BEFORE PROCEEDING WITH COMPLETE UNPACKING AND SETUP, CONSULT UNPACKING AND INSPECTION INSTRUCTIONS ON PAGE 4

model 100-A
SONIPULSE®
ACOUSTICAL AUDIO SYSTEM ANALYZER

UREI

United Recording Electronics Industries

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UREI MODEL 100-A INSTRUCTION MANUAL

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SECTION I

INTRODUCTION

1.1 DESCRIPTION

The Model 110-A SONIPULSE® (Patent No. 3.732.370) is a precision portable instrument utilizing a revolutionary approach to the problem of obtaining accurate frequency response measurements of a loudspeaker system, including its electronic components and acoustic environment. Entirely self-contained in its attractive metal attache case, the precision 100-A offers ummatched convenience of operation, combined with portability and economy. Using the SONIPULSE and suitable 1/3-octave filters, such as the UREI Model 539 Room Equalizer, a speaker system and its acoustical environment can be equalized in 30 minutes or less.

Repeatable accuracy and stable meter readings are assured because the SONIPULSE system employs a coherent signal source, rather than white or pink noise. The power generated in a given bandwidth remains constant and precise, so readings are instantaneous, there is no need for "eyeball integration" or averaging of fluctuating readings.

While your Model 100-A can be used for frequency response measurement of any audio device from 40 Hz to 16 kHz, its pulse technique is especially suitable for systems including electro-acoustical transducers: microphones and loudspeakers. Typical of such applications are:

Sound playback systems.
Sound reinforcement systems.
Sound monitoring systems.
Manufacturing quality control of loudspeakers.
Manufacturing quality control of microphones.
Acoustic absorption and transmission properties of materials.

Measurements (and correction) of the frequency response of a sound playback system, monitoring system, or of loudspeakers are performed with a SONIPULSE using a calibrated microphone with flat response, such as the AKG 451-E condenser microphone supplied as an optional accessory to the 100-A. In measuring and equalizing sound reinforcement systems, the microphone which is part of that system is frequently used with the SONIPULSE so that its response anomalies are also measured and may be included in the corrective equalization. For measurements of microphone response, such parameters as on-axis or off-axis characteristics may be compared using the calibrated microphone as a standard.

The Model 100-A is simple to operate. However the successful application can only be achieved if the user has a thorough understanding of the system. We therefore recommend carefully reading this manual.

1.2 ELECTRICAL SPECIFICATIONS

FREQUENCY RANGE:	From 40 Hz to 16 kHz.
ACCURACY:	±1 dB.
MEASUREMENT RANGE:	More than 100 dB; 20 dB display range.
INPUT GAIN:	Continuously variable.
INPUT IMPEDANCE:	Balanced and bridging, 50 to 250 ohm microphone source.
INPUT SELECTOR:	Flat and Calibrated for microphone frequency response.
MIC PREAMP OUTPUT:	5 volts rms (600 ohms).
BANDPASS FILTER:	27 contiguous channels with center frequencies at 1/3-octave increments from 40 Hz to 16 kHz according to ISO.

$$Q = \frac{f_o}{f_2 - f_1} = 4.3$$

BANDPASS OUTPUT:	1 volt rms (600 ohms).
METER:	Precision taut band movement, calibrated linearly in decibels from -10 dB to +10 dB.
METER INDICATION:	Signal level measured in the selected 1/3-octave filter channel.
METER RESOLUTION:	0.5 dB.
DISPLAY RANGE:	20 dB, can be shifted continuously over entire measurement range.
INPUT OVERLOAD INDICATOR:	LED Readout in the face of the meter scale indicating overload peaks as brief as 50 us.
SIGNAL SOURCE:	Full spectrum burst with 5 Hz repetition rate.
OUTPUT SIGNAL LEVEL:	From -50 to +4 dB (Ref. 0.775 V).
OUTPUT SOURCE IMPEDANCE:	600 ohms.

INTERNAL CALIBRATION:	Receive Response selector may be switched to internal precision reference signal to check the instrument's response and accuracy.
POWER REQUIREMENTS:	115/230 V AC, ±15%, 50/60 Hz, less than 10 W.
ENVIRONMENT:	Operating temperature from 0°C to +50°C. Storage temperature from -20°C to +60°C.

1.3 CONTROLS : Receive Response selector

Receive Attenuator and Trimmer

10-position Frequency Selector

Frequency Range

Send Level control

1.4 PHYSICAL SPECIFICATIONS

CABINET:	Rugged aluminum-covered carrying case. Contains additional space for microphone, cables, graph paper, operating manual, etc.
DIMENSIONS:	16.5 x 38 x 33 cm (6.5 x 15 x 13 inches).
WEIGHT:	5.90 kg (13 pounds).
SHIPPING WEIGHT:	10 kg (22 pounds).

1.5 CONNECTIONS

SEND SIGNAL OUTPUT:	T.R.S. jack.
MICROPHONE INPUT:	XLR-3 connector with 18 volt DC phantom supply for condenser microphones.
MIC PREAMP OUTPUT:	BNC connector.
BANDPASS OUTPUT:	BNC connector.

1.6 ACCESSORIES

Condenser microphone "AKG", consisting of C-451 E Module amplifier with especially selected and calibrated CK-2 omni-directional capsule. Flat response characteristic. 18 V DC "phantom" polarizing voltage is supplied through the SONIPULSE XLR-3 connector. (Includes plastic stand, adaptor and 25' cable with XLR-3 connectors.)

SECTION II

INSPECTION AND SETUP PROCEDURE

2.1 UNPACKING AND INSPECTION

Your Model 100-A was carefully packed at the factory, and the container was designed to protect the unit from rough handling. Nevertheless, we recommend careful examination of the shipping carton and its contents for any sign of physical damage which could have occurred in transit.

If damage is evident, do not destroy any of the packing material or the carton, and immediately notify the carrier of a possible claim for damage. Shipping claims must be made by the consignee.

The shipment should include:

Model 100-A SONIPULSE

UREI Instruction Manual (this book)

Two-part Warranty Card bearing the same serial number as the Model 100-A.

If ordered, the accessory 100-M Microphone Kit, consisting of: AKG C-451 E Preamplifier with CK-2 Omni-Condenser capsule, 25' cable with XLR connectors, plastic stand adapter, and calibration curve matching the microphone serial #. Check your order and the packing slip to verify these optional accessories.

2.2 ENVIRONMENTAL CONSIDERATIONS

The system will operate satisfactorily over a range of ambient temperatures from 0°C to +50°C (+32°F to 122°F), and up to 80% relative humidity.

While circuitry susceptible to hum pickup is sufficiently shielded from moderate electomagnetic fields, avoid using the SONIPULSE immediately adjacent to large power transformers, motors, etc.

2.3 POWERING

The 100-A may be operated from either 100-125 VAC or 200-250 VAC mains (50 or 60 Hz, single phase.) As indicated in Section 2.4, the nominal line voltage may be selected with a rear panel switch. BE SURE TO VERIFY BOTH THE ACTUAL LINE VOLTAGE, AND THE SETTING OF THE VOLTAGE SELECTOR SWITCH BEFORE CONNECTING THE 100-A TO THE MAINS.

To comply with most Electrical Codes, the 100-A is supplied with a three-wire AC cord, the grounding pin of which is connected to the chassis. Sometimes this may create ground-loop problems. If ground problems are experienced, check for the possibility of ground loops by using a 3-prong to 2-prong AC adapter. This ungrounds the SONIPULSE, and may cure the problem, but is not a substitute for proper grounding. Be aware that unless the Model 100-A is properly grounded, a safety hazard can exist. UREI accepts no responsibility for legal actions or for direct, incidental or consequential damages that may result from violation of any electrical codes.

