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DESIGN CONSIDERATIONS

PART II

●In this issue of **db**, we will complete the speculative odyssey that was begun in the July/August edition. Our goal is to continue exploring some of the most useful and cost-efficient methods of constructing (or upgrading) an electronic cottage: that such a facility might exhibit a great ease of operation, and be capable of turning out unquestionably professional product. Having divided this quest into four areas of concern, we previously dealt with the foundational considerations—*room layout* and *electricity*—in the last edition. Now let's put it all together by discussing the areas of *interconnections* and *monitoring* in the electronic cottage.

INTERCONNECTIONS

Let the reader beware from the outset: getting highest quality audio from the electronic cottage can sometimes be an arduous journey, but it is indeed possible. First, let's remind ourselves that we are working within the limitations of "unbalanced" audio, which is justly considered to be a less-than-professional standard. This stigma has almost nothing to say about the inherent quality of the device itself, but says much about what happens when we connect several devices together.

Whatever degradation in sound quality that occurs in an unbalanced system (beyond the practical S/N ratio of the mixer, outboard gear, etc.) will be primarily induced in the wiring that hooks them all together. If spurious audio (EMI or RFI) enters the signal path at any point, the unbalanced system is helpless to eliminate it. The balanced audio system, however, can factor out much of this trash even after it has invaded the system, rendering clean audio under circumstances where unbalanced systems would positively freak out. But I reiterate: it is possible to get high quality, trash-free audio from an unbalanced system. We simply need to work harder in order to get it!

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The key features of the balanced system that gives it such a high degree of immunity to interference are:

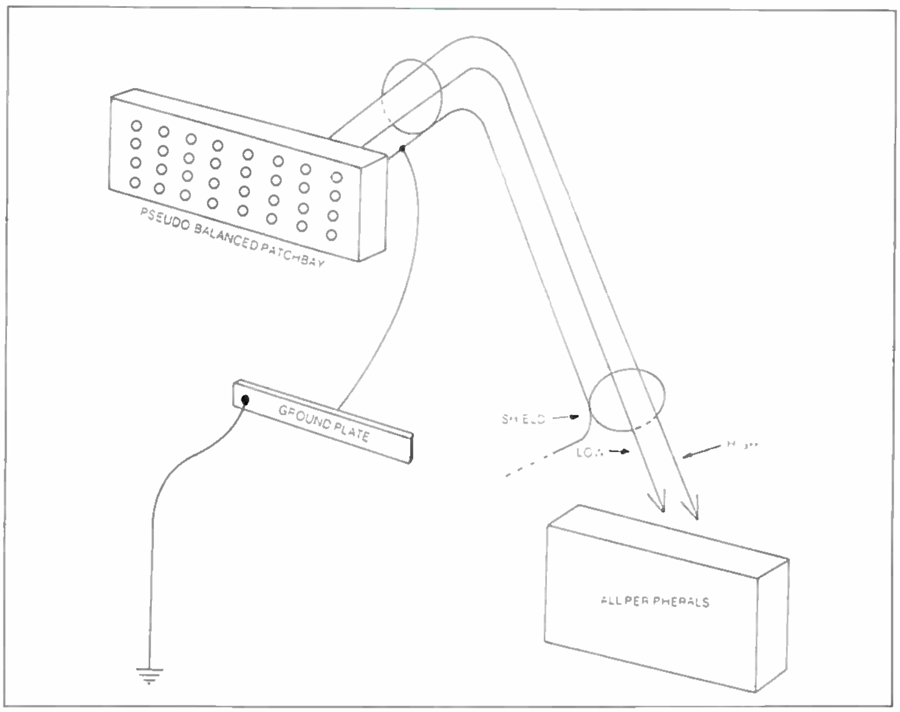
1) the differential input

2) the use of three-wire cabling (i.e., two conductors plus a shield)

3) higher nominal operating levels.

The idea here is that audio information is conducted from one device to the next down two conductors, neither of which is referenced to ground. The third lead, which is a shield (and attached to ground only at one end of the cable) is then able to intercept a good deal of extraneous noise from the environment and shunt it to ground. If any noise does makes it beyond the shield, there is a second line of defense—the differential input. It discriminates against any signal that does not bear the usual phase relationship (one side high [+], one side low [−]) at

Figure 1. The pseudo-balanced patchbay concept.



