TELOS SERIES 2101

Advanced All Digital Multi-line Multi-studio Broadcast Telephone System

USER'S MANUAL Part VI

Troubleshooting and Technical Information

Warranty Statement

Specifications

V 2.0

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1 Technical Data and Troubleshooting

If you are troubleshooting a new 2101 installation, please review the checklists "2101 Pre-commissioning checklist" and "2101 Commissioning Checklist" in Vol. 1 Part II, Section 3.

1.1 A note on rebooting the 2101 Hub

Since the 2101 Hub is a computer, there is a right and a wrong way to reboot it. Here are the ways to restart your 2101 listed in the order of most preferable to least preferable:

1. From the administration computers web browser, navigate to the *Software Update* page and click on the *Reboot* option. Technically, this is not a reboot in that this simply restarts the 2101 and 2101 web server applications. This is typically required if you have made changes to the trunk configuration. This is also the first thing to try if you have a problem effecting multiple studios (if you have a problem that only effects a single studio, you should power cycle that studio interface only)

2. Press and hold the Hub's front panel pushbutton for five seconds, then release. This is a CPU reset and should be used as a last resort, as it could cause registry corruption.

3. Disconnect power from the hub, wait ten seconds and restore power. As with option 2, this should be used as a last resort, as it could cause registry corruption.

If you are finding that frequent reboots are necessary it is important to contact Telos and work with a Support Engineer to determine the root cause of your problems; frequent reboots should not be necessary.

1.1 Overview

A consequence of modern surface-mount construction is that it is no longer possible for local repairs to be made. Special and expensive equipment is required to change parts. As well, today's equipment is very complicated and requires repair technicians to have detailed experience and training, and have access to high-end test equipment.

At the same time, the advent of overnight delivery services means that equipment can be returned to the factory for quick turn-around repair. Therefore, we do not expect you to fix this unit at the component level, and we do not include schematics of the unit in the manual. Please see page III of this manual for proper procedures on returning units for repair.

There is the possibility, however, that you could repair the unit by swapping subsystems such as plug-in modules or power supplies. The Telos customer support crew is standing-by to assist with this, if you need it.

1.2 System Software & Firmware

1.2.1 System Software & Firmware - Studio Interface

Viewing Version Information

To assist with troubleshooting, Telos customer support may ask you to provide some software version numbers. There are two major software components to the system:

- System software – This is the software that can be updated by downloading new software or changing the SIMM in the socket on the motherboard.
- Firmware – This code is analogous to a bootstrap loader and contains certain critical information about the unit. This information can only be updated or changed at the factory. It is stored in a partition of the non-volatile memory on the motherboard. Firmware changes are unlikely.

The system software affects the ISDN procedures, the user interface, general operation, and audio processing functions.

The software version, and software build date can be determined from the start-up screen.

2101 Studio Interface

VX.X. ← Software Version

TELOS SYSTEMS yyyy-mm-dd ← Software Build Date

Replacing Software (Studio Interface & Extended Hybrid/TWO only)

> **IMPORTANT TIP!**
>
> *The software of the Series 2101 Studio Interface and the Series 2101 Hub interact. Therefore, we do not recommend upgrading the software on either component without first consulting Telos Customer Support. When contacting support please provide the current version number for both the 2101 Studio Interface and the 2101 Hub.*

Electronic Download Method

In order to help upgrade to the latest features available, we are pleased to offer software updates over the internet. This process downloads the new software into a secondary bank of your 2101 Studio Interface's memory, leaving the original software intact. You then have the option of running the new software or the old software. Once you have verified that the new software is more suitable to your needs, you may make it your default.

Since the Series 2101 has software running on the System Hub as well as the Studio Interface, you should contact Telos before making upgrades. We will need to know the current version running on the System Hub as well as the version currently running on the Studio Interfaces in order to advise you.

If you do not have the proper connectivity to handle the electronic download method, contact Technical Support to request a flash SIMM be shipped with the desired software. Then follow the steps in the next section to replace the SIMM module. This method is not as convenient as it requires opening the unit.

To upgrade your unit using the electronic download method you will need the following:

- Ethernet connection to the 2101 Studio Interface from a LAN connected to the internet.
- The *Remote Cntrl* selection of the *TCP/IP Setup* menu (accessed from *System* menu) must be set to *All* or *Trusted*.
- The Studio Interface must be programmed with the following information (see Section 7.3.3 for details):

  IP Addr

  This is the IP address for this Studio Interface. As with any computer on an IP network the 2101 Studio Interface must have an IP address before it can be used over the network.

  Subnet

  This is your subnet mask. The subnet mask is to determine the size of your "local" network. All packets addressed to a destination outside this "local" area are sent to the gateway node entered in the next selection.

  Gateway

  This is the IP address of a gateway router connecting you to the internet.

  DNS IP

  This is the IP address of the DNS (Domain Name Server) you will be using.

To download the new software, and view the detailed update information see the Telos web site: www.telos-systems.com.

Manual Replacement Method

The system's software is contained within a flash memory SIMM module, similar in size and form to those used for PC memory. If you need to change this module, follow the procedure below:

1) Power the 2101 Studio Interface off and remove the AC mains power connector.

2) Remove all Philips-head screws from the top cover. Remove 1/16th inch Allen-headed screws from the *top* of the interface card slots (slot A-F) Remove top cover.

3) Remove right old software SIMM

- Turn the right side of the unit towards you (the power supply will be away from you).

- Near the back of the main PC board (PWB) you will see a power harness from the power supply to the motherboard with a 4-pin white Molex style connector and a very small PC board (the SIMM module) directly in front of it.
- Push gently outward on the two metal catches on either end of the SIMM module. It should pop toward you.
- Pivot the top of the SIMM module toward you and then lift it out. Note the grooved notch on the right end.

4) Install the new SIMM module.

- Locate the notch on the SIMM module to your right (towards the rear of the 2101 Studio Interface).
- At a 30 degree angle from vertical, insert the SIMM module into its socket. It should slide in with *no resistance*. The module will be angled towards you.
- Using your two thumbs push on the green area near either end of the SIMM module. The top of the SIMM module should pivot back and the catches on either side should click. If this does not happen remove the SIMM module (see step 3, above) and try again.

5) Reinstall top cover

6) You may need to reprogram some of your settings, check them after installing the new software.

Please return the old SIMM module to Telos, via regular US Mail. Your prompt return of this module will allow us to continue to offer you upgrades in the future, at the lowest possible cost.

1.2.2 System Software & Firmware - 2101 Hub

The software version running on the hub can be determined using the 2101 Configuration web pages.

Updating the 2101 Hub Software

The Telos 2101 Hub software is stored on a re-programmable memory. The Hub uses FTP (File Transfer Protocol) over any IP network to download new software from a FTP server into the memory module. There are two ways to obtain this software:

- You can download the latest Software Updater package with an included FTP server to a computer workstation connected to the LAN of the 2101 Hub.
- You can have the Hub connect through your ISP/Internet connection to our public FTP site directly.

We advise you to contact Telos Customer Support (have your current software version and Serial number when you call) to discuss whether the latest version would better meet your needs.

I- Installing the software from the Updater Package running on a local workstation.

Download the proper updater for your product. The link to our software download page is www.telos-systems.com/?/support/software.htm . You will need to download this application to a Windows NT based computer (Windows NT4, 2000, or XP). For ease of locating your file after downloading may want to just download this to your desktop.

Note: The new updater packages are standalone applications with no installation required. When launched the program starts a simple FTP server with the newest files embedded within it. Previous Software updaters are still available for download if you need to revert back to earlier versions. These older updaters may extract folders and files within your directory structure.

It is important that the download process to the 2101 Hub completes without interruption. If local power is not reliable, you might wish to place your Hub and your workstation on an un-interruptible power supply.

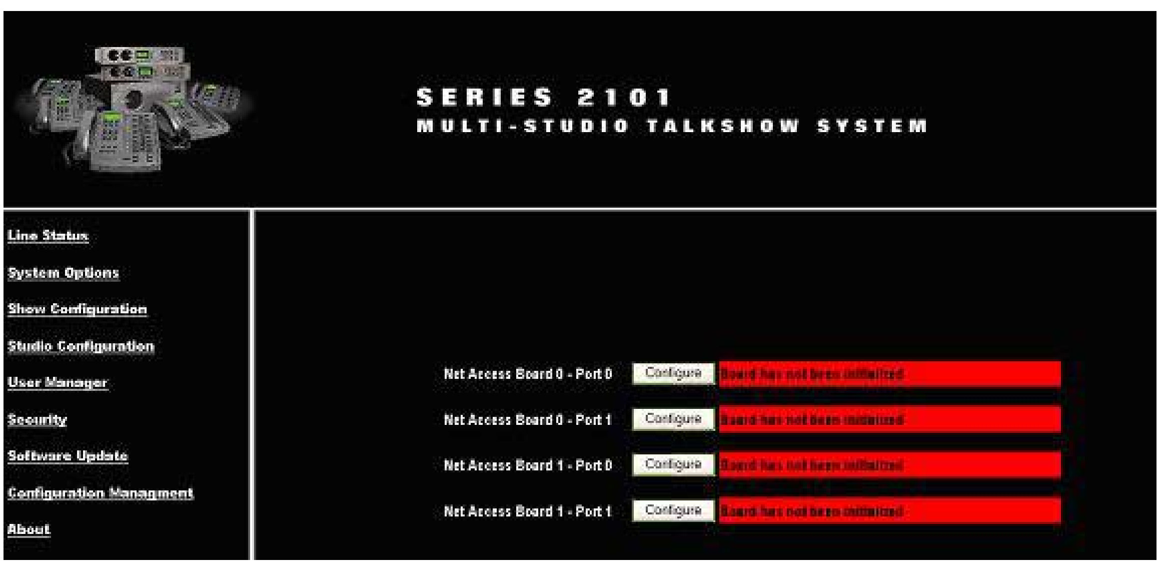
Software update step-by-step (from a Local FTP Server)

1. You will need to know the IP address of the machine that will be running the local FTP Server. You can find out your IP address by going to the MS-DOS window and typing: "ipconfig" and pressing <enter>. The information about this machine's network configuration will be displayed.

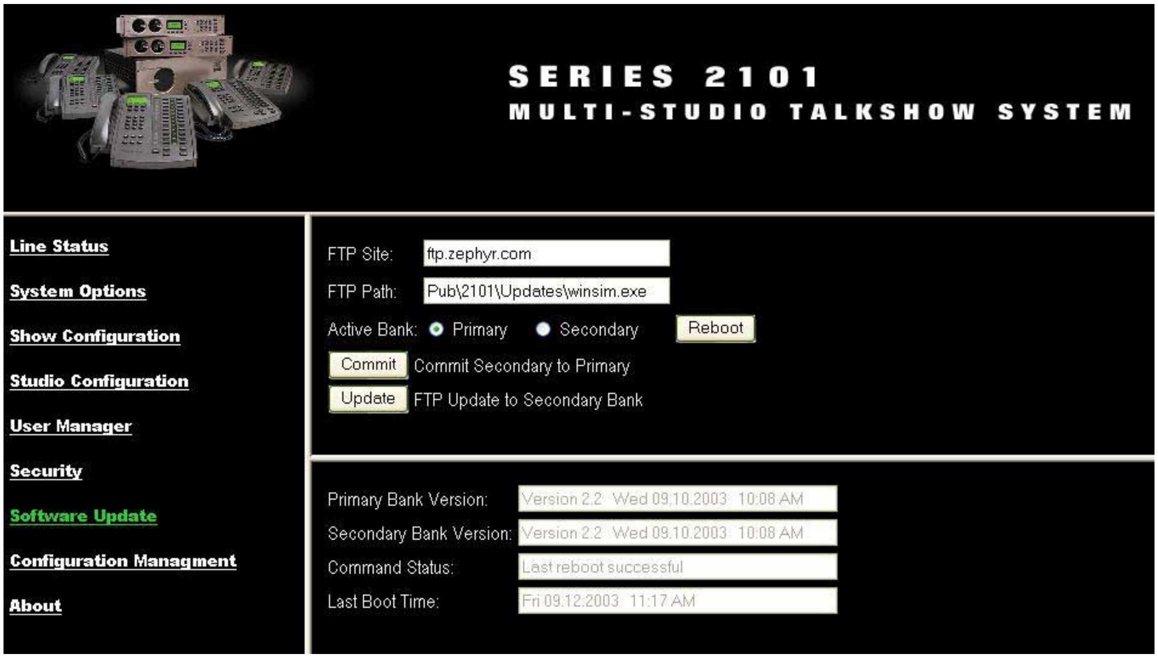
2. Run the Telos Updater package on your workstation. This will start the FTP server program. You can launch the program from your run command line or locate it through Windows Explorer and launch it by double clicking on it.

3. Connect the computer to be used for the update and the 2101 Hub to the same Ethernet network. Note that a direct connection between the Hub and the computer may be made if an Ethernet "Crossover" cable is used (see Appendix 3).

4. Connect to the 2101 Hub Configuration Web Pages and then log in (See Vol. 2, Part III, Section 4.2 for details). The following window will be displayed:



5. Click on the "Software Update link. The following screen will be displayed.



6. In the "*FTP Site*" field enter the address of the FTP server where the new software resides. As you are using the Telos FTP server on a local PC, enter the IP address of this PC. Leave the "FTP Path" to the default value of "pub\2101\updates\winsim.exe".

7. Click the "*Update*" button next to the "*FTP Update to Secondary Bank*". The "Command Status" field should show progress and completion of the FTP update. The version field for the secondary bank should update to reflect the version of the newly downloaded software. If this is not updated, or if there is any error message on the status line, the update did not happen correctly, perhaps due to a communication problem.

8. Click on the "*Active Bank*" button "*Secondary*" then press the "*Reboot*" button, and acknowledge "Yes" to reboot the 2101 application. ***Please note that this will drop any active calls!*** It is not necessary to power cycle the 2101 Hub for this application restart.

9. Verify that the "Last Boot Time" line updates to a new value. This time may not be in sync with local wall clock time, but simply indicates a relative boot time.

10. Lastly, as a double check, verify the new software version running on the Hub is the new software just downloaded into the Secondary bank by selecting "About" link on the left side of the Configuration Web Page. This shows the version of the Configuration program itself, and the software version currently running on the 2101 Hub.

11. At any time you may use the "Commit Secondary to Primary" button to copy the Secondary software bank to the Primary. Until that time, you may always choose to run your previous software in the Primary bank.

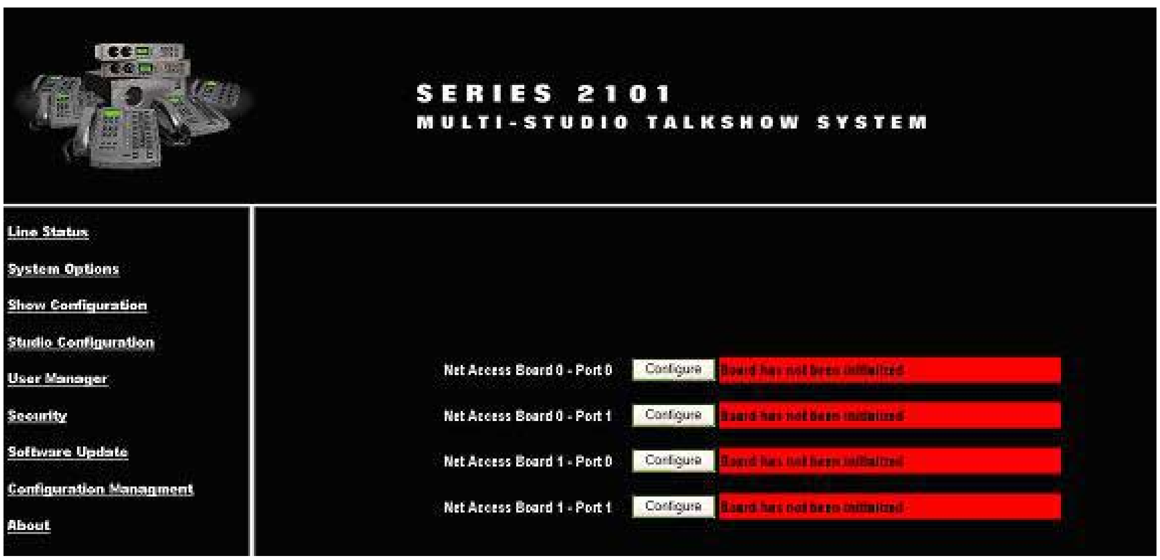
II- Updating directly from the Telos ftp site

This method may be convenient if your firewall allows FTP (usually port 21) through. In this case the 2101 Hub can download the new software directly from the Telos FTP site (ftp.zephyr.com). If your firewall does not allow direct FTP download, see section *I* above.

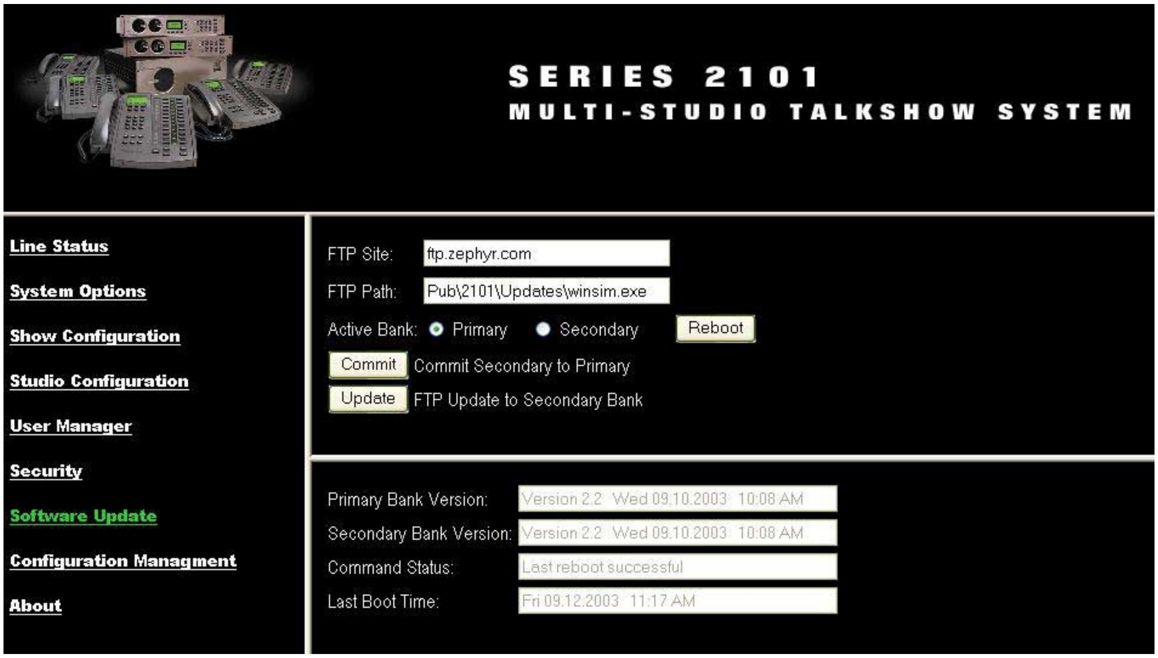
It is important that the download process to the Telos Unit completes without interruption. If local power is not reliable, you might wish to place the Telos unit on an uninterruptible power supply.

Software update step-by-step (from the Telos FTP server)

1. Connect to the 2101 Hub Configuration Web Pages and then log in (See Vol. 2, Part III, Section 4.2 for details). The following window will be displayed:



2. Click on the "Software Update link. The following screen will be displayed.



3. In the "FTP Site" field enter the address of the FTP server where the new software resides. This will normally be *ftp.zephyr.com*. Leave the "FTP Path" to the default value of "pub\2101\updates\winsim.exe".

4. Click the "*Update"* button next to the "*FTP Update to Secondary Bank*". The "Command Status" field should show progress and completion of the FTP update. The version field for the secondary bank should update to reflect the version of the newly downloaded software. If this is not updated, or if there is any error message on the status line, the update did not happen correctly, perhaps due to a communication problem.

5. Click on the "*Active Bank*" button "*Secondary*" then press the "*Reboot*" button, and acknowledge "Yes" to reboot the 2101 application. ***Please note that this will drop any active calls!*** It is *not necessary to power cycle* the 2101 Hub for this application restart.

6. Verify that the "Last Boot Time" line updates to a new value. This time may not be in sync with local wall clock time, but simply indicates a relative boot time.

7. Lastly, as a double check, verify the new software version running on the Hub is the new software just downloaded into the Secondary bank by selecting "About" link on the left side of the Configuration Web Page. This shows the version of the Configuration program itself, and the software version currently running on the 2101 Hub.

8. At any time you may use the "Commit Secondary to Primary" button to copy the Secondary software bank to the Primary. Until that time, you may always choose to run your previous software in the Primary bank.

1.3 Installing Plug-in cards

1.3.1 Installing Plug-in cards – 2101 Studio Interface and Extended Hybrid (Telos Two)

1. Power the 2101 Studio Interface off and remove the AC mains power connector.

2. Remove all Philips-head screws from the top cover. Remove the 1/16th inch Allen-headed screws from the *top* of the interface card slots (slot A-F). Remove top cover.

3. Remove the blank slot cover-plate for the slot(s) to be used by removing the 1/16th inch Allen-headed screw from the *bottom* of the blank cover.

4. Align the connector on the card to be installed with the corresponding connector on the motherboard. If the clearance with the rear panel is too tight loosen the two Phillips head screws that hold the rear panel to the rest of the chassis by a few turns. Push the card straight down. The card mate smoothly with the corresponding connector on the motherboard.

5. If the rear panel screws were loosened tighten those screws now. Install the lower 1/16th inch Allen-headed screw for the *bottom* of the new card. Make sure the top of the card is alinged at the top while tightening the lower screw.

6. Reinstall top cover and the the 1/16th inch Allen-headed screws from the *top* of the interface card slots (slot A-F). Replace remaining screws from the top of the cover.