2.4 LINE VOLTAGE SWITCH

Unless a tag on the line cord specifies otherwise, the SONIPULSE was shipped ready for operation with nominal 115 VAC power mains. In order to change this for nominal 230 V (50 or 60 Hz), slide the VOLTAGE SELECTOR switch to the 230 position. This switch is located adjacent to the fuse post and line cord entry in the accessory storage compartment. The voltage is visible in a window next to the switch slot. Be sure to change the fuse to the correct value: 1/8-amp slo-blo when changing to 230 V operation or 1/4-amp slo-blo for 115 V operation. A small screwdriver may be used to move the recessed switch.

No change is required for alternate 50 or 60 Hz supply, so long as the mains voltage is correct. The pulse frequency and gating circuits automatically lock in on either mains frequency.

2.5 EXTERNAL CONNECTIONS

2.5.1 SEND SIGNAL OUTPUT CONNECTOR: A tip-ring-sleeve jack on the front panel is marked SEND and provides for connection of the 100-A to the system under test. The output impedance is 600 ohms, and the signal level may be adjusted with the SEND LEVEL potentiometer from -50 to +4 dBm. Connections are:

TIP = Signal ±.

RING = Signal common (ground).

SLEEVE = Chassis ground.

Either a tip-ring-sleeve (ADC PJ-051 or equivalent) plug or tip-sleeve (phone) plug may be used. If the latter is used, the shield and signal common will connect to the sleeve. When using tip-ring-sleeve plugs, the best grounding practice is to connect the shield only to the sleeve, and signal common (low side) to the ring. This is especially preferable when the SONIPULSE is feeding a balanced input.

2.5.2 MICROPHONE INPUT: The RECEIVE input is a female XLR-3 with the following connections:

Pin 1 = Ground.

Pin 2 = Common.

Pin 3 = Signal ±.

In addition, pins (2) and (3) supply the phantom +18 V DC and pin (1) the ground to power the condenser microphone preamplifier (The voltage source resistors each have a value of 4.12 kohms). The microphone input is balanced, and will accept all low impedance (50 to 250 ohms) microphones, including dynamic and ribbon types. Because of the phantom DC voltage, however, unbalanced microphones <u>cannot</u> be used. Care should be taken to avoid grounding pins (2) and (3).

2.5.3 MIC PREAMP OUTPUT: This front-panel BNC connector provides access to the amplified micophone signal. (See also Section 3.7.)

2.5.4 BANDPASS OUTPUT: The amplified microphone signal is processed in the filter section in 1/3-octave bands. The output signal corresponding to the frequency selector setting is available through this front panel BNC connector. (See also Section 3.8.)

SECTION III

OPERATING INSTRUCTIONS

3.1 GENERAL

Use of the SONIPULSE is simple and straightforward, once its operation is understood. All controls are self-explanatory, and the procedure is basically the same in all applications.

Feed the Send signal from the SONIPULSE into the input of the system under test; measure the acoustical reproduction of the signal through a microphone with the readout portion of the SONIPULSE.

3.2 VERIFYING PERFORMANCE

A unique self-test feature has been incorporated in your Model 100-A permitting verification of all internal functions without additional instrumentation.

3.2.1. CONTROL SETTINGS: Set RECEIVE RESPONSE switch to CHECK. This bypasses the microphone input, and applies a reference signal (derived from the internal send signal source) into the receive preamplifier automatically.

Set controls as follows:

FREQUENCY RANGE	=	X 10
FREQUENCY SELECT	=	100 (1000 Hz)
SEND LEVEL	=	Number 2 on scale
RECEIVE ATTENUATOR	=	-20

Adjust RECEIVE TRIM control to obtain 0 dB reading on the meter. Should the overload indicator flash, reduce the Send Level until the LED stops flashing, and increase the Receive Trim to obtain a "0" reading.

3.2.2 FREQUENCY RESPONSE CHECK: Switch through all 27 1/3-octave bands, using the Frequency Select and Frequency Range switches. If the reading remains within ±1 dB, a flat response of the SONIPULSE is assured within specified tolerances.

3.2.3 ATTENUATOR AND OVERLOAD INDICATOR CHECK:

Set controls as in Section 3.2.1.

Switch the RECEIVE ATTENUATOR to -10. The meter should indicate +10 dB (±0.5 dB).

Switch RECEIVE ATTENUATOR to -30. The meter should indicate -10 dB (±0.5 dB). (Note: In this position, some fluctuation of the meter pointer is normal).

Switch RECEIVE ATTENUATOR to 0. The overload indicator should flash. Reduce the test signal gradually by turning the RECEIVE TRIM control counterclockwise until the overload indicator stops flashing. The meter reading should be between +8 and +10.

This completes the check of your SONIPULSE and verifies all internal functions.

3.3 SEND SIGNAL LEVELS

The send signal level of the SONIPULSE may be varied from -50 to +4 dB (Ref. 0.775 V). Caution must be exercised to avoid overdriving the system to be tested. An oscilloscope can be used to detect clipping of the pulses in the amplifier. With the use of extremely powerful amplifiers, it is possible to damage speakers. This could be caused by the fast acceleration and large excursion of speaker diaphragms, particularly in high frequency speakers.

With a little practice, the change in the tonal characteristic of the reproduced pulses can be detected if clipping or other overload distortion is approached. The level should be reduced somewhat below this point to assure accurate measurements. As a general rule: the SONIPULSE signal level used in testing should not be greater than necessary to obtain adequate singal-to-noise ratio during the measurement. If the reading on the meter drops at least 10 dB when the test signal is disconnected, the signal level is adequate to ensure a reading accuracy of better than 0.5 dB. In the usual environments, this will be most critical at low frequencies.

To establish the correct signal level, the following procedure is suggested:

With minimum input attenuation, check at each 1/3-octave frequency position for the highest ambient noise reading on the SONIPUSLE meter. Feed the pulse signal into the system and adjust the signal level to obtain a reading which is at least 10 dB above the highest ambient noise reading.

3.4 MICROPHONE/RECEIVE RESPONSE

When using the accessory AKG C-451 E microphone supplied with the Model 100-A, always set the RECEIVE RESPONSE switch to the CALIBRATED position. The SONIPULSE has been factory equalized to provide a flat response from 40 Hz to 16 kHz (±1 dB).

Set the RECEIVE RESPONSE switch to the FLAT position when it is desired to include other microphones in the system to be measured and equalized. Any standard low impedance (50 to 250 ohms) microphone may be used.

For loudspeaker playback system evaluation, any good low impedance microphone can be used, provided it has been calibrated against a known standard, and its response anomalies algebraically subtracted from the system response plot. Set the RECEIVE RESPONSE switch to the FLAT position.

3.5 INPUT SENSITIVITY

Caution must be exercised to avoid overdriving the readout section, which will be indicated by a flashing of the red LED in the meter face. If this occurs, the input signal should be reduced by the RECEIVE ATTENUATOR or the RECEIVE TRIM control. Please note that the overload indicator input is ahead of the 1/3-octave bandpass filter set, so an overload at any frequency will trigger the LED, regardless of the positions of the Frequency Range and Frequency Select switches. An occasional flickering of the LED will not cause erroneous readings, as the threshold of the indicator is well below the clipping level of the amplifiers.

3.6 BANDPASS FILTER SWITCHES

Any of the 27 filter bands can be selected by rotating the Frequency Select switch to the desired significant number, and the Frequency Range switch to the proper multiplier.

Example, 1,000 Hz:

Set Frequency Select to 100

Set Frequency Range to X 10

NOTE: The very low frequencies (20, 25 and 31.5 Hz) are not applicable on the X 1 range. Although readings will be obtained on these settings if the system under test provides acoustical output at these frequencies, accuracy is not assured.

The Frequency Select switch rotates through 360°. Once a convenient reference "0 dB" level has been established (usually at some midrange frequency) the entire spectrum can be scanned quickly and graph points plotted as follows:

Set the Range switch to X 1, and the Frequency Select switch to 40.

Log the reading at 40 Hz, and then proceed clockwise with each successive frequency through 160 Hz.

Continue rotation of the Frequency Select switch to 20, switch Frequency Range to X 10. Log the reading at 200 Hz, and continue through 1,600 Hz.

Repeat the previous step with the Frequency Range switch at X 100 for frequencies from 2 kHz to 16 kHz.

