TELOS SERIES 2101

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

USER'S MANUAL

PART IV

Studio Interface & Additional Hybrids

V2.0d

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Table of Contents Part IV

1 Series 2101 Studio Interface - INSTALLATION

1.1 Getting Started

If you have not done so already, you should read the material in Part I (Introduction to the Series 2101 and its components) and Part II (Series 2101 Planning & Design).

In this section we will cover the various input/output connections of the 2101 Studio Interface and what you need to know to install the system. Later, in section 2, we will cover the steps necessary to configure the unit for initial operation and test it. Section 3 has a detailed reference of all menus in the Studio Interface.

The Telos Series 2101 Studio Interface mounts in a 2RU high space in a standard 19" rack. The unit will operate in any environment where the ambient air temperature is between 0 to 40 degrees Celsius (32 to 104 degrees Fahrenheit). and relative humidity is between 0 to 98% (non-condensing). The unit will operate on any commonly available AC mains voltage between 90 and 240 volts 50 – 60 Hz. You will want to mount the unit in a position convenient for cable access for the T-Link, Ethernet, and audio connections.

IMPORTANT TIP!

Please see Part II Section 1 of this manual for Series 2101 System Design Criteria. We assume here that you are familiar with the concepts presented there.

You will need to:

- Connect the T-Link circuits from the 2101 Hub
- Connect audio, via either analog or digital (AES/EBU)
- Connect 1 or more Desktop Directors (screener phone/control surfaces)
- Connect to power
- Assign each Studio Interface a unique IP address
- Assign each Studio Interface a unique Host Name (i.e. Studio name)
- Connect to a 10/100Base-T LAN (local area network)
- Configure the hybrids' audio characteristics for the requirements of your installation
- Connect and configure optional Telos TWO hybrid (for studios which require 4 hybrids)

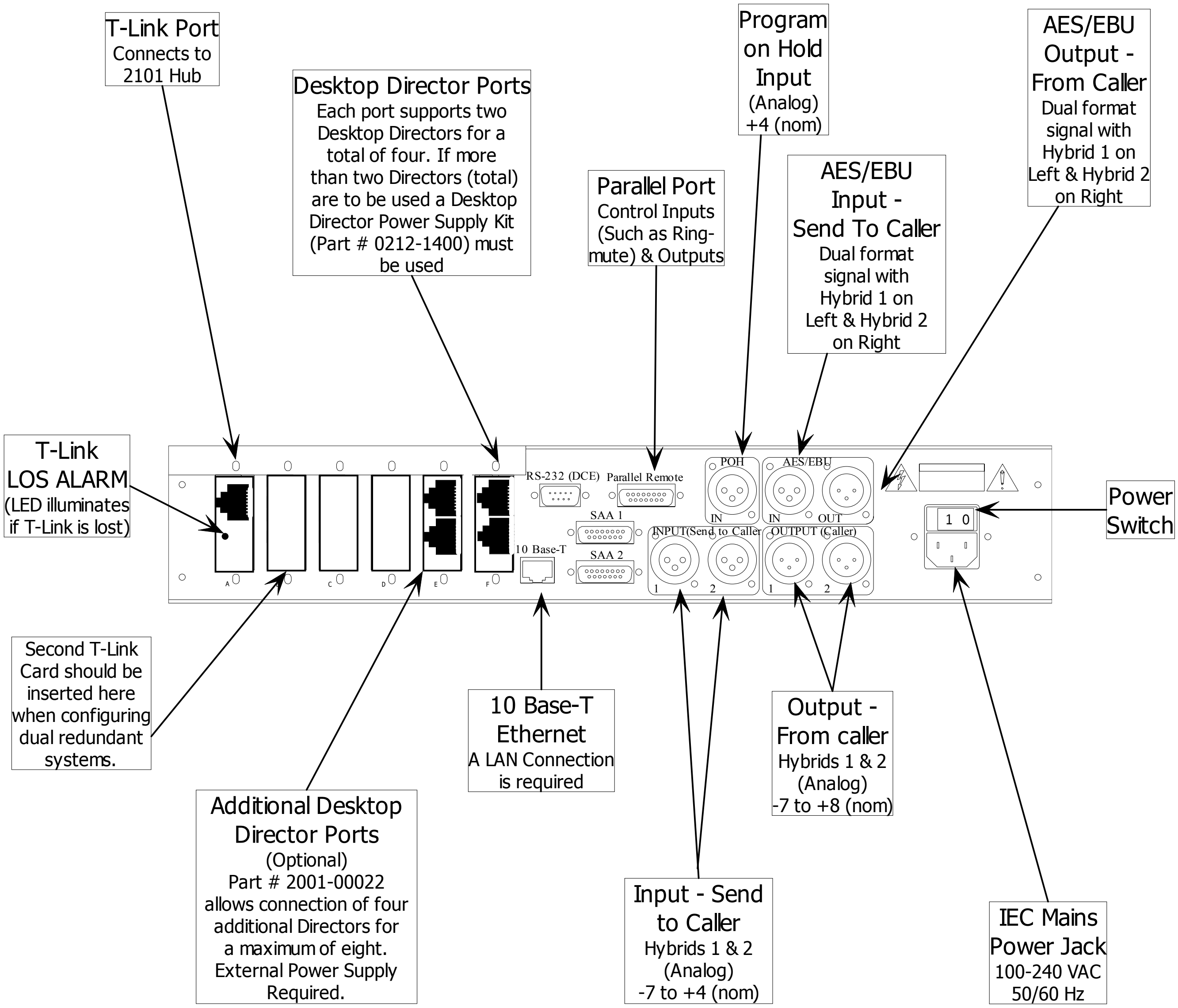
- Configure at least one Telco Trunk interface in the 2101 Hub and configure the trunks associated with that digital telephone circuit.
- Configure at least one *Show Configuration* on the 2101 Hub
- Configure at least one *Studio Configuration* on the 2101 Hub including the *Show Configuration* created above.
- Test to confirm proper operation

Studio Interface connections –

A "maxed out 2101 Studio Interface includes the following connections:

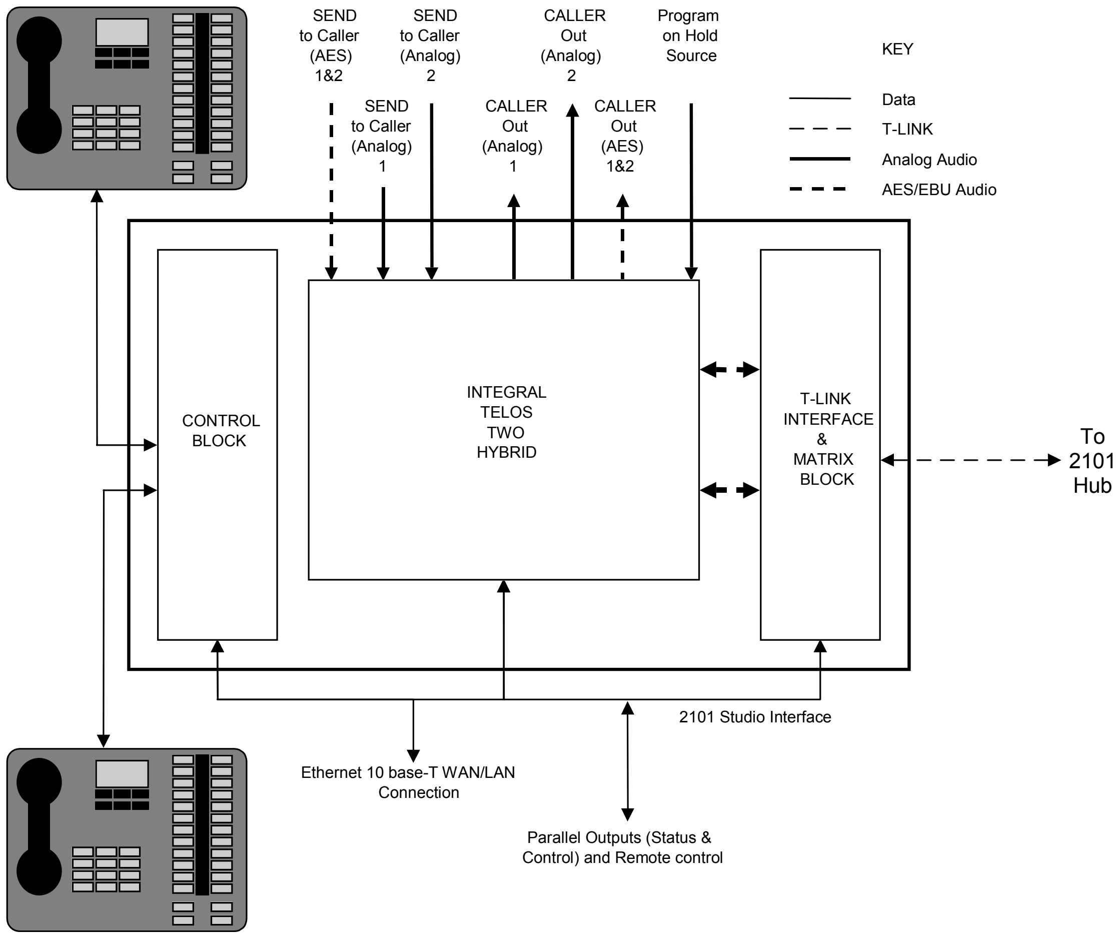
- ➔ 8 Desktop Directors (using optional power supplies and expansion card)
- ➔ 2 T-Link connections to the hub(s) (using optional T-Link Studio Interface I/O)
- ➔ 1 dual channel AES/EBU send-to-caller digital mix-minus source OR
- ➔ 2 send-to-caller analog mix-minus source(s)
- ➔ 1 dual channel AES/EBU Caller output AND/OR
- ➔ 2 Analog caller outputs
- ➔ 1 program on hold analog source
- ➔ Accessory 2101 extended dual hybrid (reduces maximum number of Directors to 6)

Note that there are several configuration choices that you will be making in order to install the system in a given facility. A diagram of the 2101 Studio Interface's inputs and outputs follows:



Rear Panel of the 2101 Studio Interface illustrating the various connectors.

Note: The stock Studio Interface includes a single T-Link Interface and a single Desktop Director Interface. Both AES/EBU and analog outputs are simultaneously active.



The input & output ports of the 2101 Studio Interface.

Note that the system can support up to Eight Desktop Directors™; Only one of the input options (AES/EBU or Analog) can be selected at a given time;

1.2 Installation checklist (Copy this checklist and use it to help organize your installation)

We suggest you look through the rest of section 1 before proceeding. Then work through this checklist as you prepare to install each 2101 Studio Interface.

❑ Bring cable for T-Link circuit(s) from the location 2101 hub to the location of the 2101 Studio Interface. This requires 2 pair for each T-Link connection. See section 1.3.1 for details.

❑ ❑ ❑ How many mix-minuses (with send-to-caller audio) will be used for this Studio Interface? See "Routing" in section 3.3.1 for more on mix-minuses.
2 1

↓ → ❑ If only a single mix-minus will be available, see "Routing" in section 3.3.1.

❑ Bring 1 or 2 mix-minus audio feeds (AES/EBU, or –7 to +4 dBu analog) from your console to the 2101 Studio Interface.

❑ ❑ How many inputs on the console will be used?
2 1

↓ → ❑ If only a single feed from the 2101 Studio Interface to the console will be used see "Routing" in section 3.3.1 for information on your options.

❑ Bring 1 or 2 of the *Output (Caller)* signals from the 2101 Studio Interface to
↓ your console.

❑ Run one of the supplied Desktop Director cables from the 2101 Studio Interface to each of the locations where the Desktop Directors will be located. See section 1.3.2.

If more than two Desktop Directors will be used see section 1.3.2.

If any Desktop Director will be more than 250 feet (75 meters) from the Studio Interface see section 1.3.2.

❑ Run a connection from the 10/100Base-T Ethernet hub to be used for the Series 2101 to the 10Base-T connector of the Studio Interface.

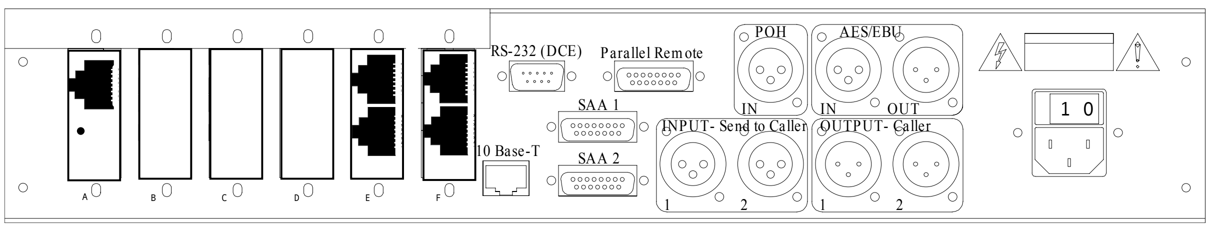
❑ If you will be using any of the 2101 Studio Interface's remote input/output options (such as the ringer mute input) to control the unit see section 1.6.1 and make those connections now.

❑ Proceed to System Programming Checklist in section 2.1.

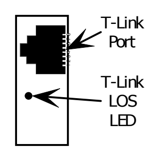
1.3 Connections to the 2101 Hub and Extended Hybrid (Telos TWO)

IMPORTANT!

Unless otherwise indicated the following instructions apply to both the Series 2101 Studio Interface and the Telos Extended dual hybrid (Telos TWO Hybrid with special 2101 software). Additional information on the Telos TWO is included in the manual shipped with that product.



1.3.1 Connecting the 2101 Studio Interface to the 2101 Hub (Studio Interface only)



The 2101 Studio Interface uses internal plug-in T-Link module(s) to connect to the 2101 Hub. These module(s) have a modular jack accessible through each rear slot, in slots A (or optionally, A& B). This will be a 8 position "RJ-45" style miniature connector. An amber (yellow) LED next to below this connector indicates a Loss Of Signal (LOS) and should not be illuminated under normal circumstances.

The Studio Interface ships with a single module in slot A, if you are building a redundant system you will need to add a second module (**Telos Part # 1701-00048**). Section 4 has information on installing or replacing these modules.

IMPORTANT!

Your T- Link connection(s) are in the first 2 Interface slots from the left side of the unit as viewed from the rear of the unit (slots A, and optionally, B). The RJ- 45 style connectors to the right side (slots E & F) are for the Desktop Directors.

T-Link interface connector – Studio Interface

PIN	FUNCTION
1	R1 (Receive/Input)
2	T1 (Receive/Input)
3	Not Connected
4	R (Transmit/Output)
5	T (Transmit/Output)
6	Not Connected
7	Not Connected
8	Not Connected

The T-Link interface is a standard modular RJ-45 style jack with the pin-out shown above.
Pin 1 is at the top of the T-Link jack.

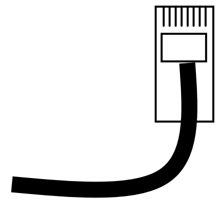
IMPORTANT!

As with any piece of modern electronic gear, it is advisable that precautions be taken to prevent damage caused by power surges. Special "T1" interface surge protectors can be used to offer some degree of protection at the T- Link interface jack. If the T- Link connection will run between two buildings this protection is required. It is the user's responsibility to ensure that adequate protection is provided.

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, T- Link, & other Cables.

T-Link interface cable (for use with Rev B T-Link card in the 2101 Hub)



T-Link Interface cable.

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

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. The above "straight through" wiring is applicable to the 2101 Hub Interface version B only.

T-Link interface crossover cable (for use with Rev A T-Link card in the 2101 Hub)

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		

T-Link cable wiring diagram. 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. This "crossover cable" is for use only when Rev A T-Link cards are used in the 2101 Hub.

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.*

1.3.2 Desktop Director™ Connections (Studio Interface only)

The 2101 Studio Interface has 2 Desktop Director™ ports located in slots F. Each port supports 1 Desktop Director™ directly. The internal power supply of the Studio Interface can power up to two (2) 12 line Desktop Directors without the need of an external power supply. A second Desktop Director can be added to each port (for a total of 4) using a Desktop Director Power Supply Kit (Telos **part # 2001-00024**) available from your Telos Dealer. Each Power Supply kit allows one additional Desktop Director. An expander kit is also required for any Desktop Director located more than 250 feet (75 meters) from the Studio Interface. One Power Supply Kit is required for each two Desktop Directors added in excess of two.

An additional two Desktop Director ports may be added by installing an optional second Desktop Director Interface card (Telos **part # 2001-00022**). See section 4 for instructions on installing this card.

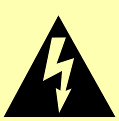
Note that the Expanded Desktop Director (for use with 24 Lines or 4 hybrids) requires a Desktop Director Power Supply Kit (Telos **part # 20001-00024)**.

A Desktop Director port is also used to connect the optional Telos TWO dual hybrid. This reduces the allowable number of Desktop Directors by two. See section 1.3.3 for information on connecting the Telos TWO to your Studio Interface.

Jack on Desktop Director™

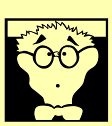
PIN	*FUNCTION*
1	Not used
2	Not used
3	S Transmit to Studio Interface +
4	S Receive from Studio Interface +
5	S Receive from Studio Interface -
6	S Transmit to Studio Interface -
7*	Power Sink –
8*	Power Sink + (Top)

The interface connector on the Desktop Director™ is a standard modular RJ-45 style jack.



