Mixers & Mike Pre-amplifiers

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Part 1

MIXERS AND MIKE PREAMPS

By Paul Stamler

In many ways, live recording is the ultimate challenge for both the audiophile and the professional. It incorporates elements of acoustical physics, electrical engineering, psychoacoustics, musical expertise, acrobatics, and (likely as not) clinical psychology. Unfortunately, makers of commercial recording equipment seem to use their gear under only controlled studio conditions, and many of the available tools are inadequate for the real-world challenges of live recording.

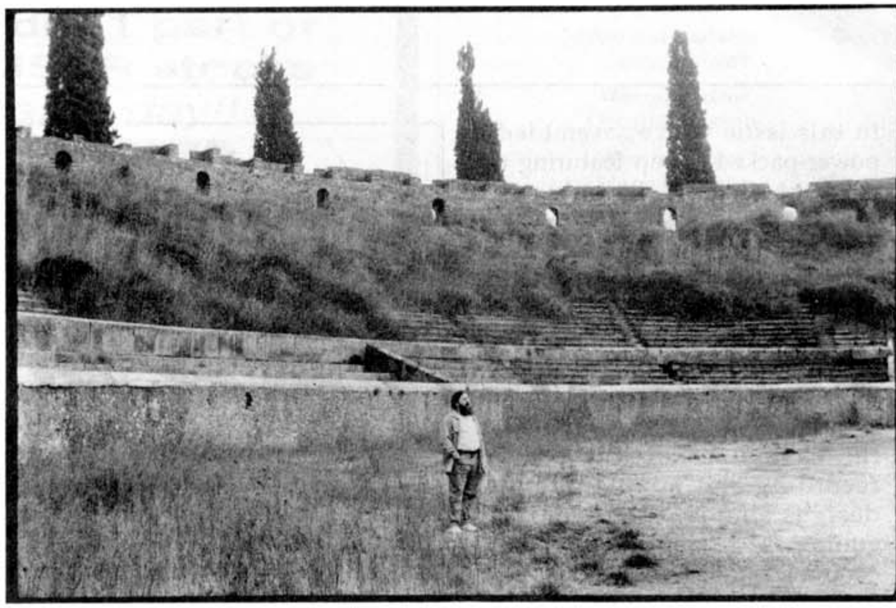

PHOTO 1: The author, singing in a loud voice at the arena in Pompeii; the monitoring system was perfect. (The song, incidentally, was "Bright Morning Stars.")

I realized this several years ago at the Frontier Folklife Festival—a three-day affair involving several stages, dozens of performers, and many rapidly changing stage setups. I was recording the proceedings at the main stage—the most demanding location (used for evening concerts as well as daytime workshops).[1]

Unfortunately, my relief man fell out of a balloon two days before the festival, breaking his leg in two places. I had to mix and record for 12 hours per day without a break at 95° temperatures. I compounded the stress by attending the festival parties every night (the rational part of my mind said I shouldn't, but who could resist Glenn Ohrlin reciting "The Face on the Barroom Floor"?). Reduced to a state of gibbering idiocy by the third day, I made several avoidable engineering errors.

After that I developed a new Law of Audio Engineering: "Never design a piece of equipment too complex for an intelligent chimpanzee to operate." It serves me well today.

Alas, it's a rule manufacturers don't follow. Judging by their products, they assume the user is an experienced engineer, working in a clean, well-lit area, and is well-rested with the leisure to think about what he/she is doing. Not always so. You may be the Einstein of engineers, but by the third day of a festival, it'll be Chimp Mode for you, too.

Another problem with commercial mixing and recording gear is that, with a few exceptions, the designs are about 15 years behind accepted audiophile practices. Long after home-equipment designers recognized the audible difference polypropylene capacitors, metal-film resistors, high amplifier slew-rates and wideband, low-impedance power supplies can make, the typical recording console (including the $100k+ studio jobs) still includes polarized tantalum coupling caps, carbon-film or carbon-comp resistors, slow amplifiers, and RC power-supply decoupling networks. Too often, manufacturers use cheap connectors, switches, and controls, and an architecture which is the antithesis of the short, straight signal path audiophiles prefer. Hence, the recording equipment is often sonically inferior to home sound-reproduction units.

So I decided to design simple, rugged, chimp-proof mixing equipment that works in the real world and reflects audiophile design practices wherever possible. I started with the most basic question.

What Does a Mixer Do?

A mixer:

—amplifies low-voltage signals from microphones to a higher voltage and (usually) accepts line-level signals at various nominal levels.

—may include controls, such as equalizers or "EQs" (tone controls with a college education), to alter the tonal character of the audio signal. A bass-cut filter is also useful, as most microphones sound boomy when located a few inches from a sound source, a phenomenon known as "proximity effect." A bass-cut filter can restore a microphone's flat frequency response and help minimize background roar, rumble, and breath noises on vocal mikes.

—controls "panning," or the placement of the signal from a particular microphone within the left-to-right spread of the stereo stage.

—"mixes" the signals together at whatever levels the operator chooses.

—provides output signals with adequate voltage and current to drive a tape recorder, sound system, or both.

—monitors the level of the signal at its output with a mechanical meter or a row of LEDs. The indicator, of whichever type, may be average-reading, peak-reading, or a combination.

—may have an "effects bus" that inserts signal-processing devices (reverberation simulators, chorus effects, and the like) into the audio path.

—can provide "phantom power" (12 or 48V DC) to run the internal preamplifiers in condenser microphones.

Also, most mixers incorporate facilities that feed a separate "monitor mix" to performers so they can hear themselves and each other.

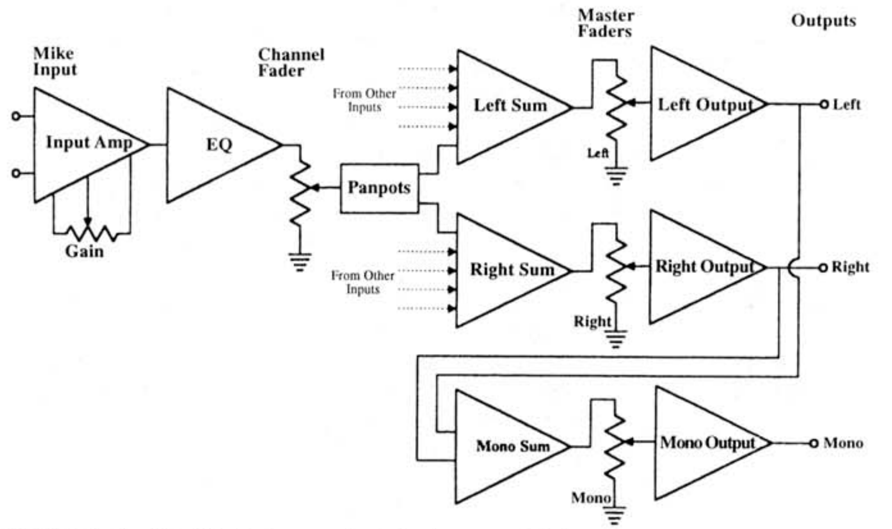

FIGURE 1: Simplified block diagram of a typical commercial mixer.

Input Issues

A typical commercial mixing board (*Fig. 1*) has several trouble spots—aside from the sonic problems caused by poor component choices—that can impair the sound. The most serious problem is overloading of the input amplifier, which happens so frequently in live performances that I'd almost classify it as the norm.

How does this happen? The IC op amps usually found in commercial boards, when run from the standard ±15V supplies, cannot produce more than about +20dBu (22V P/P). Assuming the mike preamp's voltage gain is 40dB (a typical figure), the preamp will clip if the mike puts out more than –20dBu. Unfortunately, modern microphones can yield a good deal more than that.

Russell Hamm has measured microphone output levels in typical recording situations (*Table 1*).[2] Note the figures for the Neumann U-87, which is typical of high-output condenser mikes. A loud yell at 4" can produce a signal of 0dBu, or about 2.2V P/P. (In the string-band music and blues I record, loud yells are frequent.)

On the other hand, some sessions produce very low signal levels at the mike preamp input. I recently recorded a female vocalist using a ribbon microphone. Even with the mike 3" from her mouth, the peak signal level was only about –50dBu (about 7mV P/P). Clearly, a mike preamp must deal with a wide range of signal levels.

Previous mike preamps had high gains (about 50dB); to prevent overload, attenuator pads were switched into the input circuit as needed. This was a dubious solution, as the resistive pads compromised the preamps' noise performance, and the switch contacts, carrying low voltage and current, quickly became noisy and intermittent.

Recent mixers use a variable-gain input stage, which alters the preamp's gain over a wide range, allowing proper matching to almost any mike or line level. Often a "clip light," an LED that flashes when the preamp is approaching overload, signals the operator to lower the gain. You might wonder if the operator can *hear* the preamp clipping. No. In many live situations, given the realities of ambient noise, poor monitoring, and listening fatigue, the operator is hard-pressed to manage a proper mix at all.

I prefer a more basic solution—a mike preamp that can't be overloaded by any conceivable signal. The simplest way to achieve this is to split the preamp into two separate gain stages.

The first stage precedes the fader, and thus is susceptible to overload. Op amps running on ±21.5V DC rails, or discrete amplifiers such as Erno Borbely's designs, can put out +24dBu into a reasonable load (see section on loading). Since Hamm's results indicate that they must handle a maximum input level of 0dBu, the input stage gain cannot exceed 24dB (15.8×). The rest of the gain must come after the fader, in a *secondary gain stage* (*Fig. 2*). Unfortunately, secondary gain stages, because they are downstream from the fader, contribute noise to the output.

Noise Levels

The advent of digital recording has placed new demands on mixer noise floors. A 16-bit recording system, the standard for CD mastering, can theoretically produce a dynamic range of 96dB. Since I set my nominal "0VU" level at 16dB below the recorder's maximum level, *my* theoretical signal-to-noise ratio is 80dB. Ideally, the mixer should not add any significant noise to this.

I spent several months analyzing noise levels in various mixers. Small mixers were no problem, but multiple inputs proved troublesome because ten secondary gain stages produce 10dB more noise at the output than a single stage. Until recently such a design would have been impractical, as most op amps (and, for that matter, discrete circuits) are simply too noisy. However, the new low-noise chips, such as the LT-1028A, permit moderately sized mixers (up to about a dozen inputs) without appreciable noise problems.

The gain in the secondary gain stage is determined by the *internal operating level* of the mixer—the nominal 0VU output level of the secondary gain stages and EQ amps (if any). I chose to make this internal operating level –10dBu; since –60dBu is the lowest level I ever hope to work with, the total mike preamp gain must be 50dB, with the fader wide open. The input stage gain is already fixed at 24dB, so the secondary stage gain must be 26dB (20×).

Is –60dBu a practical lower limit for mike signals? Yes. Remember that a microphone is essentially a resistive source, usually 150–200Ω, and as such generates noise. A 200Ω resistor generates a noise level of –129.6dBu, so

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the signal-to-noise ratio of a microphone with a nominal output level of –60dBu is only 69.6dB, which is barely acceptable. A microphone with a lower output level will not produce an acceptable signal-to-noise ratio, even with a perfect, noiseless preamp. This is why low-output ribbon microphones, despite often superb sonics, can be a maddening challenge to the working engineer.

Stereo and Mono

When I tape a live performance, I typically mix the live sound for a mono house mix while feeding a simultaneous stereo mix to the recorder. Most of the time, I can't perform these mixes independently. The sound level from the house speakers will leak into even the best-sealed headphones, making it impossible to judge the mix properly. I wind up mixing for the house, and hope that the stereo mix will also be correct.

Most mixers generate their mono mix by summing the left and right stereo outputs (*Fig. 1*). When I pan signal sources for a wide stereo mix, a proper mix in the house (mono) sounds wrong in stereo, as the sources in the center are low, relative to the sources on the sides.

This occurs when sound sources in a room, heard over loudspeakers, do not add in the same way as electrical sources in a summing circuit. Two identical signals, whether in a circuit or in a room, will produce a total 6dB higher than each individual signal, whereas two nonidentical signals (of similar level but different content) produce a total only 3dB higher than each signal. Panned signals added in a summing circuit are essentially identical, but when such signals are played over loudspeakers, they are no longer identical. Unit-to-unit variations in the loudspeakers and asymmetries in the listening room make the sounds from the left and right speakers quite different.

These differences vary from speaker to speaker, and from room to room. Highly phase-coherent speakers, such as Quad ESL-63s, behave more like identical sources when used in symmetrical rooms. These speakers can create a stronger center image.

Look at the simplified diagram in *Fig. 3*. Signal A comes into a panpot at a nominal level of 1V. If panned all the way left, a lossless panpot will send 1V to the left summing amplifier and 0V to the right summing amp. Meanwhile, signal B enters the panpot at a nominal level of 0.707V. If panned into the center, a typical panpot produces 3dB of loss in each channel, so it will send 0.707 × 0.707, or 0.5V, to the left and right summing amplifiers.

Finally, signal C is panned to the right channel, again at 1V. The stereo signal is at the output of the summing amplifiers. The left channel carries 1V of signal A and 0.5V of signal B, while the right channel carries 0.5V of signal B and 1V of signal C. This program feeds the recorder.

If you add the two channels together to arrive at the mono output, which feeds the PA speakers, signal A (coming from the left channel) comes through unchanged at 1V, as does signal C from the right. The two channels of signal B, each 0.5V, are effectively identical, and they add in the summing amplifier to produce a total of 1V. Signals A, B, and C will thus come through the PA system in equal volume.

Later, we replay the stereo recording in a listening room. Let us arbitrarily say that signal A, coming from the left speaker, produces a sound pressure level (SPL) of 80dBA in the room. Signal B (0.5V) is recorded on each channel at half the level of signal A (1V), so it will produce an SPL from each speaker that is 6dB lower than signal A's, or 74dBA. These signals are nonidentical, and add in the room to create a total SPL of 77dBA. To the listener, signal B (in the center) is 3dB softer than signal A (at the left) or signal C (at the right)—even though the signals were the same level in the mono PA mix.

This problem is unavoidable in any mixer which derives the mono output from the summed total of the stereo outputs. However, it can be eliminated by providing a separate summing amplifier for the mono mix, fed from the point in the signal path just upstream from the panpots. In this design, the panpot setting does not affect the mono mix. Panpots designed for "constant power" (a loss of 3dB at the center position) will produce volume levels that do not vary with panpot setting in most stereo listening rooms.

Summing Amp Overload

I have heard summing amp overload many times at live concerts, and it is always traumatic. Its genesis is subtle, and related to the output stages of the mixer.

It is estimated that an audio system requires at least 17dB of *headroom* to avoid clipping; that is, the active devices must produce a clean signal 17dB above the nominal operating level.[3] Therefore, to remain clean, a professional audio board with a nominal output level of +4dBu must put out +21dBu.

If the output stage is putting out +4dBu nominal (+21dBu peak) and its gain is 14dB (not unusual), then the nominal level at its input must be –10dBu (+7dBu peak) (*Fig. 4*). If I pull down the master fader to 30dB of attenuation (also not unusual), then the level at the top of the fader (i.e., at the output of the summing amp) must be a nominal +20dBu (+37dBu peak).

But wait, you can't do that. Even with the ±24V supply rails used in discrete-component boards, the summing

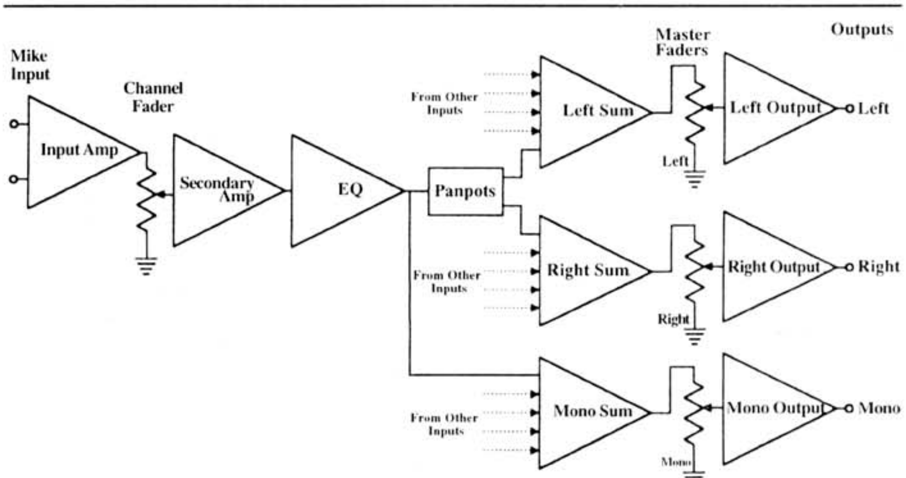

FIGURE 2: Simplified block diagram of a revised mixer.

amp runs out of headroom. It will be clipping painfully most of the time.

Unfortunately, this happens a lot at live (amplified) concerts. As the evening progresses, the engineer adds a bit from mike five, because the vocalist on that mike isn't quite audible. Now mike five is OK, but the guitar on mike 12 is too low. Up it comes. The system is pushed into overload or feedback, so the engineer hurriedly pulls down the master a few decibels. And so it goes, insidiously. I've seen master faders start the evening at 5dB attenuation, and by the intermission they're hovering around 30dB. I've done it too, and I should know better.

If you are hoping the engineer will hear the problem, you are whistling in the dark. There are too many distractions: the concert sound may be distorted by the speakers; the crowd may be noisy; or, if the engineer monitors on headphones, they may distort as well. Also, a masking phenomenon allows distortion to go unheard in an excessively loud signal. It shows up only when the recorded concert is played back at normal volume. With all of these factors, *it's sometimes all you can do to get a decent mix at all*, so distortion can easily slip by unnoticed.

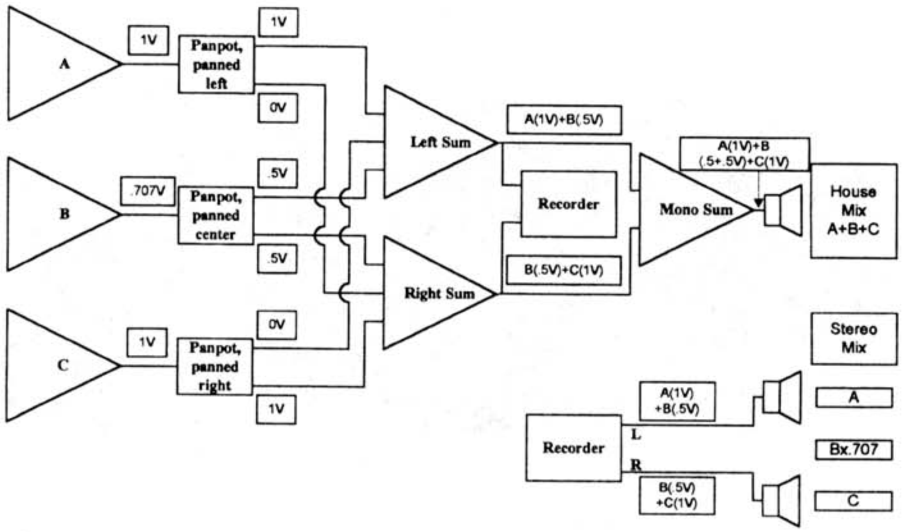

FIGURE 3: Stereo/mono problems in a conventional mixer. Remember that voltages sum linearly in an amplifier (0.5 + 0.5 = 1), but nonidentical sound pressures sum by the "root-sum-square" method in a room (84dBA + 84dBA = 87dBA, not 90dBA). I've omitted the power amplifiers for simplicity's sake.

Solutions

In theory, the prevention of summing amp overload should be a simple task. The master fader's attenuation must always be less than the gain of the output amplifier (except during a "fade-to-black" at the end of the program). One way to guarantee this is to eliminate the master fader entirely. This works in small mixers, but larger mixers are inconvenient to use without master faders.

A variable-gain summing amp does not solve the problem, but simply pushes it back to the previous stages. Variable-gain summing amps are difficult to design anyway, since input-stage bias currents and DC offsets require large electrolytic coupling capacitors to avoid putting DC on the potentiometers.

Some manufacturers use a clever approach. They tailor the fader's taper so the danger point is near the bottom of the fader's travel (or near the counterclockwise limit on a rotary fader). Although the fades-to-black can be abrupt, this works. Another approach is to adjust the gain structure so the summing amp's nominal output level is relatively low, then set the output amp's gain fairly high. This allows a good deal of attenuation, but it requires low-noise circuits for the summing and output amps.

I've found a solution as well. In my large mixer design, I've set the gain of the balanced output amp at 24dB. Each half of the balanced circuit has a gain of 18dB and produces a nominal output level of –2dBu; the output into a differential circuit is +4dBu. The fader's nominal position is (prominently marked) at 6dB of attenuation, which lets the summing amp run at –14dBu nominal (+3dBu peak). Since I very seldom do a fade-to-black on the stereo recording (I add them later, during editing), I've placed a resistor in series with the master fader to limit its maximum attenuation to 23.9dB.