3.7 PLOTTING A RESPONSE CURVE

After the send level has been established (see Section 3.3), a preliminary switching through the frequency range to be plotted will indicate the major peaks and dips in the response curve. The input sensitivity is then set to establish a convenient reference (for example, so that 1 kHz will read 0 dB on the meter). Now a step-by-step analysis of the response characteristic can be performed using the technique suggested in the previous Section, and measured values may be recorded on graph paper.

You will note that the vertical lines on the SONIPUSLE SYSTEM ANALYSIS graph paper are in standard 1/3-ocatve increments to simplify plotting (Reorder #100-G).

During equalization or when acoustical conditions are altered, a second graph may be plotted over the first for comparison, using a different color.

Due to the logarithmic characteristic of the meter indication, values reading -8 to -10 dB on the meter scale may fluctuate because of noise. To obtain a more stable reading, the input signal strength should be increased by switching the Receive Attenuator to the next lower position, and subtract 10 dB from the observed reading. Be sure to note the initial setting of the attenuator when beginning the plot and add or subtract the appropriate dB decades from readings obtained at any other settings.

Should a severe dip be encountered in the response at a given frequency, it may not be possible to obtain an on-scale reading at this frequency without the overload indicator flashing. A finding of this nature indicates a deficiency in response which is beyond correction by electronic equalization techniques: perhaps an inadquate or defect speaker, a phase reversal between two speaker elements in biamp or crossover-equipped system, or diaphragmatic absorption by the environment. (See also Section 3.10 ff).

3.8 MIC PREAMP OUTPUT

The front panel BNC connector provides access to the microphone signal after it has been amplified. This may be helpful whenever a low level mic signal requires amplification and when monitoring with an oscilloscope is desired. The maximum output level is approximately 5 volts rms, and the output impedance is 600 ohms. The overload LED in the meter indicates if the signal approaches overload.

3.9 BANDPASS OUTPUT

After the microphone signal is amplified, it is processed in the tunable 1/3-octave bandpass filter. Its output is available through the front panel BNC connector. If random noise is fed to the microphone, the bandpass provides a maximum output of 1 volt rms. This output is useful for monitoring the signal with a scope, for measurements of reverberation (with a suitable T_{60} display device), etc.

3.10 ON EQUALIZATION

3.10.1 GENERAL: The main reason for frequency response tailoring of audio systems is to compensate for response anomalies occurring in the system and its environment, and to adjust the system response curve to some desired shape. In some systems this means adjusting for best overall flatness of response through the entire audio range, and in other instances some different response shape is desirable.

Three important characteristics of audio systems may be improved by judicious application of corrective equalization techniques. They are:

(1) DISTORTION OF FREQUENCY RESPONSE BALANCE.
Sources of sound which may be amplified through a sound system start out with some specific frequency bands. If, in being processed through an audio system this balance is disturbed by variation in frequency response, then truly the sound has been distorted. For example, a clarinet has a specific characteristic with regard to the levels of the various harmonics compared to the fundamental tone. If these signals are processed through an audio system with non-linear frequency response, then the resultant sound is simply not the distinctive sound which that clarinet orignally produced.

(2) SUSCEPTIBILITY TO FEEDBACK.
In public address and sound reinforcement systems, the maximum acoustic gain that may be obtained for a microphone in the vicinity of a loudspeaker which is part of that system, will be determined by the positive feedback loop created when sound from the loudspeaker enters the microphone, reinforcing the signal level until the

system goes into oscillation. This positive feedback problem is aggravated if some frequencies in the audio range are reproduced at a higher level than the rest. If the level of these frequencies which are being reproduced at an exaggerated level can be reduced, then the sound system gain may be increased to some degree without feedback.

(3) INTELLIGIBILITY.
For an audience in an auditorium to understand a lecturer, or for a nightclub audience to hear the words which a vocalist is singing, there are two important requirements:

(a) The sound level of the source must be sufficiently loud as to be heard clearly and not masked by other sounds which may be present in the listening environment.

(b) The sound which the listener hears must be intelligible to him. It should not be distorted or garbled. There are a number of causes for poor intelligibility in a sound system. Among them are: amplifier or speaker distortion, frequency band masking, improper balance of direct and reverberant fields, and inadequate projection of the higher frequencies which are necessary for the recognition of words. The frequency band from approximately 1 kHz to 5 kHz is very important to the recognition of sibilant and consonant sounds, and if the sound system has inadequate output in this frequency band, or if other bands have considerably greater output which "cover up" the sounds in the intelligibility band, the words may simply not be understood. Equalization is very helpful to correct some of these problems.

The remainder of this section is concerned with a general description of the equalization process and some suggestions for successful application of equipment such as the UREI Model 539 Room Equalizer are given. Techniques are continuously being improved upon, therefore this discussion should be considered only as a starting point for those who are new to the field, and as a general review for those who are already proficient in the methods used.

Those interested in more complete information on the general subject of sound system design and corrective equalization are encouraged to study specialized literature* and/or participate in seminars dedicated to audio acoustics, etc.

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* Sound System Engineering, by Don and Carolyn Davis, Published by Howard W. Sams, Inc., Indianapolis, Indiana 46268, Copyright 1975.

An excellent series of continuing seminars are conducted in various cities by Don and Carolyn Davis. For further information, and a schedule of their regular seminars, contact:

"Synergetic Audio Concepts"
P.O. Box 1134
Tustin, CA 92680

3.10.2 BEFORE EQUALIZATION : Corrective equalization is only one of the solutions to the problems faced by designers or users of audio systems. Applied correctly, it is a powerful tool; but if it is assumed to be a cure-all without careful consideration of other possible alternatives, results may be disappointing. If the sound systems is not thoroughly checked out before equalization procedures are attempted, much time may be wasted. Therefore we feel that it is important to mention a few of the tests which should be made prior to the start of corrective room equalization. The list is not intended to be exhaustive, rather to point out that unless you know what the sound system IS doing, you may find it difficult (if not impossible) to make it do what it SHOULD do.

A PARTIAL CHECK-LIST OF PROCEDURES BEFORE EQUALIZATION:

(1) At normal operating levels, check the frequency response and the signal-to-noise and distortion characteristics of the individual components of the system and the system as a whole.

(2) Check phasing of signal lines from all inputs through to the amplifier outputs.

(3) Check loudspeakers to see that all units function, that there are no rattles due to loose screws or units poorly attached to walls. Check phasing of all loudspeakers, even of the several drivers within one cabinet. Check loudspeaker aiming.

(4) Check phasing of all microphones.

(5) Check that all electronic components of the system are connected correctly, that correct signal levels are observed, that loads are where they should be and not where they should not be. Check the system for oscillations, R.F. interference, and hum.

(6) Make sure you have a good system ground, and that shielding and grounding are done in accordance with good practices.

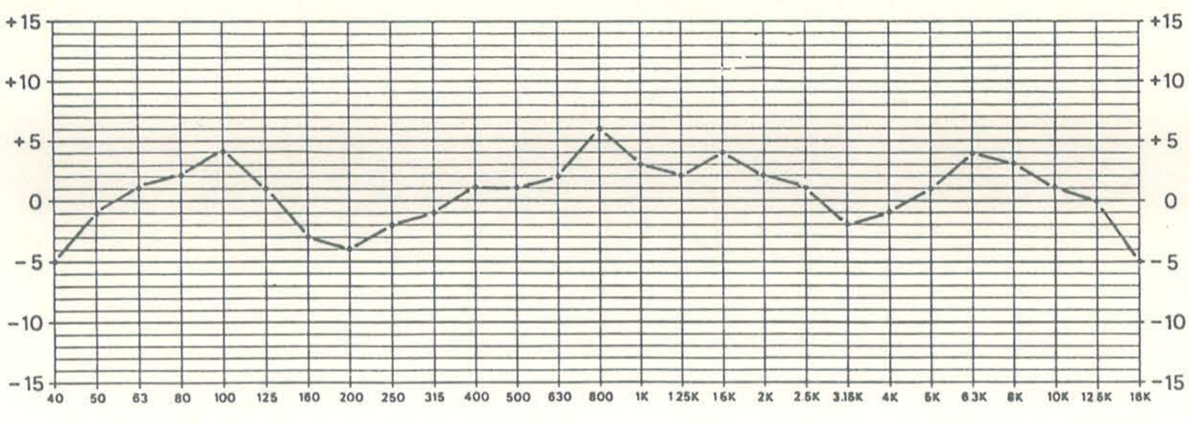
(7) Make sure that the AC electrical power to the system is of good quality, adequately protected, and that the wire size is sufficiently large to avoid excessive voltage drop when large amounts of power are drawn.

