BENCHMARK MEDIA SYSTEMS, INC.

OBSOLETE
FOR REFERENCE ONLY

MDA-102 Instruction Manual

Table of Contents

1.0 Introduction

The following introduction will familiarize the installer with the System 1000 and the MDA-102.

1.1 System 1000

The MDA-102 is one of a series of very high performance audio distribution and processing modules, a series known as the System 1000. The systems concepts utilized with the System 1000 provides for the highest flexibility, a flexibility that is unparalleled in the industry. Provisions have been made on virtually all of the Benchmark System 1000 modules for accessory daughter boards. The MTX-02 mode controller, the RGC-02 remote gain control, the OSC-01 precision oscillator; and the EQ-02 dual three band semi-parametric equalizer daughter boards are compatible with the MDA-102. The MTX-02 daughter board is of particular interest as it will decode M-S microphones when used on the MDA-102. These daughter boards may be simply added to your System 1000 modules as a field modification at any time. Please check with the factory concerning the availability and applicability of daughter boards.

1.2 MDA-102 Introduction

Specifically the MDA-102 is a combination dual "State of the Art" microphone preamplifier and dual four output line level distribution amplifier. As such, the module is capable of outstanding performance. However, to actually achieve this performance a correct understanding of the proper installation procedures is necessary. It is important that the Benchmark Media Systems application note, "A Clean Audio Installation Guide", be applied as a part of the installation process.

The MDA-102 was designed to provide the highest performance available in microphone signal preamplification and distribution. To achieve the maximum signal to noise ratio in a given installation, it is necessary to distribute the audio signals from the MDA-102 card at line level. The advantages of this are:

1) By first raising the signal level to a nominal line level at or close to the source, minimum interference is allowed to enter the system from power lines, SCR controlled stage lighting, etc.

2) A state of the art preamplifier may be had without purchasing a new audio console.

3) Longer lines may be driven than with a typical microphone, for the same interconnect bandwidth due to the lower source impedance.

4) Multiple outputs are available without any possibility of a shorted output affecting the other outputs.

A block diagram of the MDA-102 is shown in figure 1.1

2.0 Unpacking

Care has been taken in packing the MDA-102 system to assure it will withstand normal shipping conditions. Examine the equipment carefully as it is unpacked. If the shipping carton appears to have been damaged and if there are signs of physical damage, check the equipment and immediately notify the carrier and Benchmark.

3.0 Installation

The installation of the MDA-102 is a simple matter of installing the System 1000 module frames in their proper location, inserting the MDA-102 modules into their companion frames, making the input and output connections and adjusting the gain for the conditions of use. Unless the card frame was specifically ordered at the same time as the MDA-102 modules, phantom power will need to be wired in the frame, only to the specific MDA-102 module locations.

3.1 Frame Location and Magnetic Fields

As with all highly sensitive amplifier systems, electro-magnetic fields may be coupled to the electronics to the detriment of its operation. It is important to locate the amplifier system away from equipment that has power transformers, particularly high current devices such as power amplifiers. Listening to the output of the system without microphones, but with the attenuator pads engaged, under full gain will allow you to determine the presence of any magnetic field pickup. If the MDA-102 is located in an equipment rack, an arrangement of the equipment packages may have to be performed to isolate the preamplifier system from such fields. Mu-metal shielding may need to be employed if re-arrangement does not provide enough improvement. Most helpful is listening to the magnetic environment in the potential mic-pre location with a telephone pick-up coil (Radio Shack part # 44-533), mic-preamp and headphones. For example, you will be quite amazed at the new understanding of you achieve of the EMI radiation problems that exist in computer based equipment.

