- M128 : ?

G.722 : yes?

* Zephyr firmware rev 2.40/2.0/2.14 or later recommended

- **Dialog4 ReporterSet**

  L3- Mono : ?

  L3- Stereo : ?

  L2- Mono : ?

  L2- Stereo : ?

  L2- M128 : ?

  G.722 : ?

- **EELA**

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : yes?

  L2- Stereo : no?

  L2- M128 no?

  G.722 : yes?

- **Intraplex**

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : 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 : ?

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 : yes?

  L2- Stereo : yes?

  L2- M128 : yes?

  G.722 : yes?

- **Philips MPR Baby Blue**

  Must be switched to "CCS compatibility" mode for modes that use both B- channels (stereo, mono128)

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : 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 : ?

- **You/Com**

  L3- Mono : -

  L3- Stereo : -

  L2- Mono : yes+

  L2- Stereo : yes+

  L2- M128 : yes?

  G.722 : yes?

FINDING PUBLIC ZEPHYR SITES

When you need to find a Zephyr or compatible 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 high- end recording studios; some are broadcast stations; while others are small home studios. Some come with award- winning 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 the Zephyr and ISDN, including site pointers.

- **AudioBahn**

  A list of broadcasters and sound studios using codecs, maintained by Jay Rose's Digital Playroom.

  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.

  - **FTP:**
    - ftp.vortex.com:/audio/audio- networks/codec.list
  - **WWW:**
    http://www.tiac.net/users/jcrose/audiobahn
  - **Paper:**
    For a printout of the current list, send a stamped self- addressed envelope to: Jay Rose, 20 Marion Street, Brookline MA 02146- 4905. Allow time for mailing... or send an overnight airbill on your account with Fedex, UPS, or Airborne.

  The list is updated as appropriate, approximately once a month.

  Comments or additions may be sent by e- mail to: jcrose@dplay.com.

- **Digifon List**
  Dave Immer

  - **Phone**
    - 203- 254- 0869
    - 203- 256- 5723 (Fax)

- **Email**
    - immer@digifon.com
- **WWW:**
    - http://www.digifon.com

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 Zephyr. Those that cannot (mostly APT- X, but some Dolby) are able to do so via the “translation” service offered by Ednet and others. To register your site, or to order a copy of the list, you may use the form included with your unit. The cost is $19/issue (as of November ‘95).

- **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 dealer.

List of Known Working SPIDs by Telephone Company

Your SPIDs may be different! 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 is being phased in during 1996 and is also worth trying. Incorrect SPIDS will be indicated by a line status of "Wait, Init" on the Zephyr. Often the Telco will leave the last "00" or "01" off the end of your SPID when giving it to you. For the latest list check Telos Systems' web page. 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

XXXYYYZZZZ0111

or

01YYYZZZZ011

or

XXXYYYZZZZ01

Bell Atlantic

01YYYZZZZ00

or

01YYYZZZZ000

or

01XXXYYYZZZZ000

or

XXXYYYZZZZ100

Bell South

XXXYYYZZZZ0101

or

XXXYYYZZZZ0100

or

XXXYYYZZZZ01

or

01YYYZZZZ0

or

01YYYZZZZ000

Cincinnati 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

XXXYYYZZZZ0101

or

XXXYYYZZZZ0000

Pacific Bell

XXXYYYZZZZ0101

or

SPID1 XXXYYYZZZZ01

SPID2 XXXYYYZZZZ02

or

01YYYZZZZ000

or

SPID1 XXXYYYZZZZ1

SPID2 XXXYYYZZZZ2

or

XXXYYYZZZZ00

or

XXXYYYZZZZ

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

MENU'S AT-A-GLANCE; Utility Menu Summary

Users have asked for a list of the Utility screens and all options to make things easier when walking someone through the menus over the phone. For information on what each *menu item means* see section 8 (Detailed Menu Reference)

Menu Items	Options
Xmt	L3 DUAL
	L3 JSTEREO
	L3 STEREO
	G.722
	L2 HALF/24
	L2 MONO
	L2 MONO128
	L2 DUAL
	L2 JSTEREO
	-
Rcv	L3 STEREO
	L3 MONO
	G.722
	L2
	L2 HALF/24
	-
Rate	56Kbps @ 32kHz
	64Kbps @ 32kHz
	56Kbps @ 48kHz
	64kbps @ 48kHz