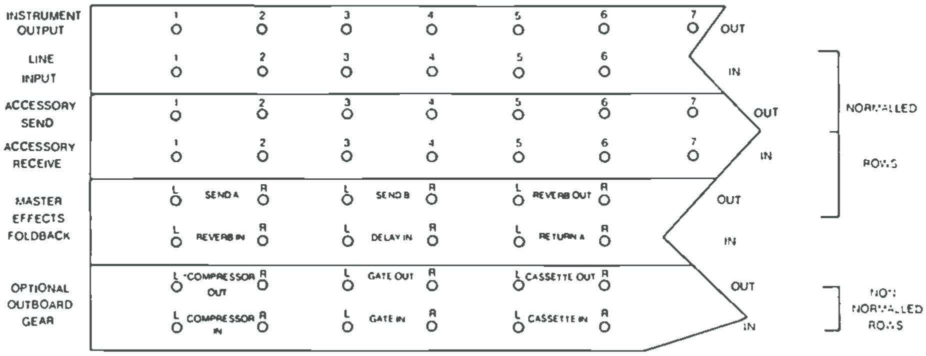


Figure 2. A sample patch bay concept

a given point in time. Recognizing only pairs of opposites, it effectively cancels any invasive signals, since they have hit both conductors at virtually the same time. This is definitely a neat trick, but for various reasons (increased cost, for one) it has been reserved for strictly pro equipment.

The higher operating level also helps out here in achieving a clean signal. Instead of using a nominal –10 dBv as most unbalanced gear does, balanced equipment sports the elevated level of +4 dBm. If extraneous sound does find it's way beyond the first two lines of defense, the elevated audio signal can make the noise seem rather unimportant in proportion to the signal, essentially masking the noise by brute force.

Now the purpose of this little side tour was not simply to extol the virtues of balanced equipment. Most electronic cottageers do not use balanced equipment. (Even if, for example, your recording console offers a +4 balanced master output, and you are recording into an unbalanced input on your mixdown machine, well good for you! You have an elevated output, but the other characteristics of a balanced system are not operating.) The point of this is to see if any of the very obvious virtues of a balanced system can be incorporated into the unbalanced studio. While it is plainly obvious that short of buying an interface amplifier for every piece of gear in the house, an unbalanced system will never function as a balanced system. But two of the three previously mentioned characteristics can actually be utilized in an unbalanced studio to improve the rejection of extraneous noise. The key to achieving this maxxed-out performance is found in a method of wiring up your studio patchbay.

Certain advantages can be gained from using a balanced type (three-wire) patch-bay in an unbalanced system. While it takes a little more forethought to integrate this into the elegant simplicity of the unbalanced (two-wire) system, in the end, it will prove a very worthwhile investment of your time.

One thing that needs to be determined before you fire up your soldering iron is the layout of your patchbay.

The theory goes something like this: Normally, in the unbalanced configuration, two-wires serve three purposes. One wire is designated as high signal carrier, the other serves as both the low signal carrier *and* shield. While this does idiot-proof the task of interconnecting equipment (neatly avoiding all ground-looping or phase problems), it also greatly compromises the unique purpose of the shielding: to intercept and divert spurious electrical currents. So if we can separate the signal carriers from the shield, it is possible to afford an extra measure of protection against invasive electricity. (Admittedly, this is not identical with the balanced system where none of the signal carriers are referenced to ground, but having a *dedicated* shield hooked into a solid ground can really make a significant improvement. Many electronic cottageers in urban areas—where RFI is extremely high—have found benefit from this technique.)

The underlying concept is very similar to the one we discussed in the last issue of db regarding electrical grounding: In order to provide the most effective drain (and avoid loops), audio shields should have a single, efficient path to mother earth. The pseudo-balanced patchbay provides a central point at which all shields can meet and be integrated into the unified system ground.

The basic procedure is as follows: Considering the patch bay as the central point and *all* devices (console, signal processors, etc.) as peripherals, wire the plugs connecting all peripherals to the patchbay, using only the red and black wires. Leave the drain wire (the shield) *detached at the device*, but make sure all drain wires are *connected at the patchbay*. All of these drain wires are then bussed together at the patchbay and solidly attached by a fat copper wire to the central grounding plate (see the previous issue of **db**).