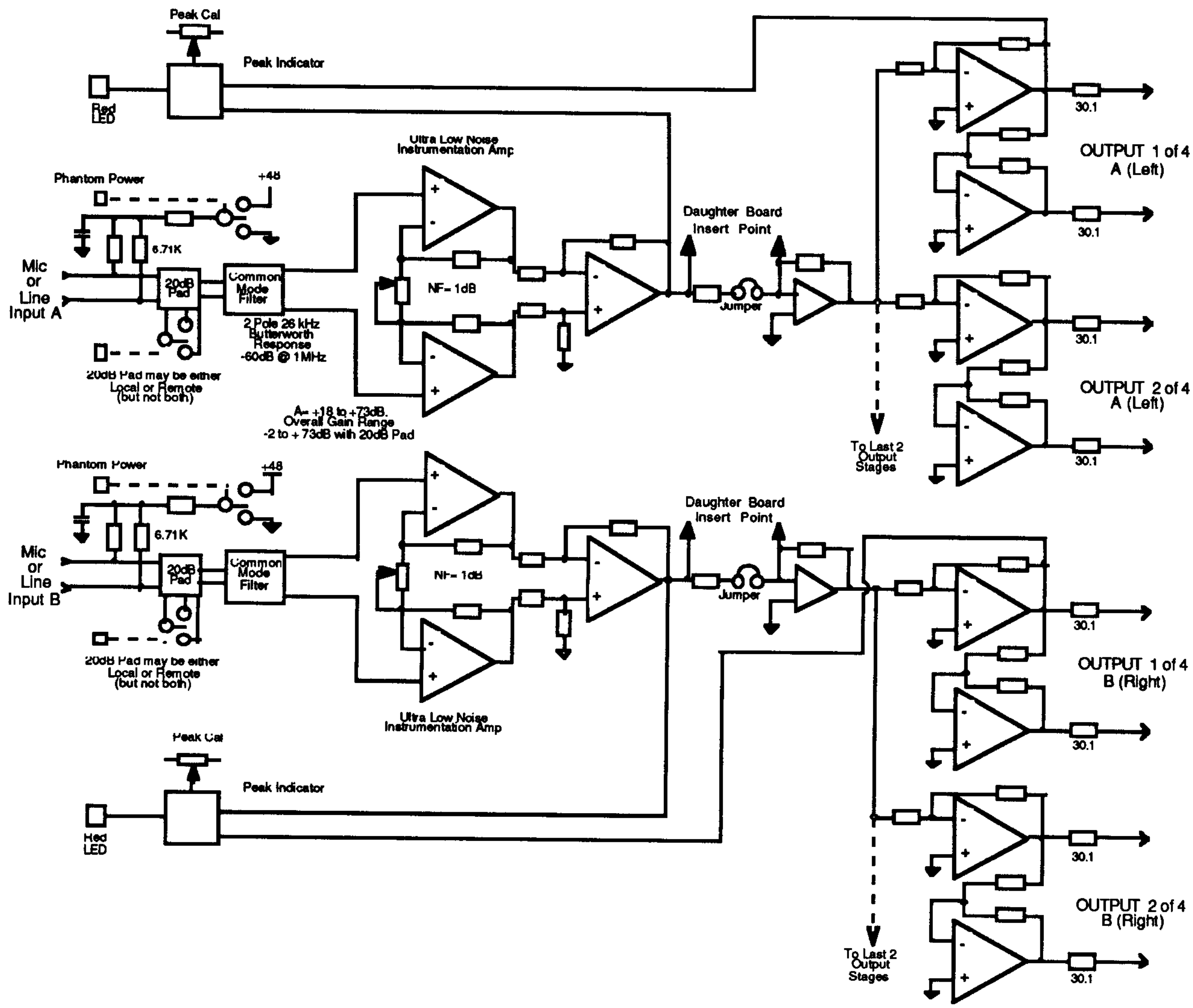

Fig 1.1 MDA-102 Block Diagram

3.2 Input Connections

Audio input is made via the card edge connector at the pins labeled input "Mic Input A" or "Mic Input B" on the inter-connector nomenclature. Please note that there are two inputs to the card, and that they are one above the other. The inputs loop through the card, and hence there are two positions labeled "Mic Input A" or "Mic Input B" from side to side. The various interconnection points are shown in figure 3.1, the MDA-102 Card Edge Connector (as seen from the rear of the card frame).

If you purchased the MDA-102 with the remote attenuator (pad) option, then the control voltage inputs for these relays must be made at the positions just below the audio input connections shown on the following diagram at pins 27 and 28. These inputs, when receiving a +12 volts, will activate the input attenuator, reducing the audio input level to the electronics by ≈20 dB. This is normally only needed for significantly higher than expected microphone output levels where a quick "fix" to input overload is needed.

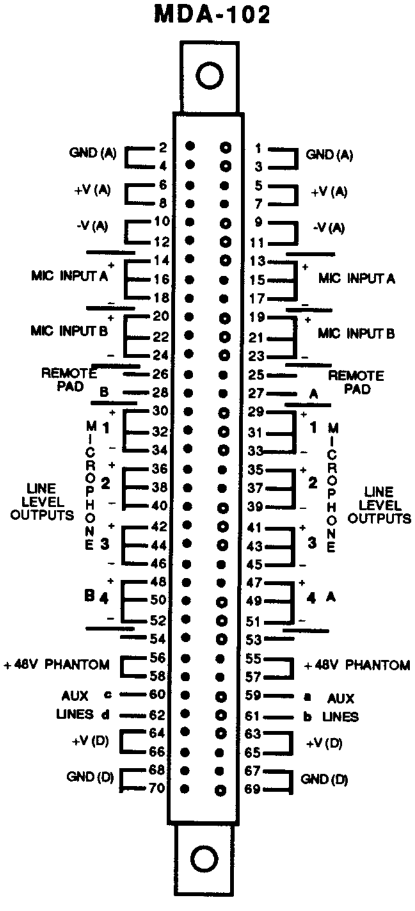

Fig 3.1 Card Edge Connector

3.3 Output Connections

Outputs are taken from outputs labeled 1 through 4 for both channel A and channel B.

While the gain of the MDA-102 will reduce to unity, the module should <u>not</u> be used as a unity gain device at normal <u>microphone</u> levels. To do so will negate the outstanding noise performance of the design. This is because the output noise of the module has a base of -93 dBu, considerably higher than that of a microphone, therefore microphone output levels should not be directly attempted.

!!! Warning !!!

> To maintain a proper signal to noise ratio with outputs that need be at a microphone level, use a passive attenuator to reduce the output level from the nominal line level to microphone level. The Benchmark LMA-1, Line to Mic Attenuator, is the proper device for this application.

Since the output impedance of the card is 60 ohms, the length of the 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 ft. of foil shielded cable may be driven with a small high frequency cutoff of 30 kHz. Since the output device has a current limit of 40 ma, cable capacitance will be the limiting factor in the actual slew rate at the receive end of the cable being driven by the MDA-102. For this reason the use of very low capacitance cable such as Mogami 2574 is recommended for long runs. It has a capacitance of 6 to 7 pf/ft. between conductors, versus 30 to 32 pf/ft. to be found in most foil shielded cables. This cable is available, typically from stock, at Benchmark Media Systems.

3.4 Connector Assembly

Either AMPMODUE or preferably the Molex™ SL (because of their physical size) series pins and housings should be used, and are available from Benchmark, for interconnection to the the System 1000 card frame. Generally these are made up as either two or three pin connectors. Larger connector assemblies may be made, but since the female pins are not zero insertion force type, you may find it difficult to attach the completed connector assembly to the card edge connector posts. On the other hand, if a condition exists that would allow a three pin connector assembly to be inadvertently pulled off the card edge connector, then the added retaining strength of a large assembly may be preferred. Large assemblies, however, have the disadvantage of not allowing additional wiring to the module position while the module is in use, whereas three pin assemblies may be added at any time.

The three pin audio signal connectors also have the advantage of being able to be physically inverted, effecting a polarity inversion of the signal. This is easily accomplished since we have specifically placed the ground or shield of the cable as 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.

3.5 Microphone Power

There are three methods for powering condenser microphones built into the MDA-102. They include the standard Neumann 48 volt phantom system, 12 volt phantom power, and the older A-B or sometimes called "T" power 12 volts system. The second two of these systems require a change in resistor values for R1101/1102 and R4104/4103. Table 3.1 details the resistors used with the different phantom power systems.