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, T- Link, & other Cables.



DEEP TECH NOTE!

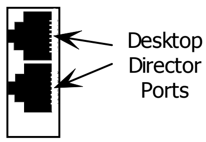
If the above wiring table looks familiar to you, that's not surprising, this looks just like the ISDN S interface on any piece of ISDN terminal gear! That's not a coincidence, our work with ISDN led to using this robust interface for other purposes as you see here.

2101 STUDIO INTERFACE Desktop Director™ Ports (interface slots E (optional) & F)

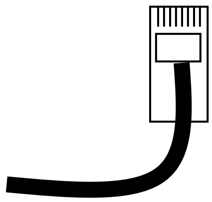
PIN	FUNCTION
1	Not used
2	Not used
3	S Receive from Director +
4	S Transmit to Director +
5	S Transmit to Director -
6	S Receive from Director -
7	PS2 Power Source -
8	PS2 Power Source +

The 2101 Studio Interface has 2 Desktop Director™ ports in Interface slots F. Optionally a second card can be installed in slot E. Pin 1 is at the top. Power is provided on pins 7 & 8 for a maximum of two Desktop Directors. If more than one Desktop Director™ is required on a single port an external power supply will be required (see below)

This interface is a standard 8-position/8-pin miniature modular jack (RJ-45 style) and has the data pairs exchanged as compared to an ISDN S interface on an NT1, or the Desktop Director. Therefore a "straight through" (non-crossover) cable is used between the 2101 Studio Interface and the Desktop Director as follows.



Desktop Director™ Cable



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

PIN	COLOUR	DESCRIPTION
1	White/Green	Not used
2	Green	Not used
3	White/Orange	Xmt (TE to NT) to Network +
4	Blue	Rcv (NT to TE) from Network +
5	White/Blue	Rcv (NT to TE) from Network -
6	Orange	Xmt (TE to NT) 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 should be wired identically.

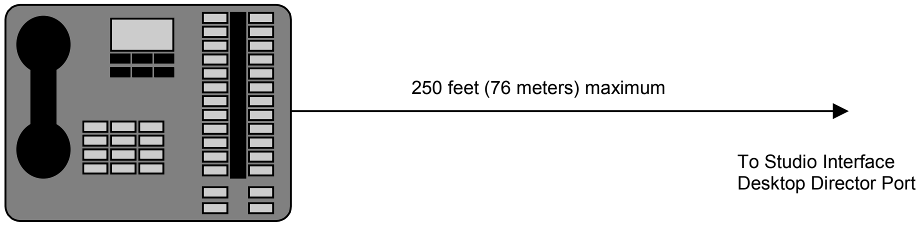
The signal to the Desktop Director™ is electrically similar to the ISDN S interface. This connection is limited to 2300 Feet (700 meters) and wiring should be configured using the extended passive bus configuration (see below for some examples).

HOT TIP!

Since category 5 wire is so common (and prices have dropped) we recommend that you use it instead of category 3 wire for the ISDN, Ethernet, and Desktop Director™ wiring.

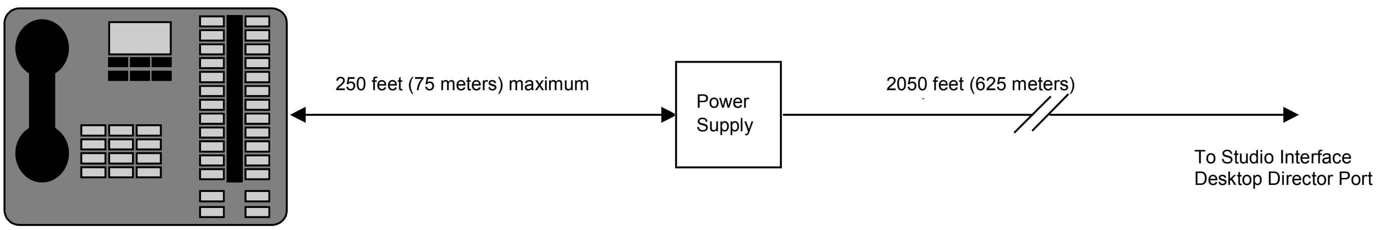
Wiring Configurations for the Desktop Director™

Each Desktop Director™ is supplied with a 25-foot cable which is used to make this connection. You can make your own cables as long as they follow the wiring shown above. Category 3 type (Cat. 3) or higher cable must be used; we suggest Category 5 type (Cat. 5) wire. The usual configuration of two Desktop Directors involves merely plugging in this supplied cable to the Director and 2101 Studio Interface as illustrated below:



The basic Desktop Director™ configuration is simple and straightforward.

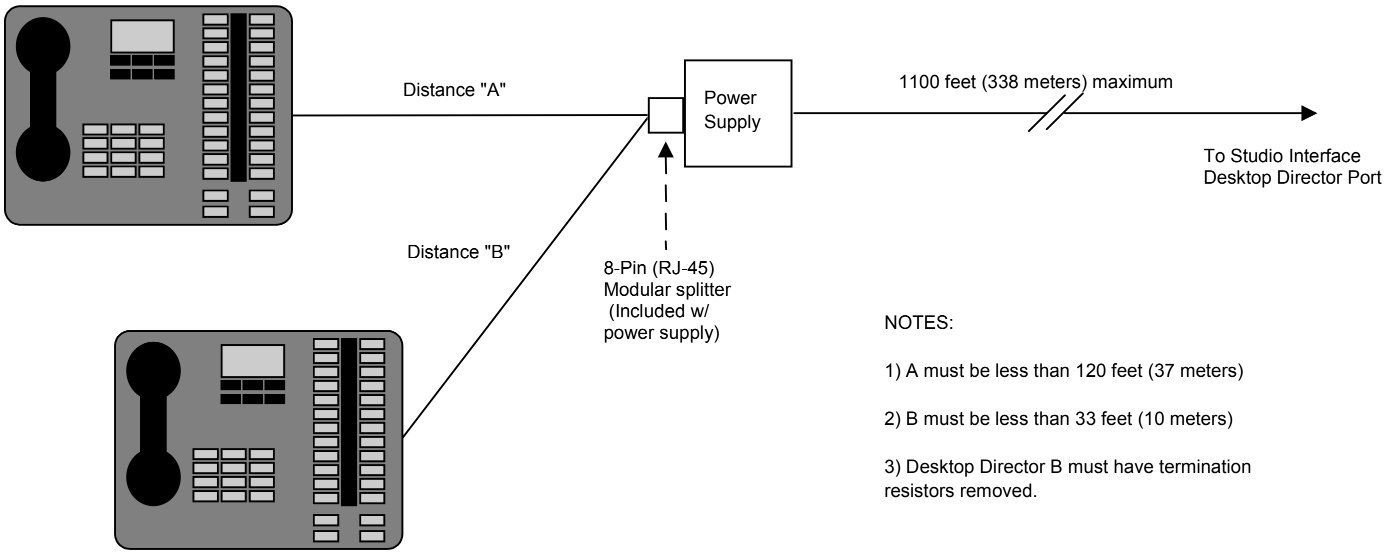
If the Desktop Director™ must be located more than 250 feet (76 Meters) from the 2101 Studio Interface you will need to use a local power supply (**Desktop Director Power Supply Kit part # 2001-00024**) to supply power to it. In that case, your configuration will look like the following:



In the extended configuration the Desktop Director™ must be within 2300 feet (700 meters) of the 2101 Studio Interface. The power supply must be 250 feet (76 Meters) or less from the Desktop Director.

Multiple Desktop Directors on a single port

This connection option is similar to the preceding, and allows two Desktop Directors to be connected to a single Desktop Director™ port of the Studio Interface (allowing for 8 Desktop Directors total). To do so you will need to use a local power supply and modular splitter (**Desktop Director™ Power Supply Kit part # 2001-00024**) to connect it. There are two reasons you might wish to do this. If you need to connect more than two Desktop Directors you would have to use this method. Or, you might choose this method to save on cable runs back to a Studio Interface located some distance away.



In the Multiple configuration one Desktop Director™ must be reconfigured to have its termination resistors removed (see below). This unterminated Desktop Director™ must be within 33 feet (10 meters) of the power supply

Desktop Director™ Termination Resistor Settings

Desktop Directors™ are shipped with the internal termination resistors enabled. When two Desktop Directors are used in a single port, the termination resistors on the Director closer to the power supply will need to be disabled. To do so follow the steps below:

1. Disconnect the Desktop Director cable from the Desktop Director.
2. Remove the handset from its resting place and turn over the Director so you can see the bottom of the unit.
3. Remove the round black cover. This can be pried off with a knife blade. Or a miniature screwdriver may be pushed firmly under the lip and then lifted out to pop the cap loose.
4. You will see a black rectangular transformer labeled T1. To the right of T1 locate and remove the jumpers labeled JP1 & JP2.

   These are small jumper blocks that can be lifted off with needle- nose pliers.
5. You'll want to save the jumper blocks. You can store each of them on one of the jumper pins.
6. Replace the access cover removed in step 3 by pushing firmly in place.

7. Reconnect the Desktop Director cable and turn the unit upright

8. Check for proper operation

1.3.3 Connecting the Telos TWO to the Studio Interface - The S/T Interface (Telos TWO only)

When connecting to the Series 2101 Studio Interface you will be using the 8-position/8-pin miniature modular (RJ-45 style) jack on the back of the TWO. Most units will have two modular jacks. This will be the lower (and smaller) of the two jacks. If a single jack is present you will use that jack.

You will be using a modular cable to connect from this jack to one of the Desktop Director ports of the Series 2101 Studio Interface. The cable used is the same cable used with the ISDN "S interface". See below for a diagram of the basic hookup.

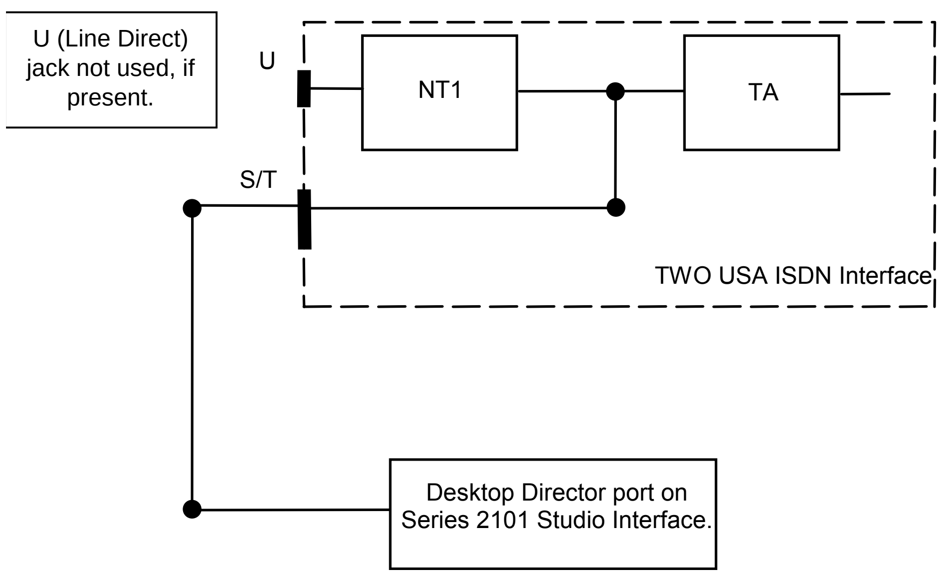
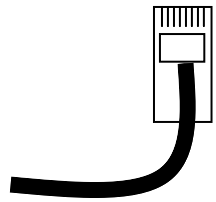


*Diagram of the connection of a Telos TWO to a Studio Interface. The TWO uses both Channels of the Desktop Director port so the port used **cannot** be shared with a Desktop Director.*

The S/T cable has wider plugs than a normal telephone cable, using 8-position/8-pin (RJ-45 style) miniature modular connectors. Four of the wires are used for the S interface and two are used to convey power (these are not needed in this application). Unshielded twisted pair category 3 (or higher) cable should be used. 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).

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 -
8@	Brown	PS 2 Power +/ground
*		Not used on Telos TWO
@		Used only in units shipped to the USA & Canada. Not needed when connecting to Studio Interface

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 10 Base- T. The wiring configuration is the same as TIA/E- 568- A T568A (T568B is electrically equivalent).

The outside pairs (Brown and Green) are often not required and may be omitted.

ISDN S interface connector

PIN	FUNCTION
1	Not Connected
2	Not Connected
3	S Transmit to network +
4	S Receive from network +
5	S Receive from network -
6	S Transmit to network -
7*	PS2 Power -
8*	PS2 Power

The S interface is a standard modular RJ-45 style jack. Pin 1 will be at the top.

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

1.4 Studio Audio Connections

IMPORTANT!

Unless otherwise indicated the following instructions apply to both the Series 2101 Studio Interface and the Telos TWO dual hybrid. Additional information on the Telos TWO is included in the manual shipped with that product.

1.4.1 Mix-Minus

What is a mix-minus?

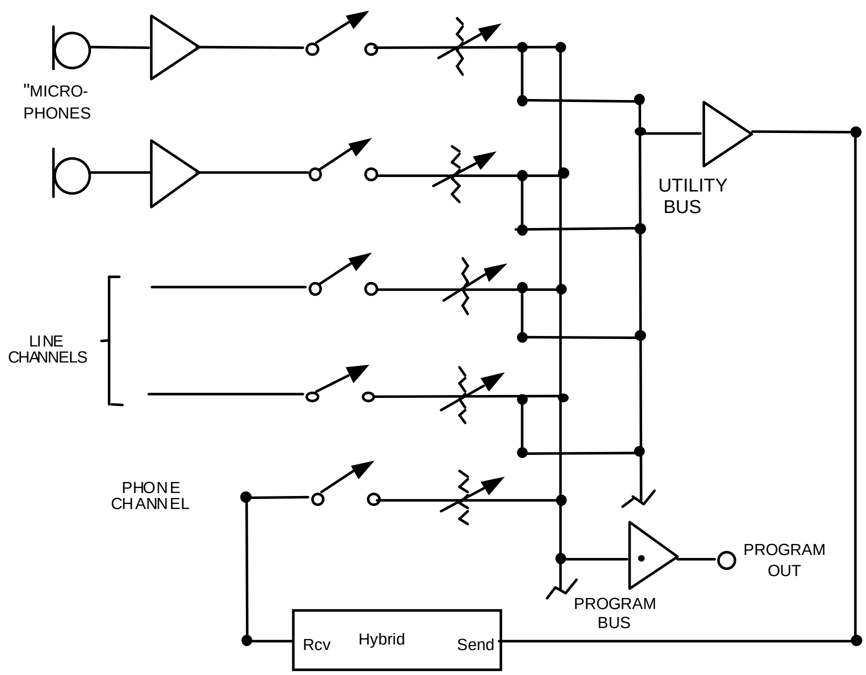
The Telos 2101 Studio Interface (and Telos TWO hybrid) must be fed send-to-caller audio which is free of the caller audio, a “mix-minus”. A mix-minus is a mix of all of your audio sources which will be placed on-air (or recorded) *except the incoming caller audio*–thus the “mix-minus” designation. The European term “mix minus one” is a clearer term for the typical situation. A mix-minus is also sometimes referred to as a “clean feed”. Most modern consoles (mixing desks) are able to handle one or two of these, but more may be difficult. In that case it may be necessary to get creative. The important thing to remember is that the hybrid must not "chase its tail," where its output makes its way somehow back to the input.

Some examples:

Example 1: Using a broadcast console (mixing desk)

A multibuss stereo broadcast console is in use. Program is used to feed the transmitter and Audition is used for some other application. A third buss, "Utility," is not in use, so we will use this to create a mix-minus for a single hybrid.

All sources, including the hybrid, will be assigned to Program, so the audience can hear them, as usual. We will also assign *most* of these sources to Utility as well, just *never* the fader representing the *hybrid's own audio*.



A typical mix minus using an extra bus on a broadcast console. In this case the Utility bus is used to feed a hybrid. Note that the channel containing the Phone audio (labeled "Phone Channel") is not assigned to the Utility bus

Note: The actual bus assign switches are not shown, just the connections.

MIX- MINUS TIP!

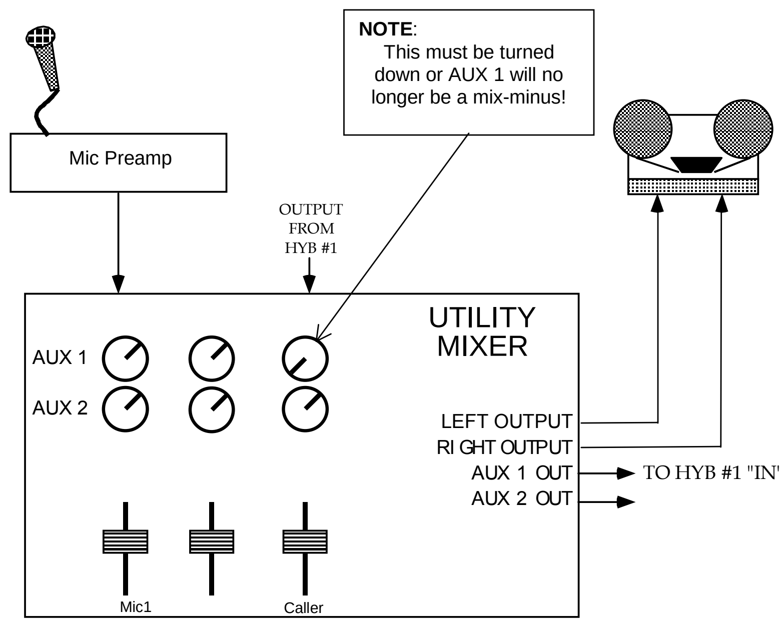
This arrangement is very flexible as it allows the operator to place any or all sources in Utility for the caller to hear. Also, if the talent wishes to talk about the caller without him/her hearing, the talent's microphone need only be taken out of Utility to do so.

