

Gassmann Studio Tutorial

This document discusses some scenarios for using the Gassmann Studio successfully, and some troubleshooting tips for when things are not so successful. This document has no table of contents or index; it's meant to be read online, reading straight through, or using the "Find..." command to find the needed topics.

I. Policies

Read the document titled "Gassmann Policies". One point in that document bears repeating here. **DON'T UNPLUG ANYTHING!!** All AC power cables, analog audio cables, digital audio cables, computer cables, and any other sort of cables are already plugged in where they are supposed to be. Don't change them, even if you think you know what you're doing. This can cause major malfunctions and loss of much time tracking down the problem. There is no reason to change any of the cables. The only place you should do any plugging or unplugging is at the analog audio patch bay, or at one of a very few other breakout points (e.g., the digital audio and MIDI breakout jacks on the MIDI rack).

II. Analog and Digital Audio

Audio signal can be transmitted in two ways: as an electrical signal "analog" of the sound itself or as a "digital" encoding of a binary representation of the sound. Nearly all the audio devices in the studio have analog sound output, and many (most) have digital output as well.

To record sound, the analog signal from a microphone must be converted to a digital signal. To hear sound, the digital signal must be converted back to an analog signal. In general, your sound quality will be highest if you keep the sound in digital form (except for listening to it, of course). See below for a list of devices that have digital inputs and outputs.

A. Analog signal

The analog inputs and outputs of every device in the studio are wired to the analog patch bay. If you want to route an analog audio signal from one device to another, you only need to connect the output of one device to the input of another using the red longframe patch cords.

Important Note: The analog patch bay jacks are for "longframe" plugs such as those on the red patch cords. They are NOT 1/4" phone jacks such as are found on most audio equipment. **DO NOT** plug anything into the patch bay jacks except the red longframe patch cords. Do not plug headphones directly into the patch bay, and do not plug any 1/4" phone plugs (guitar or instrument cables) directly into the patch bay.

If you have a piece of personal equipment you have brought into the studio, and you want to plug it into the studio, use the jacks that have been provided in the breakout panel just below the analog patch bay. There are 4 XLR jacks, 4 1/4" phone jacks, and 4 RCA jacks. These three sets of 4 jacks are each wired to the patch points labeled "Ext tie 1-4". So, for example, if you wish to plug in a pair of RCA plugs, plug them into the two leftmost RCA jacks in the breakout panel, and the signal will then appear at Ext tie 1 and Ext tie 2, and you can then patch those to whatever other device you want in the patch bay.

In the patch bay there are seven racks of patch points. Each of the patch points along the top of a rack represents the audio output of a device, and each of the patch points along the bottom of the rack represents the audio input of a device. So, for example in the top rack, in the top row of patch points, the first two jacks are the main left and right outputs of the Yamaha Motif8 synthesizer, and the two patch points just below that are the inputs 1 and 2 of the Ramsa DA-7 mixer.

The patch bays are *normalled*. This means that each output jack is internally connected to the input jack just below it. When there are no patch cords in a particular jack, the signal flows directly from output to input without a patch cord being needed. When a patch cord is inserted, that normal connection is disconnected. So, using the prior example, the main outputs of the Motif8 are already connected to the inputs 1 and 2 of the DA-7 mixer, so you don't need to make that connection explicitly with patch cords.

One *exception* to this normaling scheme is the sixth rack in the patch bay. This rack is NOT normalled.

III. Recording

A. Tape recording and hard disk recording

There are several recording devices in the studio: two stereo digital audio tape (DAT) recorders (the Tascam DA-30mkII and the Panasonic SV-3700), one 8-track DAT recorder (the Tascam DA-88), the Macintosh computer, the 192 interface (the hardware interface that goes to the computer and to the applications Pro Tools, Digital Performer, Max/MSP, etc.), a mini-disc recorder (the Tascam MD-301mkII) and a cassette recorder (the Tascam 112). All of those devices, except for the cassette recorder, are digital recording devices.

For mono or stereo recording, the Tascam DAT and the ProTools application might be thought of as the "normal" recording methods. Since computers can and do crash, it is highly recommended that you record on a DAT for backup even when using the computer as your main recording device. Alternatively (and if you have a little more time), you can record on DAT, then transfer your data to the computer in digital form

with no loss of quality. Even if your desired final output format is cassette or mini-disc (which provide lower quality than the other formats in the studio), you might want to consider recording in high quality first, then transferring your recording to the lesser output format only as a final step.

