BENCHMARK MEDIA SYSTEMS, INC.

MP-1 INSTALLATION GUIDE

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1.0 INTRODUCTION

1.1 The MicroFrame Series

The MicroFrame Series consists of a modular rack mount chassis that is only 1 Rack Unit high. The rack mount chassis in turn houses 16 modular amplifiers, with front panel access, and two power supply electronic modules. The amplifier modules are ≈ 0.8" wide and the power supply modules are ≈ 2" wide, providing a total of 18 modules in a 16.8" wide chassis that is only 13 inches deep. The very high density utilized with the MicroFrame Series provides some of the most cost effective use of rack real-estate in the industry.

1.2 The MF-1

The MF-1, a single rack unit *M*odular *F*rame, is designed to house two power supply modules and sixteen amplifier modules. This arrangement provides a redundant power configuration from a common power transformer and power entry module. The IEC type power entry device at the rear of the frame is a combination line filter, mains voltage selector (120-240 volt 50/60 Hz operation) and power plug allowing all international power cords to be used.

The MF-1C comes pre-connectorized with 50 pin D sub connectors, providing one input and two outputs per amplifier position.

Optionally, the MF-1 or MF-1C may be ordered with a +48 volt phantom power supply. This supply is mounted in the rear of the frame, near the toroidal power transformer, and provides phantom microphone power for use with the MP-1 preamplifiers. Modules may *not* be mixed indiscriminately in a frame that is wired for +48 volt microphone power. Indeed, the frame can be setup such that a mix of modules may be run in one frame. However, the presence of +48 volts at what would normally be an output pin, necessitates care being exercised in the module placement. The + 48 volt buss plugs onto the backplane at each connector. Thus, the module positions may be powered on or off individually, by simply opening the chassis and plugging or unplugging module positions.

In the basic MF-1, audio interconnections to the back plane are left to the end user with the flexibility of using rear panel connectors of your choice. The 0.025" square posts are provided on the backplane with three ports to each amplifier module position. Pre-connectorized pigtail wire assemblies are available to aid in the interconnection of the MF-1. These assemblies consist of a three position Molex® SL housing and pins attached to a length of #22 shielded pair, or #24 Mogami ultra flexible wire, ready for your multi-pin connector at the rear of the chassis.

The MF-1 is finished with a high quality black paint. The modules are held in place by the anti-vibration nylon card guides and the edge card connector, forming a very secure arrangement suitable for mobile environments.

1.3 Modules

The plug-in modules have attached front panels that correspond to the module itself, and provide a contiguous, attractive front view of the frame. The finish on the panels is the same black paint with white silk-screen graphics. Each module has a small extruded aluminum handle that has a bright anodize finish, and provides an ideal location for attaching labels as to the use of the module. Blank "modules" are available for unused amplifier positions. An extender board is also available.

1.3.1 The PS-2

The PS-2 is the power supply module for the MicroFrame Series. It provides a linearly regulated bipolar (±) 15 volts, at a maximum of 1.5 amps, to power the system. Linear regulators are used to ensure ultra low noise amplifier operation. Two power supply modules are needed for each MF-1. The module has combining diodes at the output to the backplane allowing redundant regulator operation. The MF-1 uses a common power transformer and power entry module to feed the two PS-2 power supply modules. Individual voltage trim potentiometers are provided to allow for load balancing between the two supply modules. The power supply regulator modules have transformer secondary fuses to isolate them from the transformer, in the highly unlikely event of a catastrophic failure in one of the modules.

Operation of the MicroFrame Series should be done with forced air cooling, as 80 watts of power may be dissipated by the system. Operation without forced air is permitted for short time periods, but regular operation in this manner is not recommended.

1.3.2 The LA-1

The LA-1 is one of a series of very high performance audio amplifier and processing modules. It is characterized as a one in, two out line/distribution amplifier with front panel variable gain from + 20 to - 20 dB. Unity gain has a center detent position on the potentiometer. The modules of the MicroFrame Series may all share the same chassis without degradation in performance.

1.4 The MP-1 Preamplifier Module

1.4.1 General

The MP-1 was designed to achieve the very highest performance available with microphones and other signal low level signal amplification. It is also characterized as a one in, two out mic-pre / distribution amplifier with front panel variable gain from 0 to +75 dB. The MP-1 is ideal for anyone who wants to maximize performance from analog or digital recorders, live broadcasts, and sound reinforcement, using the superior microphone technologies available today. With MP-1s at or close to their respective microphones, additional performance benefits are realized as opposed to long mic cable runs to a console. By immediately raising the microphone signal amplitude to line level, minimum interference is allowed to enter the system from power lines, SCR stage lighting, etc., longer lines may be driven and *the* "state of the art" in performance is provided for existing consoles.

In addition to the obvious use as a microphone preamplifier, the card may be used wherever amplification is needed and minimum noise is desired. One application which is finding increased acceptance is to bring up low telco levels without adversely affecting the noise floor. With the overall amplification range of 0 to +75 dB and input clip point equal to the output clip point minus the amplification, the MP-1 becomes a universal gain block with performance that is nearly perfect. In this manual we reference all voltage amplitudes to 0 dBu. 0 dBu is a voltage reference of 0.7746 volts, the voltage that is found in a 600 ohm power matched system operating at 0 dBm.

1.4.2 Features

The MP-1 is one of a series of very high performance audio amplifier and processing modules, a product line known as the MicroFrame Series. It is characterized as a one in, two out mic-preamp - distribution amplifier. The preamplifier features:

- Front panel variable gain from 0 to +75 dB
- A 1 dB noise figure
- 0.00059% THD @ 2 kHz, A = 40 dB, 22 kHz measurement filter
- A 200 kHz bandwidth
- +48 V phantom power capability
- A 20 dB switchable attenuator
- 70 Hz low cut filter (switchable)
- 300 mA output current capability

Additional features include a common mode filter that ensures very high rejection to common mode RF energy. Common mode rejection of the instrumentation stage is approximately 100 dB out to 3 kHz and 75 dB at 20 kHz.

The balanced output section utilizes a current boost integrated circuit that, in and of itself, is almost totally free from crossover distortion. When placed in the feedback loop of an NE5532, it provides all of the very desirable qualities of the op-amp but with 300 mA of output current capability. This is essential for driving long capacitive transmission lines. This high output current capability allows 150 Ω power matched transmission lines to be driven to + 26 dBm, or higher, at very low THD. Each MP-1 has either two 60 ohm outputs (unless the module was ordered specifically with an output impedance of 150 Ω), or one output and a phantom power input. The output stage gain is +6 dB. Input and output clip points, at unity gain, are +27 dBu at minimum gain, with pad inserted. The 200 kHz bandwidth provides excellent transient and square wave response without overshoot or ringing, with low phase shift at 20 kHz (≤ 8°), while still providing outstanding RF immunity. On board fuses assure the continued operation of the system in the highly unlikely event of a catastrophic failure on an amplifier module.

A bi-colored signal indicating LED provides a green indication when a signal of a pre-determined threshold is present, usually the house reference level or slightly below, and a flashing red/green indication when the module approaches overload at the output. The threshold for the peak indication is factory set to trip at +20 dBu, but may be set at any level from ≈ +16 to ≈ +26 dBu. The signal presence threshold is adjustable from - 10 dBu to +8 dBu, average, (-2 to +16 dBu peak) and is factory set at 0 dBu (+8 dBu peak).