The only drawback to this approach is the potential for the operator to accidentally put the hybrid in Utility, in which case it is no longer a mix- minus.

Example 2: Using a utility mixer

A general purpose utility mixer is used to record interviews off the telephone line using a single hybrid. The mixer's main bus is fed to the recording device. Both the microphone and the hybrid will be brought up on the faders so the interview can be recorded.

Again, we will use a second buss to feed the hybrid. In this case, we will take the Aux 1 output to feed the hybrid. We will turn up Aux 1 for the microphone but we will make sure it is turned *fully off* for the input channel with the *caller* audio.



A simple mix-minus using a spare bus.

Why do I need a mix-minus?

Adaptive hybrids, commonly used in broadcast applications since Telos pioneered the technology, work by comparing the audio sent down the line to the audio received back from the line. The system then adjusts itself to minimize send audio in the audio coming back (i.e. minimizes leakage). Thus the audio fed to the hybrid is, in effect, a reference signal. If the audio sent into the hybrid contains the caller audio (the audio we *want*) then there is no reference signal and the system cannot do its job.

CURIOSITY NOTE!

This is why we at Telos do not claim our hybrids will work without a mix minus. Sure, they might work without a mix- minus (it has been done), giving borderline performance, but that's not good enough. Do the job right and use a mix- minus!

An additional problem is the potential for feedback through the hybrid. In any real world situation there will be some leakage of the send audio into the received audio. With gain through the AGC and console, this path creates a loop and feedback can occur. The Telos 2101 Studio Interface's extremely low leakage makes this scenario unlikely, however a clean mix-minus is still essential for optimal performance.

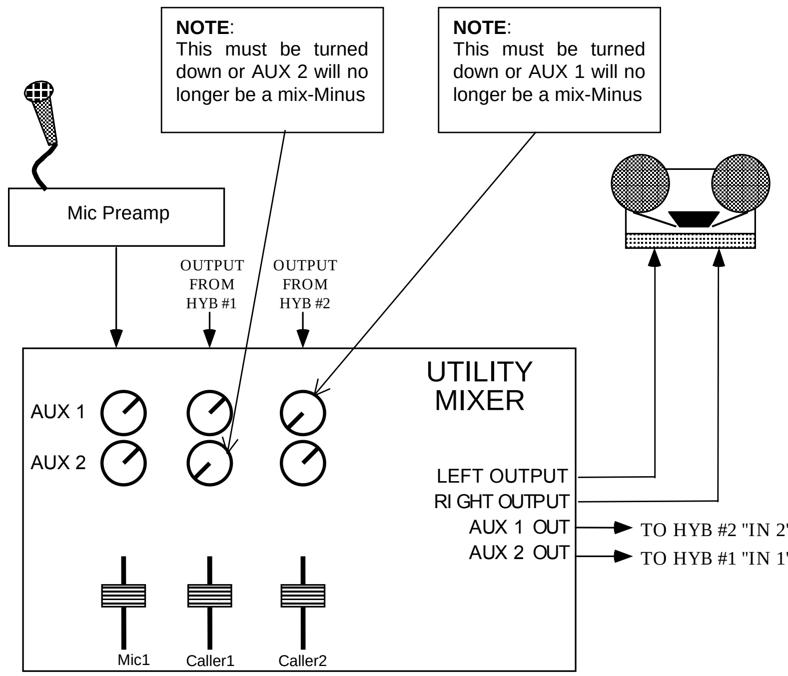
More on Mix-Minus

Simple Mix-Minus

The simplest way to create a mix-minus is to use a distribution amp to feed the mic preamp output directly to the hybrid, as well to the board (mixer, console, mixing desk). This approach lacks flexibility, and requires additional equipment if more than one mic is used and/or more than one mix-minus is required.

Sophisticated Mix-Minus

Most modern broadcast consoles have some provision for mix-minus. The best allow selective feeds to the phone system. This is useful since sometimes you want only one mic. feeding the phone, sometimes you want to three or four mics (during the morning show, for instance), and sometimes you want to play a cart machine, CD, or other device when callers need to hear and react to contest sound effects, etc. Some even provide for separate "on line" and "off line feeds". And when two hybrids are used, each caller will normally need a separate mix-minus since each hybrid will need to have the output of the other, if callers are to hear one another (actually, the 2101 Studio Interface allows for a single Mix-Minus, though it is less flexible, see section 3.3.1 for details). A simplified figure of a two-buss two-hybrid mix-minus configuration follows:



A simple mix-minus created using two spare busses.

HOT TIP!

While we are on the subject, we'll digress here for a moment. Many hybrid installation problems are caused by an inadvertent signal path which creates a loop from the hybrid's output back to it's own input. Some consoles allow this when certain control combinations are selected by the user. In some cases it may be as simple a mistake as assigning the hybrid to whichever buss is feeding the hybrid. This is the first place to look when strange or erratic performance is experienced. The quickest test is to bring up only the hybrid in question on the board and select a line. Dial tone should not appear on the send meter of the hybrid in question.

Phones and Remotes

When on remote, to save money and hassle, calls are usually received at the studio, rather than at the remote site. In this situation, caller audio must be fed to the remote talent so that they can hear and respond to callers. Moreover, the callers need to hear the talent. In many cases, the remotes are sufficiently distant that talent cannot monitor the station for the caller feed. Even if they could, the profanity delay would be a problem, since the talent needs to hear the callers pre-delay.

All perceptual codecs (Such as the Zephyr Xstream) have too much delay for talent at remote locations to hear themselves via a round-trip loop. Therefore, another mix-minus is required to feed the codecs.

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

As for the second half of the equation, the callers hear the talent because the remote feed is added to the telephone mix-minus bus. This is no problem if you have a set-up that permits selective assignment to the hybrid mix-minus.

MIX- MINUS TIP!

If you cannot create a separate codec mix- minus for occasional remotes, you may be able to borrow one of your hybrid mix- minuses and console input to use with the codec for the duration of the remote. Just make sure that your operators understand that only one of the two hybrids is active.

A problem with this arrangement is a result of a hybrid with too much leakage combined with the system delay. If the hybrid isn't doing a good job of preventing the send audio from leaking to its output, the special remote send mix-minus is corrupted. Remember, if any of the announcer audio from the remote site is returned via the monitor feed, it will be delayed by the digital link, causing an echo effect.

MIX MINUS TIP!

If you experience problems where your local monitor mix contains an echo of your own voice, then there is a mix- minus problem at the far end. This could be simply a lack of mix minus, or a contamination of the mix- minus (i.e. open speakers leaking into the mics).

Conversely, if the far end is complaining of an echo, check your mix minus locally.

The Telos Series 2101 Studio Interface really shows its stuff in this situation. Because of its superior trans-hybrid loss, the leakage is not at all likely to be a problem. When there is a problem, you can solve it by increasing the *DuplexLevel* (ducking) selection in the *Audio* menu. See section 3.3.1 for details.

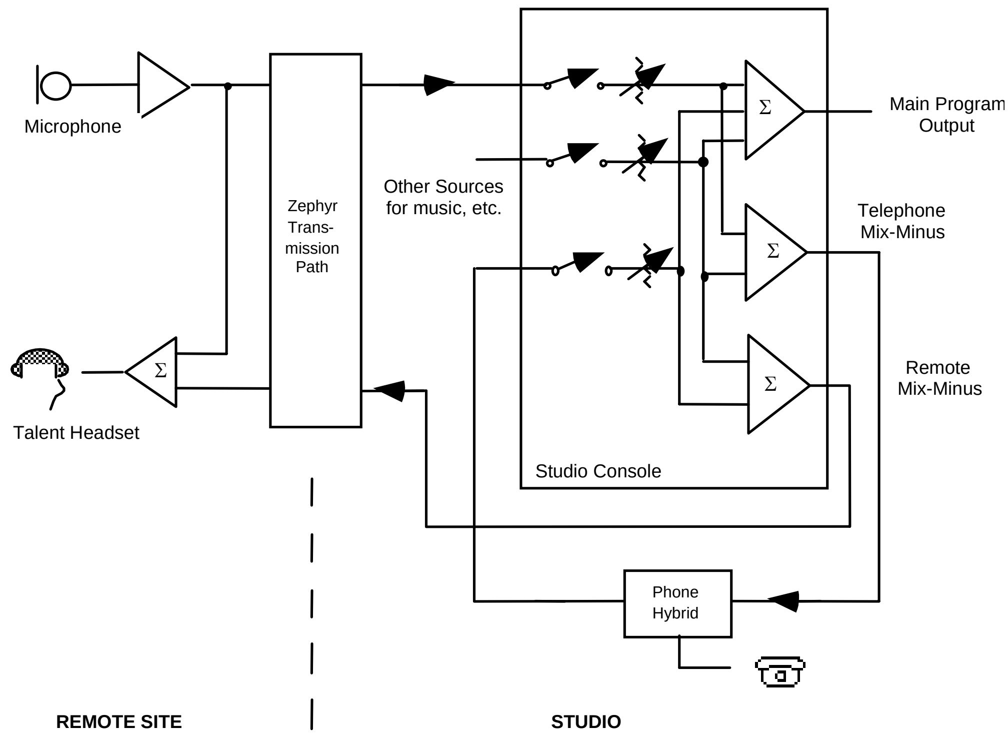
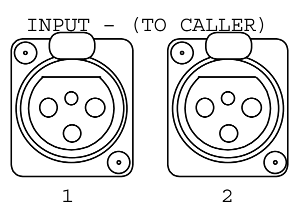


Diagram showing system set-up for remotes with delay in the transmission path and calls taken at the studio.

1.4.2 *Input- (To Caller*) Left & Right Analog Audio



PIN	FUNCTION
1	Ground
2	Audio +
3	Audio -

The analog audio inputs have the following characteristics:

- Active balanced
- Line level: -7 to +8 dBu nominal level (See section 3.3.1).
- Maximum input level (clip point): 18dBu
- Bridging ≥ 100KΩ impedance

The inputs are designed to be sourced from balanced, line level signals. Older equipment with a transformer output stage may need a terminating resistor across pins 2 and 3, consult the manual for your equipment for how to use it with high impedance inputs.

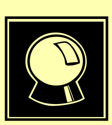
When using unbalanced sources we recommend using pin 2 and 3 to connect the signal and ground, respectively. By leaving pin 1 unconnected, you will prevent ground loops.

Clipping the inputs should be avoided, as it will cause a decrease in hybrid performance. Some consoles with high amounts of headroom may cause clipping. A 10 dB pad will help in these cases.

HOT TIP!

The signal fed to these inputs should be a mix- minus or system performance will be severely degraded.

The sensitivity of the send inputs are adjusted using the Audio menu *Input Level* selection. You will need to set these according to the levels expected at the Hybrid input. When in the *studio in* position, the left LED level meter indicates the level after the menu adjustment, but before any of the Hybrid's digital level adjustments. See Section 3.1 & Section 3.3.1 for additional information.



CURIOSITY NOTE!

The Telos 2101 Studio Interface has the more common pin- outs used for three pin XLR inputs & outputs. You can easily remember the correct signals when wiring connectors using the phrase "George Washington Bridge." Pin 1 = G = Ground, Pin 2 = W = "+" = White (typical color in mic cable, if there is no white there will be a red conductor), and Pin 3 = B = "- " = Black.

1.4.3 *POH* (Program on Hold) Input (Studio Interface Only)

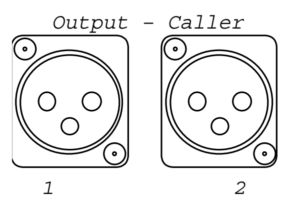
This analog input is for whatever audio you want callers to hear when put on hold, usually the main program feed for the studio associated with this Series 2101 Studio Interface. This source will be fed to any caller put on hold by a Desktop Director connected to this *Studio Interface*.

This is a three pin XLR female with the same input pins as the main inputs. Clipping point is +18dBu (+4 dBu nominal input).

HOT TIP!

You do not need to provide a Program on Hold source to the optional Telos TWO Hybrid. The Program on Hold source fed to the Studio Interface will be used for all callers placed on hold in this studio.

1.4.4 *Output – (Caller)* Left & Right Audio



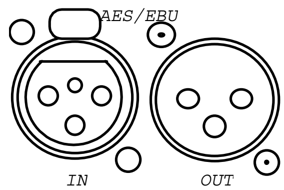
PIN	FUNCTION
1	Ground
2	Audio +
3	Audio -

The analog audio outputs have the following characteristics:

- Active differential
- Output level: -7 to +8 dBu, nominal (See section 3.3.1 for setting this parameter).
- Clip point: +21dBu.
- Impedance: < 60Ω x 2

If a single-ended (unbalanced) output is required, we recommend using pin 2 and 3 to connect the signal and ground, respectively. By leaving pin 1 unconnected, you will prevent ground loops.

1.4.5 *AES/EBU* Digital Audio In/Out



PIN	FUNCTION
1	Ground
2	Output +
3	Output 0

AES/EBU Input & Output Connections

These are inputs and outputs for AES/EBU format digital audio signals.

Input

110 Ω. This input is internally sample-rate converted, so may accept sources at any of the common rates, from 32 to 48 kHz. See section 3.3.1 for setting this parameter.

The left input of this AES/EBU stream will be fed to Hybrid 1, while the right input will be fed to Hybrid 2 (the internal routing of these two signals can be changed using the *Routing* option in the *Audio* menu). Because of this fact you may need an external AES/EBU router or combiner to provide this stream. An alternative would be to use a routing option requiring only a single mix-minus (see *Routing* in Section 7.3.1) or the analog inputs.

Such combiners are available from several suppliers, such as:

Sonifex (www.sonifex.co.uk/technical/brochures/ See # RB-SP1 under "Redbox")

Nvision (www.nvision1.com 800-719-1900 or 530-265-1021, See # NV1055).

HOT TIP!

The signal fed to these inputs should be a mix- minus or system performance will be severely degraded.

See section 1.4.1 for details on mix- minus.

Output

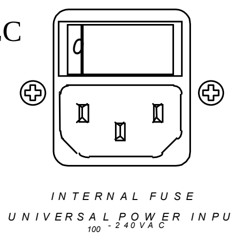
110 Ω. The output can be synced and sample-rate converted to either the frequency of the AES IN signal, or locked to the ISDN network clock/system clock (@ 48kHz), depending upon a parameter in the *Audio* menu. Even if you are using the Analog inputs the AES/EBU input can be fed a reference signal to provide output synchronisation.

Because both the left and right channels are combined on one AES/EBU signal, you may have to use a routing switcher, distribution amp, or console routing function to direct the two signals appropriately for your installation. A combined splitter combiner unit is the RB-SP from Sonifex (www.sonifex.co.uk/technical/brochures/ See # RB-SP1 under "Redbox").

An alternative would be to bridge 2 AES/EBU inputs across this output, which is generally an acceptable practice. You can find more information about setting the *AES Out* sync modes in section 3.3.1.

1.5 AC (mains) Power

The AC receptacle connects mains power to the unit with a standard IEC (International Electrotechnical Committee) power cord and provides an on/off switch. The power supply has a "universal" AC input, accepting a range from 100 to 240 VAC, 50-60 Hz. A fuse is located inside on the power supply circuit board.



IMPORTANT!

As with any piece of modern electronic gear, it is advisable that precautions be taken to prevent damage caused by power surges. Standard line surge protectors can be used to offer some degree of protection. It is the user's responsibility to ensure protection adequate for their conditions is provided.

WARNING!

This equipment is designed to be operated from a power source which includes a third "grounding" connection in addition to the power leads.

Do not defeat this safety feature. In addition to creating a potentially hazardous situation, defeating this safety ground will prevent the internal line noise filter from functioning.

IMPORTANT SAFETY INFORMATION!

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

1.6 Remote Control Ports

IMPORTANT!

Unless otherwise indicated the following instructions apply to both the Series 2101 Studio Interface and the Telos TWO dual hybrid. Additional information on the Telos TWO is included in the manual shipped with that product.

1.6.1 Parallel Remote Control (Studio Interface only)

This is a 15 pin D-Sub connector. This port provides status outputs (tallies) from the system and allows control of certain system functions using simple logic inputs or switch closures. It also has outputs to control external equipment such as an Audio Profanity Delay or Recording device. The pin outs and interface specifications are given here. The available functions for these inputs and outputs are:

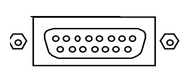
- Record Control Outputs (See Part V, section 2.3.3 for more information)
- Ring Mute In

  This input will mute the ringers on all of the Desktop Directors in Talent Mode physically connected to this Studio Interface.
- User Out (Delay Dump)

  This output will be asserted (low) when the User button is pressed. It will remain asserted until that button is released.
- Ringing Out

  This output is asserted (low) whenever this Studio Interface has one or more lines in the ringing state.
- Priority Ringing Out

  When creating a studio configuration you may choose to omit certain lines from the "Busy All" function (often referred to as "Block All"). For example, "Hot" and "Warm" lines. Lines that are not included in the Block All group will activate this output when in the ringing state.



