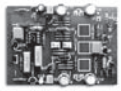


PHOTO 1: PM-2 preamp mixer/monitor.

A Purist Recording Mixer

This box is a triple threat—mike preamp/mixer/monitor—and brings you a giant step closer to experiencing the joys of recording live music.

By Dennis Colin

Named the PM-2 (for preamp/mixer/monitor), this component is designed for two recording applications:

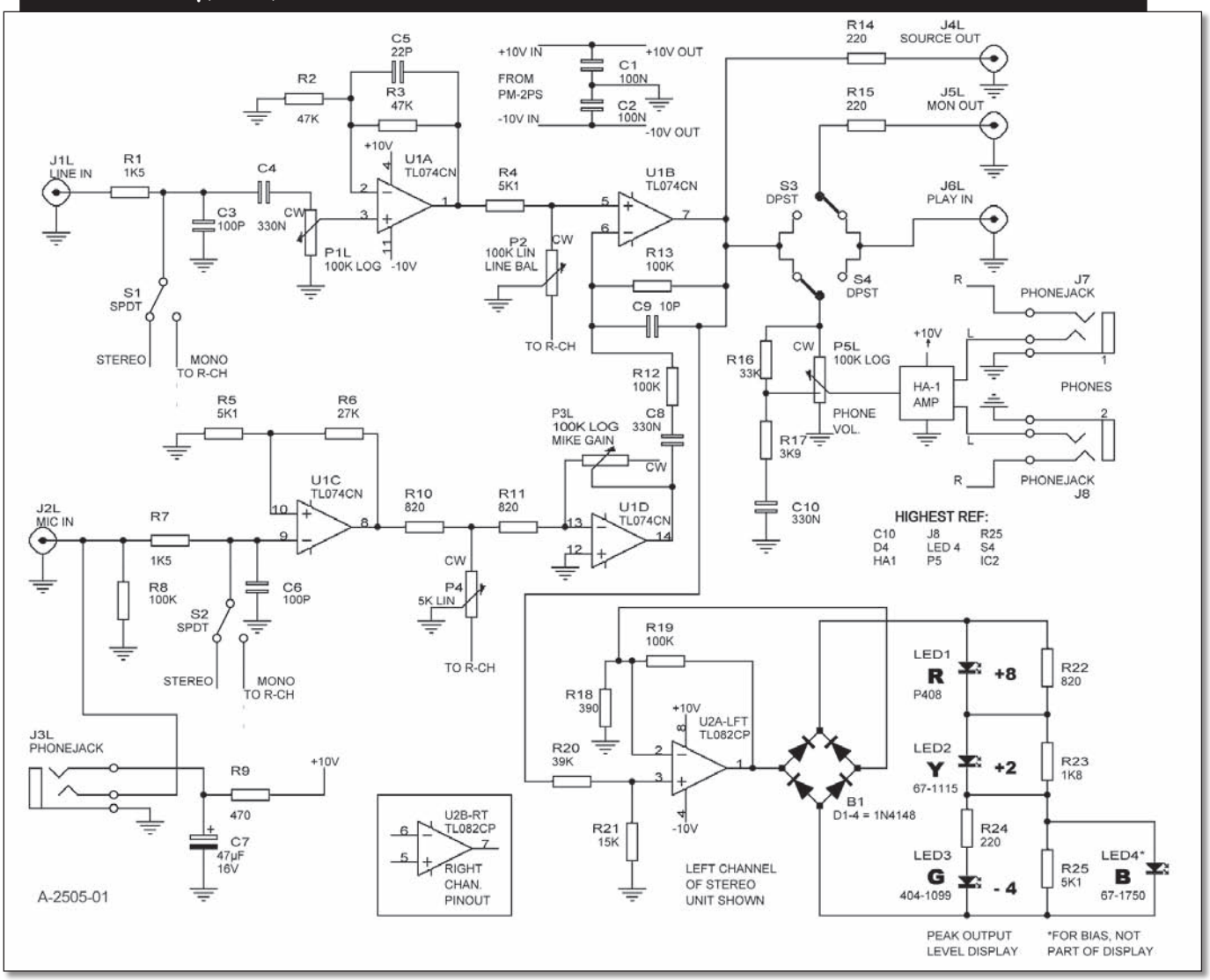
1. Live music, with a "purist" or "minimalist" single stereo pair of microphones.

2. Transfer of pre-recorded music, e.g., on vinyl or tape, to a CD (or other medium), with the option of real-time voice dubbing.

OVERVIEW

For live music recording, the simplest audio path ("minimalist") includes only a stereo pair of mikes, with their preamps directly feeding the recorder. The PM-2 audio path (**Photos 1** and **2**) com-

FIGURE 1: Preamp/mixer/monitor. Left channel of stereo unit shown.



prises a mike preamp with line-level inputs summed into the output, if desired. (The line inputs could accept additional mike/mixer outputs, a band's PA feed, a direct electric bass signal, and so on.) With a purely acoustic ensemble in a good-sounding venue, an optimally-placed stereo mike can often provide astonishing clarity, detail, stage coherence, and spaciousness.

PHOTO 2: Rear panel.

For transferring vinyl, tape, and so on to a CD, use the line inputs, but you can mix in an adjustable amount of mike signals, e.g., for voice announcements between songs, narration, or (for those so inclined) karaoke sing-along.

FEATURES

1. Level and stereo balance controls, and stereo/mono switch, for mike and line channels.

2. Inputs for hi-z dynamic mikes, or electret mikes such as the (extremely high-quality) Mitey Mike II (*SB* 4/97, Dr. Joseph d'Appolito and Richard Campbell). For such electrets, pure DC on the "rings" of stereo jacks allows remote ("phantom") powering.

3. High-quality headphone amp (outputs for two pairs), switchable between the mixer output ("source") or an external signal (usually the recorder's playback) connected to the PM-2's jacks labeled "Play In." The phone amp can drive the high-sensitivity 300Ω Sennheiser HD650 phones cleanly to 117dB SPL.

