

Remote-controlled preamplifier project: Part one: design notes

Very much a "work in progress" - most of the design has been in my head for several years, only partly realised in the amp that is part of my regular system. I am aiming towards a completely balanced topology, with digital remote control of source selection, channel balance and volume.

This document is really a set of notes for myself, to help me put some ideas in perspective before building the preamp, so it's not going to read like a White Paper! All the same, it may stimulate another reader to the odd idea or two, so I'd welcome any comments or discussion.

Preamble

My aims were to produce a box which would satisfy me in the long term, not only having a minimal sonic signature but also being easy and intuitive to use. In particular, I wanted to do the source and record switching properly first time (see below), and leave some flexibility to experiment with other "drop-in" functions like the output buffer and the volume control later on. It needs to provide as flexible as possible an interface to the rest of the system, partly to give me more freedom in building the system around it, but also to let me make changes in the future without drilling extra holes in the boxes. On top of all this, I wanted it to look good in an inconspicuous way, at the same time as providing a clear visual indication of what it is doing at any time.

As is often stated, a preamplifier has three main purposes in life: to select the input to be connected to the output, to control the listening volume and to carry out whatever amplification or equalisation is necessary for each source. Other functions may be added, for instance inversion of the output polarity, separate source selection for the record output, a headphone output, output muting and the facility to sum the channels to mono. A preamplifier may also provide a small amount of gain on the main output to compensate for power amps of below-average sensitivity, and some high-end preamps provide a choice of gain too, as well as the possibility of programming a different gain for each input.

I am building a self-contained phono stage, partly to reduce the noise, and partly because there won't be space in a single box for everything. For this reason, I can treat each input signal to the preamp more or less equally.

Source selection

I have always intended to do this with sealed relays, which removes the need to route the signal to the front panel and back. What is almost as important is to isolate the earth as well as the signal connections of the source components, and to do this both for the "listen" inputs and for the "record" inputs, if these are independent. Many source components have two-core mains cables, and are incapable of forming a hum loop, but not all.

The other significant point to remember with source switching is to be able to disconnect the record output. The input stage of a cassette deck has (one hopes!) a high impedance while it is switched on, but when powered down the input can present a nonlinear (and not necessarily high) impedance to the source. The record out selector should have a "null" position, which disconnects the output from any sources. The record output should also be buffered, to prevent the load made up of the cable to the recorder and the recorder input from loading down the source.

Input and output buffers

I have used a passive preamp for years, and appreciated its simplicity, indestructability and sonic transparency. Pete Goudreau has an excellent [article](#) listing the advantages of passive preamps (not least their cost and reliability), as well as many of the design considerations. The disadvantage of active preamps include the possibility of distortion at the frequency extremes and the importance of matching the signal level to the available dynamic range. On the other hand, using an output buffer makes the system more flexible, giving a well-defined low, and almost constant, output impedance, and allowing gains greater than unity and also giving the option of balanced outputs. The optimum gain, at any rate, is a system consideration, being determined by

amplifier and speaker sensitivity. In fact, standards exist for these, such as THX (29dB for amplifiers, for instance, and a corresponding specification for speakers), and if these are adopted the onus is put on the preamp to set the gain according to the outputs of the sources and the maximum volume the user wants in a given room.

Whether to use an input buffer is a related, but by no means identical, issue. Many commercial preamps avoid them, mostly because of the cost of doing it properly, though some designers talk about "input overload", which I suspect is irrelevant here (the input signal is generally below three volts peak, while preamp power supplies tend to be at least 10V, and usually 15V). Their chief advantage is that they isolate the output stage of the source component from the rest of the system, with the result that the former sees a high impedance, while the volume control is fed by a low impedance.

There are several choices as far as the implementation of the buffers goes. Several high-end designs use simple emitter-follower (or source- or cathode-follower) circuits with various elaborations such as constant current sources or complementary symmetry to reduce distortion. These restrict the design to unity gain, however (though Goudreau in his article gives a good justification for unity gain preamps). John Linsley Hood uses a neat little complementary buffer circuit in his Electronics Today () preamp design, with a pair of JFETs at the input and emitter followers at the output, with a preset to adjust offset voltage. Elektor (January/February 1997) has an interesting construction article by T. Giesberts for a fully discrete line-level preamp based on a power amplifier-like circuit and a battery power supply.

However, modern op-amps such as Burr-Brown's OPA-604 or Analog Devices' AD797 give vanishing figures of THD and very low noise, and allow more flexibility in design. They are also much less demanding of the power supply, since they generally have power supply rejection ratios of typically 80dB or better across the audio range (though this does decrease at high frequencies). In the end I decided to use an OPA-2604 dual op-amp for each channel for each of the input and output buffers, ensuring that the hot and cold signal lines were physically as close as possible to one another.