Phantom System	Resistor Values	Tolerance	Input A Jumpers	Input B Jumpers
48 volt	6.81 kΩ	0.01%	A-B, D-E	G-H, J-K
12 volt	1.0 kΩ	0.01%	A-B, E-F	G-H, K-L
12 A-B/T	180 Ω	1.0%	B-C, E-F	H-I, K-L

Table 3.1 Phantom Power Resistor Values

The type of microphone phantom power system used determines the jumper arrangement next to the phantom select switch on the front edge of the module. Jumper positions A-B, D-E, G-H, and J-K are the positions next to the edge of the board, while positions B-C, E-F, H-I, and K-L are the positions toward the rear of the board. The phantom power is turned on in the rear position and off in the forward position. The 48 volt system is assumed to be most common and therefore the module, resistors and jumpers, are set up at the factory for this system. Resistor values for the other systems are available from the factory.

If your card frame was not purchased as a dedicated microphone preamplifier system, then you will have to buss the 48 volt supply to each of the card positions used with MDA-102s. This may be done with wire wrapping tools and #30 wire-wrap wire, or by daisy chaining phantom via Molex connectors. The power connections are made to any one or all of pins 55 through 58. The power for 12 volt phantom and 12 volt A-B powering is derived from the +15 volt analog supply on the module itself and does not require any connection be made to the phantom power input pins at the card edge connector. Figure 3.2 shows the jumper location for the phantom selection.

!!! Warning !!!

Do not connect the phantom power buss to any module position other than the MDA-101 or MDA-102 module locations. These pin positions are assigned to other functions in different modules. Connecting phantom power to locations that are used by other modules may permanently damage that module. The same thing, of course, may happen if other modules are in advertently inserted into MDA-101 or MDA-102 locations.

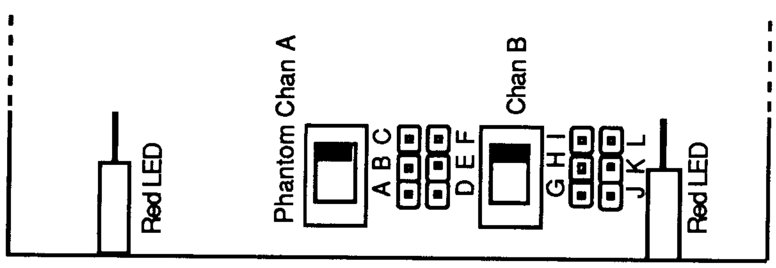

Fig 3.2 Circuit Jumper Arrangement

3.6 Additional Installation Points
The following points should be considered in the installation and operation of the MDA-102.

1. Always use the module at the gain necessary to bring the microphone output up to line level. This is necessary as we are establishing the basic S/N ratio that, in all probability, will be the limit for the rest of the system. The MDA-102 card has an output noise floor of -94 dBu at its minimum A = 18 dB. As the gain approaches 40 dB the noise floor comes up to ≈ -89 dBu. As the gain is raised further from this point, the noise floor will increase directly with the increase in gain.

2. "Catastrophic" results occur (in terms of noise) when trying to use the module as a unity gain replacement to a transformer type mic-splitter. That is, inserting the 20 dB pad and then operating the card at a gain of 20 dB, giving an overall gain of unity, then taking the necessary gain at the input to the equipment being driven. Doing this will reduce the average signal to noise ratio between 30 and 50 dB, or even further, from the 90-100 dB-typical, 120 dB-maximum, that is achievable with this module. Once again, the correct method is to take the necessary gain from the MDA-102, bringing the mic up to line level and use a pad, where necessary, at the input of the console. The LMA-1 from Benchmark, is a 40 dB pad that will mount on the back of a D3M XLR type connector. The LMA has protection against inadvertently applied phantom power from an audio console, a 20 kΩ input impedance and a 200 Ω output impedance.

3. Always use a well thought out grounding scheme with the MDA-102 system. The techniques found in the Benchmark Media application note, "A Clean Audio Installation Guide" need to be applied.

4. Be extremely careful with the use of telephone punch blocks as signal tie-points. We have been informed by a member of the telephone community that a continuous degradation at the contacts occurs with punch blocks. The point of contact between the wire and the punch block, it seems, becomes a rectifier over a period of time - not a healthy situation for high quality audio. Previous problems with hum as a part of the signal have been directly traced to poor connections made at punch blocks. Additionally, only ground switching jack fields, if any, should be used on microphones to prevent any common grounds prior to the input of the system.

4.0 Specifications
In the area of common mode rejection, the MDA-102 card has additional performance built into its circuitry, over what is normally found in microphone preamplifiers, by the addition of a common mode filter. All interfering signals see a 26 kHz LC low pass filter, and thus are down 60 dB @ 1 MHz. The desired differential signal, however, cancels the effect of the inductor, and hence is not limited by it. The performance of the preamplifier section is much superior to preamps found in most audio consoles, with THD @ 2 K Hz ≈ 0.002% (A=40 dB), and THD @ 20 kHz ≈ 0.006%. The differential bandwidth is 200 kHz, 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 dB. 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 high frequency cutoff point. The major source of the truly outstanding aural performance, however, is a result of the MDA-102s ultra low intermodulation distortion.

5.0 Operation
This section provides a description of the operation of the MDA-102 Preamplifier System. Figure 6.0 shows the control locations on the front of the MDA-102.