Some designers have tried voltage-controlled amplifiers (VCAs). Many professional boards use them because they offer several advantages. Boards designed around VCAs are immune to summing amp overload, are easy to automate (good in a multitrack installation), and, because the faders only carry DC control voltages, are free from fader-induced noise.

I've ruled out VCAs for this project. They are quite expensive and difficult to obtain through the sources available to amateur constructors. They also can have tricky problems with temperature drift and unit-to-unit variability. Finally, I don't like their sound. I've not heard a VCA-based board that was as sonically transparent as a good standard-type board.

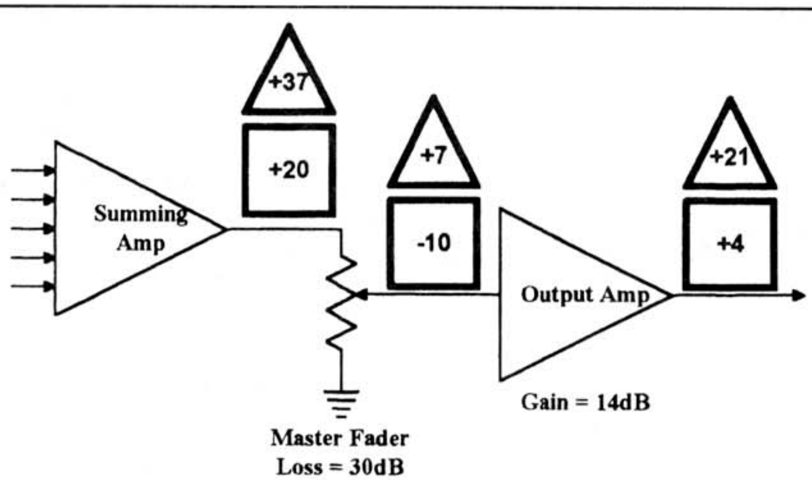

FIGURE 4: Summing amp overload. Nominal output levels (in dBu) are shown in the square boxes; peak output levels are in the triangular boxes.

Loading

Ideally, all circuits should be Class A, with output devices (transistors, FETs, or tubes) biased high enough for all stages to operate in their linear regions. Unfortunately, common integrated circuits run their output transistors at a stingy level. They will produce only about 0.2mA of output current in either direction before one of the outputs begins to turn off. This is surprisingly little current. An amplifier driving a 5k load will begin to operate in Class AB when the output level exceeds 1Vpk.

This transition is audible and is one of the explanations for the relatively poor sonic performance of ICs in many audio applications, as compared to more generously biased discrete tube and solid-state amplifiers. This is a designer's dilemma. Low-noise design requires audio circuits to operate at low impedances, but to minimize output loading (and maximize linear operation), circuit impedances must remain high.

You can approach this problem in several ways. The first is to use discrete circuits, with output devices biased at higher currents. This produces excellent results, but at a large penalty in bulk, heat generation, and cost. Probably the best application for this solution is in very small mixers (2–4 inputs), or in large consoles that reside permanently in studio control rooms. For portable multi-input mixers, integrated circuit designs seem almost mandatory.

A second approach is to augment an IC with either a discrete or integrated high-current output buffer, which works within the feedback loop, and can be biased for Class A operation into all possible loads.[4] Such integrated devices as the Precision Monolithics BUF-02 and the Linear Technology LT-1010 are used successfully in high-quality audio circuits.[5]

The cost of using buffers in a multi-input mixer quickly becomes prohibitive. I prefer to bias the integrated circuit to higher output currents. First introduced in 1978 by the ubiquitous Walter Jung, this technique uses a pulldown resistor or (preferably) a

TABLE 1

PEAK MICROPHONE OUTPUT LEVELS

Open Circuit, dB ref. 0dBu (0.775V)

INSTRUMENT	DISTANCE (INCHES)	NEUMANN U-87 (condenser)	RCA 77-DX (ribbon)	EV 666 (dynamic)
Bass drum	6	0	-9	-11
Large tom-tom	12	-1	-9	-5
Small tom-tom	12	-1	-7	-1
Piano (one note)	6	-25	-38	-32
Orchestra bells	18	-16	-33	-30
Cow bell	12	-10	-29	-15
Loud yell	4	0	Not tested	-10

constant-current source between the output and the V-terminal to force the positive-side output transistor to a higher bias level, while shutting off the negative-side transistor.[6] In effect, this changes the output stage from a push-pull Class AB circuit, with a fairly low-current transition point, to a single-ended Class A stage. The bias point should provide adequate current for Class A operation into the worst possible load at the highest possible signal level.

The cheapest, but least linear, way to do this is to connect a resistor between the output terminal and V–; with ±20V supplies, for example, a 5k resistor will produce a 4mA standing bias current. However, a better approach employs constant-current sources rather than resistors. Gary Galo suggests using IC current sources (LM-334) or FETs (2N5457) to make uninvolved constant-current sources.[7] Since both these devices have maximum voltage limits of 30V, they are not useful to us. At the supply voltages you will use to ensure sufficient input headroom, you must produce your current sources from bipolar transistors. I recommend the readily available MPS-A05 and MPS-8098 (*Fig. 5*).

Biased? Who's Biased?

In an ideal world, all solid-state circuits could be direct-coupled, and coupling capacitors would be unnecessary. Contemporary circuits haven't quite achieved this. Coupling capacitors are still needed in some places in a mixer.

The problem arises when using pots. As current passes through the track and wiper of a pot, the track surface is slightly damaged. Eventually, it becomes noisy, intermittent, or produces a subsonic "whump" as the pot is turned up and down. This can happen if the pot has a DC voltage across it, as a result of a DC offset in the amplifier immediately upstream. Using a trimpot to null out the offset voltage may work if the amplifier is stable. However, since voltage offset is often temperature-dependent, and mixers can be used in a variety of environments, I prefer using either a coupling capacitor or a servo-amplifier circuit to keep DC off the pot from that end.

The other end is more circumscribed. When a pot is followed by a bipolar-input amplifier (integrated or discrete), there is a substantial bias current at the amp input terminals. (FET-input amplifiers and tubes have negligible input currents.) If the pot is direct-coupled, this current flows through the track and wiper, and the pot will become noisy. Two problem spots in this design are the channel faders and the panpots.

To reduce noise, the secondary gain amplifier that follows the channel fader in a multi-input mixer must be at least as quiet as an NE-5534a op amp, so you need a bipolar circuit. I know of no FET- or tube-based circuits quiet enough for this application.

Unfortunately, bipolar integrated circuits that will work in this location have typical bias currents ranging from 40nA (LT-1028A) through 500nA (NE-5534a), while the discrete JE-990 has a bias current of 2,200nA. All are too high for direct coupling, so you will need a coupling cap to separate the DC from the pot. (You can use a direct-coupled Borbely-type discrete FET circuit as the secondary gain stage in a single-channel mike preamp or a three-input mixer, as the noise requirements aren't as stringent.)

The panpots present a similar problem. They feed the main left and right summing amplifiers, and the bias currents from these amplifiers can make the panpots noisy. Unfortunately, this is a difficult place to incorporate coupling capacitors.

There are two possible solutions. You can sacrifice a couple of decibels of ultimate signal-to-noise ratio by using Borbely-type circuits for left and right summing amplifiers. (The superb audio performance of this circuit makes this choice attractive.) However, this circuit is large, and might not fit in a portable mixer.

The second solution is to accept a small amount of bias current. The LT-1028A produces only 40nA, less than one-tenth the current of a 5534a. If the panpots are used infrequently, they should last a long time. (Later, I'll discuss how to use panpots less frequently.)

Running Interference

The electromagnetic spectrum becomes more of a jungle each year. New users for the ether proliferate—UHF-TV, ham and CB transmitters (legal and otherwise), pagers, cellular phones, and other high-tech excrescences (even, saints preserve us, cordless mice). The radio spectrum is now so raucous that no serious audio equipment builder can ignore the need for RFI protection.

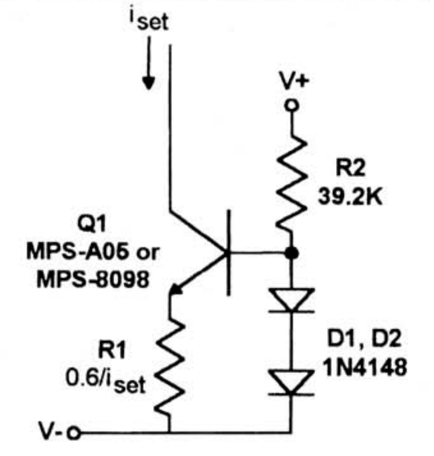

FIGURE 5: A simple constant-current source.

Along with this increase in spectrum pollution, there's an increase in garbage on the AC lines. To the ever-present problems of line transients, motor noise, and fluorescent-light starter noise, you can add SCR hash (from light dimmers) and digital noise (from computers and other microprocessor-based hardware, including digital audio equipment).

Drastic RFI protection measures are now necessary in the design of good audio equipment. Even if the offending signal does not produce obvious artifacts (such as radio broadcasts or TV sync-buzz appearing unbidden in an audio signal), RF can intermodulate with audio-frequency signals in a wideband circuit to produce potentially audible distortions.[8] I am convinced that many of the subjective differences between pieces of equipment that measure similarly on the test bench are related to RF bombardment.

I have used two laboratories to work on RFI problems. I designed home-audio equipment for my parents, and professional-level equipment for a community radio station. My parents live in a high-rise apartment two blocks from Chicago's John Hancock building, which is topped with a forest of FM, AM, VHF-TV, and UHF-TV antennas. The radio station spent its first few years broadcasting from a studio located 8′ from its 18kW FM transmitter. I quickly learned about the farther reaches of RFI treatment, including exotic ground systems, decoupling, and ferrite beads.

I also tested various input devices under these conditions, and discovered (as many have before me) that FETs are the least susceptible to RF problems, followed closely by vacuum tubes. Bipolar transistors are the worst, by a hefty margin. Therefore, I have a strong preference for building all input stages (defined as any stage that receives a signal from the outside world, including from a patch bay) using FETs or tubes.

In mixer design, this generates conflicts. The current design fashion is microphone inputs sans transformers (for reasons of cost, space, and putative audio quality). Unfortunately, both FETs and tubes give their best noise performances when driven by high source impedances, while professional microphones are low impedance (150–200Ω). A transformerless input virtually dictates bipolar input devices, while the bipolar devices' poor RFI performance, in turn, dictates the use of RC filtering networks at the inputs. These networks degrade the noise performance.

Balance

Almost all microphone inputs and professional line inputs use balanced circuits where two input lines carry identical signals of opposite phase. Interference tends to show up in phase on both lines; since the inputs respond to the *difference* between the two signals, they reject the interference while amplifying the signal. While most transformerless inputs do a good job rejecting interference at low frequencies (i.e., 60Hz hum), they deteriorate at higher frequencies. Often the rejection of RF in a transformerless balanced circuit is non-existent.

Well-made transformers can eliminate these problems. A 1:10 stepup transformer shows a 15–20k impedance to the first amplifying stage, which exceeds the minimum necessary for good FET or tube performance. Good transformers stay balanced well up into the megahertz region, and continue to reject RFI at frequencies that matter. They also provide passive bandlimiting, keeping troublesome frequencies out of the mixer circuits.

Transformers aren't panaceas. Until recently, even the best transformers degraded high-frequency transient response. A full-range audio transformer is an electrically complex device, with stray inductances and capacitances everywhere, and most transformers' phase responses are nonlinear. This created a nonuniform group delay characteristic which showed up as severe peaking at cutoff (sometimes in the audible band) and ringing on transients. Good audio transformers were also bulky, expensive, and could pick up hum via inductive coupling from the power transformers.

More recent transformer designs have solved almost all of these problems. In the late 1980s, Deane Jensen of Jensen Transformers produced improved versions that featured a "Bessel" characteristic for the high-end rolloff. Bessel-aligned transformers provide linear phase response, unvarying group delay, and no ringing or overshoot. Jensen's first Bessel-aligned transformer was the 1:10 ratio JE-115KE, which remains the best-sounding microphone transformer I have heard. It is well-shielded from induced hum, rejects RFI cleanly, and is superbly clear and transparent. It is also expensive.

It's important to apply anti-RFI measures thoroughly. I am a firm believer in ferrite beads—small inductors that slip over wires or resistor leads to provide a high impedance at radio frequencies. Use these on all input lines, and, equally importantly, on all power supply wiring to decouple RFI from the individual boards.

Power

It has become clear in the last few decades that audio circuits perform audibly better when they are fed by wideband, low-impedance, tightly regulated power supplies, which minimize crosstalk between stages and channels, as well as RFI problems.[9,10] Audio circuits behave more linearly when powered by such supplies. Multiple channels present a challenge: interaction between channels can become significant, and surprising amounts of RF garbage can be picked up on supply rails.

Decoupling networks at each card's supply inputs actually raise the impedance at high frequencies, which is undesirable. The best solution is to incorporate regulator circuits on every card. You can then decouple the card inputs, while the supply to the active devices remains low-impedance.

For solid-state circuits there are two options. Ben Duncan uses integrated regulators of the LM-317/337 variety, as does Erno Borbely in his excellent recent modules. I prefer the elegant op-amp-based regulator designed by Michael Sulzer.[11] Its output impedance is much lower at all frequencies, and when you add up all component costs, the price is comparable. Sulzer's regulator occupies a bit more board space, but I think it's worth it.

Ben Duncan has devoted considerable time to examining power supply designs. His analysis reveals that a typical raw supply is a forest of stray inductances and capacitances that produce an unpredictable array of RF resonances.[12] In addition to possible problems with RFI pickup, these resonances can be excited by the sharp transients caused as rectifier diodes switch on and off.

Similarly, the protection diodes usually installed around regulators can produce sharp resonances which couple garbage around the regulator and into the audio circuits. His supply designs use simple techniques to damp these resonances and avoid coupling. They make sense, and I've incorporated them into my own designs.

I decouple each board with ferrite beads and low-inductance ceramic disk capacitors. The latter have a poor reputation for audio use, but I've found they have no adverse effect when placed upstream from the regulator. In the case of integrated circuits, I also decouple each package with small low-inductance film capacitors, either polyester or (preferably) polypropylene, located close to the supply pins. Since I prefer to keep raw supplies away from active circuits, the cable from the power pack is shielded, and I string the wires emerging from the power pack connector in the mixer with ferrite beads.

Finally, both RFI and transient disturbances can sneak into audio circuits by way of the AC line. It's vital to prevent this by using interference filters at the AC source, and wiring 150V metal-oxide varistors (MOVs) across the AC line (use a 300V MOV if your nominal line voltage is 220V).

Component Choices

Over the years I've noticed that the use of high-quality passive components makes a major difference in the sound. I have rebuilt mixers, preamps, amplifiers, and tape decks replacing 5% carbon composition or carbon film resistors with 1% metal film resistors, and heard (along with the expected improvements in channel-to-channel matching) dramatic improvements in clarity, focus, and noise level. Metal film resistors have measurable electrical advantages: lower voltage coefficients, lower inductances, and less excess noise. Better sonic performance results.

I've experienced good results with several brands. The most readily available high-quality resistors seem to be Holco and Roederstein, which are sold in most of Europe, Asia, and US. Whenever possible, it's a good idea to use ½W, rather than ¼W or 1/8W, resistors, as voltage coefficients are lower. (For more on resistors, see Ben Duncan's series.[13])

Capacitors are remarkably complicated, and their sonic imprint on sound is considerable. Again, when you rebuild equipment with polystyrene and polypropylene capacitors replacing polyesters and (where possible) electrolytics, you will hear improvements in focus and clarity. I prefer to use as few capacitors as possible, employing low-offset circuits and offset-trimming pots to eliminate the need for coupling caps. In a mixer or mike-preamp design, I try to use only one or two coupling caps in the audio circuits and several small polystyrenes for high-frequency rolloffs and EQ.

Good capacitors are less universal than resistors. Wima and Roederstein make good ones, which are distributed throughout the world. Panasonic makes excellent low-inductance electrolytics (HFQ series) and ceramic disks that work well for power supplies and bypassing. In this country, I've had excellent luck with Rel-Cap and Mouser polypropylenes, but I'm not certain about their international availability. (Refer to Ben Duncan for more on the electrical quirks of these "strange devices."[14])

Switches and jacks can also affect the sound, as their contacts age and form oxide films. Like most metal oxides, these films make excellent rectifiers. I've measured 1.5% intermodulation distortion, independent of level, from a set of oxidized contacts on an old pair of jacks. When the jacks were cleaned, the distortion dropped to 0.065%, the analyzer's residual. Good gold plating can prevent contacts from oxidizing and ensure good performance for the lifetime of the equipment.

I've observed that contact nonlinearities affect low-current inputs. Mike connectors, in particular, are quite susceptible. I've heard the same microphone, an Electro-Voice RE-16, turn from sibilant and harsh-sounding to smooth and sweet after its contacts were cleaned with Cramolin. (This mike's output is through easily oxidized silver-plated contacts, not gold.) Excellent gold-plated connectors are available from Switchcraft (QG-P series) and Neutrik.

Similar considerations convinced me to avoid plug-in modular construction in multi-input mixers. Although it makes servicing more difficult, I hardwire signal and power connections wherever possible. I use either Teflon®-insulated, nonsilver-plated wire or foil-shielded, polypropylene-insulated cable (Belden 8450). The latter also makes good hookup wire when you pull it out of its shield, although the insulation's low melting point makes it annoying to solder. I prefer single-strand wire, both for ease of handling and for its lack of interstrand rectification problems.

Pots are a real problem. Making a consistently good device from a metal slider traveling over a resistive surface (in the presence of polluted air), while maintaining a consistent logarithmic taper, is quite an achievement. Good-quality audio-taper pots are difficult to find and quite expensive in the US. You can obtain Alps rotary pots, at a price, from several sources (including Old Colony), and the small versions sold by Radio Shack are surprisingly good. High-quality slide pots, on the other hand, are nearly impossible to find at a reasonable price. Penny & Giles faders, the industry standard, cost more than $50 apiece in small quantities. Under these circumstances, I would choose good-quality rotary pots over mediocre slide pots, despite the loss in convenience.

My global approach to this array of problems is to simplify the signal path as much as possible. I use hard-wiring between boards, solder in all active components (except tubes), and use the minimum of switching consistent with flexible operation. My revised block diagram (*Fig.* 2) shows that, with the EQ switched off, most signals pass through only two high-quality switches, five active stages, two coupling capacitors, and two pots (three if the panpots are used) between input and output. (A couple of channels might have an additional switch to provide line inputs.)

"Oh," as Columbo might say, "one more thing, ma'am." The metal shafts and knobs on pots, when touched, provide an excellent conduit for RFI into the circuitry. Use plastic knobs, OK?

Level Check

Unless you enjoy twisting your head to watch the level indicators on your tape recorder, put some on the board. Typically, they fall into two groups: mechanical meters and LED arrays. The latter, in turn, can be divided into peak-reading and average-reading circuits. (True VU meters are inherently averaging.)

I'm old-fashioned; I still prefer a set of big, well-lit VU meters to an array of flashing lights. In the first place, I prefer average-reading circuits to peak-reading. The perceived loudness of a signal correlates far more strongly with its average level than with its peak level. I recall my futile attempts to persuade engineers at an (unnamed) television station not to try to match audio levels using the peak-reading meters on an early Sony videotape recorder. They were producing audio levels (especially on speech) that could vary 10dB from moment to moment. While true VUs don't correlate perfectly with perceived loudness, they come closer than any other indicator.

I also distrust the discontinuous nature of LED arrays. Typically, I use a test tone to set up the board and recorder so the indicators on the board correlate with those on the recorder. To achieve any degree of precision with an LED array, I must tweak the levels until a particular LED just barely turns on. A moving needle on a VU meter is easier to adjust precisely. Finally, LED indicators are driven by high-speed comparators that switch on and off quite sharply, sending radio-frequency current spikes into the power supply. I'm leery of introducing another possible source of RF garbage into a piece of high-quality audio gear.

Of course, VU meters have their quirks, too. They include certain ballistic characteristics, as defined by an ANSI standard. (The cheap VU-type meters found bubble-packed at your local surplus store don't meet ANSI specs and don't provide a reliable level indicator for audio work.) To meet the standard, true VUs should be connected across a +4dBu line, in series with a specified resistor (nominally 3.6kΩ). This is fine, except that a VU meter is a decidedly nonlinear load and can cause significant distortion when you hang it across an audio signal line. Placing a buffer amp between the signal line and the meter solves this problem.

Unfortunately, real VU meters are prohibitively expensive; the current Newark Electronics catalog lists models ranging from $77–$135 apiece. I can't offer much advice to the builder on a low budget, other than to look in surplus outlets. (I was able to get mine surplus for about $5 each.)