Volume control

There are many ways to accomplish this properly, including the following:

- A potentiometer with a long shaft to allow control via the front panel without long signal leads
- As above, but with motor control of shaft rotation, via a clutch to allow the volume to be adjusted by hand.
- A switched attenuator, with a rotary switch selecting resistor networks.
- As above, but using relays rather than the mechanical switch.
- Use a monolithic attenuator such as those from Crystal, Dallas and Analog Devices.

The commonest is the first, which is often used in commercial mid-quality gear. The second allows remote control operation, and most motorised potentiometers are better quality than the default 10K carbon track, but both have the drawback of being strongly reliant on the quality and channel matching of the potentiometer. It is also impossible to obtain repeatable volume settings with either method. A related disadvantage is that it isn't easy to provide a clearly visible and precise indicator of the volume setting with a potentiometer.

My current preamp uses a rotary switch with 12 positions and metal film resistors to give 4dB steps. I use standard cheap switches, which are open to the air and so need cleaning once in a while. Steps of 4dB are definitely too coarse - you need 2dB steps as a minimum requirement, and preferably 1dB. In principle, with high-quality sealed switches with long shafts and plenty of positions, together with closely toleranced resistors, this is a good solution, but it's not ideal.

The cleanest and purest way is with relay switching, but this is pretty expensive and can be mechanically noisy. It also takes up huge amounts of space if you want to have 2dB or less between steps. Balanced working can be awkward to get right, partly because you need to double up the number of relays, but also because the signal paths for the two phases are very hard to get close enough together to get the benefits of balanced operation.

The most elegant solution is the last, using a digital chip, such as the [DS1802](#) from Dallas Semiconductors. This allows economical use of real estate in the box, digital control and often a zero-crossing detector to eliminate the "zipping" sound when you change volume while listening to music (as one often does). This approach however incurs a penalty in signal purity, as the distortion will never be quite as low as with high-quality relays with decent metal film resistors. On the other hand, quite a few high-end manufacturers, including [Jeff Rowland](#), Linn and Pass Labs, seem quite happy with digital potentiometers. Bill Schweber has written a nice [summary](#) of the pros and cons of mechanical versus digital potentiometers.

What has preoccupied me for a while now is how to turn the rotation of the control knobs into a digital up/down signal for the volume and balance control, and how do this in such a way as to allow the remote control to change the volume and balance independently and consistently. I could simply use a toggle switch with momentary action in both ways, or even a pair of push switches, but to me turning the volume up does indeed feel it ought to be a "turning" action. I also feel that in ergonomic terms the volume control should be the biggest one on the front panel, since it's the most often used. Various options are possible:

- Use a standard potentiometer, and generate a digital setting by connecting an ADC to the potentiometer wiper.
- Use a pair of small push-switches, arranged so that the turning shaft enables one or the other, and connect these to the "volume up" and "volume down" logic inputs.
- Use a potentiometer, but sprung so as to return to a central position, and use one or more pairs of comparators to generate up/down signals when the rotation is beyond given points each side.
- Use a continuously rotating control device that generates an analogue output when rotated in either sense, so that the polarity of the output defines change in output. If this is a pulsed output, all the better, as then the rate of rotation can be used to define the rate of change of the volume.

The first is the easiest to put together, but has the drawback that the position of the knob can't be changed from the remote, though a motor-driven potentiometer would remedy this. It also involves a high-frequency clock for the ADC, which I am trying to avoid as far as possible. The second idea is appealing, but isn't mechanically straightforward, nor possibly very robust - the same is true for the third, but more so. The last is the most attractive to me, as it involves no moving contacts, and I like the idea of avoiding end stops for the rotation - this solution should last for ever! The best way I can think of to put this into practice is to use optical sensors to pick up coded signals from a rotating drum on the shaft; I'm still working out how to detect the sense of rotation. Dominic Peterson pointed out a description of an [optical rotation detector](#) which works along similar principles (and it also turns out that most computer mice work this way!)

There is a fascinating discussion of volume control strategies from the rec.audio.hiend newsgroup [here](#). This includes a description of a device I hadn't heard of before - the "vactrol" is an optocoupler using a cadmium sulphide photoresistor, and the application here is to use a pair of these as an attenuator, with a clever approach to overcoming the non-linear transfer function of the device. This is something I may well try.