4. Monitor outputs for driving a stereo amp and speakers, independently switchable as above ("source" or "play").

5. Peak output level-indicating lights for –4, +2, +8dBu (0.7, 1.4, 2.8V peak).

Note: +8.24dBu = 2.00V RMS = 0dBFS = RedBook CD full scale.

If desired, you could label the level lights –12, –6, and 0dBFS. In any case, the red lights indicate instantaneous peaks of ±2.83V, which is "brick wall" digital clipping on a CD, and should never be exceeded.

PREAMP/MIXER SCHEMATIC

I used a TL074 quad op amp (U1 A, B, C, D) for the audio (**Fig. 1**); you can substitute other op amps. But note that while the TL074 output stage is usually considered to be Class AB, the maximum signal current drawn here is less than that stage's bias, so it's operating in pure Class A.

The line input is RFI filtered, DC blocked, and attenuated by P1L, R (for left, right channel), then amplified ×2 by U1A. With P2 (balance) set midway, the signal is reduced by half. This is fed to the non-inverting input of U1B, where the signal "sees" a gain of two because of R12, R13. Thus, the maximum gain (P1 up full) from line in to source out is 2 (+6dB).

MIKE CIRCUIT

The preamp (U1C, D) is admittedly not state-of-the-art in noise level with a 10kΩ source (typical hi-z dynamic mike). With 24nV/√Hz equivalent input noise density, root-sum-squared with the 12.7nV/√Hz noise of a 10kΩ source resistance, this source noise is increased by 6.5dB (the noise figure of the preamp with 10kΩ source).

However, dynamic mikes are not the best in fidelity for purist recording! I designed the PM-2 primarily for use with the Panasonic WM-60AY electret mike capsule (used in Mitey Mike II). I modified the capsule (more on this later) and incorporated a remote ("phantom") powered preamp, result-

ing in a sensitivity of 10mV/Pa and 53nV/√Hz noise density. The signal (and mike capsule noise) are thus amplified to a point where the PM-2 mike preamp's noise produces only a 0.8dB signal-to-noise ratio reduction (0.14dB with the Mitey Mike II because of its higher 25mV/Pa sensitivity).

Using the WM-60AY with its incorporated preamp (it should be called a "pre-preamp" to distinguish it from the PM-2 preamp), and plugged into the PM-2, the "A" weighted noise floor is 32dB SPL, and the 2% THD point is 133dB SPL (*much* too loud for safe hearing!), resulting in a dynamic range of 101dB, 5dB more than the (sonically questionable) 96dB dynamic range of a CD.

The output of U1C is fed, through the balance control, into the *inverting* input of U1D, then summed into (and polarity restored to non-inverting by) the output stage U1B. With P3L, R at maximum, and P4 (mike balance) midway, the maximum 2nd stage gain is 47.3. With the input stage (U1C) fixed gain at 6.3, the overall maximum mike channel gain is (measured at) 298 = 49.5dB.

This is quite a bit less gain than that of mike preamps on tape recorders, or preamps designed to capture the quietest sounds (bird songs, surveillance, and so on). But the PM-2 is optimized for music recording, where the average SPL may reach or exceed 110dB, with waveform peaks of 125dB. With maximum mike gain, the PM-2 with a Mitey Mike II will output 2V RMS (0dBFS on a CD) at 82dB SPL (**Table 1**). This is sufficient mike gain for any kind of live music; 82dB peak SPL corresponds to about 67–72dB average SPL, about that of quiet conversation.

INDICATOR LIGHTS

U2 (half of the dual op amp per channel) feeds an AC current, nearly proportional to the mixer output voltage, to diode bridge D1–4 (**Fig. 1**). Rectified, this current is fed to the series string of green, yellow, and red LEDs, each with threshold-setting resistors across them. The blue LED (4) acts as a voltage clamp for the green LED (3), helping maintain brightness balance of the three displayed LEDs (1, 2, 3). You must use the specified LEDs (available from Digi-Key) for correct thresholds and visibility balance.

HEADPHONE AMP

Figure 1 shows the phone amp (HA-1) fed from a stereo 100kΩ pot (P5L, R) with loudness compensation (10% tap, R16, R17, C10). The latter is effective but not "boomy," compensating down to about 70dB headphone SPL for low-volume reduction in LF hearing sensitivity.

Figure 2 shows the headphone amp schematic. It's the same amplifier described in my "A High Quality Solid-State Headphone Amp" (*aX* Nov. '05), except biased for ±10V supplies instead of ±13V. Q3–4 are a unity voltage-gain power stage, operating in Class A with one pair of 300Ω phones; with two phone pairs or 32Ω phones, operation is Class AB at high power, but is still sonically distortion-free. With constant-current biasing (Q1, 2), the output stage drive current is less than the op amp's output stage bias, so U1 operates in Class A.

Frequency response of the overall amp is DC –230kHz (–3dB). The 51Ω output resistance (from R12, 13) serves three functions:

1. Output is unconditionally stable.
2. Output is short-circuit proof.
3. Similar power is delivered to 300Ω and 32Ω phones (about 3dB more to 32Ω phones, but they tend to be less power-efficient than 300Ω phones).