2.0 UNPACKING

Care has been taken in packing the MP-1 modules to assure that they will withstand normal shipping conditions. Examine the equipment carefully as it is unpacked. If the shipping carton appears to have been damaged and there are signs of physical damage, check the equipment and immediately notify the carrier and Benchmark.

3.0 INSTALLATION

A correct understanding of the proper installation is necessary to achieve the capabilities built into the MP-1. It is important that Benchmark Media Systems, Inc. application note, “A Clean Audio Installation Guide”, be read, digested, and applied as a part of the installation of MP-1.

!!! IMPORTANT !!!

> Operation of the MicroFrame Series should be done with forced air cooling, as up to 80 watts of power may be dissipated by the system. Operation without forced air is permitted for short time periods, but regular operation in this manner is *not* recommended.

3.1 Packaging

Care must be taken at installation of the card frame to prevent stray electromagnetic fields from affecting the noise floor of the preamp. Magnetic leakage from adjacent equipment power transformers can cause problems. Careful arrangement of the equipment rack may be necessary for optimum performance.

3.2 Signal Connections

If you purchased the MF-1 frame without connectors at the rear of the chassis, use the following instructions to fabricate the necessary cabling. If you purchased an MF-1C with 50 pin D sub connectors, follow the connector diagrams that accompany this manual. Additionally, be sure to heed the warning that follows in the output connections section.

3.3 Input Connections

Input to the module is made at the bottom three pin header post on the backplane of the MF-1 labeled “Input”. The input impedance is 20 kΩ (balanced), or 10 kΩ unbalanced.

When feeding the inputs from an unbalanced source, it is well to treat the signal as though it were in fact a fully balanced signal. This should include wiring, patch bays, etc., right up to the output of the unbalanced equipment. At that point, the “low” (black) wire is connected to the “ground” of an RCA type connector, for instance. The shield is not connected to ground at the receiving end, but is connected at the sending end. Additionally, a star ground is used to tie all pieces of equipment together separately from the cable's shield wires. See the sections that deal with Grounding and Forward Referencing in "A Clean Audio Installation Guide". Wiring in this manner will reduce the common mode differences between pieces to a minimum and ensure that the excellent common mode rejection of the MP-1's input amplifier is able to reject residual power related differences between it and the sending equipment. The only difference between this and a balanced output is that the "low" input side is tied to the "low" output of the balanced output.

Modules may be removed and inserted into the MF-1 while the frame is powered. When pulling a module while the frame is powered, there is usually no audible disturbance in the outputs of the other modules. However, when inserting a module into a hot frame, the inrush currents that charge the power supply filter caps may produce a small "tick", much like a scratch on a record, that is generally not considered objectionable, and often not detectable in the presence of an audio signal.

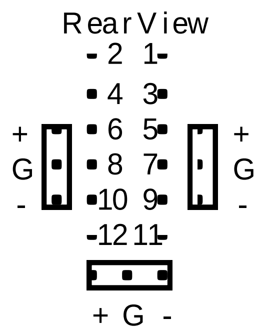

FIGURE 3.1, BASIC CONNECTOR PINOUT

3.4 Output Connections

If the MP-1s were purchased with phantom microphone power capability, only output 1 is available for use. The top pin of what would normally be output 2 is being used to bring + 48 volts into the module. If it was purchased without phantom power, both outputs 1 and 2 are active.

As a distribution amplifier, the module has two sets of build out resistors that provide the two balanced outputs. When viewed from the back of the modular frame, the outputs are to the left and the right of the edge card connector, with the exception of module position one, (the extreme rightmost position when viewed from the rear of the chassis). It has one of its outputs in the center of the edge card connector.

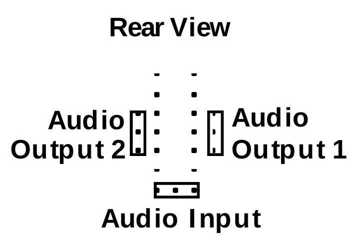

FIGURE 3.2, MP-1 SIGNAL CONNECTIONS

Feeding balanced inputs is done as it is normally. However, when feeding an unbalanced input from the MP-1, only <u>one</u> output leg and ground must be used. <u>Do</u> <u>not</u> use the second output pin. Care must be taken not to short one of the amplifier's outputs to ground. An output line shorted to ground will draw very high currents and produce overheating of the output amplifier. This will not cause catastrophic destruction of the output, since current limiting is provided within the stage, but it will produce severe distortion at the chosen output.

!!! WARNING !!!

> Do not ground one side of a balanced output when feeding an unbalanced input. This active output does not have a transformer, nor is it of the quasi-transformer type active outputs. Grounding an output will cause severe distortion and overload of the amplifier module.

Three position Molex SL connectors should be used to mate with the 0.025" square posts on the backplane for all signal inputs and outputs. Alternately, the connections may be made by wire wrapping to the 0.025" square posts.

3.5 Connector Assembly

In the assembly of connectors, be sure that the drain wire of the shielded pair is physically located as the center pin of the three pin housing. Benchmark currently sells Molex® SL terminals and housings, as well as the pre-connectorized cables.

These assemblies are made up as three pin connectors. The three pin audio signal connectors, with this wire arrangement, have the advantage of being able to be physically inverted, causing a polarity inversion of the signal. This, of course, is accomplished because ground (shield) has been specifically placed at the center pin of the assembly.

The following are part numbers for the recommended Molex connector parts.

2 pin housing	50-57-9002
3 pin housing	50-57-9003
Individual pins	16-02-0102
Crimp tool	11-01-0118

Follow the directions that came with the crimp tool you purchased for the specifics of the connector pins to be used.

3.6 Phantom Power Connections

If you have purchased the MF-1 with phantom power installed, the wiring is already in place. If you purchased an MF-1 without phantom and are now installing it as a retrofit, the following will provide the information needed. +48 volt phantom power is brought to the modules via pin 6 on the card edge connector. A DC to DC inverter is used to generate the +48 volts. It receives its power from the ± 15 volt supplies. The output of this inverter in turn feeds the module's power input pin.

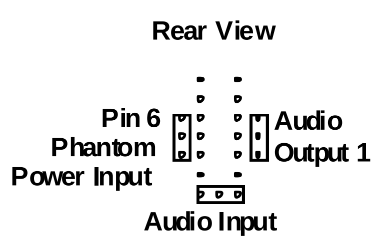

FIGURE 3.3, MP-1 PHANTOM POWER CONNECTION

3.7 Setting Levels

It is very important that you carefully set up the levels in your system. Set them so that all of the gains allow each of the various pieces of equipment and indeed, every stage within every piece of equipment, to reach their clip points at the same time. This is done by taking almost all the gain needed from the amplifier stage that has the lowest noise figure, the MP-1. This will maximize the system's dynamic range. As simple as it sounds, this is the key to an outstanding system setup, assuming the rest of the installation is correct and the equipment is of high quality.