3.10.3 MEASUREMENT TECHNIQUES: The response measurement made with the microphone at a given location with respect to the sound source will be valid for that location only. If the acoustical conditions are altered, or the speaker is moved, different results may be obtained.

Therefore, response measurements of a monitoring system (such as in a recording studio control room) should be made with a mic at the same location the listener's head will occupy.

In a large listening area, the microphone position is changed to different locations and the response variations are averaged to make what is called the RAW HOUSE CURVE (see illustration). Corrective equalization procedures are based on this curve. Many problems associated with level balance between multiple speakers in large areas, acoustic deficiencies, and speaker aiming and locating faults will show up on the curves taken in various locations in the listening area. For good results, these faults should be corrected by re-balancing of levels and re-aiming or relocating speakers -- not by equalization. In systems which use electronic crossovers or multi-amplifier configurations, the balance should be checked and adjusted between low and high frequencies and between speakers covering different parts of the listening area.

3.10.4 ADJUSTMENT OF EQUALIZERS: Room equalization is performed as a series of approximations of control settings. Looking at the sample RAW HOUSE CURVE in the illustration, we can infer that if an inverse electrical response curve were placed in the audio chain that the result would be a flat frequency response.



TYPICAL RAW HOUSE CURVE OBTAINED FROM SONIPULSE® MEASUREMENTS.

3.10.5 OCTAVE OR 1/3-OCTAVE EQUALIZERS?: The octave graphic or 1/3-octave graphic equalizers are tools with which this inverse curve is approximated. Because there are combining effects between adjacent filters, it is necessary to make settings on the equalizer and then to re-check the room response to see if the adjusments were correct. Additional adjustments and measurements may be made until the desired results are obtained.

It is important to understand the difference between these equalization filters. If the device contains filters which are one octave wide (such as the single channel UREI Model 530 or the dual channel Model 532), the user is able to adjust for broad spectrum anomalies, which in some applications is absolutely sufficient. However, the 1/3-octave equalizer, enables a much more precise adjustment to correct the frequency response.

3.10.6 SOME PROBLEMS WITH CORRECTIVE ROOM EQ: As stated earlier, the process of corrective equalization of sound systems is not a cure-all. There are some rooms which, because of very long reverberation times and poor acoustic configuration, cannot be significantly improved. There are a few others in which the acoustics and the sound system are already so well balanced that there is little need for correction. It is for the majority of situations which lie between unsolvable and relatively perfect that corrective equalization is intended.

Many of the problems that people seem to have when performing corrective equalization stem from a tendency to "overkill." It has been found in practice that it is not desirable to attempt to equalize to the "last dB" all minor anomalies in the response of an acoustical system, as this often results in an artificial or processed sound. Also, it sometimes requires extreme filter settings with introduction of unwanted phase shift, which may cause more problems than the EQ solves. In general, if the final results are within a 3 dB envelope of the desired theoretical values, the equalization should be considered successful.

Attention should also be given to the fact that different frequencies are attenuated differently when traveling through air. As the distance between the sound source and the listening position (or position of the measuring microphone) increases, it is natural for high frequencies to be attenuated more. Therefore, if equalization is adjusted to create a perfectly "flat" response, particularly throughout a large room, the resulting sound will be excessively bright. A house curve which produces a natural response in sound systems is subjective and a matter of personal preference, as well as a function of the primary use of the location. However, a flat response up to about 3 kHz and then an attenuation of 3 dB per octave above 3 kHz is frequently used in sound reinforcement. In recording studios and monitoring rooms, the response is usually flat to higher frequencies.

Widely disparate settings of adjacent controls should be avoided if at all possible. A control panel in which adjacent adjustment knobs are 8 to 10 dB apart may look interesting, but the resultant sound will not be natural. It is good to remember that the trained human ear is still the ultimate judge of any equalization.

3.10.7 FEEDBACK SUPPRESSION: An improvement of a system's tendency to feedback should be attempted after the general equalization of the frequency response is performed. SLOWLY advance the gain until the feedback frequency becomes detectable and stabilize the feedback to a constant, comfortable level. (Caution: feedback is not only annoying to the ear, but it is also dangerous to unprotected amplifiers and loudspeakers!). The control which reduces the feedback with a minimum of additional attenuation is closest to the actual feedback frequency. Experimentation by trial and error, and the resulting experience, is the best method.

Advancing the gain further will cause the next feedback frequency to occur. This procedure may be repeated as often as it seems practical. Again, extreme filter settings should be avoided.

The result will be a higher amplifier gain setting than was possible before feedback suppression. Remember: an improvement of 3 dB is equal to twice the previously available power. To prevent ringing, it is best to adjust the gain at least 3 dB below the threshold of feedback.

NOTE: Where the budget allows the expense, we suggest adding a dedicated instrument such as the UREI Feedback Suppressor to the system. It is specifically designed to "tune-out" very narrow frequency bands with continuously variable filters without noticeably altering the program material.

3.10.8 LOW CUT AND HIGH CUT FILTERS AFTER CORRECTIVE EQ: Examination of the filter's frequency response characteristic after equalization adjustments have been made may show that the band ends (low and high frequency extremes) could overdrive the amplifier or speakers if the program material contains energy at these frequencies, i.e. microphone pops, etc. Adjust tunable low cut and high cut filters until the resulting system response curve looses its "bathtub" shape and the house curve rolls off smoothly at each end.

3.11 MEASUREMENT TECHNIQUES

Several methods differeng from the SONIPULSE are in use to measure the frequency response characteristic of a sound system in its acoustic environment:

Pink noise, measured in 1/3-octave bands and displayed on a real time spectrum analyzer or suitable LED-display.

Swept sine waves, measured on an audio voltmeter or plotted on a graphic recorder.

Swept sine waves, frequency modulated with a constant percentage bandwidth (Warble Tone) generated and plotted with the UREI 200/2000 Recorder system.

Other methods, as for example "Time Delay Spectrometry," are in the development stage and may become important alternate solutions in the future.

Basically, all of the measurement methods are similar in result -- differing only in hardware. They all involve generating a known quantity of audio signal, transmitting it through the sound system, and reading the results through a calibrated microphone and readout device. The results will closely resemble each other if comparative measurements are made, presuming correct application of the various methods.

SECTION IV

THEORY OF OPERATION

4.1 DESCRIPTION

The Model 100-A SONIPULSE contains a Send Section and a Receive Section with a meter readout and the necessary power supply. For the following explanation of the system, refer to the block diagram and schematics in the back of this Manual.

4.2 SEND SECTION

The Send Section generates discrete pulses which are used:

(a) To drive the system under test
(b) To gate various functions in the Receive Circuit.

In the 10 Hz Generator, a unijunction, generates a sawtooth waveform. The power line frequency of 50 Hz or 60 Hz is used to synchronize the generator output to 10 Hz. The signal triggers three successive one-shot multivibrators which produce the gating pulses for the Receive Circuit.

In the Frequency Divider an integrated circuit multivibrator divides the 10 Hz signal producing a 5 Hz square wave. This square wave is differentiated in a Pulse Shaping Network to suppress the 5 Hz fundamental, producing exponential pulses which are then passed through the Output Amplifier to the Send Level Attenuator.