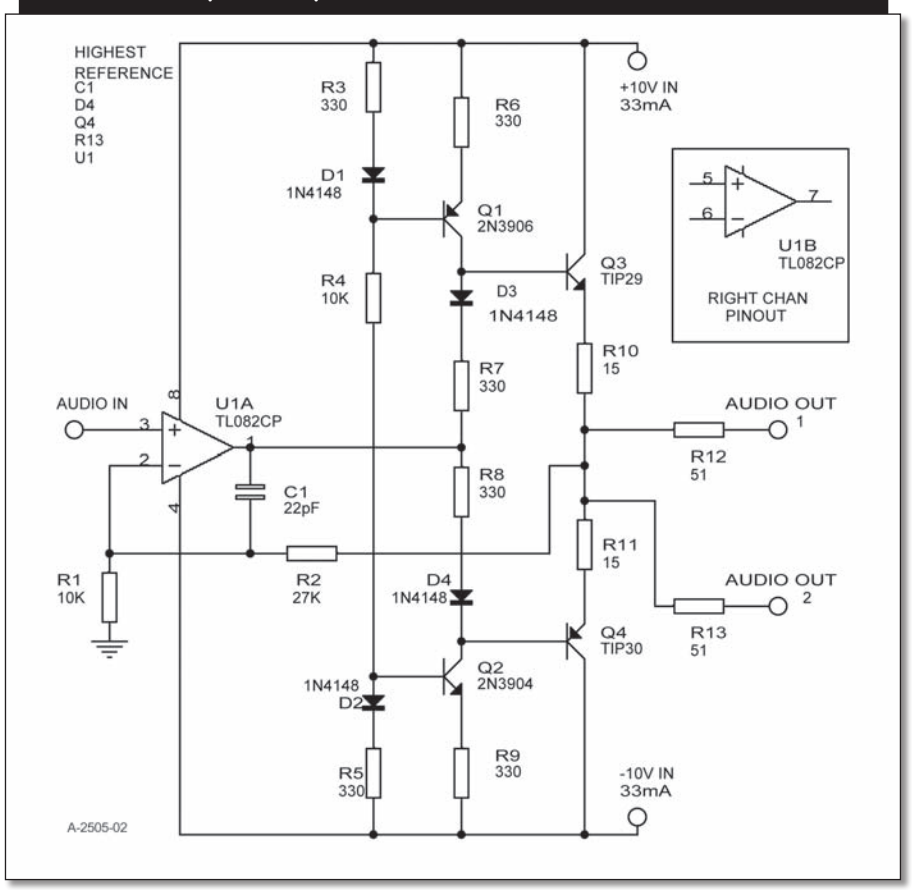
Of course, you can use the PM-2 as just a headphone amplifier (feed your

TABLE 1: Microphone Performance with PM-2.

		Mitey Mike II	Mike of *Fig. 6*
Maximum SPL	Mike Alone	130dB (3% THD)*	133dB (2% THD)
	With PM-2	125dB	133dB
Sensitivity		25mV/Pa*	10mV/Pa
Noise floor, "A" weighted		≤36dB SPL*	32dB SPL
SPL for 2V RMS (CD 0dBFS)		82dB SPL	90dB SPL
PM-2 out (maximum mike gain) Same as above for 20mV		42dB SPL	50dB SPL
(–40dBFS) out Dynamic range with PM-2		≥89dB	101dB

* *SB* 4/97, by Dr. Joseph d'Appolito and Richard Campbell.

FIGURE 2: Headphone amp.



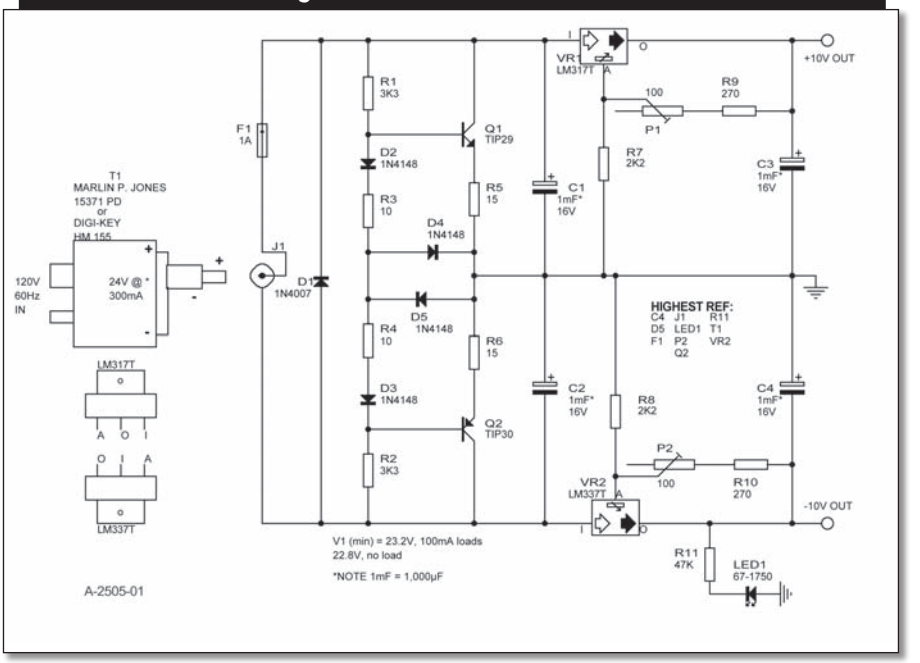
source player into the PM-2 line inputs) when not making recordings. And if you want to hear outside sounds (wife calling, phone ringing, elephant trying to get into swimming pool) while listening to loud headphone music, you can plug a microphone in and mix it into the music (and hear yourself singing along in private).

POWER SUPPLY

To render AC line hum inaudible, use a "wall wart" external 24V DC supply (**Photo 3**). But the audio circuitry (**Figs. 1, 2**) needs a split bipolar supply (±10V regulated used here). So the incoming single-ended 26V (at the 100–150mA used) needs a "DC center tap" to serve as circuit ground. You could use a series pair of resistors, but unless a small value (wasting a lot of current), the derived center voltage would be sensitive to positive versus negative supply load

PHOTO 3: 24V DC wall supply.

FIGURE 3: ±10V power supply for audio preamp/mixer/monitor PM-2. Maximum output current: 250mA if balanced; 60mA imbalance (limited by Q1, Q2 circuit). FWHK FE4823240DO30. *25.9V at 100mA, 120V RMS in Marlin P. Jones # 15371PD Jack CTR. = +. Highest Ref. C4 D5 F1 LED1 P2 Q2 R11 T1 VR2.



imbalance.

