Model 302

User Manual

Sonic Studio, LLC
www.sonicstudio.com

Sonic Studio

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Created by Sonic Studio, LLC <www.sonicstudio.com>, using Mac OS 10, Adobe InDesign and the Univers family of typefaces. We appreciate your comments and suggestions regarding this manual. Please contact us at <support@sonicstudio.com>.

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Chapter 1...Model 302 Overview

Thank you for purchasing a Model 302, the custom FireWire©–based professional audio interface from Sonic Studio. Your 302 provides an array of functions that allow you to record and mix with unprecedented quality.

Basics

The Model 302 is a portable, high quality, modular FireWire based multi-format audio converter, interface, and processor for professional audio applications. The 302 is equipped with two balanced analog inputs on Neutrik combo connectors, two channels of digital I/O, AES/EBU Type I and Type II unbalanced, two balanced analog outputs on 1/4" TRS, two balanced monitor outputs for connecting directly to power amps and self powered monitors, as well as word clock I/O plus 2 IEEE 1394a FireWire connectors that support 400 Mbs operation. All digital inputs and outputs are capable of 24 bit/96 kHz operation.

Features

1. 4 simultaneous input channels and 6 simultaneous output channels
2. Full 24 bit/96 kHz audio
3. 2 independent channels of high gain, low-noise mic-pre with switchable phantom power
4. Fully Portable Capabilities   Bus and Battery Powerable
5. Rack Mount Kit
6. 44.1, 48, 88.2, 96 kHz Sampling Rates
7. 24 bit 110 dB Dynamic Range A/D converters
8. 24 bit 120 dB Dynamic Range D/A converters
9. Selectable stereo Digital Inputs AES/EBU or S/PDIF
10. Stereo Digital Outputs AES/EBU and S/PDIF
11. Sample Rate Conversion (SRC) on Digital I/O
12. Built-in 80-bit, fully interpolated, multi-bus mixer for near-zero latency foldback of all input channels and all DAW busses simultaneously
13. Full cross point router for I/O management
14. Word Clock 1x, 256x

15. Front Panel Metering for Analog Inputs and main Outputs
16. Full console metering of every channel and mix bus
17. Total recall of every console parameter

What you need to use it

Computer:

a Power-PC based Macintosh with a FireWire Port and OS 9.1 or newer (G4, Powerbook G3 or G4 recommended). Mac OS 9.2.2 and Mac OS X recommended.

128MB of RAM

a monitor that supports 1024x768 resolution or better

Peripheral FireWire Adapter:

OHCI compliant PC-Card or

OHCI compliant PCI card

Software:

an ASIO 2 or CoreAudio compatible host (such as Cubase, Nuendo, Logic Audio, Digital Performer, Deck or Peak)

for ASIO direct monitoring-based foldback and overdubbing, an application that supports ASIO direct monitoring

What comes with it

Your 302 package contains the following items:

1. One Model 302
2. One IEC plug mold, a power cord appropriate to your country
3. One 24 volt, 48 watt, world-ready external power supply
4. One 0.5 meter IEEE 1394 6 pin FireWire cable
5. One 4.5 meter IEEE 1394 6 pin FireWire cable
6. Two rack ears w/ fasteners
7. Software CD-ROM including User Manual

If any of these items are missing from your package when you open it, please contact Sonic Studio or your dealer immediately for assistance.

Support Information

Warranty

The Model 302 hardware is covered by a manufacturer's warranty against manufacturing defects. The details of this warranty are described in a separate enclosed document. This warranty is applicable to products purchased in the USA. Products purchased in other regions are covered by a warranty administered by the distributor for that region and the terms will be as set forth in a separate document.

Registration

In order to receive warranty service, you must register the product with Sonic Studio. This may be done at any time with proof-of-purchase, but we strongly recommend that you register with us as soon as you purchase your unit. There are a couple of practical reasons for this:

Your product will be registered with us and this registration can be used as proof of ownership if your product is ever lost or stolen.

Sonic Studio aggressively updates the Model 302 on a regular basis and we would like to keep you informed of critical updates as they become available.

In order to register your 302, please e–mail or fax the following information to:

<register@sonicstudio.com>

775-330-8923 facsimile

Alternatively, if you have internet access you can use our automated registration webpage at:

http://www.sonicstudio.com/register

Service and Support

If you have problems configuring or using your 302 and you need help, please contact us. We offer free support via e–mail, as well as peer support on the Sonic Studio Users Group and MIO Users Group mailing lists.

For paid phone support, contact us at:

1-415-460-1201

For web–based support, visit:

http://www.sonicstudio.com/support/SupportRequest.txt

To subscribe to the Sonic Studio Users Group mailing list or to peruse the archives go to:

https://mail.music.vt.edu/mailman/listinfo/sonic

To subscribe to the MIO/302 Users Group mailing list or to peruse the archives go to:

https://mail.music.vt.edu/mailman/listinfo/mobileio

Finally, you can always find the latest info and updates for your Model 302 at the Sonic Studio website:

http://www.sonicstudio.com/support

Safety Compliance

This equipment has been tested and found to comply with the limits for a class B digital device pursuant to part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause interference to radio or television equipment reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by any combination of the following measures:

1. Relocate or reorient the receiving antenna
2. Increase the separation between the equipment and the receiver
3. Plug the equipment into an outlet on a circuit different from that to which the receiver is connected
4. Consult your dealer or experienced radio/television technician for additional assistance.