4.3 SEND SIGNAL CHARACTERISTICS

The Send Signal exponential pulse retains the same upper harmonic content as the original square wave from which it is derived: that is, all odd upper harmonics of the fundamental frequency. The amplitudes of these harmonics decrease with increasing frequency: the third harmonic amplitude is 1/3 of the fundamental, the fifth is 1/5, the seventh is 1/7, etc. Therefore, the harmonics in each pulse are spaced every 10 Hz, decreasing in amplitude 6 dB per octave increase in frequency, or 20 dB per decade of frequency.

4.4 INPUT CIRCUIT

Through the XLR Receive Input connector, a positive phantom supply voltage is available for the condenser microphone preamplifiers, although dynamic, ribbon or other low impedance microphones may be used, provided a balanced condition is maintained between pins (2) and (3) with respect to pin (1), ground.

The incoming signal from the microphone is applied to the input transformer and the amplified in two stages.

Two Input characteristics are provided via the Receive Response switch:

> The FLAT position is used when the microphone is to be included in the system under test, or when a calibrated microphone of known response is used (under the latter condition, any known deviations from an ideal flat response must be added or subtracted from the plotted response curve of the system under test).
>
> The CALIBRATED position of the Receive Response switch provides equalization networks in feedback circuitry of the Preamp and Postamp. These networks are factory adjusted to compensate for response anomalies of the accessory AKG C-451 E Condenser Microphone, if that accessory was ordered with the Model 100-A at the time of factory delivery. The networks have a maximum range of ± 3 dB at 10 kHz, and +12 dB at 20 Hz.

The Receive Attenuator switch has steps of 0, -10, -20 and -30 dB, and is used for amplitude ranging of the received signal. The Trim Attenuator is continuous, and provides a means of accurately setting a reference level.

The amplified audio signal is measured for its peak-to-peak value in the Overload Indicator circuit. If the limit of 22 volts is exceeded (which is well below the clipping point of the amplifier), the LED in the SONIPULSE meter face will flash.

The output of the Mic Preamp is available at the front-panel BNC connector.

4.5 1/3-OCTAVE FILTER SECTION

The received signal is passed through a low impedance follower to avoid losses and hum pickup, then is applied to the input of the 1/3-octave tunable Bandpass Filter. Various R-C networks are switched by the RANGE and FREQUENCY switches to determine the frequency to which the active filter is tuned. Each of the 27 frequency passbands is 1/3-octave wide, and centered on a standard ISO and ASA 1/3-octave frequency.

The bandwidth is determined by the Q of the filter which is set to be 4.3 according to

$$Q = \frac{F\ center}{F_2 - F_1}$$

where F_1 and F_2 represent the skirt frequencies at the -3 dB points. The bandpass output is available at the front-panel BNC connector.

4.6 BLUEING FILTER

The Frequency Range is coupled to an attenuator switch to provide a scaled input amplitude to the Blueing Filter which follows. This filter is designed to partially compensate for the Send Signal harmonics' 6 dB/octave decrease in amplitude with increasing frequency (see Section 4.3). The gain of the Blueing Filter increases 3 dB/octave, or 10 dB/decade. An additional 3 dB/octave of compensation is automatic because the power contained within the 1/3-ocatave bands increases with frequency at a rate of 3 dB/octave; this occurs because each successive octave contains twice the number of harmonics as the one preceeding.

4.7 CONVERTER

The Converter consists of a squaring circuit, multiplier and integrator. At its input the filtered audio signal is now a tone burst with frequencies passed by the 1/3-octave filter. After passing through a gate, the signal is fed into a Multiplier IC. All negative excursions of the signal are converted to postive excursions and added to the positive half of the signal, thus doubling the frequency. An IC amplifier then inverts this 2f signal which passes through a noise gate into the Integrator. The Integrator output is a DC voltage varying with the amplitude of the 2f AC input signal.

4.8 NOISE SUPPRESSOR

In the intervals between signal bursts, inherent noise could cause the SONIPULSE meter to fluctuate. This is inhibited by a Noise Suppressor, consisting of an integrated circuit noise detector and associated noise gate, which allows only the discrete signal bursts to pass.

4.9 ERROR CORRECTION

The functioning of the Error Correction feedback loop depends upon the timing sequence of the gating pulses. During the first pulse, the DC output voltage from the Integrator is sampled and connected to the meter circuit. During the second pulse the Integrator capacitor is being discharged. The third pulse then switches the input of the previously described Multiplier to ground, effectively blocking the AC signal from the circuit. Simultaneously, the remaining error voltage (consisting of offsets, temperature drift, etc.) is applied to the Error Send Amplifier, whose output voltage is stored and periodically applied through a follower as correction voltage to the input of the Integrator.

4.10 METER CIRCUIT

The Meter Circuit has a gating input which samples the DC voltage from the Integrator output during the first gating pulse. The following IC amplifier, in conjunction with a capacitor at its input, forms a sample-and-hold circuit. A specially selected transistor in the feedback loop of the next IC causes a logarithmic amplification characteristic. In the last stage, the Meter Amplifier, the signal amplitude is calibrated to the individual meter.

4.11 POWER SUPPLY

The Power Supply is conservatively designed to prevent overload and excessive heat dissipation. Standard tight tolerance regulation is employed to ensure proper operation of the SONIPULSE over a wide range of supply voltages and ambient temperatures. The supply is protected against overload, short circuits, and reverse current and voltage conditions. An AC line fuse is provided, plus a conveniently located recessed slide switch permitting alternate operation from 110-120 or 220-240 V AC mains, 50 or 60 Hz.

SECTION V

MAINTENANCE

5.1 GENERAL

The Model 100-A is an all solid-state unit, ruggedly constructed with only the highest quality components. As such, it should provide years of trouble free use with normal care. All parts used are conservatively rated for their application, and workmanship meets the rigid standards you have learned to expect in UREI products.

NO SPECIAL PREVENTIVE MAINTENANCE IS REQUIRED.

However, the following suggestions should be considered to ensure reliability.

When transporting your SONIPULSE as hand-carried luggage, treat it with the same care you would give any sensitive instrumentation. While it will probably survive being checked as baggage aboard planes, busses or trains, it is better to carry the unit aboard unless packed in a suitable protective carton with shock-absorbing packing.

Pack the microphone carefully in the foam-lined storage compartment so it will not bounce around. While the AKG microphone is rugged, a calibrated microphone should be treated carefully. Save the mic calibration curve to which your SONIPULSE was calibrated (as provided with the accessory kit). Should the microhone ever require repair because of damage, return the calibration record with the microphone for factory recalibration.

5.2 SERVICE ADJUSTMENTS

These controls have been carefully set at the factory and should not require adjustments except after service work. The performance may be checked and verified according to the procedure described in Section 3.2).

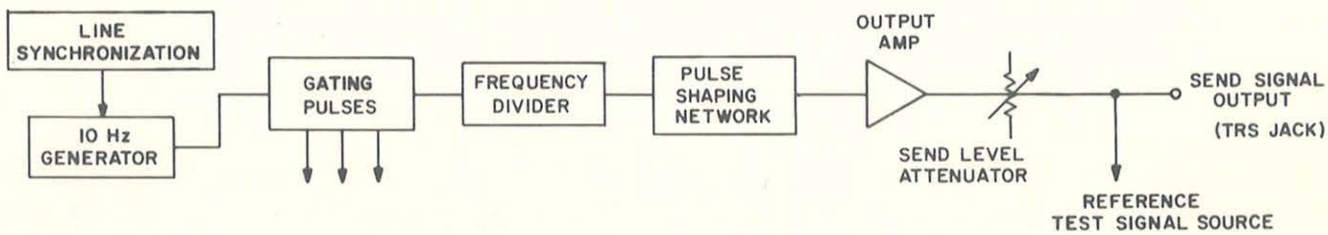
5.3 REPAIRS AND WARRANTY

The Model 100-A is factory warranted to the orignal purchaser against defects in material and workmanship for one year after initial purchase. This limited warranty must be activated at the time of purchase by returning the registry portion of the Warranty Card to the factory. Should a malfunction ever occur, the dealer from whom the unit was purchased will be glad to handle return for factory repair; alternately, for prompt service, ship the unit prepaid directly to the factory. Be sure it is well packed in a sturdy carton, with shock-absorbing material such as foam rubber, styrofoam pellets or "bubble-pack" completely filling the remaining space. Particular attention should be paid to protecting the microphone. Include a note describing the malfunction, and instructions for return. We will pay one-way return shipping costs on any in-warranty repair.