To make matters worse, I suggest using three VU meters on a multi-input mixer: one each for the left and right channels, and one that is switchable to check the mono output, the stage monitor output(s), or the echo send.

Finally, what about peak indicators? Digital recorders do not overload gracefully. If you send a digital recorder over the 0dB point, you'll hear terrible noises. If I set my recorder so 0VU on the recording console equals −16dB on the digital recorder, and carefully watch the VU meters, I don't overload the tape. Still, peak indicators mounted on the mixer may be a good idea; if you're running into the overload region on the tape, any small distortion caused by a comparator switching on and off is purely academic.

These peak indicators, if present, should incorporate a "hold" circuit, so even a momentary peak flashes the light long enough to attract your attention. Also, they should be switchable to flash at +16VU for digital tapes or +10VU for analog. I've designed such a circuit, but I don't use it in my mixer. My DAT recorder has excellent peak-reading indicators, with a numerical readout of remaining headroom. I simply glance at it periodically.

Other Buses, Other Times

In live concerts, performers must clearly hear themselves and each

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other. A performance arena unearthed at Pompeii not only projects sound effortlessly from the stage to the audience, but also folds back the sound so the performer hears him/herself perfectly (*Photo 1*). Singing in this arena (I couldn't resist), I heard a better monitor signal than I've ever heard on a modern stage, and, since the system is entirely acoustic, there is no possibility of feedback. Even first century A.D. designers understood the importance of stage monitors.

In the present, most mixers incorporate some form of monitor mixing, sometimes called "foldback." This takes two forms, "pre-fader" and "post-fader." Pre-fader monitoring produces an independent monitor mix which doesn't change when the house mix is adjusted. This allows performers to hear only what they need to, which may be quite different from what the audience hears. For example, when I perform as a folksinger, my voice must be prominent in the monitor so I can stay on pitch, but I don't need to hear my guitar as much. On the other hand, post-fader monitoring produces a monitor mix that follows the changing levels as the house mix is varied. While not often needed, it's sometimes useful.

Monitor mixing is fairly easy to implement: simply hang extra pots/gain stages in parallel with the main fader, with a buffer to avoid loading the input amplifier. Since the signal-to-noise requirements are less stringent for stage monitoring (a performance stage can be a noisy place), you can have noisier op amps and higher-value pots than you could in the main audio chain. In this mixer design, the pre-fader monitor mixes are, of necessity, upstream from the EQ amplifier. If an equalized mix is required, it must be post-fader.

Although I prefer not to use audio processing in my performances, many musicians do, and include digital delay and echo devices. A separate mix bus should be present on the mixer to feed signals into these devices. Place it post-fader, so the echo signal follows the fader setting. The return signal from the signal processor must be mixed into both the stereo and mono mixes.

Since most of these devices are mono, you should add it equally to both left and right channels. In turn, this dictates that any signals to which processing is applied should be panned to the center of the mix, so the main signal and echo come from the same place. However, that applies to the art of recording, and this article is about the science and technology of mixer design. See how they interact? In Part 2, I'll discuss some alternative designs for recording/mixing equipment. □

PREVIEW
Glass Audio

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- Diff Amp Driver With BJT Source
- Balanced Input Phono Preamp
- Old-New Bias on Bias

Part 2

MIXERS AND MIKE PREAMPS

By Paul Stamler

Part 1 (*TAA* 3/95, p. 8) discussed some of the theoretical problems with conventional designs for microphone preamps and mixing consoles, and the practical problems (human and electrical) that arise when using these designs under trying circumstances. In this article, I will describe some alternative designs that I've found useful. Please be aware that the circuits suggested here are building blocks: although originally intended for portable recording/mixing equipment, you can combine them in various ways to produce stand-alone mike preamps, small portable mixers, or even large, anchored studio mixing consoles.

You can build these basic circuit designs in several ways, using integrated circuits, discrete solid-state circuits, or even vacuum tubes. I do prefer tubes, and I've made some excellent recordings using a pair of mike preamps built onto the chassis of a Dynaco PAS-3 preamp. But designing with tubes presents a different set of challenges than designing with solid-state devices. In particular, loading issues become more difficult.

My thoughts about tube design are changing as I write, thanks to some interesting new circuit ideas recently published in *Glass Audio.* This article concentrates on solid-state designs.

Input Stage

Let's look at a simple, stand-alone microphone preamp. You can use a pair of these with a pair of crossed or spaced microphones for recording a performance in true stereo: a symphony orchestra, a chamber group, or a folksinger performing without a PA system. For simplicity's sake, I assumed that the output will feed directly into a DAT recorder, operating at standard "consumer" level (nominally –10dBu, unbalanced). This design adheres to the "chimp-proof" principles outlined in Part 1. All levels are set using the fader, which is located between the input stage and the secondary gain stage.

As I discussed in Part 1, the input stage of any microphone mixer must handle an input level of 0dBu without overloading. Since integrated circuits powered by ±21.5V rails (or discrete circuits powered by ±24V rails) can yield up to +24dBu without clipping, a fixed-gain input stage can have no more than 24dB of voltage gain.

Figure 6 shows such a stage. R101 and R102 provide phantom power for condenser microphones with internal amplifiers, and must be matched to at least 0.1% for lowest hum pickup. The input transformer, the Jensen JE-115KE, provides 20dB of voltage stepup, so the amplifier needs to provide an additional 4dB, or 1.58×. Unlike earlier versions, the current JE-115KE doesn't require an 80kHz rolloff in the amplifier; the 200kHz rolloff produced by C103 helps suppress RFI pickup and enhance the amplifier's stability.

C106 couples the amplifier to the fader, producing a low-frequency rolloff at 100Hz to compensate for microphone proximity effect, or to filter out room noise. S101 switches either C107 or C108 in parallel with C106, moving the low-frequency rolloff point to either 30Hz or 1.6Hz (effectively flat). If the amplifier's offset voltage is low enough, C108 can be replaced by a length of wire, so the low-frequency response is limited only by the transformer (the JE-115KE is flat down to below 4Hz, where my voltmeter gives up).

The input resistance in this circuit has two components: the source impedance of the microphone, as stepped up through the transformer, and the parallel resistances of the feedback network, R104 and R105. Typical professional microphones have an output impedance of 150Ω, essentially resistive. When loaded with the phantom-powering resistors, this becomes 148Ω.

The transformer has a voltage stepup ratio of 1:10; the impedance is stepped up by the square of that ratio, or 1:100. The microphone's impedance thus becomes 14.8k. This, in turn, is loaded by R103, the transformer's

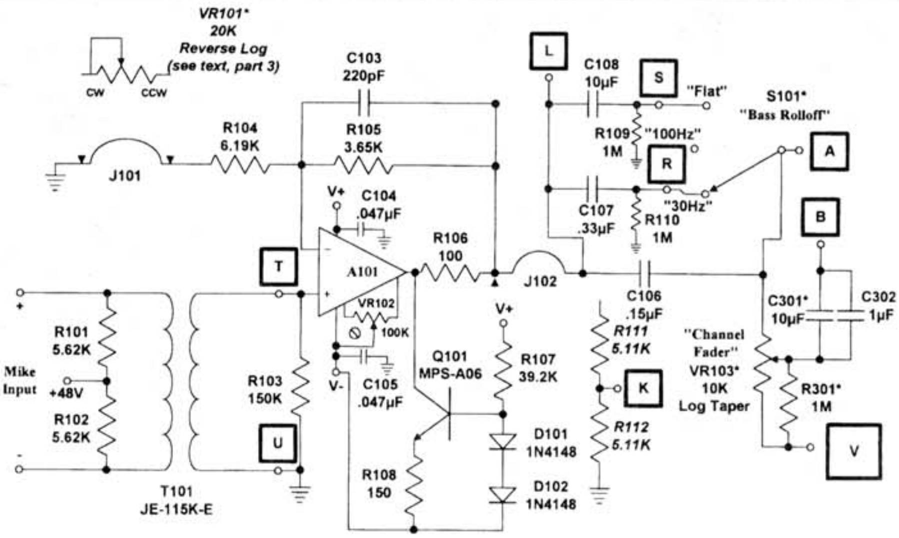

FIGURE 6: Input stage. Normally VR101 is replaced by a jumper, J101. If you choose the variable-gain option, include R111-112 and delete J102; connect terminal K to terminal C on the EQ amp, and terminal L to terminal M on the EQ amp. The current source components and trimpot aren't used when A101 is a Borbely amp; the trimpot isn't used with a 5534a. Match R101-102 to within 0.1% for best common-mode rejection. If the secondary gain stage *(Fig. 7)* uses a Borbely amp for A301, then R301 and C301–C302 may be deleted. Components marked with asterisks are located off the board.

150k terminating resistor, for a total effective source resistance of 13.5k. (For the technically alert: yes, I'm aware that it's really an impedance, not a resistance, but for many modern microphones it's so close to a pure resistance that it makes no real difference.) The parallel resistances of the feedback network add up to 2.3k, so the total input resistance is 15.8k.

Table 2 describes the performance of several well-known, low-noise devices in this circuit. Columns one, two, and three give the noise contributions of the three noise sources present: thermal or "Johnson" noise, voltage noise, and current noise. These figures are expressed in microvolts. Adding the squares of these numbers and taking the square root gives the results in column 4, the total equivalent input noise in microvolts (the dBu equivalent is column 5.) As a reference point, I've included the calculated performance of a perfect, noiseless preamp with only source resistance "noise."

Circuit & Chip Choices

The LT-1028A IC and the identically performing Jensen JE-990 discrete amplifier are washouts for this application. They have very low voltage noise specs, but their current noise makes them more appropriate for operating with source resistances in the 450–1.1kΩ range. For the higher resistances presented by this transformer, we need either a bipolar amplifier with very low current noise or an FET amplifier with virtually none.

We then have three usable alternatives: Erno Borbely's discrete Class A circuit, with low-noise FETs on the inputs; the long-in-the-tooth but still useful NE-5534a (known outside the US as the TDA-1034a); and a new IC from Burr-Brown, the relatively low-noise FET-input OPA-604. In Part 1 I described my preference for FETs in any part of a circuit that receives signals from the outside world. Their inherently linear response extends far into the regions of RF garbage, and they show much less tendency to turn into radio receivers under difficult RFI conditions.

In a stand-alone mike preamp, I recommend using the Borbely circuit. The noise figure is only 1dB worse than a perfectly noiseless preamp, and the circuit shows astonishingly low distortion figures in a variety of tests.[1] The output transistors drive the fader in Class A without difficulty, and the ±24V supplies allow enough headroom to accept a 0dBu signal at the microphone input without clipping.

The major drawbacks of the Borbely circuit are bulk and heat generation (less of a problem in a single- or two-channel preamp than in a large mixer), expense, and difficulty in locating parts. Several of the active devices are notoriously difficult for American builders to buy in small quantities, and I suspect they may be even harder to find in some other parts of the world. (Erno Borbely will sell you the parts no matter where you live, but the cost is high.) Still, I'd recommend the Borbely circuit over the others if you can afford it.

Keep in mind that my drawing shows A101 as a functional block. If you use the Borbely circuit for A101, remember to include the output stabilizing network (R21-C18 in the EB-489/104 circuit), and to omit the current-sourcing components (Q101, R107-108, D101-102), R106, and VR-102.

Given my preference for FET inputs, the OPA-604 would be my (unsurprising) second choice. This IC has reasonably low voltage noise, essentially no current noise, good high-current output stages that will handle ±21.5V supplies, and a reputation for low-distortion performance in a variety of applications. In addition, the offset voltage is extremely low and can be nulled out with a good-quality trimpot. This allows you to DC-couple the amplifier to the fader by replacing C108 with a length of wire. (Even unnulled, the typical offset is only 0.2mV; the offset drift with temperature is sufficiently low that as the ambient temperature changes from 15° to 50° Celsius, the offset will change by less than 0.3mV.)

The noise figure is still very good, only 2dB worse than a perfect preamp, and 1dB worse than a Borbely. Burr-Brown is a large corporation, with worldwide distributors, so you should be able to find the chip in most places for many years to come.

Finally, there is the much-maligned NE-5534a. This chip and its cousin, the NE-5532 dual package, received much abuse from audiophiles over the years, because they've been used in many poorly thought-out D-to-A converters. I contend that, although the 5534 and 5532 aren't the best op amps for audio any more, they are far from the worst.

The noise performance is excellent, falling between that of the Borbely and the OPA-604. The input bias current, however, is very high (remember this is a bipolar-input chip). I don't recommend DC-coupling this IC to the fader unless you use a servo-amplifier. Newer designs such as the OPA-604 have a better reputation for audio performance, which my own experience bears out. Remember that the bipolar 5534a is potentially subject to more problems with RFI than an FET-input device. Use the 5534a only if you can't get the OPA-604.

Loading

The amplifier in this stage sees six loads in parallel:

- the main fader, VR-103 (10k);
- the series combination of the resistors in the gain-setting network, R104 and R105 (9.84k);
- the two resistors that drain the free ends of the switched coupling capacitors, R109 and R110 (1M each);
- the bypass resistor for the main fader, R301 (also 1M);
- the input resistor for the second stage, R302 (100k).

All of these add up to a 4.66k load (4.55k in a multi-input mixer, which adds the 200k input resistance of the monitor buffer amplifier).

TABLE 2

INPUT NOISE
INPUT REFERRED

AMPLIFIER	JOHNSON NOISE (μV)	VOLTAGE NOISE (μV)	CURRENT NOISE (μV)	TOTAL NOISE (μV)	EQUIVALENT INPUT NOISE (dBu)
NE-5534a	2.29	0.495	0.892	2.51	−129.8
OPA-604	2.29	1.48	Negligible	2.73	−129.1
LT-1028A	2.29	0.120	2.23	3.20	−127.7
Borbely	2.29	0.651	Negligible	2.38	−130.3
Noiseless	2.11	0.00	0.00	2.11	−131.3

What is the maximum signal the amp must drive into this load? Well, the amp is designed to clip at +24dBu, or 12.3V RMS. This translates to a maximum peak voltage of 17.4V, putting a maximum of 3.82mA into our worst-case load. All the ICs we're considering will produce this current without help, but they will no longer be operating in Class A.

A typical IC chip has its output transistors biased with about 0.2mA of standing current; any higher current demand forces one of the transistors to operate in its cutoff region. As I discussed in Part 1, I prefer to operate circuits in Class A. A 4mA constant-current source (Q101 and its associated components) connected to the IC's output changes the output circuit from push-pull Class B to single-ended Class A, a technique that has been profitably employed elsewhere, with good sonic results.[2] (The Borbely circuit is already generously biased into Class A.)

Stage Two

In Part 1 we established that –60dBu is the lowest nominal output level for a practical microphone. Raising this signal to a nominal "consumer" line level of –10dBu requires a preamp gain of 50dB. Since the first stage provides 24dB, the second stage must deliver 26dB of gain (20×).

Figure 7 shows this stage. C301 and C302 keep the amplifier's bias current from flowing through the fader, while R301 keeps the capacitor's free end tied to ground in case the fader develops an open spot. (If you use an FET-input amplifier, such as the Borbely circuit for A301, you can omit C301-302 and R302.) IC302 and its associated components form a noninverting DC servo amplifier, which keeps the amplifier's offset voltage low enough to allow DC coupling. Allowing 16dB of headroom over nominal level, the maximum peak output level is 2.19V.

Assume that, in the worst case, the output is connected to a midrange equalizer whose control is set for maximum boost, and also to the input of a DAT recorder with an input resistance of 10k. Then, the load on the amplifier (including its own feedback network) is 822Ω; the maximum current drawn is 2.67mA. Again allowing for a bit of leeway, 3.7mA of output-stage bias current, provided by Q301, will ensure Class A operation even under these conditions.

Any noise generated in this stage will be amplified by 26dB, and the noise performance will vary at different fader settings. *Table 3* and *Fig. 8* show the performance of various combinations of devices in a single-channel preamp.

The results are clear-cut: the Borbely amplifier, the LT-1028A chip and the identically performing Jensen JE-990, and the NE-5534a all produce excellent noise performance, while the OPA-604 performs less well. Again, my preference is the Borbely. Its distortion is spectacularly low, and its noise performance in this stage is quite acceptable. The FET inputs also allow the omission of the input coupling caps, always a good idea when possible. If the extra decibel of noise is important,

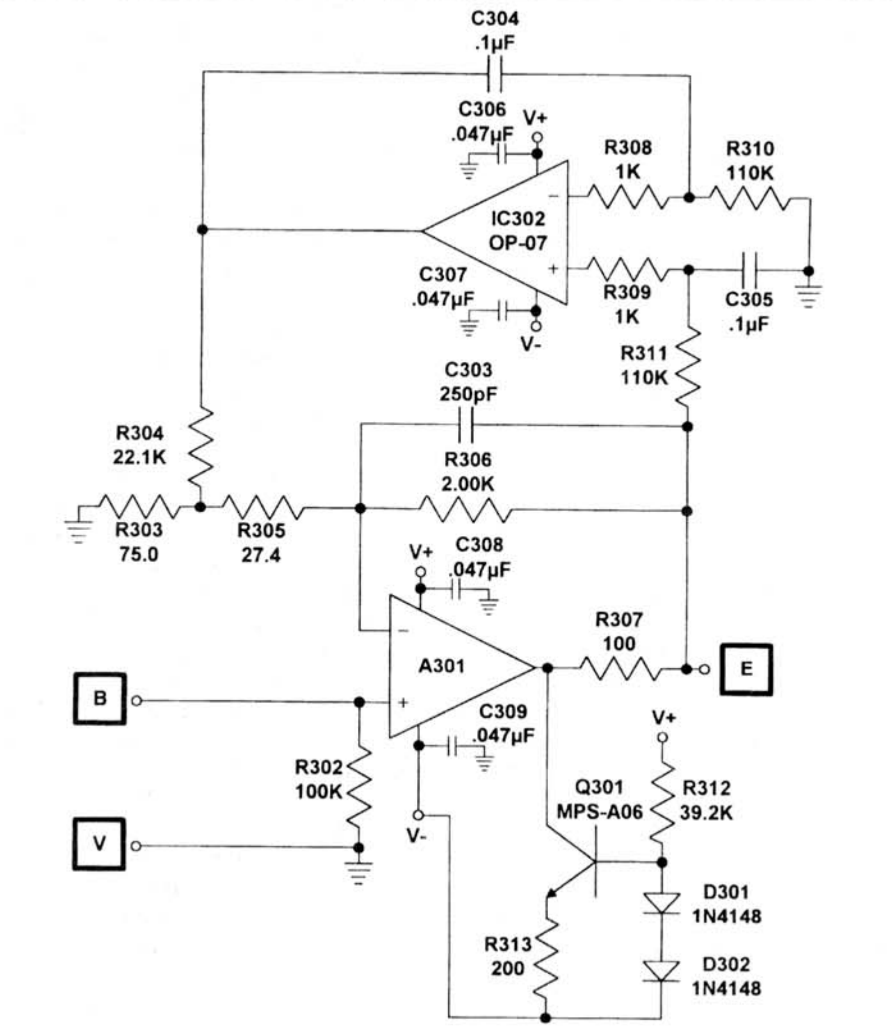

FIGURE 7: Secondary gain stage. If you use a Borbely circuit for A301, then delete the current source circuitry. If you use a 5534a, change R310-311 to 1.00M, R304 to 1.62k, and R305 to 30.1Ω.

TABLE 3

TOTAL NOISE
SINGLE-CHANNEL MIKE PREAMP

	604/1028A		5534a/5534a		BORBELY/BORBELY		604/604	
FADER POSITION	NOISE (μV)	dB REF. –10dBu	NOISE (μV)	dB REF. –10dBu	NOISE (μV)	dB REF. –10dBu	NOISE (μV)	dB REF. –10dBu
0	89.7	–68.7	83.5	–69.4	80.2	–69.7	94.5	–68.3
–6	53.4	–73.2	51.1	–73.6	50.2	–73.8	60.6	–72.1
–12	36.8	–76.5	36.8	–76.5	37.1	–76.4	46.9	–74.4
–18	29.8	–78.3	30.9	–78.0	31.8	–77.7	41.9	–75.3
–24	27.0	–79.2	28.5	–78.7	29.7	–78.3	40.0	–75.7
–30	25.8	–79.5	27.5	–79.0	28.8	–78.6	39.3	–75.9
–36	25.3	–79.7	27.2	–79.1	28.4	–78.7	39.0	–76.0
Closed	24.9	–79.9	26.7	–79.3	28.0	–78.8	38.7	–76.0

my second choice is the Jensen JE-990, or in an integrated circuit the LT-1028A, keeping the coupling caps in place for either.

