BENCHMARK MEDIA SYSTEMS, INC.

MDA-102/PA Instruction Manual

1.0 Introduction

The MDA-102/PA is a dual channel microphone preamp - distribution amplifier. On board signal routing provides the following mode control: Left only; Right only; discrete Stereo; or a Mono sum via either or both Left and Right outputs. The module has two balanced output sections, each of which have five 60 Ω outputs. One of the outputs can be used as a direct output (0 Ω) capable of driving fifteen external 60 ohm outputs (please specifiy when ordering). The MDA-102/PA has an output clip point of +27 dBu, and an input clip point equal to: +27 dBu minus the gain of the preamp. Crosstalk performance is ≈ -100 dB at 2 kHz and -75 dB at 20 kHz. The dual red LEDs monitor the peak level of the DA, and are normally set to turn on at +20 dBu.

2.0 Unpacking

Care has been taken in packing the MDA-102/PA module 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 or if there are other signs of physical damage, check the equipment and immediately notify the carrier and Benchmark Media Systems, Inc.

Please Note

Check *all* portions of the packing material for installation accessories and manuals. Filler boxes are often used to ship interconnection pigtails, instruction manuals, rack mount accessories, and small tools and fuses.

3.0 Installation

For basic installation concepts we recomend that you follow the "Clean Audio Installation Guide." The MDA-102/PA module is specifically suited for stereo microphone applications. As such the MTX-02 daughterboard is an ideal accessory that provieds the ability to remotely select between descrete stereo applications such as coincident pair, ORTF and spaced omnis. The sum and difference portions of the MTX-02 daughter board allow the use of a Mid-Side microphones with decoding happening right at the module.

Install the System 1000 frames in a location that is free from strong magnetic fields and preferably free from RF signals as well. Insert the MDA-102/PA modules into their frames. Connect the input and output cabling and adjusting the gain for the conditions of use.

3.1 +48 V Phantom Power

Due to the limited number of connector pins available on the card edge connector, phantom power is **not** connected to the module locations by way of the backplane. Therefore, unless the card frame was ordered at the same time as the MDA-102/PA modules, phantom power will most likely need to be connected to the module positions at the back of the frame. Be careful to wire **only** to the specific MDA-102/PA module locations. The phantom power pin assignment (pin 26) has other uses from module to module.

3.2 Frame Location and Magnetic Fields

As with all high gain amplifier systems, it is quite possible to couple electro-magnetic fields to the electronics. This is always to the detriment of its operational performance. It is important to locate the amplifier system away from equipment that has power transformers, particularly high current devices such as power amplifiers and the power supply of the System 1000. Listening to the output of the system without microphones, but with the 20 dB attenuator pads engaged, under full gain will allow you to detect the presence stray magnetic field pickup. At the installation level, physical distance is the only practical way to eliminate interference from magnetic fields.

A site survey can be performed with the use of a telephone pick-up coil (Radio Shack part # 44-533), portable mic-preamp and headphones. You may be quite amazed, for example, at the EMI radiation that exists in computer based equipment.