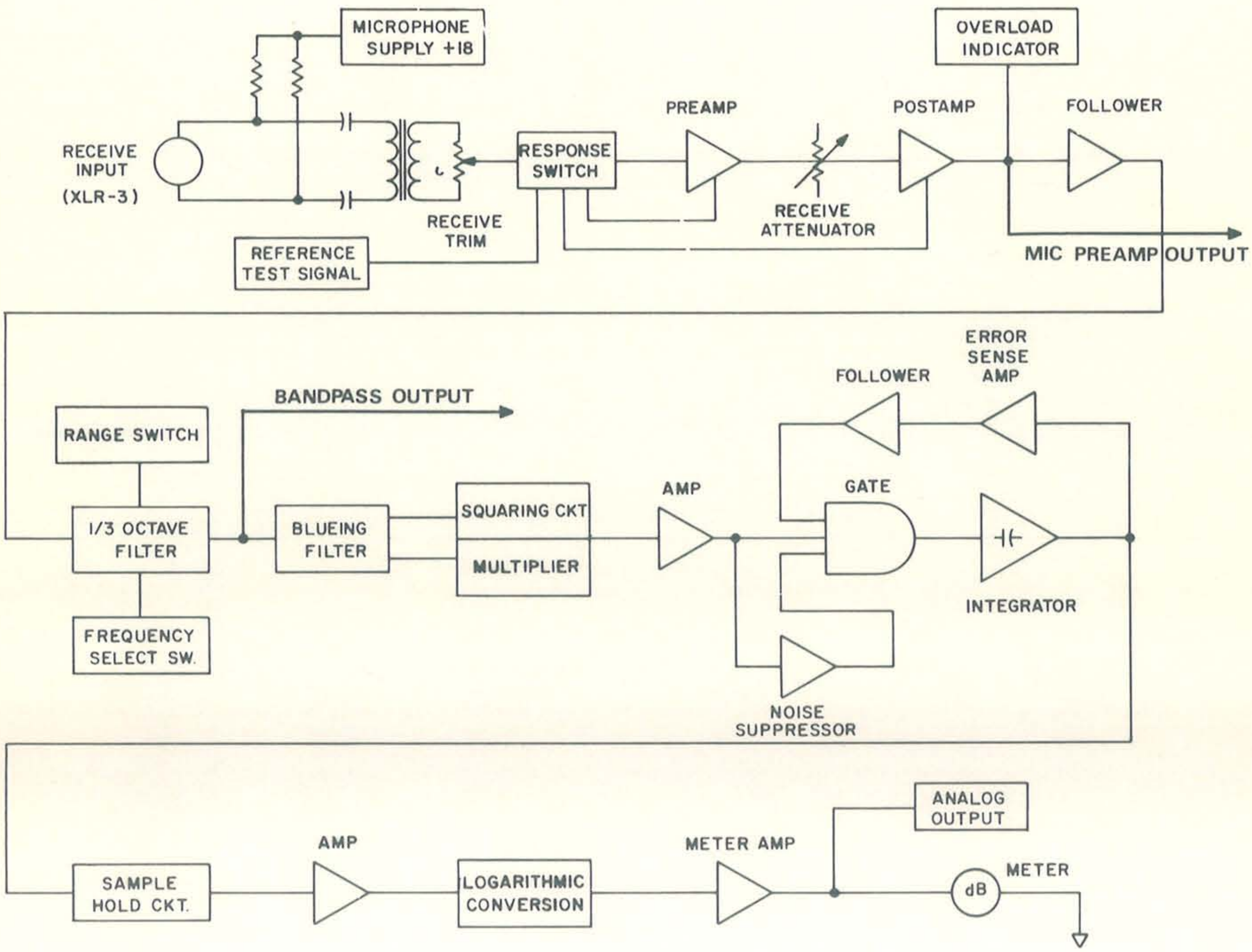
Field repairs are not authorized during the warranty period, and attempts to perform repairs may invalidate the warranty.