WARNING: Changes or modifications to this unit not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

302 Front Panel

The 302 front panel provides ten-segment metering for the 2 analog inputs and the main outputs as well as knobs and switches to control the input, monitor and headphones sections. The meters are fast PPM peak reading meters with auto-resetting peak holds.

Each input channel has the following controls:

Input gain knob

This is a 12 position, gold contact rotary switch which allows you to control the gain of the selected input.

Phantom Power enable switch

This is a push button switch which enables/disables phantom power. Push the switch IN to enable phantom power.

Trim Enable switch

This is a push button switch which allows you to control whether the attenuator trim pot is in the signal path or not. Push the switch IN to enable the trim pot. The attenuation range of the trim pot is -2 dB to -20 dB.

MIc/TRS switch

This is a push button switch which selects the input stage. The 302 has two distinct input stages: The Mic Amp and the DI Amp.

The Mic Amp is optimized for high gain and very low noise with low impedance sources like microphones. This input is connected to the XLR portion of the Neutrik combo connector. Maximum gain is 72 dB. Push the Mic/TRS switch IN to select the MIC input.

The DI amp is optimized for high impedence sources like magnetic pick-ups. This input is connected to the TRS portion of the Neutrik combo connector. Maximum gain is 63 dB. The Mic/TRS switch should be in the OUT position to use this input.

Trim Pot

The trim pot controls a passive attenuator. The attenuator is buffered between the return receiver and the A/D converter so its operation is transparent with regard to sound quality. Push the the trim enable switch IN to enable the trim pot. The attenuation range of the trim pot is -2 dB to -20 dB.

Status Display

The front panel also provides 302 system status at a glance:

Sample Rate

(nominal 44.1, 48, 88.2, or 96)

Since sample rate is determined by the currently selected clock source, it will accurately indicate the current sample rate, even when the clock source is provided by an external device.

Clock source

Internal indicates that the system is internally clocked

Word clock indicates that system is being clocked from the word clock input

256x WC indicates that the system is being clocked from a 256x clock at the word clock input

Digital In indicates that the system is being clocked from the selected digital input (AES or S/PDIF)

256x WC + Digital In indicates that the system is being clocked from the ADAT optical input

Power

Indicates that the 302 is receiving power.

FireWire

Indicates that the 302 has been successfully connected to a FireWire bus and has detected the isochronous cycle required to transmit and receive audio.

Locked

Indicates that the system clock recovery circuit is properly locked to the selected clock source. If this light is not illuminated, the 302 will not be locked to a clock and will revert to its fail–safe internal clock source. Even if the Locked light is not illuminated, the actual sample rate will still be indicated on the front panel display.

Digital I/O Section

The AES and S/PDIF lights are mutually exclusive and indicate which of the two input ports are feeding the Stereo Digital input of your 302. The Locked light indicates when the digital receiver is locked to the incoming digital audio signal.

Headphone

The 302 front panel provides access to the Headphone output and its associated level control knob as well as level control for the monitor outputs which are located on the back of the unit.

The headphone output is a TRS 1/4" jack that provides the Left Channel on the tip, the Right Channel on the ring and the ground return for the two channels on the sleeve.

302 rear panel features

Two channel, balanced MIC/LINE/INSTRUMENT inputs are available on Neutrik "Combo" connectors. Each input has:

1. 24 bit, 96 kHz A/D converters (110 dB SNR)
2. High gain, low noise mic amps with up to 72 dB of gain, fed by the XLR connector
3. High gain, low noise DI amps with up to 63 dB of gain, fed by the TRS connector
4. Switchable input impedance characteristics: MIC input 3.3k Ω, DI input 200k Ω
5. Switchable 48 volt phantom power on XLR connector
6. Balanced inserts which are post–preamp, pre–A/D
7. 2 channels balanced TRS main outputs

Each output has:

1. 24 bit, 96 kHz D/A converters (120 dB SNR)
2. Switchable nominal amplitude, +4/-10 dBu
3. Two channels balanced monitor outputs with front panel level control
4. 4 pin XLR-4M power port for use with ENG–style batteries, compatible with any power system with the following characteristics: 9 - 30 VDC, Pin 4 Hot, 15 Watts
5. Word clock input/output on BNC connectors
6. 256x "superclock" Word clock input/output on BNC connectors
7. Stereo AES Type II unbalanced "S/PDIF" input/output on RCA connectors
8. Stereo AES Type I balanced input/output on XLR connectors
9. Two FireWire IEEE 1394a ports — 400 Mbps
10. One 2.1mm DC power jack: 9 - 30 VDC, center positive, 15 Watts

Making connections to the 302

There are five classes of connections you can make to the 302 hardware:

Analog Audio

Copper-based Digital Audio

Clock Sync

FireWire

Power

Analog Audio Connections

The analog I/O connections on the 302 have been engineered for maximum flexibility. They support both balanced and unbalanced connections with a wide range of input and output levels as well as a wide range of matching impedances. This means that the Model 302 handles sources from mic level to line level and from mic impedance to guitar impedance. With that in mind, there are a number of aspects of the design that you should take into account when connecting to the 302.