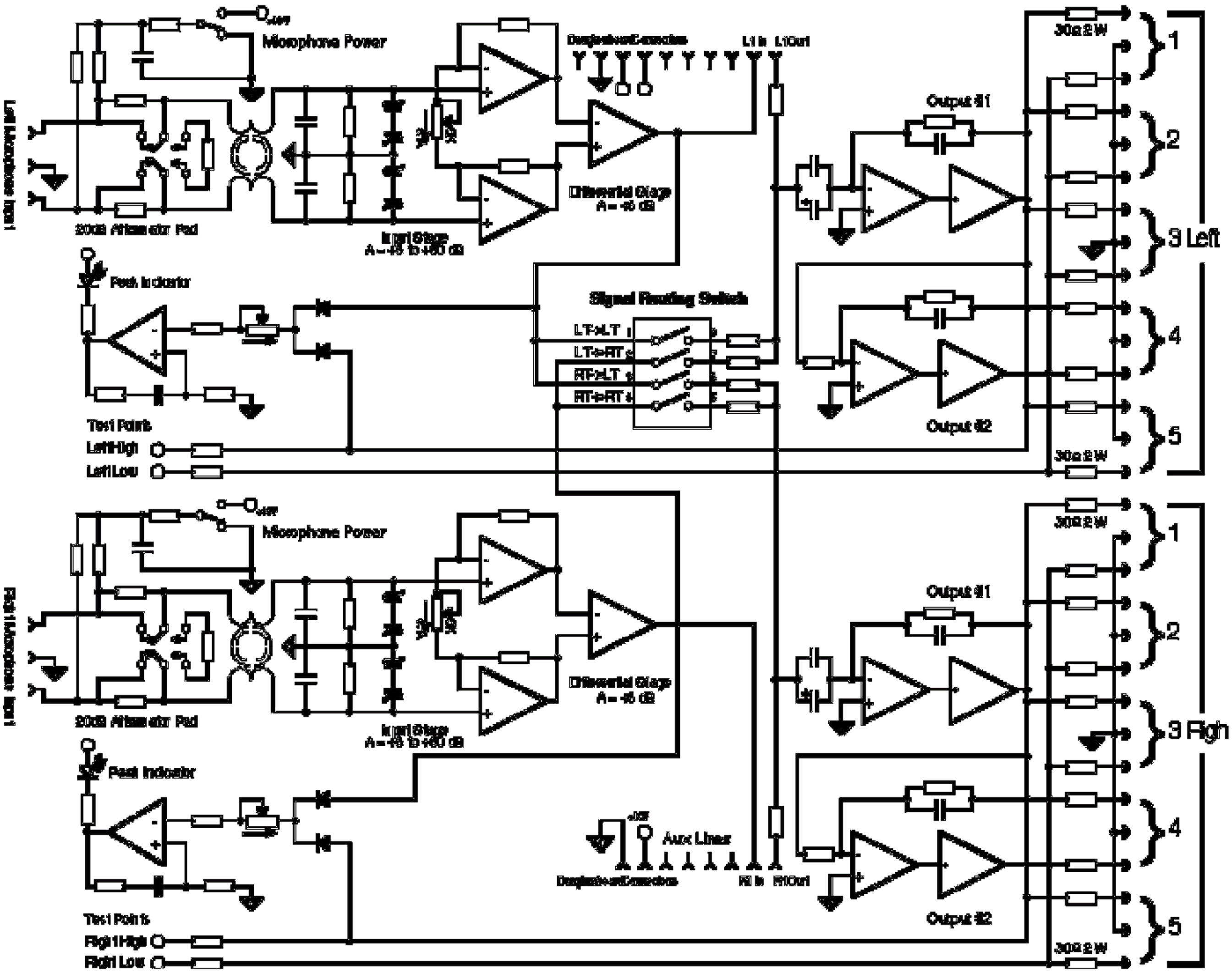

Fig 3.1 MDA-102/PA Block Diagram

3.3 Input Connections

Input to the board is made with the top two signal positions of the module edge connector labeled Left and Right. The input impedance of the MDA-102PA is ≈ 7 kΩ (balanced. Any unused inputs should be back terminated with 150 ohms or less to prevent the pickup of unwanted electromagnetic radiation.

Modules may be removed and inserted into the frame while it is powered. When pulling a board while the frame is powered, there is usually no audible disturbance in the outputs of the other boards. However, when inserting a module into a hot frame, the inrush currents that charge the power supply filter caps produce a small “tick”, much like a scratch on a record. The amplitude of the tick is generally not considered objectionable.

3.4 Output Connections

As a distribution amplifier, the module has two sets of build out resistors that provide the five balanced stereo feeds. When viewed from the back of the module frame the left side of the connector has the right outputs and the right side of the connector has the left outputs. See Figure 3.0.