In most cases, this will work without a hitch, but occasionally there will be a complication. For example, having attached the insertion points (accessory in/out) on my AKAI 12-12 to the patchbay in this manner, I found that when I pushed the "EQ-in" switch a ground loop was formed. It was however, easy enough to iron out this wrinkle by reattaching the shield at one side of the insertion point. (The "in" or the "out," it made no difference.) Fortunately, I was prepared for this by an experienced studio maintenance engineer, who advised that when wiring the plugs, the drain wires should be *detached*, but not *clipped*. Instead they should be insulated in shrink-wrapping and left intact within the shell of the plug, just in case some final jockeying was necessary. Any honest studio technician will tell you that due to differences in design philosophy between various manufacturers, anomalous conditions do crop up in wiring a studio that require a more empirical approach. We always need to be prepared for this eventuality.

PATCHBAY LAYOUT.

One thing that needs to be determined before you fire up your soldering iron is the layout of your patchbay. Naturally, for convenience and efficiency you will probably want to bring every piece of gear, every insertion point, up on patch. You want to be able to patch anything to anything else with just a short patch-cord. Truly, there are as many ways to achieve this goal as there are studio owners. Still there are some rules of thumb, some tried and proven conventions that make this task less confusing.

For example, the concept of *normalling*. Which devices or configurations of equipment will be most consistently used in a standardized way? Only you can answer that. And if your studio is a new one, your answer will probably be speculative and provisional. Nevertheless, you must define the scope of your studio (at least theoretically), out front, or else you will find yourself constantly re-patching really obvious signal paths. If your main thrust is synthesized sound, then it would be well to normal the outputs of the sound modules *through the patchbay* to the appropriate input modules on your recording console.

For this, you will need a patch bay that contains normalled-through patch points. Sometimes, you can assemble them yourselves choosing which pairs of points (upper and lower) will be normalled through and which will be non-normalling (where upper and lower points are isolated from each other). This non-normalling type is preferred for use as tie-lines, where optional equipment is to be patched in. Usually, a combination of these two basic types is best, but some people (of necessity or ignorance) use the common normalling patch bay for everything. In the latter case, if you get feedback from any device being normalled through to itself, you can just stick a "dummy plug" into either the upper or lower point, and the normal will be broken.

The following conventions are also useful for laying out a patchbay that can be comprehended quickly by the user. Your entire patchbay may consist of several rack mountable patch panels, each with two rows of points (of various numbers, commonly 2 pairs of 24). It has been found most efficient to designate the upper row for outputs and the lower row for inputs.

OUTPUT OVER INPUT.

If this convention is consistently followed throughout each panel, your patchbay will be a joy to use. (It will also be a lot harder to do dumb things like patching two outputs together).

Another worthy convention is to somehow approximate the signal flow in vertical organization of the patch bay. Usually, this would mean having "earliest" points in the signal flow of your recording console appear at the top panel and subsequent points (like the various insertion points) on lower rungs of the patch-bay. If this is not feasible in every case, then let it be a left to right movement first, then moving to the next lower rung. (See example) This of course, is harmonious with the Western mind. In any case, try to roughly map the signal flow of the console in the initial several rows. The next

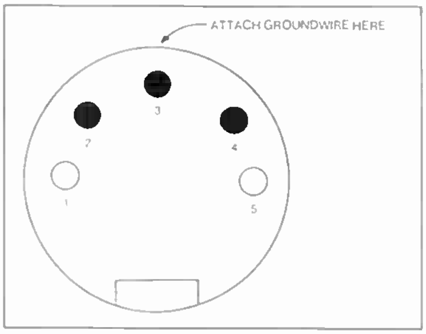


Figure 3. A do-it-yourself MIDI cable using studio hookup wire.

rows are best reserved for auxiliary send/receive circuits and their normal destinations, (such as Send A Out normalled through to your "Usual Ambience Device"). Finally, every piece of non-normalled outboard gear, auxiliary recorders, etc. should appear on patch. It is also good to have a few blank patch points wired in, terminating in a box with 1/4-inch jacks, so that the inevitable piece of borrowed or rented gear can be used to full advantage.