There are really three distinct analog input stages available in a 302 input:

The Mic amp, which is fed by the XLR portion of the Combo connector.

The DI amp which is fed by the TRS portion of the combo connector.

The TRS return jack. This is a line level input which is the shortest path to the A/D converter.

Each input path is optimized for specific sources, but each is capable of handling a wide variety of sources. For example, both the Mic amp and the DI amp are capable of receiving Line level inputs. Additionally the DI input is caple of 63 dB of gain and an be used with dynamic microphones (phantom power is only available with the Mic Amp).

Feel free to experiment with the different input paths and choose the one which works best for a given application.

Whenever possible, use balanced interconnects with the Model 302. The performance of balanced interconnects is much higher, with far more resistant to noise interference and AC electrical or mains wiring problems. The expense of balanced interconnects is not substantially higher than unbalanced cabling so, if the gear that you are interfacing with supports balanced connection, do make use of them. If you cannot utilize balanced interconnects, there are connection schemes that you can use that will maximize performance.

On input, at line level it is sufficient to simply use standard unbalanced (TS) 1/4” connections. If you are interfacing with the 302 XLR inputs, you will need to ensure that pin 3 is grounded in the unbalanced adapter cable. The 302 XLR inputs are all wired pin 2 hot and the 1/4” inputs are wired Tip hot.

Note: To use the 302 TRS input with guitar or bass, you can simply use a standard TS guitar cable “patch cord” and it will work fine. However, you can take advantage of the balanced input design of the 302 to get more noise rejection than you thought possible on a guitar input. In order to do this, you will need to make a psuedo-balanced telescoping shield guitar cable. This can be constructed with a TRS connector, an unbalanced 1/4" TS connector and balanced microphone cable. This cable will treat the guitar as a floating balanced source and provide a "telescoping" shield from the 302 ground.

Telescoping Shield Cable for Instruments

If you want to use the TRS inputs with balanced microphones, you will need an XLR female to 1/4" TRS balanced plug adapter cable. These are available commercially, or you can construct one easily. The connections are Tip to Pin 2, Ring to Pin 3 and Sleeve to Pin 1.

XLR to Balanced TRS Cable

On output, the situation is a bit more complex. If you are driving an unbalanced load, you will get the best performance by not connecting the ring of the TRS jack to ground. In order to do this, you can simply use a balanced TRS/TRS connector with the unbalanced gear. You can also construct a special cable with a TRS connector and a TS connector. In this cable, you just let the ring of the TRS connector float.

TRS to TS unbalanced cable

Alternatively, the TS connector can be replaced with an RCA connector for interfacing with gear that has RCA unbalanced interconnects.

Making the 1/4" connection

When you connect a 1/4" plug to a 302 jack, insert it straight and firmly, ensuring that the plug is fully inserted into the jack. If the plug is not fully inserted you will get level shifts, phase flips, distortion, or no sound.

To disconnect a 1/4" plug, firmly pull the plug straight out from the connector body. The connectors on 302 are stiff, so you may have to exert some force to remove the plug.

Making the XLR connection

When you connect a Male XLR plug to a 302 jack, ensure that you have aligned the pins with the connector body and insert firmly until the retention tab clicks. To disconnect the plug, press the metal retention tab flush against the box, and pull the plug from the 302.

Digital Audio Inputs

302 supports 2 channels of digital audio–over–copper connections. These connections can be made using either AES Type II Unbalanced, so called "S/PDIF" interconnects with RCA connectors, or with AES Type I Balanced interconnects using XLR connector on 110 Ω cable. Even though only one of the AES or S/PDIF inputs can be active at any given time, you can have different digital sources connected to each of the input connectors at the same time. You use the

Sonic Console application to select the active input. Audio routed to the digital outputs will be mirrored by both S/PDIF and AES outputs. This allows you to send the same stereo pair to two devices at once.

We recommend that you use the AES interconnect mechanism to establish the digital communication between the 302 and other digital devices. The jitter and electrical noise tolerance on professional AES Type I interconnects is substantially better than consumer AES Type II interconnects. The AES Type I Balanced interconnect standard is equivalent to balanced analog audio interconnections. If you need to use Type II Unbalanced (S/PDIF) interconnects, try to use the shortest cables you can and, if possible, use special purpose 75 Ω S/PDIF or video cables.

The RCA connectors used for S/PDIF are friction–fit coaxial connectors. When you connect them, ensure that they are fully inserted and tight.

The XLR connectors used for AES Type I Balanced are fully locking. When connecting to them, make sure that you align the pins and insert firmly. When you remove the connector, make sure that you release the lock by pressing the lock release button before you pull the connector out of the 302.

Integrated SRC

Normally, when working with digital audio transport, you must take care to ensure that all devices communicating with one another are synchronized to the same audio clock. While this is still an important consideration with the Model 302, the hardware provides a special feature to simplify copper-based digital connections to the box. The digital input on the Model 302 has an optional asynchronous sample rate converter (SRC) that will automatically match the sample rate of the incoming audio to the sample rate of the 302. This converter is enabled by default and you should disable it in the System section of the Sonic Console. If you have synchronized the 302 to the external source, using any of the extensive synchronization methods provided, you will generally want to disable the SRC in order to get 24 bit transparent transport via the digital inputs.