3.4.1 Direct Outs - BP-100

Additional outputs, such as the BP-100 QCP panel, may be added by use of the direct outputs These outputs are a low impedance source, therefore, build-out resistors must be added to provide isolation for however many additional outputs may be desired. (The resistor splits are conveniently on the rear of the BP-100.) No more than two outputs per channel can be shorted without excessive temperatures occurring. Therefore, due to the increasing possibility of multiple shorted outputs, we recommend that no more than 20 outputs total, or 15 additional sets of build out resistors be added to the direct outputs from an individual channel.

!!! Warning !!!

While the power amplifier stages are the same circuit design as found in the DA-101, the smaller heat sinks do not allow these stages to be used as speaker drivers.

The direct outputs for the left channel are the top two pins just below the Left input, one on the left side of the connector and one on the right side of the connector. The right channel direct outputs are just beneath the left channel outputs, and again, are on opposite sides of the connector. A three position Molex SL connector may be used horizontally to pick up the two pins from side to side, but please note that no connection is made at the center pin. Since there is no ground connection to the center pin, we recommend that one of the normal outputs have its buildout resistors removed and jumpers put in their place to create a direct output *with* a ground connection.

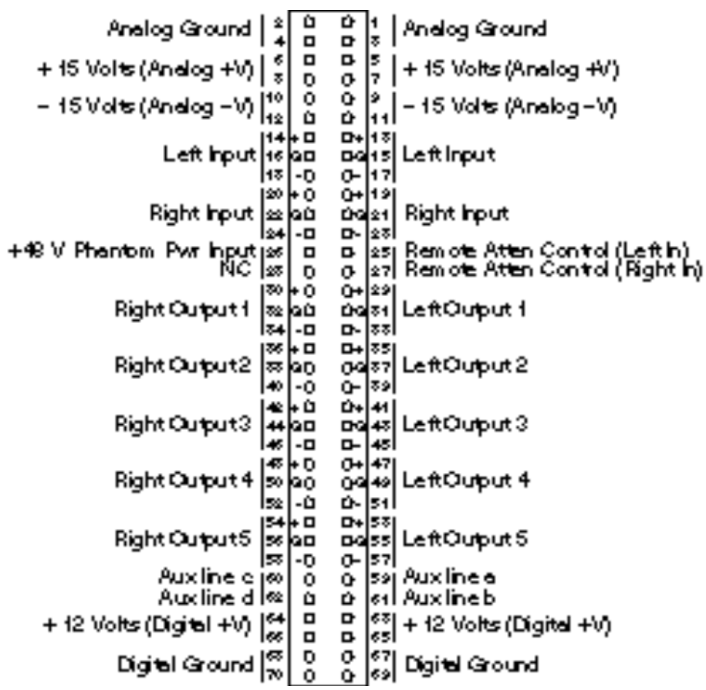

Fig 3.2 Card Edge Pin Assignments

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. If you are using the AMPMODU connectors that were previously sold by Benchmark, care should be taken when putting the connectors on the 0.025" square posts, that the connectors are not forced to travel further than what would be a comfortable seating. These connectors are not designed to go all the way to the bottom of the wire wrap pins. Forcing them further than they were designed to travel will cause physical damage to them and result in intermittent connections. This problem no longer exists with the Molex SL pins and housings.

3.6 Setup and Operation
The following is the setup procedure.

As with any piece of electronic equipment, the greatest operational performance may only be obtained by a thorough knowledge of the design of the module. Hence, we recommend studying the circuit description in section 4.0.

Input and output connections are made by way of three different methods. The first and most highly recommended is by way of the MF-300MLX pluggable rear mother-board card frame. The MF-300MLX has 16 individual three-position latching and polarized Molex® SL connectors per module position. This allows the convenience of plugging in additional outputs to modules that are “on the air.” The MF-300MLX is intended for line level and mic-pre DA applications. The output of the card frame may be taken to a BP-100 QCP patch panel for easy punch down to house wiring. Secondly, connections may be made by way of an SIB-70 interface module. The SIB-70 is a pluggable connector strip module that plugs into the rear of the MF-300 card frame. It has EuroStyle barrier strips for easy interconnection. Please contact the factory for additional information on these products. Lastly, you may plug directly onto the card edge pins of the card frame. While the connections are reliable and gas tight, it is difficult to make correct pin number identification with this method. Additionally, the wires must be carefully dressed and tied off to the wire bars of the frame to prevent an accidental removal of a connector.