4.0 OPERATION

Once the system has been installed and all levels properly set, the only place gain should be adjusted is at the MP-1. An exception may be where you would like to “fade to off” between various recording segments, since the MP-1 will not go to a “full off” condition. In such a case, the gain control of the recorder can be used to “fade to off” and then returned to its *original* preset position at the beginning of the next recording. Actual level adjustments must be made, however, at the mic-preamp to ensure optimum dynamic range.

4.1 Controls

A number of user controls are on the MP-1 module. See figure 4.1. The controls are only accessible by removing the module and then making the changes desired. These controls include: a 20 dB attenuator, S2201, the +48 volt phantom power switch, S2202, the 70 Hz low cut filter, S1201, the meter jumper W2201, the Maximum Gain Limit Trim, R1109, meter level adjustments, - O/L Threshold, R2113, and Signal Presence Threshold. R2203. The common mode trims are not generally considered user adjustments, as they require a lab test setup for adjustment. If you do not have the proper signal generator and measurement equipment, *do not* change the settings of these two controls. A procedure is included further on in the manual for the setup of the common mode controls.

4.1.1 20 dB Attenuator

The 20 dB attenuator should be used whenever very high input signal levels cause the metering LED to flash Red/Green, assuming the main level control has been set to a minimum. At minimum gain and with the attenuator “In”, the module can receive an input level of +27 dBu (line level) before clip, since the overall gain is unity.

4.1.2 +48 Volt Phantom

+ 48 volt microphone power is available only if the MF-1 card frame was purchased with the optional power supply. Additionally, if the module itself was purchased without phantom power components in favor of a second output, it will not be available to power a microphone. Microphone power should be turned on 1/2 to 1 hour ahead-of-time, to allow the formation of the dielectric in the coupling capacitors.

4.1.3 70 Hz Low Cut Filter

The low cut filter is included to help overcome the noise problems associated with room rumble and wind in a microphone. The filter is out of circuit when the switch is in the up position and in circuit when in the down position.

4.1.4 LED “Meter” Jumper - W2201

The meter jumper (W2201) arrangement is designed for three metering possibilities. These include: no metering, continuous pre-filter metering, and metering that follows the filter switch.

To eliminate metering, simply store the jumper, with one leg on the post and the second off to the right, on the right-hand post (pin 3). For continuous pre-filter metering, pins 2 and 3 should be shorted by the jumper. For metering that follows the filter switch, pins 1 and 2 should be shorted. The factory setup is for continuous pre-filter metering, pins 2 and 3 shorted. Since large amounts of room noise and wind can significantly reduce the headroom of the preamplifier, the approach of clipping should be monitored by the flashing red LED.

4.1.5 Maximum Gain Limit Trim
This control allows the maximum gain obtainable from the front panel main gain control, to be limited to whatever the engineering department deems desirable. By reducing the gain range, the "set-ability" at the main control is improved.

4.1.6 O/L Threshold Adjustment
The "overload" threshold adjustment is factory adjusted to turn on at + 20 dBu. This allows 7 dB additional increase in level before clip is actually reached. This may, however, be changed according to the preferences of the engineer. A range of ≈ + 16 dBu to ≈ + 26 dBu is available at this control.

4.1.7 Signal Threshold Adjustment
This control sets the trip point for the detection of signal presence. The signal presence threshold is adjustable from - 10 dBu to +8 dBu, average, (-2 to +16 dBu peak) and is factory set at 0 dBu average (+8 dBu peak).

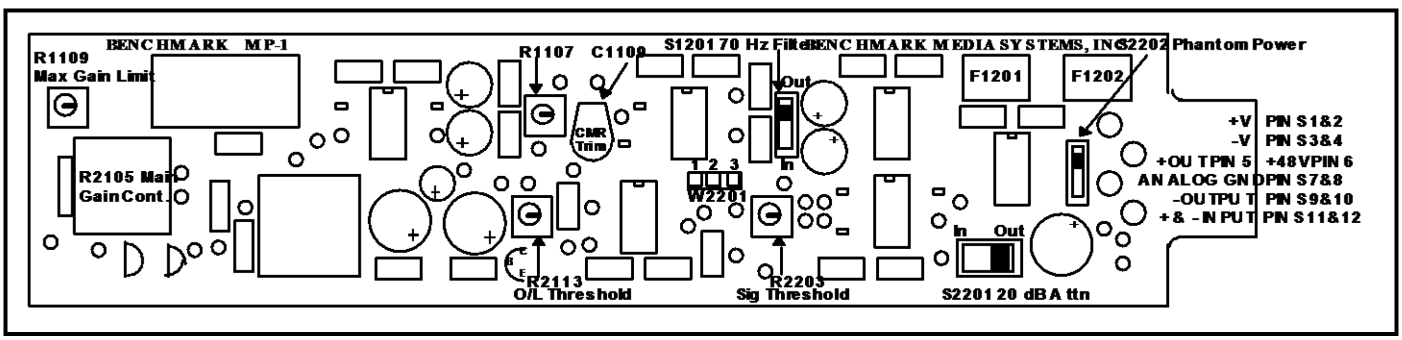

FIGURE 4.1, MP-1 MODULE

4.2 Noise Primer
Noise figure is a measure of how well an amplifier raises the intended signal without adding noise. In the case of the MP-1, the amplifier adds only one dB of noise to that of the original signal, for amplification factors greater than 40 to 45 dB. The noise figure is referenced to the Johnson noise of the resistive portion of a transducer's source impedance.

Johnson noise may be calculated from:

$$E_n = 4\sqrt{kTRB} \qquad [1.0]$$

Where:

k = Boltzman's Constant = 1.38×10^{-23}
T = temperature of resistance in degrees Kelvin (referenced to absolute zero)
(room temperature ≈ 300 Kelvin)
R = resistance = microphone source impedance
B = bandwidth = 19,980 Hz (20 Hz - 20 kHz)

From the above formula, we see that the noise of a 150 ohm resistor at room temperature is 222.9 nanovolts or -130.82 dBu, whereas, a 200 ohm resistor has a noise voltage of -129.57 dBu. Since the source impedance of a microphone is usually between 150 and 200 Ω, this is the noise floor limit.

4.2.1 Amplification of Noise

Any amplifier, while amplifying the desired signal from a microphone, will also amplify the Johnson noise from the source resistance. Therefore, the output noise of a totally noiseless amplifier, operating @ 50 dB of amplification from a source resistance of 150 ohms at room temperature, would be -80.82 dBu. The MP-1's performance under these conditions is approximately -80 dBu. At its minimum amplification (20 dB), the MP-1 has an output noise floor of -93 dBu. The noise increases slowly as the amplification is increased to 40 dB, where the output noise is approximately -88 dBu. From this point on, the noise will increase directly with the increase in amplification. Now that we have the tools, let's see how this understanding of noise applies to two specific microphones. Taking into consideration the source resistance, self noise in the case of a condenser microphone with its own active amplification, and microphone sensitivity, we can then evaluate the performance of any microphone - MP-1 system under various sound pressure level conditions.

4.2.2 Level Calculation Facts

10 dynes per cm^2 = 1 Pascal = 94 dB SPL. This is the standard sound pressure measurement reference level for microphones.