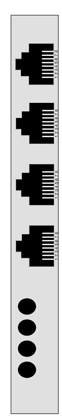
1.3.2 Installing Plug-in cards – 2101 Hub

Identifying the Cards

You will need to install various interface cards in your system. As description of each the various cards is included in the following table:

2101 Hub Interface Cards			
Telos Part Number	Description	Max # in System	Notes:
1701-00049	T-Link Hub I/O Card	8	Has 4 ports for connection to Series 2101 Studio Interfaces. May be installed in slots 1-8
2091-00013	PRI/T1 Telco Interface	2	Supports a single T1 connection or 23B+D PRI connection (1.544 Mbps) as used in North America. May be installed in slots 10&11
2091-00014	Dual PRI/T1 Telco Interface	Cards may be mixed.	Supports two T1 connections or 23B+D PRI connections (2 x 1.544 Mbps) as used in North America. May be installed in slots 10&11
2091-00015	PRI/E1 Telco Interface	2	Supports a single E1 connection or 30B+D PRI connection (2.048 Mbps) as used in Europe. May be installed in slots 10&11
2091-00016	Dual PRI/E1 Telco Interface	Cards may be mixed.	Supports 2 E1 connections or 30B+D PRI connections (2 x 2.048 Mbps) as used in Europe. May be installed in slots 10&11

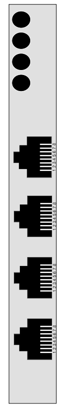
You can identify these cards by looking at the face plate and connectors. The different types of cards are pictured below: You can identify these cards by looking at the face plate and connectors.



The T-Link Hub I/O Card, Rev B Style.

Each card supports connections for up to four Studio Interfaces. The system manages the T-Link connections and any port can be used for any Studio Interface.

The four amber LEDs at the bottom indicate "Loss of Signal" and should be extinguished if a Studio Interface is present and active. The top LED corresponds to the top T-Link port and so forth.

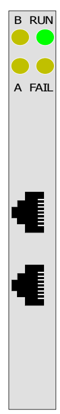


The T-Link Hub I/O Card, Rev A style.

Each card supports connections for up to four Studio Interfaces. The system manages the T-Link connections and any port can be used for any Studio Interface.

The four amber LEDs at the top indicate "Loss of Signal" and should be extinguished if a Studio Interface is present and active. The top LED corresponds to the top T-Link port and so forth.

NOTE: This version requires a custom "Crossover Cable" to connect to the Studio Interface. See Appendix 3.



A Dual Telco Interface Card.

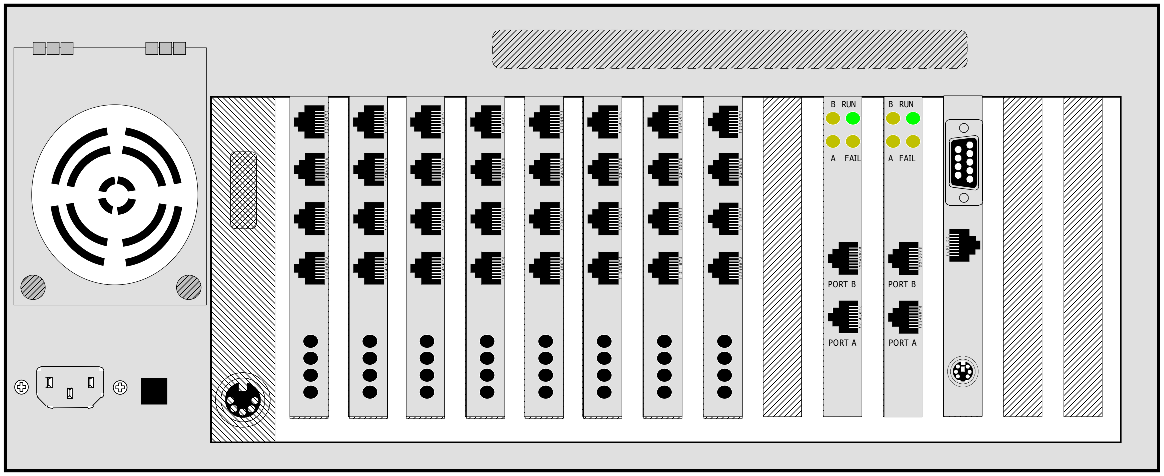
The single Interface card is the same but only has one port.

The Lower port is Port A while the upper port is port B. If only one port is to be used you must use Port A (lower port). Pin 1 is at the top.

The two amber LEDs to the left indicate "Loss of Signal" on Port A or B and should be extinguished if an active line is present.

Installing cards in the Hub

The Series 2101 Hub has 14 slots. We shall refer to them as slots one through fourteen left to right as viewed from the rear of the Hub. Note however, the slots are not numbered on the chassis of the unit.



Rear view of the Series 2101 Hub fully populated. Note the T-Link Studio I/O Cards in Slots One through Eight and the Telco Interface Cards in Slots Ten & Eleven. Position of cards may vary depending on number of Telco cards Installed.

⚠	**IMPORTANT!** *1) Slots 13, & 14 must remain unused* *2) Slots 1,2, 3, 4, 5, 6, 7, 8 (sometimes 9) are for use with T- Link Hub I/O cards* *3) Slots 9 & 10 or 10, & 11 are for use with Telco Trunk Interface cards* *4) Slot 12 is reserved for the Processor/100 Base- T Card (Provided with Hub)*

Follow these steps to install a card in the 2101 Hub:

⚠	**IMPORTANT!** • *The following instructions are to be followed by qualified Technical Personnel ONLY.* • *Handling appropriate for Static Sensitive Electronic Devices (such as personnel grounding straps) must be employed whenever the Hub is open. Damage caused by a failure to do so is not covered by the equipment Warranty.*

1. Disconnect the power cable and any other cables present from the Hub.

2. Remove the four Phillips-head screws that hold the top cover in place (two on each side of the Hub).

3. Lift the cover straight up and set aside.

4. Determine the correct slot for the card(s) to be installed (See above).

5. Remove the blank Slot Cover-plate from the slot(s) to be populated by removing a single Phillip-head screw located at the top of the Slot Cover-plate. Remove the blank Slot Cover-plate by lifting it straight up.

6. Remove the multi-conductor ribbon cable connector plugs from as many of the existing cards as necessary to allow access to install the new card(s).

7. Align the card(s) to be installed with the socket on the main board and the opening in the rear panel of the chassis. Seat the board in place by pushing firmly, straight down. If the board fails to seat properly lift it straight up and out and try again.

8. Reconnect the ribbon cable connectors to each of the cards disconnected in step 6 as well as the newly installed card(s).

9. Using the screws remove in step 5 (above) fasten the card(s) in place.

10. Replace the cover on the Hub and fasten with the four screws removed in step 2. (above).

WARNINGS!

- The interface cards and blank cover- plates in the Hub must be fastened in place with a screw as described above, or the System may not meet safety and radio frequency emission requirements.

- Each slot must contain an approved card or a blank cover plate, or the System may not meet safety and radio frequency emission requirements.

1.3.3 Replacing Hub Power Supply

The 2101 Hub has a modular dual redundant power supply. Power supply modules can be "hot swapped" as follows. NOTE: The replacement power supply voltage must be correctly set before installation.

1. Open power supply cover by turning the knurled screw head counter-clockwise until loose and then swinging the cover up.

2. Turn off the power supply module to be replaced

3. Pull firmly on power supply handle to remove power supply module

4. Configure replacement power supply module for the voltage in you're area by sliding the small switch (usually red) labeled to "115" or "230" as appropriate for your region.

5. Make sure power switch on the replacement power supply module is switched to the "off" position.

6. Orient the power supply such that the power switch is near the bottom and slide it firmly into the receptacle.

7. Turn on the power supply module and note that the green LED power indicator illuminates.

8. Close power supply cover, and re-fasten.

9. Clear power supply alarm by pressing the square reset button located next to the power supply cover.

1.4 General Troubleshooting

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 characteristics of all who successfully 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:

- Observe the behavior to find the apparent problem;
- Observe collateral behavior to gain as much information as possible about the problem;
- Round up the usual suspects;
- Generate a hypothesis;
- Generate an experiment to test the hypothesis;
- Fix the problem;
- Then, repeat, if necessary, to attack additional problems.

Let us now 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.

Step 2. Observe collateral behavior to gain as much information as possible about the problem. Does the problem only occur with a specific Studio Interface or Desktop Director? 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.

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, many computer problems stem from the same 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 will 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 known good phone or hybrid 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.

A Final Thought…

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 Telos Two 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 are ready to give up engineering, remember that the system has worked and will work. Never panic–you are smarter than it is.

The Status Screen on the Studio Interface

The status of each T-Link connection is shown on a the status screen for the slot for the T-Link card in question. This should read *active*. Press the *Status* button and then using the ▲ & ▼ buttons to cycle through the available screens based on the physical slot location of the line. You should also check the LOS LEDs, see below.

This could be caused by a bad T-Link card in the Studio Interface or Hub, or a wiring problem between the two.

T-Link card LOS LED

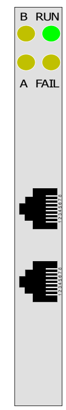
Each T-Link card also has amber (yellow) LED which, when illuminated, indicates a loss of receive signal or sync (Red Alarm). This LED should be off unless there is a Loss Of Signal (i.e. the T-Link card does not see an T-Link signal from the 2101 Hub.

A bad T-Link port at the Hub can easily ruled out by switching the cable to a different port at the hub. If your Studio Interface has a second T-Link card you can also try switching the cable there.

You should also check your wiring between the Hub and the Studio Interface.

A similar LED is present on the Hub for each T-Link connector there.

PRI/T1 Trunk card LEDs



A Dual Telco Interface Card is shown to the right.

The single Interface card is the same but only has one port.

The Lower port is Port A while the upper port is port B. If only one port is to be used you must use Port A (lower port).

The two amber LEDs to the left indicate "Loss of Signal" on Port A or B and should be extinguished if an active line is present. See the table below for additional information on the diagnostic LEDs.

Name	Color	Indicates
Run	Green	Off = Board Not Running On = Normal Activity
Fail	Amber (Yellow)	Off = Normal Operation On = Telco Interface Board failure
B (LOS)	Amber (Yellow)	Off = Normal Operation On = Loss Of Signal (carrier failure, Red Alarm or Yellow Alarm) on B
A (LOS)	Amber (Yellow)	Off = Normal Operation On = Loss Of Signal (carrier failure, Red Alarm or Yellow Alarm) on A

If the *Run* light off, or the *Fail* light is on you should retry rebooting the 2101 Hub by cycling power – wait for the system to fully reinitialize and check the LEDs again.. If this does not help, the problem could be the Hub or the Telco Trunk Interface card.

If one of the LOS lights is illuminated, the most likely cause is that the Telco Trunk is out of service. Check the diagnostic lights of the NCTE. If you have more than one Telco Interface ports, you can try the line in question in a different port and observe if the problem remains with the line or with the port.

2101 Hub configuration

Check the configuration of your 2101 Hub carefully

- Have correctly configured each of the ports of each of your Telco Trunk Interface Cards?

  Be sure to double-check the *Hardware, Circuit Type,* and *Protocol* settings. NOTE: The 2101 will allow you to create a "Custom" Protocol settings. These settings are for advanced users and are not guaranteed to work with any particular line.

- Have you correctly configured your trunks? When using a trunk type (Such as PRI) which provide DID numbers on incoming calls, you must enter each of the "DID numbers" assigned to that circuit.

  If you are using a channelized trunk (such as channelized T1) you must enter each of the phone numbers and the channel to which that number is assigned. Failure to do so will cause outbound calls to be made on an incorrect channel, causing that channel to ring busy and cause ring-no-answer for calls to the number in question.

  When using DID enabled trunks, you must enter the correct number of digits in the trunk configuration screen. You will need to ask the Telco for the correct number of digits to use. If the wrong number of digits is used, incoming calls will not be recognized by the system.

- Have you created *Show Configurations*? Don't forget that when you add a new *Show Configuration* you will also need to update your *Studio Configurations.*

- Have you created (or updated) your *Studio Configurations*? You will need to update these whenever you add a new *Show Configuration*. You will need the Host name entered in each of your 2101 Studio Interfaces the first time you create a *Studio Configuration*.

- Have you rebooted after making changes to IP Configuration, Show Configurations, or Trunk Configurations. To reboot choose *"Reboot"* from the "*Software Update*" screen. ***Note: Rebooting the system will drop any active calls!***

- Have you updated your Hub software version? If so, make sure that the 2101 Studio Interfaces have been upgraded to a compatible version as well. To reboot choose *"Reboot"* from the "*Software Update*" screen. You must reboot before the new software will take effect. See above. ***Note: Rebooting the system will drop any active calls!***

- Are you running the same show in 2 (or more) studios? ***This is not recommended.*** Instead create 2 (or more) shows with the same lines, if desired.

- If you are using T1 trunks, they must support "DP" (pulse) signaling for inbound and outbound calls. If in doubt, check with your Telco provider.

- Has the Hub been repeatedly and frequently rebooted using the front panel reset button or by power cycling the hub? In certain cases this can corrupt the registry. It is recommended that the system be rebooted by choosing *"Reboot"* from the "*Software Update*" screen. Telos Systems Customer Support can assist if the registry has been corrupted.

Audio Problems

- If the system can make and receive calls, but you get nothing but noise on the Desktop Directors and hybrids, check the *Telco* Setting in the Telco menu of each of the Studio Interfaces. This should be set to ETS 300 for European countries and other parts of the world that use A-Law coding. In the USA & Canada, and other countries that use mu-Law coding, use any other setting.
- If you experience regular, periodic, clicking or dropouts, verify that the T1 (or PBX trunk) is acting as clock master. If only one digital trunk is used, and a dual trunk card is installed, you must connect the trunk to the lower port (port A) of the interface card.
- If you get howling or strangle oscillating "darth vader" sounds check that your mix minus is correct. Also check the setting for *Routing* in the Studio Interface's Audio menu.
- If you are getting "tinny" or "hollow" announcer audio whenever callers are put on the air – first check your mix minus (see above). You may also wish to decrease the setting of the *AGC* in the Audio menu of the Studio Interface. You may also wish to use a lower (closer to half-duplex) setting of the *Duplex* setting in this same menu. Finally, you may wish to reduce the *Send Level* setting in this menu.

Ethernet Jack LEDs

Don't forget that the LAN connections between components are an integral part of the Series 2101 system.

The 10Base-T connector on the Studio Interface has two integral LED indicators that can be helpful if problems are suspected.

The green "Link" LED indicates the presence of a live physical connection to a working Ethernet hub (or other device). If the green indicator does not illuminate, you should check your network wiring. Absence of the link light could also indicate a hardware failure of the Studio Interface.

The amber "Activity" LED indicates network activity on this segment of the network. Depending on network activity this indicator may only illuminate occasionally; flickering of this light is normal.

Don't forget that your Ethernet Hub or Switch will also likely have indicator lights as well. The manual for the Ethernet Hub or Switch should have more details.

Finally, you should use a Switching Ethernet Hub to isolate traffic between the LAN reserved for the Series 2101 gear and any other connected network. Use a high quality, professional, switch, such as the HP Procurve or a Cisco product.

Using Loop Modes on the Studio Interface

This option allows the Telos 2101 Studio Interface to loop audio through the system.

- Off

This is the normal operating mode. None of the loopback paths are turned on. Studio Input is sent as send audio, and caller audio is presented at Studio Out.

- Studio

  This loops the AES/EBU input (and analog signal inputs after Analog-to-Digital conversion) back to the AES/EBU output (and analog output via the Digital-to-Analog converter). With this mode engaged, you can make signal to noise, THD, and frequency response measurements through the system to check the input and output stages.

Some diagnostic tests

Can you dial from Line 1 to Line 2?

If you can successfully call from one line to the other (you may or may not need to dial a preceding digit such as 9 first), you know that you have a working T-Link connection to the Series 2101 hub and that the hub is running. This also proves you having a working Ethernet connection between the hub and the Studio Interface is working.

Condition: Intermittent ISDN Problems

Does the problem with numbers on one Telco Trunk or does it occurs across more than one trunk?

If the problem is limited to a single Telco Trunk circuit then the problem may be with that Trunk, or with that Telco Trunk interface card. Move the Circuit in question to a different interface card, if possible (you will need to reconfigure the trunks when you do so). If the problems moves it must be with the Trunk itself.

If the problem occurs on 2 Trunk Ports simultaneously, and they are both in the same interface slot, then it is likely the problem is with that Telco Trunk interface card. Swapping cards (if your system has more than one) between two of the interface slots of the Hub will confirm or disconfirm this.

Telos Customer Support can help you to log ISDN activity to look for problems in cases where the Telco appears to be the culprit.

What is consistent about the problem (look for patterns)**?**

Keep watching for a pattern. Enlist the help of the users. We've seen apparently random problems that we eventually discovered only occurred when it rained, or on a certain day each week, or at a certain time of day, or only when another line was in use. In other cases, it was related to temperature or dirty AC power. Assume nothing and suspect everything. Leave no stone unturned while searching for the answer.

Condition: Problems with Lines Disappearing

This can happen if the 2101 Ethernet LAN is overloaded, or a poorly designed Ethernet Switch is used. We've seen several 3Com Ethernet switches that consistently lost many packets and therefore cannot recommend that brand. If you experience this problem, check your Ethernet wiring. You might also try a completely different brand of Ethernet Switch to see if this makes a difference. If you are using a hub, you may wish to try using a switch, particularly if you have a large 2101 system.

If instead of disappearing, you see the small "x" icon this means that you do not have enough trunks. If this only happens when two studios are both experiencing heavy traffic this is normal. You may need to reconsider how the PRI is provisioned if this happens frequently.

1.5 Troubleshooting Desktop Director Problems

Most problems with the Desktop Directors are cable related. First try a new cable. If that fails to solve the problem, try the Desktop Director with a short cable plugged directly into each of the Director Ports (slot E & F) on the back of the Studio Interface.

Volume 2 Part IV Section 1.3.2 has details on the allowable wiring configurations for the Desktop Directors. There are distance and wiring restrictions. Note that when two Directors share a single port, an external power supply is required, and the termination resistor must be removed on the desktop director closer to the power supply.

1.6 Gaining Access

1.6.1 Gaining Access - Studio Interface/Extended Hybrid (Telos TWO)

TO PREVENT RISKS OF ELECTRIC SHOCK, DISCONNECT POWER CORD BEFORE SERVICING

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 service personnel.

Removal of the top plate is the first step to gaining access for service. Remove the 7 (more on some units) Philips head screws.To remove the T-Link or Desktop Director modules you must remove the two 1/16 inch allen screws that mount them to the rear panel.

When replacing these modules take care to align the connector with the motherboard to avoid bending pins on the base of the module.

Replaceable modules (2101 Studio Interface)

While we do not expect you to do much repair or troubleshooting of the individual circuit boards, the system does have several removable modules which can be systematically replaced or exchanged to aid in troubleshooting and repair. The 2101 Studio Interface has the following modules:

1) Power Supply part # 1281-00007.
2) Studio Interface T-Link card Part # 1701-00048.
3) Firmware flash memory SIMM module part # 1166-00005 (specify if for "2101 Studio Interface" and required version number).

4) Dual Desktop Director Interface (slots E (optional) & F) part # 2001-00022.

   Note that the Dual Desktop Director cards have the following jumper settings. Always be sure to configure correctly before installation as follows:

Desktop Director Interface Jumper Settings:

- Jumpers J3, J4, J5, J6, J11, J12, J13, and J14 should be set to the right two pins (NT setting).
- Switch SW1 (or appropriate jumpers) should be set to the up position (-48VDC power on).

Replaceable modules Extended Hybrid(Telos TWO)

1) Power Supply part # # 1281-00007.
2) ISDN Interface card (single S and U interfaces): Part # 1701-00086.
3) Firmware flash memory SIMM module part # 1166-00005 (specify for "2101 Extended Hybrid" and required version number).

1.6.2 Gaining Access – 2101 Hub

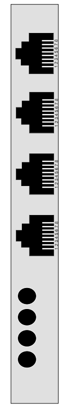
Installing Interface Cards in the Hub

Identifying the Cards

A description of each the various cards is included in the following table:

<table>
<tr><th colspan="4">2101 Hub Interface Cards</th></tr>
<tr><td>Telos Part Number</td><td>Description</td><td>Max # in System</td><td>Notes:</td></tr>
<tr><td>2101-1010</td><td>T-Link Hub I/O Card</td><td>8</td><td>Has 4 ports for connection to Series 2101 Studio Interfaces. May be installed in slots 1-8</td></tr>
<tr><td>2101-0510</td><td>PRI/T1 Telco Interface</td><td rowspan="2">2
Cards may be mixed</td><td>Supports a single T1 connection or 23B+D PRI connection (1.544 Mbps) as used in North America. May be installed in slots 10 & 11</td></tr>
<tr><td>2101-0520</td><td>Dual PRI/T1 Telco Interface</td><td>Supports two T1 connections or 23B+D PRI connections (2 x 1.544 Mbps) as used in North America. May be installed in slots 10 & 11</td></tr>
<tr><td>2101-0610</td><td>PRI/E1 Telco Interface</td><td rowspan="2">2
Cards may be mixed</td><td>Supports a single E1 connection or 30B+D PRI connection (2.048 Mbps) as used in Europe. May be installed in slots 10 & 11</td></tr>
<tr><td>2101-0620</td><td>Dual PRI/E1 Telco Interface</td><td>Supports 2 E1 connections or 30B+D PRI connections (2 x 2.048 Mbps) as used in Europe. May be installed in slots 10 & 11</td></tr>
</table>

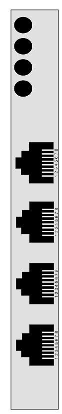
You can identify these cards by looking at the faceplate and connectors. See next page:



The T-Link Hub I/O Card, Rev B Style.

Each card supports connections for up to four Studio Interfaces. The system manages the T-Link connections and any port can be used for any Studio Interface.