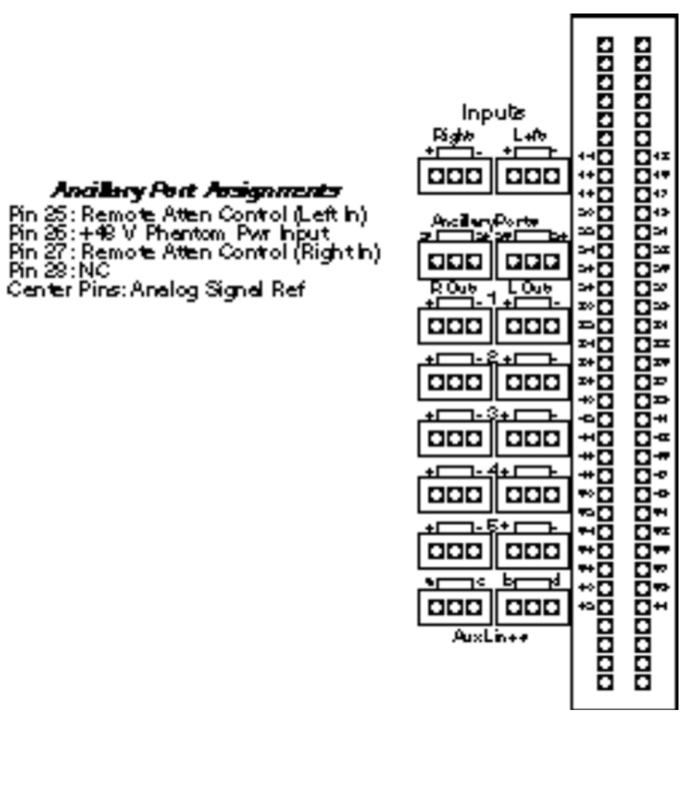

Fig. 3.1 [One of Twelve] MF-300MLX Connections

After making the proper input and output connections to the module position at the card frame, the signal flow must be set via the four position DIP switch at the center of the card. See figure 3.2

When visualizing the signal flow through the board, (viewing the board from the component side and with the LEDs on your left) it is well to remember that the first set of components, (left to right, two heat sinks and associated parts) is the left channel and the second set of components is the right (see the electrical schematic, drawing number 450133 and the component assembly, drawing number 250003). Using the DIP switch, S2401, switch position 1 routes the signal from the left input into the left power amp, switch position 2 routes the signal from the right input into the left power amp, switch position 3 routes the signal from the left input into the right power amp, and switch position 4 routes the signal from the right input into the right power amp. Therefore, the normal switch configuration for STEREO operation would be with position 1 on, 2 off, 3 off, and 4 on. For a MONO mixed output, close switch positions 2 and 3 as well. Operation with a daughter board requires all switch sections of S2401 to be in the *off* position. Signal routing is handled on the daughter board.

Most often, modules will be used as unity gain amplifiers. Switches S1301 and S4201 provide selection between fixed unity gain operation, where the front gain controls are inactive (and therefore not adjustable by non-authorized personnel), and variable gain operation. The top variable control is the left channel; the lower one is the right channel.

