The Dangerous 2-Bus Manual

Thank you for choosing products from the exciting line of ***Dangerous*** recording equipment. Many years of dependable and trouble-free performance can be expected from our gear. This has been made possible by the careful design, construction and top-shelf component choices by recording industry veterans. The designers here at ***Dangerous*** are committed to a common goal: to bring you the highest quality possible for your dollars.

This manual will assist you in the installation of the 2-Bus and the calibration of your system. In order to complete it in a timely fashion, the chapter "Mix Buss Theory" has been left out. It will be available soon and sent to registered users of the 2-Bus.

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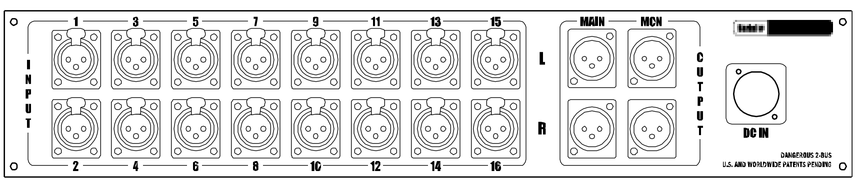
Overview

The 2-bus is a 16X2 summing amplifier designed to help the users of digital audio workstations achieve better mix performance through the use of the existing equipment in their studios. The designers and their colleagues have noted that while digital audio workstations (DAW's) offer unprecedented flexibility in multitrack recording and editing, the mixing buss in these systems generally doesn't perform up to the quality of most analog recording consoles in terms of sound quality and preservation of spatial detail. In today's portable environment, and with the cost of maintaining and housing legendary recording equipment, the choice of the big mixing desk is impossible for many users of DAW's. It is in this spirit that the Dangerous engineering team is designing and manufacturing an exciting array of mic pre-amps, summing amps, monitoring, and metering equipment designed to meet the challenge of today's recording environment.

Hooking up your Dangerous 2-Bus



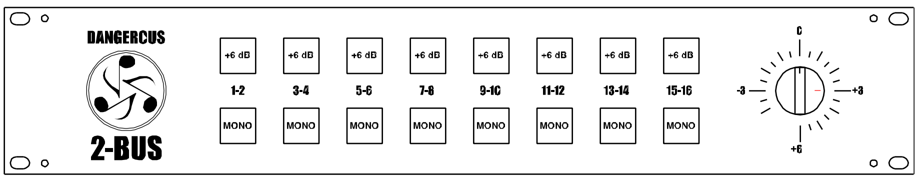
DANGEROUS 2-BUS REAR PANEL

The 2-Bus is designed to mix the outputs of one or two eight channel D/A converters (Digidesign 882, 888, Apogee AD8000, Prism Dream ADA-8, Troisi DC8-224DAC, etc….). The hook-up is simple. Use standard mic cables (or an 8- or 16-ch. snake) with XLR connectors to plug from the D/A analog output to the 2-Bus inputs. It is not necessary or desirable to lift the shields on these cables. The "Main" output feeds your 2-track mixdown medium, be it an external A/D converter, DAT machine, CD writer, analog 2-track, etc.
The "Monitor" output goes to an analog input on a Dangerous Monitor box or to the monitor section of an existing console.

It is sometimes desirable to lift the shield on the receiving end of an output cable to ensure the best noise performance. Using "ground lift" plugs on equipment is generally not the best way to achieve quiet performance. The chapter titled "Grounding Examples" goes over some of the principles of this subject and is recommended reading for anyone setting up a studio.

Users who have D/A converters with unbalanced (RCA) outputs should refer to the wiring diagrams and explanations in that chapter. The levels of both 2-Bus outputs are at a nominal +4 dBu studio line-level (1.23 volts RMS). Cables can be custom-manufactured to your specifications by Dangerous Music, Inc. by calling or contacting us through our website Dangerousmusic.com. We use the best examples of Mogami wire and Neutrik connectors coupled with quality soldering techniques to ensure exemplary performance.

Usage Examples



DANGEROUS 2-BUS FRONT PANEL

There are three types of controls on the 2-Bus - namely the "Mono" switch, the "+6 dB" switch, and a stepped attenuator to control overall output gain.