PRACTICAL WIRING TIPS.

There are many lessons to be learned about studio wiring, and the only way to really learn them is by doing them. Still, there are a few sensible pointers that will save many a costly mistake. So without any pretense to comprehensiveness, I offer these few aphorisms, anecdotes and personal recommendations, based on my experience putting together an electronic cottage.

As to wiring, never underestimate how much cable you will need.

Start with a good soldering iron. A rinky-dink iron will really hold you back, because it takes so long to heat up, and temperature sometimes varies with usage. All this impedes the rhythm of flowing on solder in a productive fashion. Never underestimate how many connections you will need to make. (In my case: a twelve-track studio with a dozen or so signal processors and 144 points up on patch, required over 1500 solder joints. Happily, there was not even one cold solder joint to be found amongst them!) A good iron is therefore, a worthy investment.

Also, take some time to keep the iron clean and well tinned. A corroded tip will never heat up quite right, so when you shut down the iron, make sure there is a good fresh gob of solder on it to protect it until the next time you fire it up. If the tip eventually gets funky anyway, just dip the (hot) tip in some flux and rub it on some fine steel wool to clean it.

As to wiring, never underestimate how much cable you will need. (In my case: the above-mentioned twelve-track studio residing in a 12-ft.x 12-ft. room, required nearly 2000 feet of wire.) Understandably then, unless you are an audiophile with a rich uncle, you will have to find a cost-effective, reasonably low-capacitance cable. Now, a few years ago I had sojourned on a wiring crew for a short period of time. The crew chief used West Penn 291 exclusively at a 24-track installation, because of the need to come in on a tight budget. Some of the staff engineers spoke derisively about this, saying it was better used for wiring telephones than pro-audio facilities. (That's really a dumb statement, because the entire legacy of pro-audio comes directly from Ma Bell.) Today, West Penn 291 is considered quite respectable stuff, showing up in the wiring troughs of many professional facilities. And at current retail prices of about $70 for a 1000 ft. reel, it's a secret that's worth sharing.

Another good investment for the would-be wirer is the shrink-rap gun. It's really nothing more than glorified hair dryer that's used to apply very hot air to slightly over-sized plastic tubing, thereby shrinking it to a snug fit. It's a tool of many uses, all of them tending to help render a more stable installation. For example, drain wires on studio hookup cable are uninsulated. After all, they are required to be in constant contact with the aluminum foil shield. At termination points (like plug ends and patch bay connections, bare wire, if accidentally twisted out of position, could hit a signal bearing lead and short it out. To ensure against that kind of trouble, all drain wires should be shrink-wrapped with the appropriate size plastic tubing. Another important use for larger shrink-rap is a permanent lamination for any cable markers.

These markers, usually found in the form of a sticky-backed tape should be utilized as a numbering system for the purposes of studio documentation. Each cable should be marked at both ends with an identifying number, and then sealed up with shrink-tubing. Such scrupulous documentation will prove of immense value if any part of your system should go down or need to be updated.

It is worthy to note, that MIDI cables can also be made from standard studio hook-up cable. While MIDI cables utilize a five-pin DIN connector, only three of the leads carry any information, at this time. So for the price of some good hardware and just a few pennies for wire, you can build your own MIDI cables, cut to exact size for your installation. To do this, you simply connect the ground (drain wire) to the center pin, and the red and black to the pins on either side of center pin. Remember *not* to attach the ground to the casing as you would a standard DIN connector, but to the center pin only.

A word about cable length. We know that there are discrete limits to how far you can push a signal in an unbalanced system without noticeably degrading it. While it varies a bit with the quality of the cable you use, practically speaking, anything beyond 25 feet will result in a perceptible loss of high frequencies. So the wise person will endeavor to keep cable lengths as short as possible. But don't get too neurotic about this. You'll need to leave about 18 inches of slack at the rear of each device, so that you can pull a piece away from the rack (to test it or do maintenance, without dismantling your whole studio. If you keep your equipment centrally located, your cable runs will naturally fall in the very acceptable 8- to 10-foot range. Finally, I mentioned that there might be a way of utilizing higher operating levels as a way of keeping your system quiet. This can be approached in two ways. On an extra-ordinary level, one can have modifications done to critical pieces of gear. The master output on my AKAI 12-12 was modified to put out a 6db hotter level, simply by changing the feedback resistor on the master output amplifier. Of course, this is a possibility for other gear, as well. From the more ordinary perspective though, running all "send" circuits hot at possible, and "returns" low as possible can contribute to a better signal-to-noise ratio. Additionally, some signals processors, such as Yamaha's SPX 90, have input/output level switches giving options of –20 dB and +4dB. These sensitivity controls can often be creatively manipulated to improve the gain structure of the effects circuit and hence, the whole system benefits. Once you start thinking in this manner, noticeable improvements can be achieved.