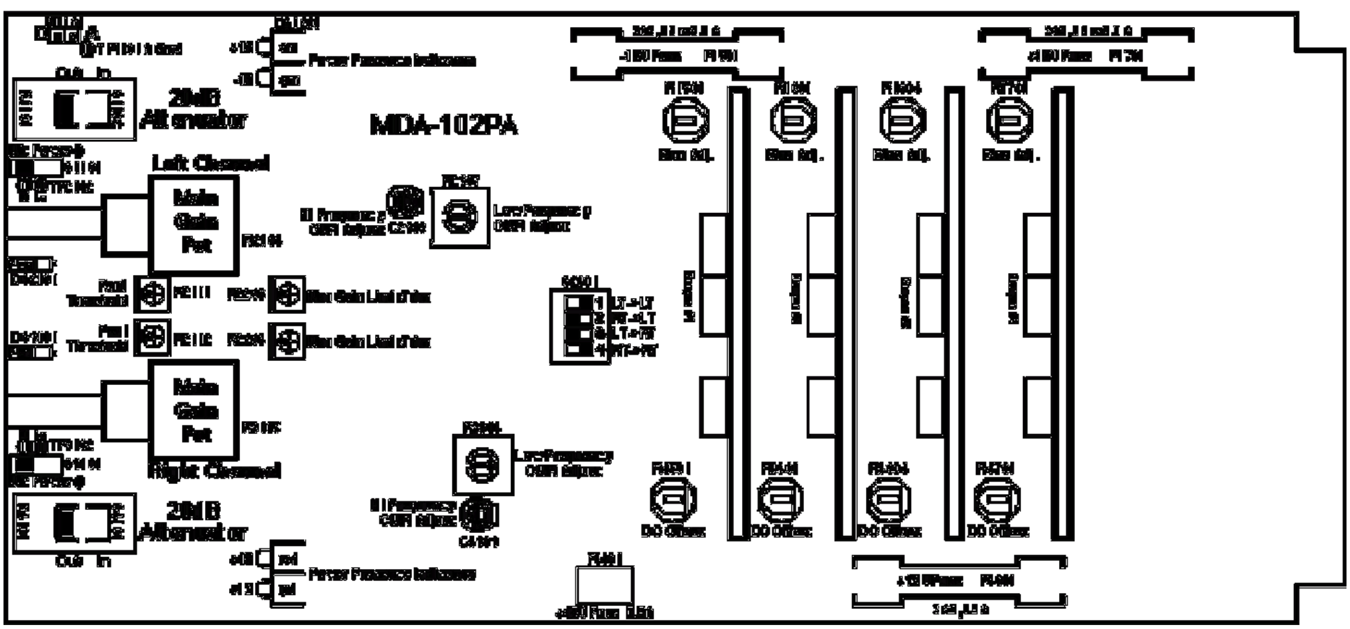

Figure 3.2 MDA-102/PA Controls

!!! Warning !!!

When using the MDA-102/PA with the MTX-02 Stereo Control daughter board, both channels must be in either the fixed unity gain position, or the variable gain position at or very near the same gain setting. Operating the gains otherwise will cause a phase difference between the signals coming into the daughter board and significantly deteriorate the L-R signal from its -90 dB null.

4.0 Specifications

In the area of common mode rejection, the MDA-102/PA 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-102/PAs ultra low intermodulation distortion.

5.0 Operation

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

5.1 General Operation

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

5.1.1 Phantom Power

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/PA 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.

5.1.2 Setting Levels

The setting of the amplification of the two channels of the MDA-102/PA 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/PA 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/PA.

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/PA, 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:

$$e_n = \sqrt{4\ kTRB} \qquad [5.1]$$

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.

$$\text{dBu} = 20 \log \frac{V}{0.7746} \qquad [5.2]$$

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/PA preamplifier’s performance under these conditions is approximately -80.0 dBu. At their minimum amplification (18 dB), each MDA-102/PA 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/PA 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/PA 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/PA 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/PA will be approximately -72 dBu.

$$-130 \text{ dBu} + 58 \text{ dB} = -72 \text{ dBu}. \qquad [5.3]$$

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.

$$+4 \text{ dBu} - (-72 \text{ dBu}) = 76 \text{ dB}. \qquad [5.4]$$

The dynamic range under these conditions is 99 dB.

$$+27 \text{ dBu} - (-72 \text{ dBu}) = 99 \text{ dB} \quad [5.5]$$

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 [5.6]

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] [5.7]

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/PA 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/PA 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 output before clip] [5.8]

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.