Q1, 2 provide a "stiff" (low impedance) center tap. Basically a Class AB amplifier output stage, the bases are biased equally above and below the single-ended input voltage's midpoint (offset by D2, 3 and R3, 4 to compensate the base-emitter drops of Q1, 2, plus a little more to establish the Class AB bias current, stabilized by R5, 6). D4, 5 limit the voltage drive across the output transistors' base-to-tap output points, in the event of a supply output-to-ground short, thereby limiting excessive ground current to protect Q1, 2.

A power op amp such as the LM675T can do this supply-splitting function (see National Semi's app notes), but it's not unity-gain stable, more components may be needed, and more bias current (than the circuit used here) may be drawn. D1 protects against reverse-polarity input DC (or an AC wall wart mistakenly used), and fuse F1 blows if all else fails. Following the supply splitter, conventional regulators and 4000µF of filter caps provide very clean ±10V supplies to the audio circuitry.

Important note: The DC input jack (not shown on schematic; power supply jack J1 on parts list) *must be insulated from the enclosure chassis*. To do this, I mounted the jack centered on a piece of 1/16″ thick plastic sheet, about 1½″ square, then attached this over the inside chassis surface, centered on a ¾″ chassis hole (**Photo 2**). I fastened the plastic with CA glue, but this broke loose after a while, so I re-glued it, then beefed it up with epoxy. I recommend mounting screws, even though I was too impatient/lazy to use them.

CONSTRUCTION

The enclosure I used is by LMB, part number UNC 5-6-6 (Digi-Key L159). It's attractive, painted textured blue and white, but possibly not large enough to accommodate a PC board holding all the parts. I needed to use two boards: the audio board (**Photo 4**) being solid copper with breadboard construction, mounted on the inside top of the cabinet, and a supply board (**Photo 5**)—a piece of balsa with parts glued on—mounted on the right side of the other cabinet half (the base). I used two- and three-pin connectors to connect the supply board to the DC input and the audio board, respectively.

Depending on the size required for a *real* PC board, you might need a larger cabinet. **Photos 6**, **7**, and **8** show my congested, impatient, Mickey Mouse wiring. Note that I didn't use the power transistors specified—I used OHS (On Hand Stock) units, but the phone amp and supply-splitter are very insensitive to transistor parameters. The TIP29 and TIP30 parts specified are well overrated for their jobs (**Fig. 4**).

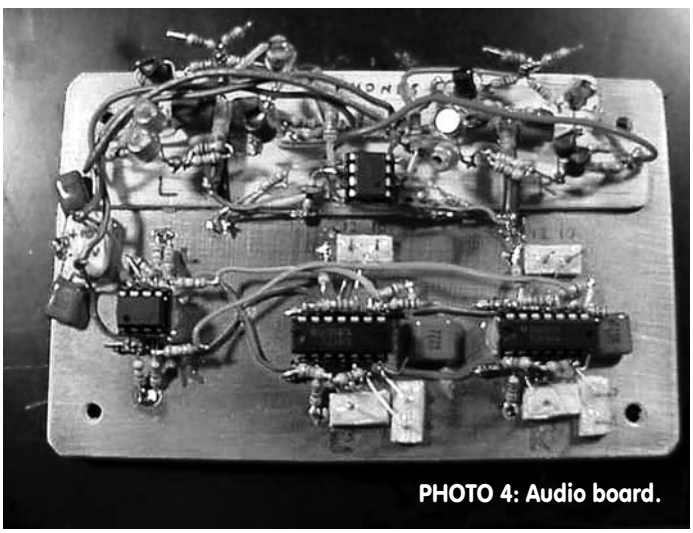


PHOTO 4: Audio board.

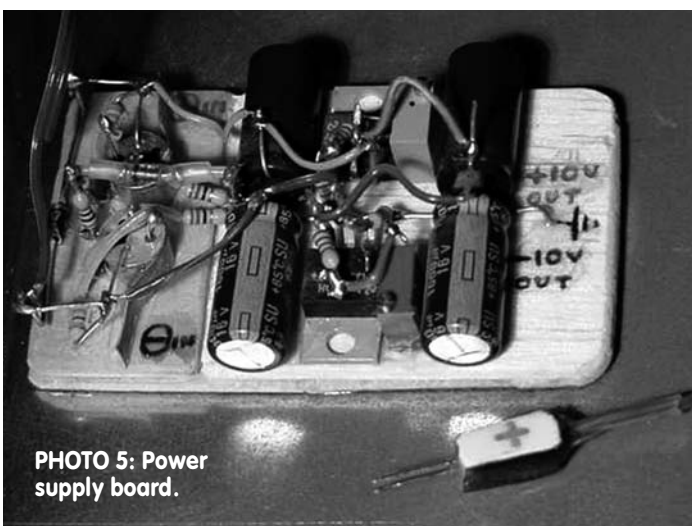


PHOTO 5: Power supply board.