The four amber LEDs at the bottom indicate "Loss of Signal" and should be extinguished if a Studio Interface is present and active. The top LED corresponds to the top T-Link port and so forth.

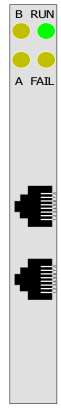


The T-Link Hub I/O Card, Rev A style.

Each card supports connections for up to four Studio Interfaces. The system manages the T-Link connections and any port can be used for any Studio Interface.

The four amber LEDs at the top indicate "Loss of Signal" and should be extinguished if a Studio Interface is present and active. The top LED corresponds to the top T-Link port and so forth.

NOTE: This version requires a custom "Crossover Cable" to connect to the Studio Interface. See Appendix 3.



A Dual Telco Interface Card.

The single Interface card is the same but only has one port.

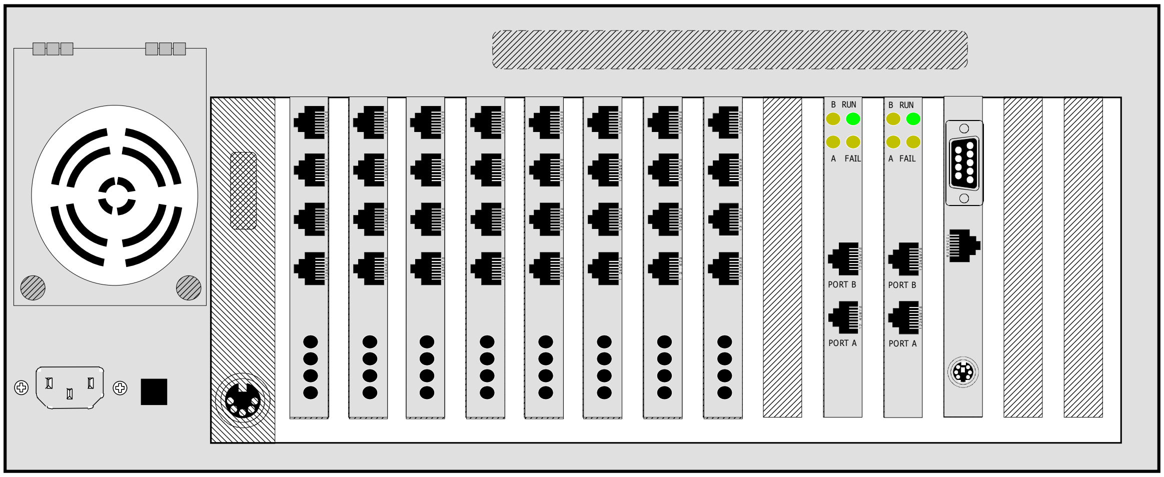
The Lower port is Port A while the upper port is port B. If only one port is to be used you must use Port A (lower port).

The two amber LEDs to the left indicate "Loss of Signal" on Port A or B and should be extinguished if an active line is present. See the table below for additional information on the diagnostic LEDs.

Name	***Color***	***Indicates***
Run	Green	Off = Board Not Running On = Normal Activity
Fail	Amber (Yellow)	Off = Normal Operation On = Telco Interface Board failure
B (LOS)	Amber (Yellow)	Off = Normal Operation On = Loss Of Signal (carrier failure, Red Alarm or Yellow Alarm) on B
A (LOS)	Amber (Yellow)	Off = Normal Operation On = Loss Of Signal (carrier failure, Red Alarm or Yellow Alarm) on A

Installing cards in the Hub

The Series 2101 Hub has slots 14 slots. We shall call them slots one through fourteen left to right as viewed from the rear of the Hub. Note however, the slots are not numbered on the chassis of the unit.



Rear view of the Series 2101 Hub fully populated. Note the T-Link Studio I/O Cards in Slots One through Eight and the Telco Interface Cards in Slots Ten & Eleven. Position of cards may vary depending on number of Telco cards Installed.

⚠	**IMPORTANT!** *1) Slots 13 & 14 must remain unused* *2) Slots 1, 2, 3, 4, 5, 6, 7, 8 (sometimes 9) are for use with T- Link Hub I/O cards* *3) Slots 9 & 10 or 10, & 11 are for use with Telco Trunk Interface cards* *4) Slot 12 is reserved for the Processor/100 Base- T Card (Provide with Hub)*

Follow these steps to install a card in the 2101 Hub:

⚠	**IMPORTANT!** *- The following instructions are to be followed by qualified Technical Personnel ONLY.* *- Handling appropriate for Static Sensitive Electronic Devices (such as personnel grounding straps) must be employed whenever the Hub is open. Damage caused by a failure to do so is not covered by the equipment Warranty.*

1. Disconnect the power cable and any other cables present from the Hub

2. Remove the four Phillips-head screws that hold the top cover in place (two on each side of the Hub)

3. Lift the cover straight up and set aside

4. Determine the correct slot for the card(s) to be installed (See above)

5. Remove the blank Slot Cover-plate from the slot(s) to be populated by removing a single Phillip-head screw located at the top of the Slot Cover-plate. Remove the blank Slot Cover-plate by lifting it straight up

6. Remove the multi-conductor ribbon cable connector plugs from as many of the existing cards as necessary to allow access to install the new card(s).

7. Align the card(s) to be installed with the socket on the main board and the opening in the rear panel of the chassis. Seat the board in place by pushing firmly, straight down. If the board fails to seat properly lift it straight up and out and try again

8. Reconnect the ribbon cable connectors to each the cards disconnected in step 6 as well as the newly installed card(s)

9. Using the screws removed in step 5 (above) fasten the card(s) in place

10. Replace the cover on the Hub and fasten with the four screws removed in step 2 (above)

WARNINGS!

- The interface cards and blank cover- plates in the Hub must be fastened in place with a screw as described above, or the System may not meet safety and radio frequency emission requirements.

- Each slot must contain an approved card or a blank cover plate, or the System may not meet safety and radio frequency emission requirements.

My my my – a blank page. Do you suppose it was intentional?

2 2101 System Specifications

General

Interface for connecting the digital telephone lines to professional studio equipment.

Line Connectivity

Interface:

- Up to 32 2.048 Mbps T-Link connections Series 2101 Hub to Studio Interfaces.
- Desktop Director Interface (144kbps) on Studio Interface supports 4 Desktop Directors per Studio Interface. Expandable, by adding a second interface, to support 8 Desktop Directors per Studio Interface. Optionally can be used for attachment to a Telos TWO hybrid (for 4 hybrid studios). Power internally provided for 2 Desktop Directors per Studio Interface.

Telco Trunks: (the following are supported depending on the selected of Telco Trunk Interface)

- T1/23+D PRI (North American T1/PRI); PRI- NI-1, DMS Custom, 5ESS Custom; T1 - E&M Wink Start DP
- E1/30B+D PRI (E1/ETS-300 PRI); E1
- Telco Trunk Interface cards support one or two digital circuits. Up to two Telco Trunk Interface cards are supported, permitting a maximum of 4 digital trunks.

Telephone Coding Modes

- Speech Coding; mu-Law or A-Law (Telco set to ETS-300). 300-3,400Hz.

Switching matrix and Conferencing

- DSP audio routing and switching
- 10 Conference Bridge Resources are flexibly allocated among lines, hybrids, and Desktop Directors when button-mash conferencing. Each line, hybrid, or Desktop Director included in a button-mash conference consumes one Conference Bridge Resource (no resources used for single-line connections). For example:
  - 1 hybrid + 9 lines
  - 2 hybrids + 4 lines each
  - 2 hybrids + 3 lines each + 1 Desktop Director + 1 line
- Transfer: Trombone transfer on a given PRI is possible (uses 2 trunks)

Analog Audio Specifications (2101 Studio interface and Studio Interface)

Send Input (Analog)

XLR female, pin 2 high. Active balanced, with RF protection.

Input level; Adjustable, -7 to +8 dBu (nominal) level.

Headroom before clipping; +13

Bridging, >100KΩ impedance.

Analog-to-digital converter resolution; 20 bits

Receive Output (Analog)

XLR male active differential. Pin 2 high.

Output level; adjustable from –7 to +8 dBu (nominal)

Clip point; +21dBu

Output impedance; <60Ω X 2

Frequency Response (± .5dB, 50 to 20kHz; swept sine procedure)

- Measured from analog Input to output with unit in loopback mode.

Noise – Input (-93 dBFS, A weighted)

- Measured analog in to AES out in loopback mode.

Noise – Output (-80 dBFS, A weighted)

- Measured AES in to analog out in loopback mode.

THD+N - Input (< 0.06% typical)

- Input: <0.06% typical; Measured at 0 dBu @ 1 kHz analog in to AES out in loopback mode.
- Output: <0.01% typical

Digital-to-analog converter resolution; 16 bits

AES/EBU Digital Inputs/Outputs (2101 Studio interface and Telos TWO)

Sample rates supported; 32, 44.1, and 48kHz.

Rate conversion; Input and output independently selectable.

Input clock; AES input or ISDN network clock.

Input Level; Adjustable -27 to –12 dBfs

Output Level; Adjustable –27 to –12dBfs

Desktop Director Ports (per studio interface)

2 ports, permitting connection of 4 Desktop Directors standard. Optionally expandable to 4 ports for connection of 8 Desktop Directors. Extended Hybrid (Telos TWO) reduces number

of available Desktop Directors by two) External power supply required for more than 2 Desktop Directors per studio interface.

Control Ports (2101 Studio interface)

RS-232; 9 Pin D-sub connector

DCE connector supports asynchronous data, 8 bits, no parity, 2 stop bits, 2400-38,400 bits per second.

Ethernet; 10Base-T: for intercommunication with Series 2101 Hub

General purpose Input/Outputs

15 pin D-Sub connector with 5 status outputs and 4 control inputs.

Control Ports (2101 Hub)

Ethernet; 100 Base-T: for intercommunication with Series Studio Interfaces

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3 Warranty and Application Caution

TELOS SERIES 2101 STUDIO INTERFACE 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 that satisfy regulatory requirements in their country of use.

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TELOS SERIES 2101

Advanced All Digital Multi-line Multi-studio Broadcast Telephone System

USER'S MANUAL Part VII

Appendices

1 – Telephone Tutorial

2 – Glossary of Telephone Technology

3 – Modular Cable Guide

4 – Suggested Reading

5 – Ordering Digital Trunks for use with Series 2101

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Table of Contents Part VII

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1 Appendix 1 – Telephone Technology in the Digital Age - A Tutorial for broadcasters

The following discussion is centered around the Telecom industry and technology in the USA, however the majority of these principles will apply anywhere.

The 2101 is really a sophisticated telephone system. As such it is similar to an Electronic Key Telephone systems designed for general business purposes. We will use the generic term "Key System" in the following discussion. Note: Words in **bold** are defined in the glossary (Appendix 2).

Introduction to telephony; Where we came from.

On the Seventh of March 1876, Alexander Graham Bell received a patent for the telephone. His patent was described as "Improvement in Telegraphy ..." The Bell Telephone Company was formed in July of 1877. While many at the time scoffed at this invention, and Bell had a hard time getting investment capital (the powerful Western Union refused his offer to sell his patent for $100,000 in late 1876/early 1877), the public loved the telephone and acceptance grew rapidly. By December 1877, Western Union had formed the American Speaking Telephone subsidiary to compete with Bell.

Bell's original apparatus was similar to a modern dynamic microphone. A permanent magnet and a piece of moving metal interacted to induce current in a coil (the magneto effect). The original system used the same transducer for both sending and receiving. At the far end the electrical energy was converted back to vibrations thereby creating acoustic waves (i.e. sound). As early as 1878 telephones had become available in places as far west as San Francisco, with the first exchange installed there in the January of that year.

Transducer development

Once it became clear that the telephone was, to use a modern term, a "killer app," a number of other parties came up with alternative methods of telephony in order circumvent Bell's patents. A lot of energy and inventiveness (not to mention money) were invested in "wire" telephones which work on the same principle as two cans connected with a piece of string (or wire), since this was clearly a legitimate way to beat Bell's patent. Indeed, patents were even issued for **exchanges** for this technology! Inventor Thomas Edison (who had yet to invent the phonograph & electric light) and others were contracted by Western Union to develop new and improved telephonic technology. Edison invented the carbon transmitter (or microphone) which became widely used for telephonic applications until the mid 1980's. The carbon microphone works on the principle that carbon granules that are loosely packed will vary widely in resistance when disturbed. By impacting the carbon granules using a mechanical connection to an acoustic diaphragm, the resistance becomes proportional to the acoustic energy hitting the diaphragm. A battery was used to provide a source of current (note that the original Bell technology was "talk powered"). This technology worked well as a transmitter, but was entirely inappropriate as a receiver. When coupled with one of the Bell transducers (as a receiver) this transmitter worked better (higher levels and less distortion) than using two of the Bell devices.

The infant telephone industry grew rapidly. While the Bell Company had a head start, the giant Western Union rapidly caught up. Eventually, the courts determined that Bell's patent had a rather wide scope and a settlement between the two companies was made in November of 1879. The Bell Company got the rights to the Edison transmitter technology, along with a network of 56,000 telephones! Western Union agreed to stay out of the telephone business with the Bell Company paying them 20% of telephone rental fees until the Bell patents expired. Two years later Bell bought Western Union's telephone manufacturing division, Western Electric. It became the sole supplier of equipment to the Bell System until divestiture in 1984. Lucent Technologies is the direct descendant of that company.

The standard transducers used became the Edison carbon style transmitter and a Bell style receiver. Note that there were numerous improvements along the way, however this basic technology was used for nearly all telephones right up to the 1980s. Many such phones are still in service today. Also note that the telephone required batteries (originally large wet cells, later the more compact and less dangerous dry cells were used) until conversion to "common battery" system beginning in the 1890's.

Signaling - Alerting

The original telephones, as mentioned above, each had a single transducer which the user moved back and forth from mouth to ear to communicate. The typical arrangement was two or three telephones directly wired together. However it became evident quite early that there needed to be some way to alert the far end that someone desired to talk. The earliest approach was to yell into the phone and hope someone heard you at the far end! The first technological solution to the "call alert signaling" requirement was to tap the telephone transmitter with a pencil or a small hammer. Once phones began to have separate transmitters (mounted in a wooden box) and receivers (mounted in a wooden or rubber hand-piece) it became possible to build a "thumper" into the box with the transmitter. This allowed the user to pull a knob and a hammer would strike the diaphragm of the transmitter.

Very soon thereafter Thomas Watson, Mr. Bell's original assistant, came up with the idea of using a hand-cranked magneto to create an AC voltage on the line couple with a simple electromechanical ringer at the far end for signaling. A condenser (capacitor) served to isolate the ringer coil from the DC battery circuit. This technology already existed at the time, however its application to the telephone was much needed and worked well. Users could hear if someone wanted to speak to them on the telephone, even if they were not right near it at the time.

Switching - Early manual exchanges

As mentioned above, the early telephones lines were generally two or three phones wired in parallel. For instance (to use a dated example appropriate to those times), an affluent family might have a telephone from the home to the husband's place of business. And possibly another phone from the home to the local store so the wife could call and request that provisions be delivered. It became rapidly clear that the utility of the telephone was directly proportional to the number of people to whom one could talk. For instance, to use our example above: If the businessman were to call and inform his wife that he would be bringing home an Associate for dinner, the wife could then phone the store and get more provisions. If the residence had only one line (to either location) this would not be possible.

However, the practical matter of many individual lines running to many other locations was problematic. If the system were to work as desired, every time a new phone were added lines would be required to each existing phone. Also, each time another phone was bridged onto a line the level dropped. So these would need to be separate lines, possibly with a phone on each end.

The solution to this dilemma was the Central Telephone Exchange. Instead of running lines from subscriber to subscriber, each line went from the subscriber's location to a central location, the telephone exchange. Here the lines were terminated on a "switchboard". The first such telephone switchboard was installed in Hartford CT in 1878 with 21 subscribers.

Switchboard technology evolved rapidly. The typical switchboard worked as follows. Each subscriber line came into a "jack" (originally a "jackknife switch", but the term remains). An electro mechanical device called a "drop" bridged the line. When the subscriber cranked their magneto the drop released a small metal hinge indicating to the Operator at the Central Exchange that the customer was requesting service. The Operator could then plug into that jack and ask the customer what they desired. The Operator could then ring the requested party (using a hand magneto at first, later large "ringing machines" driven by water wheels or a steam engines were used to provide ringing current). If the called party answered, the Operator would proceed to connect the two parties together using a "cord" or connecting cable. The operator would then reset the drop. When the parties were finished they would ring their magnetos and the operator would come back on line to check to see if they were finished. If so, the connection would be "pulled down" by unplugging the cable.

Note that there is a second signaling function implicit in the above arrangement. The calling party would tell the Operator to whom s/he desired to be connected. This signaling function is an example of "in-band signaling". In other words, the "signaling" occurs within the same channel as the communication itself.

Later, in 1891 the automatic exchange (**switch**) was developed. The system rapidly evolved to where a rotating dial pulsed out DC pulses that then drove a series of stepper switches of one kind or another. These dial "pulses" are another form of "in-band signaling". The automatic system was adopted fairly early by the **Independent** Telco's, but was not widely adopted by the "Bell System" until much later in the 1920's through the 1940's.

The economics of telephony

Central exchanges had a number of important advantages for the telephone companies:

1) Only one pair of wires were required to each subscriber's premises.

2) Simplified plant requirements; The Telco could run extra cables from the central office to the vicinity of a growing area in advance. The designers knew in advance that all circuits will terminate at the central telephone exchange (now often called a Central Office). They could therefore have these circuits pre-installed and terminated at a central frame ready to be hooked up to the switching equipment prior to receiving a call for new service.

3) They could build and share "toll" circuits and other facilities among users. For example, one or two toll circuits (called toll **trunks**) could be built from town A to town B. Any subscriber in either town could take advantage of this facility, most likely there would be an added "toll charge" for this call. The cost of building the trunk was thereby shared among all users who had need to call between the two towns. Even if a wealthy subscriber had been able to afford a line between the two towns, under the pre-exchange system s/he would have been limited to speaking to a single location at the other end. And no one else would have had access to this facility without disturbing this individual.

An important principle soon became evident when building telephone exchanges. While many people might have telephones, at any given time only a few are actually using the telephone. So the number of "paths" through the switchboard (i.e. the number of Operators and the number of cords) can be far lower than the number of lines coming into the switch. The science of Traffic Engineering attempts to determine the number of paths required to keep the amount of "**blocking**" (inability to complete a given call at a given time) to an acceptable level. It applies to both **trunks** and to **switch** paths. There is no simple solution to this problem as economic issues must be considered as well. Each trunk added between Town A and Town B will reduce **blocking**. However, the more trunks put in, the greater the number which will be unused except during periods of peak demand. The inevitable result is that in order to maintain economical service *some* blocking will occur ("all circuits are busy... please hang up and try again later..."). The state regulators determine level of blocking that is acceptable and the Telco must meet that level or risk being punished.

These principles are important to understand as they determine the very basis of economical telephony. Shared facilities are essential to economical operation. Note that the same principles apply to private telephone systems. However in this case *you get to decide* what level of **blocking** is acceptable rather than the regulators. As you add more trunks to your **PBX** you will reduce the chances of users being unable to get an "outside line", however you will need to pay for the **trunks** every month. And a bigger **switch** will allow more simultaneous paths through the switch, however a bigger switch costs more money.

There are a few disadvantages to the central exchange method as well:

1) If you desire to talk to your neighbor, the audio must still run from your location all the way to the **exchange** and from there back to your neighbor's

location. This method can have a significant effect on signal level and noise floor.

Even should you purchase a dedicated line from your facility to a neighbor's location, the Telco infrastructure will still mandate that the line go through the **central office**. Readers who have dealt with high fidelity conditioned audio loops are well aware of the frustration involved with having your audio routed many extra miles with the resulting performance and reliability limitations

2) Some **blocking** is inevitable

3) Single point of failure. A fire or equipment failure at the **central office** will affect multiple users.

An introduction to Private Telephone Systems

Many readers will recall seeing old motion pictures where multiple phones are shown on a worker's desk at a business establishment. Film writers often took advantage of the absurdity of talking on more than one phone at a time, or trying to determine which of several phones was ringing, for comic effect. However there was a time when this was exactly how telephones were used in the office. Two different approaches were developed to solve these problems, both of which still exist today. **Key telephone systems** and the Private Branch Exchanges (**PBX**).

Key Telephone Systems

The **key telephone system** is a direct evolution from having multiple phones on the desk. The earliest system had a series of "keys" (switches) mounted in a box, which allowed one to choose which of several lines was connected to the phone. One position of the key "hung up" the line while another connected it to the phone. Usually an intermediate position of the switch allowed one to place a call on "hold". However there was still the problem that if more than one person was sharing these lines that each person would not know if a line was in use without checking it, thereby interrupting anyone who might be using it. In addition, callers could be forgotten on hold. Not only did this annoy the customer, but that line would not be available for use until the mistake was discovered. "Supervision" between stations showing the status of lines was required.

True **key telephone systems** have a Key Service Unit that all lines are connected to. Standard **loop start lines** are normally used. All phones in the system connect to this KSU. Each phone in a Key system typically has access to several lines, *through the KSU*. The phones and KSU work together as follows. Each phone has an indicator for each line to which it has access. These indicators allow the user to see that state of each line to which they have access. For instance, if a line is ringing, the user can determine which line(s) it is by looking at the indicators. Or if a user desires to dial out s/he can select and idle line by looking at the indicators and selecting a line shown as idle. Once the line is selected and the handset lifted it is as if this phone is actually connected to the line itself. However, the KSU will provide supervision so that other users are aware of the status of this line. In fact, with the early "mechanical" key system such as the

1A1 and 1A2 the user's phone is actually connected to the line through a "hard" connection (switch contacts) when a line is in use.