PIN	FUNCTION
1	Ground
2	Priority Ringing Out
3	Record Start Out
4	Record Stop Out
5	N/C
6	<Reserved>
7	Ring Mute In
8*	+5 VDC (400 mA max)
9	N/C
10	User Out (Dump)
11	Ringing Out
12	N/C
13	<Reserved>
14	<Reserved>
15	N/C
*	This power pin is in parallel with pin 3 of the SAA ports

Note that the +5 volts DC on pin 8 is in parallel with pin 3 on both SAA ports. Failure to limit combined current on these pins to 400mA could cause hardware damage or system instability.

CURIOSITY NOTE!

The DB- designation for D- Sub connectors is from Cinch Corp's part numbering system. The proper Cinch designation for a 15 pin D- Sub connector is a DA- 15p (plug) or DA- 15s (socket) not DB- 15!

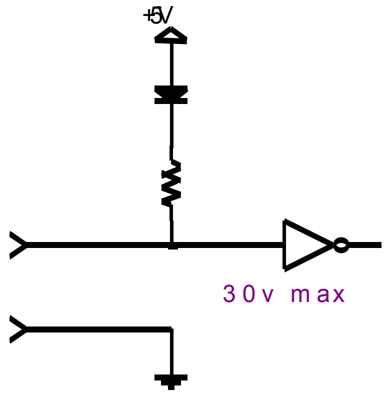
Other Cinch designations for D- Sub connectors are; DE- 9, DB- 25, DC- 37 and DD- 50.

Our thanks going out to Mike Schweizer, for contributing this information.

Inputs

- The function for each of the inputs is shown on the previous page.
- All inputs are specially treated to accept either a voltage (up to 24 VDC), or a closure to ground, which may be provided by switches, relays, or logic outputs. The inputs are active low.

- A built in 1kΩ pull up resistor is provided so TTL outputs can be directly interfaced. See below for a simplified schematic of the input circuitry.



Parallel logic input circuit

DEEP TECH NOTE!

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

Outputs

Outputs are open collector to ground and can sink up to 400 mA of current. Their functions are shown on the previous page.

These will require a pull-up resistor to function with other logic inputs. Some equipment has the pull-ups built into their control inputs – check the device's manual to be sure. If there is no pull-up in the interfaced equipment, you'll have to add one. An appropriate value is 2.2KΩ.

Current should be limited to 400ma maximum per output with total output restricted to 1 amp (200ma each output if all five will be used). A safety diode internal to the output component limits all external control voltages to 5 VDC.

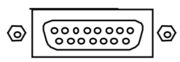
If used with a relay or LED, then tie your external 5 VDC power source ground to pin 1 (or use the 5 VDC power supplied on pin 8) and run this power source through your device, with a resistor in series to limit maximum current to 400mA or less. For external devices requiring greater than 5 VDC for operation you need to buffer the input using some other 5 VDC device to prevent an over-voltage situation on the output pins. Excess current from this over-voltage will damage the outputs.

HOT TIP!

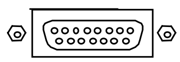
These pins are ignored on a Telos TWO connected to a Studio Interface.

1.6.2 Special Auxiliary Access ports

These ports are unimplemented and are reserved for future use.



SAA 1

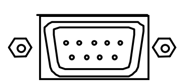


SAA 2

The input and output electrical characteristics are the same as the general purpose inputs and outputs, see section 1.6.1. Note that the +5 volts DC on pin 3 has both SAA ports in parallel with pin 8 on the Parallel remote control port. Failure to limit the combined current on these pins to 400mA could cause hardware damage or system instability.

1.6.3 RS-232 Serial Remote Control

The RS-232 serial port supports asynchronous data, 8 bits, no parity, 2 stop bits, 2400-38,400 bits per second. The Clear to Send (CTS) handshake signal is provided, however any handshake signals from the external equipment are ignored. This port is primarily for troubleshooting purposes and should be used as directed by Telos Customer Support. Connector is a 9 pin D-Sub connector with the following pin-out:



RS-232
DCE

PIN	FUNCTION
2	Rx (Studio Interface to Computer)
3	Tx (Computer to Studio Interface)
5	Ground
8	CTS (Studio Interface to Computer)

Using a 9 pin D-Sub connector, this is an RS-232 serial port using the standard PC-style format, configured as if it were a modem.

HOT TIP!

This port is configured for DCE (Data Communications Equipment) as opposed to DTE (Data Terminal Equipment) operation – meaning that it looks like a modem, not a computer. Therefore, a "reversing" or "null- modem" cable is required to connect to a modem, while a standard "straight through" cable is required to interface to a computer. Cables may be easily fabricated or may be obtained from a computer store.

The following tables show the recommended cable configuration for connecting the 2101 Studio Interface or Telos TWO to 9 pin & 25 pin DTE (computer) serial ports.

CABLE PIN OUT 9-PIN			
Computer (DTE)	**Description**	**Direction**	**Studio Interface (DCE)**
1	*Data Carrier Detect (DCD)	◄	8
2	Receive Data (RD)	◄	2
3	Transmit Data (TD)	►	3
4	*Data Terminal Ready (DTR)	►	Not used
5	Signal Ground (SG)		5
6	*Data Set Ready (DSR)	◄	8
7	*Request to Send (RTS)	►	Not used
8	*Clear to Send (CTS)	◄	8
9	*Ring Indicator (RI) *Optional connection, may not be required	◄	Not used

The serial interface only provides one handshake output signal, CTS (on Pin 8). When standard cables are used it may be necessary to disable "handshaking" or "flow control" in the serial port setup screen of the software.

CABLE PIN OUT 25-PIN			
Computer (DTE)	**Description**	**Direction**	**Studio Interface (DCE)**
2	Transmit Data (TD)	►	3
3	Receive Data (RD)	◄	2
4	*Request to Send (RTS)	►	Not used
5	*Clear to Send (CTS)	◄	8
6	*Data Set Ready (DSR)	◄	8
7	Signal Ground (SG)		5
8	*Data Carrier Detect (DCD)	◄	8
20	*Data Terminal Ready (DTR)	►	Not used
22	*Ring Indicator (RI) *Optional connection, may not be required	◄	Not used

The serial interface only provides one handshake output signal, CTS (on Pin 8). When standard cables are used it may be necessary to disable "handshaking" or "flow control" in the serial port setup screen of the software.

1.6.4 Ethernet Connection

Ethernet 10Base-T connector for connection to an Ethernet Local Area Network. This allows the Studio Interface to communicate with the Series 2101 Hub. This connection is required for the system to operate. The Series 2101 uses IP (Internet Protocol) for its communications. Therefore each 2101 Studio Interface and each Series 2101 Hub will need an IP address and subnet mask (more on entering this later in Sections 2.2.2 & 3.3.3).

Since this connection is essential, we suggest that a separate local area network (LAN) be used for the Series 2101 equipment. Each Studio Interface can be accessed using a Telnet connection over this LAN to configure or monitor the unit. Section 3.4.1 has a list of commands which can be sent over the RS-232 or Ethernet port to control or program the system.

This can be linked to the Internet or your office LAN. If connecting it to another LAN a Switching Ethernet Hub should be used to isolate traffic between the two LANs.

Computers running Assistant Producer for 2101 will also use this LAN to connect to the Series 2101 to allow call screening and control.

IMPORTANT TIP!

Assistant Producer for 3.5 is required for use with the Series 2101, earlier Assistant Producer versions will not work with the system.

HOT TIP!

An IP address and Ethernet connection to the 2101 Studio Interface are required.

However, no IP address, or Ethernet connection, is required to the Telos TWO extended hybrid during normal operation. A connection will be required during software updates however.

Finally, this port is also used for software upgrades over the Internet. For details see section 3.3.3.

10 Base-T Ethernet Connector

This is an industry standard 10Base-T Ethernet connector on the usual 8 position/8 pin miniature modular (RJ-45 style) jack.

The 10Base-T connector 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 light you should check your network wiring. Absence of the link light could also indicate a hardware failure of the Studio Interface.

The amber (yellow) "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.

Blank, blank, blank, I tell you this page is blank! This writing? It does not exist.

2 Studio Interface Configuration

2.1 Pre setup information

The following form is intended to help you collect the information you will need about your installation for entry it into the Studio Interfaces. Before proceeding through the steps outlined in this section, please perform the initial installation (hooking up cables, etc) as described in section 1, *with the exception of connecting the units to the LAN*. See the Installation Checklist in section 1.2. NOTE: An IP address and Ethernet connection are optional for the Telos TWO extended hybrid.

Preinstallation Work Sheet

Please record the information for each Studio Interface on a photo-copy of this form:

Studio Interface Location	Host Name for this Studio Interface	IP Address for this Studio Interface	Notes:
		. . .	
		. . .	
		. . .	
		. . .	
		. . .	
		. . .	
		. . .	
		. . .	
		. . .	

System Programming Checklist (Copy this checklist and use it to help organize your installation)

❑ Set *Audio-Input Source* for *Digital* (AES/EBU) or *Analog*. See section 3.3.1

❑ Set Input & Output levels. See sections 2.4 & 3.3.1

❑ Enter TCP/IP & Subnet addresses. See sections 2.2.2 & 3.3.3

IMPORTANT TIP!

For configuration purposes, every time the IP address Is being changed, the Studio Interface reboots Itself automatically.

❑ Enter a unique Host name. See sections 2.2.2 & 3.3.3

❑ Connect the Studio Interface(s) to the Ethernet LAN

❑ Set up hybrid audio options. See section 3.3.1

❑ Set up 2101 Hub and create at least one *Show Configuration*. See Part III

❑ Review Desktop Director operation. See Part V

❑ Configure 2101 Hub (See Part III)
  - ❑ Configure digital circuits and trunks
  - ❑ Create at least one *Show Configuration*
  - ❑ Create at least one *Studio Configuration* using this *Show Configuration*

❑ Test Desktop Directors and Studio Interfaces

2.2 Initial Installation

We presume you have at least glanced through the material in section 1 and gone through the steps shown in the Installation Checklist (section 1.2). By now you should have your console mix-minus(es) connected to the 2101 Studio Interface, and the Studio Interface's outputs connected to faders on your console (mixing desk). The 2101 Studio Interface should be connected to the 2101 Hub and the Hub must be programmed with at least one Show Configuration. Both the Hub and the Studio Interface must be connected to an Ethernet LAN and have IP addresses programmed. The Studio Interface and Hub (as well as any external Desktop Director power supplies) should have power and be turned on. The LCD screen should be on and you should see the default screen similar to the one illustrated below.

Series 2101

V2.2

TELOS SYSTEMS yyyy-mm-dd

2.3 Telco Setup

2101 Studio Interface

We'll need to configure your 2101 Studio Interface for the appropriate speech coding method for the country you are in. This is determined by the setting Telco option in the Telco menu and must be set correctly for the country of use. In the USA & Canada set it to any option **except** ETS-300. In Europe set it to ETS 300. See section 3.3.2 for information on other countries.

Telos TWO Hybrid

If the optional Telos TWO hybrid is present it will need to be programmed as well. This is determined by the Telco option in the Telco menu. Set it to "Natl I-1" when used in the USA & Canada or to "ETS 300" when used in Europe. See section 3.3.2 for information on other countries.

Telco Setup (Studio interface & Telos TWO):

- Press the TEL button

You will see a screen which looks similar to the following:

Telco Settings

Telco	Natl.I-1
Slot A	T-Link
Slot B	Empty
Slot C	Empty

- Make sure the dark cursor bar is over *Telco* as shown (if not use the ▲ & ▼ keys to move the cursor) and press the *Select* button.
- Press the ▲ button until the appropriate protocol is shown and press the *Select* button again to confirm. See section 3.3.2 for additional information on the correct choice.

2.4 TCP/IP & Hostname Setup (Studio Interface [required] & Telos TWO [optional])

- Press the SYSTEM button

You will see a screen that looks similar to the following:

System Settings

TCP/IP Setup	→
Software Update	→
Contrast	[||||||||||||| 12 ||||||||||]
Backlight	[|||||||||||||||||| 20 ||||||||||||||]

- Make sure the dark cursor bar is over *TCP/IP Setup* as shown (if not use the ▲ & ▼ keys to move the cursor) and press the *Select* button.

You will see a screen that looks similar to the following:

TCP/IP Settings

↵

IP Addr	0.0.0.0
Hostname	
Subnet	0.0.0.0

- Move the dark cursor bar to *IP Addr* using the ▼ key and press the *Select* button. Press the ▲ repeatedly to erase any existing entry for *IP Addr*.
- Enter an unique IP address using the numeric keys. You should consult with your network administrator for IP addresses for the Series 2101 components. Note that an IP address has 4 numbers (each is between 0 and 255) separated by periods ("."). Either "#" or "*" on the keypad will produce a period. Press *Select* to confirm entry.

IMPORTANT TIP!

For configuration purposes, any time the IP address Is being changed, the Studio Interface reboots Itself automatically.

- Move the dark cursor bar to *IP Addr* using the ▼ key and press the *Select* button.
- Enter a subnet mask using the numeric keys. You should consult with your network administrator for a subnet mask for the Series 2101 components. Note that an IP address has 4 number number (each is between 0 and 255) separated by periods ("."). Either "#" or "*" on the keypad will produce a period. Press *Select* to confirm entry.
- (Studio Interface Only) Move the dark cursor bar to *Hostname* using the ▲ key and press the *Select* button. Press the ▲ repeatedly to erase any existing entry for *IP Addr*.
- (Studio Interface Only) Enter a unique name for this studio Interface. Each studio Interface must have a name. This name will be used when selecting a show configuration from the Desktop Director (see part V). The rule for naming the Studio Interface is to keep it short and simple, use NO spaces and NO special characters.

- A word on entering alphanumeric characters

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

For example:

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

To move to the next character in a field, tap any other key or use the ▼ button. To back up use the ▲ button. As usual the *Select* button confirms entry

2.5 Is this Studio Interface working?

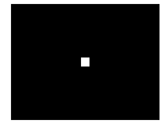
You will need to make sure the 2101 Hub is up and running first. See Part III (The System Hub) for details. You will need to have programmed the Hub for the incoming Telco trunks to be used, and have created one or more *Show Configurations* and one or more *Studio Configurations* (See Part III). Verify that the startup screen (as described above) appears on the Studio Interface and attached TWO (if present). Verify that send to caller audio (your mix minus(es)) from your console appears on the left meters. If this is the case, you know that this Studio Interface (or Telos TWO) has successfully booted up.

From a Desktop Director connected to *this* Studio Interface change the Director's mode to *Talent* mode using the LCD display and function buttons. See Part V (The Desktop Director) section 2.1.2 for details.

IMPORTANT TIP!

If you are testing a studio with both a Studio Interface and an attached Telos TWO Hybrid, you will need to use an Extended Desktop Director (Telos ***part # 2001-00072****), which has 4 rows of line buttons) or you will be unable to access hybrids 3 and 4 (the two hybrids on the Telos TWO).*

Successful connection to a *Show Configuration* will be indicated by the idle line state on the Desktop Director for each line used in the selected *Show Configuration*. The idle line state looks as follows:



Press the *Status* button then press the ▼ button. You should see a screen that says *Slot A - Status* at the top. If you do not see this screen don't panic, just repeatedly press the ▲ or ▼ button until you see this screen.

You should see a screen indicating that an active T-Link connection is present as well as statistics on active channels, etc.

Once you confirmed this we can proceed to make a call and verify that the Studio Interface and 2101 Hub are, in fact, working. We will start out with the simplest case, a call from your "line 1" to your "line 2"as follows:

- Confirm that the Desktop Director™ you will be using has power and says "*Talent*" at the top of the screen. If it says "*Producer*" follow the directions in Part V section 2.1.2 to change this Director to "*Talent*" mode.
- Choose a *Show Configuration* using the LCD screen. See Part V section 2.1.3 for details on selecting a show. If no show configurations appear check the LAN connections at this Studio Interface, 10/100 Base-T LAN hub and the 2101 Hub. Check the Telos 2101 hub is running and configured (see Part III).
- First pick up the handset and push the line button in the left row for each line active in this *Show Configuration*, listening for dial tone. If this does not happen see Part VI for information on troubleshooting your problem.
- Next, pick up the handset and press the left *Line 1* button to select that line for dialing. You should now hear "inside" dial tone on the handset.
- Enter the phone number for line 2 using the numeric keypad of the Desktop Director.
    - You should now hear a ringing sound and see the ringing Status Symbol (an expanding square) on Line 2. Push the right *Line 2* button to answer this call. The → Icon should appear for line one.
    - Drop this test call by pressing the right Drop button .
- If this happens, your "lines" 1 and 2 are working). If this does not happen see Part VI for information on troubleshooting your problem.
    - Now repeat this test by calling from "Line 2" to "Line 3", "Line 3" to "Line 4" etc until all lines have been tested.
- If you do not plan on proceeding to the next section and setting your levels now, be sure to push the *Drop* button after the final call.

