The Metz Preamp

The design aspect of my audiophile hobby has been dominated by power amplifiers, starting with tubes back in high school to a variety of FET designs. One thing I had in the back of my mind for a long time was to design a preamp – but somehow I always made do with a receiver as the source selector and control center for my system. For the last ten years though, the ‘ole Sherwood receiver has been apart at least three times to nurse dirty controls and switches, and I always had some nagging doubts about the ultimate quality of it’s sonic performance. The time had come to tackle the preamp.

Key Goals

As I set out on this design, I began with a set of goals. First and foremost, I am both a serious amateur audiophile in pursuit of performance and an engineering pragmatist – I wanted a straight forward stereo source selector, monitor selector, volume control, and line amp, and not much more – this is not a home entertainment center/surround sound/video switcher/MP3/Espresso machine – just good clean stereo. Key things I wanted to achieve are:

- Clean, long life, reliable source switching
- Electronically controlled, non-deteriorating volume control
- Impeccable sonic performance and low noise and crosstalk

In summary, the design features you will find in what follows are:

- 8 inputs – MM phono, Tuner, CD1, CD2, Tape1, Tape2, Aux1, Aux2
- 4 record outputs with monitor select for CD2, Tape1, Tape2, Aux1
- Gold dual contact sealed relay source selection
- Front panel headphone output
- Front panel 1/8th inch appearance of Aux1, for an IPOD or similar device
- Optical encoder volume/balance control
- PIC microprocessor control
- Bypass-able bass and treble controls
- Mute
- 2 sets of controlled outputs to amplifiers
- Ground plane main/audio PCB for low noise and crosstalk
- 2 power switched 5 volt outputs to drive relay controlled power strips for system components

Design considerations

Beyond the general feature choices, there were three key design considerations. The first was the method of input selection – can I do this entirely solid state? Short and simple, based on distortion characteristics and cross talk, the answer is no. I researched all the available analog switches, including those meant for high end audio applications, as well as do-it-yourself discrete FET solutions, and nothing is adequate as far as cross talk is concerned, primarily due to semiconductor parasitic capacitances. They also introduce a small amount of distortion due to R_{ds}sat modulation. The next best, mechanical, alternative was sealed relays. An extensive search led to the choice of the Panasonic AGQ200 series, the 12V variety requiring relatively low coil currents, and featuring gold plated bifurcated contacts. These should provide a life expectancy well in excess of most of us!

Next was the volume control – definitely not a potentiometer, and resistor ladder networks still involve mechanical contacts. The choice I made was an optical encoder in conjunction with the highly respected Burr-Brown designed TI PGA2310 volume control chip. Given the cost of an optical encoder, and the desire for simplicity, I then chose to combine the volume and balance functions in one control, with a button to select between them, and LEDs to indicate the settings. Of course, once the digital volume control choice has been made, you already have one foot firmly stuck in the mud of the op amp/discrete design debate, which brings us to the last key design decision:

Op amps v.s discrete: Like a good audiophile snob, I began this project firmly and faithfully committed to a discrete FET amplifier design approach, and scratched out multiple pages of 'J74 and 'K175 – based phono and line stages, and prototyped them. They all worked quite nicely. (I'm sure some of you out there are thinking it should have been 12AX7s, ha, ha!)

When I starter to look at PCB layout and put together a parts list, I went "holy schnockers, what a load of discrete parts!" What a forest this board is going to be! Do I really want to do this?" That's when I decided to have a bake-off with an op amp alternative, especially since I already had working discrete FET prototypes. I went ahead and designed OPA2134 based phono and line stage alternatives, and ran them side by side with the FET prototypes. That raised the question of what kind of tests to run. The first were standard THD measurements, with results in Figure 1. While these look great for both designs, they don't address dynamic performance, and I knew this was just not going to be sufficient in the world of discrete v.s. op amp mythology and never ending controversy. From here, one can dive into IM, TIM, and various other esoteric measurements – but I wanted a more dynamic and convincing test, something that would resonate more with the ethereal golden ears hyperbolic adjective comparison world – so this is what I came up with (see Figure 2):