The distinctive characteristics of a Key Telephone system are as follows:

1) Each **line** on the phone has a direct correspondence to actual lines from the Telco. In older key systems such as the 1A1 and 1A2 six conductors for each line go to each phone on which it appears!

2) A given line can, and often does, appear on more than one telephone set. Which lines appear on which telephones is determined when the system is installed. This relationship is more-or-less permanent.

3) The system communicates line status to the users who supply the "intelligence" of the system. It is up to the user to select an unused line before dialing out. If s/he does not s/he may interrupt a conversation in progress or inadvertently answer a call ringing in. Likewise, if a line is ringing it is up to the user to select the ringing line before answering.

4) There is no way to directly "call" from one phone in the system to another. Any number of users may select an "intercom line" (a common circuit with just talk-battery on it) and talk, however this does not go through the KSU or really constitute a "call" (i.e. a switch connection). Or one user may call out on one line to another line of the system, and hope that the desired party answers the ringing line. For these reasons a public address system is often used with a Key system to allow announcements to users that they have a call on a certain line.

To summarize: Key systems allow multiple phones to efficiently share phone company lines. Each line has an identity (the phone number) but the telephones do not.

Of course modern Key Systems often offer features that tend obscure some of these characteristics. For instance, modern key systems may only require 2 or 4 conductors to each phone. Some modern systems may automatically select a ringing line, etc.

Private Branch Exchanges (PBXs)

A private branch exchange is just like a telephone company exchange. It has one or more "Telco **trunks**" to the phone company (note that the Telco may still call these "**lines**", so it can get confusing). And it has one or more "**station lines**" to telephone terminals. Just as with telephone company exchanges, a switchboard and Operator were originally used to place calls. Later, Private Automated Branch Exchanges were developed where direct dialing out of the system became possible without an Operator.

With traditional **PBX**s an attendant (Operator) was also required to answer all incoming calls and route them to the intended party.

Key characteristics of a PBX

1) Each phone has its own "station line" from the **PBX**. While sometimes phones will have more than one line, never does more than one phone share the same line.

2) Inside calls between stations of the PBX are possible without going through the telephone company **trunks**.

3) **Trunks** (to the Telco **switch**) are shared between all users. To make an "outside" call the user dials a special digit (typically a "9" or "8"). If the **PBX** has an available **trunk,** the user is connected it. The user then hears or "draws" dial tone from the Telco switch and can proceed to dial the number. Unlike a Key system, the user has no indication until after s/he picks up the phone and dials the special digit if a trunk is available.

To summarize; **PBX** telephones have an identity of their own. They are extensions with a unique extension number. Each has access to the PBX. The PBX has trunks to the phone company. The PBX actually switches calls through itself based on the user's demands, not based on a pre-configured wiring plan. Station-to-station calls as well as station-to-trunk calls are possible.

Modern business telephone systems can still generally be categorized as Key systems or PBXs. Note that the advanced features available on many modern Key systems can blur the difference unless one looks carefully.

Centrex

In the 1970s, with the deregulation of the telephone companies, end users began to have the right to purchase telephone systems from suppliers other than the AT&T owned phone companies. The end user suddenly had many more choices. To compete with the PBX and Key systems, now available from outside vendors, the phone companies began to offer another option called "Centrex" service. This option became the only option the phone companies were permitted to offer once the divestiture of AT&T was complete in 1984. Only since the 1996 Telecommunications Act has been implemented have the major Telcos been permitted to provide equipment at the user's premises such as Key and PBX equipment. Therefore Centrex was an important product for the telephone companies during the 1980's through 1990's.

Centrex is really very simple. Each user has a phone, and each phone has a line from the phone company and a telephone number. However, in order to meet the needs of businesses, Centrex lines have a number of important features. Centrex is designed to act much like a PBX. To dial another person in your Centrex Group you need only dial 4 digits (in some arrangements 3 digits). These intra-Centrex calls do not occur any per-call or per-minute charges. An unlimited number of these calls are included as part of the Centrex package.

Other Centrex features include:

- Transfer to another Centrex line
- Hold
- Busy/no-answer forwarding to another Centrex line
- User specified call forwarding
- Call pick-up (ability to pick up a call ringing on another Centrex line).
- Hunt groups

- Conferencing

One feature that is usually available on Key and PBX systems is generally not part of Centrex, music-on-hold. This feature is sometimes available, but is complex to order and install since the audio must be transported to the telephone company **central office**.

Centrex is sometimes used as a way to bridge between two systems, For instance, if your on-air and business systems both used Centrex you could transfer calls from one system to the other using the Centrex transfer function.

The major advantage to Centrex for most users is the convenience of "one stop shopping".

The single biggest drawback to Centrex is the fact that it is typically used with basic telephones. The user must "hookflash" the line (hang up the line for less than 0.9 sec) and then enter a string of digits to request the above features. It is confusing for the users, and the proper codes for a given feature are easily forgotten. Recently a number of "smart phones" have become available for use with Centrex. These phones have feature buttons to make things easier for the user, although the actual signaling is via current interruptions and **DTMF** tones.

Digital Phone Systems and the broadcaster.

Introduction, early digital systems

In the 1970-80's both **Key systems** and **PBX** systems began to offer systems where the internal workings of the system (and often the link to the proprietary telephone sets as well) worked with digitized audio. By using digital audio paths these systems could offer more advanced features (such as high quality conferencing) and better prices. Part of the drive to do so was the availability of low cost digital components due to the surge in popularity of the computer and related digital technologies.

The switch to digital systems was not always embraced by those in the broadcast industry for several reasons. Once DSP (digital signal processing) became available, the adaptive hybrid became possible, allowing true full duplex on-air audio without corruption of the "announcer" audio. When Steve Church (our founder) designed the first adaptive DSP hybrid, the Telos 10, he designed it to work with the 1A2 Key systems that were ubiquitous at the time. It could also be easily adapted to direct connection to a single phone line or a home-built line switcher.

The new digital phone systems required a special "analog port" adapter to talk to any analog device (including faxes and modems), which complicated matters. More importantly, it was found that the Analog to Digital to Analog conversions of the signal as it passed through the system severely degraded hybrid performance. One potential solution, to connect to the system digitally, was not feasible since the digital phone ports were proprietary. Manufacturers would not release details, and each used their own scheme.

Therefore the best approach for many years was to use direct lines from the phone company.

Digital trunking

As businesses began to buy digital switches, the practice of bringing digital circuits to the user's premises from the Telco became more practical. While very high volume users had sometimes used digital trunks in the past, often converting them to analog for use with analog PBXs or other equipment, the newer digital PBX's could frequently support direct connections to these digital trunks. Usually T1 Telco circuits are used for digital trunking purposes, although PRI is another option.

The advent of the digital **PBX** connected directly to a digital **trunk**, combined with the demand for better quality analog ports by modem users, has made it practical to operate high quality adaptive hybrids off many PBX systems. Since the network is nearly 100% digital today, this means that the caller's voice is digitized at his/her telephone company central office and transmitted digitally all the way into the PBX. There, it is converted back to analog. If the analog port is of good quality the small amount of analog cabling between the PBX and the hybrid is insignificant and very good hybrid performance usually results. This arrangement is well suited to situations where a single hybrid or pair of hybrids is used for interviews or production purposes.

ISDN PRI is the ideal solution for flexibility. The **D channel** protocol allows for sophisticated operation difficult to achieve in any other way. Incoming calls requests are delivered to the **PBX** which can then accept or reject them based on its own requirements. When a call is accepted, the PBX determines which channel to accept it on. This allows the same sort of efficiency enjoyed by telephone company trunking arrangements for quite some years. No **channel** is used just to return a busy signal.

DID

Recall that telephones on a **PBX** have an identity of their own. It is possible to have the identity of a PBX phone be a "real" telephone number rather than just an extension number. A block of numbers can be purchased from the Telco for this purpose. When combined with a special "**DID**" (Direct Inward Dial) **trunk**, a specific extension of a PBX can be dialed by users outside the system. Traditionally DID trunks could only be used for calls inbound to the PBX (hence the word "inward" in the name) which limited their affordability to large PBX owners only. With the advent of digital trunking "two-way-DID" has become available. Using **PRI** (or sometimes **T1**) these trunks can handle both inbound and outbound traffic and still have the ability for outside users to dial specific extensions as desired.

Buying dial tone from a phone company other than your ILEC (Incumbent Local Exchange Carrier)

Under the provisions of the 1996 Telecommunications act, the local dial tone business in the USA has become much more competitive. Deregulation is the trend throughout the world. This author believes this is a good thing in many ways and that competition will benefit all, even those who stay with the traditional provider. However, it is primarily the well informed consumer who will reap the most benefit. The following are some of the questions you should ask as part of deciding if you should change providers:

1) Does the provider merely resell the services of some other company? Do they own their own **switch**ing equipment?

2) Does the provider own **trunk**ing to your neighborhood, or will they be leasing carrier/copper/fiber from a **ILEC**?

3) What are the charges (per minute and/or per call) for local calls? It is likely these charges will be structured somewhat differently from your existing carrier. Scrutinize your usage (if you have a PBX ask your vendor to generate usage statistic reports for a couple of months) and calculate what your actual charges will be under the new plan.

4) What duration is their minimum contract? Do longer term contracts offer a meaningful discount?

5) Does the provider offer a 24 hour repair hotline? Most incumbents do not. Will they give you the number direct to their switch technicians?

6) Contact some references and see how satisfied they are with the customer service and company in general.

Our experience is that the new **CLEC**s (Competitive Local Exchange Carriers) are often more "friendly" to do business with. Often, their smaller bureaucracies allow you a better chance to get through to the person you need to deal with easily. Employees often appear to care about you as a customer. However, there can be a drawbacks too; these leaner companies have less redundancy and may have only one expert in a given specialty in your region to rely on. And, if they are leasing local loop facilities from the ILEC, they may be slower to get things installed (or repaired) as they must rely on the (sometimes unwilling) cooperation of the ILEC. And those providers that merely "resell" the services of another company are completely at the mercy of the companies who service's they sell.

Remember, that in today's competitive marketplace you may very well have the ability to negotiate with the vendor. This may even be true with the ILEC.

A few words on Long Distance Providers (Interexchange Carriers)

It has been over 15 years since basic long distance services were deregulated. Just as with the choice of dial tone providers, you should choose a long distance provider based on something other than just price. Most stations would not buy the cheapest components for the air chain. Remember that the quality of your caller audio can reflect on your reputation.

1) Does the provider merely just resell the services of some other company? Do they own their own network?

2) Does this long distance provider offer "**Circuit Switched Data**" connectivity? If you are making outbound long distance calls with a codec (such as the Zephyr or ZephyrXstream) this is required. Be aware that most do not. If in doubt, get their 10xxxxx dial-around code (in the USA) and try some calls before changing

carriers. Get the number for their "switched data service center" who can troubleshoot problems with switched data calls. You may decide to change your carrier for all lines except the lines used for your codecs.

3) Is this carrier using any form of "voice compression" on calls? If so, audio quality and hybrid nulling may be unsatisfactory. Voice compression can also greatly slow fax transmission. Again, if in doubt, ask for the dial-around codes and try some calls. Be critical – you will be placing these calls on the air.

4) Look at both per minute as well as fixed monthly charges when comparing rates. You will need to determine your current usage to calculate what your actual charges will be.

5) Consider international long distance needs carefully. If you do not need it you may wish to get a plan that does not allow international dialing. If you need it you must be sure that the rate for international calls is also reasonable. Again, using your actual past history is the best way to make informed judgments.

6) Carefully examine the details of systems that require dialing a security code. Systems which require a **DTMF** tone sequence will probably not work for Circuit Switched Data connections (for calls with a codec) as there is no audio channel to convey these tones on this type of call. Also, some of these system can create user confusion and inefficiency if staff must make many long distance calls.

7) Be wary of long-term contracts. Rates continue to fall regularly. You should be checking the marketplace every 18 months or so and renegotiating rates accordingly. Negotiate your cancellation charges as well as your per-minute rates.

Always be sure to go back to your current provider and see if they are willing to renegotiate their rates. This is a particularly good strategy if your main motive for changing is to get lower rates and your current service is working well.

8) Contact some references to see how satisfied they are with the prospective company and their customer service.

Broadcast on-air requirements for telephone systems

On-air telephone systems require the ability to "stack up" callers who can then be put on the air as needed to maintain the continuity and artistic needs of the show. Easy access to each caller is essential to allow the show to progress smoothly. The "Key" approach works best here because multiple callers can be accessed with certainty and immediacy. In addition, call screener systems (both manual and computerized) can refer to a caller on a particular "line" and be easily understood. Therefore, most broadcast telephone systems are either based on a key system such as the 1A2 (preferred because it is easy to use, non proprietary, and offers "hard" connections rather than using electronic switching of the audio path) or a specialized self-contained system that follows the "Key" style of operation (such as The Telos Series 2101, TWOx12, One-x-Six, or DIM).

When a digital **PBX** with digital **trunks** is available, we have frequently seen a number of analog ports fed from the PBX to a 1A2 key system, or directly to one of the self contained Telos systems mentioned above. This approach generally works quite well. However, with the Telos TWOx12 digital version several new all-digital options are now available and offer superior performance.

Behind the PBX

Many of the current generation of **PBX**s can support **ISDN PRI** equipment. While the physical interfaces are well defined, there are some issues regarding protocol compatibility. You should consult the list elsewhere in this manual for PBXs which have been tested with the 2101.

The "behind the PBX approach" has a number of important benefits beyond the benefits of a fully digital path from the caller's central office:

1) When more than a few **trunks** are required, digital trunks frequently cost less than analog trunks. By combining your on-air and business-related telephone traffic, you are better able to take advantage of these economies.

In addition, you may choose to have separate digital trunks going to your long distance carrier. This permits substantial savings on your long distance bill. If the same long distance also handles your inbound toll free lines for on-air use, you should be able to achieve savings on your inbound "800" long distance rates as well.

2) Callers to the main **PBX** number(s) can usually be transferred to the on-air system just as they can be transferred to any other extension in the system.

3) Caller ID is more likely to be supported on a PBX BRI or PRI port than a PBX analog port.

4) Using **DID** (Direct Inward Dialing) numbers and hunt groups on the station side of the PBX, you can gain considerable flexibility and control vs. a Telco hunt group on analog **trunk**s.

For instance, by using the "called number" information available on incoming calls the PBX could route calls to one of multiple inside hunt groups. Instead of ordering a separate group of lines for your Morning Show and syndicated Talk Show you can share the digital trunks from the Telco and just route the calls within the PBX appropriately.

Bringing it all together

New standards-based approaches to digital telephony such as ISDN BRI & PRI offer substantial new opportunities for the broadcaster. When selecting a PBX for your broadcast facility you should be sure to investigate the following:

1) Can the system support **ISDN BRI** or **PRI** (system 2101 only) on the user side (extension side) of the switch? If so, is this standards based, or proprietary? Are both **circuit-switch-data** and **circuit-switched-voice** connections possible from these extensions? What **ISDN protocol** family is used?

2) Does the system support digital Telco trunking (i.e. **T1**, **E1**, **PRI**)? Can the system support **two-way-DID** over digital trunks? Can a **hunt group** of extensions be accessed using a single DID number? Can the same hunt group be accessed using a second DID number?

3) Can station personnel readily change hunt group configurations? Add new stations? Is this level of self management included in the training package or is this extra? This is important not only as your station (or group) evolves, but as a tool to be used for special events or as part of your disaster recovery plans.

4) Can multiple station-side **BRI**'s be in a **hunt** group?

5) Can a user on a "feature" telephone transfer a call to a BRI/PRI extension? Can a user on an analog port transfer a call to other extensions on the system using a hook flash transfer?

6) Are high quality analog ports available for the system? Do they support "high speed" data (33.6 or 54 Kbps)?

We hope this tutorial has served to make you aware of the issues involved in selecting and managing a telephone system for broadcast use in the 2000's. While it is likely that there is more than one solution that will adequately meet your needs, we hope that this discussion will help you build a system that meets your needs and also takes advantage of the economies possible when the newer technology is applied to today's broadcast facility.

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2 Appendix 2 – Glossary of Useful ISDN & Telephone Technology

You'll get better results from the Telco if you understand, and speak, the lingo! We have tried to include the typical acronyms used by Telco personnel. We've put the definition under the most commonly used acronym. For Intel's ISDN glossary check out: http://support.intel.com/support/isdn/ISDN_Glo.htm

AMI – Alternate Mark Inversion. A T1 line coding method. This is the older of the two commonly available. See: line coding, T1. See also: B8ZS.

ANI – Automatic Number Identification- A system, originally designed for use by Interexchange carriers (IEC's) that transmits the "billed party number" along with a call. Note that the billed party number is not necessarily the number of the line placing the call. ANI predates SS7 and can operate in with analog as well as digital trunks. See also: CLID and Caller ID.

Asynchronous Data - A form of serial data communications that is not clocked. To keep the bit stream synchronized, start and stop bits are used which cuts down on throughput. RS-232 computer data is commonly asynchronous data. In contrast to synchronous data. See also: Synchronous Data.

B Channel - Bearer Channel. One of the multiple user channels on an ISDN circuit. Used to carry user's data; i.e. coded audio data in the case of Zephyr.

B8ZS- Bipolar 8 (with) Zero Substitution. A T1 line coding method. This is the more modern line coding method of the two commonly available. See: Line Coding, T1. See also: AMI

Bearer Channel- See: B Channel

Behind the PBX- This is our own term, and refers to when one privately owned phone system is tied to another privately owned phone system. The most common application is when a key system is connected to analog ports of a PBX. When it involves one PBX behind another, it is a limited Tandem application. See: Tandem Switch and Tandem Tie Trunk Switching

Bell Labs. The basic research facility that was AT&T's primary research facility. Bell Labs was spun off with Lucent Technologies. Many very important discoveries were made at Bell Labs including the transistor, communications theory, and radio astronomy. The future of Bell Labs seems bleak at the time of this writing.

Bell Operating Company. See: BOC. See also: RBOC.

Bellcore- BELL COmmunications REsearch. See Telcordia. The research and development organization owned by the RBOCs. Bellcore represents the RBOCs in developing standards for Telco equipment and in testing equipment compliance to those standards. Bellcore also offers educational and training programs open to all interested parties. Now Telcordia. See: Telcordia

BERT - 1) Bit Error Rate Test- A test for digital lines which involves looping a data path and sending a test pattern. Data returning is compared to the sent data to check for errors. Depending on the "Test Pattern" used, BERTs may or may not uncover problems. A line which only has occasional problems will need a BERT of sufficient time duration to catch that intermittent problem. A five minute BERT of an ISDN BRI circuit will only catch severe problems. 2) A **Bit Error Rate Tester**. The test equipment used to perform a Bit Error Rate Test.

Billing Telephone Number- The main phone number which all calls on a PRI are billed to. This information may be required when configuring a PRI PBX.

Bit Error Rate- The basic measure of errors on digital transmission paths. It is usually expressed as the number of errors per number of bits. For example, the allowable bit error rate on a BRI circuit is 1 x 10 E7 (one bit error in 10^7 bits or 1 error in 10 million bits).

Bit Error Rate Test-See: BERT

Bit Rate- The capacity of a digital channel. ISDN calls are set up at a given bit rate, either 64Kbps or 56Kbps. The bit rate cannot be changed during a call. See Kbps.

BLEC- Building Local Exchange Carrier. A LEC who covers the occupants of a single building (or a small group of buildings) only. Often Telecom services are provided by a BLEC as a service or incentive to potential tenants. If a BLEC offers Long Distance Service it is covered by the same regulations as any other LEC. See: LEC.

Blocking- When a circuit switched call cannot be completed. The percentage of blocked calls to the number of calls attempted forms the basis of a statistic called "grade of service". While it is economically infeasible to build a network that would have no blocking, the Telcos are held accountable by the utility commissions to keep blocking below tariffed levels. The concept of blocking cannot be applied to packet networks (they just lose packets, instead), only circuit switched networks.

Blue Alarm- Also called an Alarm Indicating Signal (AIS). A keep-alive signal sent if a problem occurs mid-span in a T-carrier system. The blue alarm signal is required because in some cases T-1 repeaters will become unstable if inadequate 1's density is not maintained.

BOC - Bell Operating Company. One of the regional telephone companies that were owned by AT&T before divestiture in 1984 (i.e. New England Telephone, Ohio Bell, etc). The 22 BOCs were divided among the RBOCs at divestiture. See also: RBOC.

Both Way Trunk- see: Combination Trunk

BRI- ISDN **Basic Rate Interface**- The common form of ISDN with 2 "Bearer" Channels and one "D" channel. In the USA & Canada ll three channels are on a single copper pair and encoded with type 2B1Q coding.

BRITE- Basic Rate Interface Transmission Extension. A technology were ordinary T-1 trunks (or any other digital carrier system) are used to extend ISDN BRI service. See Repeater.

BTN- See: Billing Telephone Number.

Business Office- The part of the phone company where you call if they mess up your bill, to report problems, and to order service. Not necessarily technically literate.

Call Wading - Also known as Call Waiting, this feature allows you accept calls from, and talk to, additional callers during a conversation without any equipment other than an ordinary telephone. It is a CLASS feature for loop start lines.

So called because once you have more than one call involved, it becomes nearly impossible to know who you are talking to as you try to switch between (wade through) the calls. See also: CLASS

Called Party Address- This is the destination phone number of a call delivered to a switch. For instance this could be the CLID of a call delivered to a PBX using DID or two-way trunks. See also: DID.

Caller ID- A CLASS feature on an analog line that provides the number of the calling line as a burst of FSK data (modem tones) following the first ring. Also called Calling Line Identification . See also CLASS.

Calling Line ID- See: CLID. See also: Caller ID.

Calling Party Control- See: CPC.

CAS – Channel Associated Signaling. A bit-based signaling method used on digital lines (such as T1) that is periodically inserted into the low order bit also used for the audio transmission. See Robbed Bit Signaling.