If you can't obtain the LT-1028A, the NE-5534a is the fallback IC. However, there are a few special considerations if you use this device. The basic design for the stage biases the output transistors at 3.7mA. With the NE-5534a's normal standing current of 4.5mA, this means the package dissipates 353mW when powered by ±21.5V supplies. The LT-1028A doesn't mind this much heat generation, but the NE-5534a's maximum-rated dissipation is 300mW at lower temperatures, and 247mW at 45° Celsius (not an unreasonable temperature in real-world conditions).

To keep the chip from cooking, I suggest biasing the output transistors at only 1mA instead of 3.7mA, thereby limiting the package dissipation to 237mW. Unfortunately, this means that the stage ceases to operate in Class A for signals more than 7.5dB above nominal level. You can see why the NE-5534a is my least favorite chip in this application.

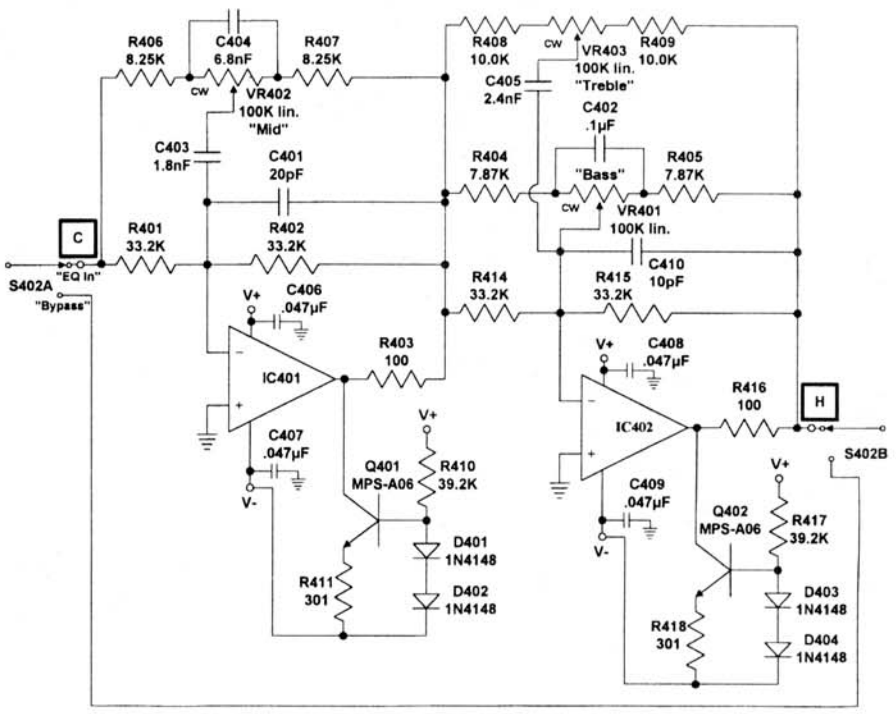

FIGURE 9: EQ stage for a stand-alone mike preamp.

Equalizer Circuits

I seldom use an equalizer in simple two-microphone setups, which are the principal application of a stand-alone mike preamp. Instead, I rely on microphone choice and positioning to obtain the tonal balance I need. (I do sometimes use the high-pass filters to eliminate room rumble and other low-frequency problems.)

Equalizers are occasionally useful, though, and it's a good idea to have them handy, provided they don't degrade the sound when not being used. I've included a simple equalizer in the preamp unit, with a bypass switch that connects the recorder or output stage directly to the output of stage two when the EQ isn't needed.

Most equalizers are variants of Peter Baxandall's famous tone controls, designed in the 1950s. They work well and don't require fancy tapers on the control pots, but they do invert the polarity of the signal. This can be a problem in a circuit where the EQ is switched in and out. You don't want your polarity to reverse every time you throw the switch. There are two solutions to this problem: either keep the amplifier in circuit all the time, switching the EQ components in and out as needed, or use two equalizers so the second circuit restores the inverted signal to normal.

For the stand-alone preamp I chose the latter solution. Since I seldom use the equalizers in this application, I prefer to keep the most-used signal path (EQ out) as short and simple as possible. (In multiple-input mixers, my choice goes the other way.)

The equalizers themselves (*Fig. 9*) are fairly standard designs, lifted with minor modification from Walter Jung's *Audio IC Op Amp Applications.*[3] I lowered the amount of boost and cut to 10dB, since I can't imagine a situation requiring more EQ than that.

For active devices, the OPA-604 is the obvious choice: the FET inputs allow DC coupling to the pots; the offset voltage is low, and can be nulled further by a trimpot; and the chip will work with up to ±24V supplies, allowing it to share a regulator with a Borbely amplifier.

The Borbely amp also is an excellent choice, but it seems like overkill in this application. Input bias currents rule out the NE-5534a and probably the LT-1028A; alas, most of the other FET amplifiers out there are rated at ±18V maximum supply voltage. (A possible exception is the CA-3140, an early FET

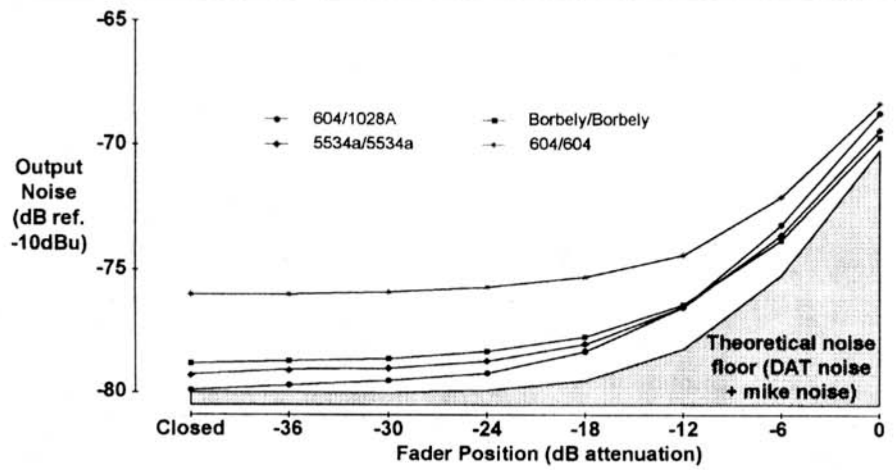

FIGURE 8: Output noise for a single mike preamp, using various active devices. In the legend, the devices are listed in the order "Input amp/secondary gain amp."

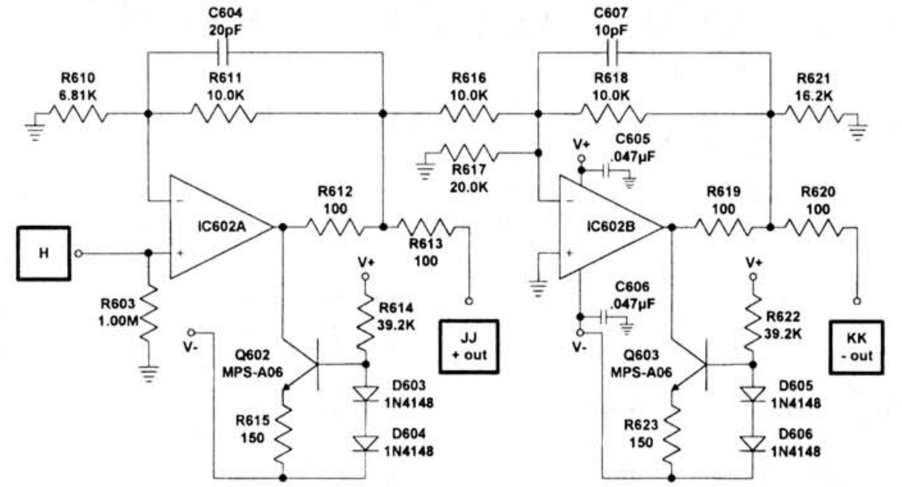

FIGURE 10: Balanced output stage for a stand-alone mike preamp.

design that sounds very good with proper attention to its loading.)

The loading on the first amplifier is fairly light. In the unlikely event that the midrange control is turned to maximum cut, while the following bass and treble controls are at maximum boost, the worst-case load is 2.44k. As with the second stage, the highest possible output signal is 2.19V, which translates to a maximum output current of 0.9mA. Output transistors biased to 2mA should keep the amplifier comfortably in Class A.

The second EQ stage's loading is similar. If you set the bass and treble controls to maximum cut and hang a DAT machine with an input resistance of 10k on the output (along with a balanced output stage), the worst-case load is 2.58k. The maximum output current is 0.85mA; again, a 2mA output bias is ample.

Output Stages

Up to this point I've assumed that the mike preamp is feeding a standard "consumer" input: an unbalanced connection with a nominal level of –10dBu and an input impedance of 10k or higher. I still prefer to work into this type of interface. The higher voltage (and current) demands of "professional-level" inputs can lead to higher distortion levels from active devices. Since I seldom need to feed long, unshielded lines from a simple mike preamp, I don't need the noise rejection that comes with a balanced interface.

In recent years some audiophiles have claimed significant improvements in sound when preamps and power amps are run with balanced connections. They may be right, if they use a balanced, floating connection to isolate power amp ground currents from the low-level circuits of a preamp. However, I've never heard any improvement when connecting line-level devices (i.e., mike preamps and tape recorders) to each other with a balanced cable, and, if possible, I prefer to avoid the additional active circuits and/or transformers necessary to produce a balanced output.

Still, you may think (or hear) differently, and it's worth having a balanced output available in case you need it, so I've designed one (*Fig. 10*). The circuit is simple and straightforward. While it is balanced, and operates at the standard professional studio level of +4dBu, it is not "floating"; that is, the signal remains tied to the mixer's system ground, which connects to the ground of the tape recorder. For the rare occasions when I need floating outputs, I keep a pair of high-quality 1:1 isolation transformers in my kit.

Loading is problematic. Standard studio practice in the past dictated designing equipment with a balanced, floating input, transformer coupled, with a "terminated" input impedance of 600Ω and an operating level of +4dBu. No longer. Most pro equipment now uses "bridging" inputs, usually involving differential amplifiers instead of transformers, with an input impedance of 5k or greater. Under these conditions, assuming we still allow 16dB of headroom over nominal operating level, the worst-case output current from each half of the output circuit is 2.4mA. An output bias of 3.5mA will be sufficient.

Driving a 600Ω line requires 18mA per side. The OPA-604 chip will deliver this easily, but not in Class A. Most of the time, I suspect this won't matter. These days, the only time you're likely to run into a 600Ω impedance is when you're driving a telephone line, which usually sounds terrible to begin with, so Class A operation becomes a moot point. If you need high-quality Class A operation into 600Ω, incorporate a high-current IC buffer (such as the LT-1010) into the feedback loop of each output amp, or use a pair of Jensen amplifiers instead of ICs.

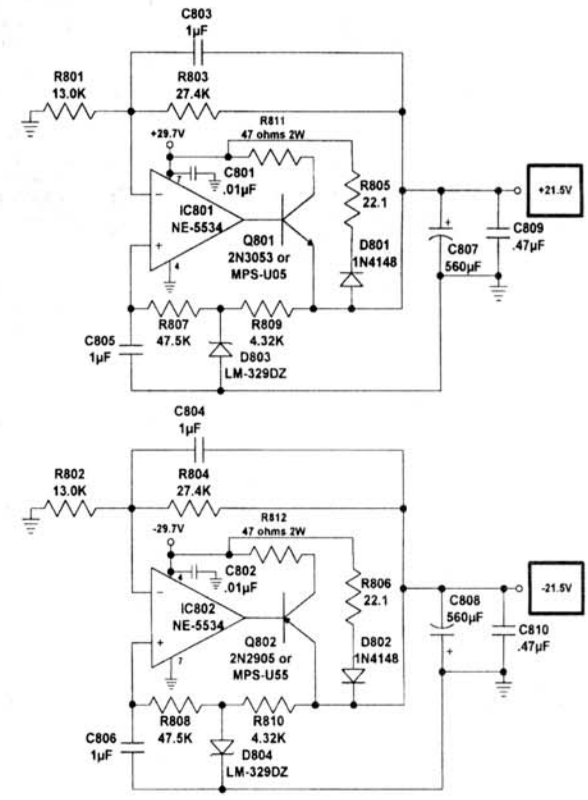

FIGURE 11a: Rail voltage regulators. You may substitute 1N4736A zener diodes for D803-804. If you need 24V outputs, change R801-802 to 11k. Use ferrite beads on the +29V and –29V wires from the preregulator card.

Supply Regulators

As I discussed in Part 1, I've become a convert to on-card power supply regulation. On-card regulation is usually implemented using IC regulators; however, when I calculated the cost of regulator circuits using LM-317/LM-337 adjustable regulators (fixed regulators have unacceptably high output impedances), I discovered that the total was very close to the cost of a Sulzer op-amp-based regulator and associated components. Since the Sulzer regulator has a generally lower output impedance than IC regulators, the choice was clear.

Figure 11a shows my version of the Sulzer regulator.[4] It's essentially stock, except for changed resistor values to reflect a different zener diode and a higher output voltage. The zener is actually a synthesized zener, the LM-329, with a stable output voltage of 6.9V. If you're on a tight budget, or can't get the LM-329, then a standard 6.8V 1W zener will do. The resistor values shown produce an output voltage of ±21.5V (perfect for ICs) when used with an LM-329. If you're using a Borbely or Jensen amplifier, which run from ±24V rails, change R801-802 to 11k. (Note that the OPA-604 op amp will operate on ±24V rails and can therefore share a regulator with the discrete circuits.)

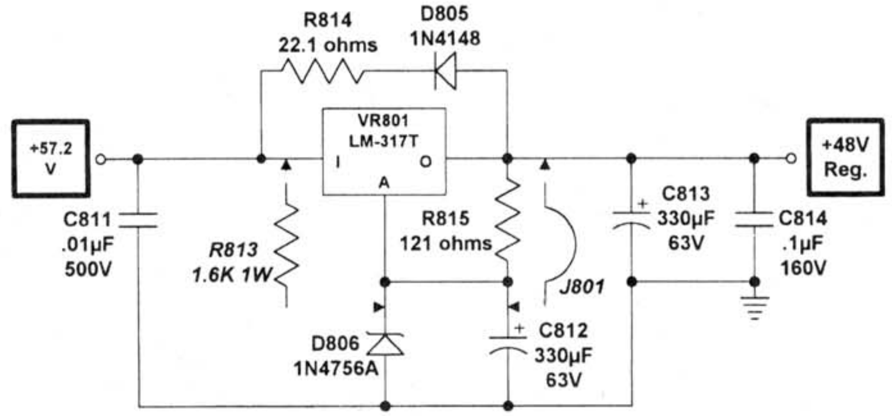

FIGURE 11b: Phantom power voltage regulator. If you prefer a simple zener-regulated circuit, delete VR801, R814-815, D805, and C812, and include R813* and J801*. Use a ferrite bead on the +57.2V wire from the preregulator card.

I changed Michael Sulzer's original design by avoiding tantalum capacitors, which have a bad sonic reputation in audio design circles (deservedly, in my opinion) and a poorer long-term reliability record than I prefer. I chose high-quality, low-impedance, aluminum electrolytics for supply decoupling, and polypropylene film bypasses to minimize RF problems. For the important rolloff cap in the regulator's feedback circuit, I chose another polypropylene unit for good high-frequency performance and maximum long-term stability. Michael Sulzer suggests substituting electrolytic capacitors of much higher value for tantalum caps, to minimize impedance at high frequencies.

I don't recommend this for the feedback and zener decoupling caps. When I tried it, the circuit latched up at turn-on. This stressed and blew the pass transistors. Modern high-quality aluminum electrolytics have sufficiently low high-frequency impedance that you can substitute them for tantalum caps of equal value.

If you build a preamp using Borbely's EB-489/104 cards, you may elect to use the LM-317/337 regulator circuits laid out on the cards. While the impedance isn't as low as a Sulzer regulator's, you have the advantage of separate regulators for each stage (you can power the EQs and the output stages from the second stage's regulator or use a separate Sulzer regulator).

The original Sulzer regulator includes a pair of resistors at the input for RFI protection and current limiting. In his revised version, he suggests deleting these resistors and using an IC preregulator. I've split the difference. Although I've used a preregulator (see next section), I prefer to keep the resistors. The pass transistors I use here, while excellent, are a bit fragile, and I have seen them blow on turn-on, as the current rushes into the electrolytic decoupling capacitors. For maximum reliability, and extra RFI decoupling, keep the resistors. Use a heatsink on the pass transistor.

You can substitute MPS-U05 and MPS-U55 transistors in this circuit. However, they are hard to find, so they are not my first choices for that reason. Still, they can withstand a lot more abuse than the specified transistors. If you expect to be abusing these regulators, you might wish to locate these transistors. *Figure 11b* shows the phantom supply regulator, which floats an LM-317 on a zener diode. (If you wish to economize here, the simple zener regulator shown as an alternative will probably work as well most of the time.)

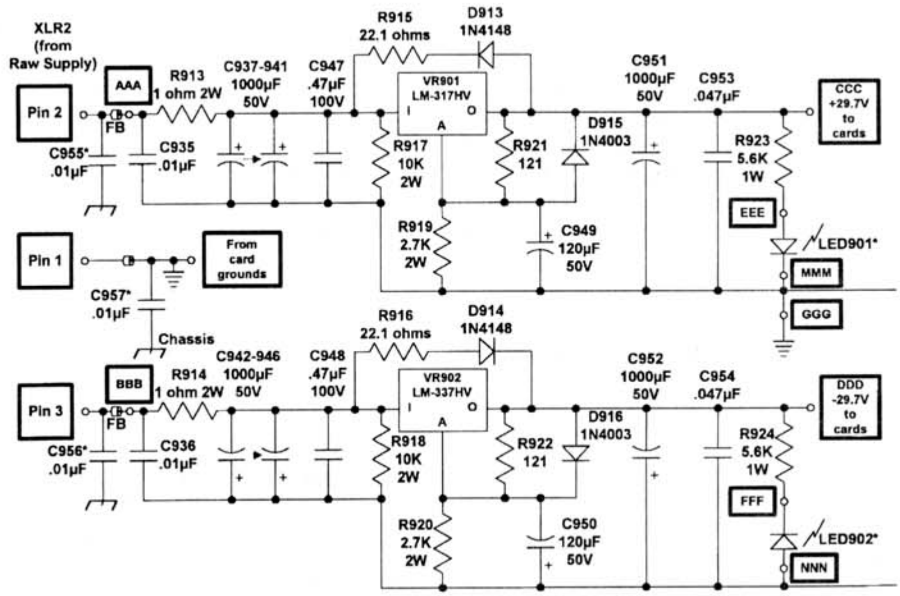

FIGURE 13a: Preregulators for rail voltages. If the final regulated voltage is 24V, change R919-920 to 3k, 2W.

Raw Supplies

The English designer Ben Duncan spent much time investigating the sonic impact of power supplies on audio equipment and reached some provocative conclusions. I won't summarize all his findings, but the gist of them is that parasitic capacitances and inductances in diodes, capacitors, and other devices form networks that are highly resonant at radio frequencies. These inadvertent resonances can be

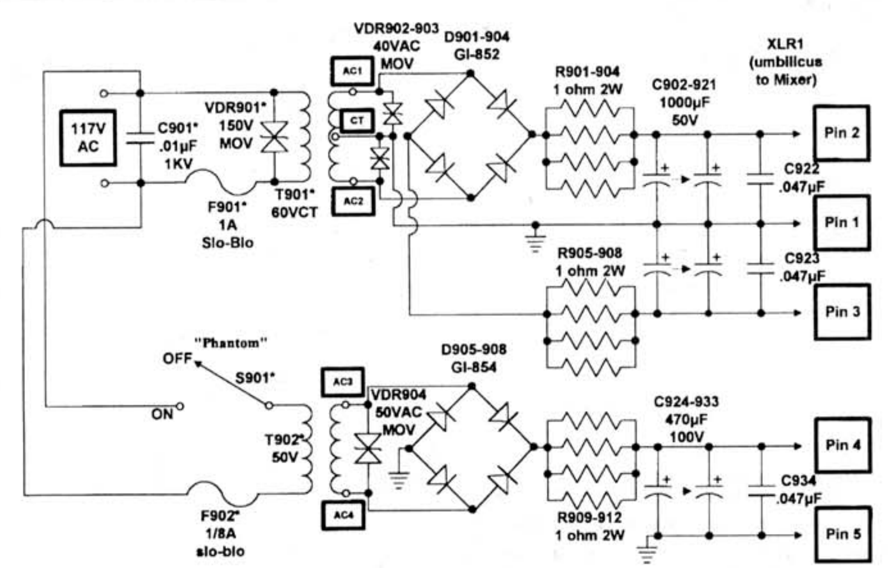

FIGURE 12: Outboard raw power supplies. Components marked with asterisks are located off-board.

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1. E. Borbely, "Measuring Nonlinear Distortions, Part III," *TAA* (3/90), 8.