Clock Sync

Clock sync is a serious consideration in any digital audio system.

If you are recording analog sources with 302, you can simply use the unit's high-quality internal clock source to drive the converters. This is the easiest method.

If you need to interface with other devices digitally or ensure sample accurate sync with video sources, the extensive clock synchronization capabilities of 302 will prove to be more reliable (and better sounding) than most higher priced alternatives.

There are three different ways to get external clock information into the unit:

1. Sending a 1x word clock signal into the WC Input BNC.
2. Sending a 256x word clock signal into the WC Input BNC.

3. Sending an AES or S/PDIF signal into the Digital input.

The BNC word clock input port is a 75 Ω terminated coaxial input. It should be driven by a 75 Ω source driver and interconnected with 75 Ω coaxial cable. If you do not use proper cabling and source drive, you will introduce reflections on the word clock cable which will propagate jitter into the recovered word clock and your final output. This is true whether you use the port as a 1x WC input or a 256x WC input, but becomes more important when the clock signal is 256x.

1x is generally appropriate for use with devices that provide a word clock output. If your device provides a 256x output, you may find that you get better results using that clock signal. The Digidesign© line of ProTools© products use 256x as their “Superclock” clocking reference.

The AES recommended procedure for distributing clock is to use an AES clock signal. The AES clock signal is an AES digital audio signal with no audio activity. The Model 302 uses only the AES preambles for clock recovery, so it is immune to data–dependent jitter effects. This means you can reliably use the Digital Input as a clock source with or without audio data.

Finally, you can recover clock from the AES Type II Optical input sent to the 302 from an ADAT device. In general, if you want to use the “ADAT optical” input to receive audio, you should choose the ADAT optical input as the clock source.

FireWire

FireWire© is Apple’s registered trademark for the IEEE 1394 high speed serial bus. FireWire started as an Apple technology to replace a variety of interface ports on the back of the computer. After promulgating a number of closed proprietary technologies in the early days of the Macintosh, Apple determined that open standards were better for the Mac, for the industry, and for Apple itself. On that basis they opened their technology for standardization under the auspices of the Institute of Electrical and Electronics Engineers, Inc. (IEEE), an international organization that promotes standards in the field of electronics. FireWire was standardized as IEEE 1394 and promoted for open licensing in the industry.

The first widespread adoption of the technology was for DV camcorders where space was at a premium and bus powering was not percieved as a real issue since all camcorders have batteries. Sony designed an alternative version of the standard 6 pin FireWire connector that provided 1394–based communication with 4 pins in a much smaller form factor. This version of the connector sacrificed bus power support and mechanical stability for reduced space requirements. Sony dubbed this version “i.Link©.” This became the de facto standard in the DV world, and was later added to the IEEE 1394 standard. Both i.Link and FireWire refer to the same underlying standard and are completely interoperable. Obviously, i.Link connectors and FireWire connectors cannot be used together without adapters.

The Model 302 uses the 6 pin implementation of FireWire for bus power support. The unit ships with two 6 pin to 6 pin cables, one that is 0.5 meters long, about 18 inches, and the other 4.5m long or about 14.5 feet. If you want to use your 302 with a 4 pin FireWire device, you will need to purchase a 6 pin to 4 pin adapter cable. These cables are available from a wide variety of retail sources. If you are using a 4 pin cable to connect any device to the computer with 302, bus power will not be available.

The 6 pin FireWire connector is keyed by its shape, as one end of the connector is pointed. The FireWire ports on the Model 302 point downwards toward the bottom of the box. It will be very difficult to insert the connector upside down, but it is possible if you force it. If the plug is inserted into the socket upside down, the socket will be destroyed. ***Never force a FireWire connector into a FireWire socket!***

Devices connected to the FireWire bus are auto–configuring. You do not need to set IDs, DIP switches or in any way configure the devices in order to facilitate communication between devices or to configure of the bus.

FireWire devices on the same bus must be connected in a tree structure with no loops. This means that devices can be connected to each other in any order, and any device with multiple ports can act as a chain or a hub for other FireWire devices, but you should never be able to get from one device to another by more than one path. If you construct a loop in the bus, it will not operate properly and you will not be able access some or all of the devices on the bus.

Although you are able to attach devices in any order on the FireWire bus, the order of attachment will have an impact on performance. Most current model FireWire devices support 400 Mbs operation, but many older devices may only support 100 or 200 Mbs operation. These devices act as a bottleneck in the bus and limit the speed of any bus traffic that flows through them. In order to maximize performance, you want to ensure that low speed devices are not used to join high speed devices. In practice this generally means that you should attach your 302 directly to your computer or through a high speed FireWire hub.

To connect your 302 to your computer, simply connect the 302 and the computer with a FireWire cable. The FireWire bus provides a path for all communications between the computer and the 302: audio, control and metering data.