Cause Code- A code returned by switching equipment to ISDN equipment as part of the call control signaling protocol. The cause code indicates that a network call-related event has occurred or has failed. Since these codes actually come from the network, the fact you are getting a cause code is an indication that that the ISDN circuit is operational at some level.

CCIS- Common Channel Interoffice Signaling. A signaling system where network information such as address and routing information are handled externally to the actual communications (voice) path. SS7 (Signaling System 7) is the internationally standardized CCIS system. Deployment of CCIS increased efficiency since no communications (voice) channels are used merely to report an "all trunks busy" or "far end busy" conditions. It also decreased toll fraud substantially since it removed the potential for access to the signaling information that was inherent to in-band signaling schemes. CCIS also enables CLASS features as well as sophisticated re-routing features for "intelligent network" applications. See also: In-band Signaling and SS7.

Central Office- See: CO

Centrex - Central Exchange Service. An enhanced business telephone service intended to offer most of the features of a PBX but where the lines are all from the LEC out of a public switch. Offers CLASS-like features for business users such as 4-digit "inside" dialing, hold, transfer, attendant, etc.

CEPT- Conference on European Posts & Telecommunications. This is a European standards body that formerly set the standards for telephone interfaces for 26 countries.

CEPT Format- The usual rate and frame format for E1 circuits. 2.048 mbps. See: E1.

CEPT Rate- See CEPT format. See also: E1.

Channel- An actual path you can talk or send data over. This is what you are paying the phone company for. For instance, ISDN BRI lines can be ordered with 1 or 2 active channels and these channels can be configured for voice calls (CSV), data calls (CSD) or both (alternate CSD CSV). A channel does not necessarily have it's own unique telephone number. See: ISDN.

Channel Associated Signaling – See: CAS. See also: Robbed Bit Signaling.

Choke Exchange- A telephone exchange, which is assigned to Radio and TV stations, Promoters, and other users that will be receiving large numbers of simultaneous calls. The idea is to group all of these users on a single exchange so when all routes into that exchange are in use "normal" users (on other exchanges) will not experience blocking of incoming or outgoing calls. Trunks from other local exchanges into the choke exchange are deliberately limited to just a few paths so callers will get an "all trunks busy" instead of completely blocking their local exchange. However, when one of the choke exchange users experiences a large number of calls (as when your station runs a contest) the other choke exchange users will be blocked because all trunks into the choke exchange will be busy.

In the modern network, using CCIS signaling such as SS7, actual trunks are not used to convey "busy" or "all trunks busy" conditions. Thus blocking due to a station contest should not occur as the busy status in response to a call attempt is conveyed over the separate SS7 network. Therefore, the need for choke exchanges has pretty much disappeared. Nonetheless, many Telco still insist that Broadcasters use special choke lines for call-in lines.

See: blocking and concentration.

Circuit- A physical path through which electrical signals can pass. It consists of a network of conductors and other components, separated by insulators. Technically this term cannot be applied to fiber optic or other "non-metallic" paths. See also: channel

Circuit Switched Data- See: CSD

Circuit Switching- A system where a dedicated channel is allocated to the users of that call for the duration of that call. That channel is allocated for the duration of the call regardless if information is being transmitted at any given moment. Bandwidth through the channel is fixed, at no time may this bandwidth be exceeded. If this bandwidth is not used it is wasted. While inherently inefficient, the dependable and reliable nature of circuit switching makes it ideally suited to real-time voice and audio/video conferencing applications. When over loaded Circuit Switched networks will respond "all circuits are busy… try again later". This is in stark contrast to packet switched networks or to systems where statistical multiplexing is used. See: Statistical Multiplexing and Packet Switching.

CLASS- Custom Local Area Signaling Services. A variety of enhanced features (usually on analog lines) that take advantage of the ability of modern SS7 technology's ability to transmit information about the calling party. CLASS includes such features as Caller ID, Automatic Callback, Call Trace (initiated by subscriber, Selective Call Rejection, etc.

Clear Cause. See: Cause Code.

CLEC- Competitive Local Exchange Carrier. Your local telephone service provider who is one of the new-generation providers rather than a RBOC or Independent. A CLEC is really just an independent, albeit one formed after the divestiture of AT&T. See: LEC and Independent.

CLI - Calling Line Identity. European term for CLID. See: CLID.

CLID- Calling Line Identification. This is the ISDN and SS7 equivalent of Caller ID; I.E. the number of the calling party. See also: Caller ID and ANI.

CO- Central Office. The Telco facility to which your local telephone circuit lead. Contains "Switches" and "Trunks" as well as the local telephone circuits.

Codec- COder/DECoder. A device which takes digitized audio and/or video and "codes" it in order to reduce the transmission bit rate and which can also simultaneously "decode" such coded audio. Strictly speaking, a codec does not include an ISDN terminal adapter and related equipment.

Simple codecs are also used in digital telephony. The most common A-Law and Mu-Law codecs used with PCM use a simple companding scheme to reduce channel noise.

COL - COnnected Line number. European Term. The number to which you have connected. This may not be the number you dialed if call forwarding is used.

Combination Trunk- A trunk (channel) which can both make and receive calls. This generally refers to analog ground start or loop start trunks, although the term can be applied to ISDN BRI or PRI channels as well. Each combination trunk normally has a telephone number, although they are frequently part of a hunt group and only one number may be published for that group. Also called a Both Way Trunk. This is not the same as a Two-way DID trunk. See also: DID Trunk, Hunt Group and Trunk.

Common Channel Interoffice Signaling- See CCIS

Competitive Local Exchange Carrier- See CLEC.

Concentration- The basic premise is to share facilities wherever possible. For instance, while there may be thousands of customers served by a given Central Office, there will be substantially less than that number of calls which can be handled simultaneously. And, even fewer long distance calls can be made simultaneously. The art of Traffic Engineering is to have enough capability that calls are rarely blocked, but not any more than that. See also: Choke Exchange and Blocking.

Confusatrex – Also known as "Centrex", this is an enhanced loop start line frequently offered by telephone companies to their business customers. It offers the appeal of no on-premises equipment and no maintenance contracts.

So called because the user must be able to accurately "hookflash" the line and then press a 2 or 3 digit code to activate the desired feature. Since this is a tricky operation at best, many users will warn their customers "if I lose you just call back".

Fortunately, smart phones are now available that can initiate the hookflash and dial the code, making Centrex a much more usable than in the past. ISDN Centrex does not suffer from these problems. See: Centrex and CLASS.

CPC- Calling Party Control. Sometimes referred to as "CPC Wink" or "disconnect supervision". A call supervision feature on an analog loop start line that provides the ability for a CO (Central Office) to signal the called party when the calling party hangs up. CPC allows the PBX, key system, or telephone answering device to reset the line so that it is ready to accept or initiate another call. CPC is accomplished by either a loop current drop or reversal. With some CO equipment, it is also provided if the called party drops the call. See also: MCLD.

CPE- Customer Premise Equipment- Customer owned equipment located at his/her facility, such as a CSU or terminal. In the USA and Canada, an NT1 or CSU are considered part of the CPE.

CPN - Called Party Number - European Term. The number that has been dialed. See: Called Party Address.

CSD- Circuit Switched Data- A dial-up data communications channel which, once established, looks like a transparent data pipe. Also, the type of ISDN service required to utilize this capability of an ISDN circuit. In contrast to CSV.

CSU- Channel Service Unit. The NCTE used in the USA & Canada to terminate a T1 line. Typically the CSU must be provided by the end user. See: NCTE . See also: DSX1

CSU/DSU- A device which incorporates the functions of a CSU (Channel Service Unit) and a DSU (Data Service Unit) Most commonly it interfaces between a T1, Switched-56, or Dedicated Digital Service circuit and a user's data equipment such as the Zephyr.

CSV- Circuit Switched Voice- A dial-up communications circuit for voice grade communication. Also, the type of ISDN service required to use this capability of an ISDN circuit. In contrast to CSD.

Custom ISDN (USA & Canada)- An ISDN protocol which pre-dates National ISDN-1. In most cases, National ISDN-1 is also available. The Northern Telecom DMS-100 switch Supports "DMS Custom Functional" ISDN. The AT&T/Lucent 5ESS switch Supports "Custom Point-to-Point" (PTP) and Custom Point-to-MultiPoint (PMP). The ISDN protocol has no relation to where one may call. Telos equipment does not support PMP.

D Channel- Data/Delta Channel. Depending on who you ask, it is Data or Delta. The channel which handles ISDN network-related data between the user's equipment and the Telco switch. Used to carry data to set up calls and receive calls. Some Telco's also allow users to use the D channel to access the packet data network, with appropriate terminal equipment.

D4- See: Superframe. See also: Line Format.

Data Channel – See D channel

DCE- Data Communication Equipment. When using serial communications such RS-232, V.35, or X.21, the DCE is the device sending/receiving from the Telco line. i.e.: a modem or CSU/DSU. In contrast to DTE.

DDS- Digital Data System- See Dedicated Digital Service.

Dedicated Circuit- A permanent channel between two locations. As opposed to a Switched Circuit.

Dedicated Digital Service- a "Hardwired" or "Nailed Up" digital circuit which is permanently connected between 2 points. Typically 56Kbps, 64Kbps. Dedicated digital lines are frequently cheaper than ISDN for full time service. Also called Digital Data System.

Delta Channel – An alternative term for the D channel. See D channel.

DID- Direct Inward Dialing. The ability for an outside caller to dial to a PBX extension without going through an attendant or auto-attendant. See also: DID Number and DID Trunk.

DID Extension or DID station- A specific phone within a PBX which can be called from the public telephone network without going through an attendant or auto-attendant.

DID Number- A phone number used to route calls from the telephone network to a specific phone in a PBX (the DID extension). DID requires special DID trunks or ISDN PRI "two-way DID" trunks. Blocks of DID numbers (typically 10 or 20) are purchased from the LEC for use on the PBX. The number of DID numbers usually substantially exceeds the number of trunks in the system.

DID Trunk- A Direct Inward Dialing Trunk. A trunk (channel) which can only receive calls. A group of telephone numbers (DID numbers) are associated with a given trunk group, however there is no one-to-one correspondence between the individual channels and these numbers. The PBX uses the DID number given it by the phone company to route the channel to the correct DID extension within the PBX. This allows some or all PBX stations to receive calls directly without going through an attendant (or auto attendant) Note that there are usually more DID numbers than there are DID trunks. See: DID number and DID extension.

Direct Inward Dialing- See DID

Directory Number (USA & Canada)- Your seven digit telephone number (without the area code), as found in the telephone directory.

Directory 1&2 (Zephyr)- The Utility menu on the Zephyr where the 7 digit Directory Numbers can be entered during set up.

DNIS- Dialed Number Identification Service- A service, typically offered by a long distance company on 800 lines, that provides the number dialed by the caller. This allows a caller to receive specific treatment depending on the number dialed.

DP - Dial Pulse. A method off sending address information by either 1) Causing brief interruptions in loop current, or 2) Causing brief changes of state of a bit on a digital circuit using Channel Associated Signaling. In other words, "rotary" or "pulse" dialing. See also: DTMF and CAS.

DSØ- Digital Signal Level Zero. The smallest unit of measure of the standard rate hierarchy used by the Telcos (i.e. all other rates are a multiple of the DSØ rate. For example, the T1 rate is 24 times the DSØ rate and the E1 rate is 32 times the DSØ rate). 64 kbps. See also: B channel.

DS1-Digital Signal Level 1. The second level up the digital rate hierarchy used by the Telcos. This is 24 times the DSØ rate for a total of 1.544 mbps. See also: DSØ and T1.

DS2-Digital Signal Level 2. Data rate of 6.312 mbps (4 times the DS1 rate). See also: DSØ and DS1.

DS3-Digital Signal Level 3. Data rate of 43.232 mbps (28 times the DS1 rate or 7 times the DS2 rate). See also: DSØ and DS2.

DSL- Digital Subscriber Line. Typically refers to an ISDN line or a T1 line, although the term is also frequently used to mean the next generation beyond ISDN. Sometimes xDSL is used to indicate that the writer is referring to any of a number of emerging DSL technologies.

DSU- Data Service Unit. See CSU/DSU.

DSX-1- Digital Cross Connect level 1. Defined as part of the DS1 (T1) specification and is a closely related signal. The type of signal switched by a Digital

Cross-Connect System (DACS). DSX-1 is also the type of signal that arrives at the user side of a CSU on a T1 line. A DSX-1 cable is limited to 655 feet (200 meters).

DTE- Data Terminal Equipment- When using serial communications such RS-232, V.35, or X.21, the DTE is the device sending/receiving from a modem or CSU/DSU. In contrast to DCE.

DTMF – Dual Tone Multi Frequency. The standard tone-pairs used on telephone terminals for dialing using in-band signaling. The standards define 16 tone-pairs (0-9, #, * and A-F) although most terminals support only 12 of them (0-9, * and #). These are also sometimes referred to as "Touch Tones". Note that while digital data terminals have the same symbols, ISDN uses "common channel signaling" (over the D channel) and therefore does not necessarily generate any tones at all. However many terminal still generate the tones since they will still be used on occasion to access services (such as voicemail or automated attendant) at the far end using in-band tones.

E1- A common type of digital telephone trunk widely deployed outside the US and Canada. Has 31 available 64Kbps channels (called DS∅'s) plus a sync/control channel for a total rate of 2.048 mbps. See also: CEPT format.

E-1- See: E1

EOC – Embedded Operations Channel. A low rate signaling channel included in certain digital circuits to allow commanding NCTE into diagnostic states such as loopback and to convey performance reports. Sometiems called a "facilities data link'. Typically this information is handled at a very low level (e.g. at the frame level). The ISDN U interface and ESF framed T1 have EOC's. See also: FDL.

ESF- Extended Superframe. A type of Line format supported on T1 circuits. The Telco determines the line format and line encoding of your line. See: Line Format

ETS 300- The pan European telephone standards standardized by ETSI. Often used to refer to the ETS 300 ISDN protocol; this protocol is used throughout Europe and has been adopted in many other countries outside the USA & Canada. See also: MSN.

ETSI - European Telecommunications Standards Institute

Euro ISDN- See ETS 300.

Exchange- Another name for a Central Office (most often used in European countries). Also used in the USA & Canada to refer to a particular 3-digit prefix of a 7-digit telephone number. See: CO.

Extended Superframe. See: ESF.

FDL- Facilities Data Link. A bi-directional data link available on T1 circuits when the ESF line format is used. The FDL is primarily used by the Telco to poll the CSU for error statistics.

Four Wire – A circuit path using separate pairs for send and receive. This term is also used when referring to digital channels that inherently have discrete send and receive paths, regardless of the number of pairs (or other media) used.

Frame – A unit of data which is defined by the specific communications protocol used. See Line Format, T1.

FX – Foreign Exchange. An FX circuit has two terminations. The FXO termination connects to a CO while the FXS termination, at the other end, connects to a Station set (e.g. a Telephone). See: FXO and FXS.

FXO – Foreign Exchange Office Interface. The end of an FX line normally connected to a CO. The FXO termination of an FX line appears to the CO as if it were a telephone set.

FXS – Foreign Exchange Station Interface. The end of an FX line normally connected to a telephone set. The FXS termination of an FX line appears to the telephone (or PBX) as if it is a line from the CO.

Glare – On a POTS line an incoming call is signaled by periodically applying an AC ring voltage to the line. Since there is a semi random period before the ring, and pauses between rings, it is possible to seize a line which is "about to ring" (and answer a call) when attempting to place an outgoing call. When this scenario happens it is called glare. Glare is much less likely if Ground Start or ISDN trunks are used. See Ground Start Trunk.

GR-303 – See: SLC-96

Grade of service- This is simply the ratio of calls blocked to total calls in a decimal form. Therefore, a grade of service of P.08 would represent 8% blocking. Telephone tariffs regulate the acceptable average grade of service which must be provided on public networks. See also: Blocking

Ground Start Trunk – A type of telephone trunk where the request to make an outgoing call (i.e.

request for dial tone) is made by briefly grounding the Tip conductor. Many PBX system use Ground Start trunks as they are less prone to glare than Loop Start trunks. Ground Start lines are sometimes used with equipment designed for Loop Start lines. This may or may not – generally it serves to prevent outgoing calls while incoming calls work normally. Telcos may call these a "ground start line". See: Loop Start Trunk. See also: Glare.

HDB3- **High Density Bipolar 3**. An E1 line coding method. This is the more modern line coding method of the two commonly available. See Line Coding, T1. See also AMI and B8ZS

Hunt group- A group of telephone channels configured so that if the first is busy (engaged) the call goes to the next channel, if that channel is busy it goes to the next channel, etc. Hunt groups may hunt from the highest to the lowest, the lowest to the highest, or on some other arbitrary pattern. But the order of hunting will usually be fixed, beginning with one channel and working through ("hunting") until an unused channel is found. The term may have originated back in the old manual switchboard days when the operator literally hunted for an unused jack to plug a cord into. This arrangement is very common in business scenarios where a single incoming number (the Listed Directory Number) is given to the public, but multiple incoming channels are supported. See: LDN.

Hybrid – A device which converts from a two-wire signal such as POTS lines (or a 2-wire intercom) to a four-wire system (separate send and receive paths) such as used in the pro-audio world. While this task is theoretically quite simple, the fact the impedance of most phone lines varies widely across frequency complicates matters. The Telos 10 telephone system was the first practical DSP based hybrid and applied the then brand-new technology to this problem.

IEC -1) InterExchange Carrier. "Long Distance" carrier. Handles Interlata and interstate calls. Also referred to as IXC. **2) International Electrotechnical Committee.** A European standards body best known for the power plug now used throughout the world for AC power cords for use on office equipment and computers.

ILEC – Incumbent Local Exchange Carrier. A local Exchange Carrier which entered the marketplace before the enactment of the 1996 Telecom act;. i.e. a telephone company which is either an Indi or an RBOC. See: LEC and CLEC.

IMUX – See Inverse Multiplexing

In Band Signaling- A signaling system where network information such as address and routing information are handled over the communications (voice) path itself. Usually the information is represented in the form of tones, however DC loop current signaling also qualifies as In Band Signaling. See also: CCIS.

Incumbent Local Exchange Carrier. See: ILEC. See also: CLEC & LEC

Independent – Any of the phone companies in existence at the time of divestiture that were not affiliated with the Bell System. See: RBOC, LEC, and CLEC.

Indi- See Independent.

Interconnect Company- A vendor of telecommunications CPE other than a BOC or AT&T. This term was originated by AT&T and was meant to be derisive towards the fledgling industry when the courts said it was OK for end users to buy equipment from someone other than the Bell System. This industry flourished, in spite of AT&T's disdain, and ironically, the RBOCs were not allowed to sell CPE under the terms of the break up of AT&T. With the current state of deregulation, the RBOCs are slowly re-entering this business.

Interexchange Carrier- See IEC

Inverse multiplexing- A method of breaking apart a data stream into two parts for transmission, and later recombining them, that does not involve the telephone network. Inverse multiplexing is the most common way of achieving this function in Codecs operating at 128kbps or less over ISDN.

IOC Capability Packages - ISDN Ordering Code system. This system was devised by the National ISDN User's Forum and Bellcore to simplify ordering new ISDN lines in the USA and Canada. Using a single code specifies all line parameters. The Telos Zephyr, ZephyrExpress, TWO, and TWOx12 support IOC package "S". Search for document NIUF 428-94 at NIUF's web page http://www.niuf.nist.gov/ for more information.

ISDN - Integrated Services Digital Network- A relatively new and highly flexible type of telephone service which allows dialing on digital channels with multiple bi-directional "Bearer" channels each with a capacity of 56 or 64 Kbps and a single bi-directional "D channel". ISDN is unique in that it offer dial up point to point synchronous data connections (CSD) as well as a fully digital connection to the voice network (CSV). See: BRI and PRI.

ISDN Protocol - The "language" used for communication between the Telco's switch and the customer's Terminal Adapter. Each ISDN circuit has one protocol, and the protocol has no effect on where or whom one may call. See: ETS 300, National ISDN, and Custom ISDN.

ISDN 2- A term used in Europe for ISDN BRI. Also called SØ. Not to be confused with National ISDN-2. See: BRI.

ISDN 30- A term used in Europe for ISDN PRI. Also called S2M. See: PRI.

ISG – Incoming Service Grouping. See: Hunt Group

IXC- IntereXchange Carrier- See: IEC

Kbps- KiloBits Per Second. Measure of digital channel capacity.

Key Telephone System – A system that allows multiple telephones to share multiple pre-determined telephone lines. The system provides indicators to allow the user's to understand the status of each line available on a given phone. In its most basic form it is up to the user to provide the intelligence to select an unused line, or answer a ringing line, for example. The Telos 10, Direct Interface Module, One-x-Six, TWOx12 and Series 2101can be classified as Key Telephone Systems. See also PBX.

LATA- Local Access and Transport Area. The area within which calls are routed by your Local Exchange Carrier (LEC). Under the divestiture of the Bell System calls going outside of this area must be handled by an Interexchange carrier (IEC). With the latest round of de-regulation the usual IEC companies are being allowed to compete in the IntraLATA long distance market and LECs are beginning to be permitted to handle InterLATA calls.

LDN- Listed Directory Number. When a number of Telco channels share the same hunt group, it is customary to give out only one phone number for the group, although generally each channel will have its own number. The number given out is the "Listed Directory Number" since that is the number that would be listed in the Telephone Directory and given to customers. Sometimes called a Pilot Number. See also: DN and Hunt Group.

LEC- Local Exchange Carrier. Your local telephone service provider which is either an RBOC or an Independent. In other words, a traditional phone company. In contrast to CLEC or IEC.

Line- An electrical connection between a telephone service provider's switch (LEC or CLEC) and a telephone terminal or Key system. An electrical connection between a telephone service provider's switch and another switch is called a trunk. Note that some types of physical lines offer more than one channel. I.E. a BRI circuit has 2 channels, called B channels. This term is a confusing one, so we try to avoid using it. See: Channel. See also: Station Line.