HOT TIP!

Many business telephone lines have per- minute usage charges. Don't forget to drop the call once your testing and configuration are complete.

2.6 Levels, Levels, Levels

Ok, we assume that you just made a call to yourself in section 2.5, above (if not, please follow those steps now). Congratulations, you just have some tweaking to do and your basic installation & configuration of this Studio Interface will be complete!

- Bring up some audio on the console (mixing desk). Spoken word is best, as it tends to have a lot of dynamics, but music will work. While tone will work for the coarse adjustments (set it to the red-green boundary point), we do not recommend using it for final adjustments. Adjust the levels on the board to the levels set by your typical operator! NOTE: If your board operators typically run "into the red", you should do the same! For now make sure the faders for both of the hybrids are TURNED OFF.

IMPORTANT!

If your board operators typically run "in the red", you should do the same! We've said this twice since clipping on the hybrid's inputs must be avoided or hybrid performance will be greatly reduced.

In some cases a 10 dB pad may be required.

- Set the left LED meter to *STUDIO IN*, if necessary, by pressing the button located on this meter. You should see the audio from your console on these meters (the send-to-hybrid 1 audio will be on the LED bargraph labeled *1* and the send-to-hybrid 2 will be on the LED bargraph labeled *2*.
- Observe the levels on these two meters. You will note that the highest peak level reading is represented by a "floating" LED that persists near the top of the meter. Since our goal is to set the levels so that peak levels do not hit 0dBfs, this peak reading is what we need to watch. Section 3.1 has more information on the 2101 Studio Interface's metering.
- First set the *Input Level* selection found in the *Audio Settings* menu to a **nominal** level close to that used in your facility (i.e. –7, 0, +4, or +8 dBu). This control sets the level for both inputs. If the inputs are not reading identically, you will need to adjust them at the output of the console. Note

that the lower this selection is set, the higher the levels on the unit's meters, as this setting represents "nominal" input level.

Next adjust your console output (mixing desk) level (for the mix-minus) if necessary. We suggest that you keep the peak level below a -6 to -7 reading, this will allow some room for error on the part of your operators. If you are using a tone to set levels you may need to set them considerably lower. NOTE: Clipping must be avoided or hybrid performance will suffer.

IMPORTANT!

To ensure clipping is not taking place, set the Input Level to +8 and look at the meters. If the last few LEDs on the meter illuminate, insert a pad in- line with the input source.

- Set the right meter to *STUDIO OUT*, if necessary, by pressing the button located on the meter. You should see the audio from the phone line on these LED bargraphs (since we are connected hybrid 1 to hybrid 2 the audio will be reversed. The audio sent to hybrid 1 will return on hybrid 2 and will be on the LED bargraph labeled *2* and the audio sent to hybrid 2 will return on hybrid 1 and will be on the LED bargraph labeled *1*. Bring up the fader for hybrid 1 on your console.
- Adjust the 2101 Studio Interface's *Output Gain* adjustment found in the *Audio Settings* screen. This sets the level for both outputs. Once you have set hybrid 1 pull that fader down and bring up the fader for hybrid 2. If the sources are not reading identically you will need to adjust the input trim of the console on one of the two input channels.
- You can now drop the call you made in the previous section. Push the right *Drop* button.

HOT TIP!

If all of your studio's have identical audio facilities you can write down the settings determined above and enter these settings into all of the Studio Interfaces in your facility.

2.7 Time to test it out!

Ok, you made it! Another hour and you can go home for the day! First we will check that the mix-minus for each hybrid is correct. Then we will make a test call.

- Confirm that the Desktop Director you will be using has power and says "*Talent*" at the top of the screen. If it says "*Producer*" follow the directions in Part V section 2.1.2 to change this Director to "*Talent*" mode.
- Next, pick up the handset and press the **left** *Line 1* button to select that line for dialing. You should now hear dial tone on the handset.
- Enter the phone number for a telephone line that has audio present, using the dial pad (calling either the time or weather service works well for this).
- Press the **left** *Line 1* button and hang up the handset to place the call on hybrid 1.
- Bring up hybrid 1 on the console. You should hear the audio from the phone line.
- Look at the left meter on the 2101 Studio Interface and verify that this audio is ***not*** present on the "*1*" LED bargraph. If this audio is present here you are not feeding hybrid 1 a mix-minus and you ***must*** fix this problem before you proceed.

IMPORTANT!

If you are using one of the routing schemes that requires only a single mix- minus you should ***not*** *have seen audio on either the "1" or "2" bargraphs in the previous test. In that case you can skip the next test.*

See section 3.3.1 for detailed information about the routing options.

Now we will verify the second mix-minus.

- Press **right** *Line 2* button to move the call made above to hybrid 2.
- Bring up hybrid 2 on the console. You will hear the audio from the phone line.
- Look at the left meter on the 2101 Studio Interface and verify that this audio is ***not*** present on the "*2*" LED bargraph. If this audio is present here you are not feeding hybrid 2 a mix-minus and you ***must*** fix this problem before you proceed.
- Drop the call by pushing the *Drop* button .

IMPORTANT!

Repeat the above steps with Hybrids 3 and 4, if present.

Ok, now we just need to test the system as whole. We suggest calling your spouse or partner and reporting that you *will* be home for dinner after all!

1. Confirm that the Desktop Director you will be using has power and says "*Talent*" at the top of the screen. If it says "*Producer*" follow the directions in Part V (The Desktop Director) section 2.1.2 to change this Director to "*Talent*" mode.
2. Choose a Show Configuration using the LCD screen. See Part V (The Desktop Directors) section 2.1.3 for details on selecting a show. If no show configurations appear check the LAN connections at this Studio Interface, 10/100 Base-T LAN hub and the 2101 Hub. Check the Telos 2101 hub is running and configured (see Part III).
3. Makes sure the handset is on hook and that the speaker phone button is not illuminated. Press the **left** *Line 1* button to select that line for dialing. You should now hear local dial tone through the console (mixing desk). Enter the phone number for the line you wish to call.
4. Bring up hybrid 1 on the console. You will hear the audio from the phone line.
5. Put on some headphones, bring up the microphone, and greet the party at the far end.
6. Your voice should sound normal in your headphones, without any tinny or hollow artifacts, and the caller should sound clear.
7. Move the call to Hybrid 2 by pressing the **right** *Line 1*.
8. Repeat steps 4 through 6 using hybrid 2.
9. Drop the call by pushing the *Drop* button .

What now?

Your basic installation is complete. Next you will want to look over other sections of this manual to further familiarize yourself with the 2101 Studio Interface and decide how to set up the following parameters:

- Desktop Director Operation (Talent mode) Vol. 3, Part V Section 4
- Desktop Director Operation (Producer Mode) Vol. 3, Part V Section 3
- Receive EQ, AGC & Noise Gate Section 3.3.1
- Duplex Level (caller ducking) Section 3.3.1
- Send EQ, AGC Section 3.3.1
- Internal Audio Routing Section 3.3.1
- Feedback Control (when using open speakers) Section 3.5.4
- Creating new Show Configurations Vol. 2, Part III
- Configuring Telco Trunks Vol. 2, Part III

3 Detailed Configuration & Reference

IMPORTANT!

Unless otherwise indicated, the following instructions apply to both the Series 2101 Studio Interface and the Telos TWO dual hybrid. Additional information on the Telos TWO is included in the manual shipped with that product.

After power-up and initialization, a status screen similar to the following appears…

Series 2101

V2.2

TELOS SYSTEMS yyyy-mm-dd

When you see this screen;

then the Studio Interface (or Extended Hybrid) has completed its boot cycle.

You may confirm that the T-Link (Studio Interface) or S interface (Extneded Hybrid) connection is ready with the Status screens. Press the *Status* button to display these.

A few words about the 2101 Studio Interface, Extended Hybrid (Telos TWO), and their hardware family

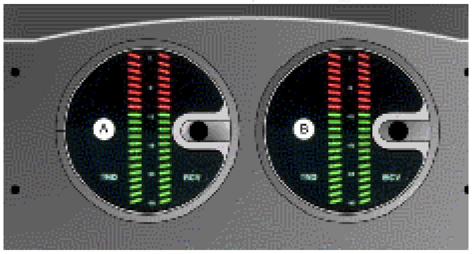
You will note a high degree of similarity between the 2101 Studio Interface, Telos TWO, and several other recently developed Telos Systems products. In order to keep your costs as low as possible, we have chosen to use many assemblies on more than one product.

Therefore, there are a few buttons on the front panel of the Studio Interface that are not used. The four buttons to the right side of the numeric keypad and the 3 buttons to its left fall in this category. These buttons are also not used in the Telos TWO when it is part of a Series 2101 system. The lowest LED to the left of the numeric keypad is used. When this LED is flashing it indicates that the system is writing to nonvolatile memory. You should not turn the unit off if this LED is blinking. When this LED is flashing you may notice some sluggishness of the front panel controls. This is normal and only occurs in the rare instance where the system is storing user information.

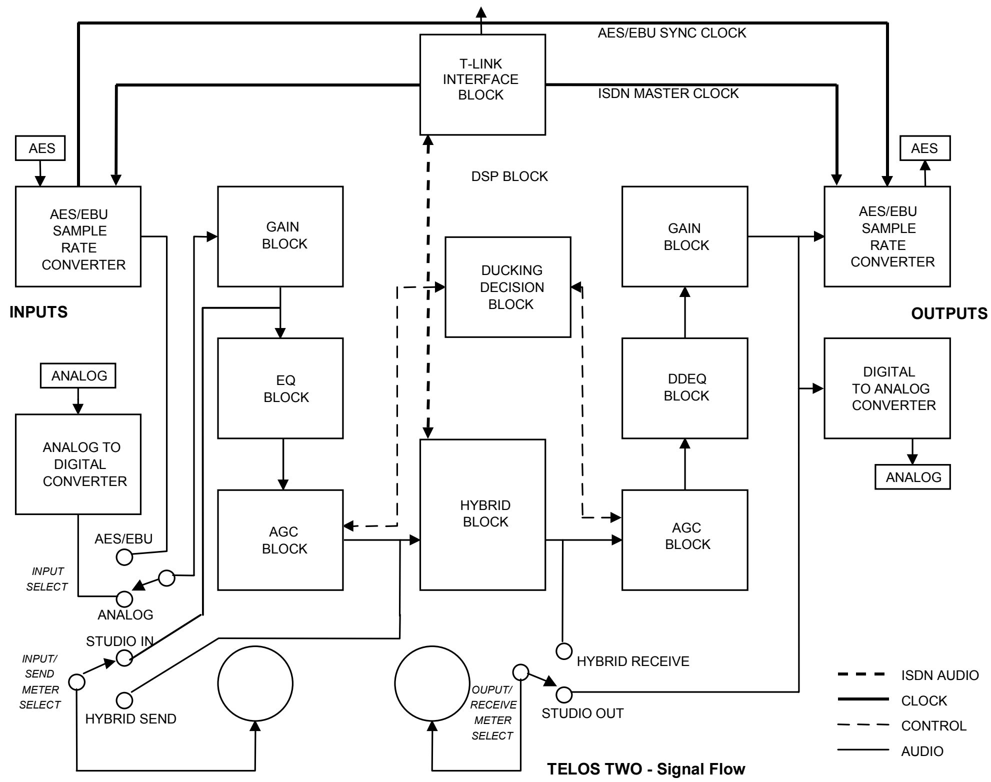
HOT TIP!

Note that while we suggest powering the system off and back on in case of problems, the system should be stable and this should not be required on a routine basis. If you experience frequent difficulties contact Technical Support for assistance, we are here to help.

3.1 Level Metering



The Telos 2101 Studio Interface and Telos TWO hybrid include two dual LED dual meters. As is customary with digital equipment, these are peak reading meters with the topmost segment representing 0dBfs (digital 0). The following block diagram indicates where the metering takes place. Note that all metering occurs in the digital domain. The metering is designed so that the highest peak reading persists for a few seconds to aid in perceiving short peaks. As with any digital system there is no level beyond 0dBfs and you must not attempt to exceed that level or severe distortion will result. Since voice tends to be very dynamic, plus the are huge variations in level due to varying operators, we suggest that the input level be configured such that the red LEDs rarely, illuminate.



Simplified Block diagram of the 2101 Studio Interface's integral Telos TWO dual hybrid. For the sake of simplicity only one audio channel is shown.

The leftmost meter can display either the *Studio Input* levels (*Send - to caller* 1 & 2) **from** your console or the *Hybrid Send* levels (Line 1 and Line 2) **to** the phone line. The *Hybrid Send* levels are active even if a line is not connected. Pressing the small button on the meter selects between *Studio In* and *Hybrid Send*. This level is controlled by the *Input Gain* and *Hybrid Send* Level selections in the *Audio* menu. See section 2.4 for step-by-step instructions and section 3.3.1 for more information on setting this level.

So what's the difference between *Studio In* level and *Hybrid send* level? Well, we're glad you asked. The *Studio In* level represents the raw digital level after the analog to digital converter (or the AES/EBU level after sample rate conversion) and gain adjust stage, while the *Hybrid Send* level represents the signal after the AGC/Limiter and EQ stages. The *Hybrid send* meter can be used to see how much limiting is occurring and fine tune your level. Once the *Studio In* level is set using the *Input Gain* adjustment (from the *Audio* menu) you can then view the *Hybrid Send* meter to see the levels after the AGC. This setting will allow you to see remaining headroom when the system is in operation. See the diagram above for details.

The right meter displays either the *Studio Out* levels (Left and Right) **from** the Studio Interface **to** your console or the *Hybrid Receive* levels (Line 1 and Line 2) **from** the telephone network. Pressing the small button on the meter selects between *Studio Out* and *Hybrid Receive*.

Once again, an explanation is in order. *Hybrid Receive* level represents the level of the caller *before* any receive AGC action. The *Studio Out* levels show the caller audio signal which is headed **to** your console, **after** the equalization, AGC, and gain adjustments have been made. See diagram above for details.

HOT TIP!

*You will expect to see a wide variation in levels on the **Hybrid Receive** meters, we've seen over 25dB variation between different calls on the same line. If you find large variations in caller levels when monitored at the **Studio Out** meter, you should increase the **Receive AGC** function slightly as follows:*

- *Press the **Audio** button*
- *Press the ▼ button seven times until the cursor is over the **Receive AGC** option and push the **Select** button*
- *Use the ▲ button to increase AGC activity (or the ▼ to decrease AGC activity) then push **Select** to confirm your choice*

The levels displayed are in decibels relative to the full-scale limit of the Telos 2101 Studio Interface (dBfs). Thus, 0dB represents the maximum level that can be passed through the system, above which clipping will occur.

The metering includes a peak level function. The LED segment indicating the peak level remains on for approximately one second, holding the peak value.

3.2 Status Displays

Pressing the *Status* button displays the various status screens. These status screens display information about the state of the Studio Interface and various user settings, but do not allow you to make any changes. The ▲ & ▼ buttons cycle through the screens.

Line Status

This screen shows the status of the T-Link channels.

3.3 Advanced: Setting the 2101 Studio Interface's or Extended Hybrid's Configuration

You use the various configuration settings to adjust audio levels, set the ISDN parameters, etc. The buttons located near the LCD display on the rack-mount chassis are used for this purpose. Note: the Desktop Directors each have menus which allows adjustment of the ringer and headset volume, LCD contrast, *Producer* vs *Talent* mode and other user settings. See Part V for details on the Desktop Director™ operation

Each of the four buttons below the LCD display calls up the menu item group associated with the button's name. The menu groups are:

- AUDIO
- TEL
- SYSTEM
- STATUS

Status, described above, is not really a menu, it presents screens for viewing only; there are no items that can be changed. See section 3.2 for details on this menu.

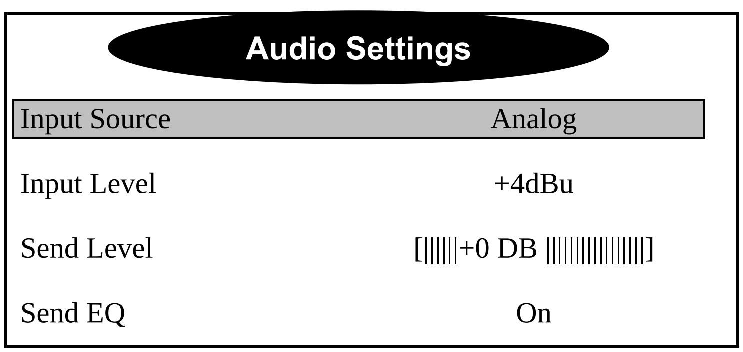
Menu Navigation & Item Selection

After you select a menu group, you can use the ▼ & ▲ arrow buttons to scroll through the available items. You may select a highlighted item for modification by pressing the *Select* button. Once an item is selected, you can use the ▼ & ▲ arrow buttons to can change the value of the setting.

When you reach the value you desire, press the *Select* button again to lock it in and return you to the item selection/scroll mode.

3.3.1 The Audio Menu

This group consists of sixteen settings that control the audio levels, mixing, etc.



Input Source

The selection chooses either *Analog* or *Digital* (AES/EBU) as the source of *Studio In* audio to be sent to callers.

Input Level (Nominal Level)

Adjusts the nominal level **from** the console sent **to** the built-in hybrids. This setting ranges from -7 to +8dB in 4 steps. This adjustment is before the *Studio Send* Input level meter. It is before the send AGC, but because that AGC is limiter-like in its operation, you will be able to change the average send level effectively.