The Model 302's audio transport takes advantage of FireWire's support for isochronous transmission, in which the 302 can reserve a dedicated amount of bandwidth on the bus for moving audio samples. Since the audio must be transmitted on a regular basis to ensure continuous playback and recording, the isochronous mode is ideally suited to this task.

Control changes and meter data are transmitted using asynchronous transactions on the FireWire bus. This transmission approach makes use of the unreserved bandwidth on the bus and competes with processes like FireWire hard disk access for time. Under normal circumstances, this is completely transparent to the user. If the bus becomes overloaded, you may find that disk accesses and meter updates slow down. If you are experiencing bus overloads, you can always add a second FireWire bus with a third-party HBA. A Host Bus Adapter, either PC-Card or PCI, depending on your machine), will offload one or more devices to the second bus.

Power

The 302 ships with a world-ready, 24 volt, 2 amp power supply. You can plug this supply into any AC power source from 90 to 240 VAC, 50 to 60 Hz, using an appropriate IEC power cord, and it will supply the proper power to the 302 on the 2.1 mm coaxial power connector. The Model 302 will automatically supply the extra power to the FireWire bus. This means that the 302 and its power supply can be used to power other bus-powerable FireWire devices including disk drives, hubs and other Model 302 units. Note that Model 303, 304 and 305 units require external power.

One of the Model 302's great strengths is the flexibility of its power system. The 302 can be powered from any DC source including bus power, in the range of 9 to 30 VDC as long as it provides 12 Watts of power. The DC input on the 302 is a 2.1 mm coaxial power connector, center positive and a 4 pin XLR connector, pin 4 Hot. If you are powering the unit with a third party power source and it supplies 9 VDC, the power source will have to provide 1.4 amps of current. If you are powering the unit with 12 VDC, the power source will have to provide 1 amp of current, etc.

Since the Model 302 is DC powered, you can also power the device using the FireWire bus or another DC source. As mentioned above, the 302 uses 12 Watts, so any device supplying bus power must be capable of sourcing that much power. Most desktop Macs provide more than enough power for your Model 302 and one other low power device. Most laptops provide enough power for the 302, but not enough for a 302 and another bus-powered device at the same time. If you are using a PowerBook, you should not expect to be able to power both the 302 and a hard drive from the computer. The power capabilities of individual computers vary, so you will have to test the complete system to determine exactly how much your configuration can handle.

If you find that the computer is not capable of powering the 302 or does not provide enough run time, you may want to explore using an external power source. Check with Sonic Studio for details on different battery powered solutions for the Model 302.

As with all electronic devices, when connecting an external power source to the 302, you should first connect the power source to 302 while it is in an unenergized state (e.g. not connected to the mains or switched off). After the connection to the 302 has been made, you should energize the power source.

If you connect an energized power source to the Model 302's 2.1mm power connector, you may see a small spark when you make the connection. This is due to surge current and is normal. While this will not damage the 302 in any way, to avoid the spark just connect the power connector to your 302 *before* connecting the power source to the wall.

Sonic Console

The Sonic Console is the nerve center of your Model 302. Functioning as a standalone application in OS X, Sonic Console provides full control of every aspect of your Model 302. The Console allows you to rapidly and easily adjust the system sample rate, Digital I/O source, and system clock source. It also allows you to assign ASIO output channels and hardware input channels to the integrated 80 bit, fully interpolated, multi–bus, near zero latency hardware mixer.

Sonic Console's powerful mixer supports both mono and stereo busses, with solo and mute functions for all input and master channels. The mixer bus outputs are routable to any of the hardware outputs, allowing you to easily create multiple simultaneous mixes for send/return busses and multiple live main and monitor mixes. Various applications include foldback support for multiple performers, separate monitor feeds for studio, tape, and control room, with separate mixes for front of house, archive recording, and monitors for live shows.

Sonic Console also contains a patchbay router, which allows you to quickly select the source being fed to any output. The patchbay provides easy configuration of stand-alone operation, mix mults, direct outs and various combinations thereof to suit the needs of the moment.

Sonic Console Output Patchbay

In order to simplify work flow and optimize the extent of system control, Sonic Console supports comprehensive preset management on both a global and individual control level.

The preset management pop–up controls within Sonic Console allow you to configure various aspects of your 302 and save that configuration information for later recall. Various applications include storing routing configurations for monitor setups, mixer configurations for stem and scene recall, storing analog level standards for interfacing with external gear and managing different mastering standards.

Global configuration snapshots allow you to save each and every aspect of a 302's configuration for later, total instant recall. This is useful for pre–configuring a 302 and bringing back the configuration once you're at the gig, managing separate location setups, or for saving complex studio routing setups for quick changeover.

Sonic Console Overview

The Sonic Console application consolidates all of the controls for the Mobile I/O hardware into one easy to use window. The 302 has an extremely large number of user configurable parameters and it is very important that you have instant access to the ones that you need. The console provides a thinly layered interface to the entire system and keeps you from having to deal with "Window Overload."

The Sonic Console window has a view panel selector bar that runs along the top of the window. This bar indicates which of the console view panels is currently active. You can tell which panel is active because the button in the bar is "pushed in." To switch to one of the other panels, simply click on the name of the panel you want to use. The view will change instantly to the one that you have selected.