After a discussion of how to get the audio signal, we'll discuss how to record it.

B. Microphones

There are two dynamic microphones (AKG D2000S), two mic clips, two mic stands, and two mic cables supplied in the cabinet in the studio lobby. These mics are decent cardioid dynamic mics, but not of exceptional quality. They're OK for recording voices and instruments that provide a strong source signal without sudden dynamic plosives. For better quality mics, you can check out two AKG C1000S condenser mics from the Arts Media Center. The Gassmann and the Arts Media Center also possess other high quality microphones. If you think you need a special application mic, consult with AMC director Ross Whitney. The cabinet also contains a plosive screen for voice recording and a pair of headphones for monitoring.

The expected recording location is the room next door to the Gassmann Studio, MOB 19. A key to MOB 19 is in the cabinet in the studio lobby. To record in this room you will need at least one person to be the control room technician (in the Gassmann Studio) in addition to the recording artist(s). If you want to record yourself, and you have nobody to act as technician, it is also possible to record right in the Gassmann, but you will want to turn off as many of the various fans as possible: the room ventilation fans, the rack ventilation fan (the red switch above the main power switch), the computer and the external hard disk drives.

C. Recording procedure

To make a standard stereo recording, here is the recommended procedure:

- 1) Set up two microphones in the desired location in MOB 19. Plug them into the jacks on the wall marked Mic 1 and Mic 2. (If you're recording in the Gassmann Studio itself, plug the mics into the first two XLR jacks on the breakout panel underneath the analog audio patch bay.) Plug the headphones into one of the output jacks of the Tascam headphone amplifier so the artist can hear instructions from the technician or any pre-recorded playback.

- 2) In the Gassmann, turn on the monitoring camera, using the wall switch labeled "Camera". Then turn on the RCA video monitor. (It will take a moment to cycle through all of its channels. Just let it do that.) Now you should see MOB 19 on the monitor. If not, check that the RF modulator is set to "Channel 3" and "Mod".

3) The microphone signals going into the two MOB 19 wall jacks marked “Mic 1” and “Mic 2” appear in the patch bay (bottom rack) as the first two output jacks, labeled “Mic tie 1” and “Mic tie 2”. Note that those jacks are normalled to the t.c. electronic Gold Channel microphone pre-amplifier. This means that the audio signals from your two mics will go straight to the Gold Channel by default. (Handy, no?) The Gold Channel has the best quality preamplification and analog-to-digital conversion in the studio, so this is usually the best way to go. The alternative is to patch your microphone signal to one of the first 8 channels of the DA-7 mixer, which also has high quality preamplification and conversion.

4) On the Gold Channel, make sure the Input for each of the two channels is set to “Mic” (rather than “Line” or “Digital”). If you are using condenser microphones (such as the AKG C1000S), turn 48v phantom power on for each channel. If you are using dynamic microphones (such as the AKG D2000S), turn 48v phantom power off. Test the input levels by having the artist(s) play their loudest sounds into the microphones. Adjust the level using the Pad switches (to adjust the level in large 20 dB increments) and the level knobs and (for finer adjustments) until the loudest source sounds show just under 0 dB (maximum) on the input monitor LEDs. (Make sure the monitor LEDs are set to show the input signal, not the output signal.) The different Gold Channel presets have particular settings for equalization and compression of the incoming signal. Read the manual for more information on compression, EQ, and presets. Select the preset you want, or design your own and store it.

5) Now that the signal has been converted to 24-bit digital form by the Gold Channel, you need to route the digital signal to the proper locations using the Z-Sys Router software. A detailed explanation of how to do this merits a tutorial of its own, so see the section below on Digital Audio Routing. For now we will assume that you have read that. Open the RouterControl application from the Dock (or from the “Applications (Mac OS 9)” folder). This will open the Router software, a virtual audio patchbay. (Since it’s an OS 9 application, the Classic mode will begin to run.) In the RouterControl application, choose Open... from the File menu, and open the “DigitalAudioSetup “ file in the /Users/Shared/ folder. In the router matrix, assign the source Gold Channel to the destination ProTools 1&2, also assign it to the (Tascam) DA-30mkII DAT machine, and also assign it to channels DA7 25&26 of the mixer so you can hear the output of the Gold Channel directly. Assign source ProTools 1&2 to destination DA7 17&18, ProTools 3&4 to DA7 19&20, and so on, so you can hear the output of ProTools. Assign the source DA-30mkII to destination DA7 27&28 to hear the output of the DAT recorder. (Its analog outputs are already normalled to inputs 13&14 of the DA-7, so that will work just fine, too.) After you have made those connections in the software, send the configuration to the Router (command-R), then quit the application.