$$+4 \text{ dBu} - (-93 \text{ dBu}) = 97 \text{ dB} \quad [5.9]$$

and a dynamic range of 120 dB.

$$+27 \text{ dBu} - (-93 \text{ dBu}) = 120 \text{ dB} \quad [5.10]$$

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.

$$+4 \text{ dBu} - (-11.82 \text{ dBu}) = 15.82 \text{ dB [above the +94 dB SPL]}$$

$$15.82 \text{ dB} + 94 \text{ dB SPL} = 109.82 \text{ dB SPL [for +4 dBu out of preamp]} \quad [5.11]$$

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/PA preamplifier which is adjacent to the recorder. What is the <u>lowest</u> <u>peak</u> 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/PA to achieve this performance.

Solution
Since the output of the MDA-102/PA 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/PA 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

$$+21 \text{ dBu} - 95 \text{ dB} = -74 \text{ dBu}. \quad [5.12]$$

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

$$-74 - (-111.82) = 37.82 \text{ dB} \quad [5.13]$$

If both the recorder and the MDA-102/PA 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-102/PAs output noise voltage 3 dB lower than the noise voltage of the recorder. Thus a 35 dB amplification factor is selected. Therefore the peak <u>input</u> clip is

$$+21 \text{ dBu} - 35 \text{ dB} = -14 \text{ dBu} \quad [5.14]$$

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,

$$-14 \text{ dBu} - (-29.82 \text{ dBu}) = 15.82 \text{ dB} \qquad [5.15]$$

That is,

$$94 \text{ dB SPL} + 15.82 \text{ dB} = 109.82 \text{ dB SPL [system clip point]} \qquad [5.16]$$

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

$$109.8 \text{ dB SPL [system clip]} - 95 \text{ dB [dyn range]} = 14.8 \text{ dB SPL} \qquad [5.17]$$

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, <u>very</u> 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,

$$+21 \text{ dBu} - (-7.78 \text{ dBu}) = 28.78 \text{ dB} \qquad [5.18]$$

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

$$109.82 \text{ dB SPL} - 28.78 \text{ dB} = 81.04 \text{ dB SPL.} \qquad [5.19]$$

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 acoustic 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/PA 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 musical 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/PA Preamplifier Circuit Description

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

6.1 General

The superior performance of the MDA-102/PA is the result of careful attention to each element of the circuit design, particularly 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. This provides better RF immunity than is available with a transformer, 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 x.x the MDA-102/PA 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.

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/PA.

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/PA, 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.

.4 The Signal Routing Switch

The output of the gain stages feeds the signal routing switch. This switch and the daughter boards are some of the elements that give the Benchmark System 1000 modules their outstanding versatility. Normal operation of the MDA-102/PA module switch is outlined in Section 2.4.

.5 The Power Amplifier

The power amplifier stage consists of an operational amplifier, which in turn drives a power-current boost stage. The current boost stage uses complimentary symmetry topology with power transistors that have a rating of 50 watts and an f_t of 50 MHz. The driver transistors for the Darlington pair are 200 MHz transistors. This provides 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 output transistors have 0.27 ohm emitter ballast resistors which reduce thermal instability. These resistors also provide a convenient point to measure the quiescent current in the output stage. The stage is adjusted to allow approximately 13 mA of quiescent current to flow, to minimize crossover distortion. Drive current for the output stage is provided by the active current source/current sink combination. These current sources are held at the proper bias voltage by the V_{be} multiplier. This multiplier circuit exhibits a constant voltage from collector to emitter. The voltage is the ratio of the resistors in the voltage divider string, a ratio of approximately 2.7, times the base-emitter voltage of the transistor. This voltage is made to vary by the temperature of the output transistors as a negative feedback factor, to maintain quiescent current stability in the output stage.

The outputs of the current boost stage feed an L/R/C output stabilization network, a type recommended by Neivell Theil of Australia. The major advantage of this network over others is that the capacitor is directly across the direct output, and acts as a shunt for pickup of RF power by output wiring. The current boosted outputs feed two sets of ten 30 ohm build-out resistors, for five stereo balanced outputs. The compensation capacitors (across the feedback resistors) have been chosen for a nominal cutoff frequency of ≈300 kHz. This allows the overall bandwidth of the module to be ≈150 kHz.