View Panel Pane Selector Bar

Under the view panel selector bar is the currently selected view panel. You control the various aspects of the box with the controls in each view. The Sonic Console has three main panels:

1. Analog I/O Panel
2. Mixer Panel
3. Routing Panel

Analog I/O Panel

This panel provides full control and metering of all of the analog I/O that the box provides. The top half of the view is dedicated to inputs and the bottom half is dedicated to outputs. You access this panel by clicking on the "Analog I/O Control" button of the view panel selector bar.

Channel Label

This simply labels which channel is associated with the channel strip.

Channel Level Meter

This is a peak reading, high-resolution, fast PPM meter. It shows the post converter level of the input signal of the associated channel. The peak hold bar indicates the highest level seen on the channel since the last reset. You can reset the peak hold by clicking on the meter. These meters are simply high resolution versions of the meters shown on the front panel of the box. All the meter data is generated by the 302 hardware.

Optimizing Input levels

The Analog to Digital converters (ADC) in most devices function best, with the lowest distortion and best sound, when the peak level is around -6 dBFS. This is true of the ADCs in the Model 302. Since you have full level control of the input with the gain trim knob, you will find that you get the best quality recordings if you set the nominal peak level of the input at about -6 dBFS. In addition to providing the best recording quality, it has the added benefit that you will operate with 6 dB of headroom before clipping. There is no drawback to optimizing your levels in this way, and plenty of benefit.

Digital Input Meters

To the right of the Analog Input control section is the Digital Input Meter section. This group of meters provides level metering for all of the digital inputs on the 302. These meters have the same response characteristics as the analog input meters and show you the audio activity on digital input channels 1-2, going from left to right.

System Controls

The System block provides controls that adjust various system level aspects of the 302 hardware:

The Clock Source popup menu controls the system clock source used by the hardware for digital synchronization and driving the converters:

Clock Source Pop-up Menu

Internal: causes 302 to use its internal clock. You must select this if you want to set the sample rate from the 302. If any other clock source has been selected, the console will not allow you to change the sample rate since the sample rate is determined by the external clock source.

WC (44/48): directs 302 to clock off of an external Word Clock Source at single baseband rate (e.g. fs = 32-50 kHz)

WC (88/96): directs 302 to clock off of an external Word Clock Source at double rate (e.g. fs = 64-100 kHz)

WCx256 (44/48) directs 302 to clock off of an external 256fs Clock Source at single rate (e.g. fs = 32-50 kHz)

WCx256 (88/96) directs 302 to clock off of an external 256fs Clock Source at double rate (e.g. fs = 64-100 kHz)

DigIn (44/48) directs 302 to clock off of the selected stereo digital input at single rate (e.g. fs = 32-50 kHz). This allows operation of the digital input without SRC, and from devices that must supply clock.

DigIn (88/96) directs 302 to clock off of the selected stereo digital input at double rate (e.g. fs = 64-100 kHz). This allows operation of the digital input without SRC, and from devices that must supply clock.

The Sample Rate popup menu allows you to select the sample rate when you are using internal clock. The 302 must be running on internal clock for the Sample Rate popup menu to have any effect. If the 302 is running from an external clock source, you cannot select the sample rate since it is determined by the external clock source.

Sample Rate Popup Menu

The WC Out popup menu allows you to select the output clock signal the 302 generates on its WC Out BNC connector. The available choices are 1x and 256x. The 1x signal is appropriate for driving devices that accept a Word Clock signal. The 256x signal is appropriate for driving devices that accept 256x or SuperClock signals. Refer to the documentation for the external device to determine what is the most appropriate clock reference for it.

WC Out Popup

The DI Source popup menu allows you to select the active input for the digital input pair. The choices are AES and S/PDIF. This selector physically switches the input to the digital audio receiver between the RCA and XLR input connectors.

DI Source Popup

The DI SRC button enables and disables the asynchronous sample rate converter (SRC) in the 302 digital audio receiver. When the SRC is engaged when the button is illuminated yellow, the digital audio receiver will automatically synchronize the input signal to the 302 system clock over a wide range of sample rate ratios.

DI SRC Buton

This allows you to, for example, digitally transfer a sample from a CD player into a 96 k session without any clocking problems. If you want to make bit–transparent transfers, you will need to disengage the SRC and ensure that the 302 and the external device are both using the same digital audio clock via one of the 302 synchronization mechanisms.

The Lock indicators show which elements of the 302 clocking system are properly locked. The clocking system must be locked for the unit to behave as expected. If the system is not locked, audio will play at the wrong rate and will be distorted or noisy. Under normal circumstances, the system should always be locked, but if you have selected an external clock source and the clock signal is not present, corrupted or out-of-range, the system may unlock. There are indicators for the system and the digital input.

Analog Output Control

The bottom half of the panel is dedicated to the hardware outputs of the 302.

Channel Label button

This simply labels which channel is associated with the output.

Channel Level Meter

This is a peak reading, high-resolution fast PPM meter. It shows the pre–converter level of the output signal for the associated channel. The peak hold bar indicates the highest level seen on the channel since the last reset. You can reset the hold by clicking on the meter. All meter data is generated by the 302 hardware.