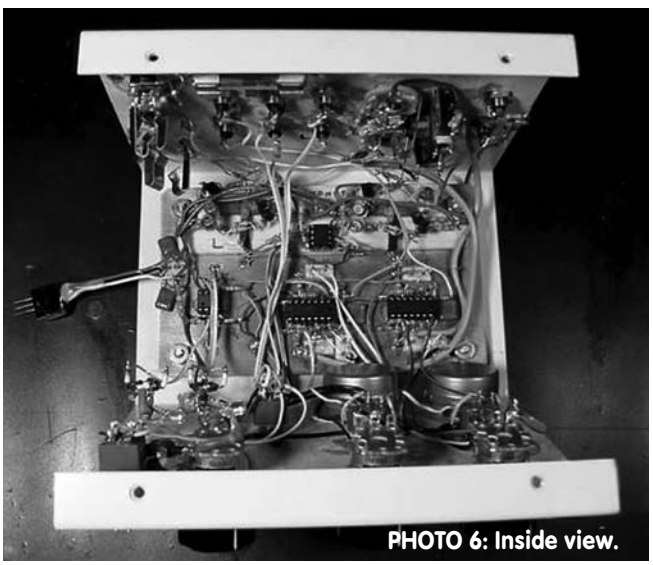
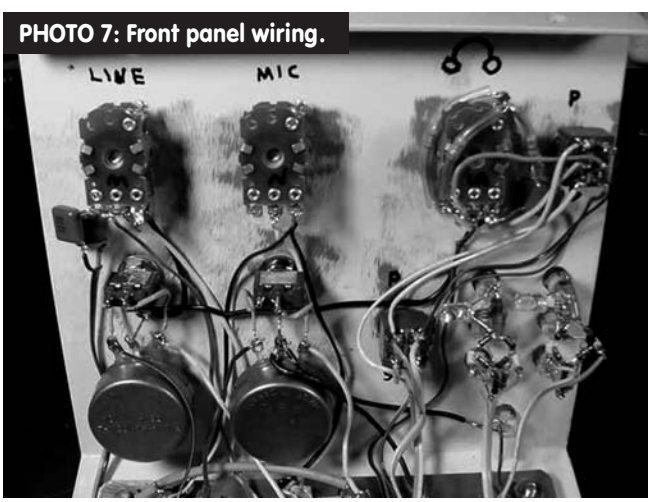


PHOTO 6: Inside view.

PHOTO 7: Front panel wiring.



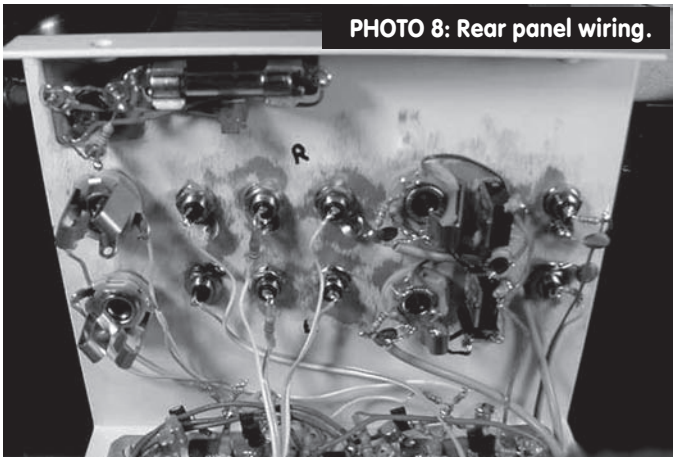


PHOTO 8: Rear panel wiring.

SCRAPE THAT PAINT

You must grind off the thick paint (on the cabinet type suggested) to bare metal surrounding all controls and switches, and audio jack mounting holes (you must ground the pot and switch shafts to avoid hum pickup, and so on). You could mount the audio board with screws, but instead, to avoid protruding screw heads aesthetically upsetting the nice smooth cabinet top, I:

1. Located the four board corner mounting hole locations on the inside cabinet top surface.

2. Ground off the paint on a ½″ circle around the marked locations.

3. Silver-epoxied four ½″ #6 machine screws' flat heads to the chassis. "Circuit Works CW2400" is a room-temperature curing conductive epoxy made by Chemtronics (Kennesaw, GA 770-424-4888).

4. After that cured, I beefed up the joints by applying two-part epoxy around them.

This worked well—the joints have held up, and the screws are well-grounded, which with nuts and lockwashers clamping the audio board's solid copper ground plane, provide a very solid audio ground. I used, and recommend using, a layer of ⅛″ thick balsa wood (or rubber sheet) between the board and chassis to prevent board flexure or possible rattling.

MICROPHONE INPUT JACKS

I used (**Photo 2**) ¼″ phone jacks for hi-z dynamic mikes, and ⅛″ stereo jacks for electrets (+10V DC on the "rings"). I used ⅛″ jacks (and plugs on my mikes) so neither mike type could be plugged into the wrong jack. But the ⅛″ plugs and jacks I used are sometimes intermittent (117dB SPL transient blasts in the phones!), so I recommend ¼″ stereo phone jacks for the electret mike inputs. If you plug a dynamic mike into the electret jack, the mono plug sleeve will short the 10V DC. But no harm is done; the 470Ω resistor (**Fig. 1**, R9) will safely limit the short current to 21mA.

A HIGH DYNAMIC RANGE RECORDING MICROPHONE

The Panasonic WM-60AY (**Photo 9** $2.64 from Digi-Key, P/N P9959) is used in Mitey Mike II (*SB* 4/97). Its response is so flat (typical ±1dB, 20Hz–20kHz) that *Stereophile* uses a MMII for speaker measurements.

Figure 5 shows what's in the electret capsule, measured data, and a circuit model. The internal FET source, after a 75Ω resistor, is originally grounded. The MMII article describes how to break this connection, so that you can add an external source-to-ground resistor. This greatly increases the maximum SPL capability.

I modified this and mounted the capsules with heat-shrink tubing (careful—

PHOTO 9: Modified WM-60AY mike capsule with connector.

FIGURE 4: Transistor pinouts.