.6 Peak Indicator Stage

The peak indicator is a half wave detecting comparator with AC-coupled feedback. It has the feature of being able to monitor the levels of a number of circuit points at once. When any of the monitor points exceeds a predetermined threshold, the comparator trips and begins to oscillate. The trip point is determined by the input resistor string. When the inverting input signal rises above ground potential, the comparator trips. R2102 and R1304 are the calibration trims for the peak indicators. They have a range of approximately +16 to +26 dBu. The factory calibration point is +20 dBu, unless otherwise requested.

This completes the MDA-102/PA circuit description.

.0 Service and Calibration

.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.

.1 Circuit Board De-Soldering

Printed circuit boards are *very* 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 Hakko 470. The proper technique with these stations is to apply the de-soldering tip to the area to be un-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. When the solder there has become liquid, apply the vacuum while moving the hollow tip and 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.

.2 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.

.3 Power Amplifier Bias Calibration

Troubleshooting and testing should be done with current limited power supplies. The bias set potentiometers should be initially adjusted to mid position, if any part of the power amplifier sections have been replaced. When turning on the power, the average current should not exceed 200 mA, and will more typically be about 100 mA. A millivolt meter, such as the Fluke 8050A, should be connected between the emitters of the power transistors, across the two emitter ballast resistors, by connecting the probes to the tops of the vertical 0.27 ohm resistors of each power amplifier. Adjust the bias trim resistor until approximately 7 millivolts is dropped across the two resistors. This establishes the normal quiescent bias current at approximately 13 mA. If the unit fails to exhibit control over bias current, it is possible that the V_{be} multiplier has shorted or a solder bridge may exist at that point on the board.

If the unit fails to have a current of 200 mA or less at power up and cannot be adjusted down to that current level, the following procedure should be used in troubleshooting.

Other than a direct short between the + and - power rails, the only possible current path that will allow high currents to flow is through one of the power amplifier stages. It is often possible to find the offending power amp section by comparing the temperatures of the individual heat sinks.

Barring solder problems, the only reason for high current drain is a defective device which must be located and replaced. The best way to test the transistors, using an ohmmeter, is by removing the devices from the board. The collector of the power transistor is the center pin. The emitter is the right pin, while the base is the left. With the small signal transistors, the device should be held with the flat side facing the technician and the leads down: in this position the pin-out is, from left to right, emitter, base, collector.

Before removing any transistors for testing, a few other checks can be made that will narrow the fault window. An open V_{be} multiplier will allow all of the drive current to pass into the bases of the darlington transistors that drive the power output transistors, and will turn them on hard. This can be checked by shorting out the Vbe multiplier, collector to emitter. If the device is open or the required bias voltage has changed, the current drain should drop.

.4 Common Mode Rejection Null

The common mode rejection trims should never need to be readjusted once they have been set at the factory. This is a passive bridge, and normally the characteristics of the operational amplifier used do not affect the accuracy of balance on this bridge. When replacing the operational amplifier, therefore, we strongly recommend that you measure the common mode rejection *before* making any adjustments to those trims.

The process of nulling the common mode rejection must be done with the gain network selection switch in the unity gain position.

1. Feed an unbalanced signal with a level of +10 dBu, referenced to ground, into the inputs of the channel being adjusted. This signal must be exactly the same on both inputs. This is best achieved by using an oscillator with a single ended output, tying the ± inputs together and, in turn, to the single ended output.