+ 4 dBu = standard operating line level for most radio and recording studios.

Preamp minimum gain = 20 dB

Preamp maximum gain = 73 dB

Preamp peak clip = +27 dBu (±15 volt power supplies)

Preamp noise floor at (A = 20 dB) = -93 dBu

Preamp noise figure @ A $\geq$ 45 dB = 1 dB

4.2.3 RE-20

The sensitivity of an Electro-Voice RE-20 is 1.09 mV (-57 dBu) at 94 dB SPL input. If we use a voltage amplification of 60 dB, the noise output from the MP-1 will be approximately -70 dBu (-130 equivalent input noise of the preamp + 60 dB of gain). The output amplitude from the preamp at the measurement reference level of 94 dB SPL will be (-57 + 60) = +3 dBu. At just one more dB, 95 dB SPL, we have the sound pressure level necessary to give us an output of +4 dBu, the nominal line signal voltage reference.

The average signal-to-noise ratio at 95 dB SPL will be 74 dB, (+4 ref. - [-70] dBu noise floor). The peak SPL that the system can handle is +118 dB SPL, since the maximum output amplitude is +27 dBu (+27 peak clip -[+4 ref.] = 23 dB level increase possible) 95 dB SPL + 23 dB increase = 118 dB SPL max. At this amplification and SPL, the preamp will reach its output clip point and yield an overall dynamic range of 97 dB.

As you can see, these figures are all derived from simple dB additions and subtractions.

4.2.4 MKH-40

Let's examine the performance of the MP-1 with the high performance Sennheiser MKH-40. This microphone has a very high sensitivity of 25 mV (-29.8 dBu) at one Pascal, and very low self noise of 12 dBa (a condenser microphone with internal electronics). The self noise, therefore, is 1.99 microvolts, “A” weighted. This translates to -111.82 dBu. If we use 20 dB of amplification at the MP-1, the combination output noise is ≈ -90 dBu (20 kHz noise bandwidth). At the reference SPL of 94 dB, the output of the microphone is -29.82 dBu. Add 20 dB of amplification from the MP-1 and the preamp output amplitude is -9.8 dBu @ 94 dB SPL. Therefore, to get +27 dBu out (amplifier peak clip), the input SPL must be:

+27 dBu -(-9.82 dBu) = 36.8 dB additional dB SPL to be added to the reference (94 dB) SPL with the resulting 130.8 dB input SPL.

Thus, we have an average signal to noise ratio of 94 dB (+4 -[-90]) and a peak signal to noise ratio (dynamic range) of 117 dB (+27 -[-90]) at the peak acoustic input of 130.8 dB SPL. This is still 3 dB below the 134 dB SPL / 0.5% THD point of the microphone. 107.8 dB SPL average is required at this amplification for an output of +4 dBu ({+4 -[-9.4]} + 94). This dynamic range is beyond the best commercially available digital performance to date. The *average* signal to noise ratio is approximately what is available from the full dynamic range of the 16 bit process.

4.2.5 Problem

Let’s assume that you desire to feed a digital recorder that has a dynamic range (peak signal-to-noise ratio) of 95 dB and input clip point of +21 dBu into an unbalanced input. Indeed, you plan to feed the recorder unbalanced, with the mic pre physically near the recorder. What is the lowest peak sound pressure level that the Sennheiser MKH-40 microphone can receive and still maintain the full dynamic range of the recorder? Also, what amplification is necessary from the MP-1 to achieve this?

4.2.6 Solution

First, since the output of the MP-1 will be used unbalanced, and the recorder input clip point is +21, then ±15 volt supplies of the PS-2 are quite sufficient to power the MP-1, since 15 volt supplies yield a balanced output clip point of +27 dBu and an unbalanced (one output side and ground) clip point of +21 dBu. Using the MP-1 as an unbalanced output device, we give up 6 dB of amplification range. Therefore, in this application, the card has an overall amplification range of -6 to +67 dB. Since the dynamic range of the recorder is 95 dB, then the recorder noise floor is +21 dBu - 95 dB = -74 dBu. If the microphone self noise is -111.82, then the maximum amplification that we can use is (recorder noise floor - self noise) - 74 -(-111.82) = 37.82 dB. Actually, if we make both the recorder and the MP-1 have the same noise voltages, they will add and we will experience a 3 dB loss in dynamic range. Hence, it is well to keep the MP-1’s output noise voltage 3 dB lower than the noise voltage of the recorder and adjust our output level by moving our microphone closer to the music source. Therefore, let's select a maximum amplification factor of A = 34 dB. With 34 dB of amplification, the peak input voltage that the MP-1 can receive is +21 dBu - 34 dB = -13 dBu. Once again, from the microphone sensitivity figure given by the manufacturer, we find that @ 94 dB SPL into the mic, we have an output voltage of 25 mV = -29.82 dBu. Therefore, we can have a 16.82 dB higher SPL than this reference where we reach the clip point. Preamp input clip - sensitivity = additional SPL over reference, -13 -(-29.82) = 16.82. That is 94 + 16.82 = 110.82 dB SPL, the SPL clip point.

If we assume that this hypothetical recorder is a "semi-pro" device and that a nominal input level of -10 dBV (≈ -8 dBu) is what will give a 0 indication on the device's meter, then the average SPL (we're assuming an average type meter when in fact most R-DAT meters are peak type) that we should have at the microphone is 110.82 dB SPL - 31 dB = 79 dB SPL. Actually 31 dB is probably an excessive amount of headroom for most recordings. 20 dB is more realistic, but you must verify that this is sufficient for yourself.

While the above calculations are hypothetical, and do not take into account the ambient acoustic noise of our recording environment, they are close to the real world. These calculations demonstrate how the microphone and the preamp set the dynamic range of the system. This also demonstrates the need, in this digital recording era, to use recording environments with extremely low acoustic ambient noise levels, if we expect to realize the recorder's full dynamic range.

The amplification required for most microphones will typically be 40 dB or greater, and the preamp section will most often be the limiting factor in the output noise of the electronics, prior to any recording or transmission medium.

5.0 SPECIFICATIONS

Input Section:

Type -	Active instrumentation type input with ultra low noise transistors and a common mode filter.
Attenuation -	20 dB passive attenuator, switch selectable
Microphone Pwr -	+48 V Phantom (Optional) switch selectable
Com Mode Filter -	Differential mode bandwidth - >500 kHz, Common mode bandwidth - 26 kHz 2 pole LC filter > 60 dB attenuation @ 1 MHz
Input Impedance -	No attenuation: 8 kΩ balanced with phantom pwr network, 20 kΩ without phantom network. With attenuation: 1.5 kΩ.
C M Rejection -	100 dB @ 1 kHz typ, 80 dB min. 75 dB @ 20 kHz typ., 60 dB min., 20 Hz - 20 kHz.

Output Section:

Type -	Active balanced ground referenced output with current boost
Output Impedance -	60 Ω: 150 or 600 Ω (special order)
Output Current -	300 mA peak per output leg
Quantity -	Two (2) outputs per module or one (1) output and phantom microphone power.
Differential Phase -	0.5° @ 20 kHz max., 0.35° typ.