...if you go this far, remember not to restrict the current, by sticking any resistive obstacles in its way.

MONITORING

Monitoring in the electronic cottage bears many things in common with the professional recording studio. There is the need for a clean, flat, power amplifier, accurate speakers, and solid low-resistance interconnections. But beyond this, the similarity ceases. Given that the electronic cottage work space is generally not the realization of an acoustical consultant, but the simple, mundane reality of available space, we can say that many issues the pro-studio owner will fret over, are irrelevancies to us. Not because they are unimportant, but because they are unattainable.

One area of great concern in the acoustically designed room, is accuracy in the low end. State-of-the-art studios go to great expense to achieve this goal, with varying degrees of success. But such noble hair-splitting would be a endless *megilah* in the electronic cottage. So instead, I believe, most of us would opt for establishing electronic accuracy to the point of the speakers (an easy task), and then spending whatever time it takes to establish a genuine intimacy with our environment. After all, we are the "rugged individualists" of the audio industry, aren't we?

As mentioned, accuracy out to the speakers is not hard to establish. We don't need high power or elaborate crossover schemes. A clean stereo power amp of at least 50 watts should be all you need. Remember that mixing in the EC is best done at moderate to low levels in order to keep room resonances to a minimum. The best way to work with an untuned room, is not to work with it at all. Learn to utilize near-field monitors, exclusively. It is a skill that needs to be developed if consistency in mixing is to be achieved.

The issue of monitor speakers in the electronic cottage boils down to a matter of prudent personal preference. My choice for main monitor speakers are TOA 280-MEs. They are three-way sealed enclosure type speakers, with an 8-inch woofers, tweeter, and super-tweeter. They are virtually flat from 20k down to about 90 Hz. (TOA makes a slightly larger monitor the 312-ME which is flat down to about 60 Hz.) I also installed TOA 22-MEs (very similar to the Auratone cubes) as secondaries, and as tertiary speakers, I bought Roland SRS-80s, because they were cheap and provided additional insight from the perspective of your "average" two-way speaker. How do I know what's happening on the low-end when my best monitor is flat only to 90 Hz? Well, it may sound esoteric, but by cross referencing information gleaned from all three speakers, one learns to quite accurately deduce what is going on in the low-end; even in regions below the flatness curve of the monitors. The task is initially spending a lot of time in the experimental mode.

Most of what I said thus far, is purely discretionary. But there are a few hard-core facts I would like to mention. Don't hook up your amp to speakers with anything like "normal" speaker wire. If there was one place in your system where you should go for broke, here it is. Audiophile speaker cable (like Monster and other similar brands) is really quite expensive, but from amp to speakers should be a relatively short run, and it will be worth the expense. It has been shown that power, frequency response, and phase relationships are disturbed by inferior cabling—so go the extra mile. (If finances are really tight, some fat zip cord, 12 gauge or larger, will work decently.)

And if you go this far, remember not to restrict the current, by sticking any resistive obstacles in its way. I am speaking, of course, about things like switch boxes—which are necessary contrivances if you would use multiple speakers with a single amplifier. If you pull one apart, you may find it's built with wire so thin, you could use it to darn your socks! Instead of living with this electrical bottleneck, re-wire the beast with the same wire you chose for your speakers, (or at least the fattest wire that can be soldered onto the lugs in the switch box). The difference this makes will be quite discernable, so it's worth another of evening of fun with your soldering iron.

If you are currently planning or revamping your EC, we hope this series has been helpful to you. Stay tuned for more insights about life in the Electronic Cottage. db