BLOCK DIAGRAM

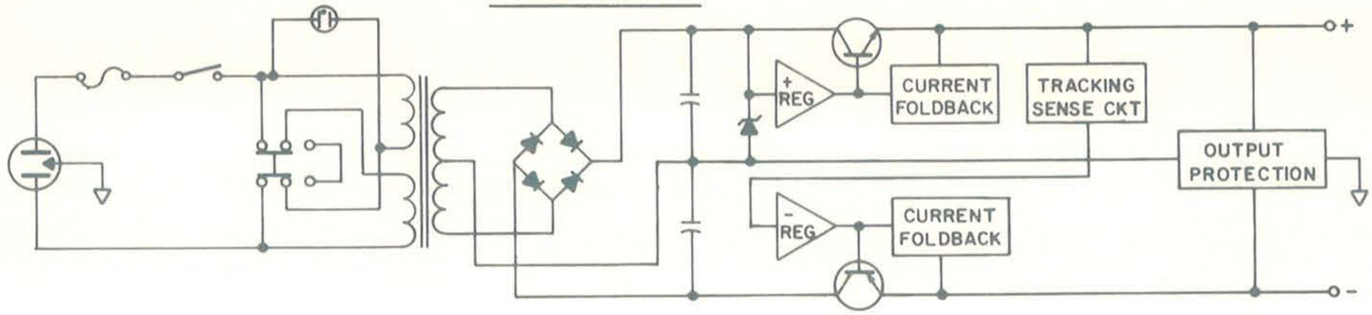
SEND SECTION

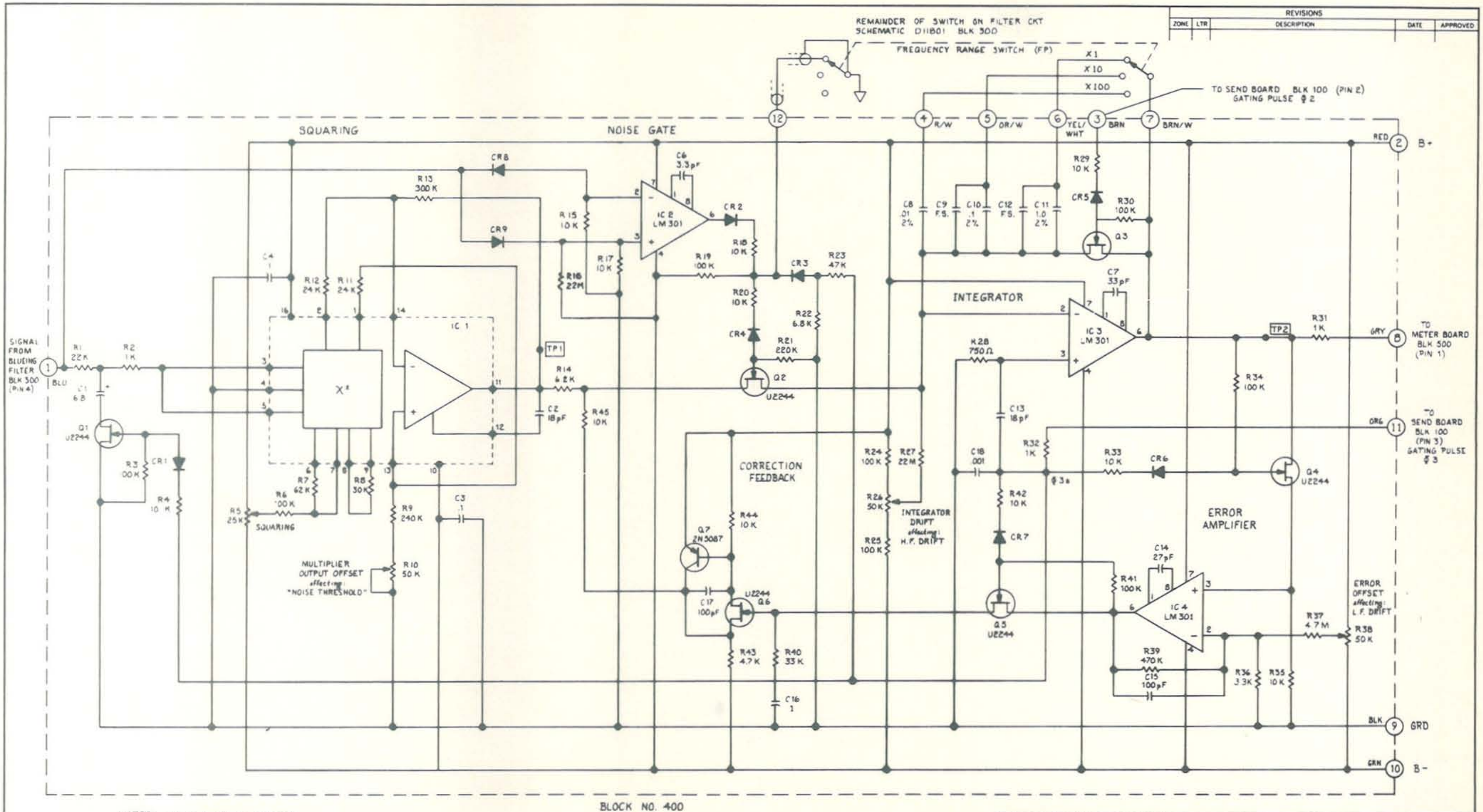


RECEIVE SECTION



POWER SUPPLY





NOTES: UNLESS OTHERWISE SPECIFIED

1. CAPACITOR VALUES ARE IN MICROFARADS
2. RESISTOR VALUES ARE IN OHMS ± 5% 1/2 W
3. REFERENCE CONVERSION P.C ASSY B11814
4. REFERENCE TEST PROCEDURE T.P. 11814
5. ALL DIODES URET 13-0135

REVISIONS				
ZONE	LTR	DESCRIPTION	DATE	APPROVED

NEXT ASSY	USED ON
APPLICATION	

UNLESS OTHERWISE SPECIFIED DIMENSIONS ARE IN INCHES TOLERANCES ARE FRACTIONS ± DECIMALS XX ± XXX ± ANGLES ±

MATERIAL

FINISH

DO NOT SCALE DRAWING

CONTRACT NO

APPROVALS	DATE
DRAWN BDM	9-4-74
CHECKED	

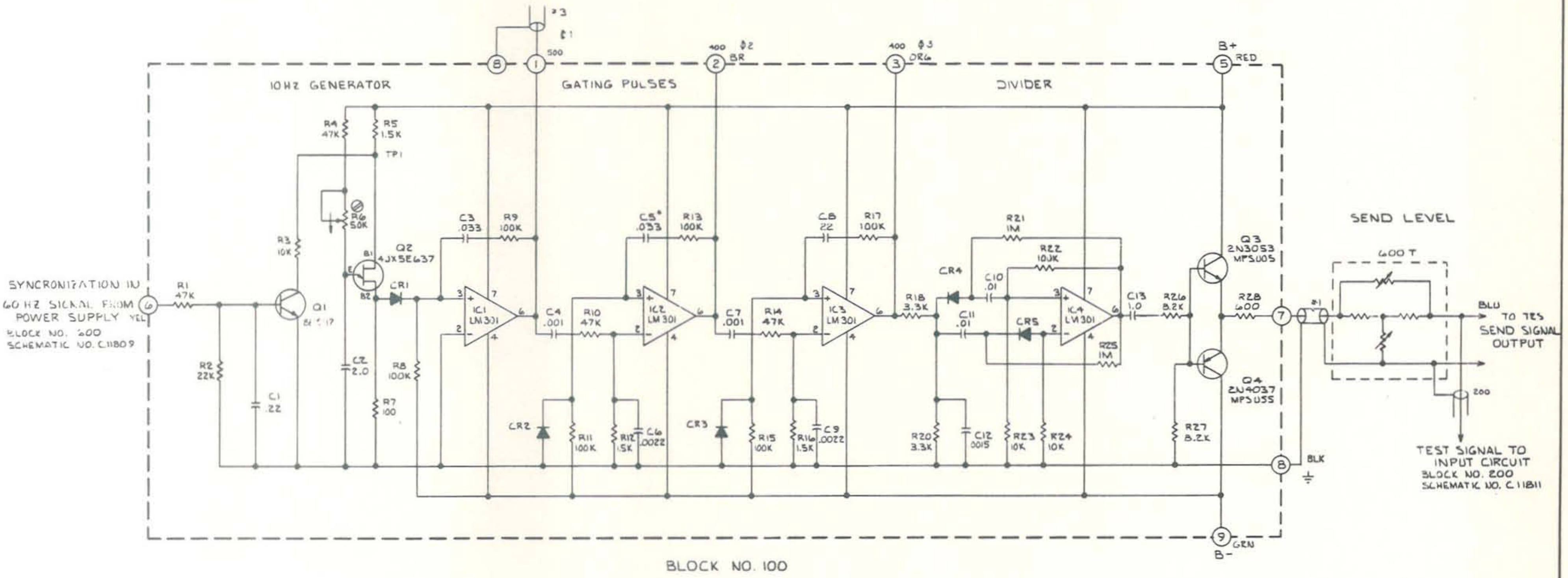
UREI UNITED RECORDING ELECTRONICS INDUSTRIES 11922 VALERIO STREET NO HOLLYWOOD CALIFORNIA 91605