The switches on the front panel on a 2-Bus are laid out so that the inputs can be treated as stereo pairs (panned hard left-right) or as individual inputs panned up the middle. This function is selected by pushing the "Mono" button of any channel pair to put both inputs of the pair "up the middle" of the mix. For sounds that are panned somewhere in between the two extremes, a pair of outputs is used and the "Mono" button is not engaged.

For example, Let's say that one wants to mix a drum kit with bass, guitars, a lead vocal, and reverb on all of the above. To keep things simple, assume that our engineer has a Pro-Tools system with one 888 I/O box for a total of 8 analog feeds into the 2-Bus. This typical mix has 8 tracks of drums, one bass, two guitars, one vocal, and a reverb return for a total of fourteen tracks in the system. Usually, a DAW user would mix the song then spit the resulting stereo feed to two additional tracks internally or out of the 1st digital output of the 888 onto a DAT machine. By using 4 sets of outputs in Pro-Tools, the audio performance can be increased by assigning the drums to PT outputs 1&2, the bass and vocal to outputs 3&4, and the guitars to outputs 5&6. The bass and vocal want their own channels so they are panned hard left and right in Pro-Tools and the "Mono" button for channels

3-4 is pressed on the front panel of the 2-Bus to put the two “up the middle”. Reverb is sent from the tracks to a plug-in and the result routed to outputs 7&8. The panning on the drums and guitars is determined by the setting of the panners on the Pro-Tools desk. The “Mono” button pans the bass and vocal to the center of the mix. The “Main” output is sent to an external A/D converter then to a DAT machine for recording. This mix has the same balance as the one done internally, but because the summing to a stereo pair is being done externally in a high-quality analog environment rather than digitally, all of the computer’s processing power can be spent on the individual tracks, and the results are incredible imaging, clarity, punch, and better overall musicality. (This designer has some theories about why this is true explained in the chapter titled “Mix Buss Theory”.)

The “+6 dB” switch adds gain to a channel pair if needed. (There are reasons why it may be desirable to turn up the fader levels 6 dB in the computer instead of pushing the +6 dB button. The pros and cons of this action are treated in the “Mix Buss Theory” chapter.)

The output “Gain” control has a range of 10dB in .5 dB steps. It is at unity for any individual channel when fully open (clockwise) but the recommended setting is to start at what is labeled “0” on the knob (12:00). This allows for ample maneuvering room to achieve optimal level to your mixdown medium.

Calibration

The Dangerous 2-Bus comes to you fully calibrated. For a more in-depth look at the calibration and alignment of the recording studio environment, see the “Studio Calibration” chapter.

The following chapters provide a more detailed look at some aspects of recording studio setup and wiring.

Studio Calibration

In order to enjoy the full benefits of the 2-Bus, it is necessary to align the D/A converters that feed it. This can be done with the aid of an AC voltmeter (available from Digi-Key, Techni-Tool, Radio Shack, et.al.) and the digital oscillator found inside the DAW.

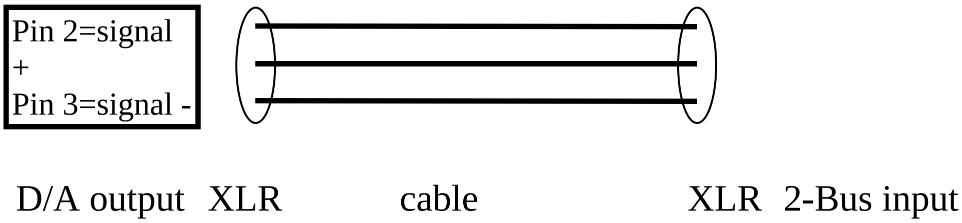
A 1 kHz signal at a level of -14dB full scale should be fed to the individual channels of the D/A converters from the DAW oscillator and the voltage at the output connectors adjusted to read 1.23 volts AC (+4dBu) across pins 2&3 of the XLR on professional level systems. Semi-pro systems (ones with RCA connectors) usually get adjusted to .245 volts RMS (-10dBu) measured between the center pin and the shell of the connector. If there are no adjustments for level on the device than it is best not to worry as long as the levels are the same within several millivolts. This adjustment sets the maximum operating level from the converters to +18dBu and will avoid clipping professional outboard gear. The 2-Bus inputs clip above +25dBu so headroom is not a problem on individual inputs.