2. G. Galo, "Ask *TAA*," *TAA* (2/90), 47.

3. W. Jung, *Audio IC Op-Amp Applications*, 2nd ed. (Howard Sams, 1978), 132–136.

4. M. Sulzer, "A High Quality Power Supply Regulator for Operational Amplifier Preamplifiers," *TAA* (2/80), 8.

5. B. Duncan, "A State-of-the-Art Preamplifier: AMP-02, Part 2," *Hi-Fi News & Record Review* 34:11 (Nov. 1989), 45.

SOURCES

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82205 Gilching, Germany
49/8105/5291, FAX 49/8105/24605

Burr-Brown
6730 Tucson Blvd., Tucson, AZ 85734
(602) 746-1111, FAX (602) 889-1510

Jensen Transformers
7135 Hayvenhurst Ave., Van Nuys, CA 91406
(213) 876-0059, FAX (818) 763-4574

Linear Technology Corp.
1630 McCarthy Blvd., Milpitas, CA 93505
(800) 637-5545

Mouser Electronics
958 North Main St., Mansfield, TX 76063
800-992-9943

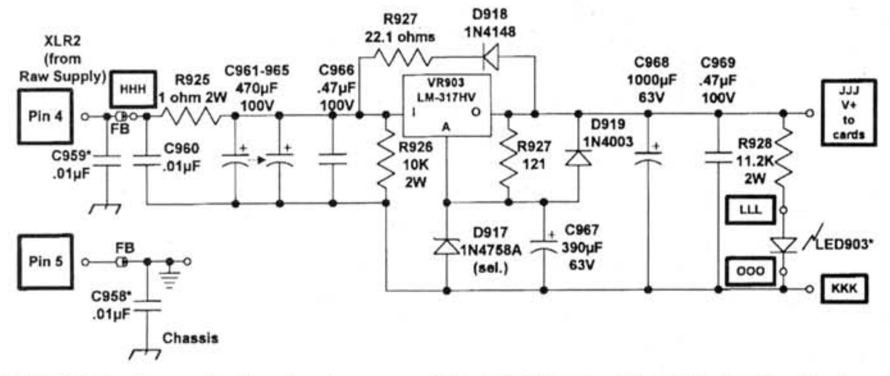

FIGURE 13b: Preregulator for phantom power. Select D917 to be within 0.5V of 56V, in-circuit.

excited by RFI from the environment, or by the sharp spikes normally produced when a rectifier diode turns on into a capacitative load.

To eliminate this problem, Duncan designed an antiresonant power supply.[5] By incorporating small resistors between the diode bridge and the main filter capacitors, decoupling the second filter stage with another small resistor, and using an array of smaller filter capacitors instead of a single large one, Duncan made an exceptionally quiet raw supply that is free from RFI breakthrough. The usual spiky ripple waveform is replaced by a near-sinusoid, without the high-frequency components that are apt to leak into audio stages.

In addition, all regulator stages are bypassed, not with the usual 1N4000-type diodes, but with small-signal diodes with much smaller stray capacitance.

My raw supply (*Fig. 12*) is similar to Duncan's, with voltage ratings scaled up to match the higher end-voltages desired. I've also incorporated LM-317HV/LM337HV preregulators to provide stable, low-impedance voltage to the on-card regulators. I prefer to have the voltage drop across the on-card regulators remain constant despite fluctuations in line voltage.

There are three preregulators: two large ones (*Fig. 13a*) to power the active devices, and a lower-current unit (*Fig. 13b*) to feed the 48V phantom power regulators. Note that the latter preregulator "floats" on a 56V zener diode. For this supply to function properly, the diode must be within 1% of 56V. Buy several and select the closest one, using a digital voltmeter. (The zeners are cheap.)

To avoid inducing hum in the input transformers, mount the raw supply in an external case with a shielded umbilicus connecting it to the main chassis. I use a 5-pin Cannon/XLR connector, and decouple the raw power inputs with ferrite beads and ceramic-disk capacitors. (The shield connection on the input socket is connected to the chassis with another ceramic disk.)

The main power transformer can be a toroid or a conventional transformer, but it must have a high current rating and correspondingly low

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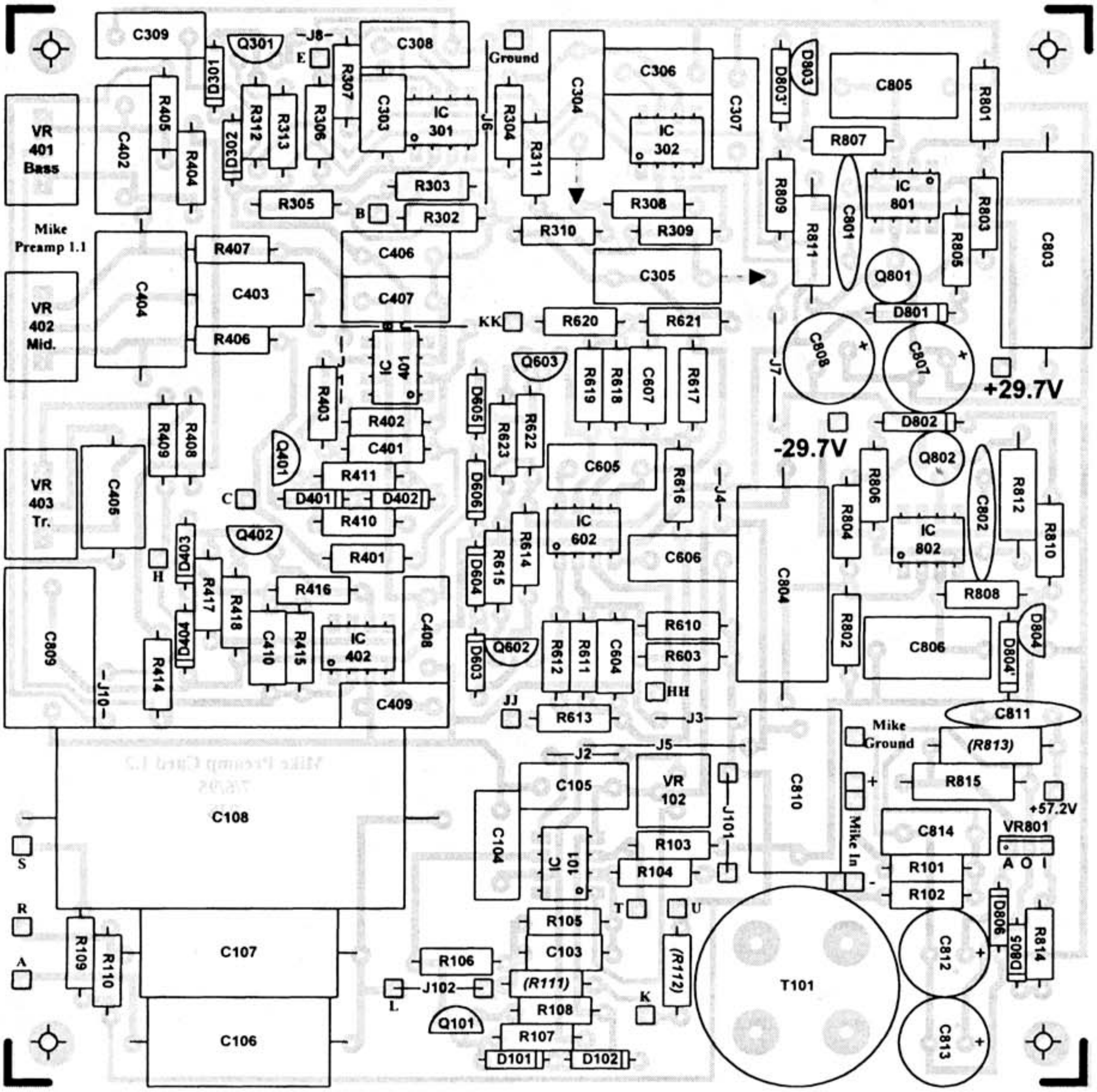

FIGURE 14: Mike preamp card: printed circuit board and parts layout. The transformer mounts in the four screw-holes on the board; two are for mounting screws, two are for wire leads. There are ten jumpers (in addition to J101 and J102, used to select "chimp-proof" operation) that are indigenous to the board, numbered J1-J10.

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series resistance. I suggest a 1A transformer for a single- or dual-mike preamp, and *at least* 2A for a 10-input board. Half an amp will suffice for the phantom-power supply.

A 60V center-tapped transformer (or two 30V windings wired in series) will theoretically produce 42.4V DC. Allowing 2.4V drop in the bridge rectifier and 1.25V drop across the series resistors in the raw supply, the preregulators are driven with 38.8V DC. Since the output of the preregulators is 29.7V DC, the typical drop across the preregulator is 9.1V DC.

If the line voltage runs 10% low, the voltage across the preregulator works out to 4.8V DC, quite adequate for the LM317/LM337 chips to regulate properly. However, if the line voltage runs 10% high, the voltage across the preregulator becomes 13.3V DC; in a multi-input mixer drawing 1A, this means the chip must dissipate 13.3W. I strongly recommend that the preregulators in such high-current applications be TO3-cased units, and that they be mounted in hefty heatsinks.

The line input includes a voltage-dependent resistor and ceramic-disk capacitor to protect against overvoltage and surges. I also recommend including ferrite beads on the AC lines. (It's amazing how much garbage actually rides the wires: look at them with an oscilloscope sometime!) If you include a commercial RFI filter at the input, be sure it's rated at more than 200W.

Component choices make a critical difference in this supply's performance and reliability. I've specified Panasonic's low-ESR capacitors, which should be available everywhere. Conventional capacitors are far less effective at filtering high-frequency noise. The bypass capacitors should be polypropylene or polyester; use the easily obtainable Panasonic caps or another low-loss capacitor. American builders might try the excellent polypropylene caps available from Mouser Electronics.

The resistors should be metal film or metal oxide (carbon film and carbon composition resistors can change value drastically over time.) Don't skimp on the power ratings. I've assumed worst-case operating conditions (over- and undervoltages from the power company, St. Louis summer temperatures) and then left considerable further margin. In my experience, this kind of overkill assures reliability. One warning: the 1Ω resistors will not withstand having the supply's output shorted, so it might be a good idea to carry half a dozen in your repair kit.

Let's Start Building

Figure 14 shows a circuit board layout for a single-channel mike preamp, using integrated circuits and parts placement. (Borbely can supply cards for constructing his circuits; you need the EB-489/104 card. If you choose to use the Jensen JE-990 discrete amplifier, you're on your own.) Full regulation is included on the card, and only the switches, pots, and jacks are mounted externally.

Figure 15 is a layout and placement guide for the preregulator circuit card. *Figure 16* is a layout and placement guide for the raw supply card.

Proper grounding is important for low noise and clean signal. I like the

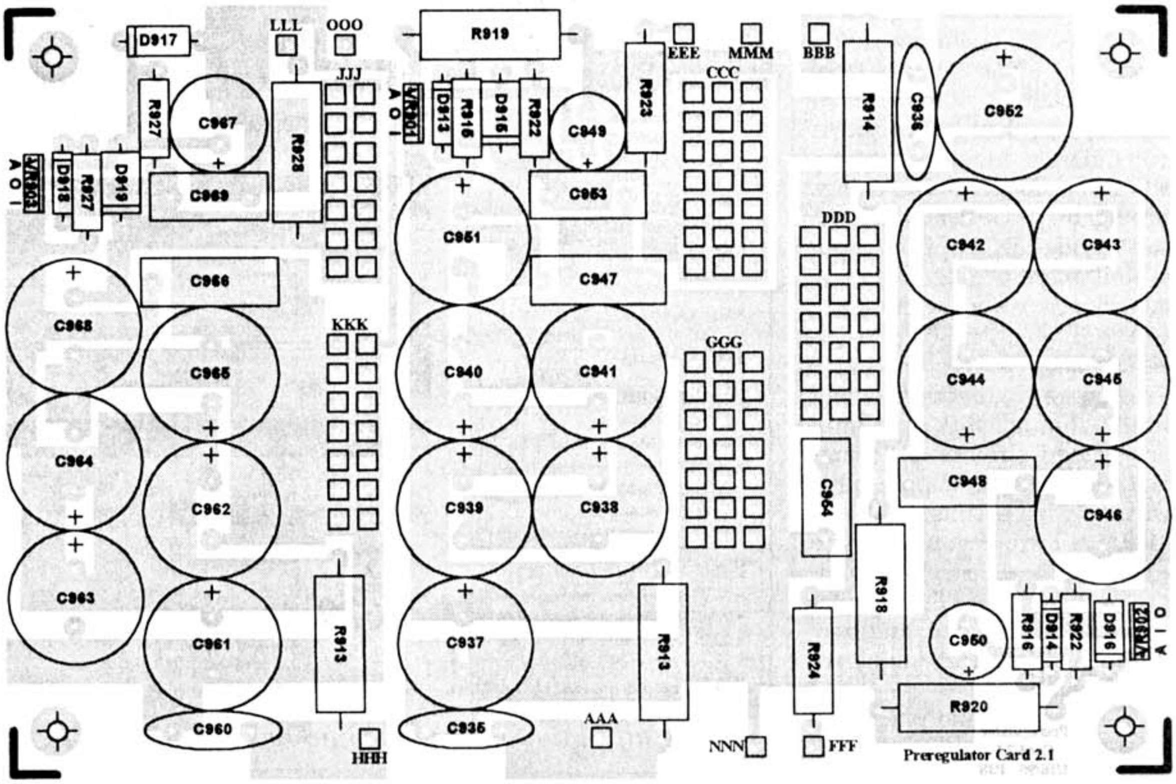

FIGURE 15: Preregulator card: printed circuit board and parts layout. This design can also be used for other applications: the plus and minus regulators for audio circuits, and the additional plus regulator (used here for phantom power) to feed digital circuits. Use heatsinks on the voltage regulators. In a large multi-input mixer (or other high-current application), use TO3-cased regulators for VR901 and VR902, and mount them next to the card on hefty heatsinks. Keep the leads short.

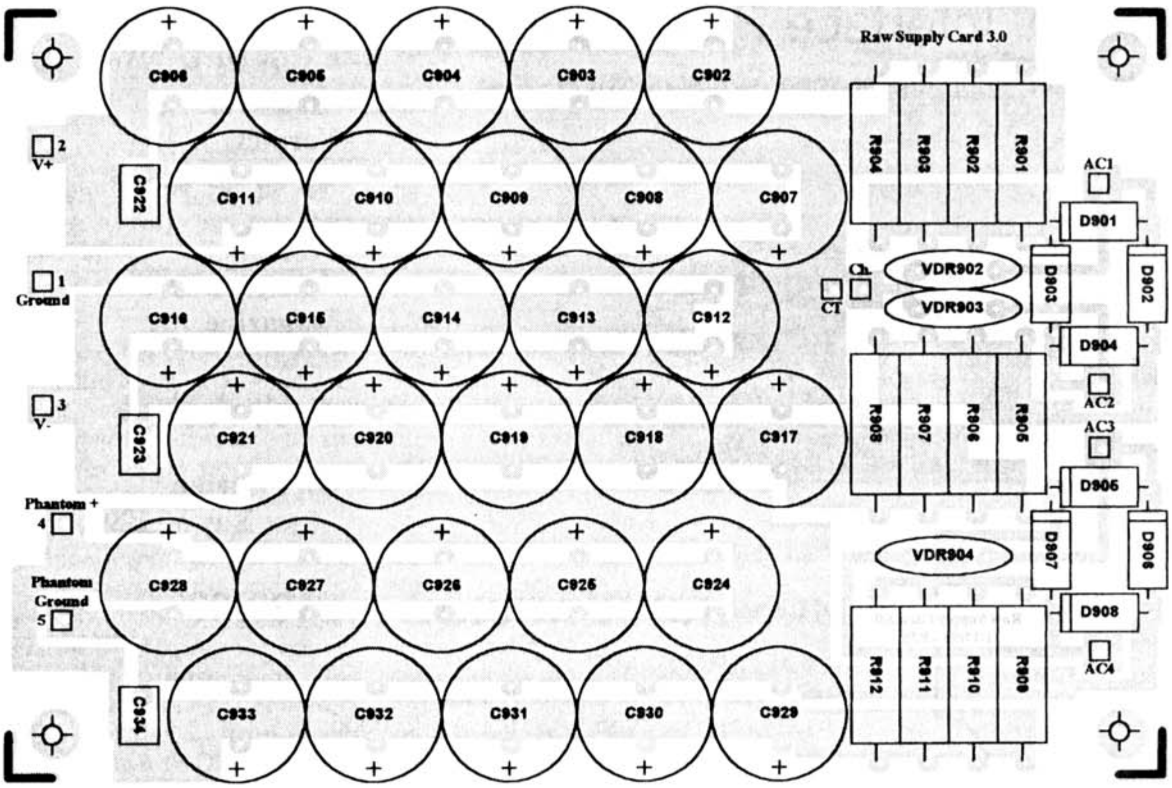

FIGURE 16. Raw supply card: printed circuit board and parts layout. Like the preregulator card, this can be used to power other projects. For high-current applications, mount D901-908 and R901-912 ¼″ above the surface of the board, to allow some air circulation.

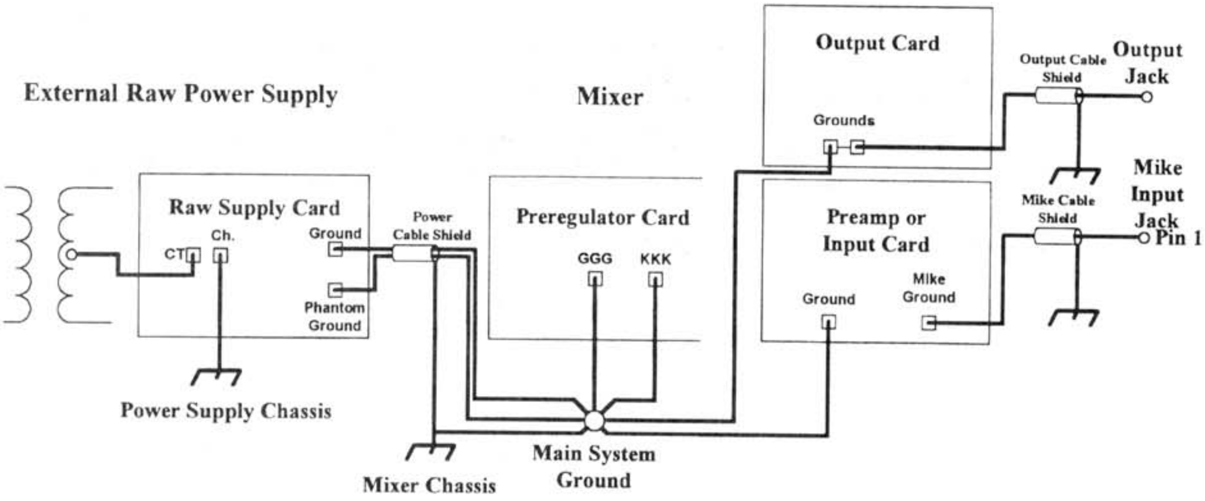

FIGURE 17: System grounding. The cable shield on the power umbilicus is connected to the XLR plug's shell at the mixer end only. Similarly, the shields on input and output cables are connected to chassis ground at one end only. RCA and ¼″ jacks should be isolated from the chassis with shoulder washers. The Main System Ground is a terminal strip, mounted close to the XLR connector where the power umbilicus plugs in.

grounding scheme in *Fig. 17,* which will work for multi-input mixers as well as for this stand-alone preamp. The modular design of this preamp allows you to build it in several ways. You might try a dual-channel mike preamp, a pair of single-channel preamps powered by a common raw supply, or two preamps with separate supplies. If you decide to build a preamp using the Borbely modules, but wish to include the option of using equalizers, you can use the board in *Fig. 14,* using the EQ stages, balanced output stage, rail voltage regulators, phantom power regulator, and the input transformer, but omitting the input and secondary gain stages. The Borbely modules can be powered by the Sulzer regulator, or by the modules' built-in IC regulators.

I'll conclude this series next issue with a closer look at multi-input mixers. ■

Part 3

MIXERS AND MIKE PREAMPS

By Paul Stamler

Multi-input mixers are more difficult to design than single-input mike preamps. The summing stages, which combine the various inputs into one or two outputs, present nasty traps, and the designer is constantly struggling to control noise levels. I find it useful to work backward, from the output stages to the input. Noise figures are easier to calculate when you know how much fixed noise is already in the system.