Overall:

Gain Range -	0 to +73 dB (including attenuator)
THD @ A=40 dB -	0.0009% typical 2 kHz (Measurement filters = 22 Hz and 22 kHz)
20 kHz Ø shift -	- 8° typical
Bandwidth -	200 kHz min.
Unity Gain Noise -	-93 dBu typical
Noise @ A = 50 dB -	- 80 dBu typical
Supply Current -	60 mA Quiescent max., 60 mA typical, 70 mA max. with LEDs flashing, 70 mA typical
Slew Rate:	8 V/μ Sec

Metering:

Type -	Bi-color LED.
Signal Presence -	Green, threshold range = -2 to +16 dBu tone setup, -10 to +8 dBu with program material.
Peak -	Flashing Red/Green, threshold Range = +16 to +26 dBu tone setup.

5.1 Noise Performance Evaluation

5.1.1 Source Impedance

Specific procedures are necessary for the proper evaluation of noise performance from a microphone preamplifier. To obtain the correct noise level measurements, two specific criteria must be met:

[1.] A 150 source impedance

[2.] A 20 kHz measurement bandwidth.

The preamplifier must have the proper source impedance at its input. If the input of the preamplifier does not "see" the normal source impedance, it will amplify the noise of the parallel combination of the internal 10 kΩ bias resistors and the 6.81 kΩ phantom power resistors. An input termination can be made up with an XLR type connector and the appropriate resistor. When making an input termination, be sure to use a carbon film or a metal film resistor, not a carbon composition resistor. Carbon composition resistors have a phenomenon known as "excess noise", and this will yield a noise figure that is misleading.

Alternately, and in most cases the best choice, you may engage the 20 dB attenuator, since that presents a 150Ω source impedance to the input stage of the preamplifier. The noise specifications of the MP-1 are made referenced to a 150Ω source and thus, you will have the correct specifications. The one case where an input termination is preferable to the internal pad is when the noise pickup and microphonics of the microphone transmission line is to be evaluated.

5.1.2 Noise Bandwidth

The second criterion that must be met for correct noise evaluation is the limitation of the measurement instrument's bandwidth to 20 kHz. All the Benchmark noise specifications are 20 kHz bandwidth measurements. If the test instrument that you are using does not have an internal filter for this purpose, you will need to construct a device that will give the same results as a 20 kHz "brick wall" filter. We have available an article by Deane Jensen, entitled "A 20 kHz Low Pass Filter for Audio Noise Measurements." If you should need to construct such a filter, this would be a good circuit to follow. If you need a copy, please call the sales department and request it.

Remember, when measuring noise, the output noise (for gains of ≈45 dB and higher) of the preamplifier will be the 20 kHz bandwidth Johnson noise of the source impedance (-130.8 dBu for a 150Ω source over 20 kHz), plus the amplification of the preamplifier, plus the noise figure of the preamplifier. Therefore, the expected noise floor of a preamp operating @ 50 dB of amplification, would be -130.8 dBu, +50 dB, plus the noise figure of the preamp, which in this case is ≤1 dB, or Total noise = ≈ -80 dBu. The only way to achieve lower noise performance is to operate both the microphone (source impedance) and the amplifier at much lower temperatures. Since we are talking temperatures in degrees Kelvin, the implication is a move toward cryogenic temperatures.

5.1.3 Potentiometer Noise

Occasionally, we receive reports from customers that gain control potentiometers are noisy. On a rare occasion this may indeed be the case, but most often it is not. Although the input transistors are matched for V_{be}, a very small difference usually remains. As a result, rapid gain changes will often cause a momentary DC voltage across the potentiometer as the large coupling capacitor in the emitter circuit is changing charge. The same result will occur with any potentiometer. A part of the "problem" is the very wide gain range that is designed into the MP-1.

If the potentiometer is varied slowly, as you would in a live recording session, most often there will be no resulting noise. Only when the potentiometer is varied *slowly* and noise is still heard, does the gain potentiometer indeed have a problem.

The magnitude of this "problem" is often exaggerated by listening to the output of an MP-1 with a headphone or speaker amplifier at full amplification, where a total of 100 dB or more of amplification is possible. At this degree of amplification, it is usually quite easy to hear microphonics in microphone cable, depending upon the quality of the cable.

Limiting the gain range of the front panel potentiometer to less than the maximum of +73 dB improves the "setability" of the potentiometer. This also reduces the "apparent noise of the potentiometer" under rapid gain changes. The 100Ω trim potentiometer (in series with the 2.7Ω end stop limit resistor, the front panel pot and the capacitor network), allows you to set the maximum amplification obtainable at the front panel pot to just above what will be regularly needed.

5.1.4 Front Panel Pot Gain Range

Unless otherwise specified, the front panel potentiometer's fully clockwise max. gain is set to +60 dB. You may adjust the front panel pot's fully clockwise "max. gain", either up or down from this 60 dB point to the +73 to 75 dB maximum allowable, or to as low as +43 dB on the minimum side, via the 100Ω trimmer. A max. gain setting of +50 dB will provide the most linear action from the potentiometer, and if you are using high output condenser microphones and do not need more amplification, this may be the ideal setting. Additionally, if 73 dB is not enough gain, the 2.7Ω resistor may be bypassed or a lower value of resistance can be substituted. By totally eliminating the resistor (with a short), the maximum gain can be as high as 93 dB. Be advised that the preamplifier may, or may not, be stable at that gain. If the unit is unstable, additional feedback compensation capacitance may be needed around the first stage operational amplifiers. 47 pF is the present value of compensation capacitance; 56, 68 or even 82 pF of capacitance at both locations may be necessary to stabilize the amplifier.

5.1.5 Additional Points to Note

Whenever phantom power is to be used, it is well to turn it on one half to one hour ahead of time. This allows the full formation of the coupling capacitor's dielectric, which reduces a small noise source that would be generated by leakage currents.

The amplifier, as noted above, may be used for boosting marginal line amplitude signals with minimum output noise. For example, when operated at 30 dB of amplification, the noise output of the card is still -92 dBu (150 ohm source).

Since the output impedance of the card is 60 ohms, the length of line that may be driven is three to four times longer than can be driven with a 150 to 200 ohm microphone drive impedance for the same small signal high frequency cutoff point. For example, almost

2800' of foil shielded cable may be driven with a *small signal* high frequency cutoff of 30 kHz. When long lines are to be driven 1000' or more, we recommend the use of very low capacitance cable such as the Mogami 2574. It has a capacitance of 6 to 7 pF/ft between conductors, versus the 30 to 32 pF/ft to be found in most foil shielded cables, and thus will preserve the high frequency slew capability of the system. Mogami cable is normally available from stock at Benchmark Media Systems, Inc. See the "Clean Audio Installation Guide" for more information on slew rate limiting due to cable capacitance.

5.1.6 Conclusion

The amplification required for most microphones will typically be 35 dB or greater, and the preamp section will most often be the limiting factor in the output noise of the electronics, prior to any recording or transmission medium. Therefore, the majority of amplification needed, consistent with desired headroom, should be taken from the MP-1, since it has the lowest noise figure of any of the amplifying stages.