1) Assume the discrete FET designs were golden – the standard of comparison – after all, they were adaptations of Borbely preamp/line stages that have appeared in print here. In any case, they could be considered and accepted by many as the "best effort" approach.
2) Arrange the FET designs to be run in parallel with the OP amp designs, fed by a common source, and provided with a precision input attenuation and output level matching capability.
3) Feed the common input with real, dynamic and varied music source material, not static signals.
4) Take the level matched outputs and feed them to my best Tektronix 2 channel oscilloscope amplifier, with one channel in the invert mode, and both in add mode, so that any amplitude or phase difference between the two inputs would show up clearly on the screen.
5) Turn the scope amplifier channel gains up to the highest level they will handle without distortion or overload. I used a 7A26 plug in on my 7904A scope, and this point was roughly at a gain of 80dB, which would make differential artifacts visible down to at least –90dB (~ $1/10^{th}$ division). The reason this can be done is because the outstanding design of the 7B26 plug in performs the invert/add functions far enough to the front end of the amplifiers that much of the gain appears after the subtraction. If you don't believe this works, try it, and you will be amazed – it takes some effort to match the channel gains to get a null, and when they are slightly off, all that gain applied to a real difference will immediately make the scope display go off the screen!
6) The result is a flat line if both input signals are identical, with any difference in the two showing up as blips or trace deviations on the otherwise flat scope trace.
7) In addition to just looking at it (where you night miss something temporally), use the adjustable scope trigger level to "catch" any deviations between the two signals within, say > -90dB.
8) The final result? For the test comparison between the discrete FET line amp stage and the OPA2134 version, there was no discernable difference in output signals, for all the different music inputs tested, within a discernable amplitude of -90dB below the input levels. In other words, they did, within a negligible error band, produce exactly the same output for the same input.

The unavoidable conclusion is that the op amp stages perform as well as the discrete FET alternatives, with real dynamic music signals, and greatly simplify the design. To seal the deal, I put the two prototype line amps in my system, between the CD player and amplifiers, with an a-b switch. The result, again, was that both sounded phenomenal, and indistinguishable.

The only thing left to do was to come to grips with the results and go ahead with the project!

Block Diagram Overview

Figure 3 is a block diagram of the preamp. One of eight inputs is selected by a relay network. The phono input passes through the phono amp stage first to apply RIAA compensation and to bring it essentially to the same level as the other line level inputs. Four of the inputs also appear at the monitor select relays. The left and right selector relay outputs may be bridged for mono by the mono relay, while the signals at this point are also fed to a pair of line buffers for the 4 record outputs. Following this, the main/monitor relay selects the signal to be passed on from the main inputs or the monitor selection.

A pair of buffers feed the volume control chips, which are followed by the bypass-able tone controls. This then is also the controlled amplifier output. The pair of headphone amplifiers are also driven from this point.

The center of control is a PIC 16F877A microcontroller, which receives all of the switch inputs and the optical encoder input as parallel I/O, and controls LEDs, the volume control chip, and relay drive chips via an SPI serial interface.

Power is provided by a toroidal transformer, accepting 120 or 240 V input, and providing regulated +/- 15V for the analog circuits, and +12 and +5 for the digital circuits, with separate grounds.

Detailed Design

Input and monitor selection is performed by a set of 8 and 5, respectively, Panasonic AGQ20012 relays. (see Figure 4, main board schematic) These were chosen on the basis of their: 1) Small size, 2) low coil current requirement, 3) low resistance, gold plated contacts, and 4) “duplex” bifurcated contacts, to ensure a long noise-free operating life. The same relays are used throughout, for the Mono/Stereo, Monitor/Main, and tone control bypass switches. All the relay coils are driven by a pair of Alegra A6821 SPI serial input parallel I/O drivers, U3 and U4 – very convenient devices! All control push buttons are lighted with LEDs, which, in the case of those associated with relay functions, are also energized by the relay drivers.