6) Put a DAT tape in the DA-30mkII. Make sure the input is switched to “Digital” and “AES/EBU”. Put the deck in “record” and pause the tape. The digital output signal from

the Gold Channel should be shown identically in the LED signal indicator of the DA-30mkII.

7) Open the Pro Tools application (from the Dock). Create two new audio tracks. By default, they will receive their input from channels 1 and 2 of the 192 and will send their output to channels 1 and 2 of the 192. Choose the Hardware... command from the Setups menu and ensure that the input preference for channels 1 & 2 is set to "Digital" (not analog) and that the clock source is set to "Internal". This will cause it to get its signal in digital form, and slave itself to the Word clock generator. If the digital outputs of ProTools are set to 9-16 (the default), you will want to change the output bus of your audio tracks 1 and 2 to go to outs 9-10. Now click on the "rec" button in each of the two audio tracks. If all is correct, you should see the signal in the virtual LEDs identical to the output signal of the Gold Channel.

8) In the DA-7 mixer, you should see signal coming in on channels 17&18 (from Pro Tools), 25 & 26 (from the Gold Channel), 27 & 28 (from the Tascam DAT), and 13 & 14 (analog from the Gold Channel). Route whatever channels you want to hear to the L&R bus, route the L&R bus to Monitor A, and turn up the Monitor A pot to hear the sound. (If you're in doubt how to do any of that, read the DA-7 manual.) In this case, the DA-7 signal is in digital form before it even gets to the DA-7, so for best results the DA-7 should be slaved to the clock of the incoming digital signal. So, press the Dig I/O button on the DA-7 to call up the clock source window. The clock source should be set to "44.1 Internal".

Troubleshooting: If you're not getting signal where you're supposed to be getting it—that is, if steps 6, 7, and 8 above are not working because the signal is not arriving where it should—here are some things you can try.

- a) Redo step 5 above, and make sure you have routed the digital signals correctly.
- b) On the Gold Channel, make sure that the digital signal format is set to "AES/EBU" (not S/PDIF).
- c) On the Gold Channel, make sure that the preset you are using has the Aux Send/Return set to "None".
- d) Review the instructions in steps 6-8 to ensure that you've followed them correctly.

9) To communicate with the artist in MOB 19, press the "Talkback" button on the mixer and speak. The level of the talkback signal you send will be determined by the level you set with the pot for Monitor B output on the mixer. Monitor B L&R outputs are normalised to the Cue Line L&R jacks in the patch bay. Therefore, the Monitor B outputs should be audible to the artist via headphones plugged into the Tascam headphone amplifier in MOB 19.

10) Once you have set the levels the way you want, start (unpause) the DAT in record mode, click on the record and start buttons in Pro Tools, and cue the artist to begin.

When the recording is finished, stop the DAT and ProTools, listen to the results to ensure that no unwanted noise is included in the recording.

D. Digital Audio Routing

All equipment that sends or receive digital audio signal does so via the Z-Sys 32-32r digital audio router. The Z-Sys performs the needed format conversions between S/PDIF and AES/EBU formats, but you still might need to know what format each individual device uses, just to ensure that the device itself is set to send and receive digital audio in the correct format. Here is a chart of digital audio devices and their digital formats used in the studio. (Note that the order of devices in this list corresponds to the order in which they are patched to the Z-Sys ports, just for convenience of reference.)

<u>Device</u>	<u>Format</u>
Pro Tools 1&2*	AES/EBU
Pro Tools 3&4*	AES/EBU
Pro Tools 5&6*	AES/EBU
Pro Tools 7&8*	AES/EBU
DA-88 1&2*	AES/EBU
DA-88 3&4*	AES/EBU
DA-88 5&6*	AES/EBU
DA-88 7&8*	AES/EBU
DA-7 17&18 (Slot 1 1&2)**	AES/EBU
DA-7 19&20 (Slot 1 3&4)**	AES/EBU
DA-7 21&22 (Slot 1 5&6)**	AES/EBU
DA-7 23&24 (Slot 1 7&8)**	AES/EBU
DA-7 25&26 (Slot 2 1&2)**	AES/EBU
DA-7 27&28 (Slot 2 3&4)**	AES/EBU
DA-7 29&30 (Slot 2 5&6)**	AES/EBU
DA-7 31&32 (Slot 2 7&8)**	AES/EBU
DA-7 Main Out L&R (Rec In L&R)	AES/EBU
DA-30mkII	AES/EBU
SV-3700	AES/EBU
Gold Channel	AES/EBU
K2000	AES/EBU
External breakout panel XLR jacks	AES/EBU
DA-7 Rec out	S/PDIF
Motif8	S/PDIF (output only)
MD-301mkII	S/PDIF
PCM80	S/PDIF
M300	S/PDIF
Marantz CD	S/PDIF (output only)
External breakout panel RCA jacks	S/PDIF