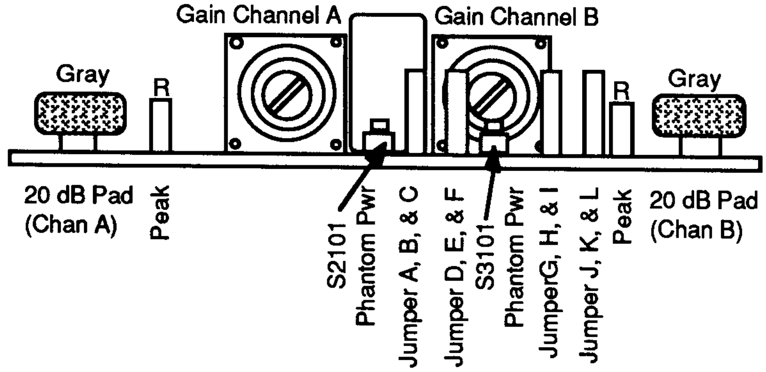

Figure 6.0 - MDA-102 Front View

5.1 General Operation

The operation of the MDA-102 is generally intuitive. However the following items may not be immediately apparent.

5.1.2 Phantom Power

While you may never notice a problem, if Phantom power is to be used, the phantom power switch should be turned on approximately one hour before the use of the system. This minimizes the possibility of noise in the system, by allowing the proper formation of the dielectric in the phantom power coupling capacitors. This in turn ensures no DC leakage currents, the source of noise in capacitors.

If a microphone is wired to feed two MDA-102 modules or sections, and phantom power (Neumann style) is used to power the microphone, both modules should have their phantom power switches turned on to maintain proper voltages in the circuits.

5.1.4 Setting Levels

The setting of the amplification of the two channels of the MDA-102 is accomplished with the two physically large potentiometers whose shafts protrude toward the front on the module. The upper (or left unit in the illustration above) potentiometer controls channel A and the lower unit controls channel B. The range at the shaft of the gain control potentiometer is from +18 (extreme counterclockwise rotation) to +73 dB, unless a smaller range is ordered (extreme clockwise rotation). Additional gain reduction may only be accomplished by use of the 20 dB attenuator. With the attenuator in circuit, the amplification range is from -2 to +53 dB. However for optimum noise performance amplification with the attenuator in circuit should be limited to a range of from -2 to +18 dB.

Practically the amplification is set with the intended microphone in place and the sound source that will be using that microphone also in place. Level checks are then made and the amplification of the MDA-102 set accordingly.

The output clip point of the microphone preamplifiers is typically as high or higher than the input of the device receiving the signal. This provides the greatest amount of headroom possible.

With any recording, the maximum dynamic range of the recorder should be utilized. Digital recorders, in spite of their dynamic range improvements, should be used as close to digital clip as possible to ensure the greatest bit utilization and hence the greatest possible accuracy. This presents a conflict between maximum utilization of the media and adequate overload margin to account for the dynamics of the recorded material. It is imperative to do a "run through" for level setting with the orchestra or group being recorded to ensure that the levels have been set accurately for the maximum SPL of the material being recorded. Once this has been accomplished, an extra 5 to 10 dB should be included to account for the psychological difference between a rehearsal and the live performance, particularly with non-professional musicians.

One way to ensure the maximum utilization of the recorded "bits" is to include some form of companding on the input of the digital recorder. Either the Dolby® SR™ or the ANT "telcom c4" are high quality noise reduction systems which will force the utilization of the upper (highest accuracy) bits when recording with a digital system. Additionally the dynamic range can be improved beyond the 16 bit limit (≈92 to 96 dB), up to 118 dB. This will fully exploit the dynamic range capability of the MDA-102.

5.2 Discussion of Noise and Microphone Performance

The engineers at Benchmark Media Systems have expended considerable effort to produce an amplifier with 1 dB noise figure so the user may enjoy the highest signal-to-noise ratio possible. It is appropriate, therefore, that a discussion of noise and microphone performance be included.

5.2.1 Noise Primer

Noise figure is a measure of how well an amplifier amplifies the intended signal without adding noise. In the case of the MDA-102, each amplifier adds only 1 dB of noise to that of the original signal for amplification factors greater than 40 dB. The noise figure is referenced to the Johnson noise of the resistive portion of a transducer source impedance.

Johnson noise may be calculated from:

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

Where:

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

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 20 kHz bandwidth noise voltage of -129.57 dBu.

$$\mathrm{dBu} = 20 \log \frac{V}{0.7746} \qquad [2.0]$$

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 <u>totally noiseless amplifier</u> operating at 50 dB of amplification from a source resistance of 150 ohms at room temperature would be -80.82 dBu. The MDA-102 preamplifier's performance under these conditions is approximately -80.0 dBu. At their minimum amplification (18 dB), each MDA-102 preamplifier has an output noise floor of -94 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.

Given the source resistance, or self noise in the case of a condenser microphone, and the sensitivity, an evaluation of the performance of any microphone with the MDA-102 preamplifier systems under various sound pressure level conditions can be accomplished. The following examples utilize two different microphones to apply the understanding of noise.

In the following examples there are a number of reference points that will be used, a brief explanation of them is in order.

1. The first reference is the standard sound pressure level at which microphone output voltage, that is sensitivity, is referenced. That, most commonly, is 94 dB SPL. This is the same as 10 dynes/cm sq. or 1 Pascal of pressure. This is a fairly loud sound pressure level, but by no means a "Rock" SPL.

2. The second reference is the equivalent input noise of the MDA-102 microphone preamp, which is -130 dBu. All amplification figures are added to this figure to give the output noise of the preamplifier.

3. The third reference point is the operating voltage level of the audio system, and for these examples +4 dBu has been chosen, since it is the most common in recording and radio. +8 dBu is common for television.

4. The fourth reference point is the output clip level for the MDA-102 operating with ±15 volt power supplies, and that is +27 dBu.

Additionally, signal-to-noise ratio is defined as the difference between the system noise floor and the average operating voltage level (in this case +4 dBu). Headroom is defined as the difference between the average operating level and the peak clip point, and in our example it would be (+27 dBu) - (+4 dBu) = 23 dB. Dynamic range is defined as the sum of these two figures, i.e. the difference between the noise floor and the output clip point.

It should also be remembered that an increase or decrease in dB of sound pressure level at a microphone will result in a corresponding change in output voltage level (in dB) both at the output of the microphone and at the output of the preamplifier. Therefore these dB changes may be added or subtracted directly between the acoustic and the electronic environments.