The phono preamp (Figure 5) is of the MM type, as I have no MC equipment – if you need MC, that design aspect is left as an exercise to the reader. The MM design consists of a low impedance buffer stage driving a passive high frequency roll-off, R5 and C4, followed by an active low frequency boost stage, providing standard RIAA compensation.

The record output buffer as well as the volume control chip input buffer are both simple x2 gain stages (Figure 6). Both are preceded with 75kHz low pass and 10Hz high pass filters, to remove low and high frequency input noise. The volume control chip requires a source impedance < 600 ohms to meets it’s distortion specs – therefore the buffer.

SPI serial input also controls the TI/Burr-Brown PGA2310 volume chip. More on how this is functionally used – later, in the firmware section. Mute capability is also provided.

Output from the volume control goes to the pair of amplifier drive outputs, via the tone control bypass relay, which is normally in the bypass mode. The bass and treble tone control stage consists of a buffer, driving a double feedback loop Baxandall +/-10dB boost/cut stage, with a 1kHz rollover frequency (Figure 7).

An op amp/FET power stage (Figure 8) drives the headphone output, and is capable of providing several watts to speaker impedance loads, if desired. The power stage includes a bias adjustment, which must be performed as part of initial unit testing.

The controller (controller board, Figure 9) is a PIC 16F877A, with internal program and user Flash memory and SPI serial control. Select, monitor, and other push button inputs appear directly as parallel I/O. The optical encoder connects as a two bit input. Volume position is indicated by a set of 24 LEDs arrayed around the encoder, while balance is depicted by a linear 4 + 4 left/right green/red LED display (Figure 10). The firmware is described later, and is available for download, or programmed into a chip, as shown in the parts list.

An Avel/Lindberg toroidal power transformer feeds a pair of adjustable series regulators set for +/-15 volts for the analog circuits, while the digital supplies are generated by fixed +5 and +12 volt regulators Figure 11). Adjustables are used for the analog supplies as they are quieter than the fixed variety. Grounds for the analog and digital supplies are kept separate, except for a single, well-chosen tie point, under the volume control chip.The power transformer provides windings for only one set of outputs. The digital supply windings are added to the transformer as described in the assembly guidelines.

Fundamentally, when the preamp is plugged in, the power supplies and the controller are always on, and the power on/off function (power button) turns on the analog operation. In the off mode, all relays, and therefore input selects are de-energized, and the volume is muted. In the on mode, input selections are recalled from EEPROM and applied, but volume always starts off at zero. A feature of the design is the inclusion of a pair of power on/off controlled +5 volt outputs, which can be used to control power to other system components. In my case, I added a relay to a power strip, into which my amps, CD players, etc. are plugged, so the whole system comes up together – no need to flip an array of switches!

PCB Partition and Design

Circuitry is distributed over three boards. The main board (Figure 12) houses all the analog circuitry, some of the control switches, and the very convenient Kobicon dual gold plated RCA input and output jacks. The key features of the design here are a multilayer board that includes ground plane under analog signals, and very careful layout and power distribution to minimize crosstalk and noise.

The controller and input select switches are on the second board (Figure 13), a partition chosen for convenience of physical layout of the switches and to keep the digital functionality separate. Finally, the optical encoder and associated volume and balance indicating LEDs and their serial to parallel drive chips are mounted on a separate board (Figure 14), again to facilitate the mechanical design and minimize cabling. Boards are interconnected with ribbon cables.

Firmware Design and Functionality

Figure 15 shows a high level flow chart of the firmware. The core is a 4 state machine, driven by the de-bouncing states of button pushes. This is, in turn, embedded in a main loop that responds to optical encoder state changes. Each time around the loop, the state machine checks to see if a button has been pushed, and if so, it is timed for de-bouncing, and then checked for release, before the handling routine for that button is entered. Then, the power button and optical encoder state are checked, and the unit's operating condition is updated accordingly. Optical encoder state changes result in increases or decreases of volume, or left/right balance adjustments if the control is in the "balance" mode. More details including flow charts for button functions, optical encoder reading, volume/balance adjustment, and volume and balance LED setting calculation and driving, can be found on the web site referenced in the parts list.