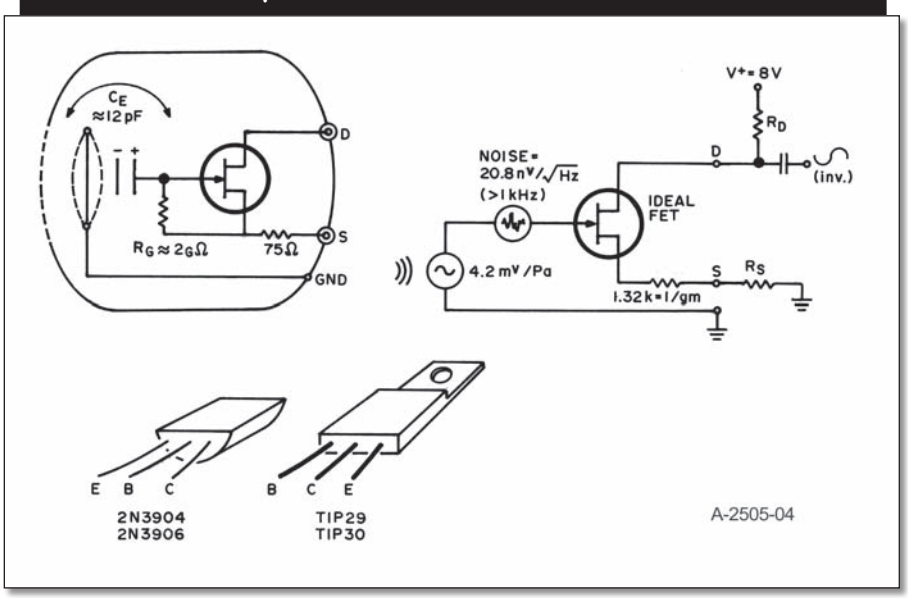
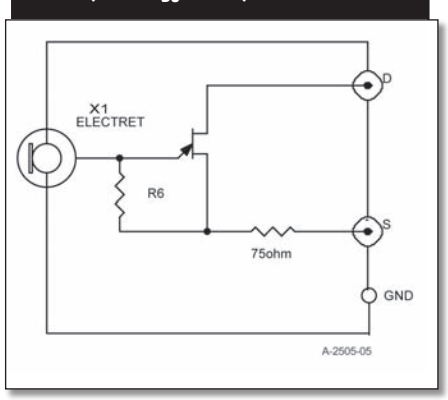


FIGURE 5: Panasonic WM-60AY electret mike. When the original connection between S and gnd. is broken, nearly constant-current (300 ± 50μA over ext. Rs range of 0 – 20kΩ) establishes that Rg is returned to FET S, not gnd. Circuit Model Accuracy ≈ ±1dB for sensitivity and noise (>1kHz) calcs. (4V ≤ V_{DS} ≤ 10V)



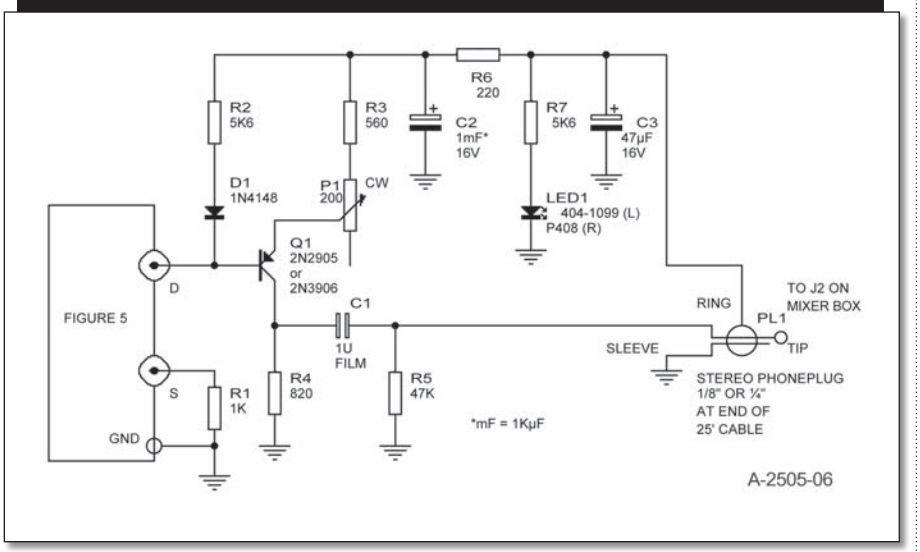
low heat!) to a 3-pin connector, which plugs into a receptacle at the end of a copper tube attached to a 3¼" × 2 ⅛"× 1 ⅝" Bud box (**Photo 10**). The "mike wand" end of the box is covered with ½" thick foam rubber to prevent HF reflections back to the mike. The bottom of the box has a standard receptacle for mike stands.

PHANTOM POWERED PREAMP

Figure 6 shows the preamp I built into the mike box (**Photo 11**). The circuit functions as a very linear current mirror, amplifying the mike sensitivity to 10mV/Pa. Maximum SPL (2% THD point) is 133dB. The preamp output

PHOTO 10: Mike assembly.

FIGURE 6: High-performance microphone. Low noise current-mirror preamp in 3¼" × 2 1/8" × 1 5/8" Bud Box, attached to 13" × ¼" O.D. copper tube, "mike wand."



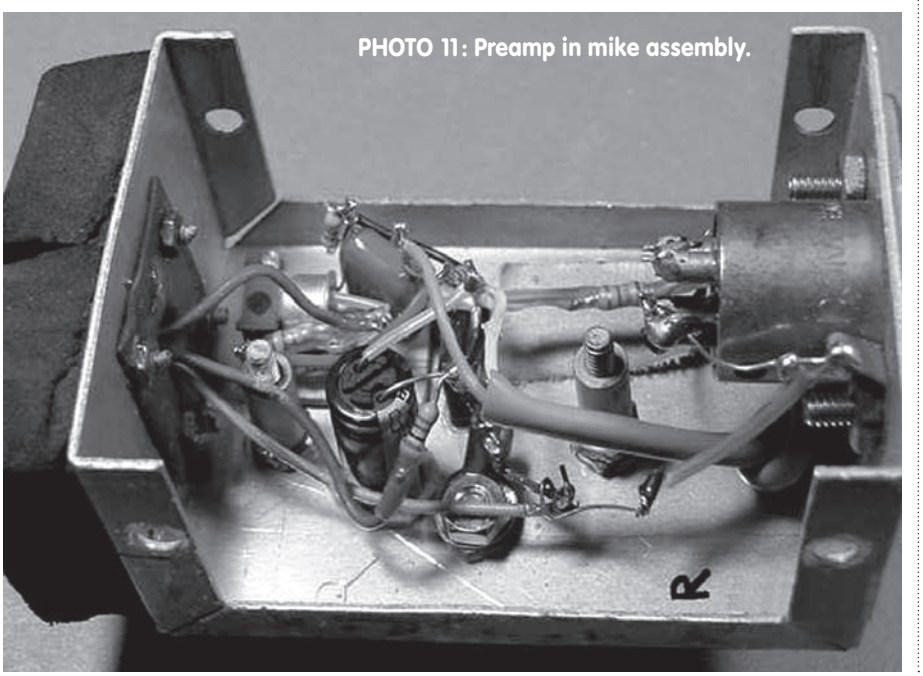


PHOTO 11: Preamp in mike assembly.

impedance is 787Ω, seemingly high. But it takes 2460pF loading (82′ of 30pF/ft. cable) to cause 0.25dB loss at 20kHz. This output topology, though, has advantages over a low-z emitter follower or push-pull op-amp output: no amount of capacitive or other loading will produce slew-rate, crossover, or other distortion. The output stage is basically a "solid-state single-ended triode."