Grounding Examples

To achieve maximum performance from the 2-Bus, cables need to be wired correctly. Custom cable sets can be ordered from Dangerous Music, Inc. if necessary. Store-bought cables can work just fine. Below are diagrams of frequently encountered wiring scenarios to help explain some of the possibilities. We’ll start with professional, balanced, +4dBu systems encountered in most studio situations and then cover some contingencies that happen when interfacing -10dBu, semi-pro, unbalanced equipment.

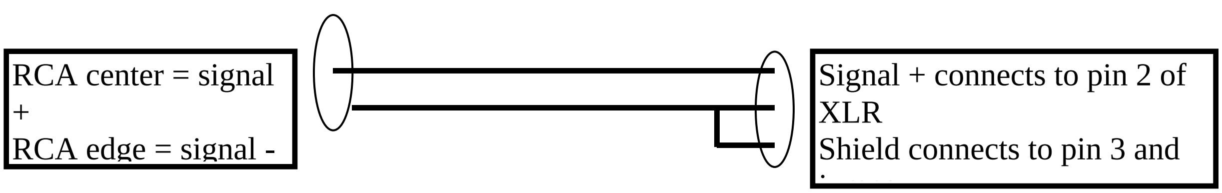
The balanced world (XLR- type connectors) is the easiest to deal with because the audio signals and shield are treated separately down the cable. Properly wired systems that use balanced interfaces can achieve impressive signal to noise ratios with the least amount of bother. Unbalanced equipment (RCA and 1/4” type connectors) can be made to perform very well but sometimes requires more effort because the signal reference (low side, or ground) and the cable shield (and usually, the chassis) of such equipment is shared. This sharing can cause hum or buzzing problems for several different reasons that can usually be remedied with some logical thinking and judicious soldering.

The 2-Bus inputs are designed to make the unbalanced interface easy to pull off properly if some simple rules are followed, but first, let’s explore the balanced input interface.



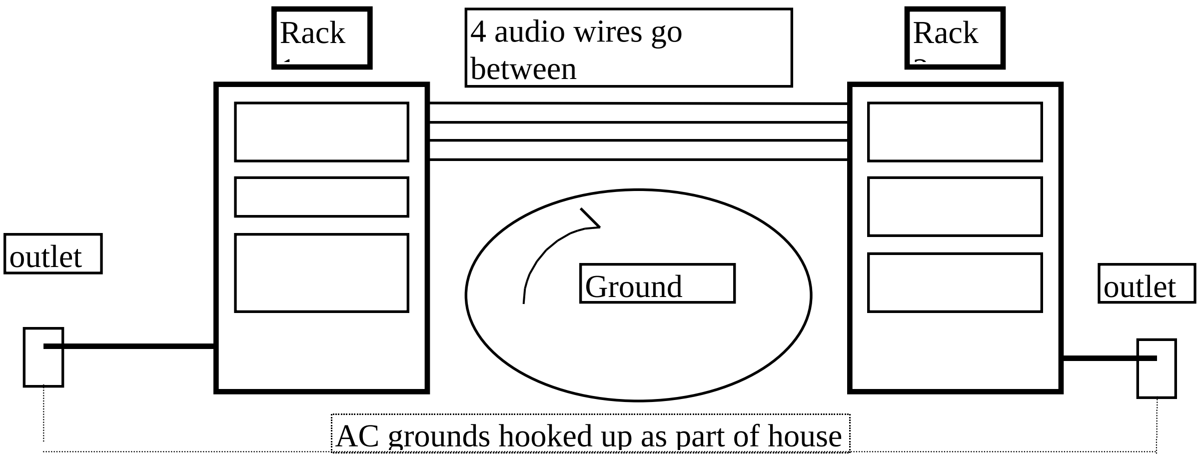
Standard mic cables are used to plug the D/A into the 2-Bus input. Shields (pin 1) should <u>not</u> be lifted as this is done at the appropriate place inside the 2-Bus to keep the input cable shield from conducting ground current. This is the beauty of the balanced connection. Pins 2&3 carry the signal across them (transverse mode) and noise that gets through the shield is picked up equally by both signal conductors (if they are a twisted pair). This “common mode” noise is canceled by the differential action (subtraction) of the instrumentation amplifier in the first stage of the 2-Bus. Since the shield is not connected at both ends, current does not flow down the shield wire and no ground loop results from this interface. Audio goes through, noise is canceled, and grounds stay inside their respective pieces of gear. Beauty exists.