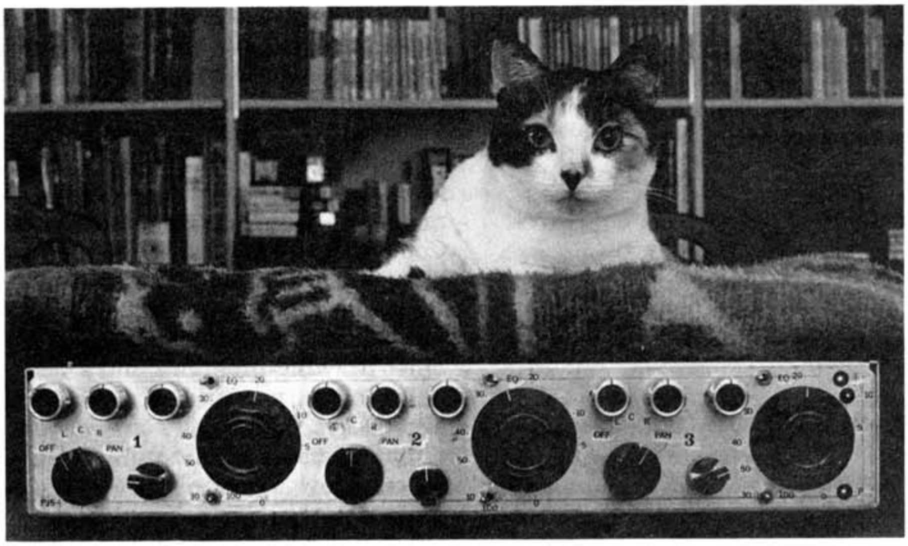

PHOTO 2: Prototype mixer, with three inputs and two outputs. This version doesn't have monitoring or echo buses. Note the different shapes for the EQ, selector, and pan knobs. In a dark place it helps to be able to sort out functions without relying on labels or color codes. (The cat's name is Sadie.)

Output Stages

Like the single-input preamp, each output stage of a multi-input mixer must be able to drive two outputs simultaneously: unbalanced (nominal −10dBu) and balanced (nominal +4dBu). It should also be able to drive a VU meter without loading the output or causing distortion. Unlike the single-input preamp, a multi-input mixer needs a master fader to control the overall program level.

In Part 1 I discussed how running the master fader at too low a level leads to severe overload in the summing amplifier. I've designed two solutions for this problem. The first is to make the gain of the output stages fairly high; the signal level at the master fader slider is only −20dBu for nominal output levels. Assuming the summing amplifier clips at +22dBu, we can lower the fader to −24dB while maintaining 17dB of headroom. Since the summing amp has 1dB more headroom than the tape, it will not limit the system's headroom.

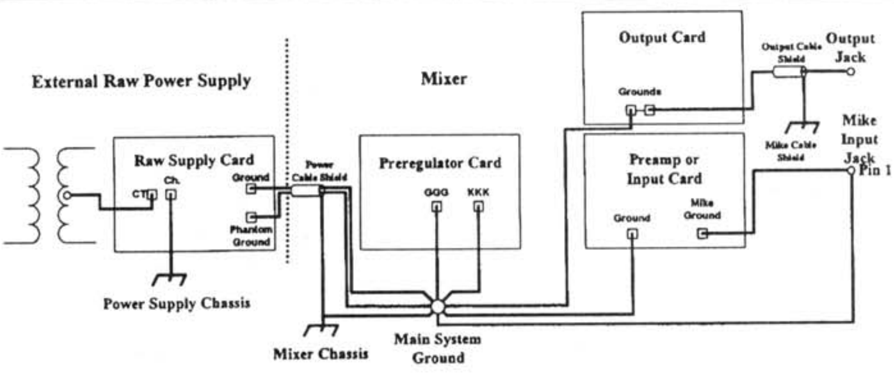

FIGURE 18 (revision of Fig. 17, Pt. 2): System grounding. The cable shield of the power umbilicus is connected to the XLR plug's shell at the mixer end only. Similarly, the shields on output cables are connected to chassis ground at one end only. If the cable connecting the mike input to the card uses three internal conductors plus a separate shield, connect the shield to chassis at the jack, leaving the other end unconnected. If it has only two conductors plus shield, connect as shown, using the shield for ground. RCA and ¼″ jacks should be isolated from the chassis with shoulder washers. The "Main System Ground" is a terminal strip, mounted close to the XLR connector where the power umbilicus plugs in.

The master fader normally sits at −6, so there is room for 18dB of fudging before trouble sets in. This ought to be enough, but since I tend to be a "suspenders-and-belt" designer, I've added one refinement in the stereo master channels. I seldom use the stereo masters for a fade-to-black, preferring instead to add my fades when editing, and thus I find it unnecessary to pull down the master faders to infinite attenuation.

The stereo masters ride on a pair of resistors, nominally 1.37k, which provide a maximum attenuation of 23.9dB when the fader is closed. (Pots have fairly wide tolerances, so you should select these resistors individually. Either choose one that is as close as possible to $0.0685 \times R_M$—where R_M is the measured value of the master fader—or one that gives equal attenuation in both channels with the fader set for 6dB.)

If you need complete attenuation in the stereo outputs, you may eliminate these resistors. In that case I'd recommend painting a red line at the −24dB point as a warning not to pull the master down any farther.

The output stages (*Figs. 19a* and

TABLE 4

FIXED-NOISE SOURCES OUTPUT REFERRED (μV)

	DAT	OPA-604	NE-5534A	LT-1028A	BORBELY
Recorder	24.5				
Output Amp		6.65	5.11		
Summing Amp			14.6	11.5	17.2
EQ Amps (10)		8.78			
Total fixed-noise sources (including DAT and EQ amps):					
LT-1028A summing amp, OPA-604 output amp			28.8μV		
Borbely summing amp, OPA-604 output amp			31.9μV		
5534a summing amp, 5534a output amp			30.3μV		

19b) are fairly straightforward. While the balanced circuit is not the most sophisticated, it continues to produce clean signal from its noninverting pin, even when the inverting pin is shorted to ground.

Table 4 lists fixed-noise sources, as their contributions appear at the −10dBu output. Even the noisiest amplifier—the OPA-604—produces only 6.65μV of noise compared to the DAT recorder's 24.5μV. Remembering that dissimilar noise voltages add by the root-sum-square method, the total noise from the DAT and the OPA-604 will be 25.4μV, a degradation in the noise figure of only 0.31dB. Clearly, the choice of amplifier makes little difference here.

For reasons of space, I suggest using integrated circuits, although with some determination you could probably fit in a Jensen JE-990 discrete circuit. The OPA-604 is the obvious first choice. The negligible input current inherent in FET input devices allows the coupling capacitors to be replaced with jumpers. The NE-5534 will also do the job, but you should definitely use the coupling capacitor at the input to prevent the fader from becoming noisy. Unless you're certain all the recorders you'll be using will have coupling caps, I'd recommend the output caps as well. This applies to the JE-990 and to any other bipolar-input amplifier, whether integrated or discrete. The Borbely will work without coupling caps, but it doesn't really like low load impedances.

Load Factors

What about loading? On the −10dBu output, I've arbitrarily assumed that the worst possible external load is 10k. Add a 20k trimpot for the meter buffer, plus 6.32k for the feedback and input resistors, and the total load is about 3.24k. Allowing our usual 16dB of headroom over −10dBu, the maximum peak voltage is 2.19V. This draws 0.68mA, which biasing the amplifier's outputs to 2mA of standing current amply provides.

The balanced outputs are trickier. The maximum output from each side of the amplifier is 5.49V. If the mixer is operated into a standard 600Ω studio line, the load is 300Ω per side, and the maximum current draw is 18.3mA. The OPA-604 and the NE-5534 chips can emit this much current, but not in Class A.

A typical studio recorder has a balanced input impedance of 5k, or 2.5k per side. Adding the output stage's feedback and input resistors makes the total load 1.97k. Under maximum output conditions, this draws 2.79mA of current. A 3.0mA bias on the chips' output stage should work fine.

The meter buffer's trimpot adjusts the nominal level to −16dBu, and the amplifier boosts it to +4dBu. The (nominally) 3.65k resistor provides the necessary damping for the meter's

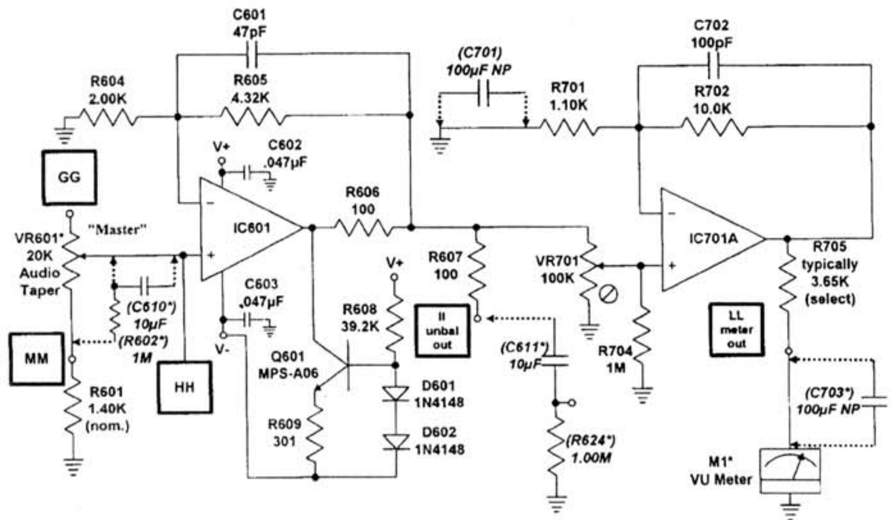

FIGURE 19a: Unbalanced output stage and meter buffer for multi-input mixer. If you are using NE-5534 amplifiers for A601–701, include C701 and replace J601 and J604 with C610/R602 and C611/R624. Trim VR701 for a 0VU reading when terminal II is delivering −10dBu (0.245V) on a 400Hz tone.

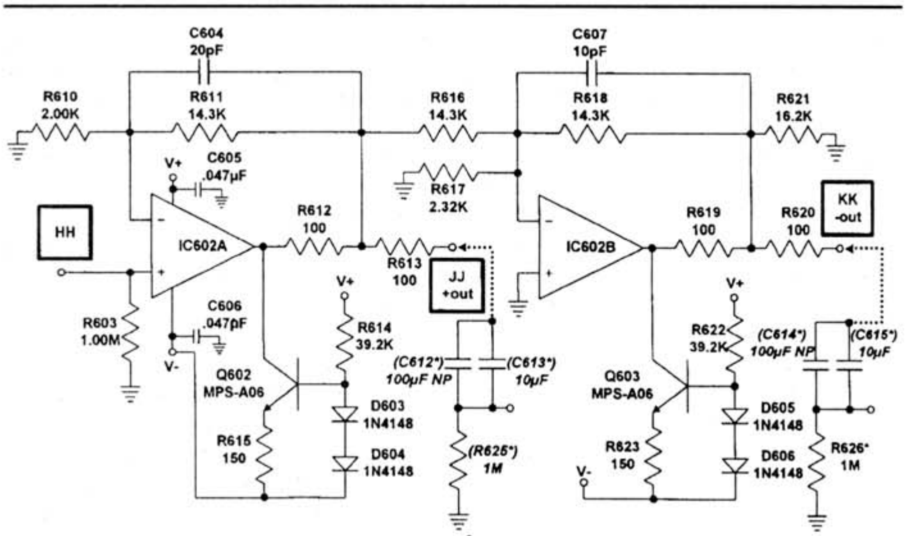

FIGURE 19b: Balanced output stage for multi-input mixer. If a 5534 amplifier is used for A602–603, J602 and J603 should be replaced by C612–613 and C614–615. These should be parallel combinations of 500μF NP electrolytics and 10μF polyester or (preferably) polypropylene.

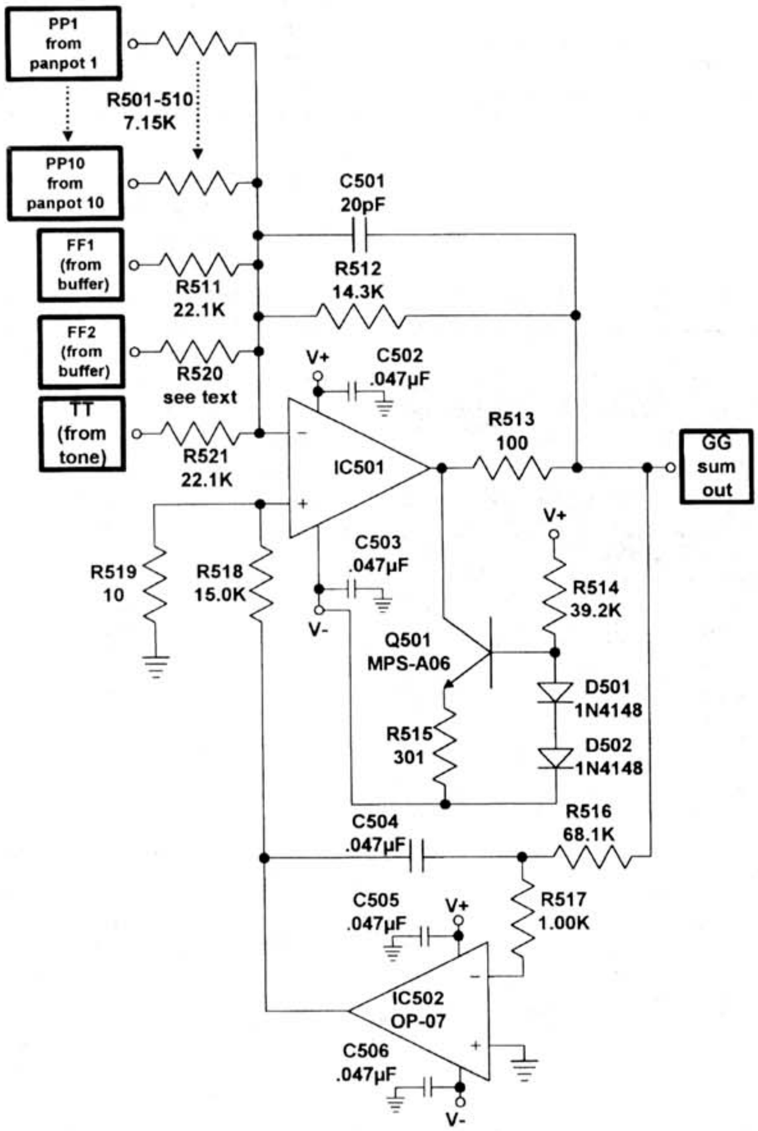

FIGURE 20: Summing amplifier stage. The component values shown are for the left and right summing amps. See *Table 5* for component values for the mono (house) monitor and echo summing amps.

movement. If you're using an OPA-604 or other FET input amplifier, you may eliminate C612, C614, R603, and R624–626; C610–611, C613, and C615 may be replaced with jumpers.

The master fader should be a high-quality, conductive-plastic dual pot, audio taper, without detents (available from Old Colony Sound Lab, Sonic Frontiers, Welborne Labs, and Triode Labs). You need only one, and it's worth getting a good one. If you must have a straight-line fader, Penny & Giles is the industry standard. They also make superb, expensive rotary pots.

Summing Stage

The stereo summing stage (*Fig. 20*) presents a difficult design challenge. Its noise contributions are threefold: the amplifier's voltage noise, its current noise, and the Johnson noise of the summing resistors. The internal sources are critical. The summing amplifier's noise gain is not the same as its signal gain. You find the former by calculating $R_F/R_{IP} + 1$, where R_F is the feedback resistance and R_{IP} is the parallel combination of the summing resistors. The internally generated noise is amplified by the loop gain, not the (much lower) signal gain.

Johnson noise can also be significant. Normally, the noise performance of paralleled resistors is that of a single device. In a summing amplifier, however, the resistors should be viewed as discrete noise sources whose contributions add in the usual root-sum-square manner.

Table 4 shows the noise contribution of the summing stage with three amplifiers. While all three are very good, the LT-1028A is clearly the quietest. Even the worst—the Borbely—is only 0.89dB noisier than the LT-1028A when the DAT recorder, output amp, and EQ amps are factored in. (The total fixed-noise sources, in various combinations, are also listed.)

I'd recommend the LT-1028A unreservedly, except for one problem: its bipolar inputs typically produce 25nA of bias current. The summing amp's inputs are direct-coupled to the panpots, and may become noisy over time.

Engineering is the art of compromise, and, while no perfect solution to the bias current problem exists, several alternatives are possible. One is to accept a slightly higher noise level and use a Borbely amplifier for the summer. Aside from the noise, the main disadvantages are bulk and cost. Another option is to couple the panpot to the summing resistors with large (about 22µF) polypropylene capacitors. The disadvantages are again bulk and cost. Finally, given that the LT-1028A's input current is small for a bipolar IC, and that it will be distributed between several panpots, you could simply leave it directly coupled and accept the eventuality of replacing pots. That's what I've done.

If the LT-1028A isn't available, the NE-5534a will do a fair job. Its typical input current is an order of magnitude higher than the LT-1028A's, and I'd suggest placing coupling capacitors between the panpots and the summing resistors.

Loading is fairly light, even under extreme conditions. If the master fader is lowered to –23.7dB, the summing amp's maximum output is 11.9V. The load on the summer is about 8.4k. This translates to a maximum current of 1.4mA, which a bias of 2.0mA will handle nicely.

Buffer amplifiers enable several mixers to be cascaded or line-level signals patched in. These are summed with echo-return signals, and fed to the stereo summing amps via R511.

The summing/output card can also be used for the mono (house) summing circuit, and for summing the monitor and echo-send signals. In such cases the resistor values change (*Table 5*). (Extensive parts lists, offering alternative parts values, are available by sending a 9″×12″ self-addressed envelope, with $1 for postage and handling, to *Audio Amateur*,

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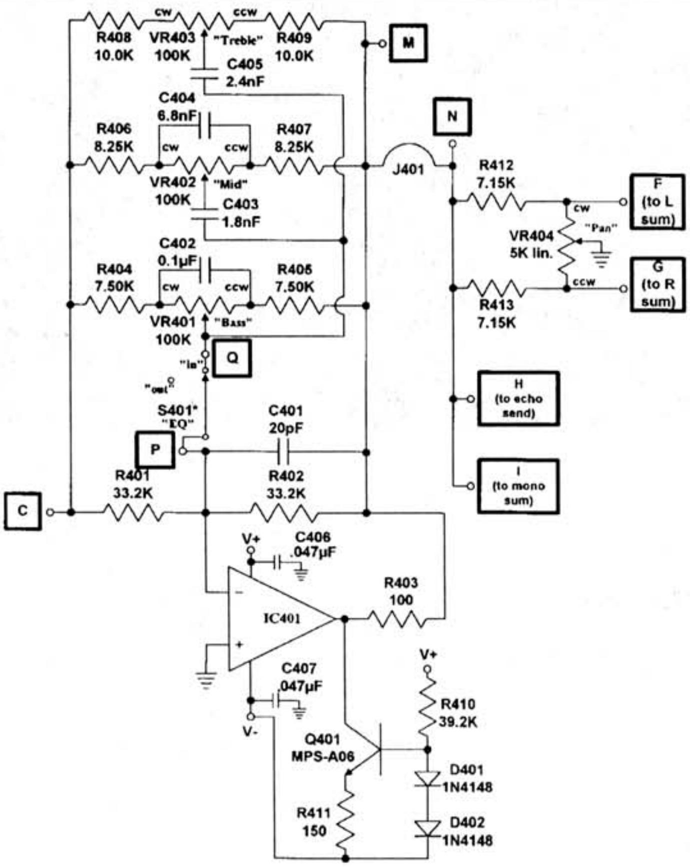

FIGURE 21a: EQ stage and panpots for multi-input mixer. If the variable-input-gain option is chosen, delete J401 and connect terminal M to terminal E on the secondary gain stage.

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Editorial Dept., PO Box 576, Peterborough, NH 03458.)

EQ Stage

For simplicity, I chose to use a single-amplifier EQ stage for the multi-input mixer modules, rather than the double-amplifier stage of the single-mike preamps. In a mixer with ten or so inputs, fewer components mean greater reliability and better sound. I decided to leave the EQ stage in-circuit and switch the functions as needed. While this design may not be minimalist (an extra stage is always in the chain), it ensures that the channel always passes signal, even if the switch breaks. The EQ stage and the summing stage both invert polarity. The net effect is noninverting for signals originating at the mixer's input.

The circuit (*Fig. 21a*) is a straightforward equalizer similar to those described by Walter Jung.[1] It incorporates shelving boost and cut at 200Hz and 2kHz, plus a peaking-and-dipping function at 1.8kHz, a frequency I've found useful in backing off the "forward" sound of certain condenser microphones. This mid-range frequency can easily be made switch-selectable, although I use the equalizers so infrequently that I haven't bothered.

I see few alternatives to the OPA-604 op amp in this circuit. It is direct-coupled to the EQ controls, so the amplifier should have FET inputs. And it must tolerate high supply voltages because the mixer runs on ±21.5V rails. The only other candidates I know about are certain versions of the CA-3140 chip (still a remarkably good-sounding amplifier if properly used). The Borbely circuit is a good choice in a studio mixer, but it's probably too bulky for a portable unit.