Network	ISDN
	V.35
AES In	NO (ANALOG)
	S/R CONVERT
	SYNC TO NET

	-
AES Out	NO CONVERT
	32 KHz
	44.1kHz
	48 KHz
	EXTERNAL
	AES IN

Auto Answer	YES
	NO
	-
Loopback	OFF
	NEAR
	FAR
	-
Status Out	RCV LOCK
	LINE 1
	LINE 2
	LINE 1 & 2
	LINE 1 OR 2

Store Setup	1 ... 50
Category	NAME
	NUM1
	NUM2

SPID 1 & 2 /MSN 1 & 2	SPID/MSN entry fields as required

Directory 1 & 2	Directory number entry fields (rarely required)

Telco	Natl I-1

	ETS300
	PTP

Panic Dial	NO
	1 ... 50
	1-4
Compatibility	Used only for certain subscription services

LCD Contrast	1 ... 9
LCD Backlight	1 ... 9
Ancil Chan	Normal
	None
	Right
	Both
	-
Version Info	2.69/2.00/2.32 <for example>
	-

ISDN BRI, ZEPHYR, AND YOU

This section applies to users in North America and is intended to help you when it comes time to order your ISDN line. We assume you have read the description of ISDN in the manual and are familiar with ISDN's basic concepts. If you have not read that section, doing so now will help you better understand the information that follows.

ISDN (Integrated Services Digital Network) is the information superhighway at your front door. Because ISDN has only been recently introduced, 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 it's 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 the Zephyr menus and LEDs.

The Zephyr's internal interface (sometimes called by it's 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) Residential 800- 204- 7332	http://www.ba.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; large business 702- 688- 7100	
NYNEX	Call your account representative. If you do not know who s/he is call; 800- GET- ISDN (800- 438- 4736)*	http://www.nynex.com
Pacific Bell	800- 4PB- ISDN (800- 472- 4736)	http://www.pacbell.com
	For questions or assistance 403- 944- 8130	
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 visit Telos System's web site at http://www.zephyr.com .

IOC Capability Packages

More and more telcos are using ISDN Ordering Code (IOC) capability packages for ISDN ordering. If your telephone company uses these you need only tell them you need IOC Capability Package "S". If you do not require the Zephyr's ability to call a regular (POTS) telephone you may specify Capability Package "R"

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 the Zephyr, 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 protocols. This number is given to the user by the phone company and *must be entered correctly* into the Zephyr 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 Zephyr 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. The Zephyr allows outgoing voice calls on either of the channels if you have CSV. The Zephyr's voice capability exists on both channels, even when your Zephyr is a mono version. 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.

IIf you do not require the Zephyr's ability to call a regular (POTS) telephone you may omit the CSV option.

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.

The Zephyr provides standard rate adaptation (officially known as CCITT V.110) from 56 to 64 kbps when required. Bit rate adaptation happens automatically within the Zephyr 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" 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 "4 wire" S interface may be on user equipment. Zephyrs shipped outside of the USA and Canada have the "S" interface only, as do units sold prior to 1996. 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 Zephyr. 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 Zephyr 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 10288 for AT&T, 10222 for MCI, or 10333 for Sprint. If the local telephone company insists that you contact the long distance company directly, we advise that you just call and give them your phone numbers, just 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 as follows:

AT&T

800- 222- 0400 Business

800- 222- 0300 Residential

LDDS

LDDS recently announced ISDN availability and has not been tested. Contact Telos customer support for the latest information.

MCI

800- 727- 5555 or 800- 888- 0800 Business

800- 444- 3333 Residential

Sprint

800- 326- 1015

The Faxable ISDN Order Form

Following are forms which can be used to place orders for ISDN lines. These should give the phone company all of 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 when he or she puts in your line and ask the installer to verify with the switch programmer that your line is configured as ordered.

FAXABLE ISDN BRI LINE ORDER FORM

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:____________________________

We request an ISDN Basic Rate Interface (BRI) line for use with the Zephyr hi-fidelity audio codec. This device is used to transmit audio using digital telephone services. It *requires* Circuit Switched Data (CSD). The Zephyr 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.

Zephyr 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 the Zephyr, Telos Systems, at +1 216.241.7225 for any additional required information. Ask for " ISDN Customer Support."

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 the Zephyr

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

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

Terminal Type: A

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.

NOTE for EWSD running NI-97: Switch must be programmed CLID=DN.

Give us:

1) Two SPID numbers, depending upon number of active B channels

2) Two Directory Numbers

Manual History

Version 2.9 First ISDN and Layer II version. MS Word.

Version 2.91 Fixed pagination so that chapter text starts on odd pages.

Version 2.92 Insert added with AES Pin outs and Rev F schematics and specs.

Version 3.0 Reformatted with new fonts, icons. Revised up to V2.69, Rev F circuit descriptions.