6.0 CIRCUIT DESCRIPTION

6.1 General

The superior performance of the MP-1 is a result of careful attention to detail in every element of the circuit design. For example, the common mode rejection circuitry of the MP-1 has additional performance built into it, over what is normally found in microphone preamplifiers. With the addition of a common mode filter, all interfering signals see a 26 kHz LC low pass filter, and are down 60 dB @ 1 MHz, while the desired differential signal is unaffected. The performance of the preamplifier section is much superior to preamps found in most audio consoles, with THD @ A=50 dB, 20 Hz - 20 kHz = 0.002%. The differential high frequency cutoff is 200 kHz at any amplification, thus insuring low phase shift @ 20 kHz and superb transient performance. The EIN of the card is approximately -130 dBu, when operated at a minimum A=40 to 43 dB. As we have seen, this is a noise figure of less than one dB.

6.2 Input Stage

Following the signal flow through the preamplifier, the input signal first encounters the phantom power circuit, which consists of a pair of 6.81K ohm resistors that feed power to the microphone line. This power is turned on or off by the small vertical slide switch near the card edge connector. Next is the 20 dB pad which consists of two 750 ohm resistors plus one 150 ohm resistor, and is activated by the somewhat larger horizontal slide switch a little farther forward on the module. Next in line are the input coupling capacitors. These capacitors are aluminum electrolytics, rather than tantalum, for their superior dielectric absorption characteristic. Dielectric absorption is the distortion producing mechanism in capacitors. Additionally, all aluminum electrolytics are bypassed with film capacitors for superior high frequency performance. The next section is the common mode choke. This consists of a highly symmetrical dual winding on a common toroidal core, whose inductance on each winding is 38 millihenries. The operation of the choke is such that with a differential signal, the magnetic fields created by the two highly symmetrical, but opposite, windings will cancel one another, and the net induction is zero. However, with a common mode signal, the inductance remains, and in conjunction with the 1000 pF capacitors and 10K ohm termination resistors, these elements form a two pole Butterworth low pass filter. The filter has a 26 kHz corner frequency and a 12 dB/octave roll off. Its amplitude response is down 60 dB at 1 MHz, providing excellent RF rejection.

We next encounter the ultra low noise input transistors. These transistors are protected by small signal diodes across the emitter-base junction. It is important to prevent these transistors from ever going into their emitter-base reverse biased zener mode. If this should happen, the low noise capability of the transistors is destroyed and the devices are then more prone to failure.

The transistors are a part of a low noise active differential amplifier. This topology is an emitter feedback differential amplifier. The noise figure of the transistors is less than 1 dB referenced to 150 ohms and is still less than 2.5 dB referenced to 10 ohms. These transistors were developed for use as moving coil head preamplifiers, and their noise performance is maintained by the absence of series input resistors. As a result of coupling the input signal directly to the bases of the transistors, the only noise producer in the signal path is the intrinsic base resistance of the transistors. This has been reduced to a very low amount by their design. Since the common mode filter has no appreciable series resistance, it also does not add noise to the circuit. The source resistance of the transducer becomes the major noise producer. The amplification of this stage is established by the ratio of the resistance between the emitters to the feedback resistance coming to the emitters from the outputs of the op-amps. The values selected allow for an amplification range of 8 to 61 dB at the differential pair. The values also set the bias currents for the stage in conjunction with the fixed voltage bias point at the non-inverting input of the op-amps.

6.3 Differential Converter

The headroom of the input stage has been reduced by the bias configuration 6 dB from what would normally be available to a stage with a ± 15 volt supply. Therefore, the differential input converter has +6 dB of amplification to equalize its clip point with the input stage. The output stage has additional amplification of 6 dB from the inverting follower, yielding an amplification range of +20 to +73 dB for the electronics. The 20 dB pad gives an overall range of 0 to +73 dB.

The differential converter also converts the differential signal from the first stage to a single-ended signal. This stage rejects any power supply noise and low frequency common mode signals that may be present. The common mode rejection is adjusted by using both the resistive trim and the capacitive trim and is made with an input frequency of 2 kHz. Typically, the null that can be achieved is -105 dBu.

6.3.1 Common Mode Rejection Adjustment.

Common mode rejection adjustment must be performed using an oscillator with a "single ended" output (ground referenced). A function generator will work very well. Additionally, you will need a sensitive, wideband analog, AC type voltmeter. The voltmeters found in distortion analyzers with sensitivities down to -100 dBu or better are good. A digital voltmeter is *not* a proper choice for this measurement. Most preferable, however, is a logarithmic type voltmeter, such as those formerly made by **dbx** and Valley People. Unfortunately these are no longer available, except perhaps on the used market. Additionally an oscilloscope is very advantageous in this test setup. The oscilloscope should receive its signal from the output of the *voltmeter*, where the voltmeter is acting as a preamplifier for the scope. The scope should be triggered from the "Sync" output of the function generator.

Connect the test equipment to the Differential Input Amplifier as shown in figure 6.1. Although the output of the MP-1 is balanced, the input of most general purpose AC voltmeters, is unbalanced (single ended). When feeding an unbalanced input, use only one output leg and ground. Do *not* ground the second output leg.

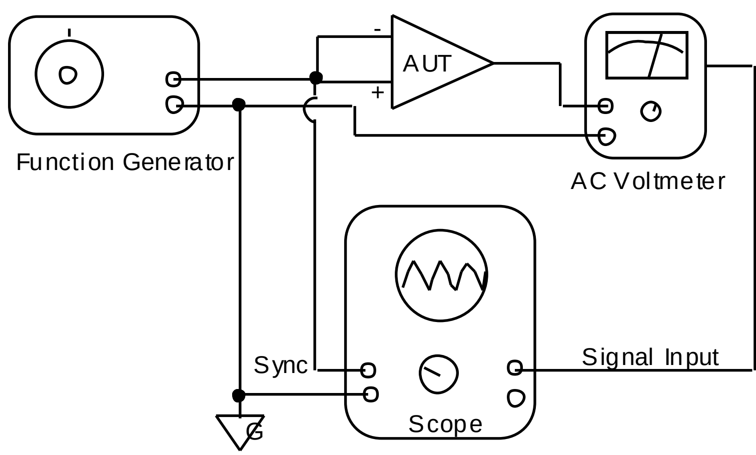

FIG. 6.1, COMMON MODE REJECTION TEST SETUP

The common mode rejection adjustment is actually a series of iterative adjustments between the resistive portion of the bridge and the capacitive portion. This adjustment should be performed at a frequency of 2 kHz.

Adjust the trim resistor for the best null possible at the output of the MP-1, while observing the voltmeter. If you have an oscilloscope as a part of your test setup, observe the phase shift of the remnant signal as you pass through a voltage minimum. The phase change is a more easily observable parameter when near the voltage minimum than is the voltage itself. Set the potentiometer for the center of the phase shift, and then proceed to the capacitive adjustment and adjust for a further reduction in output voltage. You will most likely need to switch the voltmeter to progressively lower ranges during this process. You will find the adjustments progressively more sensitive. Observing the oscilloscope will allow a significantly faster achievement of the desired null.