Digital Output Meters

To the right of the Analog Output controls is the Digital Meters section. This group of meters provides level metering for all of the digital outputs on the 302. These meters have the same response characteristics as the analog output meters, and show you the audio activity on ADAT output channels 1-8 and digital output channels 1-2, going from left to right. Please note that when the box is clocking at 2x rates (88.2-96 kHz), the ADAT output uses pairs of optical channels to transport the audio and only channels 1-4 will show activity, channels 5-8 will not display signal. Both channels of the stereo digital output remain active at all sample rates.

Box Info

The Box Info section of the panel, in the lower right-hand corner of the window, shows you information about the currently connected and selected 302 unit. This section displays the serial number, model information and firmware revision of the connected box, as well as the DSP load for the unit. All of this information can be useful in trying to track down any connection problems that may arise.

Box Info

If there is no information displayed in the Box Info section, the software is not communicating properly with the 302 hardware, or there is no 302 present on the FireWire bus. If the FireWire light on the front panel of the 302 is illuminated but the box information does not appear in the Console window, it is very likely that the software has not been installed properly. If this is the case, please refer to the installation instructions for details on how to properly install the software. Also, please ensure that there are no copies of the Driver in the same folder as the Sonic Console application or any of the ASIO drivers that you will use with an ASIO host.

If, on the other hand, the FireWire light on the front panel of the 302 is not illuminated, the box is not communicating properly with the computer. Please check the cabling of your 302 and other devices on the FireWire bus and make sure that everything is connected correctly. If that does not properly establish the connection, try rebooting your computer. As a last resort, try connecting only the 302 to the computer to ensure that communication can be established.

Mixer Panel

The Analog I/O panel provides all of the bread-and-butter functionality that you would expect from a flexible audio interface like the Model 302: what you need to get the job done quickly and easily. The Mixer panel provides features that you won't find anywhere else.

The controls in the top half of the mixer panel are the same analog input controls described in the "Analog I/O View" section. The bottom half of the mixer panel provides a control surface for the multiple integrated hardware mixers running on the 302. You access this panel by clicking on the "Mixer" button of the view panel selector bar at the top of the Console window.

Mixer Controls Overview

Each mixer has faders for Level, with Solo and Mute buttons for each channel. Each stereo mixer also includes a pan control knob for each mono channel. For stereo inputs, such as the default digital inputs or your DAW output, will not have a pan knob. Instead, the two individual channels that make up the pair will be hard panned, and the fader meters will appear as stereo meters.

At the far right of each mixer surface is the Master Fader. This fader will be Mono for Mono Mixers and Stereo for Stereo Mixers. Above the Master Fader is a bus master mute button that allows you to mute the entire mixer.

Each mix bus can have up to 36 inputs assigned to it. All of the analog, digital, and ADAT hardware inputs are available in each mixer. You can also assign any or all of the ASIO playback channels from your DAW to each mixer. Since there are 18 hardware inputs and 18 FireWire return busses that are routed from the ASIO driver on the host computer, you can mix up to 36 channels on each bus.

The default state of the hardware input channels in the mixers is faders at unity, mutes on, solos off and center panned. Note that channels are muted by default. The playback channels from your DAW are unmuted in pairs per bus. DAW 1/2 will be unmuted on Mix 1/2, DAW 3/4 will be unmuted on Mix 3/4, et cetera.

Mixer Pane Tabs

Use the tabs above the pan knobs to select the mixer you want to control. Each tab represents a mixer you have configured in the routing panel. When you click on a tab, the controls will be instantly updated to reflect the state of the associated mixer. In this way you can quickly switch back and forth between multiple independent mixes, each with a full mixing interface.

Each mixer also has a Parameter Pop–up control associated with it. Unlike the Input and Output controls, the parameter control pop–up does not have an explicit user interface element representing it. Instead, the pop–up is accessed from the mixer's tab itself. To pop-up the menu, you either click and hold the tab, or control-click the tab. The parameter popup menu allows you to maintain a library of standard mix configurations, scenes, and setups. It also provides a very quick method for copying mixes from one bus to another. See "Parameter Popup Controls" for more information.

Channel Faders

The Channel fader controls the relative level of the input channel in the mix. It works just like its non-virtual counterpart. The calibration numbers to the left of the fader knob provide an accurate guide to the amount of gain that will be applied by the fader. The label area above the fader knob displays which input channel the fader is controlling. The labels default to the console's default names for the channels, but you can rename the channels to more meaningful names using the naming controls in the Routing Matrix. See "Configuring Channel Names in the Matrix" for more information.

The exact amount of gain for the fader is displayed in the small black window above the fader channel label. If you want to set a channel gain precisely, simply click in the small black window and type in the desired gain in dB numerically. Hitting the return key or clicking outside the entry box makes the new setting take effect. Hitting the tab key will make the new setting take effect and will move to the next numeric entry field in the mixer.

The gain range for each channel is from -∞ (muted) to +10 dB. The resolution of the gain control is extremely fine and gain changes are interpolated in the mixer, so there is no zipper noise when you change gain.

Command–clicking the fader knob will allow you to make fine adjustments to the fader level by dragging the mouse up and down. Option–clicking the fader knob will reset the fader to unity gain. Clicking the meter associated with the fader will clear the peak holds for that channel.