*The listing “Pro Tools” in the above chart, in the Z-Sys RouterControl software, and in the analog patch bay actually refers to the Digidesign 192 hardware interface. Since this interface is most commonly accessed via the ProTools software, in common parlance we say that we’re routing the audio to or from “Pro Tools”. In fact, though, we’re really routing the audio to or from the 192, which can be accessed by several applications on the computer, including Pro Tools, Digital Performer, Max/MSP, and Reason.

**DA-7 input channels 17-24 are via AES/EBU card Slot 1 on the mixer, and DA-7 input channels 25-32 are via AES/EBU card Slot 2 on the mixer. The mixer signal that gets sent to the outputs of each slot are configurable in software in the mixer (in its Digital I/O window), but by default we have set each of the slots to send out the signal assigned to Bus outputs 1-8. So, in the Z-Sys routing software, what’s listed as DA7 Slot 1 1&2 will normally mean the signal that is being sent to Bus 1 and Bus 2 in the mixer.

The “DigitalAudioSetup” document for the Z-Sys RouterControl software (which you can open easily by choosing RouterControl from the Dock, then opening the file “DigitalAudioSetup” in the /Users/Shared/ folder) is set up as a matrix of sources and destinations, like a virtual patch bay. The sources of digital signal are listed along the left side, and the destinations are listed along the top. You can route one source to as many destinations as you want, but a destination can only receive from one source. To route a source to a destination, click on the name of the source and the name of the destination you want, then type command-M to make the connection. When you have made all the connections you want, type command-R to send those connections to the router. Here is a more thorough step-by step instruction.

1) Plan mentally or on paper how you want the digital audio signal flow to be constructed. For example, if you want to apply an effect to your synthesizer sound, then record it in ProTools, you might say to yourself, “I want the output of the Motif8 to go to the inputs of the PCM80 effects processor, and I want the output of the PCM80 to go to inputs 1&2 of ProTools, and I want digital outputs 1&2 of ProTools to go to channels 17&18 of the DA7 mixer.”

2) Open the RouterControl application from the Dock (or from the “Applications (Mac OS 9)” folder). This will open the Router software, a virtual audio patchbay. (Since it’s an OS 9 application, the Classic mode will begin to run.) In the RouterControl application, choose Open... from the File menu, and open the “DigitalAudioSetup “ file in the /Users/Shared/ folder.

3) In the router software, assign the sources to the proper destinations. To do this, you click on a source, click on a destination, and type command-M to make the connection. Do this for as many sources and destinations as you want to connect. If you wanted to do the routings described in Step 1, you might make the following connections: Motif8

>> PCM80, PCM80 >> ProTools 1&2, ProTools 1&2 >> DA7 17&18 (Slot 1 1&2). Note: In this example we have assigned the output of ProTools to the mixer in order to hear the sound once it has been recorded. If you're routing the digital output of ProTools to an input of the DA-7, then you must be sure that in the mixer window for those channels you don't route that sound to some outputs of the mixer that you're sending to ProTools while you have record enabled in ProTools!

4) Once you have made the source-destination connections you want, type command-R to send the configuration to the router. If the connections show up with black dots in them in the software, that means the router has received and implemented those connections. Quit the application.

5) Now you must also be sure you're routing the signal correctly within the DA-7 mixer. This is done by selecting the destinations you want in the mixer window for each channel you're using. In channels 17 and 18 of the mixer (where ProTools is coming in) you should set the destination as L&R, with channel 17 panned full left and channel 18 panned full right.

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Here are some other topics that should be covered in future tutorials. If you think you understand the topic well, you're invited to create a Tutorial document of your own to explain the topic, and put that document in the Users:gassmann:Documents folder.

- playing back a notated score
- recording a MIDI sequence
- adding reverb to a recording
- transferring sound from a tape to a computer file, and then to a CD
- why can't I hear a sound?