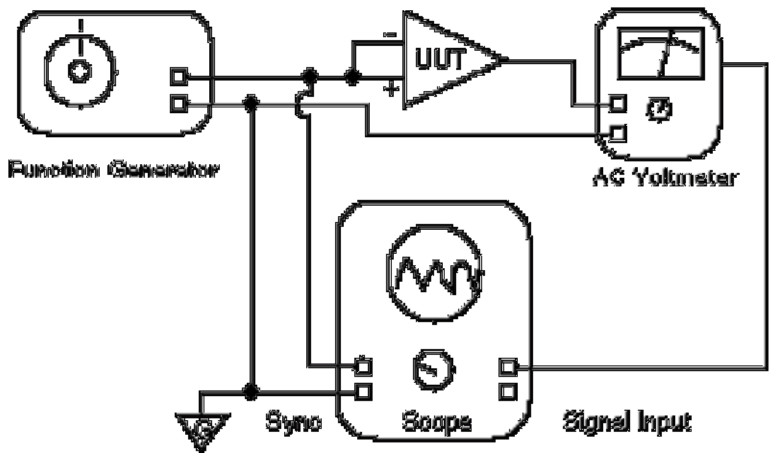

Fig. 5.4 Common Mode Test Setup

2. Send a 2 kHz Signal to the input and adjust the resistive portion of the diff-amp bridge for a minimum output. Use either a logarithmic level meter with a sensitivity down below -100 dBu, such as the Audio Precision System One, or a very sensitive linear meter, such as the Amber 3501 distortion and noise meter. Once a minimum on the resistive trim has been achieved, null the capacitive trim. Two or three iterations between these trims should be sufficient to achieve the best broadband null possible. A null of better than 100 dB at 200 Hz and typically 109 dB (-99 dBu), and better than 75 dB at 20 kHz, typically 80 db (-70 dBu), usually is achievable with the current P.C. layout.

.5 Bar Graph Meter

Troubleshooting the bar graph meter is quite straightforward. The LEDs are arranged to work in groups of four and are turned on by successive current sinks within the chip. First, LED 1's current sink turns on, then LED 2's sink turns on and the sink for LED 1 turns off, placing the two LEDs in series, thus reducing the internal power dissipation within the driver chip. This continues until a group of four have been turned on by the last current sink in the string, and about 9.6 volts is across the LED string. Then, the next group of four starts with the same process. Troubleshooting the LED string is easy, once you recognize that the string is turned on by its last activated current source. For example, if the middle four LEDs in the meter are extinguished when they should be on, if is safe to say that one of the four devices is open or is mounted backwards. You can find the offending diode by shorting successive diodes one at a time in the string of four. When you get to the problem device, shorting it will light the rest of the string.

The first two LED strings operate at about twice the current of the last string. Therefore, the last green LED and the three yellow LEDs are high output devices to compensate for the lower current. The philosophy behind the design of the chip is that the user would have red LEDs in the last positions, which are much more efficient than either green or yellow LEDs, and thus the higher current isn't necessary.

The time constants necessary to approximate a VU meter action are set by the parallel 10 µF capacitor and 10 kΩ resistor connected between pin four of the meter chip and ground. The current through the LEDs is set by the resistor from pin 2 to ground. Audio comes into the chip on pin number 3. The meter amplifier is a standard inverting amplifier with the calibration potentiometer as a part of the feedback network. The resistors are set up for a calibration range of -5 dBu to +10 dBu. The calibration potentiometer is adjusted, so that, with the desired system reference level coming out of the board, the first yellow LED just comes on.

The peak comparator, as described above, is an oscillating comparator by virtue of the fact that AC coupled hysteresis is applied around the device. The diodes form an analog OR circuit. Initially, the output voltage of the comparator is near the + supply voltage, in the off state. The comparator is held in the OFF state by the bias that is applied to the inverting input until an input peak overcomes the preset bias. When the comparator trips, the output voltage swings to the opposite supply rail. The 0.1 µF capacitor, in turn, pulls the noninverting input negative, holding the comparator in the ON state. The capacitor recharges with opposite polarity through the two 220 kΩ resistors and, when the threshold is passed, the device turns off, and is again held off by the charge on the capacitor, until the capacitor recharges to its original state. The action of this circuit is as a pulse stretcher, which allows the operator to “see” very short peaks as they occur.

This completes the MDA-102/PA service instructions. 052097

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