Unloaded, the preamp BW (–3dB) is 5Hz–1.7MHz. Driving 50′ of 30pF/ft. cable, 20kHz loss is only 0.1dB. Both the preamp and (as configured) mike capsule have very high power-supply rejection ratio (PSRR), so with the regulated and R-C filtered DC fed into one conductor of the two-wire + shield cable, you hear absolutely no AC line hum or crosstalk from the other stereo channel of the PM-2 mixer.

Figure 7 shows the noise spectrum of the mike and preamp. **Table 1** shows the performance of this mike, and MMII, feeding the PM-2 mixer. With the mike of **Fig. 6**, maximum SPL (2% THD) is 133dB, corresponding to an average music waveform SPL of about 118–123dB, louder than a large rock band up close (and *much* too loud for safe hearing!). "A" weighted dynamic range of this mike/mixer combo is 101dB.

THE INPUT SIDE OF *AX*

audioXpress deals almost entirely with music playback technology. And as every reader knows, even with the best playback system in existence, the sound is at the mercy of the recording, so many of which use too many (and too close) mikes, not to mention the incessant addiction to processors (EQ, compression, digital "enhancements," and so on) that engineers believe they need.

But when you capture (as opposed to "produce") your own recording of live music with a simple (but well-optimized through much experimentation) minimalist mike pair placement, and a direct path to the recorder, then *what you hear* (on neutral playback equipment) is *what was there*.

TABLE 2: PM-2 PERFORMANCE DATA.

(Note: Red Book CD Full Scale = 0dBFS = 2.0V RMS = ±2.83V peak)
Maximum output: 5.6V RMS, ±8.0V peak
Output impedance: 220Ω
Line Input
- Gain: 0 to 1.94 (+5.8dB)
- Input z: 100kΩ
- Frequency response: +0, -3dB, 4.8Hz - 123kHz; -0.14dB at 20kHz
- Equivalent input noise: Density = 62nV/√Hz, 8.8μV RMS 20kHz BW
- Noise floor = –107dBFS wideband, –109dB "A" weighted

Mike Input
- Gain: 0 to 298 (+49.5dB)
- Input z: 100kΩ
- Frequency response: LF –3dB at 6Hz
  - HF: (maximum gain) –3dB at 42kHz, -0.9dB at 20kHz
  - (maximum gain –6dB) –3dB at 68kHz, -0.3dB at 20kHz
  - (maximum gain –10dB) –3dB at 88kHz, -0.18dB at 20kHz
- Equivalent input noise: density = 23.6nV/√Hz, 3.3μV RMS 20kHz BW
- Noise floor: "A" weighted, re 10mV/Pa mike sensitivity = 22dB SPL; re 25mV/Pa = 14dB SPL
- Noise figure (NF) = reduction of signal/noise ratio from mike:
  - mike/preamp of **Fig. 6** (10mV/Pa, 53nV/√Hz): NF = 0.79dB
  - Mitey Mike II (25mV/Pa, ≈ 132nV/√Hz): NF = 0.14dB
- Maximum input to avoid clipping 1st amp stage (before attenuation) = 0.90V RMS
  = 133dB SPL w/10mV/Pa, 125dB SPL w/25mV/Pa
- Mike input dynamic range (with "A" weighted noise) = 111dB

Miscellaneous
- L/R balance variations: ±0.4dB, attenuation range maximum to –40dB

Headphone amp
- Maximum output: 4.93V RMS = 81mW, 300Ω phones
  2.03V RMS = 129mW, 32Ω phones
  117dB SPL with Sennheiser HD650 phones (300Ω)
- Output z = 50Ω
- Signal level (from PM-3 mix output) for maximum phone output = 1.56V RMS = -2.2BFS
- Frequency response: +0, -3dB, DC –230kHz
- Loudness compensation: maximum at phone volume attenuator = –20dB, constant below –20dB: +3dB at 100Hz, +7dB at 36Hz. Within ±2dB of LF hearing reduction from ≥100dB SPL to 70dB SPL, down to 20Hz.