The common mode adjustment involves passive components that form a bridge around the active operational amplifier, and this adjustment is independent of the operational amplifier. If an amplifier integrated circuit ever needs replacing, the common mode rejection adjustments will *not* need to be performed. The only possible exception to this would be the equivalent of a lightning strike that would take out or change the values of the passive components.

6.4 70 Hz Low Cut Filter

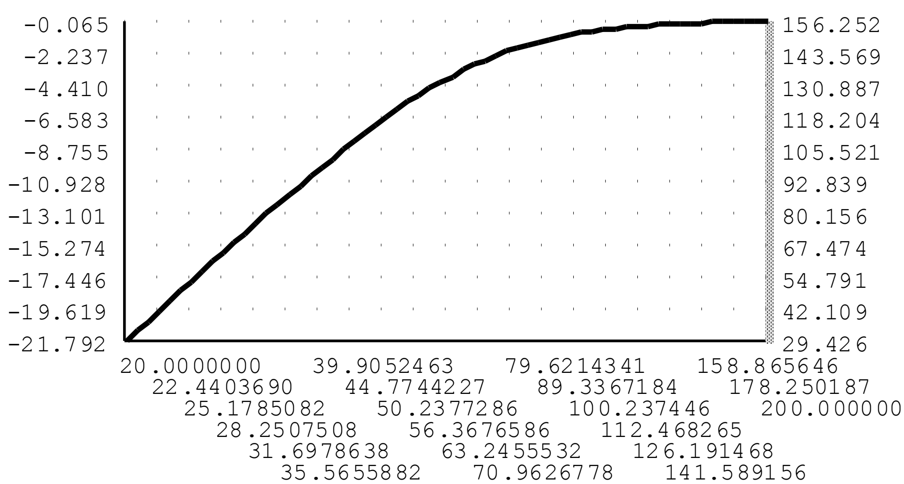

FIG 6.2, FILTER PERFORMANCE 20 - 200 HZ

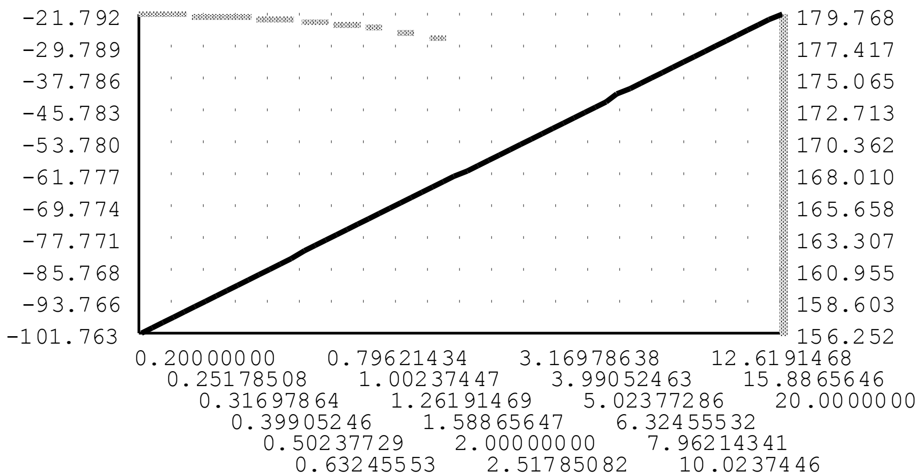

FIG 6.3, FILTER PERFORMANCE 0.2 - 20 HZ

A selectable low cut filter has been included in the module to reduce the effects of room and wind noise. The design of the filter is a standard two pole Sallen and Key type with a cutoff frequency, f_3 = 70 Hz, at a cutoff rate of 12 dB per octave. The filter is down 12.3 dB at 35 Hz, 21.8 dB at 20 Hz and 33.8 dB at 10 Hz. Switch S1201 allows the output stage to select either pre-filter or post-filter signals. In the performance graphs, amplitude is the left Y axis and phase is the right Y axis, where amplitude is the solid black line and phase is the gray line.

6.5 The Signal Indicating LED Circuit

The Signal indicating LED is a bi-colored LED with red and green sections. The green section is used to indicate signal presence, and the red section is used to indicate the approach of a signal clip condition. Both halves of the FET input TLO72 are used as voltage comparators, and both have hysteresis built into their circuits.

The signal presence indicator uses one half of the TLO72 and associated components. The input audio is rectified and the DC voltage is compared with that of a preset (threshold) trimming resistor. This circuit yields a steady green indication when the desired signal threshold has been reached or exceeded.

The operation of the green signal presence LED is as follows. A single 1N4148 small signal diode operates as a half wave rectifier. The turn-on threshold of the diode is precisely at the lowest DC voltage represented by the lowest signal level detection limit. This is serendipitous, in that it permits implementation of a half wave rectification without the high parts count required by active circuits. DC from the diode is stored on the 0.1 μF film capacitor. Charge is held on the 0.1 μF capacitor to prevent a flashing condition with the instantaneous absence of audio, such as between words in dialog. The discharge time to an “off” condition will vary with the peak level of audio, and thus the peak voltage on the capacitor. Typically, however you will see a variation from 1 to 4 seconds to an “off” state. The 10 MΩ resistor provides a discharge path for the 0.1 μF capacitor, as well as providing bias current source for the FET input operational amplifier.

The peak indicator is an oscillating half wave detecting comparator. When the audio input exceeds a predetermined level, the comparator trips and begins to oscillate. The factory calibration point is +20 dBu, unless otherwise requested. The output of the comparator, in addition to driving the red portion of the LED, also drives a transistor that shunts the current from the green LED, causing an alternating red/green flash when the signal exceeds the trip point.

The peak comparator is an oscillating comparator, by virtue of the AC coupled hysteresis applied around the device. Initially, the output voltage of the comparator is near the + supply voltage, in the “off” state. The trip point is determined by the resistor string R2114, R2113, R2112, and diode CR2103. When the inverting input signal rises above ground potential, the comparator trips. R2113 is the calibration trim for the peak indicator. It has a range of approximately +16 to +26 dBu. The comparator is held in the “off” state by the bias that is applied to the inverting input through R2112, until an input peak overcomes this preset bias. When the comparator trips, the output voltage swings to the opposite supply rail. The 0.1 µF capacitor, in turn, pulls the non-inverting input negative holding the comparator in the “on” state. The capacitor recharges with opposite polarity through the two 220 kΩ resistors and when the new threshold is passed, the device turns off and is now held off, again by the charge on the capacitor, until the capacitor recharges to its original state. This action is that of a pulse stretcher, which allows the operator to “see” very short peaks as they occur.

All of the circuitry of the signal indicators is high impedance and has no measurable effect on the audio signal quality.

6.6 Output Stage

The output amplifier stage consists of an NE5532 dual operational amplifier, each half of which drives a current boost stage. The current boost stage is an LM6321 unity gain buffer amplifier that has a 50 MHz bandwidth, a 500 V/μ sec slew rate, and a 300 mA output current capability. This performance provides a very low phase shift in the current boost stage, which in turn protects the phase margin of the op-amp driver once the loop is closed. The buffer itself has an excellent output stage design that is virtually free from crossover distortion, prior to being enclosed within a feedback loop, and has almost perfect performance when placed in a loop. Its high current capability allows two output splits to be taken from the low impedance point of the amplifier. The current boost buffer will run warm to the touch without a load and with a heavy load it will be hot, that is, over 135° F. This is not a problem, as the device will withstand temperatures up to 100° C and above.