Line card- The circuit in the Telco switch to which your line is connected. On an ISDN circuit the line card performs a role analogous to the NT1 in adapting to and equalizing the circuit to establish OSI Layer 1.

Line Coding, T1- The clock signal for T1 is derived at the far end from the data bits themselves. Therefore, T1 lines have certain restrictions as to the data allowed. No more than 15 zeros shall be sent in a row; and average density of 12.5% ones must be maintained. The CSU is responsible to ensure that these requirements are met. The line encoding method, AMI or B8ZS determines exactly how these requirements are met while still allowing recovery of the original data at the far end. Your Telco will determine the method used on a specific circuit. B8ZS is preferred. E1 circuits have similar restrictions. HDB8 is preferred for E1 circuits.

Listed Directory Number- See: LDN

Line Encoding, T1- See: Line Coding, T1.

Line Format, T1-Modern T1 circuits usually use either Superframe (sometimes called SF or D4) or Extended Superframe (sometimes called ESF) line formatting. The type of framing used is determined by your Telco. ESF is preferred. See: ESF and SF

Local Access and Transport Area- See: LATA

Local Exchange Carrier- See: LEC and CLEC

Long Distance- If your local Telco is a former Bell Operating Company then any call outside of your LATA or any Interstate call is considered long distance and is handled by an IEC. The above is true regardless of whether you are referring to a dedicated line or a dial up call. 2) However, under the current state of deregulation, toll calls within a LATA may now be covered by the IEC, and in some cases RBOCs are being permitted to handle InterLATA calls. See: IEC

Loop- The telephone circuit from the CO to the customers premises. Generally refers to a copper cable circuit.

Loop Qualification- Process of actually measuring the loss on a prospective ISDN line to see if it can be used for ISDN service. The actual loss on the line (usually measured at 40 kHz) is the determining factor whether ISDN service can be offered without a repeater. Generally, ISDN is available up to 18,000 feet from the serving Central Office. It may not be available within this range, or may be available further from the CO. Only a loop qualification can tell for sure. Not all Telcos will extend ISDN lines with repeaters.

Loop Start Line - A plain old telephone line. The telephone terminal signals the "off hook" condition by allowing DC current to flow. See: Ground Start Trunk. See also: Glare.

Loop Start Trunk – A plain old telephone line connected to a PBX switch. See Loop Start Line. The PBX signals the "off hook" condition by allowing DC current to flow. Ground Start Trunks are generally preferred for use on PBXs to prevent glare. The Telco may call this a "Loop Start Line". See: Ground Start Trunk. See also: Glare

LOS- Loss Of Signal. An LED or other indicator that illuminates if a signal is absent. This terminology is commonly used with T1 equipment.

LT - Line Termination - The electrical and protocol specifications for the Central Office end of an ISDN line. If you wish to connect an ISDN terminal (such as a Zephyr Xstream) to a PBX the PBX must support LT ISDN. See: also NT and Line Card

Lucent Technologies- Company which now makes the former AT&T 5ESS switch, as well as various other piece of Telco gear and semiconductors. Lucent was split off from AT&T in 1996 and owns Bell Labs. As of approximately 1999 Lucent sold their telephone set manufacturing business (and the right to use the AT&T name on telephone sets) to V-Tech. V-Tech is using the AT&T name and line as their high-end line. In 2000 the PBX and business telephone division was split off to become Avaya.

MCLD- Modifying Calling Line Disconnect. The parameter on the Lucent 5ESS switch that determines if CPC is active. Should be set to "Yes" if CPC is required. See: CPC

MSN- Multiple Subscriber Number. This is a telephone number associated with an ETS 300 BRI line. Providers of ETS 300 often give you three MSNs with a BRI, although additional MSNs can be purchased. An ISDN terminal will "ring" (provide an alerting signal) only when calls are made to the MSN (or MSNs) entered in that terminal. If a terminal has no MSNs entered it will "ring" whenever there is a call to any of the MSN's on that BRI. See: ETS 300 and DN

National ISDN (USA & Canada)- The USA "standardized" multi-platform ISDN protocol. The first version was National ISDN-1. As of mid 1996, National ISDN-2 had been implemented in some areas, and is fully backward compatible with National ISDN-1.

NCTE – Network Channel Terminating Equipment. NCTE is a general term that can be applied to a CSU or NT1 or other equipment terminating a digital line at the customer's premises. In many countries the NCTE is provided by the Telco. The USA is not one of those countries.

Network Channel Terminating Equipment. See: NCTE.

NFAS- Non Facility Associated Signaling. A D channel that is used to carry signaling information for a PRI that is not the PRI with the D channel. For example, a 23B + D PRI and a 24B PRI could share the same D channel using NFAS. See also D channel and CCIS.

NIUF- National ISDN User's Forum. A user's group formed under the National Institute of Technology (NIST) in the USA. The NIUF was a neutral forum where the switch manufacturers and Telcos could get input from users and CPE manufacturers regarding the implementation of National ISDN. Among NIUF's successful projects have been the IOC ordering codes. Their web page is at http://www.niuf.nist.gov/ . See: National ISDN.

Nortel- Manufacturer of the DMS-100 switch as well as many other pieces of telecom and network equipment. Now called Nortel Networks.

Northern Telecom- The Canadian company which was once the manufacturing arm of Bell Canada (it was called Northern Electric back then). Now called Nortel Networks. See: Nortel

NT - Network Termination - The electrical and protocol specifications for the user end of an ISDN line. See also: LT

NT-1- An alternative expression for NT1. See NT1

NT1- Network Termination Type 1. The termination at the customer premises of an ISDN BRI circuit. The NT1 performs the role of line termination of the "U" interface and Codes/Decodes from the line's 2B1Q coding scheme. The customer end of the NT1 interfaces to other equipment using the "S" or "T" interface. The NT1 is frequently part of the "Terminal Adapter" and is built-in to Zephyr, Zephyr Xstream, Zephyr Xport, Telos TWO and TWOx12 systems sold in the USA & Canada. See also: NCTE

NTBA- Network Termination Basic Access. The term used for NT1 in some countries. See NT1. See also NCTE.

Packet Switching. Packet Switched networks are most commonly associated with Computers, Local Area Networks, and the Internet. In a packet switched network the raw stream of data is broken into individual pieces, called packets. Each packet is routed through the data network, individually and may, or may not, take the same path through the network.

This is somewhat analogous to taking the pages of a book and sending each page as a letter through the postal system. The page numbers would allow reassembly of the book no matter what order the pages were received at the far end. The end user does not know or care that the packets may travel a variety of routes. If a given page did not arrive in a reasonable length of time, one could request that this page be re-sent. Most packet switched systems allow packets to be discarded if the network capacity is exceeded (the postal system is not supposed to do this). This is accommodated by the higher-level protocol, which knows to request that a packet be re-sent if it does not arrive. Therefore, the typical behavior of a packet switched network when overloaded is that throughput decreases (i.e. the network "slows down") as the percentage of discarded packets increases. In stark contrast to Circuit Switched networks. See: Circuit Switching

PBX- Private Branch Exchange. A privately owned telephone switch. Basically, a PBX is a private "business" telephone system which also interfaces to the telephone network. In some circles, 'PBX' implies a manual switchboard whereas 'PABX' (Private Automatic branch exchange) implies a PBX that supports dialing by end users.

Many PBX's can now offer ISDN BRI service, usually over the S Interface. A few vendors are now offering BRI over the U interface as well. PRI over DSX-1/T1 or E1 is also offered in some cases. Be wary of these ISDN protocols since they have not been as well tested as the versions running on "public" switches. They may or may not work with a given piece of CPE.

PIC- Primary Interexchange Carrier (USA). Or pre-subscribed Interexchange Carrier. This is your default "1+" carrier used for interLATA calls. In most areas you may have two PICs, one for interLATA calls, and one for intraLATA long distance calls (in which case it stands for Primary Intraexchange Carrier). In some areas intraLATA long distance calls are still handled by your RBOC, in most you now have a choice. You may be able to discover who you current interLATA PIC is by dialing 700 555-4141.

PMP (USA) - AT&T "Custom Point to Multi-Point" Custom ISDN Protocol. Not supported by Telos products. See: Custom ISDN and ISDN Protocol

POP - Point Of Presence. The local facility where your IEC maintains a switch. This is where your long distance calls get routed so that your IEC can handle them. Also used to describe the local access point of an Internet Service Provider.

Port - This is a pretty general term. Newton's Telecom Dictionary 10th edition defines a port as "An entrance to or an exit from a network". Many phone equipment vendors refer to ports as the physical interface between a Switch and a Line or Trunk. Product literature often refers to the number of ports on a phone system. In this context it refers to the number of phones or lines (or sometimes the combination) the system supports.

POT- Plain Old Telephone. A black, rotary-dial desk phone. Usually a Western Electric model 500 set. Outdated term.

POTS – Plain Old Telephone Service. Regular old-fashioned analog loop start phone service.

PRI – ISDN **Primary Rate Interface**- A form of ISDN with 23 "B Channels" and one "D channel". All 24 channels are on a single cable. Functionally related to T1 telephone circuits. In Europe PRI has 30 "B Channels" and one "D Channel" and one "Sync channel".

Provisioning -The act of configuring an ISDN or other telecommunications path. Also refers to the complete line configuration information.

PS2 Power - Power provided on pins 7 and 8 of the "S" interface cable. This power is used so that a NT1 can provide power to a terminal (usually a phone). In some cases it is used to allow a terminal to power an NT1. The USA versions of the Zephyr, ZephyrExpress, and Telos TWO supply PS2 power in the "S" jack. This power arrangement is also used in the Telos TWOx12 and 2101 Studio Interface to power Desktop Directors.

PTP (USA) - AT&T "Custom Point to Point" Custom ISDN Protocol. Point-to-Point lines have only one incoming phone number which must be dialed twice to connect to both lines (the first call goes to "line 1" and the second call rolls over to "Line 2". See: Custom ISDN and ISDN Protocol

RBOC- Regional Bell Operating Company (USA). One of the regional companies formed when AT&T got out of the local telephone business. Each RBOC (or "baby bell") owns a number of the former "Bell Operating Companies". The Bell Operating Companies are the traditional local phone companies (pre-1984), except where one's service is from an "Independent" (non bell) telephone company or a CLEC. Due to their former association with the Bell System RBOCs are regulated by the FCC differently than are independent Telcos or CLECs. In many cases the Bell Operating Company structure is no longer used. For instance, here in Ohio we now deal directly with the RBOC, Ameritech, while the old Bell Operating Company, Ohio Bell Telephone, no longer exists. Another trend is mergers among the RBOCs (and in some cases the independents as well). See: CLEC and LEC

RD- Receive Data. Data coming from the network, or DCE towards the DTE. Also, a light on a modem or CSU/DSU which lights to indicate presence of this signal.

Red Alarm- An alarm state on a T-carrier circuit that indicates that the incoming signal (at the network interface) has lost frame for more than a few seconds. Normally a Yellow alarm is then returned (i.e. sent back) if a Red alarm is present. A Red Alarm indicates a loss of inbound signal; a Yellow alarm indicates (indirectly) a loss of outbound signal. See also Yellow alarm, Blue alarm, and LOS.

Regional Bell Operating Company- See RBOC

Repeater- A device intended to extend ISDN telephone service (or other digital service) to sites further from the central office than could normally be served. ie: beyond 18,000 feet. ISDN repeater technologies include "BRITE", "Virtual ISDN", "Lightspan", and "Totalreach". Some Telcos do not use repeaters. Compatibility between a given NT1 (CPE) and a repeater is less certain than if that CPE were directly connected to the switch.

Robbed Bit Signaling- A signaling scheme that "borrows" bits on each T1 channel for use as signaling channels. On SF T1's there are two bits, the A bit and the B bit in each direction. On ESF T1's there is also a C and D bit in each direction, although they are rarely used. Using these bits, various older analog trunk interfaces can be emulated over a T1. For instance, address signaling using 10 pulse per second (rotary style) digit groups over these bits. Since robbed bit signaling interferes with the least significant bit, only 7 bits can be used for sensitive data applications, leaving only a 56kbps channel for data applications. See also: CAS and CCIS

Rollover – See Hunt Group

S interface- The electrical interface between the NT1 and the Terminal Adapter or other ISDN equipment. ISDN equipment with built-in NT1's do not necessarily provide access to the S interface (the Zephyr, Zephyr Xstream, Zephyr Xport ZephyrExpress and Telos TWO do). Multiple devices can share an NT1 by connecting on the S interface. Also known as the S passive bus.

S∅- European term for ISDN BRI. See BRI and ISDN2.

S2M- European term for ISDN PRI. See PRI and ISDN 30

Sealing Current- Unlike telegraphy, teletypewriter and POTS lines, most digital lines (such as ISDN) use a voltage rather than current mode of operation. Sealing Current allows a controlled amount of current to be passed through a telecom circuit for purposes of "healing" resistive faults caused by corrosion. Bellcore specifies sealing current on the ISDN U interface. The Siemens EWSD switch does not provide sealing current. Most other ISDN capable switches used in North America do.

SF- Superframe. A type of Line format supported on T1 circuits. The Telco determines the line format and line encoding of your line. ESF is the preferred Line Format on T1 circuits. See: Line Format

Silence Suppression- See Statistical Multiplexing.

SLC-96 – A Subscriber Loop Carrier Circuit system manufactured by AT&T (now Lucent). This allows a remote terminal to support a number of remote subscribers with over a shared digital connection to the CO. SLC-96 has its own version of T1 framing between it and the CO. SLC-96 and similar "SLIC" systems may or may not perform a concentration function. The interface is the Bellcore TR-008 or the newer GR-303 interfaces that are specialized versions of T1 intended to allow transparent transport of analog CLASS features such as Caller ID, Call Waiting, etc. The GR-303 interface is specifically intended to be used as a common point of interconnection between alternative equipment, technologies, and/or networks (i.e. voice-over-DSL, voice-over-IP, etc) and the public switched network. See the following link for additional information from Telcordia: http://www.telcordia.com/resources/genericreq/gr303/index.html See also: SLIC

SLIC- 1) Subscriber Line Interface Circuit, see "Line Card". 2) The equipment used with the AT&T (Lucent) SLCC Subscriber Loop Carrier Circuit, a system used to multiplex a number of subscriber loops onto a single circuit to reduce fixed costs. 3) Also sometimes used generically for other brands of similar equipment.

SPID- Service Profile Identifier (USA & Canada) - On the "National ISDN", "AT&T Custom PMP" and "Custom DMS" ISDN BRI protocols, the Telco switch must receive correct SPID(s) from the CPE before it will allow access to ISDN service. Intended to allow multiple configurations on ISDN lines shared among different types of CPE equipment. While your SPID may include your area code and telephone number, the SPID is distinct from the telephone number. For the National ISDN, Custom PMP, and DMS custom ISDN protocols, the equipment requires that the user to program SPIDs into it. Custom PTP and ETS 300 protocols do not require a SPID. No SPIDs are required for ISDN PRI protocols.

SS7 - Signaling System 7. The internationally adopted Common Channel Interoffice Signaling (CCIS) system. Previous to SS7 the Bell System used SS6 which did not support the International Standards. SS7 does. It allows for substantially flexibility and power in dynamically routings calls. An SS7 database lookup is how a call to a mobile telephone user can handled transparently despite the fact that the user's location may change. Also used to determine what carrier should handle a given toll free call. See also: CCIS

Station Line – A telephone circuit from a PBX to a telephone on that PBX. Since this is a telephone-to-switch connection it is considered to be a "line". See: Line and Trunk

Station Side - The user side of a PBX. The side of the switch that the telephones are attached. Also, occasionally called the 'line side'. The main reason for distinguishing between this and the trunk side is that certain customer related features (Such as Hold and Transfer) are inapplicable to trunks. See also Trunk Side.

Statistical Multiplexing- A method of improving effective bandwidth of a Telco channel. Statistical Multiplexing takes advantage that there are typically many pauses in a conversation. By taking advantage of this fact, and not sending the pauses, improvements in efficiency can be made. Also referred to as silence suppression. See: circuit switched

Subscriber- The customer of a Telecommunications company. This term dates back to when a local Telephone Company was formed at the specific request of a group of customers who agreed in advance to "subscribe" to the service.

Superframe- See SF

Switch- Telephone switching device which "makes the connection" when you place a call. Modern switches are specialized computers. ISDN service is provided from a "Digital" switch, most commonly (in the USA & Canada) an Lucent model "5ESS", Northern Telecom model "DMS-100", or Siemens model "EWSD". The switch, and related software running on it, will determine which ISDN protocol(s) will be available to customers connected to it. See also: PBX

Switch Type (ZephyrExpress)- The utility menu item where the ISDN Protocol is selected. See: Telco Setting and National ISDN.

Switched Circuit- A channel which is not permanent in nature, but is connected through a switching device of some kind. The switching device allows a switched circuit to access many other switched circuits (the usual "dial up" type of telephone channels). Once the connection is made however, the complete capacity of the channel is available for use. As opposed to a dedicated circuit or a packet based connection.

Switched-56- A type of digital telephone service developed in the mid 1980's which allows dialing on a single 56Kbps line. Each Switched-56 circuit has 1 or 2 copper wire-pairs associated with it. Switched-56 is being rapidly replaced with ISDN, which is cheaper and more flexible. See also: CSU/DSU

Synchronous Data- A form of serial data which uses a clock signal to synchronize the bit stream. Since, unlike asynchronous data, no start and stop bits are used, data throughput is higher than with asynchronous data. ISDN and T1 use Synchronous data. See also: Asynchronous Data

T-Link- A proprietary Telos 2.048mbps channelized link. This link uses the DSX-1 electrical protocol and has 30 channels at 64kbps each.

T1- A common type of digital telephone carrier widely deployed within the US, Canada, and Japan. Has 24 64Kbps channels (called DS∅'s). The most common framing scheme for T1 "robs" bits for signaling leaving 56kbps per channel available.

T-1- An alternative expression for T1. See: T1

TA- **Terminal Adaptor**. The electronic interface between an ISDN device and the NT1. The terminal adapter handles the dialing functions and interfaces to the user's data equipment as well as to the NT1 on the "S" or "T" interface.

Tandem Switch- A switch which is between two others. It connects two trunks together. Long distance calls on a LEC line go through a long distance tandem that passes them through to the long distance provider's switch.

Tandem Tie Trunk Switching- When a PBX switch allows a tie line call to dial out of the switch. For example, if switch "A" in Arkansas has a tie line to switch "B" in Boise, Boise could use the tie line to make calls from switch "A".

TD- Transmit Data. Data coming from the DTE towards the DCE or network. Also, a light on a modem or CSU/DSU which lights to indicate presence of this signal.

TEI - Terminal Endpoint Identifier. An OSI Layer 2 identifier used by an ISDN terminal to communicate with the serving CO. Most equipment uses a dynamic TEI assignment procedure.

Telco- Telephone Company. Your local telephone service provider. In the 21st century you generally have a choice of Telcos if you are a business in a major metropolitan area in the USA. Competition is coming to the Telecom industry around the world.

Telco Setting (Zephyr, Zephyr Xstream, Zephyr Xport, TWO, TWOx12)- The menu selection where the ISDN protocol is selected. Choices are Natl I-1 (for National ISDN and DMS-100 Custom Functional ISDN), AT&T Cust or PTP (for AT&T Custom Point-to-Point), and ETS300 (for Euro-ISDN).

Telcordia Technologies- Formerly BellCore. The research and development organization owned by the telephone companies. Telcordia represents the phone companies in developing standards for Telco equipment and in testing equipment compliance to those standards. Telcordia also offers educational and training programs open to all interested parties. BellCore was sold to SAIC in 1997. Telcordia is responsive to both RBOCs and independent Telcos. Their web site is: http://www.telcordia.com . See: GR-303

Telephone Number- See DN and MSN

Telos Customer Support +1 216.241.7225. Generally available from 9am to 6pm EST, but you can try at other times. You may also ask for assistance by emailing to Support@telos-systems.com

Telos Test Line(Xstream, Zephyr & ZephyrExpress)- 24 Hour testline which can be called to verify correct operation of your Zephyr or ZephyrExpress codec. Numbers are +1 216-781-9310 and +1 216-781-9311. The test line transmits in dual/mono mode at 32KHz and can be received in L3 Mono using one B channel or L3 Stereo using 2 B channels. Telos also maintains an identically configured test line in Freising, Germany at 49 81 61 42 061 (dial this number twice).

Terminal Adapter- See: TA

Tie Line- See: Tie Trunk

Tie trunk- A Trunk between two PBXs. Note, a tie line is a dedicated circuit, not a switched circuit. See: Trunk

Trouble Ticket-A Telco "work order" used to track Customer Repairs within the Telco. If you call someone "inside" the Telco's repair department they will need this number to proceed. It will also be needed whenever you call to check on the status of a repair. Always ask for this number when initiating a repair request.

Trunk- A communications path between two switching systems. Note that many trunks may be on a single circuit (if that circuit has multiple channels). The trunks most users will deal with are between the Telco switch and a PBX. However ,a Tie Trunk can connect two PBXs. See also: Tie Trunk and Trunk Group

Trunk Group- A number of telephone channels which are functionally related. Most common is the Hunt Group. Other common types include Incoming Trunk Groups and Outgoing trunk groups. See also: Combination Trunks and Two-way DID Trunks

Two-way DID trunk- An ISDN PRI equipped for direct inward dialing. Most trunks are related to a given phone number, either alone or as part of a hunt group. In the case of a "normal" (ie analog) DID Trunk a group of phone numbers are associated with that DID trunk (or group of trunks) and incoming calls include the DID number, so the PBX can route that call to the correct DID extension. These are one-way (i.e. inward only) trunks. This is exactly how ISDN PRI functions, with the DID information coming in over the D channel. There is a big difference between a normal DID Trunk and a Two-way DID trunk over ISDN PRI. For one thing, ISDN PRI is digital. More importantly is that you cannot dial out over a true DID trunk and you can dial out over a PRI.