For mono channels, the current gain is applied to the channel before the pan. For stereo channels, the current gain is applied uniformly to both of the hard panned input channels.

Channel Meters

Each channel fader has a meter or a pair of meters for stereo channels associated with it. The meters are calibrated consistently with the fader calibration. Each meter is a peak reading, fast PPM, pre-fader meter. The peak hold bar shows the biggest peak since the last reset. Clicking on a meter will clear the peak holds for that channel.

Channel Pans

Each mono input in a stereo mixer has a pan knob above the fader. The pan knob allows you to control the relative amount of the input channel that is placed into the two busses of the stereo mixer. Panning hard left (L100) means that the channel will appear at full volume in the left or odd bus of the mixer. Panning hard right (R100) means that the channel will appear at full volume in the right or even bus of the mixer. When the channel is center panned, the signal appears at a decreased volume, -3 dB down in both channels, such that the volume of the total signal in both channels is equivalent to full volume in one channel. As you pan from left to right, the signal is distributed between the two channels so that the total volume remains constant.

As with the channel fader, the black window above the channel pan knob provides a precise readout of the current pan position, and clicking on the black window will allow you to type in an exact pan amount. Negative numbers (-100 to 0) indicate left pans, positive numbers (0 to 100) indicate right pans and 0 represents center pan.

Option–clicking the pan knob will return it to center pan, and command– click-dragging the knob will allow you to adjust the pan position in fine increments.

Mute Buttons

Each input has a Mute button, labeled "M," associated with it. When the mute button is engaged and illuminated white, the channel will be muted in the mixer. The mute is interpolated, so muting a channel will not cause audible clicks.

Option–clicking a mute button will set all of the mutes on the mix bus to the same state as the button you click. This allows you to quickly mute all the channels or quickly unmute all the channels on a bus.

Solo Buttons

Each input has a Solo button, labeled "S," associated with it. When the solo button is engaged and illuminated red, only channels that have been soloed will be mixed by the mixer. The other channels will effectively be muted. As with the mute button, the gain changes associated with soloing or unsoloing a channel are fully interpolated and will not cause audible clicks.

Command–clicking a solo button will exclusively solo the associated channel. A command–click will automatically clear the solo state of all other channels on the mix bus. Option–clicking a solo button will set all of the solos on the mix bus to the same state as the button you click. This allows you to quickly solo all the channels or quickly un-solo all the channels on a bus.

Mixer Master Fader

The Fader that appears on the right side of the mixer pane is the Mix Master Fader. This fader controls the overall bus level of the mix bus. Operationally, it is exactly the same as the channel faders. See "Channel Faders" for more information.

Each Mix bus in the system has 24 dB of headroom above full scale. If the summing point of the mix bus is clipping, you can pull the mix out of clipping by dropping the Master fader, as long as the sum point is clipping by less than 24 dB.

Mixer Master Mute

Each mix bus has a master mute button above the master fader that allows you to mute the output of that mixer. This mute is interpolated.

WIDE Mixers

As was described before, the integrated mixers in 302 are WIDE, That is, they allow you to mix every available input channel together, both hardware and FireWire channels. In the widest case, the mixer will have 36 faders to allow you to control the gains for all of the input channels.

The Sonic Console window is only wide enough to accommodate 18 faders plus the Master fader. If you have enabled more channels than will fit in the window, the Console will automatically display a scrollbar at the bottom of the Mixer pane. Use this scrollbar to control which faders are visible at any given time. The width of the scrollbar indicator shows you how many of the enabled channels are visible at any given time. The scrollbar will automatically be hidden if you reduce the number of inputs to the mixer below 19 channels or if you switch to a different mixer that has less than 19 channels enabled.

Mixer Applications

Note: The Model 302's support of near-zero latency mixing of every channel opens up a huge variety of applications that cannot be achieved with standard interfaces or, at the very least, require external gear or major work–arounds to accomplish. Some examples are:

Stem-based Mixing

In this mixing technique, you mix disparate elements of the program to separate sub-mixes called stems. You might mix drums to one stem, instruments to another, and vocals to a third. Then the relative balances of the mix can be addressed later in a macroscopic way (during mastering, for example). This also enables remixing the project easily without having to go back to the multi-track master. Since you will be creating stems on individual busses in the DAW, you need to sum the stems for monitoring. This is easily accomplished with 302's mixer.

Monitoring Effects Sends When Outboard Gear is Unavailable

You may find that you want to continue editing or mixing while you are away from the studio. You can use the WIDE mixer in the 302 to mix in DAW effects send busses for monitoring without having to reconfigure your DAW session.

Multichannel Foldback Mixes

Near-Zero Latency monitoring of external effects.

Most singers need some reverb or other effects to get the feel right during their performance. With 302 and the WIDE mixer, you can split off a send from the performer's input channel, send it to an external effects unit, and mix the effect return into the performer's foldback mix -- with virtually no latency.

Multiple WIDE Mixes

Since the Model 302 supports multiple WIDE mix busses simultaneously, you can form multiple, individual foldback mixes for multiple performers at the same time. In addition, each mix has its own complete mixer control surface, so you don't "scan and decode" multiple aux send knobs.

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