Input Gain *Related to input Levels*

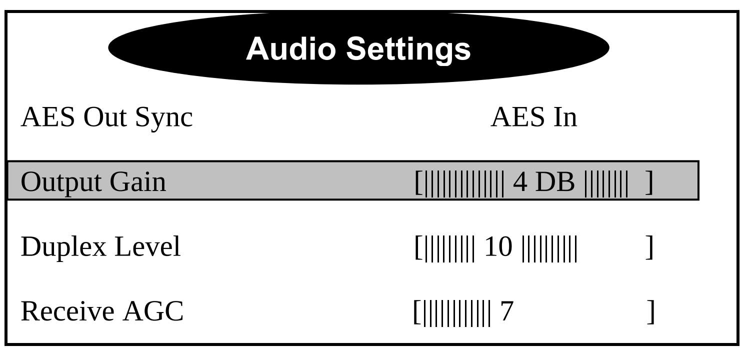
Input Gain Setting	-7	0	+4	+8
Analog Input Level (nom/clip) dBu	- 7/+6	0/+13	+4/+17	+8/+18
AES Input Level dBfs	- 27	-20	-16	-12

Send Level

This setting adjusts the level of the hybrid into the Telco line and is calibrated with 0dB representing the maximum allowable send level into the line (USA). This should normally be left at the default setting of 0. Reducing this setting can improve announcer audio quality by reducing leakage. If you are using analog (POTS) lines you may wish to set this somewhere in the range of –4 to –2 dB.

Send Eq

Turning on this equalizer simulates the audio response of a telephone handset microphone and will improve intelligibility. Sometime high-quality microphones sound "too good" and the low-frequency audio the pick-up can sound unnatural on a telephone handset. Section 3.5.1 has additional information on this function.



AES Out Sync

This item selects the synchronization source for the AES/EBU Output. It can take one of two values.

- 48 kHz

  AES/EBU Output rate is locked to a precision 48kHz clock derived from the ISDN line.

- AES IN

  AES Output is synchronised to the clock frequency of the AES/EBU source connected to the AES/EBU input. Even if you are using the analog inputs the AES/EBU input can be fed a reference signal to provide output synchronisation.

Output Gain

This item adjusts the gain to both the Studio Analog and AES/EBU outputs. The value is in dB relative to +4 dBu on the analog outputs and –16 dBfs (dB relative to full scale) on the AES output. The default value is 0 dB.

The system allows for 13dB before clipping, so the clipping point can be determined by adding 13 to the readings below. The following table shows the relationship between the *Output Gain* setting and the output levels.

Output Gain *Related to Output Levels:*

Output Gain Setting	-11	-4	0	+4
Analog Output Level (nom/clip) dBu	- 7/+6	0/+13	+4/+17	+8/+21
AES Output Level dBfs	- 27	-20	-16	-12

Duplex Level

This control adjusts the amount of ducking that occurs to the received telephone audio based on the level of send-to-caller audio. A value of 16 means that the telephone audio is not ducked (full duplex) or attenuated, while 0 applies full attenuation when send audio is present (half duplex). Settings in the range of 10-12 are typical. Section 3.5.1 has more information on this function. This function is desirable for a variety of reasons:

- Allows the announcer to "override" the caller by causing the received caller audio to be attenuated (ducked) when the announcer speaks. This is often desired for aesthetic effect and allows the announcer to remain in control of the conversation.
- Permits open loudspeaker monitoring of callers with reduced feedback or problems
- Dynamically improves "apparent" trans-hybrid loss to reduce send to receive leakage when necessary.

Receive AGC

The *Receive AGC* section serves to improve level consistency of incoming caller audio. Nominal levels of the telephone network vary as much as 30dB from call to call. The Telos 2101 Studio Interface uses a dB-linear approach to AGC with a feed-forward topology. This provides a consistent sound regardless of the drive level. The smart compressor approach used normalizes levels while retaining the natural dynamics of the caller's voice.

This selection controls the "aggressiveness" of the AGC process, taking values from 0 (off) to 15 (max). Values of above 10 are quite aggressive, so we strongly suggest you start off with a value below 9.

More aggressive AGC is more audible, but is also more effective at maintaining consistent levels. Adjustment of this setting simultaneously changes as number of parameters within the AGC function, such as attack & release times, thresholds, and compression ratio.

An important feature of the 2101 Studio Interface's AGC is the fact that it is cross coupled to other sections of the hybrid so that it can reliably distinguish between the real caller audio and residual hybrid leakage. This allows for considerable amounts of gain on low level callers while preserving excellent hybrid performance.

Audio Settings

Receive EQ — Adaptive
Rcv EQ Low — [||||||+0 DB |||||||||||||||||]
Rcv EQ High — [||||||+0 DB |||||||||||||||||]
Noise Gate — Normal

Receive EQ (Dynamic Digital EQ – DDEQ)

Telephone audio frequency response varies widely as many factors can effect it (we've measured the response on a number of calls and the results were revealing). Consequently, some form of receive equalization is desirable. See section 3.5.2 for additional information. The *Receive EQ* selection controls the type of equalization applied to the received telephone audio as follows:

- *Off* - The caller audio is passed through without modification. The *Rcv EQ* values (see below) are ignored in this mode.
- *Fixed* - This is a simple manual equalizer mode where EQ values selected for LF and HF are applied constantly.
- *Adaptive* (Dynamic Digital) - This is a three band dynamic equalizer. The proper values of high and low frequency boost or cut is automatically determined and set. The *Rcv EQ* values (see below) are ignored in this mode.
- *Fix + Adap* - The Dynamic EQ uses the *Rcv EQ* values (see below) as "target levels". Otherwise, the functioning is the same as *Adaptive*.

Rcv EQ Low

Amount of low frequency boost or cut applied to the *Hybrid Rcv* audio. Works in conjunction with the *Receive EQ* selection, see above. Settings from –8dB to +8dB are possible with 0dB representing flat response.

Rcv EQ High

Amount of high frequency boost or cut applied to the *Hybrid Rcv* audio. Works in conjunction with the *Receive EQ* selection, see above. Settings from –8dB to +8dB are possible with 0dB representing flat response.

Noise Gate (Expander)

Enables or disables a noise gate applied to the caller audio. This has the effect of reducing the level of noise on the caller audio when it falls below a threshold. On some very low-level calls, this attenuation may be inappropriate because you may be able to hear the caller fade away when the level falls below the threshold. It has three settings as follows:

- *Off* – Disables the noise gate
- *Normal* – A slow setting, with moderate noise attenuation
- *Aggressive* – A faster setting with deeper attenuation

Audio Settings

Routing   Separate

Feed Thru 1   Off

Feed Thru 2   Off

Hybrid Mix   [|||||||||||||||-12 DB]

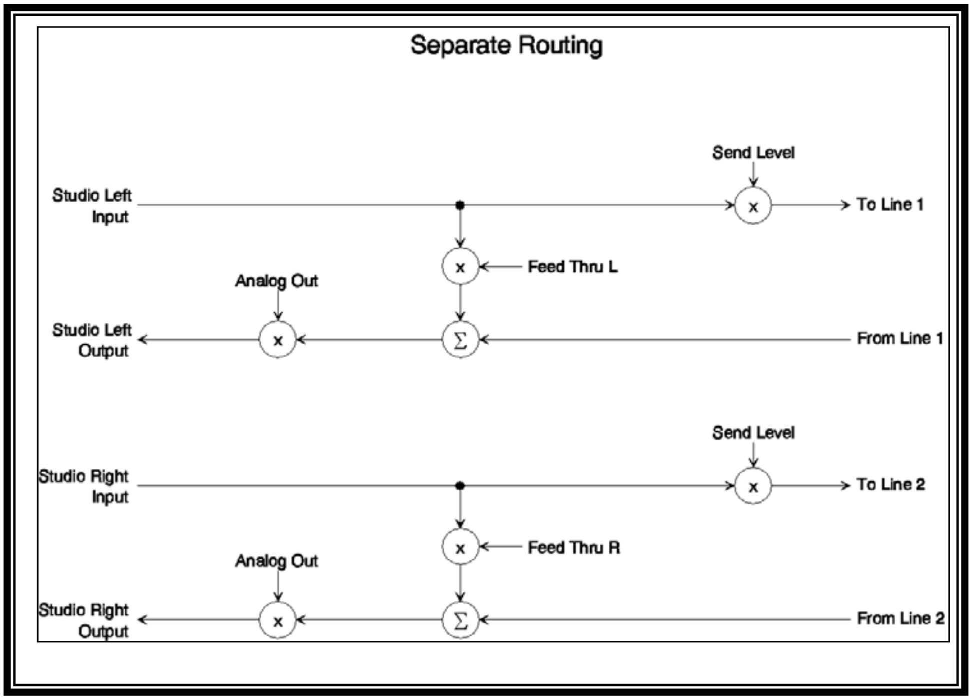
Routing

This item controls the internal mix options for the audio between the studio and the caller. This feature is powerful as it can feed one mix-minus to both hybrid sections (thereby conferencing them together), or mix the receive from both hybrids, or create a stereo panned mix of the two hybrids receive audio. This option is important as it determines how many mix-minus feeds you will require (see section 1.4.1 for more on mix-minus) and the number of input channels required on the audio console (mixing desk).

The most popular settings for this selection are *Separate* (where the 2101 Studio Interface's dual hybrids are completely independent) and *Mix Wide* (see page 61). Note that the *Feed Thru* audio path, shown on the following diagrams, is controlled independently of the routing option chosen.

- Separate Routing

  This option provides two fully independent hybrids, with the Left Studio Audio "connected" to Hybrid 1 and the Right Studio Audio to Hybrid 2. In this mode, the two callers cannot hear each other, except through your external mix-minus configuration. The diagram below illustrates this configuration:

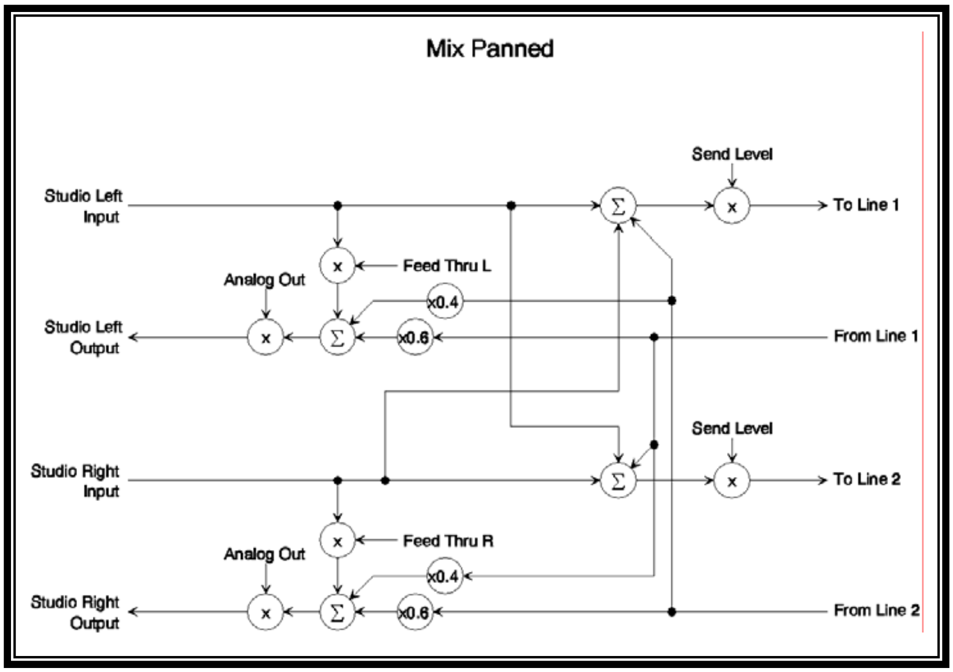


Separate routing treats hybrid 1 and hybrid 2 as completely independent hybrids. The only correlation between the two are the input and output gain levels, and certain other options that control both hybrids. The audio paths, however, are completely isolated from each other.

The *separate* routing choice requires that 2 mix-minuses be provided and requires 2 separate faders on the console (mixing desk).

- Mix Panned

  This option causes the to *Studio Input* channels to be mixed equally and the resulting monaural send mix sent to each caller. The caller audio, however, is not mixed equally to the studio outputs. *Studio Out 1* has a higher level of line 1 caller audio than line 2 caller audio; approximately 1.4 times or +3 dB. The converse is true for *Studio Output 2*. The two callers will be able to hear one another through the hybrid. The *hybrid mix* control in the *Audio* menu determines the level of this cross-feed path. See diagram below:



*The **Mix Panned** gives a stereo output with the two callers "panned" to either side. Each caller hears a mix of the two studio audio inputs plus any caller on the other hybrid.*

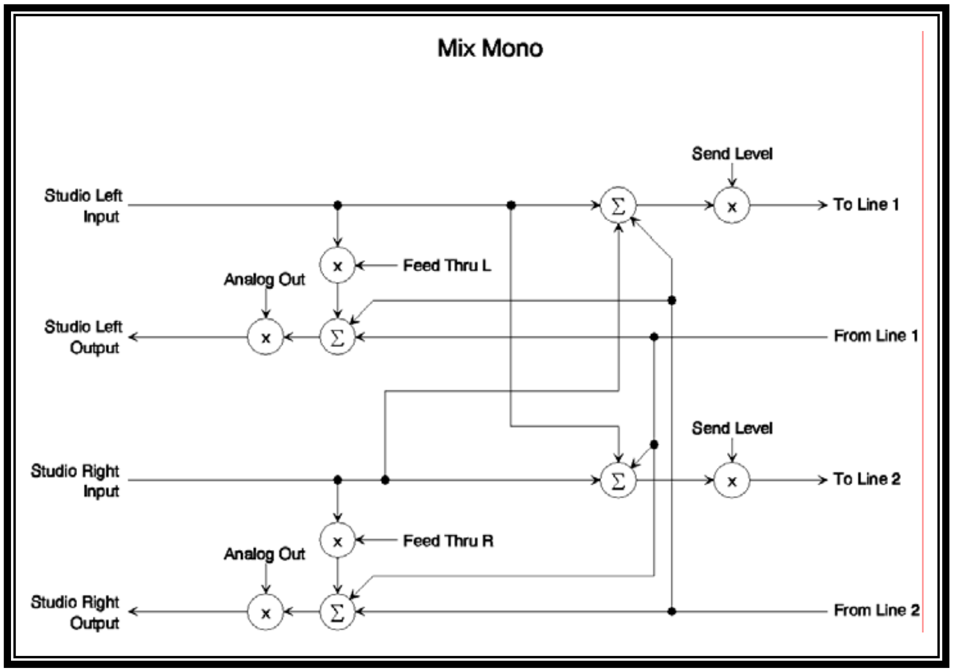
This configuration requires only a single mix-minus (with neither hybrid's audio present). This mix-minus should be fed to both *studio in* inputs. It requires a single stereo fader on the console (mixing desk). The only significant drawback to this arrangement is that the only way to control the relative gain of the two callers is to adjust the balance control on the console input module, if present.

IMPORTANT TIP!

*When using any of the **MIX** routing options it is important to feed a signal to both inputs, even if this is the same signal to both. Failure to do so will cause significantly reduced send- to- caller levels.*

- Mix Mono

  This routing option causes the two *Studio Send* Input channels to be mixed equally, and the resulting monaural send audio is sent to each of the two callers. The caller audio is mixed equally and output on each of the Studio Outputs. The two callers will be able to hear one another through the hybrid. The *hybrid mix* control in the *Audio* menu determines the level of this cross-feed path. See below.

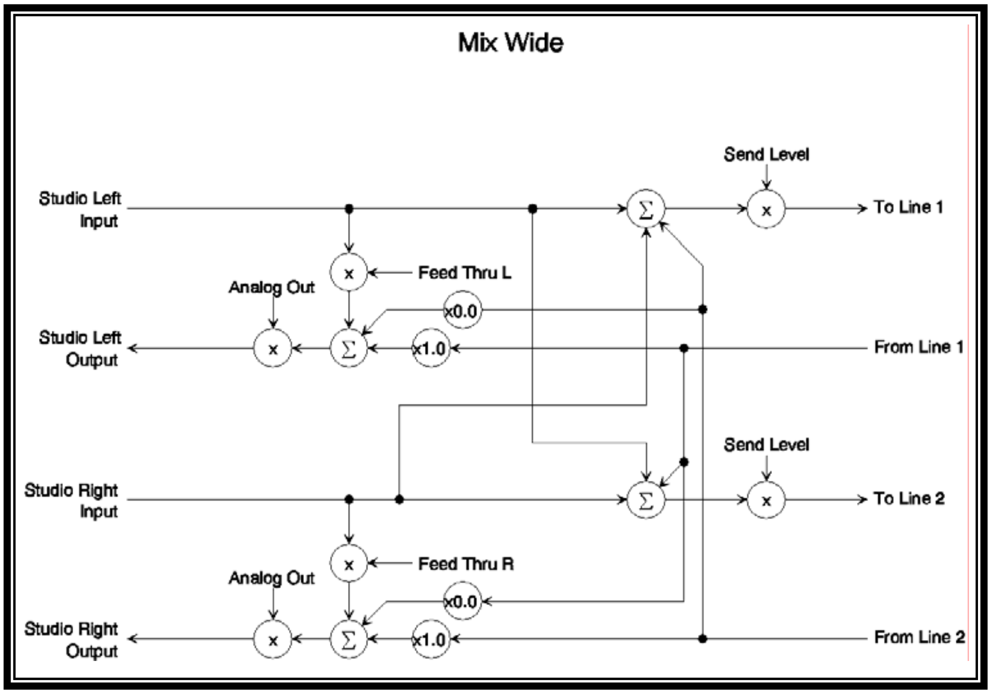


*The **Mix Mono** routing option provides identical audio, with both callers, on both studio outputs. Each caller hears a mix of the two studio audio inputs plus the other caller.*

This option requires a single mix-minus fed to both inputs (with neither hybrid's audio present) and requires only a single mono fader on the console. The disadvantage here is that the operator has no control of the relative levels of the two callers and must depend on the AGC to keep these relative levels equal.