A T1 using the E&M Wink Start signaling protocol can provide similar Two-way DID functionality, when ISDN PRI is not available.

Two Wire – A circuit path where only a single pair of wires is used. A hybrid is used to convert from two wire to four wire circuits. No hybrid is perfect, and those used by the phone company can be poor. However, the hybrids in the Telos TWO/2101 family are approaching perfection!

U Interface- The interface between the ISDN "BRI" line and the "NT1". This can be considered the ISDN "phone jack" in the USA & Canada it is a 2B1Q coded signal and is frequently in the form of a RJ-11 or RJ-45 style telephone jack.

Outside the USA and Canada, the U interface is not standardized, since the NT1 is provided by the Telco. In many cases it is 2B1Q, but an individual Telco may choose to use some other method.

V.35- A serial data interface/connector pin out for synchronous data. V.35 uses balanced signal and data lines. Many Zephyr/Zephyr Xstream models support V.35 using part #9812 cable.

Variant- The particular protocol (i.e National ISDN-1 or ETS 300) running on a specific switch. Not all variants are valid for a specific switch. The switch brand and model plus the variant defines the ISDN protocol. See: ISDN Protocol

Virtual ISDN- An alternative to repeaters which uses a local Telco Switch to act as a repeater and which then sends the signal onto another switch which supports ISDN. See also: Repeater

Work Order- See Trouble Ticket

X.21- A serial data interface for synchronous data popular in Europe. X.21 uses balanced data and unbalanced signal lines. Many Zephyr/Zephyr Xstream models support X.21 using part #9822 cable.

revised:

03/2005

3 Appendix 3 – Cable Guide

This appendix is intended to be a reference tool that covers various modular telephone-style connectors used in Telos products.

IMPORTANT!

Be very careful to label wiring used with RJ- style connectors. In some cases, they have power on them, and there is a risk of applying this power to something which does not expect it. For this reason we suggest that you use different colored cables (or connector boots) on the Desktop Director cables versus your Ethernet & ISDN Cables.

Pin Numbering on RJ style jacks

It is recommended practice to install modular wall jacks such that the pins are at the top of the cavity when viewed. This helps to protect the contacts from water when baseboards are mopped and from dust. When the jack is oriented in this position (i.e. looking into the jack with the contact pins at the top) the pins are sequentially numbered left to right. Note that some jacks may not have all pin positions populated, however the numbering would still begin with the first position. For instance the RJ-11 style jack is a 6-position 4-pin jack. Therefore it has pins 2,3,4, & 5. Pins 1 & 6 are missing.

Note that all but a few of the following cables use 8-position 8-pin miniature modular connectors (RJ-45 style) on both ends. In today's rapidly changing facilities it is worth considering using a universal system for most of these cables. This strategy allows the most flexibility since the same wiring configuration could be reused for a different purpose without doing anything but documenting the change!

IMPORTANT!

The T- Link cables must be custom wired. While the standard color code wiring may be used at one end, the other end must reverse certain connectors.

Therefore, we recommend that you use the TIA/EIA-568-A T568A cable specification. This specifies a Category 5 (Cat. 5) cable with 4 pairs wired straight through (both ends wired identically) as follows:

PIN	COLOR
1	White/Green
2	Green
3	White/Orange
4	Blue
5	White/Blue
6	Orange
7	White/Brown
8	Brown

TIA/EIA-568-A T568A color codes. Cat 5 cable Is used and both ends are Identical. The color conductor of each pair may or may not have a white stripe. The other half of the pair may be either white or white with a colored stripe. We will use this color coding in our examples.

Note that TIA/EIA-568-A T568B is electrically equivalent, however the Green and Orange pairs are reversed as follows:

PIN	COLOR
1	White/Orange
2	Orange
3	White/Green
4	Blue
5	White/Blue
6	Green
7	White/Brown
8	Brown

TIA/EIA-568-A T568B color codes. Note that this is electrically Identical to TIA/EIA-568-A T568A, however the green and orange pairs are switched.

Hot Tip!

While TIA/EIA- 568- A T568A and TIA/EIA- 568- A T568B are electrically equivalent, care must be taken that both ends of a given cable utilize the same system. For that reason we strongly recommend choosing one of the two standards and using It throughout your facility.

Desktop Director™ to 2101 Studio Interface Cable Pinout

The Desktop Director interface is electrically very similar to the ISDN S interface "extended" configuration. Wiring practices and termination resistor rules for the S interface apply. In this configuration we are using the optional PS2 power convention so the pair on pins 7 & 8 is needed. This cable uses 8-position 8-pin miniature modular (RJ-45 style) plugs. The wiring uses 3 of the 4 pairs described in TIA-568A/B.

PIN	COLOUR	DESCRIPTION
1	White/Green	Not used
2	Green	Not used
3	White/Orange	Xmt (TE to NT1) to Network +
4	Blue	Rcv (NT1 to TE) from Network +
5	White/Blue	Rcv (NT1 to TE) from Network -
6	Orange	Xmt (TE to NT1) to Network -
7	White/Brown	PS 2 Power - 48 VDC
8	Brown	PS 2 Power ground (+)

Desktop Director™ cable wiring diagram. TIA 568A or 568B standard Category 5 cables may be used. Both ends are wired identically.

Standard 10Base-T Ethernet Cable Pinout

Traditionally Ethernet 10Base-T cables use the Green and Orange pair of the TIA/EIA-568-A T568A or TIA/EIA-568-A T568B wiring configuration. The full 4-pair TIA/EIA-568-A T568A or T568B configurations will work just fine.

Crossover 10Base-T Ethernet Cable Pinout

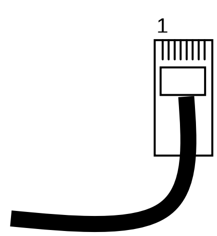
When connecting a 10Base-T device (such as a computer or the TWOx12) to a hub, a "straight through" cable is used. Note that this implies that the jacks on the hub are different than those on the computer, since the "output" of one must correspond to the "input" of the other.

Therefore the standard cable will not work if you wish to connect 2 computers together without a hub. This can be accomplished with the cable below.

10Base- T Crossover Cable		
PIN	***COLOR***	
1	White/Green	3
2	Green	6
3	White/Orange	1
4	Blue	Not Used
5	White/Blue	Not Used
6	Orange	2
7	White/Brown	Not Used
8	Brown	Not Used

ISDN S Interface Cable Pinout

The S interface is standardized around the world and in most countries is the only designated ISDN termination for end users. It specifies an 8-conductor 8-position miniature modular (RJ-45 style) plugs with 4 pairs as shown below (Identical to TIA/EIA-568-A T568A). Unshielded twisted pair category 3 (or higher) cable should be used. We recommend category 5 cable. The cable details are shown below:



BRI S Interface cable.

Note: Pin 1 is to the left when viewed as above (with pins facing you and at the top).

ISDN S interface connector (Worldwide)

PIN	COLOR	DESCRIPTION
1*	White/Green	PS 3 Power +/ground (Optional)
2*	Green	PS 3 Power – (Optional)
3	White/Orange	Xmt (TE to NT1) to Network +
4	Blue	Rcv (NT1 to TE) from Network +
5	White/Blue	Rcv (NT1 to TE) from Network -
6	Orange	Xmt (TE to NT1) to Network -
7@	White/Brown	PS 2 Power 48 VDC – (Optional)
8@	Brown	PS 2 Power +/ground (Optional)
*		Not used in Telos Products
@		Not used on the TWOx12

ISDN S interface cable wiring diagram. Both ends are wired identically.

HOT TIP

This cable has 4 twisted pairs wired "straight through" just like the cable normally used for Ethernet 10Base- T. The wiring configuration is the same as TIA/EIA- 568- A T568A (T568B is electrically equivalent). Since the 2101 Studio Interface does not use or provide power the pair corresponding to pin 1&2 and 7&8 may be omitted, if desired.

T-Link Cable Pin-out

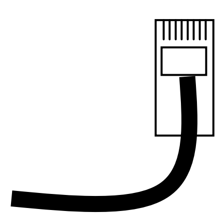
This connection requires 2 pairs. Unshielded twisted pair category 3 (or higher) cable should be used. We recommend category 5 cable.

This cable uses 8-conductor 8-position miniature modular (RJ-45 style) plugs with 4 pairs as shown below (Identical to TIA/EIA-568-A T568A). The cable details are shown below:

PIN	*COLOR*
1	White/Green
2	Green
3*	White/Orange
4	Blue
5	White/Blue
6*	Orange
7*	White/Brown
8*	Brown
* = Not required	

The T-Link connection can be made using a standard EIA/TIA-568-A T568A category 5 cable. Optionally, the cable can be wired with only two pairs as indicated above.

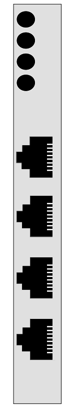
T-Link interface cable



T-Link Interface cable.

Note: Pin 1 is to the left when viewed as above (with pins facing you and at the top).

Crossover T-Link Cable



This cable is only used with the T-Link cards provided with the earliest Telos Series 2101 Hubs. If your Hub has T-Link cards that look like the illustration to the right, you will need to use this cable between the 2101 Hub and the Studio Interface.

This special cable wired as shown below. You should use unshielded twisted pair cable rated Category 3 (Cat. 3) or higher. We suggest you use Category 5 (Cat. 5) cable.

PIN	*COLOR*	*DIRECTION*	*PIN*
1	White/Green	←	4
2	Green	←	5
3*	White/Orange		3*
4	Blue	→	1
5	White/Blue	→	2
6*	Orange		6*
7*	White/Brown		7*
8*	Brown		8*
* Not used on the T- Link Interface; may be omitted			

T-Link cable wiring diagram for use with 2101 T-Link card version A. Pins 1/2 are swapped with 4/5 as shown. Note that this is a special cable and both ends are not wired identically. Note that the T-Link interface does not require all 8 conductors.

HOT TIP!

*This cable must be fabricated on site and is not included with the Series 2101. Note that this is **not** the same as an "Ethernet Crossover cable".*

T1 / E1 Telco trunk cable Pin-out

The Telco Trunk Connection requires two pairs of Category 3 (Cat. 3) or greater cable. We recommend Category 5 cable. A 8-position/8-pin miniature modular (RJ-45 style) jack is required at the Hub end.

If, as is usually the case, the NCTE also uses this jack all that is needed is a 2 pair twisted pair cable wired to the TIA/EIA-568-A T568A or T568B standard.

PIN	*COLOR*	*DESCRIPTION*
1	White/Green	Receive from Network (Ring)
2	Green	Receive from Network (Tip)
3	White/Orange	Not Used
4	Blue	Transmit to Network (Ring)
5	White/Blue	Transmit to Network (Tip)
6	Orange	Not Used
7	White/Brown	Not Used
8	Brown	Not Used

Telco Interface pin functions and wiring diagram. Pin 1 is the top pin of the jack.

HOT TIP

This cable has 4 twisted pairs wired "straight through" just like the cable normally used for Ethernet 10Base- T. The wiring configuration is the same as TIA/EIA- 568- A T568A (T568B is electrically equivalent). Since the 2101 Hub does not use or provide power the pair corresponding to pin 1&2 and 7&8 may be omitted, if desired.

ISDN U interface Cable Pin-out

The U interface, specified in the USA & Canada is specified between the Telco and the user-provided NT1. The U interface is a reasonably robust interface meant for use on existing Telco copper plant. It is a single copper pair. It is provided on the center pair of either a 6-position 4-pin miniature modular jack (RJ-11 style) or a 8-position 8-pin miniature jack (RJ-45 style). The standards specify the compatibility between the 6 position plug and 8 position jack so either can be used. For this reason we prefer the 6 position connectors on cables as they will work with either type of receptacle.

ISDN U interface connector (USA & Canada only)

PIN	FUNCTION	PIN
1	Not Present	6
2	N/C	5
3	Line (R)	4
4	Line (T)	3
5	N/C	2
6	Not Present	1

Typical ISDN U interface cable. This is identical to a POTS telephone cord. Polarity should not matter.

Blank as the new fallen snow is this page.

4 Appendix 4 – Suggested Reading

ISDN/T1

ISDN – A Practical Guide to Getting Up and Running; William A. Flanagan; Flatiron Publishing, Telecom Books, New York, 1996; www.telecombooks.com .

ISDN for Dummies; David Angell; IDG Books, Foster City, CA, 1995.

T1: A Survival Guide; Matthew S. Gast; O'Reilly, Sebastopol, CA, 2001.

T-1 Networking (5th edition); William A. Flanagan; Telecom books, New York, 1997; www.telecombooks.com .

Telephony

The Telephony Book – Understanding telephone systems & services; Jane Laino; Telecom books, a division of Miller Freeman, USA, 1999; www.telecombooks.com . This book is a general introduction, however it is a good starting point for those new to Telephony.

ABCs of The Telephone –Anatomy of Telecommunications; ABC Teletraining; www.abcteletraining.com . A great introduction to all aspects of Telephony.

ABC's of The Telephone – Background for Switching (2nd Edition); ABC Teletraining; www.abcteletraining.com . A good historical account of all the functions any switch, including a PBX, must handle.

ABC's of The Telephone – Principles of Switching (vol 10); ABC Teletraining; www.abcteletraining.com . In depth hard-core switching information.

ABC's of The Telephone – Principles of Traffic and Network Design ; ABC Teletraining; www.abcteletraining.com . In depth hard-core traffic and network planning information.

National Association of Broadcasters Engineering Handbook (9th Edition); National Association of Broadcasters, Washington, 1999; www.nab.org . Chapter 3.10 is written by our founder, Steve Church and is available on the Tech Talk section of our website at: http://www.zephyr.com/techtalk/index.htm .

Newton's Telecom Dictionary (18th Edition and counting) Harry Newton; Telecom books, New York; www.telecombooks.com . This book is more of an Encyclopedia than a dictionary. It is well worth the space on your bookshelf.

Wiring

Mike's Basic Guide to Cabling Computers and Telephones; By Mike Gorman; Praire Wind Communications, USA, 1997. This books covers a lot of ground. Cover "typical" installations for both Telephones and Data. Excellent illustrations offer practical insight.

Technician's Handbook to Communications Wiring; Jim Abruzzino; CNC Press, Chantilly VT, 1999. This book is concise yet contains a lot of great information including proper technique for working with Cat. 5 cable and connectors.

Other

Murphy's Law and Other Reasons Why Things Go Wrong!; Arthur Block; Price/Stern/Sloan Los Angeles; 1977

The Dilbert Future-Thriving on Stupidity in the 21st Century; Scott Adams; Harper Business. New York; 1997

5 Appendix 5 – Ordering Digital Trunks for use with Series 2101

This Document provides information to help order Digital Telco Trunks used with the Series 2101 Hub. This section is duplicated at the following link and will contain any current changes. http://www.telos-systems.com/?/techtalk/isdn_order.htm

For additional information about ordering PRI service, you can call your Regional Bell Operating Company (RBOC) or other local exchange carrier for assistance. Telephone numbers for a number of carriers in the USA & Canada are listed later in this chapter.

5.1 Service Ordering Overview

5.1.1 PRI

The general steps for provisioning PRI service for your Dual T1 card are listed below.

Contact a telephone service provider to order PRI ISDN service. Telephone numbers for ordering PRI ISDN service providers are listed below. The required information includes:

Switch Type used in Central Office (CO)

This is the brand and model of telephone switching equipment used by the Telco to provide the PRI Service. See also *Variant for Switch Type*, below.

Variant for Switch Type

This is the specific ISDN protocol used by the CO switch for this ISDN line. Some switches support more than one type of protocol. The combination of *Switch Type* and *Variant* defines the protocol to be used.

Line Frame Format

Modern T1 circuits usually use either Superframe (sometimes called SF or D4) or Extended Superframe (sometimes called ESF) line formatting. The type of framing used is determined by your Telco. ESF is preferred in the USA & Canada. The preferred frame format for E1 or E1-based PRI is CRC-4 Multiframe with Si=FEBE.

Line Coding

The clock signal for T1 is derived at the far end from the data bits themselves. Therefore, T1 lines have certain restrictions as to the data allowed. No more than 15 zeros shall be sent in a row; and average density of 12.5% ones must be maintained. The NCTE/CSU is responsible to ensure that these requirements are met. The line encoding method, AMI or B8ZS determines exactly how these requirements are met. B8ZS is preferred as it allows complete recovery of the original data at the far end. Your Telco will determine the method used on a specific circuit. B8ZS is preferred in the USA and Canada. E1 circuits have similar restrictions. HDB8 is preferred for E1 circuits.

Having correct and complete information about the above items is essential to successful PRI ISDN service with the Series 2101 Hub.

IMPORTANT!

Users in the USA, Canada & certain other countries are expected to provide a Network Channel Terminating Equipment (Channel Service Unit or CSU). We suggest purchasing a unit that uses 8- position 8- pin (RJ- 45 style) connectors to simplify connection.

PRI TRUNK GROUPS (Group Sizing, Virtual Facilities Groups)

Incoming calls on a PRI are assigned a PRI channel by the Telco switch. If the 2101 "rejects" the call, the channel is released within a few milliseconds, but at least briefly, the trunk is not available for other use.. This means that a large number of calls into the system can prevent access for outbound calls, even if they are associated with a different "Show Configuration". Therefore, a way to restrict the number of call setup messages coming from the Telco is desirable. In the USA the mechanism to do this is to create multiple "trunk groups" (5ESS) or to use "group sizing" (DMS-100) on the PRI to restrict the number of inbound calls to specific number(s) at the CO switch. In the UK a similar function is called "Virtual Facilities Grouping".

The way these work is that after a specific number of calls to specific number(s) are in progress, the network rejects further calls to that specific number(s). Thus, further calls are blocked at the network, rather than using PRI resources just to be rejected by the 2101.

This strategy is particularly needed in the usual case where the number of "appearances" of a particular is substantially less than the number of channels of the PRI (23 in the USA & Canada and 30 in most other countries. Lets use a few examples to explain:

Lets assume we have a 2101 with three telephone numbers.

1111 Hotline

2222 Warm line

3333 Contest Line

One studio exists, with 10 appearances of 3333 and one appearance each of 1111 and 2222

If this were an analog system, the Hotline and Warm line would not be lines from the choke exchange. Therefore, we would not need to worry that the Hot or warm lines could be affected by heavy contest calling on the contest line.

However, in the PRI example above, the following could very well happen during periods of high contest traffic. The Hotline and Warm line could go from the "Idle" status (dot icon) to the "no-line" (blank icon) or "in use elsewhere" (X icon) status.

This is a natural result of the following facts:

A) There is little or no "choking" in this scenario outside of the 2101

B) Each incoming call is assigned, temporarily, a PRI B channel until it is rejected by the 2101.

Consequently, the first 10 calls come through and are put in "ringing" status on lines assigned to line 1111 of the system. The next 13 calls fill up all available channels on the PRI. At this point, the Telco will give additional callers FAST BUSY. However, at this point there are no available channels, so the icons for lines 2222 and 3333 will go blank (or show the "x" symbol) to indicate there are no longer channels available.

However, since a contest is underway, as soon as the 2101 "rejects" one of these 13 calls the Telco will pass us a new call for that idle channel. Hence the congestion continues.

Only after call volume has diminished would lines 2222 and return to the idle state. This is normal, if the system is configured as described above. It can be a bit misleading to see the dots go away, but it is "expected" behavior.

There is a solution. Here's how this would work for our example:

- Trunk Group 1 has 2 channels and 2 phone numbers: 1111 and 2222
- Trunk group 2 has 10 channels and 1 phone number: 3333

Now the drawback is that we cannot make a new show configuration with greater than 10 appearances of 3333 (we could, but we would never get more than 10 calls).

Therefore, we might wish to create a third trunk number and give it another number, 4444 so we have the ability to use more of the PRI.

We could then have a contest simultaneously on 4444 and on 3333 and still never interfere with 1111 and 2222 since 10+10+2 is less than 23.

When planning your installation be sure to give this some thought, as the total number of channels across the trunk groups cannot exceed 23 (assuming a single USA PRI the number would be 30 for ETS-300 PRI's). One of the advantages of Series 2101 is flexibility, and shared facilities can save money. In some cases, a certain degree of overlap may be acceptable. Here is a more sophisticated example, where 2 stations (and AM and an FM) share a 2101.

1111 Hot line for AM and FM

1112 Warm Line AM

1113 AM Morning Show

1114 AM Afternoon Show

2221 Hot line for AM and FM

2222 Warm line FM

2223 FM Morning Show

2224 FM Afternoon show

We could configure the PRI trunks as follows, and still be conservative:

- Trunk Group 1:
  - Number of channels: 4
  - Phone numbers: 1111, 1112, 2221, and 2222
- Trunk Group 2:
  - Number of channels: 10

  - Phone numbers: 1113 and 1114 (we can assume that the morning show and the afternoon show won't be on at the same time)
- Trunk Group 3:
  - Number of channels: 9
  - Phone numbers: 2223 and 2224 (same assumption here)

We could be a bit bolder, if we wish, if we know that both the AM and FM will rarely, if ever, have heavy usage at the same time. However, we would have to accept that occasionally one station could partially swamp out the other. See example, below:

- Trunk Group 1:
  - Number of channels: 3
  - Phone numbers: 1111, 1112, 2221, and 2222

  Note: you will never be able to call into both hotlines and both warm lines, just 3 of the 4 total).
- Trunk Group 2:
  - Number of channels: 10
  - Phone numbers:1113 and 1114 (we can assume that the morning show and the afternoon show won't be on at the same time).
- Trunk Group 3:
  - Number of channels: 10
  - Phone numbers: 2223 and 2224 (same assumptions here).

Of course, other scenarios are possible. Just remember that you can usually have as many groups as you want, however the total number of channels (from all of the groups) must not exceed the total number of channels (23 or 30). Each group can have as many phone numbers assigned to it as you like, however number assigned to the same group have the possibility of causing the all-channels-in-use phenomenon described above.