An unbalanced D/A driving a 2-Bus presents no problem due to the differential action of the input stage.



These two scenarios cover the receiving side of the 2-Bus. The driving side uses the same principles but optimum system performance usually requires the use of the dreaded voltmeter plus possibly a soldering gun. Don’t worry, read on. Let’s look at the scenario where a 2-Bus drives the balanced input of an A/D converter (either on the DAT machine, or a separate component).

Equipment manufacturers are required by CE standards to connect pin 1 of an input XLR to chassis ground at the connector in order to get the coveted sticker that lets them sell gear in Europe. This can cause ground loops if the shield wire is allowed to hook both ends of the cable up. The problem stems from the fact that two grounds in a system are never at the same potential. They can be close if the two pieces if gear in question are in the same rack or a heavy gage wire is used to bolt both the chassis together. Some people in desperation resort to using AC plug “ground lifts” to defeat the safety grounds (the third pin on an AC cord) in a random fashion until the system quiets down. This in our view is an unacceptable method of curing ground loop buzzes or hums. The diagram illustrates the problem and a solution follows.



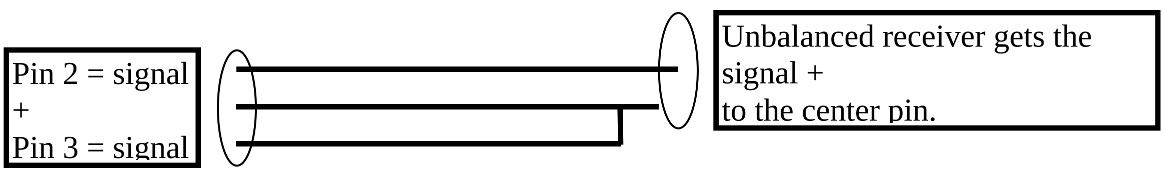
If the audio cables between the racks connect the grounds together via the shield pins, and the racks are at even slightly different potentials (on different circuits, one draws heavy juice with amplifiers, long distance from each other, etc.) the shield of the audio cable will try to equalize the potential difference between the two racks.

Juice will flow down the shield of the audio cable and broadcast hum into the signal wires the shield was supposed to protect. This situation manifests itself as the all too familiar buzz of a ground loop. The intensity depends on many variables but can go from unnoticeable to raging. The simple way to avoid this problem is take the voltmeter and switch it to measure continuity (the unit beeps when the leads are touched together).

Disconnect the cables from the back of the A/D and put one of the leads into pin 1 of the input XLR connector. Touch the other lead to the AC ground pin or a convenient chassis screw and see if the VOM beeps. If it does, then pin 1 is grounded and the shield of the interconnecting cable should be unsoldered and taped (so as not to inadvertently short to the shell of the XLR) at the male connector. Some equipment manufacturers provide a header inside the gear to lift the pin 1 connection. In this case the VOM in the above test would not beep and the audio cable can be left alone. This test can be performed on most balanced line level interfaces to clear up ground loops without applying AC ground lift adapters (which are generally unsafe to use unless the gear in question is grounded by some other method).

If you are unfamiliar with VOM and soldering techniques, this whole procedure can be avoided and the system just hooked up to hope for the best. In small installations where there are no great distances for signals to travel or everything is in the same rack, resulting ground currents cause hum below the desired noise floor. Indications on the A/D meters of greater than -70 dB (-60dB? you be the judge) with no music going should be cause to pick up the phone and get some help from a friend who knows how to use a VOM. It's really not that hard and the extra work can make a big difference in the quality of your mixes. The self noise of a Dangerous 2-Bus is better than -80dB in a bandwidth of 22Hz-22kHz and with the A/D aligned at -14dBfs for an input of +4dBu provides a better than 94dB dynamic range for clean, quiet, punchy recording, yeah baby.