Channel balance

This is generally problematic in mid-to-high-end amplifiers. Historically, manufacturers like Naim made the absence of balance controls or tone controls a selling point, on the justifiable grounds that extra potentiometers in the signal path were detrimental to sound quality, and that with decent quality source and speakers they were unnecessary most of the time. While tone controls have yet to make a reappearance in the high end, many systems still sport a balance control somewhere, and the availability of higher quality potentiometers, not to mention good quality digital potentiometers (and cheap microprocessors for the user interface) has meant that manufacturers have been able to keep the balance control without embarrassment.

The advantage of using relay switching of volume or digital potentiometers is that with independent control of the attenuation of both channels, balance control can be added in software, rather than by adding any extra components. The only drawback of this approach is that you are still restricted to the attenuation levels provided by the potentiometer - usually in 1dB steps. This means that it is still difficult to get the sum of either the amplitude or the power from the two channels to stay constant as the balance control is moved from left to right

or vice versa. On the other hand, this isn't critical to sound quality, and the result will probably be perfectly satisfactory even if only a little care is put into the choice of attenuation versus balance setting.

The ideal way to translate from a balance code and volume setting into an attenuation setting for each channel is to use a pair of EPROMs, each with a composite address formed from the volume code and the balance code, and have a saved datum for each address. This allows one to test any given mapping and change it simply by reprogramming the EPROM. I plan to install a pair of sockets for the EPROMs, and while I don't have facilities for programming these at the moment, I can make a couple of simple digital circuits which add or subtract the balance code from the volume setting.

Balanced working

I always liked the idea of using balanced connections to reduce induced noise. The working (passive, single-ended) preamp I use is actually quite noisy, as there is a rather sprawling mess of wire between the input sockets (left rear), the source selector (left front), the volume control (right front) and the output sockets (centre rear), which conspires with some large mains transformers elsewhere in the equipment stack to generate a nice hum on the main outputs. Using relay switching for source selection, and with the volume control sorted out, hum shouldn't be a problem within the box, but there's still plenty of opportunity for picking up induced noise in the interconnect leads between the boxes.

My original picture was of balanced-to-single-ended conversion at the inputs, doing the source and volume control with unbalanced circuitry and then providing balanced outputs. Reading about high-end preamps from Krell, Audio Research, Jeff Rowland and others, it occurred to me that the most elegant and purest way was to stick to balanced operation all the way through, but I was still thinking in terms of using relay-switched attenuators or motorised pots for volume control, which would be either too expensive or too impractical to double up. When I discovered the digital potentiometers from Dallas Semiconductor and Analog Devices, however, I realised that I could go down this road after all without too much of a sacrifice in sound quality (I hope - I haven't got round to doing proper listening tests yet). An additional advantage of this wholly balanced operation is that when a balanced signal is passed through an active circuit with a common power supply any noise leaking in through the power supply is only amplified in common mode, and so is almost cancelled out when the balanced signals are recombined. This way the power supply rejection ratio (PSRR) of the circuit is considerably enhanced.

I plan to have a mixture of balanced and unbalanced inputs (I can't see myself getting a cassette deck with a balanced output, for instance), and have an input buffer configured so that the cold line for the rest of the system is taken either from the cold input or derived from the hot input via an inverter, depending on the type of input selected.

Polarity switching

This is often wrongly described as "phase inversion", which implies some kind of time delay, and this name is only really appropriate for sinusoidal waveforms. Regardless of the label on the switch, in practice this normally means swapping each of the outputs with an identical signal, but with the polarity inverted. The presence of balanced outputs means that I can simply swap hot and cold outputs, while for the unbalanced outputs I just choose whether to connect the output to hot or cold.

Mono output

This normally involves summing (averaging) the left and right channels, and feeding the result identically to each channel's output. It's not that useful in practice, except for testing connections and positioning the speakers in the room, though it can give a small improvement in signal-to-noise ratio for mono recordings. If it's possible to generate this output without adversely affecting the quality of the signal, it might be worth it. It can be available as a separate output (for a subwoofer, for instance), or generated at the main outputs instead of the stereo signal. The latter is more useful as it lets you switch between mono and stereo without unplugging anything.

The summing process is done by feeding both signals to one point via a T-network of resistors, either to ground or to the virtual ground in an op-amp circuit. It can generally be carried out in any one of four places, namely at the inputs (before the input buffers), before the volume control, between volume control and output buffer, or after the output buffer. All of these involve a compromise - either extra impedance is inserted in the signal path or an extra buffer stage is needed. In particular, monoing immediately preceding or following the volume control means losing the accuracy of the potentiometer. The most acceptable compromise seems to me to do it right after the input selector, where a well-defined impedance isn't obligatory, although of course it relies on both channels of the source component having the same output impedance. On the other hand, this doesn't have any effect on the signal in the normal stereo mode.

Power supply

For the moment I'm using a pretty conventional linear power supply, with 7,000uF of capacitance per channel, high-speed rectifiers and a pre-regulator to give a 26V DC output. I take care with the ground paths, and use dual mono construction.