5.2.2 Electro-Voice RE-20 Microphone

The sensitivity of an Electro-Voice RE-20 is 1.09 mv (-57 dBu) at 94 dB SPL input. If a voltage amplification of 58 dB is used, the output noise of the MDA-102 will be approximately -72 dBu.

$$-130 \text{ dBu} + 58 \text{ dB} = -72 \text{ dBu}. \qquad [6.1]$$

If the sound pressure level is sufficient to give an output of +4 dBu then the average signal-to-noise ratio will be 76 dB.

$$+4 \text{ dBu} - (-72 \text{ dBu}) = 76 \text{ dB}. \qquad [6.2]$$

The dynamic range under these conditions is 99 dB.

$$+27 \text{ dBu} - (-72 \text{ dBu}) = 99 \text{ dB} \qquad [6.3]$$

The sound pressure level necessary to achieve an average output of +4 dBu is +97 dB SPL.

If @ 94 dB SPL

-57 dBu [*Mic output*] + 58 dB [*Gain*] = +1 dBu [*Preamp out*]

Then:

94 dB SPL + 3 dB SPL will yield +1 dBu + (+3 dBu)

That is:

a 97 dB SPL input will yield +4 dBu preamp output [6.4]

The peak SPL that the system can handle is +120 dB SPL. At this amplification (58 dB) and Sound Pressure Level the preamp will reach its output clip point of +27 dBu.

+4 dBu [*system reference*] + (+23 dB) [*headroom*] = +27 dBu [*peak clip*]

+97 dB SPL + 23 dB = +120 dB SPL [*peak clip*] [6.5]

Please note that this does not necessarily represent the peak output capability of the microphone, it may be higher or lower than this SPL for a given distortion point.

5.2.3 Sennheiser MKH-40-P48

The performance of the MDA-102 with the new high performance Sennheiser MKH-40-P48 will now be examined. This microphone has a very high sensitivity of 25 mv/Pascal (10 dynes/cm sq. = 1 Pascal = 94 dB SPL), and very low self noise (a +48 volt phantom powered condenser microphone with internal electronics) of 12 dBa. The self noise, therefore, is ≈1.99 microvolts. This is -111.82 dBu.

Using 18 dB of amplification at the MDA-102 or preamplifier, the combination output noise is approximately -93 dBu (20 kHz noise bandwidth).

At the reference SPL of 94 dB the output of the microphone is 25 mv or -29.82 dBu. Add 18 dB of amplification and the output amplitude from the preamp is,

-29.82 +18 = -11.82 dBu [*preamp output @ 94 dB SPL*]

Therefore, to get +27 dBu out, the input SPL must be,

+27 dBu - (-11.82 dBu) = 38.82 dB [*additional mic output before clip*] [6.6]

to be added to the reference SPL of 94 dB SPL with the resulting 132.82 dB SPL.

Thus there is an average signal-to-noise ratio of 97 dB.

+4 dBu - (-93 dBu) = 97 dB [6.7]

and a dynamic range of 120 dB.

+27 dBu - (-93 dBu) = 120 dB [6.8]

at a peak acoustic input of 132.82 dB SPL. This is just below the 135 dB SPL / 0.5% THD point of the microphone.

109.82 dB SPL average is required at this amplification for an output of +4 dBu.

+4 dBu - (-11.82 dBu) = 15.82 dB [*above the +94 dB SPL*]

15.82 dB + 94 dB SPL = 109.82 dB SPL [*for +4 dBu out of preamp*] [6.9]

Problem

If a digital recorder has a dynamic range of 95 dB, an input clip point of +21 dBu into an unbalanced input, and it is desired that the recorder be fed unbalanced with the MDA-102 preamplifier which is adjacent to the recorder. What is the lowest peak sound pressure level that the Sennheiser MKH-40-P48 microphone can receive and still maintain the full dynamic range of the recorder. Also what is the amplification necessary from the MDA-102 to achieve this performance.

Solution

Since the output of the MDA-102 will be used unbalanced, the output clip point of the system (one output lead and ground) is +21 dBu. This conveniently matches with the recorders input clip point of +21 dBu.

By using the MDA-102 as an unbalanced output device 6 dB of amplification is lost, therefore, in this application the preamplifier has an overall amplification range of -8 to +64 dB.

Since the dynamic range of the recorder is 95 dB then the recorder noise floor is

+21 dBu - 95 dB = -74 dBu. [6.10]

If the microphone self noise is -111..82 then the maximum amplification that can be used is (recorder noise floor - microphone self noise)

-74 - (-111.82) = 37.82 dB [6.11]

If both the recorder and the MDA-102 have identical noise voltages they will add resulting in a 3 dB loss in dynamic range. As a result, it is well to keep the MDA-102s output noise voltage 3 dB lower than the noise voltage of the recorder. Thus a 35 dB amplification factor is selected. Therefore the peak input clip is

+21 dBu - 35 dB = -14 dBu [6.12]

From the microphone sensitivity figure given by the manufacturer at 94 dB SPL into the mic, the output voltage is 25 mv which we previously found to be -29.82 dBu. The result is an 15.82 dB increase allowable in SPL from reference. Preamp input clip - output at the sensitivity rating = additional SPL over reference,

-14 dBu - (-29.82 dBu) = 15.82 dB [6.13]

That is,

94 dB SPL + 15.82 dB = 109.82 dB SPL [*system clip point*] [6.14]

Also, if the maximum SPL that the system can receive is 109.82 dB, then the noise floor of the system is,

109.82 dB SPL [*system clip point*] - 95 dB [*dynamic range*] = 14.82 dB SPL [6.16]

This is the equivalent acoustic noise floor of the system.

In other words the ambient noise floor of the recording environment must be below 14.82 dB SPL to utilize the dynamic range of the recorder, very stringent requirements indeed.