As we said earlier, you'll need to discuss this with your Telco representative. If they don't understand exactly what you are referring to just make the point that what you want to do is to prevent calls to a single number from blocking all PRI channels. After all, you might need to make an emergency call on the Hot or Warm line even if a contest was in progress.

INCOMING DIGITS (PRI)

The 2101 will use the incoming "Called Party Number" to route the call to a particular active "Show Configuration". Series 2101 can support any number of digits for the Called Party Number. You will need to use the same number of digits when creating your show configurations. For this reason we prefer the full number be delivered (in the USA & Canada this would be 10 digits). If the Telco prefers to send only 7, 4, or 3 digits, this presents no problems. In those cases, just use the appropriate number of digits when creating your "Show Configurations".

NETWORK CHANNEL TERMINATING EQUIPMENT (NCTE or CSU)

Each T1 or E1 must be terminated by a device generically referred to as a Network Channel Terminating Equipment (NCTE). See section 5.2, below for more information on this. The NCTE perform several functions including:

- Line amplification and conditioning of the signal
- Protects your 2101 Telco Trunk Interface Cards from overvoltage or transients present on the digital circuit
- Protects the digital circuit from possible problems with a malfunctioning Telco circuits

Whom to Call

You may order PRI service from either your local telephone company or a long distance carrier. Check with your business account executive first. If they are unsure, you can try one of the numbers below.

COMPANY	TELEPHONE NUMBER	WORLDWIDE WEB
Ameritech/SBC	800-TEAMDATA (800-832-6328)	www.ameritech.com
Bell Atlantic	800-204-7332	www.ba.com
(see below for former NYNEX regions)		
Bell Atlantic North	Call your account representative. If you do not know who s/he is, call;	
(former NYNEX)	800-GET-ISDN (800-438-4736)	www.nynex.com
Bell South	615-401-4347 (Lorilynn Smith)	www.bell.bellsouth.com
Cincinnati Bell	BRI 513-566-DATA (513-566-3282)	www.cincinnatibell.com
	PRI 513-397-1616	
Natco	800-775-6682 ext 288	www.natcotech.com
Nevada Bell		www.sbc.com
Pacific Bell/SBC	800-4PB-ISDN (800-472-4736)	www.pacbell.com
Rochester Tel	800-851-5626	www.rtc1.com
SNET	800-430-ISDN (800-430-4736)	www.sbc.com
Stentor (Canada)	800-578-4736 (Fax server, document 200 has a list of local numbers)	
	For questions or assistance call 403-945-8130	
Southwestern Bell	800-SWB-ISDN (800-792-4736)	www.sbc.com
US West	BRI 800-246-5226	www.uswest.com
	PRI 206-447-8306	
	For questions or assistance 303-441-6988	
Verizon	See Bell Atlantic or Bell Atlantic North	www.verizon.com

For countries outside the USA and Canada see:
www.gbmarks.com/html/international.html

5.1.2 Ordering T1 or E1 service (see above for PRI)

Ordering a channelized T1 or E1 is relatively simple. Each channel acts as if it were a separate phone line. As we discussed in Part I, you will lose quite a bit of flexibility with a channelized T1 since you will only be limited to no more than 24 (T1) or 30 (E1) phone numbers. When ordering channelized service you will need to specify the following information:

Line Frame Format

- For T1 or PRI/T1 you should specify ESF (Extended Super Frame)
- For E1 or PRI/E1 you should specify (CRC-4 Multiframe with Si=FEBE)

If the format listed above is not available, please consult with Telos Systems Technical Support.

Line Coding

- For T1 or PRI/T1 you should specify B8ZS (Bit 8 Zero Substitution)
- For E1 or PRI/E1 you should specify HDB3 (High Density Bipolar 3)

If the line coding specified above is not available, please consult with Telos Systems Technical Support.

Signaling Type

- For T1 you should specify Robbed Bit E&M Wink start
- For E1: Please consult with Telos Systems Technical Support

If the signaling type specified above is not available, please consult with Telos Systems Technical Support.

Hunt Groups

You will need to specify how many hunt groups, if any you wish, and which channels are associated with each.

Phone numbers & Channel Assignments

You will need to get from the Telco the phone number and channel assignment for each channel of a Channelized T1. You will need to get a list of the blocks of "DID numbers" assigned to you PRI. You will be programming this information into the 2101 Hub using the *Configuration Utility for 2101*.

Incoming Digits (T1)

When using "E & M Wink Start" signaling on a T1, it is possible to get a "Two Way DID" arrangement where the DID number is provided for incoming calls over the T1. In this case, the 2101 uses this information to route the call to a particular active "Show Configuration" just as with a PRI. Series 2101 can support any number of digits for the Called Party Number. You will need to use the same number of digits when creating your show configurations. For this reason we prefer the full number be delivered (in the USA & Canada this would be 10 digits). If the Telco prefers to send only 7, 4, or 3 digits, this presents no problems. In those cases, just use the appropriate number of digits when creating your "Show Configurations".

5.2 NTCE, Clocking, power and other issues

NETWORK CHANNEL TERMINATING EQUIPMENT (NCTE or CSU)

Each T1 or E1 must be terminated by a device generically referred to as Network Channel Terminating Equipment (NCTE). The NCTE performs several functions including:

- Line amplification and conditioning of the signal.
- Protecting your 2101 Telco Trunk Interface Cards from overvoltage or transients present on the digital circuit.
- Protects the digital circuit from possible problem with a malfunctioning Telco Interface Card or 2101 Hub.
- Provides various diagnostic capabilities such as error reporting and loopback modes to allow the Telco to test your line.
- Provides a signal when the 2101 Hub is turned off or disconnected, preventing "alarms" at the Telco Central office. This is important since the Telco may "busy out" (turn off) your T1 if such alarms persist. For this reason a NCTE should be powered from an uninterruptible source of power.

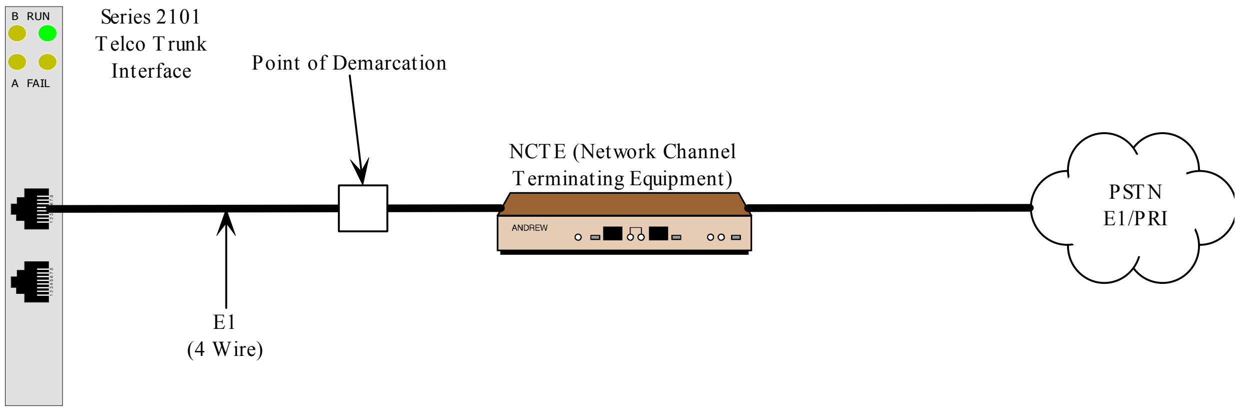
In most countries, this is provided by the Telco, and is often powered by them. In the USA & Canada, the NCTE is called a CSU (Channel Service Unit) and is provided by the end user and you are responsible for powering it (see Volume 2 Part II Section 2.1 and Part III Section 3.1.1 of this manual). Be aware that not all CSUs provide a power supply, so you should investigate this before purchasing a unit. Volume 2 Part II Section 2.1 lists a source for suitable CSU's.

Customers in countries where the user must provide the NCTE should be sure to order this. Check with your sales person to see if this was included in the sales order.

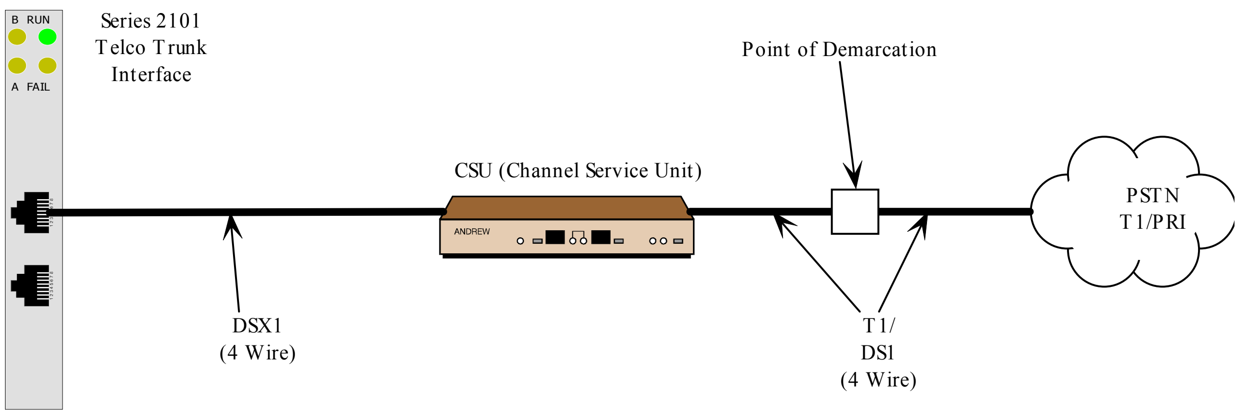
NOTE: The CSU collects error statistics every 15 minutes. It is advantageous for troubleshooting purposes to have a CSU that can display this information, or has a serial port to allow access via computer.

Wiring the Network Channel Terminating Equipment (NCTE) or Channel Service Unit (CSU)

See below for information on NCTE. The Trunk cards in the Series 2101 connect to the digital trunk circuits from the Telco as shown. The point where customer owned equipment interfaces to the telephone network is often referred to as the "point of demarcation" or "demarc". Depending on the on local practices, the demarc will be on one side or the other of the NCTE as shown below.



Outside the USA and Canada the NCTE is generally the responsibility of the Telco. Therefore, the handoff from the Telco to the end user (the point of demarcation) is after the NCTE.



In the USA and Canada the NCTE is called a CSU (Channel Service Unit) and is the responsibility of the end user. Therefore, the handoff from the Telco to the end user (the point of demarcation) is before the CSU.

HOT TIP!

Some service providers sell or lease their own CSUs, and they may require you to use their equipment.

Clocking

The Series 2101 expects to derive clock from the E1, T1 or PRI from the Telco. Since all trunks from the Telco are clocked from the same controlled precision clocks, this works fine.

In the rare cases where the Series 2101 has connections to a PBX as well as the Telco the Series 2101 should be clocked from the Telco only (connect the Telco to Port A and the PBX to port B). In this case, the PBX must also be clocked to the Telco.

In the case where the 2101 is connected only to a PBX, then clock will be from the PBX. In this case the PBX must derive clocking from the Telco, not from the Series 2101 Hub.

When a dual Telco Trunk Interface card is present, use Port A if only one port is to be used. By default the system Telco card l nearest to the CPU card is board 0 (and the other card would be board 1). Port A of Board 0 must have a connected line for clock purposes.

In summary, the Series 2101 normally slaves off the network (Telco clock) either directly or indirectly. Failure to provide proper clocking will cause "clicks" due to buffer overflow/underflow on the digital trunk links.

Power to NCTE/CSU

The Series 2101 trunk interface is powered by the system. It does not use or provide power. The Telco will normally arrange for power to the NCTE if they own it. They should tell you if a source of power is required. If power is required a separate power source would be required since the 2101 does not provide power.

In the USA, Canada, and other locations with customer provided CSU's or NCTE's this equipment must be provided power. Not all CSU manufacturers provide the power supply with the CSU, so be sure to check this when purchasing a CSU.

You should purchase a CSU with an internal backup battery or plug the CSU into an Uninterruptible Power Supply (UPS). Battery backup ensures that the CSU continues to exchange signals with the CO even if your site loses power. The CSU should remain powered at all times to prevent service disruptions. If the CSU loses power and stops transmitting signals, the CO switch reports a Red Alarm on the line. If this condition persists, the service provider may "busy out" your line. In this situation, you will have to call the service provider to restore your service.

ISDN PRI Service Order Summary

What to Order-USA & Canada

Ordering a PRI ISDN line is similar to ordering a T1 line; there are few features or options to consider. When you call to order PRI service, ask for Primary Rate ISDN (23B+D) service provisioned as follows. See order forms later in this chapter.

- Extended Superframe (ESF) framing
- B8ZS line coding.
- Each B-channel provisioned for circuit-switched voice (CSV).
- The D-channel on channel 24 of the PRI.
- No packet handlers other than the D-channel. Some service providers offer special packet data transfer capabilities (such as X.25 messages) over the D-channel; these capabilities are not usable with the Series 2101 and could interrupt service.
- 64 kbps clear channel service end-to-end for every call. Calls that start on ISDN lines may be routed over older T1 digital lines that use a different form of call

signaling that "robs" bits from the channel. This bit robbing reduces the available channel bandwidth to 56 kbps. If your service provider cannot guarantee clear channel service, you may need to turn on rate adaption.

- ISDN Protocol; one of the following protocols
    - Lucent 5ESS Switch; Lucent Custom or National ISDN-1
    - Nortel DMS 100 Switch; Nortel Custom or National ISDN-1
    - Siemens EWSD; National ISDN-1

See later in this Appendix for an ISDN Order Form suitable for Ordering ISDN in the USA & Canada.

ISDN TIP!

Some customers of AT&T have been offered ISDN service off the 4ESS switch, a long distance switch.

Since the 4ESS ISDN PRI is not conformant with the Bellcore 'National ISDN' or Lucent 'Custom 5ESS' ISDN standards, we do not recommend this option. In some cases, AT&T can give you service off a 5ESS switch. Be sure to be clear about this matter, as several customers have been told that the 4ESS protocol is "the same as the 5ESS protocol" which is simply not true.

What to order *outside* the USA & Canada

Use the following information for countries that use the "Euro-ISDN" (ETS 300) protocol. If you are outside the USA and your Telco does not provide "Euro-ISDN" according to the ETS-300 protocol please contact your Telos Representative.

Ordering a PRI ISDN line is similar to ordering a E1 circuit; there are few features or options to consider. When you call to order PRI service, ask for Primary Rate ISDN (30B+D or ISDN-30) service provisioned as follows. See order forms later in this chapter.

- CRC-4 Multiframe with Si=FEBE (far end block error) framing
- HDB3 line coding.
- LAPD signaling; D channel on Channel 16 and 31
- Each B-channel provisioned for circuit-switched voice (CSV).
- No packet handlers other than the D-channel. Some service providers offer special packet data transfer capabilities (such as X.25 messages) over the D-channel; these capabilities are not usable with the Series 2101 and could interrupt service.
- ISDN Protocol; Public Network Switch
    - ETS-300 (ITU-T) compliant

FAXABLE ISDN ORDER FORMS FOR ETS 300 PRI Circuits

Following form may be used to place orders for PRI ISDN (ISDN-30 or S2) circuits in countries using the ETS 300 protocol. This should give the Telecom all of the information they need.

If you need Virtual Facilities Groups, or other means of restricting inbound traffic to specific numbers, you must discuss this with your Telco at the time you place the order.

Complete the top portion of the form and send both pages to the Telco.

ISDN PRI LINE ORDERING INFORMATION
ETS 300 (**not** for use in USA & Canada)
Telos Series 2101

To: Telecom

Attention:____

From:	Location for line:
Company:__________________________________	Company: __________________________________
Address: __________________________________	Address:___________________________________
City / Post Code:_____________________________	City / Post Code:_____________________________
Phone:_____________________________________	Phone:_____________________________________
Contact: ___________________________________	Contact:____________________________________
Long distance carrier:________________________	Number of ISDN-30 (PRI) circuits required: __

Number of Directory Numbers (MSN or DID numbers) needed: ____________

Date needed: ____________________

We request the above number of ISDN Primary Rate Interface (ISDN-30, PRI, S2M) circuits for use with the Telos Series 2101 digital telephone system. This device interfaces audio equipment to digital telephone services. It ***requires*** Circuit Switched Voice (CSV).

The Series 2101 has an integral PRI interface that supports the ETS 300 (Net3) protocol.

We require a standard E1 (ITU-T G.703) 2.048 mbps 4-wire interface on a ***standard, eight-pin/4-conductor RJ45-style modular jack***, of which only four conductors will be used.

Frame Format: Frame format should be CRC-4 Multiframe with Si = FEBE. If this frame format is not available please notify the customer.

Line Coding: Line coding should be HDB3 (High Density Bipolar 3). If this line coding is not available please notify the customer.

Modular Jack :
The interface to the Series 2101 should be 120 ohm cable terminated in a 8-position 8-pin miniature modular jack (RJ-45 style) wired as follows:

Pin	Description
1	Receive Ring
2	Receive Tip
3	No Connection
4	Transmit Ring
5	Transmit Tip
6	No Connection
7	No Connection
8	No Connection

PRI LAPD Protocol: The series 2101 supports ETS-300 (TS014) protocol.

Virtual Facilities Grouping: The customer may wish to have different MSN's in different virtual facilities groups to preserve channels for operations, by preventing calls to their "contest" or "request" numbers from blocking access to the operations numbers. This must be decided, on a case-by-case basis with your customer. Usually 3-4 special numbers will be associated with one group of 3-4 channels and the remaining numbers assigned to one or two additional groups.

Please provide the customer with the following information:

Switch Type and Variant: A Generic Switch with ITU-T variant are the defaults for ETS-300 on the Series 2101.

MSN Numbers/DID numbers: Please provide the customer with a list of all Telephone numbers for this PRI circuit.

Number of digits on incoming calls: Please provide the number of digits that will be sent for Called Party Number on incoming calls.

Facilities Groups: If multiple facilities groups are provisioned, please provide a list of the groups, the size of each, and the DID numbers associated with each.

You may call the manufacturer of the Series 2101, Telos Systems, in the USA at +1 216 241 7225 for any additional required information about ISDN compatibility. Ask for Telos Series 2101 Customer Support.

FAXABLE ISDN ORDER FORMS For PRI ISDN circuits in the USA & Canada

Following forms should be used to place orders for PRI ISDN circuits in the USA & Canada. Since ISDN is complex in those countries, it is necessary to use these forms or you may experience problems. 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 technical support is here to help. You may also wish to look at our troubleshooting information in the 2101 manual.

If you need Trunk Groups (or Group Sizing), or other means of restricting inbound traffic to specific numbers, you must discuss this with your Telco at the time you place the order.

Complete the top portion of the form and send both 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 (at the central office) that your line is configured as ordered.

ISDN PRI LINE ORDERING INFORMATION
USA & Canada
Telos Series 2101

To: Telephone Company

Attention:____

From:

Company:___________________________________

Address: ___________________________________

City / State / ZIP:______________________________

Phone:______________________________________

Contact: ____________________________________

Long distance carrier:__________________________

Location for line:

Company: ___________________________________

Address:____________________________________

City / State / ZIP:______________________________

Phone:______________________________________

Contact:_____________________________________

Number of PRI circuits required: ________

Number of Telephone Numbers (DID numbers) needed: _____

Date needed: ______________________

We request the above number of ISDN Primary Rate Interface (PRI) circuits for use with the Telos Series 2101 digital telephone system. This device interfaces audio equipment to digital telephone services. It ***requires*** Circuit Switched Voice (CSV).

The Series 2101 has an integral PRI ISDN interface that supports these protocols:

AT&T 5ESS:

- -Custom 5ESS
- -National ISDN-1

Northern Telecom DMS-100:

- -DMS Custom
- -National ISDN-1

We can use any of the protocols given above. Please let us know which protocol you will provide and the brand and model of switch. If the switch or protocol is not listed above, notify the customer. ***NOTE: Proprietary signaling protocols, such as TR 41459 are not supported.***

The customer will provide the CSU unless other arrangements have been made through you. We require a DS1 1.544 mbps 4 wire interface ***standard, 8-position 8-pin miniature modular jack (USOC RJ48C)***, of which only the four conductors will be used (see below).

Frame Format: Frame format should be Extended Super Frame (ESF).

Line Coding: Line coding should be B8ZS (Bipolar 8 with Zero Substitution). If this line coding is not available, please notify the customer.

D Channel: D Channel must be Channel 24. Non-Facilities-Associated Signaling (NFAS) is not supported; one D-channel for each PRI is required.

Modular Jack : The interface to the Series 2101 should be a 8-position 8-pin miniature modular jack (RJ-45 style, USOC RJ-48C) wired as follows:

Pin	Description
1	Receive Ring
2	Receive Tip
3	No Connection
4	Transmit Ring
5	Transmit Tip
6	No Connection
7	No Connection
8	No Connection

Trunk Groups (Group Sizing)**:** The customer may wish to have different DID numbers associated with different Trunk Groups, to preserve channels for operations by preventing calls to the "contest" or "request" numbers from blocking access to the operations numbers. This must be decided, on a case-by-case basis with your customer. Usually 3-4 special numbers will be associated with one group of 3-4 channels and the remaining numbers assigned to one or two additional groups.

Please provide the customer with the following information prior to installation:

Switch Type and Variant (protocol). Customer needs to know the actual brand and model of the switch as well as the protocol family used.

Directory Numbers/DID numbers: Please provide the customer with a list of all Telephone numbers for this PRI circuit.

Number of digits for Incoming Calls: Please provide the number of digits that will be sent for Called Party Number on incoming calls. 10 Digits for incoming calls is generally preferred.

Trunk Groups: If multiple trunk groups or group sizing are provisioned, please provide a list of the groups, the size of each, and the DID numbers associated with each.

You may call the manufacturer of the Series 2101, Telos Systems, at +1 (216) 241-7225 for any additional required information about ISDN compatibility. Ask for Telos Series 2101 Customer Support.

The End!

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