This arrangement ought to give pretty good isolation from mains borne rubbish (though I haven't had the opportunity to measure it). All the same, the ultimate power supply would seem to be a set of rechargeable sealed lead-acid batteries. I don't expect to get this together for a while, since it's definitely an expensive luxury (though that could surely be said about high-end audio in the first place!). To feed the preamp, phono stage and active crossover would need six separate channels of dual supplies of 20V or more, as well as a decent recharging circuit. I haven't even tried to work out how much that would cost yet, never mind the space and the weight... Anyway, each of my boxes will have DC inputs for their eventual batterification. (I like that word, but last time I checked, I seemed to have the only site on the Net to use it.)

There are other, less expensive, power supply improvement options. One is to use one or more isolation transformers to provide a balanced AC input to the system apart from the power amplifiers - this will not only decrease the power radiated by the wires but will also reduce the noise significantly. Another possibility is to have a remote mains power supply to feed the DC inputs to the preamp, phono stage and active crossover. This should clean up the EM environment around the circuitry, and has the advantage of being completely interchangeable with a battery supply once sockets and switching are installed in the preamp.

Recent experience with the PS Audio Power Plant, which is an AC-AC converter, has stimulated me to think about the effect of mains-borne noise: my phono stage and CD player definitely sounded clearer and smoother when powered through the PSAPP. This device obviously has several effects on the mains signal, but I'm not sure which of them makes the most difference - it could be the decreased source impedance, the removal of harmonic distortion or the filtering.

Inputs and outputs

For flexibility I am installing as many inputs as I could possibly expect to use, and catering for balanced and unbalanced sources. This amounts to four balanced inputs and four unbalanced. All of these are selectable for listening through the main outputs, with the correspondence between sockets on the rear panel and the more functionally labelled selectors on the front determined by internal jumpers.

There is a single tape output, corresponding to the "record" select control. I assume that one of the unbalanced inputs is from the tape device, so all inputs except this one are routable to the record output.

I am using two stereo pairs of main outputs to the power amplifier or active crossover, one balanced and one unbalanced. I would have liked to have provided a "passive" unbalanced main out, direct from the volume control wiper, but there was simply not enough space on the back panel. It would have been nice to have a pair of identical doubled-up balanced outputs for passive bi-amping, but again there wasn't enough room, though I can always use the unbalanced output if I want to do this.

There are also a couple of DC inputs (one pair of 24-0-24V for analogue supply, and one pair of 5V inputs for control functions). The preamp circuits can be connected either to these or to the internal power supply, with a switch to select them.

Hardware

At the moment the preamp is housed in a plain two-part aluminium box from Maplin (as similarly are the active crossover and the preamp power supply), with controls simply bolted onto the front. It does its job, but it's ugly, with a couple of gaping holes where redundant controls are missing, and I've never put effort into proper labelling.

I shall shortly put together some more substantial - and definitely more domestically acceptable - boxes with decently finished sheet aluminium chassis and gloss-painted MDF side cheeks. The panels will be damped with Dedshete, to reduce the audibility of clicking relays and humming transformers. I also plan to line the insides of the boxes with copper foil (if I can find it) to help exclude RF junk.

The only domestically acceptable enclosures I've come across for DIY audio equipment have been computer-type rack boxes, with those macho handles a la American high end, and these are not only very expensive but too wide for most support systems. I'm sure there must be a market for reasonably finished boxes in the standard audio system dimensions at a sensible price. On the other hand, I know audio DIY-ers like to stamp their personality on just about everything...

Remote control

Digital control of all the source select and volume control facilities makes remote control an obvious possibility. I have already built a rudimentary remote controlled preamp based on the (now obsolete) MV501/MV600 chips, and I just have to make a more domestically acceptable remote than the present chunk of MDF!

Standby facility

The more complicated an audio system becomes, the more boxes collect on the stack. In particular, the use of monobloc power amps (or even pairs of stereo power amps for an active system like mine) means that when I go out I put the mute on the preamp, switch off the tuner (or record deck or whatever) and switch off both the power amps. What would be nice would be a central standby switch, which would keep the basic signal circuits powered up in the various parts of the preamp and active crossover, but cut the power to the power amps and the control and display circuitry, while muting the outputs of each box for safety's sake. Building the system myself makes this possible, since I can use a set of connections between the boxes (rather like the control bus used by Denon and others in their separates systems) to unify the control system.

There is another less obvious benefit to this. If I eventually get round to building a battery power supply for the low level circuits, the standby state will provide an ideal signal for the batteries to recharge between listening sessions.

Alex Megann, May 2000

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