If it is assumed that this hypothetical recorder is a "semi-pro" device and that a nominal input level of -10 dBV is what will give a 0 indication on the device's meter (-10 dBV = -7.78 dBu), then the drop in level that will yield a 0 indication is,

$$+21 \text{ dBu} - (-7.78 \text{ dBu}) = 28.78 \text{ dB} \quad [6.17]$$

and then the average SPL (assuming an average type meter) at the microphone would be,

$$109.82 \text{ dB SPL} - 28.78 \text{ dB} = 81.04 \text{ dB SPL}. \quad [6.18]$$

While the above calculations are hypothetical, they are quite close to the real world. They demonstrate how the microphone and preamp can set the dynamic range of a system. They also highlight the need, in this digital recording era, to use recording environments with extremely low acoustic noise levels to realize the recorder's full dynamic range. In most cases air handling equipment will need to be turned off during the actual recording, and even this may not be enough improvement if the site is near a transportation terminal. In practicality, the microphones need to be as close as possible to the acoutic source, given the constraints of coverage patterns and instrumental balance, to take advantage of the higher SPL and to minimize the effect of the ambient noise.

The amplification required for most microphones will typically be between 20 and 40 dB but may be as high as 60 to 70 dB with ribbon microphones, and the preamp section will often be the limiting factor in the output noise of a console or other electronics prior to any recording or transmission medium. The majority of amplification needed, consistent with desired headroom, should be taken from the MDA-102 since it has the lowest noise figure of any of the amplifying stages under these amplification factors.

5.2.4 Miscellaneous

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 as might be done with instruments or telco lines, the noise output of the preamplifier is still ≈92 dBu (150 ohm source), yielding a dynamic range of 119 dB, yet with 30 dB of gain. This is far better than is possible with a normal line level amplifier.

6.0 MDA-102 Preamplifier Circuit Description

The following is a generally complete description of the circuitry comprising the MDA-102 preamplifier.

6.1 General

The superior performance of the MDA-102 is the result of careful attention to each element of the circuit design, particularly in the design of the first amplifying stage. The choice of an active amplifier circuit as opposed to the use of a transformer input, allows freedom from the significant disadvantages of audio transformers, being low frequency saturation and most importantly the use of a noninverting amplifier topology that most often suffers from common mode non-linearities. The major disadvantage of the active type input, that of RF susceptibility, has been dealt with by the use of the common mode filter, with no degradation in noise performance. The following description will give insight into the operation of the amplifier. Following the description with the schematic in hand, document 450057, will aid in its understanding.

6.2 Phantom Power Circuitry

As mentioned in section 3.4 the MDA-102 is capable of three different condenser microphone powering methods. The first and most common is the Neumann 48 volt method. This method sends its current out both input wires to the microphone and returns the current via the shield of the cable. Its advantage is that any residual mains related ripple will be rejected by the common mode rejection capability of the microphone preamplifier.

Second is the low voltage, 12 to 15 volt, version of this same scheme. This is convenient when high voltage for the original Neumann system is not available as it derives power from the incoming +15 volt supply.

The last option is the "A/B" or "T" power system. This is not used very much anymore, except in the film industry, because ripple that may be present on the DC power is added directly to the signal from the microphone. And since this signal will be amplified up to 70 dB the purity requirements placed upon the power source are extremely stringent.

Every one of the three powering methods requires different series resistors (R1201/1202 and R4201/4202) to couple the power from the low impedance power source to the microphone line. All three systems require the resistors be matched to 0.01% to maintain the common mode rejection capability of the amplifier. The only possible exception might be the A/B-T power system. Resistor positions (R3106 and R4105) are provided to allow the microphones current drain, when using the A/B-T power system, to drop the voltage to 12 volts, and at the same time allow C4202 and C4201 to perform a superior job in cleaning up the incoming power. Resistors R3101/2105 and R3104/3105 de-couple the potentially low impedance of the phantom resistors from a microphone while still discharging the input coupling capacitors. These resistors may be increased in value or even removed if the need exists for a higher input impedance (up to ≈10 KΩ per leg) is needed.

6.3 Input Stage

The input signal first encounters the phantom power circuit which will normally consist of a pair of 6.81K ohm resistors that feed power to the microphone line. This phantom power is turned on and off utilizing the miniature black slide switches (S2101 and S3101) located on the front edge of the MDA-102.

Next in line is the 20 dB pad which consists of two 2K ohm resistors plus one 200 ohm resistor. The 20 dB pad is activated by either the gray pushbutton switch (S1201/ 4201) located on the MDA-102, or by the remote control relay K1301/4301. The relay is activated by the application of +12 volts at the input of the control lines located at the card edge connector.

Next are input coupling capacitors (C2401/3202 and C3302/3201). These capacitors are a very low leakage variety of aluminum electrolytic rather than tantalum. They are noted for their superior dielectric absorption characteristic (distortion producing mechanism in capacitors). Additionally, as with all aluminum electrolytics in this module, these input capacitors are bypassed with large film capacitors (C2402/3301 and C3303/3101) to improve the high frequency dielectric absorption properties. Low leakage electrolytic capacitors are used to ensure low noise since any leakage currents in these capacitors results directly in input noise to the amplifier.

The next section is the common mode choke (L4401). This consists of a highly symmetrical dual winding on a common toroid core with an inductance of 38 millihenries per leg. The operation of the choke is such that with a differential signal the magnetic fields created by the two highly symmetrical windings will cancel one another. This results in a net inductance of zero for differential signals, whereas common mode signals see the 38 mH inductance. The choke coupled with the 1000 pf capacitors (C3301 and C3302) and 10.0 kΩ termination resistors (R3303 and R3304) form a two pole Butterworth low pass filter whose corner frequency is 26 kHz with a 12 dB/octave roll off.

Additionally four zener diodes, two on each input (D1401-1403, 1501 and D4401-4403, 4501), protect the input transistors from over voltages. The voltages of these Zener diodes are chosen such that the sum of the forward drop and the zener voltage (≈ 6.7 volts) is less than the zener turn-on voltage of the input transistors. This protects the ultra low noise transistors. It prevents them from going into the emitter-base zener mode which would destroy their low noise capability and also make the devices more prone to failure.