The noise contribution of the equalizer stage is negligible. With an OPA-604, it adds only 0.42dB to a ten-input mixer's fixed noise. Again, the amplifier is fairly lightly loaded. Under the worst possible circumstances (i.e., all three EQ controls set to full cut) the total load is 1.19k. Since the nominal voltage at the EQ amp's output is −10dBu, the maximum voltage with 16dB of headroom is 2.19V. This draws 1.84mA, which a 2mA bias will easily cover.

The panpots should be high-quality, linear-taper types, preferably conductive plastic. Bourns pots have an excellent reputation. A lower-cost alternative is offered by Mouser Electronics.

If there's room in the mixer, you might add a pair of ganged switches. I discovered myself panning hard left, center, and right a good deal, so I rigged a double-pole, five-position switch to select the resistors which correspond to the panpot in these three positions (*Fig. 21b*). (Other settings are for continuous panning or muting.) I did this because a good switch is both more reliable and better sounding than even a good pot; it also saves wear and tear on the panpot when I'm not using it, which is most of the time.

Secondary Gain Stage

Designing this stage was something of a nightmare. Because it is always in-circuit, its noise is perpetual, fairly substantial due to significant gain (26dB), and multiplied as a result of multiple channels.

Table 6 is a list of variable noise sources, measured at the mixer's output, for various channel-fader settings. The columns on the left show the noise produced by the input stage. The center columns list the noise produced by the secondary gain stage. The worst noise occurs with the fader at the −6dB position, where it presents its maximum source impedance to the amplifier (2.5k or one-fourth of the fader's value). This maximizes both

TABLE 5

SUMMING/OUTPUT CARD PARTS VALUES

PART	LEFT/RIGHT	MONO (HOUSE)	MONITOR/ECHO
C501	20pF	15pF	15pF
R501–510	7.15k	10.0k	10.0k
R511	22.1k	14.0k	10.0k
R512	14.3k	6.32k	6.32k
R520	Not used	14.0k	49.9k

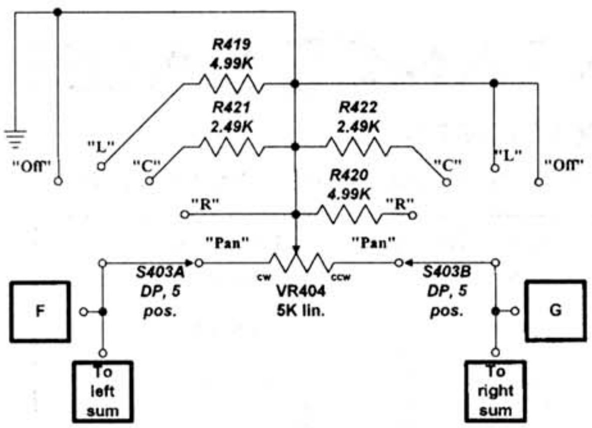

FIGURE 21b: Alternative hookup for panpots. All parts, including VR404, are mounted off-board.

the fader's Johnson noise and the amplifier's current noise.

There is a considerable difference between the LT-1028A chip and the NE-5534a in this application (the OPA-604 and Borbely amps are too noisy to be used here). Because the LT-1028A has much lower voltage noise, it offers better noise performance at most fader settings. The figures are roughly comparable only at –6dB—and there, input noise predominates. The columns on the right in *Table 6*, which describe the composite performances of an input stage and a secondary gain stage, bear this out.

I tried two different amplifier permutations: my preferred combination of an OPA-604 on the input and an LT-1028A in the secondary gain stage, and the alternative of 5534As in both positions (suggested if the preferred parts are unavailable or unaffordable). With the pot wide open, the performances are roughly comparable; however, the 604/1028A combination is the hands-down winner at lower settings, and especially with the fader turned off. (The last column is discussed in the sidebar).

The design of the secondary gain stage is essentially the same as for the single-channel mike preamp, with a bit heavier loading. In the worst-case scenario (i.e., all EQ pots set to maximum boost—pretty unlikely) the total load is 718Ω, for a maximum output current at 2.19V of 3.05mA. With only two controls turned up, the maximum output current is 2.83mA, and I am comfortable setting the bias on the secondary gain stage to 3mA.

One oddity is worth noting. With classic star grounding, all ground connections are returned to a single central point, usually the main power supply ground. In this stage, however, connecting the channel fader's ground terminal to the central system ground can generate hum. I've had better results connecting the fader ground to the grounded end of R301 and R302.

While we're in that neighborhood, two parallel coupling capacitors (C301 and C302) are mounted off board. For economy's sake, these caps could be a 10μF polyester and a 1μF polypropylene, but I've found that the difference between this setup and an all-polypropylene combination is not subtle. It is worth the expense for the tighter, cleaner bass, cleaner transients, and reduced hash in the highs. The best—and most expensive—capacitor in this position is probably the MIT Multicap. It contains its own bypass capacitors, making C302 unnecessary.

Input Stage

This is the same design I used for the single-channel mike preamp, using ICs. The coupling capacitors are arranged so that a 6dB/octave bass rolloff can be added at 100Hz or 30Hz, or it can be switched to 1.6Hz (effectively flat). If you're using OPA-604 chips, you may be able to trim the offset voltage low enough to allow replacing C108 with a jumper, for truly flat response. Just keep an eye on that offset voltage: it doesn't take much DC to produce a noisy pot or thumps when you fade up and down.

What should you use for a channel fader? Good straight-line audio-taper pots are notoriously hard to find and very expensive; cheap ones are infuriatingly unreliable. Unless you can afford high-grade devices (such as Penny & Giles or Duncan), and can do the machine work necessary to create slots for them in your front panel, I suggest using rotary faders. Quality single-channel rotary pots are available from Sonic Frontiers, Welborne Labs, Penny & Giles, and—astonishingly—Radio Shack. Their Alps 10k audio-taper pot is a decent low-budget choice. Rotary pots are harder to read at a glance than most linear faders, but good ones can be adjusted more accurately after some practice.

Whichever type of fader you choose, I strongly suggest custom-labeling the mixer's faceplate. A standard practice is to label the fader's attenuation so that wide open is 0dB and subsequent settings are –10dB, –20dB, and so on. Another option is to label the gain of the microphone preamps, from the input jack to the secondary gain stage output, with wide open being 50dB. Either method will work, but I do suggest individually calibrating each fader, using a sine-wave generator and an AC voltmeter. Removable labels allow you to recalibrate when you replace a fader. I use rub-on letters, which I then lacquer; they are quite durable, but can be removed with thinner and a Q-Tip.

There usually isn't much call for line-level inputs. Instruments that produce line-level signals (i.e., synthesizers) are normally connected through a "direct box," which buffers the signal and converts it to mike-level, balanced. You might install a switch just before the channel fader (in place of J102) to allow connection of an unbalanced line-level signal. (An 80.6k resistor in series with the switch

TABLE 6

VARIABLE NOISE SOURCES OUTPUT REFERRED (μV)

FADER POSITION	INPUT NOISE, 1 CHANNEL		STAGE 2 NOISE, 1 CHANNEL		INPUT + STAGE 2 NOISE, 1 CHANNEL		
	604	5534A	1028A	5534A	604/ 1028A	5534A/ 5534A	CONV. 604/1028A
0	86.2	79.1	4.37	10.5	86.3	79.8	83.0
–6	43.1	39.6	19.8	21.0	47.4	44.8	41.7
–12	21.5	19.8	17.1	19.0	27.5	27.4	21.7
–18	10.8	9.89	13.2	16.0	17.0	18.8	12.1
–24	5.39	4.95	9.99	13.7	11.4	14.6	7.94
–30	2.69	2.47	7.76	12.3	8.21	12.5	6.37
–36	1.34	1.24	6.31	11.5	6.45	11.5	4.72
Closed	0.00	0.00	4.37	10.5	4.37	10.5	2.18

lowers the signal 14dB.) On the other hand, introducing unnecessary switch contacts into the signal path can degrade both sound quality and reliability. A good compromise is to include the "Mike/Line" switches on only a few channels—perhaps the last few—of a multi-input mixer.

Monitor/Echo-Send Circuits

The monitor circuits (*Fig. 22*) are basically duplicates of the secondary gain stage, slightly simplified and isolated with a buffer to avoid loading the input amplifier. Since this circuit is usually heard only on a noisy stage, I've relaxed my performance criteria a bit. The gain stages after the pots are specified as halves of an NE-5532a dual IC; the buffer amp can be half of either an OPA-2604 (the dual version of the OPA-604) or an NE-5532. With the latter, the buffer should include coupling capacitor C216.

To create a monitor mix that is close to the house mix, a good starting point, set the monitor mix pots to the same gain level as the fader for their channel. It helps to calibrate the pots in decibels, using the same scale as the channel fader.

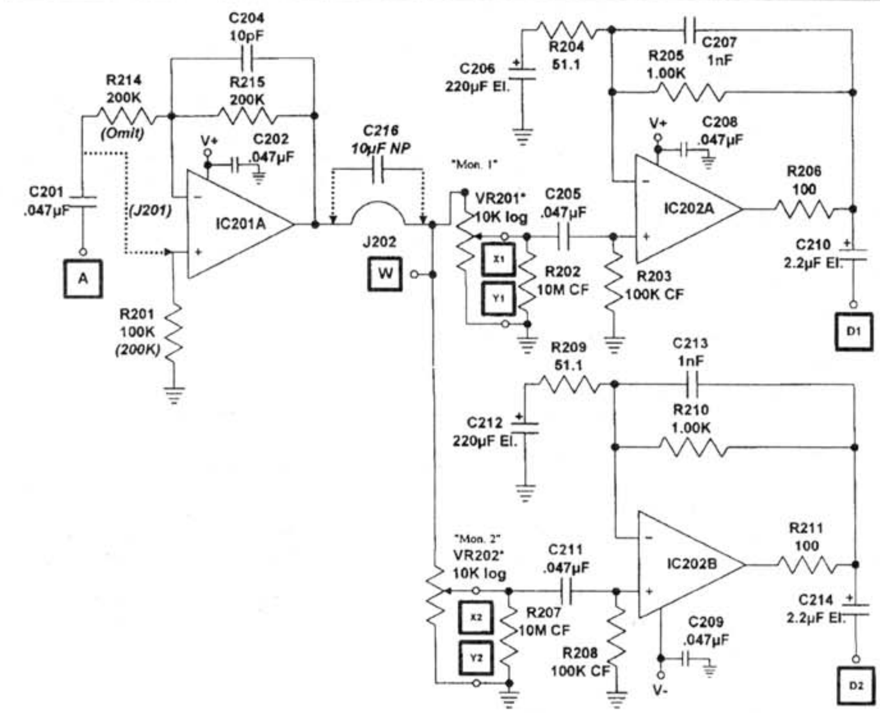

FIGURE 22: Monitor circuits. If A201a is half of an NE-5532, use C201, C204, and R201.

The echo-send pot (*Fig. 23*) is normally connected to the EQ amp's output, so that it follows the gain setting and equalization of the stereo and mono outputs. Again, the output is buffered, the buffer amp being the other half of the dual IC used with the monitor circuits. If this is an NE-5532, include C215 to isolate the pot from the buffer's input bias current.

The monitor and echo-send summing/output boards are identical to the left and right boards. The resistor values are different, however, as shown in *Table 5* (and in the parts lists available from *Audio Amateur*).

Buffer Amplifiers

You may need to cascade several mixers together for concerts with multiple microphones. To make this easier I've designed auxiliary inputs to the summing amplifiers. Since these amps invert polarity, inverting buffer amps are also included (*Fig. 24*).

These can simultaneously accept +4dBu balanced and −10dBu and +4dBu unbalanced stereo auxiliary mixer signals, as well as +4dBu and −10dBu unbalanced echo-return signals. The right- and left-channel buffers also feed their output signals into the mono (house) summing amp.

Once again, I recommend OPA-2604 amplifiers for the buffers, as their noise contribution is negligible. Loading is fairly light, and the maximum current is 0.36mA.

System Integration

How do these sections perform in a real mixer? In particular, how much noise is added to the signal by various active devices? I set up a hypothetical ten-channel mixer, including fixed-noise sources, with the highly artificial constraints that all channel faders are set to the same level, and all panpots have their left channels at maximum. I then calculated the noise level produced at the left-channel outputs for various fader settings, and summarized the results in *Table 7*. Several permutations of active devices are represented, with the noise output given in microvolts and the signal-to-noise ratio in decibels, relative to nominal operating level.

With all faders wide open, the performance is uniformly terrible, although this is not a failing of the mixer. The inherent Johnson noise of the microphones' own 150Ω impedances accounts for most of the noise. It would be present even in a perfect, noiseless mixer (if anyone has ever built one).

Differences become apparent at lower fader settings. The preferred configuration—OPA-604/LT-1028A on the inputs, LT-1028A/OPA-604 on the summer and outputs—and its variant (Borbely amp for the summer) provide excellent performance. The all-5534a version is significantly poorer but still more or less acceptable, and the "conventional" design (discussed in the sidebar) is the quietest. For comparison, I've plotted the data in *Fig. 25*, along with the output of a hypothetical noiseless mixer.

The data helped point me in the right direction, but I was curious to see how various configurations would perform in actual recording environments. Accordingly, I set up four "benchmark" situations that simulated real working conditions for a multi-input mixer:

1. two faders set for 44dB of channel gain, two set for 32dB, six closed (representing two performers, each with a dynamic vocal mike and a condenser instrument mike);

2. one fader set for 44dB, one set for 32dB, eight closed (single performer, same setup);

3. four faders set for 32dB, six closed (this could be two performers with condenser vocal and instrument

TABLE 7

TOTAL NOISE TEN-INPUT MIXER

FADER POSITION	604/1028A 1028A/604 NOISE (μV)	dB REF. −10dBu	604/1028A BORBELY/604 NOISE (μV)	dB REF. −10dBu	ALL-5534a NOISE (μV)	dB REF. −10dBu	CONVENTIONAL 604/1028A NOISE (μV)	dB REF. −10dBu
0	274.4	−59.0	274.7	−59.0	254.3	−59.7	263.9	−59.4
−6	152.7	−64.1	153.3	−64.1	144.9	−64.6	135.1	−65.2
−12	91.6	−68.5	92.7	−68.4	91.8	−68.5	74.4	−70.4
−18	61.0	−72.1	62.5	−71.9	66.8	−71.3	47.9	−74.2
−24	46.0	−74.5	48.0	−74.2	55.2	−72.9	38.2	−76.1
−30	38.8	−76.0	41.1	−75.5	49.9	−73.8	35.2	−76.9
−36	35.3	−76.8	37.9	−76.2	47.4	−74.3	32.4	−77.6
Closed	32.0	−77.7	34.8	−77.0	45.1	−74.7	29.6	−78.4

TABLE 8

BENCHMARK TESTS

TEST	604/1028A 1028A/604 dB REF. −10dBu	604/1028A BORBELY/604 dB REF. −10dBu	ALL 5534a dB REF. −10dBu	CONVENTIONAL 604/1028A 1028A/604 dB REF. −10dBu
1	−70.0	−69.9	−69.8	−71.1
2	−72.3	−72.1	−71.6	−73.4
3	−74.6	−74.2	−73.0	−76.2
4	−76.7	−76.1	−74.4	−77.7

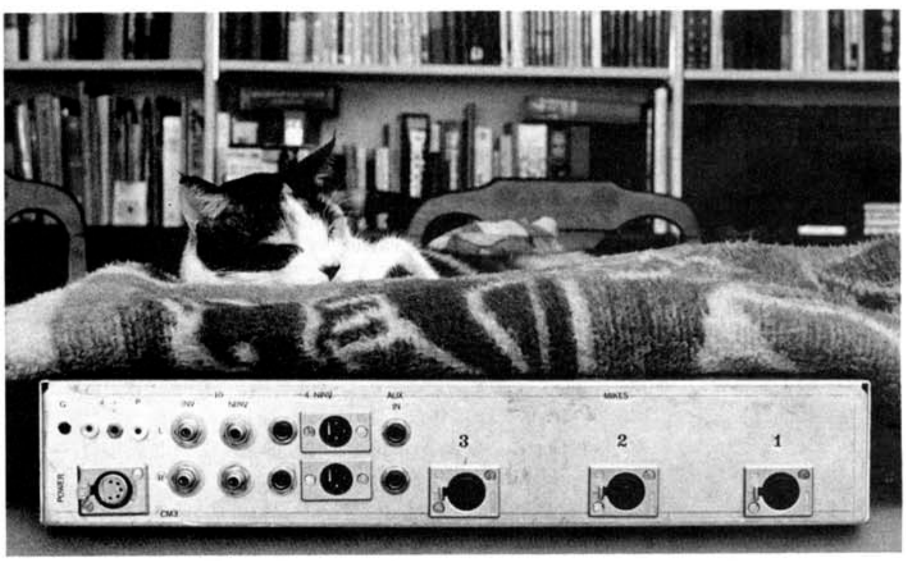

PHOTO 3: Back view of the prototype mixer. I later added two output jacks from the first and third input preamp channels, so I could use the unit as a straight stereo mike preamp without summing, output, or buffer stages. The stereo image was sharper, clearer, and better focused.

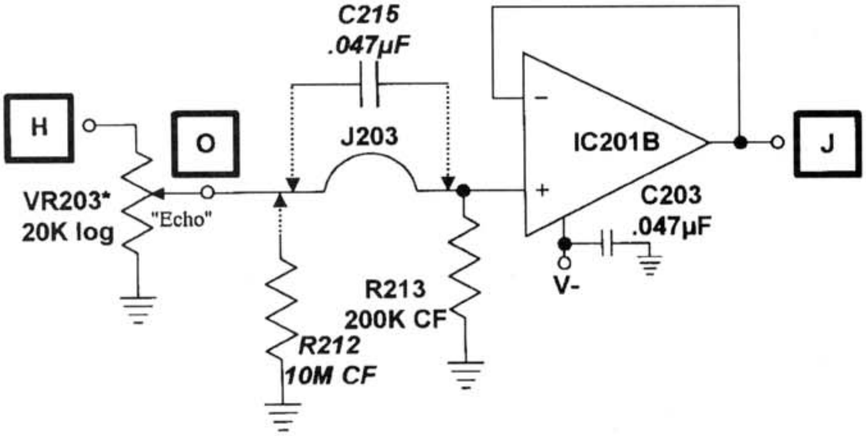

FIGURE 23: Echo-send circuits. If A201b is half of an NE-5532, use C215.

mikes, or perhaps a multi-miked classical setup);

4. one fader set for 32dB, eight closed (a crossed-pair stereo recording, probably classical, with a single condenser mike per channel).

The results (*Table 8*) reveal little variation in the first test, where the Johnson noise of the dynamic microphones in the 44dB channels is prevalent. The second test is similar, although the differences are more apparent; the third and fourth tests are fairly clear. All the results are acceptable, but the all-5534a configuration is marginal. Given its poor sonic reputation, and the need for coupling capacitors to isolate its high input currents, I recommend the 5534a as a fallback: use it only if you can't obtain the preferred components.

Input Cards

Figure 26 shows the circuit board design for the input-channel cards, incorporating the input, secondary gain, and EQ stages, the monitor buffers and amplifiers, the echo-send buffer, and on-card regulators for the amplifiers' rail voltages and the phantom power voltage. In the design of the printed circuit board and parts placement I tried to retain as many options as possible. The layouts accommodate different amplifiers, with or without coupling capacitors.

Jumper wires connect the EQ stage to the secondary gain stage and the panpot. This allows you to place the EQ stage elsewhere in the chain, as discussed in the sidebar, "Losing My Nerve."

Component choice is critical for a good-sounding mixer. As before, I recommend high-quality, metal-film resistors (Holcos or Roedersteins). Their sound is clearer, excess noise lower, and tolerances tighter.

A mixer built with the preferred active components will have only two coupling capacitors in the main circuits. I like audio-grade polypropylenes (Rel-Caps and MIT Multicaps) in the critical locations. Low-inductance polypropylenes are good for supply-rail bypassing, but if you're strapped for funds, polyesters will also fit the board layout. In either case, the Panasonic caps sold by Digi-Key work well. You can get by with polyesters and electrolytics in the monitor amplifiers.

I have mounted C&K switches (from Digi-Key) with gold contacts off the board. They should be connected with short lengths of high-quality hookup wire. The EQ pots and pan-pots can also be mounted off-board, or on-board in the case of Bourns or Mouser EQ pots. This allows you to use the pot shafts to mount the board to the chassis.