- Mix Wide.

  As shown below, this audio routing option sends the same monaural audio to each caller as with Mix Mono and Mix Panned. The difference is that the Left Studio Output consists of only hybrid 1 received audio and the Right Studio Output only hybrid 2 received audio. The two callers will be able to hear one another through the hybrid. The *hybrid mix* control in the *Audio* menu determines the level of this cross-feed path.



*The **Mix Wide** routing option, gives independent caller audio outputs. Two independent caller outputs allow for maximum mixing control. Each caller hears a mix of the two studio audio inputs plus the other caller.*

This option requires only a single mix-minus (with neither hybrid's audio present) but allows for 2 two mono faders on the console (one for each caller). **This is the most popular option when only one mix-minus is available**, as it allows independent control of the two caller levels at the console.

MIX- MINUS TIP!

Recall that in section 1.4.1 we discussed a simple mix- minus consisting of a feed directly off the mic preamp. In that chapter we mentioned the drawback to that approach was the need for external equipment if more than one mic were used. If you have only 2 mics you can use that simple approach by feeding the audio from each mic preamp into one of the ***Send (to caller)*** *inputs and using the* ***Mix Mono****,* ***Mix Wide****, or* ***Mix Panned*** *routing option on the 2101 Studio Interface. Since the two inputs are mixed within the Studio Interface with these settings, each caller will hear both mics.*

Feedthru 1 & 2

For special applications. Should be set to *Off* at all times.

Hybrid Mix

This option allows adjustment of the amount of "cross-feed" of one hybrid's output to the other. The range is from –12 to +8. 0 represents no gain or loss between callers and is the usual setting. If you have problems with callers unable to hear each other you should increase this setting slightly. The setting of this control makes no difference if the *Routing* is set to *Separate,* since that routing option does not cross feed the caller audio.

Audio Settings

Feed Thru 2	Off
Hybrid Mix	[|||||||||||||-12 DB]
Studio Adapt	Off
Adapt Burst	On

Studio Adapt

This selection enables (*on*) or disables (*off*) the acoustic adaption process used for feedback control when the system is used with open speakers. This should be set to *ON*, whenever open speakers are employed with microphones. Default is *OFF*.

HOT TIP!

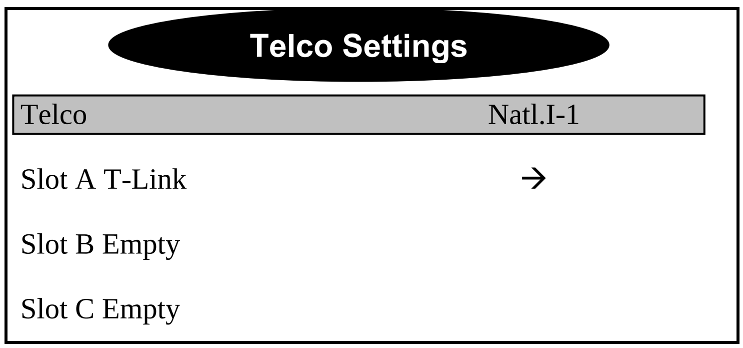
When using open speakers it is best not to change the speaker or mic positioning while the system is operating as this will require the system to readapt and temporary feedback may occur while this is happening.

Adapt Burst

Turns the adaption tone sent to the caller *Off* or *On*. Should normally be set to *On*. Turning this selection *off* will prevent the tone from being sent. Under those circumstances the hybrid sections will adapt only when send – to-caller audio is present. If this option is *off* the announcer audio during the first few syllables of send - to caller audio may sound "hollow" or "tinny" as adaption takes place.

3.3.2 The Tel Menu

This menu allows selection of a Telco setting which determines compatibility with the telephone network in your country. Note that the items for Slot A, Slot B, Slot C, Slot D will say "T-Link", or "EMPTY" depending on how your system is configured.



Telco

This item allows for the selection of the ISDN protocol standard and takes one of four values:

- ETS 300 (A-Law)
- Natl I-1 (u-Law)
- DMS Cust (u-Law)
- 5ESS Cust (u-Law)

The correct setting will depend on where you are located:

IMPORTANT!

We've collected information on the speech coding modes for as many countries as possible. Please understand that inclusion on the following list does not imply that the Series 2101 has been tested or approved for use in any particular country on the list. Your Telos representative or Sales Engineer is your best source of information regarding tested compatibility in various countries.

Location	Speech coding type ("Telco" setting)
Americas Brazil Costa Rica Mexico	A-Law (Set to ETS 300)
Americas Canada Dominican Republic Puerto Rico United States of America Venezuela	u-Law (Set to any option but ETS 300)
Europe Belgium Czechoslovakia France Germany Greece Hungary Italy Netherlands Russia Spain Switzerland United Kingdom	A-Law (Set to ETS 300)
Middle East Egypt * Saudi Arabia	A-Law (Set to ETS 300) * Except for Cairo which uses µ-Law

Pacific Rim Australia China India Indonesia New Zealand Philippines Singapore Thailand	A-Law (Set to ETS 300)
Pacific Rim Hong Kong Japan Korea Taiwan	u-Law (Set to any option but ETS 300)

Slot A, Slot B, Slot C, Slot D (Series 2101 Studio Interface only)

Your T-Link cards are in slots A & B. A card will only be present in slot B only if you have added an optional second card. When the system is turned on it scans the slots and the type of card is shown after the slot letter in this screen. For instance: Slot A T-Link. The other settings in this menu may be disregarded and should be left blank.

3.3.3 The System Menu SYSTEM

This menu consists of sixteen items that control basic system settings.

System Settings

TCP/IP Setup	→
Software Update	→
Contrast	[||||||||||||| 12 ||||||||||]
Backlight	[|||||||||||||||| 20 |||||||||||||]

TCP/IP Setup (sub-menu)

Selecting this option displays the *TCP/IP Settings* sub-menu with the following options. Selecting ↲ will return you to the *System Settings* menu.

TCP/IP Settings	
↲	
IP Addr	0.0.0.0
Hostname	
Subnet	0.0.0.0

IP Addr

This is the IP address for this Studio Interface. As with any computer on an IP network the each Studio Interface must have an IP address before it can be used over the network. Since this address must be unique, you'll have to ask whoever manages your network to give you this number.

Actually, an IP address is a series of 4 numbers (each number being between 0 and 255) separated by periods. This is a numeric-only field; enter the numbers using the dial pad. Both # and * will give you a period. As usual, the ▲ key will backspace if you make a mistake or need to delete an old entry.

Hostname (Studio Interface Only)

You must assign a unique name to each Studio Interface. Allowable characters are a-z, A-Z, and 0-9. Do not use spaces or special characters when creating Hostnames. This name will be used when selecting a *Show Configuration* on the Desktop Directors. Keep the name short and simple.

A word on entering alphanumeric characters

To enter text characters, press the key with the appropriate letter printed on it. Keep tapping it to cycle through the possible choices. For example, tapping the 2-key in a text field will first enter A. Repeated taps will change that to a B, then C, then the lower-case letters, then the number 2. Enter a space by pressing 1 three times. Punctuation characters appear on the * and # keys. Pressing # button 2 times will give you the "." character. To move to the next character in a field, tap any other key or use the ▼ button. To back up use the ▲ button. As usual the *Select* button confirms entry.

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. Therefore the same subnet should be used for the Series 2101 Hub and all Studio Interfaces.

Just as with the IP address, the Subnet mask is a series of 4 numbers separated by periods. Enter the numbers using the dial pad. Both # and * will give you a period. This is a numeric-only field; enter the digits as described above. As usual, the ⏶ key will backspace if you make a mistake or need to delete an old entry.

Gateway

This is the IP address of a gateway router connecting you to the internet. It is an IP address just as with the previous two items and is entered in the same way. You will need to get this from your network administrator or internet service provider. This is a numeric-only field; enter the digits as described above.

DNS IP

This is the IP address of the DNS (Domain Name Server) you will be using. It is an IP address just as with the previous items and is entered in the same way. You will need to get this from your network administrator or internet service provider.

Remote Ctrl (This command reserved for future use)

This is a security feature and determines whether control via the RS-232 port, Telnet, or neither is allowed.

Trusted (This command reserved for future use)

Software Update (sub-menu)

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.

Selecting this option displays the *Update Software* sub-menu with the following options. Selecting the ↵ will return you to the *System* menu.

Update Software	
FTPsite	ftp.zephyr.com
Boot Bank	Primary
Update software…	
Commit Software	

FTPsite

This is the site from which new software will be downloaded. Normally the default value of "ftp.zephyr.com.com"should be used unless you are instructed to do otherwise by Telos technical support. This can be either an IP address or (if a DNS IP address has been entered) any alphanumeric internet domain. See above for information about IP address and about entering alphabetic information.

Boot Bank

The 2101 Studio Interface has two independent banks of flash memory for software storage. This menu selection determines which bank (and hence which software version) will be loaded at startup. The *Primary* bank is the default choice and is the only bank with software when the unit is shipped.

When a new software version is downloaded it is written into the *Secondary* bank. After a successful download this selection will be changed to *Secondary* and the new software will be booted the next time power is cycle off and on. However, you may change the selection back to Primary should you determine that you prefer the original software. This allows you to compare the two versions and determine which version better meets your needs.

Update software…

This menu brings up a query window which asks: "*Are you sure you want to download new software into this unit?.*" If you do not wish to do so you should press *Select* or any of the Menu keys to abort. Since the Studio Interface works in conjunction with software running on the Series 2101 Hub, we do not recommend that you update the software without consulting Telos Customer Support first (we'll need the version of the software running on your 2101 Hub as well as the current version of software running on the Studio Interfaces). If you do wish to download new software press ▲ or ▼ to highlight *Yes* and then press *Select* to begin the download process. You will see the message "*Please wait while new software is being downloaded and burned into the internal SIMM memory…*" Once the download is complete the unit will reboot and the new software will be running. If the download is unsuccessful the system will display an error message. Please write down the complete error message, check your network connections and TCP/IP setup options (from the *System/TCP/IP Setup* menu). You may wish to check with your network administrator and make sure the IP address entered is not being used by some other device on your network. If the problem persists, contact Telos Technical support.

HOT TIP!

If you get the error ***Cannot resolve host name*** *the problem could be anything from your IP settings to your network cable. If the green LED on the 10Base- T connector is not lit then you are probably not connected to a working network hub.*

Commit Software…

This menu brings up a query window which states: "*Are you sure you want to commit the secondary software for permanent usage?*" If you do not wish to do so you should press *Select* or any of the Menu keys to abort. If you do wish to do make the new software your *Primary* software press ▲ or ▼ to highlight "yes" and then press *Select* to begin the process.

The software version in the *Primary bank* will be deleted in this process.

In order to prevent you from making software from a faulty download your *Primary* software you can only perform this operation when running the *Secondary* software version. If the secondary software will not execute for some reason, this protects you from deleting your primary bank software unless the downloaded software is capable of successfully operating.

HOT TIP!

If you get the error ***Cannot perform operation from this bank*** *you must change the* ***Boot Bank*** *selection in the* ***Software Update*** *sub menu off the* ***System*** *menu to* ***Secondary*** *and reboot the system to test and evaluate the new software before you are permitted to commit to it.*

Reboot System

This command is intended to save you a trip to the back of the rack. Choosing this selection reboots the system just as would cycling power off and then back on again.

The complete software process typically consists of these steps:

1. Download software and boot into secondary bank.

2. Execute secondary software for some evaluation period of time. During that time no new software can be downloaded.

3a. If the new software does not meet your needs (or you wish to run the old software for comparison) you can go back to the old version. Do this by changing the *Boot Bank* selection in the *Software Update* sub menu off the *System* menu to *Primary*, and reboot to go back to the old software.

3b. If the new software is ok make it the permanent one by "committing" it to the primary bank. After that new software can be downloaded in the future.

4. If the new software did not meet your needs, you can download another version once you have set your "Boot Bank" menu back to "primary" without first “committing” the version previously downloaded.

Contrast

Adjusts the contrast of the LCD. It takes values between 0 and 20. The usual setting for most situations is from 14 to 17.

<table>
<tr><th colspan="2">System Settings</th></tr>
<tr><td>Backlight</td><td>[|||||||||||||||||| 20 |||||||||||||||]</td></tr>
<tr><td>Click Volume</td><td>[||||||||| 5]</td></tr>
<tr><td>Serial Bitrate</td><td>19200</td></tr>
<tr><td>Loop Mode</td><td>Off</td></tr>
</table>

Backlight

Adjusts the brightness of the backlight for the LCD screen. The value 0 is dimmest and 20 is brightest. The usual setting for most situations is from 12 to 20.

Click Volume

Adjusts the volume of the audible “click” generated when a front panel button is pressed. The value 0 is off and 20 is loudest (quite loud, in fact). We prefer settings from 0 to 5.

Serial Bitrate

Adjusts the bit rate of the RS-232 serial interface. It accepts values between 2400 baud to 57600 bits per second.

Loop Mode

This item allows the 2101 Studio Interface to loop audio from various points in the system, primarily for testing purposes. Section 8.3 has more information on troubleshooting.

- Off

  This is the normal operating mode. No loopback path is 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 after Analog-to-Digital conversion) back to the AES/EBU output (and analog output via the Digital-to-Analog converter). This can be used to proof the audio input and output sections.

3.4 External Control

Certain functions may be accomplished by each of the following methods:

- The 2101 Studio Interface's Parallel Control Inputs and Outputs. See also section 1.6.1.
- RS-232 serial connection can be used to program the system and monitor various functions. See also section 1.6.3.
- An Ethernet connection:

  A Telnet session can access the same functions available over the RS-232 port

  Client software (such as our Assistant Producer for 2101 call screening system) can connect to IP Port socket 5001 (decimal). A Windows API is available for programmers wishing to program to this interface. It can be downloaded from www.telos-systems.com. See also section 1.6.4.

3.4.1 RS-232 Control

Section 1.6.3 has details on the physical interface. Since the 2101 Studio Interface is intended for use only with the Telos Series 2101 Hub, there are no user commands that should be accessed through the RS-232 or Telnet commands. The following connection information is for the rare case where Telos Customer Support may require you to do so.

First connect the Studio Interface to a computer's serial port. Since the Studio Interface has a DCE interface you will need to use a straight through cable.

You will be using a computer "Terminal Emulation" program such as Windows HyperTerminal or ProComm for IBM compatibles or White Night or Z-term for the Mac. Configure this software to 8 bits, no parity, and 1 stop bit. Set the bit rate (may be called baud rate) to match the *Serial Bitrate* selection in 2101 Studio Interface's *System* menu.

Since HyperTerminal comes with Windows, we will give you some tips on using this particular software:

Start up HyperTerminal.

Click on the "new" button on the tool bar. You may see a prompt asking if it is ok to continue, click "ok". You should now see a **Connection Description** dialog box requesting that you name the setup and select an icon. Enter the desired name such as "Studio Interface on com 2" and click on an icon (we like the electron symbol which is last). Now click on "ok".

You will now see a **Phone Number** Dialog box. Enter the following information:

- Country code – Setting does not matter
- Area Code – Setting does not matter
- Phone Number – setting does not matter
- Connect Using - select the com port connected to the 2101 Studio Interface

Now click on "ok". Next you will see a **Com Properties** Dialog box. Enter the following information:

- Bits Per Second - set to match *Serial Bitrate* setting in the 2101 Studio Interface's *System* menu selection.
- Data Bits - set to 8
- Parity – set to none
- Stop Bits – set to "1"
- Flow control - set to "none"

Now click on "ok".

Once you have made the physical connections and programmed your terminal emulation software, you should see a blank terminal window. If all is ok you should see your own characters echoed back when you type. If not, check the settings and make sure you are properly connected. You may need to restart HyperTerminal if you have changed serial port setting. You may need to contact your software vendor for advice.

IMPORTANT TIP!

Many of the commands available in the following section are very powerful and can interfere with normal operation of the Studio Interface if used incorrectly. Please note that you should avoid using a command that you do not understand. Note also that the most complete source of documentation for these commands is the online help, accessed as shown below. Telos Customer Support will not be able to assist you beyond what is shown in that help file.

Serial Port Commands

Note that in the following section we will use **bold type like this** to indicate what your would type and `Courier type like this to show what the system would reply.` If two parameters are separated by a "/" character this represents that you should choose one parameter form the list. As usual ignore quotes ("or') when typing entries.