Next is the low noise active differential amplifier. This topology adds a single transistor (Q2401/2401 and Q3401/3402) input stage to each of two high quality operational amplifiers (package U2401 and U3401). This allows the formation of a high gain emitter-coupled feedback differential-amplifier. The noise figure of the transistors is less than 1 dB referenced to 150 a Ω source and is still less than 2.5 dB referenced to 10 a 10 Ω source. These transistors were developed for use as moving coil head preamplifiers and their noise performance is maintained by virtue of the fact that no series input resistors are used prior to the devices.

By coupling the input signal directly to the bases of the transistors the only noise producer in the signal path is the intrinsic base resistance ($r_{b'b}$) of the transistors and this has been reduced to an extremely low value in the design of the transistor (approximately 2 ohms).

Since the common mode filter has no appreciable series resistance it does not limit the noise performance of the transistors. The source resistance of the microphone is the major noise producer. The amplification factor of this stage is established by the ratio of the variable resistor between the emitters of the two transistors and the feedback resistors from the outputs of the op-amps. The values selected allow for an amplification range of 6 to 58 dB at the differential pair. The collector currents for the transistors are set by the fixed bias voltage at the non-inverting inputs of the op-amps and by the collector and feedback resistor networks. Coupling capacitor pair C1301 and 1202 (C4301 and 4203) prevent the amplification of DC signals and the accompanying 1/f noise increase. The polarity of the electrolytic capacitors in this capacitor pair, is chosen at the time of test and is a function of the DC offset voltage caused by the variation in V_{be} of the input transistors. Should one or both of the input transistors be replaced the polarity of the offset voltage should be determined (at the terminals of the electrolytic capacitor) and the capacitors polarity inverted if need be. If the polarity is not correct and the offset voltage is large, a phenomena we call gain drift will occur. This manifests itself as a continuing slide in gain for a number of seconds after the gain potentiometer is released, and can be very frustrating when trying to achieve precise gains with the pre-amp.

6.4 Differential Converter

The differential converter consists of capacitors C2601/2504 and C3504/3601, resistors R2602 through R2606, R3603, 3605, 4601-4603 and the A sections of U2601 and U3601. The output clip point of the input stage has been reduced by the bias configuration 6 dB from what would normally be available to a stage with a 30 volt supply. Therefore, the differential input converter has 6 dB of amplification to equalize the clip points of the two stages.

The differential converter converts the differential signal from the first stage to a single-ended signal. This stage rejects low frequency common mode signals and power supply noise that may be present.

The output of the differential converter is the input signal source for any daughter board accessories. Additionally it is a signal pickoff point that feeds the circuitry for the red peak indicating LED.

The common mode rejection is set by the resistive (R2604, R4602) and capacitive (C2606, C4601) trimmers in these circuits. Typically, a null of -100 dB can be achieved from low to mid-band and -60 dB or better null at 20 kHz.

6.5 Patch/Daughter Board Insertion Point

The line level signal may be patched out of the board, in a single ended fashion, for processing such as compression and/or equalization, if desired. This is accomplished by removing the signal jumper (W1601, W3601) and wiring the output of the differential converter to one of the aux lines. An input resistor (R1602, R3601) must be installed at the summing node of the output driver op-amps to correctly establish the gain of that stage. 4.99 kΩ is the correct value for unity gain.

6.6 Output Driver

The output driver consists of R1601 (R3602), jumper W1601 (W3601), R2601 (R3604), C2602 (C3605) and the B halves of U2601 and U3601.
The output drivers feed each of the inverting inputs of the four associated output stages from a low impedance source. It also provides the input for the signal return from any daughterboard accessories. As such jumpers W1601 and W3601 allow the interruption of the direct signal, additionally resistors R1602 and R3601 may or may not be needed depending upon whether the daughterboard output is a voltage source or a current source. Most daughter board outputs are current sources. If it is not needed then it must be either jumper bypassed or replaced with a jumper wire.

6.7 Output Stages

There are four output stages per section and they are have a fixed gain of +6 dB by virtue of the fact that two unity gain stages with opposite swings have a voltage doubling effect. This yields an amplification range for the MDA-102 of from less than +18 to greater than +70 dB. The 20 dB pad increases the overall range to greater than 72 dB (-2 to +70).

The output stages are separate op-amp drivers consisting of U1701 through U4702 and the amplifiers associated components. Each of the output stages use two sections of a dual operational amplifier to form a balanced output stage. Both parts of each output stage are unity gain inverters using 10.0 kΩ input and feedback resistors, with 39 pF feedback capacitors around the first section, and 22 pF feedback capacitors around the second section. Both halves of each output stage have 30.1 Ω buildout resistors to provide a balanced output impedance of 60.2 Ω.

The DC offset voltage from the output amplifiers of the MDA-102 is typically less than 10 millivolts.

6.8 Peak Indicator Circuitry

Peak indication is given by two red LEDs that are next to the two 20 dB pad switches, and their associated circuitry. They are set up to indicate the approach of peak clip at the output of the input stage and/or at the output stage of the DA section. The input diodes form an analog "or" circuit in that either signal input, or both, can turn on the indicator. Since they sense both points, the effect of any daughterboard accessories that may be present will also be included in their operation. The peak indicator may be set to turn on in a range of from ≈ +16 to ≈ +26 dBu, and are factory set for an output level trip point of +20 dBu unless otherwise specified. The range may be modified, if desired. Check with the factory for the appropriate resistor value changes for the desired range.

The peak indicators consists of a dual operational amplifier U1201 with both sections operating as voltage comparators for the respective A and B sections, diodes D1201 and D1202 (D4201, D3202, input trimmer R2104 (R3103), input resistor R2102 (R3202), bias resistor R2103 (R3203) and bias diode D2101 (D3201). Feedback is provided by resistors R1104 (R1204) and R2101 (R2201) as well as C1101 (C1201). The output of the comparator swings between the positive and negative power supply rails. Resistor R1103 (R4102) limits the current that may flow through diode D1101 (D4101). The peak indicator is an oscillating comparator due to the A.C. coupled hysteresis applied

The common mode rejection is set by the resistive (R2604, R4602) and capacitive (C2606, C4601) trimmers in these circuits. Typically, a null of -100 dB can be achieved from low to mid-band and -60 dB or better null at 20 kHz.