Nonconformist Connections

In a decision at odds with standard pro-audio practice, all connections are soldered to the input cards. My rationale is that even good edge connectors become noisy or intermittent, suffer metal fatigue and break, or oxidize and dirty up the sound. (Incidentally, you should see some of the crud I've removed from gold-plated edge connectors.) Of course, my solution makes it harder to reach the card should it need fixing, which is why I prefer not to mount pots and switches on the boards. As mechanical parts, they are the most likely to go bad, and are easier to replace if they aren't soldered onto a circuit board.

The transformer mounts in two holes on the input card. Connect it to the input XLR connector with a balanced, shielded cable, attaching one end of the shield to the "chassis" terminal and leaving the other end unconnected. Both Neumann and Canare microphone cable sound good and are sufficiently flexible to enable moving boards without damage. For hookup wire, I like the solid-core, Teflon®-insulated, linear-crystal wire made by Hitachi, or the (expensive) silver-plated wire that can be dissected from Apature BL-2 speaker cable. The inner conductors from Belden 8450 cable are also good, but the polypropylene insulation melts easily.

Ferrite beads on all power-supply wires protect against RFI. I also thread them on the signal conductors of the shielded cable that runs from the XLR connector to the input card, about 1cm from the connection. For maximum protection, and to avoid crosstalk between input channels, use shielded cable between the input card and the various summing cards. Again, be sure the shield is connected to ground only at one end. Cable

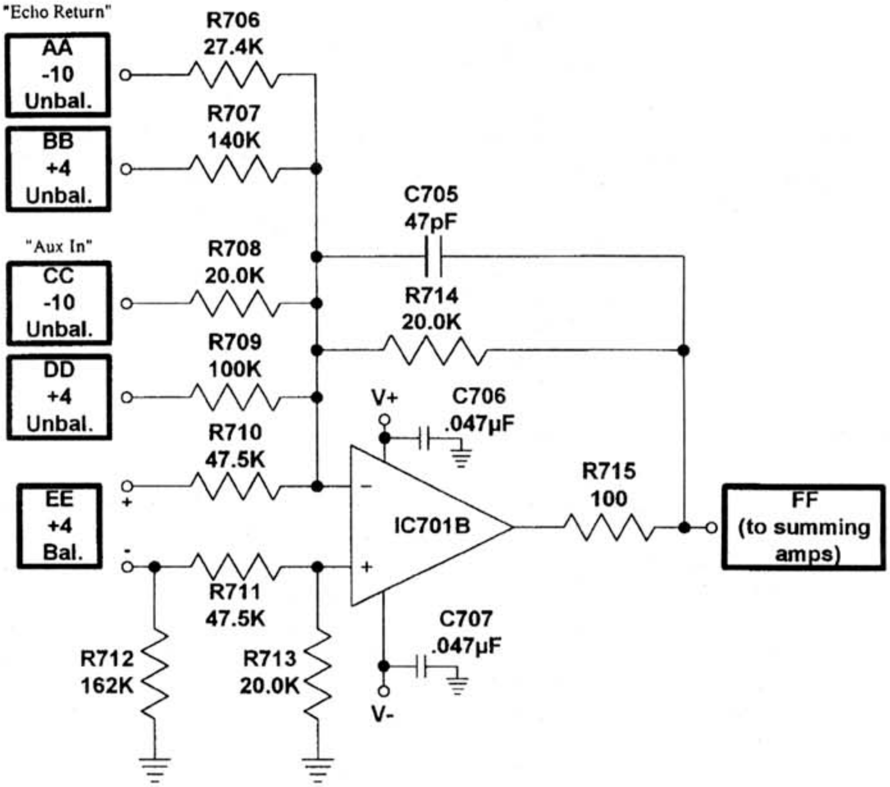

FIGURE 24: Buffer amplifier. The output (FF) on the left card is connected to FF1 on the left summing amplifier, and to FF1 on the mono summing amplifier. On the right card the output (FF) is connected to FF1 on the right summing amplifier and FF2 on the mono summing amplifier. Since most echo devices have mono outputs, the echo-return inputs (AA and BB) are connected in parallel on the left and right cards. Don't install the buffer amp circuit on the mono (house) card.

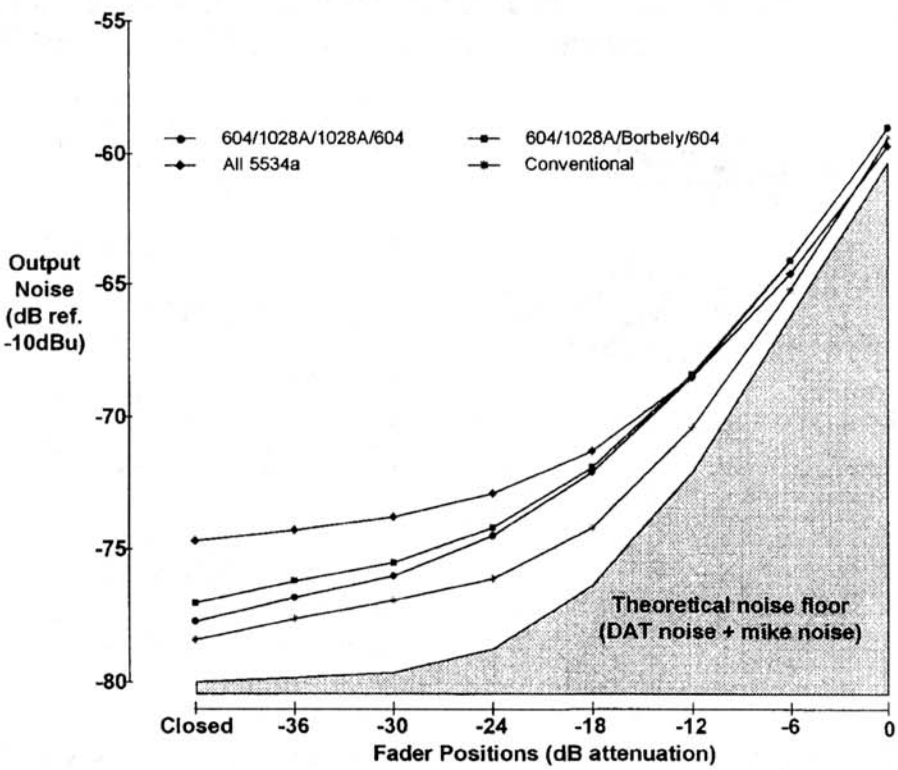

FIGURE 25: Total output noise for a ten-input mixer, using various active devices. They are listed in the order "Input amp/secondary amp/summing amp/output amp"; see the sidebar for a description of the "conventional" design.

capacitance may be a factor in the connections between the panpots and the left/right summing cards. If you keep it below 270pF, the bandwidth will be greater than 200kHz (worst case). To preserve good stereo imaging, these cables should all be the same length.

The on-card power supply is a modified version of the well-known Sulzer regulator, as described in Part 2. This regulator's performance is dependent upon the amplifier's gain-bandwidth product: the faster the op amp, the lower the impedance at a given frequency. I've had good results with the NE-5534, the National Semiconductor LF-357, and the Harris HA-2625. High-speed chips may become unstable in this circuit. If you try them, be sure you can look at the regulator outputs with a 200MHz or faster scope. The rolloff capacitors in the feedback circuit are polypropylene or polyester; the electrolytics are low-series-impedance caps from Panasonic. Try to avoid substitutions.

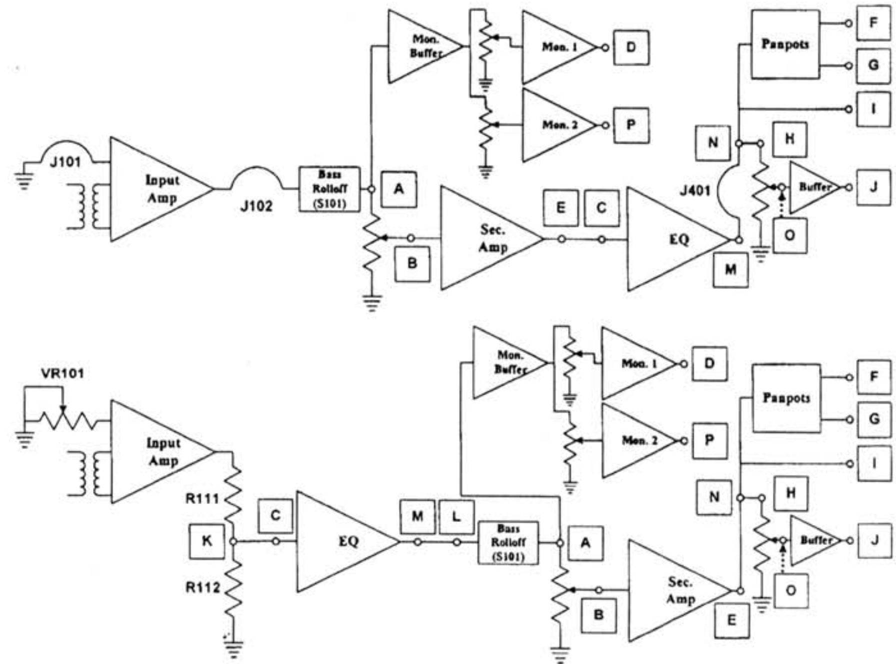

FIGURE 26: Input-card block diagrams. The upper hookup is used for the chimp-proof configuration, the lower for the more conventional design.

The supply inputs are decoupled with either ceramic-disk or stacked-film capacitors. The former enjoy a terrible reputation among audiophiles, but I suspect their presence upstream from the supply regulator won't degrade the sound. Besides, they are the most effective caps for filtering RFI.

The phantom power regulators are designed to use floating IC chips. A simple zener regulator will also fit on the board if you are economizing. You

TABLE 9

INPUT CARD PARTS VALUES

PART	MIXER	
	CHIMP-PROOF	CONVENTIONAL
C103	220pF	150pF
J101	Used	Not used
VR101	Not used	20k reverse log
R104	6.19k	261
R105	3.65k	5.11k
R111–112	Not used	4.99k
R304	22.1k (1.62k)	24.3k (1.78k)
R305	26.7	200

Assumes that an OPA-604 is used for A101, and an LT-1028A is used for A201. If an NE-5534a is used for A201, use the values in parentheses. Because of input current problems, don't use an NE-5534a for A101 in a conventional mixer.

may not need fancy devices anyway, since phantom supplies are almost always reregulated in the microphone, using a zener diode.

Somewhat surprisingly, I find I prefer the old-fashioned, cheap phenolic circuit boards to the more modern glass epoxy ones in any application where they aren't exposed to much heat. They're easier to drill, and I swear they sound cleaner (lower dielectric absorption, for one thing). Teflon® boards are the best, but the price is sky high.

Don't neglect the most vulnerable connection in the mixer: use high-quality, gold-plated XLR input jacks, such as Switchcraft QGP series or Neutrik. Silver-plated jacks oxidize.

Summing/Output Cards

These cards include summing amplifiers, output amplifiers (balanced +4dBu and unbalanced −10dBu), summing amp buffers, VU meter buffers, and on-card power-supply regulators. The same cards, with different resistor values (*Table 5* and the parts lists available from *Audio Amateur*), are used for the various outputs: left/right, mono (house), monitor, and echo-send.

I have been describing a ten-input mixer because it is a size I find useful. But I've also left a couple of extra pads open on the summing cards, which accommodate up to 12 inputs.

Most of my suggestions about input card component choice also apply to the output cards, such as using shielded microphone cable to connect the cards to the output jacks, which should be gold-plated XLRs. For unbalanced inputs and outputs, I recommend ¼″ three-conductor phone jacks, with the "ring" connection grounded. These jacks never go *blaaap* when a cord is accidentally pulled halfway out (unlike RCA or two-conductor phone jacks).

Don't cheap on the VU meters. Professional-quality VUs—running at least $50 apiece—have controlled ballistics which allow the display to correlate fairly well with perceived loudness. Cheap meters may be labeled "VU," but they don't meet the ANSI standard, responding either too quickly or too slowly. Try to find good surplus units, or strip them from old pro-

Audio Amateur will send you parts lists of this three-part project and copies of the board patterns detailed in Parts 2 and 3, if you send us a 9″×12″ self-addressed envelope with one dollar to cover postage and handling, along with a note indicating your request, to: *Audio Amateur*, Editorial Dept., PO Box 576, Peterborough, NH 03458.

Losing My Nerve

Nothing is as simple as we'd like. The chimp-proof mixer is a good design, capable of excellent performance; however, I'd be remiss to ignore the advantages of more conventional designs.

The first is ergonomics. In the chimp-proof mixer, a single knob controls a channel's level in the mix. This might require that one fader be set at –30dB, with the adjacent one at –5dB. To create a monitor mix which is close to the house mix, each monitor pot must have the same setting as its associated channel fader. Inconvenient.

In a conventional mixer, the gain trimpot is set to give a normalized voltage at the top of the channel fader. The usual sound check procedure is to place all the faders at a standard level (usually between –6dB and –12dB, depending on the manufacturer), then set the gain trimpots for the basic mix. Any deviations from standard (such as instrument changes) are controlled by moving the channel fader, but you can easily return to the basic mix by adjusting all the channel faders to standard level. Similarly, once the gain trimpots are set, you would adjust all the monitor pots to the same level (i.e., straight up) to set the basic monitor mix.

The second advantage to conventional designs is EQ stage placement. It follows the secondary gain stage in the chimp-proof mixer. This guarantees that the EQ stage will run at a nominal level of –10dBu, which provides plenty of headroom, even with 10dB of EQ dialed in. But it won't let you feed an equalized signal to the monitor mix; if the microphone needs to be cleaned up, the house will hear the improvement but the performer won't. (A surprising number of commercial boards, although conventionally designed, still take the monitor feed from before the EQ amp—why?)

Best of Both Worlds

It's possible to partially remedy both of these problems. I've designed the input cards in such a way that you can implement either the fully chimp-proof design or a suitable compromise.

Figure 26 shows the signal flow for both designs. In the chimp-proof mixer the signal goes through the input amp and the bass rolloff filter, and splits to feed the monitor pots and the channel fader. It then passes through the secondary gain and EQ stages to feed the panpots, mono summing amp, and echo pot.

The more conventional design has a variable-gain input amp, which is followed by a pair of dropping resistors, the EQ amp, and the bass rolloff filter. This feeds the channel fader and monitor pots. The fader is followed by the secondary gain amp, which feeds the panpots, mono summing amp, and echo pot. The input amp is made variable gain by substituting a 20k reverse-log taper pot (VR101) for J101 and changing resistor values (*Table 9* and parts lists available from *Audio Amateur*).

The lowest practical gain for A101 is 2dB, and the highest is 26dB. Because the transformer has a 1:10 turns ratio, the hottest microphone signals will bring the amplifier to +22dBu peak. This may be fine for the input amp, but not for the EQ amp. After all, when the EQ pots are set to boost certain frequencies, they create gain. The hottest signal, combined with several decibels of boost, could conceivably create clipping in the EQ amp.

The two 4.99k resistors, R111–112, alleviate this problem by dropping the signal level 6dB before it enters the EQ amp. This means you can use up to 8dB of EQ boost before clipping becomes a possibility. While it's no longer perfectly chimp-proof, only the worst combination of super-hot microphones and excessive boost will actually cause trouble.

Note that the signal feeding the monitor buffer is now inverted, since it's been flipped by the inverting EQ amp. Since an even number of inversions is required to preserve polarity, convert the monitor buffer into a voltage follower by deleting R214, installing J201, and changing R201 to 200k.

Does the variable-gain input stage allow you to set all the faders to nominal level? Not quite, but close. The minimum gain for the input amplifier, including the transformer, is 22dB. With the resistor values chosen in the secondary gain stage, the fader can be set to a nominal –12dB position for most circumstances; however, it will need to be progressively lowered for situations where the channel gain is less than 21.5dB (corresponding to a microphone input level greater than –31.5dBu). Similarly, the fader needs to be raised for the softest mikes, where the input level is less than –54dBu. In practice most microphones work at nominal settings most of the time, so the monitor pots can also be set to nominal.

Not-So-Perfect Arrangement

This arrangement is a compromise; a fully adjustable input stage would require a major circuit redesign. Most FETs, especially those in integrated circuits, are noisy. To stay quiet they need a high-ratio input transformer. This dictates that the minimum gain for the input stage approximates the voltage step-up from the transformer, since noninverting op amps cannot go below unity gain.

An excellent, quiet variable-gain input stage can be made from an LT-1028A, using a 1:2 or 1:4 ratio transformer. Unfortunately, the LT-1028A is a bipolar input chip. Its input bias current would cause VR101 to become noisy very quickly. (It would also create low-frequency *whumps* when the gain was adjusted.) To prevent this you would need to insert a large electrolytic coupling capacitor—on the order of 1,000μF—in series with the pot. I prefer not to use these capacitors in audio circuits, as they degrade the signal more than I'm willing to accept.

I present you, therefore, with two alternative designs for the mixer's input card. Which would I choose? The answer depends on the circumstances. For recording in relatively controlled settings, such as concerts and clubs where there's time to do a sound check, I'd probably opt for the conventional hookup. But, for more wild-and-woolly situations where mixing is done on the fly and devil take the hindmost, I'd choose the chimp-proof design.

Of course, you could always build a half-and-half mixer: inputs 1–6, say, could be variable gain, and 7–12 could be chimp-proof. After all, you never know when you'll find yourself up a tree.—**PS**

SOURCES

Digi-Key
701 Brooks Ave. So., Thief River Falls, MN 56701
(800) 344-4539, FAX (218) 681-3380

Mouser Electronics
958 N. Main St., Mansfield, TX 76063
(800) 992-9943

Newark Electronics
4801 N. Ravenswood Ave., Chicago, IL 60640
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Old Colony Sound Lab
PO Box 243, Peterborough, NH 03458
(603) 924-6371, FAX (603) 924-9467

Sonic Frontiers
2690 Brighton Rd., Oakville, ON L6H 5T4
(800) 769-0747

Triode Labs
(312) 871-7459

Welborne Labs
PO Box 260198, Littleton, CO 80126
(303) 470-6585, FAX (303) 791-5783

fessional gear. I prefer to use three meters on a mixer for left, right, and mono levels (the latter is switchable to monitor or echo-send). If your budget is tight, omit the third meter and make one switchable stereo unit.

Meter calibration is crucial. Run a 400Hz tone through the mixer, and set the −10dBu output you're adjusting to read −10dBu (0.245V) on an external meter. Now, with the meter connected through a 3.62k resistor, set the trimpot to produce +4dBu (1.23V) at the output of the meter buffer. If the meter doesn't read exactly 0VU, substitute different resistors until it does. A 5k trimpot and a digital multimeter may be useful here.

Raw-Supply Cards

The power-supply cards designed for the simple microphone preamps described in Part 2 also work for a multi-input mixer. The preregulator chips, however, dissipate a lot more power in this application. They should be TO-3 units, mounted in ample heatsinks on the mixer's back panel. It might be a good idea to substitute the LM-317 and LM-337 with the Linear Technology LT-1085 and LT-1033 3A adjustable regulators. The power transformers and fuses should likewise be rated for more current. Be sure to use ferrite beads on all incoming leads, and on the external power pack's outgoing leads.

A five-pin XLR connector attaches the external power supply: Pin 1 is the main ground; Pins 2 and 3 are the plus and minus rail voltages, respectively; Pin 4 is the phantom supply's ground connection; and Pin 5 is its plus connection. Use a shielded multiconductor cable, with the shield connected to the chassis ground of the power supply box but not to that of the mixer.

Overall Construction

Portable mixers can be housed in many types of enclosures. I like the slanted Bud "turret racks" sold by Newark Electronics for attaching to rack cabinets. Most of the circuit cards mount on the front panel, leaving the back for connectors and heatsink-mounted preregulators. You may prefer a flat-topped unit or a rectangular box. I've occasionally toyed with building a wooden frame to which I could attach a standard rack panel, but the lack of RFI shielding put the kibosh on that idea. However you build your mixer, be sure to allow some ventilation, perhaps by making one of the rear panels from perforated metal.

I prefer to mount my (rotary) faders on a panel which is separate from the main circuit cards. This lets me replace them without removing the whole caboodle from the enclosure, or in the case of slide faders without redrilling an entire front panel.

If you decide to build a studio mixer with these circuit designs, the sky's the limit. You could use discrete circuits (Borbely or Jensen) throughout, or adapt them to feed a multitrack recorder.

Remember that multi-input mike mixers are complex systems. If you find that odd combinations of stages are interacting in peculiar ways, grounding problems are a likely cause. Follow good procedures: always ground cable shields at one end only.

I offer these designs as raw material; they can be interconnected in many different ways. I'd love to hear about how you've used them. Feel free to write (c/o *Audio Amateur*), and I'll try to answer as quickly as possible.

Good luck, and happy recording! ■

REFERENCE

1. W. Jung, Audio IC Op-Amp Applications, 2nd ed. (Howard W. Sams, 1978): 131–136.

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