You should now be able to get the most up-to-date listing of commands by typing "help ?" as follows:

>> **help ? <return>**

```
?

Available commands:

Codec Dump R4 R2 R1 RW4 RW2 RW1 W4 W2 W1 DTMF DSPrw DSPw DSPr
AudSet AudSource A

udInGain AudSendEQ AudAESsync AudOutGain AudDplx AudAGC AudReq
AudReqLo AudReqHi

 AudNoise AudRoute AudFeed AudHybMix AudStudAdapt AudBurst
NVclear NVdump FTPboo
```

```
t FTPcommit FTPupdate FTPpath FTPsite IPdns IPgw IPmask IPaddr
IPid IPset APinfo

 DDinfo LnMSN LnDN LnSPID LnTelco LnSet LnStudio LnStat
LnState LnCIDset TaskInf

o Version Log help ? Reboot


Type '? <command>' to get specific help.

Type '? *' to get detailed list of commands.

'#' starts a decimal number, '%' a binary number, '$' a hex
number.

Current number base is 16. Line editing commands are:

  ^R repeats last command, ^L & ^K recall previous commands.

  ^S and ^D move cursor one character left or right.

  ^A and ^F move cursor one word left or right.

  ^X deletes character, ^T word under cursor.

  ^P reprints the current command string.

>>
```

For the most up-to-date detailed list of commands type "?" followed by the name of a command. For detailed information on all commands type "? *" <return> as follows:

>>**? * <return>**

? *

Name.......: **AudSet**

Parameter..: **AudSet**

Description: Display current audio settings

Name.......: **AudSource**

Parameter..: **AudSource** [**A**/**D**]

Description: Set or display audio input source

Name.......: **AudInGain**

Parameter..: **AudInGain** [**-7**/**0**/**4**/**8**]

Description: Set or read audio input gain

Name.......: **AudSendEQ**

Parameter..: **AudSendEQ** [**0**/**1**]

Description: Display send EQ settings or turn [off (0)/on (1)]

Name.......: **AudAESsync**

Parameter..: **AudAESsync** [**I**/**E**]

Description: Set AES output synchronization to [Internal/External]

Name.......: **AudOutGain**

Parameter..: **AudOutGain** [**value**]

Description: Set Audio output gain to [value]

Name.......: **AudDplx**

Parameter..: **AudDplx** [**level**]

Description: Display duplex level or set to [level]

Name.......: **AudAGC**

Parameter..: **AudAGC** [**level**]

Description: Display receive AGC or set to [level]

Name.......: **AudReq**

Parameter..: **AudReq** [**setting**]

Description: Display or set value of Receive EQ

Name.......: **AudReqLo**

Parameter..: **AudReqLo** [**value**]

Description: Display ReceiveEQ low or set to [value]

Name.......: **AudReqHi**

Parameter..: **AudReqHi** [**value**]

Description: Display Receive EQ High or set to [value]

Name.......: **AudNoise**

Parameter..: **AudNoise** [**0**/**1**]

Description: Display Noise Gating setting or change to [0 (off)/1 (on)]

Name.......: **AudRoute**

Parameter..: **AudRoute** [**value**]

Description: Display Routing or set to [value]

Name.......: **AudStudAdapt**

Parameter..: **AudStudAdapt** [**value**]

Description: Display current Studio Adapt setting or set to [value]

Name.......: **AudBurst**

Parameter..: **AudBurst** [**0**/1]

Description: Display Adapt Burst setting or change to [0 (off)/1(on)]

Name.......: **NVclear**

Parameter..: **NVclear**

Description: Delete contents of nonvolatile RAM

Name.......: **NVdump**

Parameter..: **NVdump**

Description: Display raw contents of nonvolatile RAM

Name.......: **FTPboot**

Parameter..: **FTPboot** [**bank**]

Description: Display or change boot bank

Name.......: **FTPcommit**

Parameter..: **FTPcommit**

Description: Commit new software to primary bank

Name.......: **FTPupdate**

Parameter..: **FTPupdate**

Description: Initiate remote software update

Name.......: **FTPpath**

Parameter..: **FTPpath** [**path**]

Description: Display or change FTP path

Name.......: **FTPsite**

Parameter..: **FTPsite** [**site**]

Description: Display or change FTP site

Name.......: **IPdns**

Parameter..: **IPdns** [**DNS**]

Description: Display or change DNS server

Name.......: **IPgw**

Parameter..: **IPgw** [**gateway**]

Description: Display or set gateway

Name.......: **IPmask**

Parameter..: **IPmask** [**mask**]

Description: Display or change subnet mask

Name.......: **IPaddr**

Parameter..: **IPaddr** [**address**]

Description: Display or set IP address

Name.......: **IPid**

Parameter..: **IPid** [**MAC**]

Description: Display or set MAC

Name.......: **IPset**

Parameter..: **IPset**

Description: Display current TCP/IP settings

Name.......: **APinfo**

Parameter..: **APinfo**

Description: Display information on Assistant Producer connections

Name.......: **DDinfo**

Parameter..: **DDinfo** [**director**]

Description: Display information on all or a given director

Name.......: **LnTelco**

Parameter..: **LnTelco** [switch]

Description: Display current switch type and options or set switch type

Name.......: **LnSet**

Parameter..: **LnSet**

Description: Display current line settings by card

Name.......: **LnStudio**

Parameter..: **LnStudio**

Description: Display Studio Status

Name.......: **LnStat**

Parameter..: **LnStat**

Description: Display current line status by card

Name.......: **LnState**

Parameter..: **LnState**

Description: Display internal line states

Name.......: **TaskInfo**

Parameter..: **TaskInfo** [**task**]

Description: Display information on all tasks [or a specific task]

Name.......: **Version**

Parameter..: **Version**

Description: Display system version information

Name.......: **Log**

Parameter..: **Log** [**source**]

Description: Display logging options or toggle [source]

Allowable sources:

0 : All Errors	1 : Initialization	2 :Phone Line Msg
3 : Phone Line Ctrl	4 : Phone Line Remote	5 : Phone Ln Net Req
6 : Main ISDN	7 : ISDN Layer 2	8 : ISDN Layer 3
9 : ISDN IE	10 : ISDN stack	11 : POTS hardware
12 : Caller ID Parse	13 : Caller ID Dump	14 : Socket Control
15 : GUIThreadIface	16 : DD LCD Control	17 : DD Proxy
18 : DD Proxy Helper	19 : DD to Studio	20 : HDLC Msg Creator
21 : HDLC Msg Parser	22 : HDLC Buffer	23 : NonVolatile
24 : Storage Manager	25 : Storage Man Proxy	26 : StorMan Pxy Helper
27 : Application	28 : App Messages	29 : App Threads
30 : Services	31 : Service Thread	32 : App Advertise
33 : Advertise Send	34 : Rem. Obj. Updater	35 : Global Directory
36 : Net Objects	37 : Hybrid	38 : Audio Setting
39 : Audio Connection	40 : Remote Audio Conn	41 : Remote Audio Ctrl
42 : Remote Audio Proxy	43 : Hub Link Traffic	44 : Link Timing Info
45 : Line Man. Local	46 : Line Man. Group	47 : Line Man. Remote
48 : Line Man Net Req	49 : Studio	50 : Studio Proxy
51 : Studio Net Req	52 : Studio Line Ctrl	53 : RUDP7 layer
54 : HDnt logs		

Type 'log -1' to disable all logging.
Type 'log' to display a list of possible log sources.

Note that multiple log modes can be enabled simultaneously.

Name.......: **help**

Parameter..: **help**

Description: List available commands

Name.......: **?**

Parameter..: **?**

Description: List available commands

Name.......: **Reboot**

Parameter..: **Reboot**

Description: Restart this Studio Interface

3.4.2 Ethernet 10Base-T control

A Telnet session can access the same functions available over the RS-232 port (see section 3.4.1 for a list of commands available). Simply use your favorite Telnet client to Telnet to the IP address of the 2101 Studio Interface you wish to control.

Or client software (such as our *Assistant Producer for 2101* call screening software) can connect to IP Port socket 5001 (decimal). See also section 1.6.4.

3.5 System Audio Processing and Feedback Control

3.5.1 Send Audio Processing

The Studio Interface's send audio processing consists of the following functions:

- Sample rate conversion
- High-pass filter
- Frequency shifter
- Send equalizer
- Send AGC/Limiter

Sample Rate Conversion

AES/EBU sources are sample rate converted to the 2101 Studio Interface's internal sample rate of 48kHz. 48Khz sources are re-clocked to the 2101 Studio Interface's internal clock (which is locked to the T-Link clock whenever at least one T-Link signal is present).

High-Pass filter

A high pass filter with a 250 Hz crossover frequency improves hybrid performance and enhances intelligibility by removing unnecessary low-frequency information from the audio present at the Studio Interface's input. This function is not defeatable.

Frequency Shifter

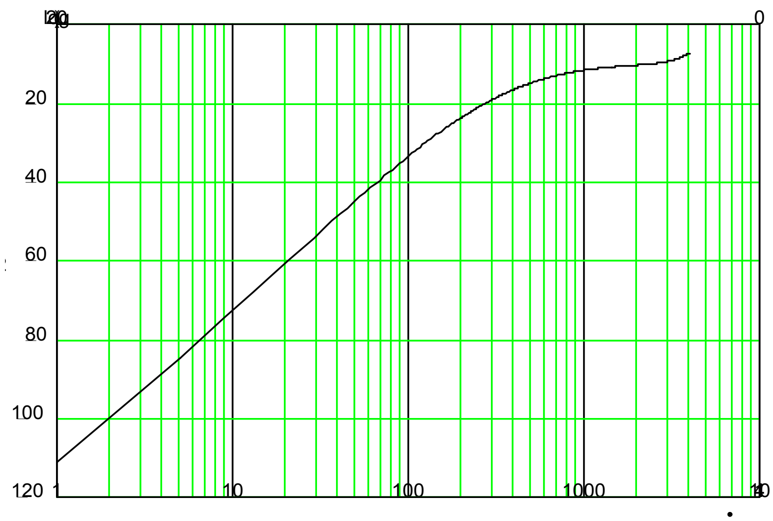
The frequency shifter (pitch shifter) inserts a small, unnoticeable, shift in frequency to the send audio to prevent feedback buildup when the system is used with open speakers. See below for additional considerations regarding feedback control.

Send Equalizer

Applies equalization to the audio sent to the caller. Choices are:

- Off - No equalization (default)
- On - Equalization for speech

Enabling this equalizer (On setting) simulates a telephone handset microphone, and will improve intelligibility. Sometimes high-quality studio microphones sound "too good" and the low-frequency audio they pick-up can sound unnatural on a telephone handset.



*Equalisation Characteristics of the **Send EQ***

Send Automatic Gain Control / Limiter

This AGC helps maintain consistent audio levels to the caller. This function is not defeatable. This is a carefully crafted algorithm utilising a feed-forward topology with near inaudible artifacts below the limit threshold. At moderate levels it is "AGC-like" while at high peak levels it is "limiter-like" in its effect.

3.5.2 Receive (Caller) Audio Processing

The 2101 Studio Interface's Receive Audio Processing consists of the following functions:

- High-pass "hum" filter
- Automatic Gain Control
- Noise gate
- Dynamic equalization
- Sample rate conversion

High-pass "hum" filter

This filter removes hum and other unwanted low frequency noise from the caller audio. This filter has a crossover frequency of 250 Hz.

Receive Automatic Gain Control

This AGC serves to improve consistency of caller audio levels. This function is very important to effective hybrid performance as caller levels can vary by as much as 30 dB. This AGC is an advanced dB linear feed forward topology providing a consistent sound independent of drive level. Its "smart" response normalizes levels while retaining the natural dynamics of the caller's voice.

An important additional feature of this AGC is that it is cross-coupled to other sections of the Studio Interface and can therefore reliably distinguish between caller audio and hybrid leakage. This allows a more aggressive gain control for bringing up low-level callers while still preserving excellent hybrid performance.

Noise Gate

Turning on the noise gate enables the built-in downward expander. The downward expander reduces low level line noise when no caller audio is present. And reduces low level leakage thereby improving hybrid performance. This function is cross-coupled with the AGCs and the duplex system.

Digital Dynamic Equalizer

Telephone audio frequency response varies widely as many different factors can affect it (we've measured the response on a number of calls and the results were revealing). Consequently, some form of receive equalization is desirable. See section 3.3.1 for details on how to adjust the DDEQ system.

The Digital Dynamic EQ process used on the Telos is the most sophisticated equalizer available in a telephone hybrid. The purpose is to allow you to customize the sound of callers on your station. All processing is performed in the digital domain. The *Receive EQ* selection controls the type of equalization applied to the receive telephone audio as follows:

- Off

The caller audio is passed without modification. The Rcv EQ values (see below) are ignored in this mode.

- Fixed

This is a simple manual equalizer mode where EQ values selected for LF and HF are applied constantly.

- Adaptive (Digital Dynamic)

This is a three band dynamic equalizer. The proper values of high and low frequency boost or cut is automatically determined and set. We've chosen frequency breakpoints, time constants, and other characteristics to optimize the tonal quality of varied telephone callers. The Rcv EQ values (see below) are ignored in this mode.

- Fix + Adap:

The Digital Dynamic EQ uses the Rcv EQ values (see below) as "target levels". This allows you to customize the quality of the sound, but still get a consistent quality to the audio from caller to caller. Otherwise, the functioning is the same as Adaptive.

3.5.3 Duplex "Ducking" System

The ducking functions occur on both the send and caller audio. This function is important for several reasons:

- Provides an "aesthetic" control over the caller that many programmers prefer. When the announcer speaks, the caller is ducked, or reduced in volume dynamically. This is particularly true when the caller is "carrying on" and the programmer wishes to get a "word in edgewise" so the show can continue.
- Reduces feedback when a loudspeaker is necessary. Because the ducking system operates on both audio paths, and because we designed it to be symmetrical in its gain reduction action, the "feedback gain" is kept constant and is reduced by the amount of the ducking level selected.
- In the loudspeaker case, reduces audibility of the caller signal that couples through the acoustic path, into the microphones, and is returned the caller resulting in a more natural sound for the caller.

The duplex system inserts a controlled loss (ducking) into whichever audio path (send or receive) is not active at the moment. When the caller is speaking, this loss is inserted in the announcer path, when the announcer is speaking the loss is inserted in the caller gain is reduced. The effect is somewhat "seesaw-like" with the total loss always being constant.

As a guideline, you will need a lower *DuplexLevel* setting when using open speakers. This helps reduce the occurrence of feedback and also reduces the chances of the caller hearing themself via the speaker to mic path which has an "unnatural" quality and can be disturbing. In a "morning zoo" type of scenario the setting should be even lower as there will be multiple mics used with open speakers. Fortunately, this scenario coincides with a more duplex style of operating – where pre-recorded "bits" may be played to callers and must be heard without interruption. The default setting is 10.

You may choose how much of this effect you prefer, from *HALF* (0) which makes the system operate like a "one way at a time" speakerphone to *FULL* (16) which disables ducking. A typical value for many applications would be in the range of 8-12.

3.5.4 Feedback Control

We've carefully designed the Studio Interface so that problems with feedback should be rare. However, despite excellent trans-hybrid loss and a number of other features, you can probably induce feedback if you try sufficiently hard (try cranking up your studio monitors). Some suggestions for what you should do if feedback does occur follow:

- Adjust the *DuplexLevel* selection in the Audio Menu to a lower setting
- When mic processing is used, connect the hybrid in such a way that it gets an unprocessed mic signal. Our Omnia ToolVox microphone processor has an output just for this purpose.

The problem here is that the mic processing combines with the 2101 Studio Interface's internal AGC to increase gain in the feedback path. Depending on the mic processor used, the feedback margin can be reduced by many dB. The Studio Interface's internal AGC has an internal adaptive smart-gate function to prevent inappropriate gain increase, however it is thwarted by this additional processing.

If it is not possible to get an unprocessed mic signal, try to set the mic processor in such a way that room noise is not "sucked-up" during pauses. You can also try reducing slightly the output the mic processor (or set the *Input Level* selection in the *Audio* menu to the next higher setting, thereby reducing levels into the unit). The level to the caller should be ok since the Send AGC will compensate, however you will force that AGC towards its maximum gain and thereby reduce how much gain it can add to the feedback path.

- Try repositioning the mics or speakers. It also helps, of course, to use directional mics. Customers have reported good results with cardioid mics such the EV RE-20 and the Shure SM-7.
- Add equalization to the monitor path. Acoustic (and electrical) resonances usually result in pronounced peaks in the feedback response. Since there are likely to be only a few pronounced peaks in this response characteristic, flattening out those peaks with an equalizer can significantly improve the feedback margin. A spectrum analyzer connected to the output of your mic processor will help detect the frequency of those peaks.
- If necessary, soften acoustic room reflections by adding curtains or wall treatments.

The discussion above is intended to help in those situations where open speakers are a requirement. Whenever possible, of course, it is best to use headphones to listen to the caller audio. We have found the best scheme is to have the phone monitor speaker mute when the mic is turned on. If your mix minus has provided a means for the mic signal to be active even when the mic channel is turned off, the system can be used like a speaker phone when the taking calls off air. When the call is on the air the speaker mutes, however the talent will generally be wearing headphones when on mic in any case.

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