The D.C. offset voltage from the output amplifiers of the MP-1 is typically less than 2 millivolts.

This completes the MP-1 circuit description.

7.0 TROUBLESHOOTING TECHNIQUES

Armed with the knowledge of the circuit descriptions given above, standard trouble shooting techniques should be used to determine first, the general area of a malfunction, and then more specifically, the actual offending components. A review of the most basic of these techniques follows.

> 1. It is best to trouble shoot a module at a work bench using current-limited lab power supplies. Set the current limit of the power supplies to 125 mA. This will protect the module and still allow the location of failures to be made. A special connector pigtail interconnect system will have to be constructed, to provide power as well as signal inputs and outputs. Edge card connectors are available from Benchmark.
>
> 2. The highest percentages of failures within systems, without a doubt, are due to component failures with semiconductors. Another source of failure is with electrolytic capacitors whose dielectric may dry out. Improvements in capacitor manufacturing techniques, however, have reduced this problem.
>
> 3. Since most failures are catastrophic rather than a gradual degradation of performance, make a close visual inspection of the module. Look for discoloration of components and possible shorts between vertical leads of resistors, diodes, etc., and shorts on the PC board itself. Discoloration indicates excessive heat, and is most likely associated with component failure. Remove any component that has obviously failed, i.e. carbonized resistors or IC packages that are cracked, etc. A stereo optical magnifier such as the "OptiVISOR" #7, manufactured by Donegan Optical Co., will allow close inspection of the module and rapid identification of physical problems.

4. If fuses are blown, replace them and power up the module. If there are short circuits on the module, the current limiting of the power supplies will prevent further failures. The presence of a short will be shown by the current limiting of the power supplies. If this occurs, allow the supplies to run at their preset current limit. Look for any components that are operating hot to the touch. This will often show up shorts when there are no visual symptoms. Next, trace the current flow using a DC millivolt meter to view small voltage drops across PC board traces that are carrying relatively high currents. This technique will provide the precise location of shorts. Even sub-hairline shorts in bare boards can be easily located with this method. A pair of very sharp needle pointed probes, such as Huntron Microprobes™, are ideal for piercing the solder mask on the PCB. Meaningful voltage drop can be measured over only 1" of PCB trace.

Typically, one can just keep their hand on a surface at 130°-135° F. (≈ 40° C) When in the modular frame, no component should run with a temperature beyond which one can comfortably keep their finger. There is one exception, that of the output buffer amplifier. However, it should be operated on the test bench without a load, and thus at a relatively cool temperature.

5. Remove any components, i.e. transistors or integrated circuits, that are experiencing overheating. Most often at this point, the power supplies will come out of current limiting, and the module will function, at least in part. If further problems exist after the power supplies come out of current limiting, they can most often be found by performing the normal voltage checks throughout the circuitry.

7.1 Circuit Board De-Soldering

Printed circuit boards are <u>very</u> easy to damage by excessive heat. Unless you have developed the specialized skills necessary to remove and replace components, we suggest that you leave the task to someone skilled in these techniques.

When servicing printed circuit boards, we strongly recommend the use of a vacuum de-soldering station. The proper technique with these stations is to apply the de-soldering tip to the area to be unsoldered and wait for the solder to thoroughly melt. You can be sure of a thorough melt by observing the top side of the board or by observing when the component lead moves about freely. <u>Only</u> when the solder there has become liquid, apply the vacuum while moving the hollow tip with the component lead in a circular motion. By rotating the lead, with the tip against the board, but <u>without</u> applying pressure to the pad, you are able to most thoroughly remove solder in the plated-through hole. In turn, the component will often drop out of the board when you are finished. If you push the tip against the board, it will often destroy the bond between the pad and the board, then requiring an eyelet to be inserted in the board before it can be used.

!!! WARNING !!!

If the solder is not thoroughly removed from the plated-through hole, attempting to remove the component will bring with it plating from inside the hole. This may destroy the usefulness of the board, depending on the size of the hole and the ability to replace the plating with an eyelet. If you find that your attempt to completely remove the solder from the hole and pads has failed, do not attempt to re-heat the area with the de-soldering tool, as this will <u>overheat</u> the pad, and not the area that is in need. As a result, the board is usually damaged. Rather, re-solder the joint, and then go back and apply the proper technique, by allowing the solder in the joint to thoroughly melt <u>before</u> applying vacuum. This technique uses new solder as an efficient heat conductor to the total area, eliminating hot spots.

7.2 Circuit Board Re-Soldering

Here is an effective technique that ensures highly reliable hand solder joints. It involves heating the component lead first, since it usually has the higher mass and is often in less danger of being damaged from excessive heat.

First, brush a small amount of liquid flux on the board and component area. This greatly aids in the solder flow through the hole, providing nice fillets on both sides of the board. We much prefer water soluble flux, both in the solder core and liquid, because of its easy clean up, and freedom from the need for (and soon to be banned) chloro-floro hydrocarbon solvents.

Next, apply a small amount of solder to the tip of the soldering iron at the same time as you apply the iron to the component lead. This provides a good solder joint between the lead and the iron. This good junction between the iron and the component lead is necessary for efficient heat transfer.

The iron should be approximately 1/8" above the board. When the lead has come up to temperature so that it melts solder when placed against it and has good wetting, slide the soldering iron down the lead and heat the printed circuit board pad while applying a controlled amount of solder to the joint. All of this should take no more than two or four seconds.

If the component that is to be installed has leads that are oxidized, it will be necessary to pre-clean them. This may be done with either a Scotch-Bright® abrasive pad, a fine bristle fiberglass brush, or even woven flexible wire strap, among other methods.

8.0 WARRANTY

Benchmark Media Systems, Inc. warrants its products to be free from defects in material and workmanship under normal use and service for the period of five years from the date of delivery. This warranty extends only to the original purchaser. This warranty does not apply to fuses, lamps, batteries, or any products or parts which have been subjected to misuse, neglect, accident, or abnormal operating conditions.

In the event of failure of a product under this warranty, Benchmark Media Systems, Inc. will repair, at no charge, the product returned to its factory. Benchmark Media Systems, Inc. may, at its option, replace the product in lieu of repair. If the failure has been caused by misuse, neglect, accident or abnormal operating conditions, repairs will be billed at the normal shop rate. In such cases, an estimate will be submitted before work is started, if requested by the customer.

The foregoing warranty is in lieu of all other warranties, expressed or implied, including but not limited to any implied warranty of merchantability, fitness or adequacy for any particular purpose or use.

Benchmark Media Systems, Inc. shall not be liable for any special, incidental, or consequential damages.

A return authorization is required when sending products for repair. They must be shipped to Benchmark Media Systems, Inc. prepaid and preferably, in their original shipping carton. A letter should be included, addressed to the customer service department, giving full details of the difficulty.

This completes the MP-1 service instructions.

1) Low Noise Electronic Design, C.D. Mothchenbacher & Fitcher, 1973, John Wiley & Sons, Inc., ISBN 0-471-61950-7

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