Driving an unbalanced input of an A/D is done by using balanced cable at the 2-Bus end and tying pin 1&3 together at the A/D end and applying that connection to the shell of the RCA connector as in the following diagram.



It may be necessary to bolt the A/D in the same rack as the 2-Bus to obtain optimum performance. This is the potential downside of unbalanced equipment, but this method can be effective. Isolating the A/D from the rack would be the next step. Using an AC ground lift adapter and letting the offending A/D (or DAT machine) get its ground down the signal line would be the last resort although this technique can still work OK. Some DAT decks with unbalanced inputs have two pronged power cords and electrically isolating them from the rack and providing ground down the audio cable is a situation that has worked well and is really functionally equivalent to the last resort mentioned above. Use the VOM set to continuity mode to check isolation from the rack (with all the cables unplugged from the deck while testing)

Volts, Decibels, Bits (16), and Bits (24)

The voltage list is for a sine wave AC measured root mean squared (the way a volt meter would). The numbers are rounded off so a direct comparison to a calculation could be slightly different. Professional levels in a studio are referenced to 1.228VACRMS. This is the voltage out of a console when a VU meter says “0dB”. The dB scale is a logarithmic scale that is easier to deal with in a working world of audio than a voltage scale because loudness in hearing also follows a log scale. 6dB is twice the voltage.10dB is roughly twice the loudness. 20dB is 10 times the voltage. The 16 bits column represents the number of bits on a CD or the number of bits that it takes to represent the voltage at a given level. The 24 bit column represents how a theoretical 24 bit converter would represent the given voltage. I don’t know of any microphone preamp, console, or A/D converter that can deliver 144dB of dynamic range but perhaps it is good to represent the noise floor with several bits. It is certainly not a bad idea to process digital audio information with at least a 24 bit system if number manipulation (DSP) is taking place.

Volts	dBu	(16)Bits	(24)Bits
6.20	+18	16	24
3.10	+12	15	23
1.55	+6	14	22
1.228	**+4**	**13+**	**21+**
0.775	0dBu	13	21
0.3875	-6	12	20
0.245	**-10**	**11+**	**19+**
0.1938	-12	11	19
0.0969	-18	10	18
0.04844	-24	9	17
0.02422	-30	8	16
0.01211	-36	7	15
0.00605	-42	6	14
0.00303	-48	5	13
0.00151	-54	4	12
0.000757	-60	3	11
0.000378	-66	2	10
0.000189	-72	1	9
0.000095	-78 dither		8
0.000047	-84		7
0.000024	-90		6
0.000012	-96		5
0.000006	-102		4
0.000003	-108		3
0.0000014	-114		2
0.0000007	-120		1
	-126 dither		

In a digital system that uses binary counting (PCM being popular) every bit doubles the number of the possible voltages that can be present at the output of a D/A converter. A 16 bit number represents 65536 different possible voltages in a PCM converter assuming no DC offset and a noise floor at the least significant bit. The smallest step is roughly 94uV. Stack 65536 90uV steps on top of one another and you get the 6.2 volt maximum. The step size for a 24 bit word under these same conditions is about .7uV, a small step indeed. The point of this exercise is to illuminate the relationship between voltage, dB’s, and bits in a PCM system.

2-Bus Specifications

Frequency Response ....1 Hz-100kHz within 0.2dB
Total Harmonic Distortion.................0.005% in audio band
Intermodulation Distortion.................0.005% IMD60 4:1
Crosstalk @ 1kHz ....-97dB
Crosstalk @ 10kHz ....-91dB
Noise floor ....-80dBu total energy in audio band
Max level ....+26dBu
Nominal operating level.....................+4dBu (1.228 volts)
Input impedance.................................25kohm balanced
Output impedance ....<20 ohms (600 ohm drive capable)
Gain accuracy.....................................better than .05dB @ 1kHz for any gain setting
Power consumption............................20 watts
Warranty ....2 year labor, 2 years parts. Subject to inspection. Does not include shipping damage or abusive operation