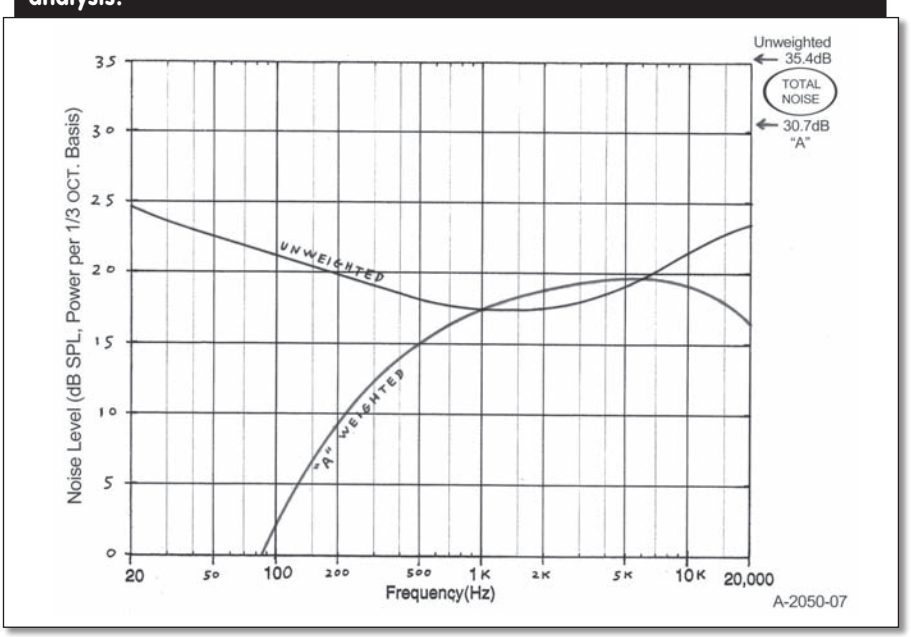
HIGH-PERFORMANCE MICROPHONE

Sensitivity = 10mV/Pa
Frequency Response = ±1dB, 10Hz–20kHz
Maximum SPL = 133dB at ≤2% THD
Pattern = OMNI up to 20kHz
Noise: 52.9nV/√Hz, 1–20kHz
7.29μV, 1–20kHz
11.7μV, 20Hz–20kHz
Noise Floor: 35.4dB SPL, Flat
30.7dB SPL, "A" weighted
Dynamic Range, "A" noise to 2% THD = 102dB

With RD = 5kΩ, V^+ = 8V

R5 Ω	Sens mV/Pa	SPL, dB 2% THD	Noise SPL, dB "A"
0	12.7	125	30.4
300	10.4	127	
500	9.6	130	
700	8.75	131.5	
1k	7.76	133	30.6
1.5k	6.46	135	
2k	5.51	137	
3k	4.30	140	
5k	2.94	144	32.0

FIGURE 7: Noise level, WM-60AY mike and PM-2 MP preamp, 1/3 octave analysis.



Then, if the music and its venue acoustics were very good, you'll have a recording that provides:

1. An accurate, undoctored reference to best evaluate your playback system's fidelity (you might learn more than you wanted to know, or you might be pleasantly surprised).

2. The supreme enjoyment of extremely natural, clear, and three-dimensionally spacious music reproduction. Once you start recording live music, you'll never want to stop. And, hanging around musicians could make you become one—but don't quit your "day job" just yet! *aX*

PM-2 Parts List

Total Qty.	Schematic Ref.				
	Mixer	Phone Amp	Supply	Description	Source
2	C9			Cap, 10pF	
4	C5	C1		Cap, 22pF	
4	C3,6			Cap, 100pF	
2	C1,2			Cap, 0.1µF	
6	C4,8,0			Cap, 0.33µF, 100V poly	Digi-Key EF 1334
2	C7			Cap, 47µF 16V	
2			C1,2	Cap, 1mF, 25V	
2			C3,4	Cap, 1mF, 16V	
20	D1,2,3,4	D1,2,3,4	D2,3,4,5	Diode, 1N4148	
1			D1	Diode, 1N4007	
1			F1	Fuse, 1A, FB	
8	J1,4,5,6			Jack, RCA	
2	J2			Jack, ¼" phone, mono or stereo	
6	J3,7,8			Jack, ¼" phone, stereo	Digi-Key SC1094
1			J1	Jack for T1 (DC in)	Digi-Key SC1048
2	LED1			LED, red	Digi-Key P408
2	LED2			LED, yellow	Digi-Key 67-1115
2	LED3			LED, green	Digi-Key 404-1099
3	LED4		LED1	LED, blue	Digi-Key 67-1750
1	P4			Pot, 5klin	Digi-Key RV4N502C
1	P2			Pot, 10klin	Digi-Key RV4N103C
3	P1,3,5			Pot, 100k	Alps 100k Ax2 aud, stereo Radio Shack 271-1732
2			P1,2	Trimpot, 100Ω	
2		Q2		NPN, 2N3904	Digi-Key 2N3904RLRAOSCT
2		Q1		PNP, 2N3906	Digi-Key 2N3906RLRAOSCT
3		Q3	Q1	NPN, TIP29	Digi-Key TIP29BOS
3		Q4	Q2	PNP, TIP30	Digi-Key TIP30COS
2			R3,4	RES, 5% ¼W 10Ω	
6		R10,11	R5, 6	RES, 5% ¼W 15Ω	
4		R12,13		RES, 5% ¼W 51Ω	
6	R14,15,24			RES, 5% ¼W 220Ω	
2			R9,10	RES, 5% ¼W 270Ω	
12			R3,5,6,7,8,9	RES, 5% ¼W 330Ω	
2	R18			RES, 5% ¼W 390Ω	
2	R9			RES, 5% ¼W 470Ω	
6	R10,11,22			RES, 5% ¼W 820Ω	
4	R1,7			RES, 5% ¼W 1.5k	
2	R23			RES, 5% ¼W 1.8k	
2			R7,8	RES, 5% ¼W 2.2k	
2			R1,2	RES, 5% ¼W 3.3k	
2	R17			RES, 5% ¼W 3.9k	
6	R4,5,25			RES, 5% ¼W 5.1k	
4			R1,R4	RES, 5% ¼W 10k	
2	R21			RES, 5% ¼W 15k	
4	R6		R2	RES, 5% ¼W 27k	
2	R16			RES, 5% ¼W 33k	
2	R20			RES, 5% ¼W 39k	
5	R2,3		R11	RES, 5% ¼W 47k	
8	R8,12,13,19			RES, 5% ¼W 100k	
2	S1,2			Switch, SPDT	
2	S3,4			Switch, DPDT	Digi-Key 360-1156
1			T1	"Wall Wart" DC supply	Marlin P. Jones 15371 PD or Digi-Key HM155
				24V 300mA	
2	U1			TL074CN	
3	U2		U1	TL082CP	
1			VR1	Pos. Reg, LM317T	
1			VR2	Neg. Reg, LM337T	
1				Enclosure	See text