6.5 Patch/Daughter Board Insertion Point

The line level signal may be patched out of the board, in a single ended fashion, for processing such as compression and/or equalization, if desired. This is accomplished by removing the signal jumper (W1601, W3601) and wiring the output of the differential converter to one of the aux lines. An input resistor (R1602, R3601) must be installed at the summing node of the output driver op-amps to correctly establish the gain of that stage. 4.99 kΩ is the correct value for unity gain.

6.6 Output Driver

The output driver consists of R1601 (R3602), jumper W1601 (W3601), R2601 (R3604), C2602 (C3605) and the B halves of U2601 and U3601.
The output drivers feed each of the inverting inputs of the four associated output stages from a low impedance source. It also provides the input for the signal return from any daughterboard accessories. As such jumpers W1601 and W3601 allow the interruption of the direct signal, additionally resistors R1602 and R3601 may or may not be needed depending upon whether the daughterboard output is a voltage source or a current source. Most daughter board outputs are current sources. If it is not needed then it must be either jumper bypassed or replaced with a jumper wire.

6.7 Output Stages

There are four output stages per section and they are have a fixed gain of +6 dB by virtue of the fact that two unity gain stages with opposite swings have a voltage doubling effect. This yields an amplification range for the MDA-102 of from less than +18 to greater than +70 dB. The 20 dB pad increases the overall range to greater than 72 dB (-2 to +70).

The output stages are separate op-amp drivers consisting of U1701 through U4702 and the amplifiers associated components. Each of the output stages use two sections of a dual operational amplifier to form a balanced output stage. Both parts of each output stage are unity gain inverters using 10.0 kΩ input and feedback resistors, with 39 pF feedback capacitors around the first section, and 22 pF feedback capacitors around the second section. Both halves of each output stage have 30.1 Ω buildout resistors to provide a balanced output impedance of 60.2 Ω.

The DC offset voltage from the output amplifiers of the MDA-102 is typically less than 10 millivolts.

6.8 Peak Indicator Circuitry

Peak indication is given by two red LEDs that are next to the two 20 dB pad switches, and their associated circuitry. They are set up to indicate the approach of peak clip at the output of the input stage and/or at the output stage of the DA section. The input diodes form an analog "or" circuit in that either signal input, or both, can turn on the indicator. Since they sense both points, the effect of any daughterboard accessories that may be present will also be included in their operation. The peak indicator may be set to turn on in a range of from ≈ +16 to ≈ +26 dBu, and are factory set for an output level trip point of +20 dBu unless otherwise specified. The range may be modified, if desired. Check with the factory for the appropriate resistor value changes for the desired range.

The peak indicators consists of a dual operational amplifier U1201 with both sections operating as voltage comparators for the respective A and B sections, diodes D1201 and D1202 (D4201, D3202, input trimmer R2104 (R3103), input resistor R2102 (R3202), bias resistor R2103 (R3203) and bias diode D2101 (D3201). Feedback is provided by resistors R1104 (R1204) and R2101 (R2201) as well as C1101 (C1201). The output of the comparator swings between the positive and negative power supply rails. Resistor R1103 (R4102) limits the current that may flow through diode D1101 (D4101). The peak indicator is an oscillating comparator due to the A.C. coupled hysteresis applied around the device. This assumes that the input signal does drop below the threshold point after seeing a peak to allow for reset.

7.0 Trouble Shooting and Repair

The MDA-102 was designed and manufactured to the strictest commercial standards. As such the probability of failure is very small. However should you experience difficulty the following procedures should be employed.

7.1 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 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 limiting of the power supplies to 150 mA for the analog supplies and 100 mA for the logic supply. This will protect the module and still allow the location of failures to be made.

2. Since most failures are catastrophic in nature rather than a gradual degradation of performance, make a close visual inspection of the module for any discoloration of components and possible shorts on the PC board itself. Discoloration would indicate excessive heat, most likely from a component failure. Remove any component that has obviously failed, i.e. carbonized resistors or I.C. packages that are cracked.

3. 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 any further failures, and the presence of a short will be shown by the current limiting of the power supplies. Allow the module to operate in this condition.

4. Look for any components that are operating too hot to the physical touch. This will show where the shorts are when there is no physical symptoms. Typically one can just keep their hand on a surface at 130° F. With one exception, that of the PS-101, all of the components of the System 1000 are meant to operate at temperatures lower than this.

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 in part. If further problems exist after the power supplies come out of current limiting, they can most often be found by performing voltage checks through the circuitry.

7.2 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, such as the Pace MBT-100.

The proper technique with these stations is to apply the tip to the area to be de-soldered and wait for the solder to thoroughly melt. You can be sure of a thorough melt by observing the top side of the board. <u>When</u> 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 without 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 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. 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 overheat 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 before applying vacuum. This technique uses new solder as an efficient heat conductor to the total area, eliminating hot spots.

7.3 Circuit Board Re-Soldering

NASA has developed an effective technique that ensures highly reliable solder joints. It involves first heating the component lead, since it usually has the higher mass, by applying a small amount of solder to the tip of the soldering iron at almost the same time as you apply the iron to the component lead. This will allow some flux to make it to the component lead. The iron should be approximately 1/8" above the board. When the lead has come up to temperature so that it melts the 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 a couple of seconds. If the component that is to be installed has leads that are oxidized, it will be necessary to clean them. This may be done with either a Scotch Bright® abrasive pad or fine bristle fiberglass brush, among other methods.

This completes the MDA-102 Instruction Manual

Ed. 28 Oct 1988

BENCHMARK MEDIA SYSTEMS, INC.
3817 Brewerton Road North Syracuse, NY 13212
(315) 452-0400