Some particulars of the Firmware implementation that manifest themselves operationally are as follows:

- Input select, monitor, and mono/stereo switch states are retained in non volatile memory, and are re-instated after a power down/up cycle
- Balance is returned to center, volume is set to zero, tone controls are bypassed, and mute is turned off after a power cycle
- The very first time a unit is powered up, all inputs are de-selected
- Monitor buttons toggle – when one is pressed repetitively, it toggles on and off. If one is on, and another is pressed, the new monitor is selected, and the previous one cancelled
- The volume control chip inherently is adjustable over a gain range of +31.5dB to −95.5dB in 255 0.5dB Steps. This is far too much gain on the top end, as my gain plan calls for approximately +xxdB overall. The optical encoder volume generates an 8 bit word from 0 to 255, which the processing firmware translates to –95.5db at the low end and +yydB for full volume.

Assembly Guidelines and Considerations

Probably the most important aspect of construction is surface mount soldering technique, which will not be taken up here in detail, except to say that a high quality small tipped soldering iron, fine (< = .032") solder, copper braid solder wick, a good magnifier, a bright work light, and steady hands are essential. A very careful inspection and review process is recommended before powering anything up.

Figure 16 is the parts list.

In terms of parts choices and substitution – hey, this is the DIY world – if you know what you are doing, go to it!. Parts I would not recommend substituting are:

- The PGA2310 volume control chip
- The OPA2134 op amps, except if you know what needs to change to maintain stability/compensation for an alternative
- The Power FETs in the headphone amp
- The optical encoder, unless you chose one with the same encoding scheme and same number of transitions per rotation
- The poly/film capacitors, except with similar types – do not substitute ceramics for these

Areas where improvisation is encouraged include:

- The select push buttons: Their only requirement is a momentary contact and an associated LED. The ones shown are accommodated in the PCB layout, but others may be used and hand wired.
- Input RCA jacks – the board is laid out for the particular Kobicon items, which are very convenient, but others may be hand wired as your particular mechanical design accommodates.

The power transformer deserves some note: The digital power supplies require a separate winding, for which I found the addition of about 300 turns, 28ga, center tapped (150+150) to be about right. These can be easily added to the toroidal transformer specified. Alternatively, you can generate the digital supplies from the same AC winding that is used for the analog supplies, but this may compromise the hum/digital noise performance some what, although I have not tested this. One could also use a second transformer, or find/specify one that has the necessary windings to begin with.

I found installation of all the surface mount parts first the easiest approach to assembly. It allows the boards to sit the flattest, and with least access interference for installation and soldering of parts as you go along.

Before powering up the preamp, set both headphone amp bias adjustments to minimum bias. Then, plug the preamp in and measure the voltage across R9A4R while adjusting R11A4R, until 60 to 80mV is reached, corresponding to a bias current of 30 to 40mA. Repeat for the left channel.

My enclosure is shown in Figures 17 and 18. It consists of two mating pieces of 0.050” aluminum, one forming the bottom and back, the other the top, and a 1/8th” thick front panel. Figure 19 shows the assembled interior.

Performance

Performance data is shown in Figure 20. The distortion figures, as one might suspect from the op amp data sheets, is respectable, although I put the greatest faith in the before-mentioned music comparison tests, and actual listening tests. The later were performed with my HHB850 CD player, a pair of my own design FET power amps, and a pair of my own design speakers, employing KEF drivers. The results were exceptionally pleasing, but to claim any more here would be self-serving.

The ground plane main board appears to also have paid off, given the noise and cross talk figures.

About the Author

Reinhard Metz has B.S. and M.S. degrees in Electrical engineering from the Illinois Institute of Technology and The University of Illinois, respectively. He has worked over 20 years as an engineer and manager at Bell Laboratories, AT&T, and Lucent technologies. In 2001, he left Lucent and started an engineering/consulting business specializing in circuit design, particularly power circuits. The business also provides technical expert services for intellectual property legal cases. He currently is designing high power induction heating inverters and other analog circuits for CookTek LLC. Throughout his career, Reinhard has maintained an avid audio hobby, including many of his own designs. He has 6 patents granted and 5 pending.