BLOCK 400

CONVERSION CKT. SCHEMATIC SONIPULSE

SIZE	MODEL NO	DRAWING NO
D	100 A	D 11802

SCALE SHEET OF

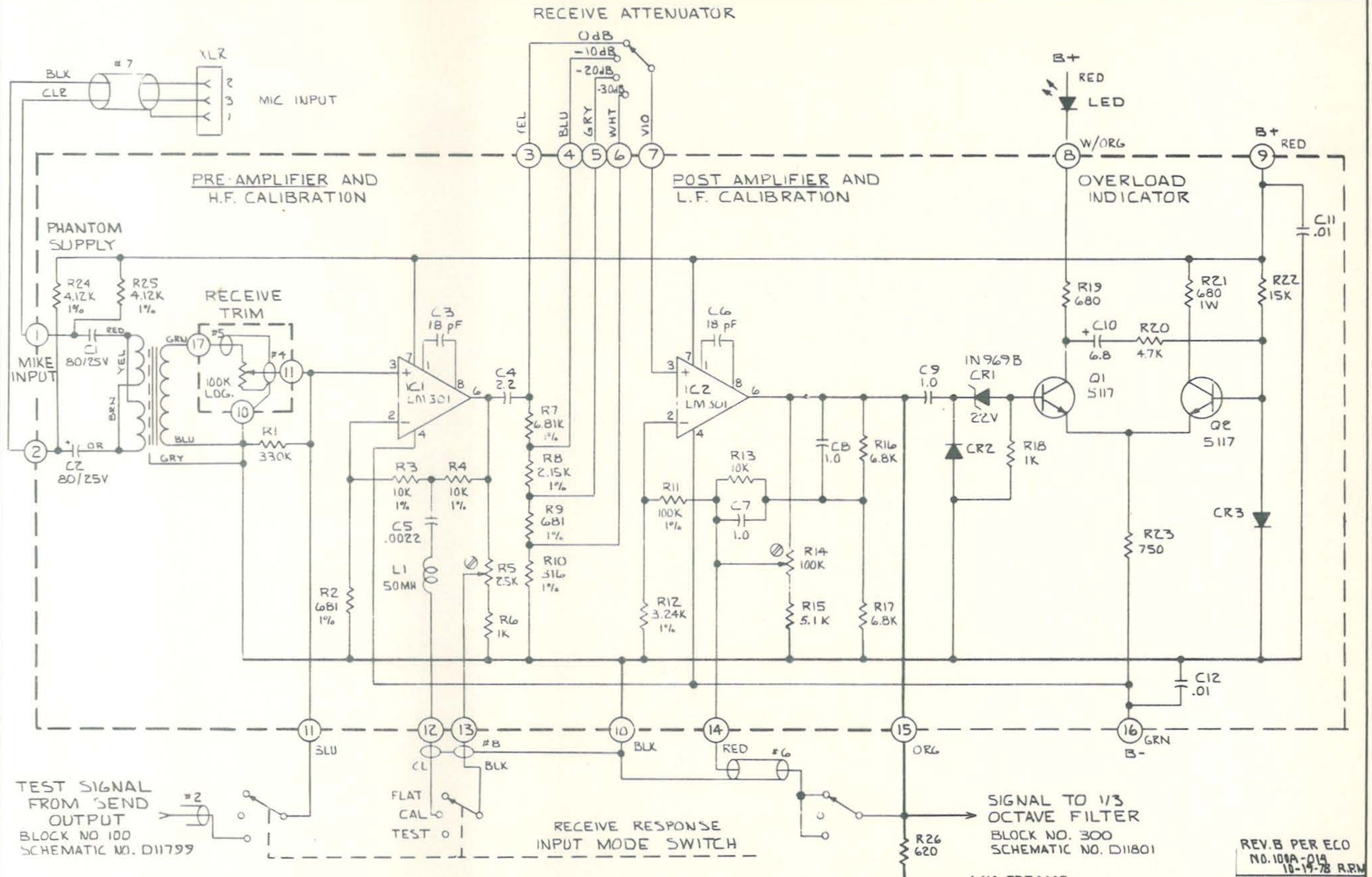


REV. A EFFECTIVE SERIAL # 250

6. ALL DIODES UREI 13-0135
5. C5, C8 & C12 USE SMALL SIZE TO CLEAR DIMENSIONS OF FREQUENCY RANGE SWITCH AFTER ASSEMBLY
4. REFERENCE TEST PROCEDURE TP11818
3. REFERENCE SEND CIRCUIT ASSY B11818
2. RESISTOR VALUES ARE IN OHMS, ±5%, 1/2W
1. CAPACITOR VALUES ARE IN MICROFARADS

NOTES: UNLESS OTHERWISE SPECIFIED

BLOCK 100	REV A 7-2-75 BDM CHANGE C8 WAS .033μF NOW .22μF
UNITED RECORDING ELECTRONICS INDUSTRIES	SEND CIRCUIT SCHEMATIC SONIPULSE
MODEL 100-A	DATE 6-20-72
LOS ANGELES CALIFORNIA	CHECKED L.E. — D11799 A



BLOCK NO. 200

NOTES: UNLESS OTHERWISE SPECIFIED

1. CAPACITOR VALUES ARE IN MICROFARADS
2. RESISTOR VALUES ARE IN OHMS, ±5%, 1/2W
3. COMPONENTS WITHIN BROKEN LINES ARE ON P.C. BOARD
4. REFERENCE MIC INPUT P.C. ASSY B11810
5. REFERENCE TEST PROCEDURE T.P. 11810
6. ALL DIODES UREI 13-0135

REV. B PER ECO NO. 100A-014 10-19-78 R.R.M.

REV A PER ED 0148 SN 325 → 11-9-76 BDM

REVISIONS

BLOCK 200

UNITED RECORDING ELECTRONICS INDUSTRIES	MIC. INPUT SCHEMATIC SONIPULSE	
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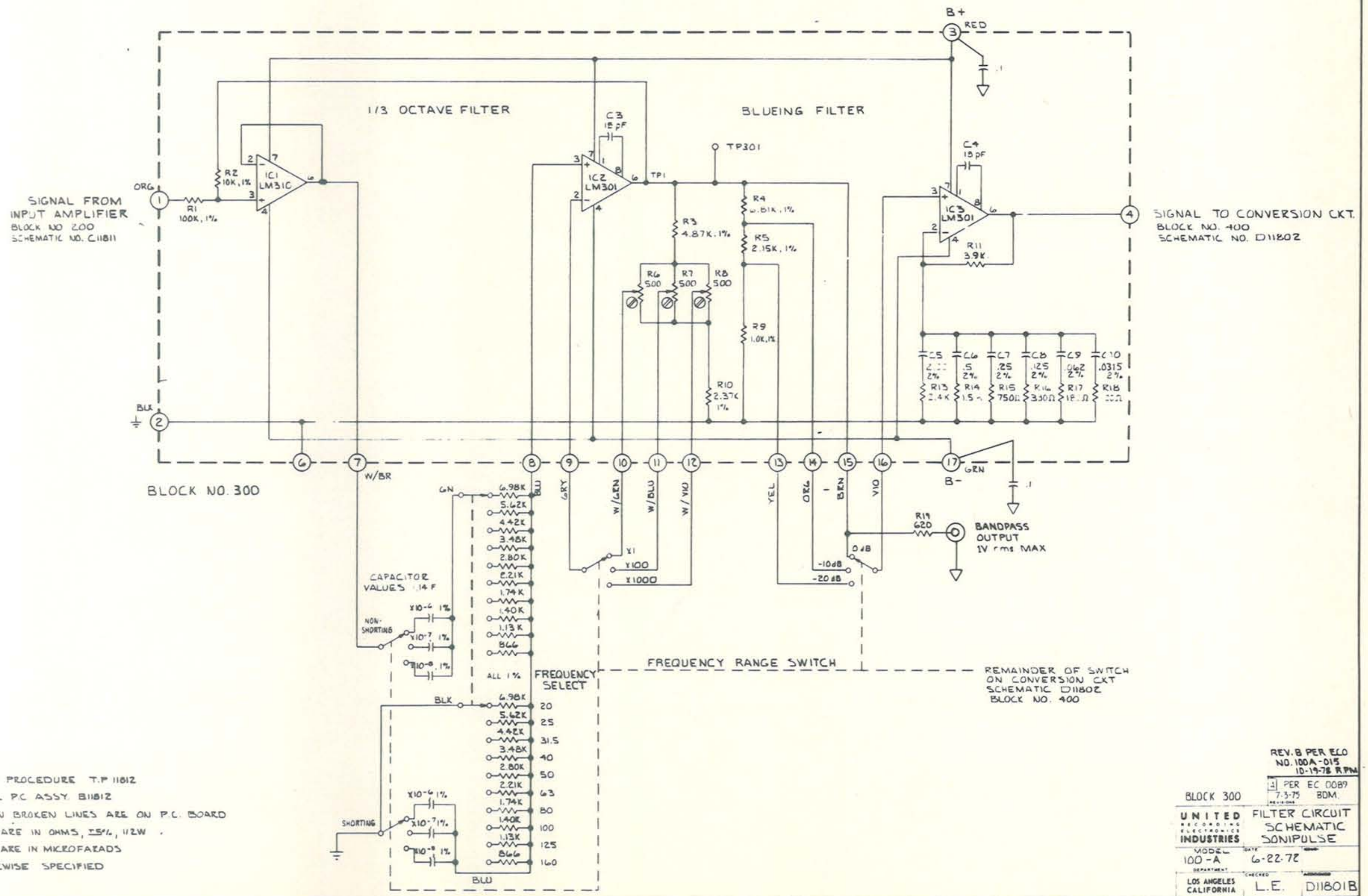


5. REFERENCE TEST PROCEDURE T.P 11812
4. REFERENCE FILTER P.C. ASSY. B11812
3. COMPONENTS WITHIN BROKEN LINES ARE ON P.C. BOARD
2. RESISTOR VALUES ARE IN OHMS, ±5%, 1/2W
1. CAPACITOR VALUES ARE IN MICROFARADS

NOTES: UNLESS OTHERWISE SPECIFIED

REV. B PER ECO NO. 100A-015 10-19-78 RPM

1 PER EC 0089 7-3-75 BDM.

BLOCK 300

UNITED RECORDING ELECTRONICS INDUSTRIES

FILTER CIRCUIT SCHEMATIC SONIPULSE

MODEL 100-A

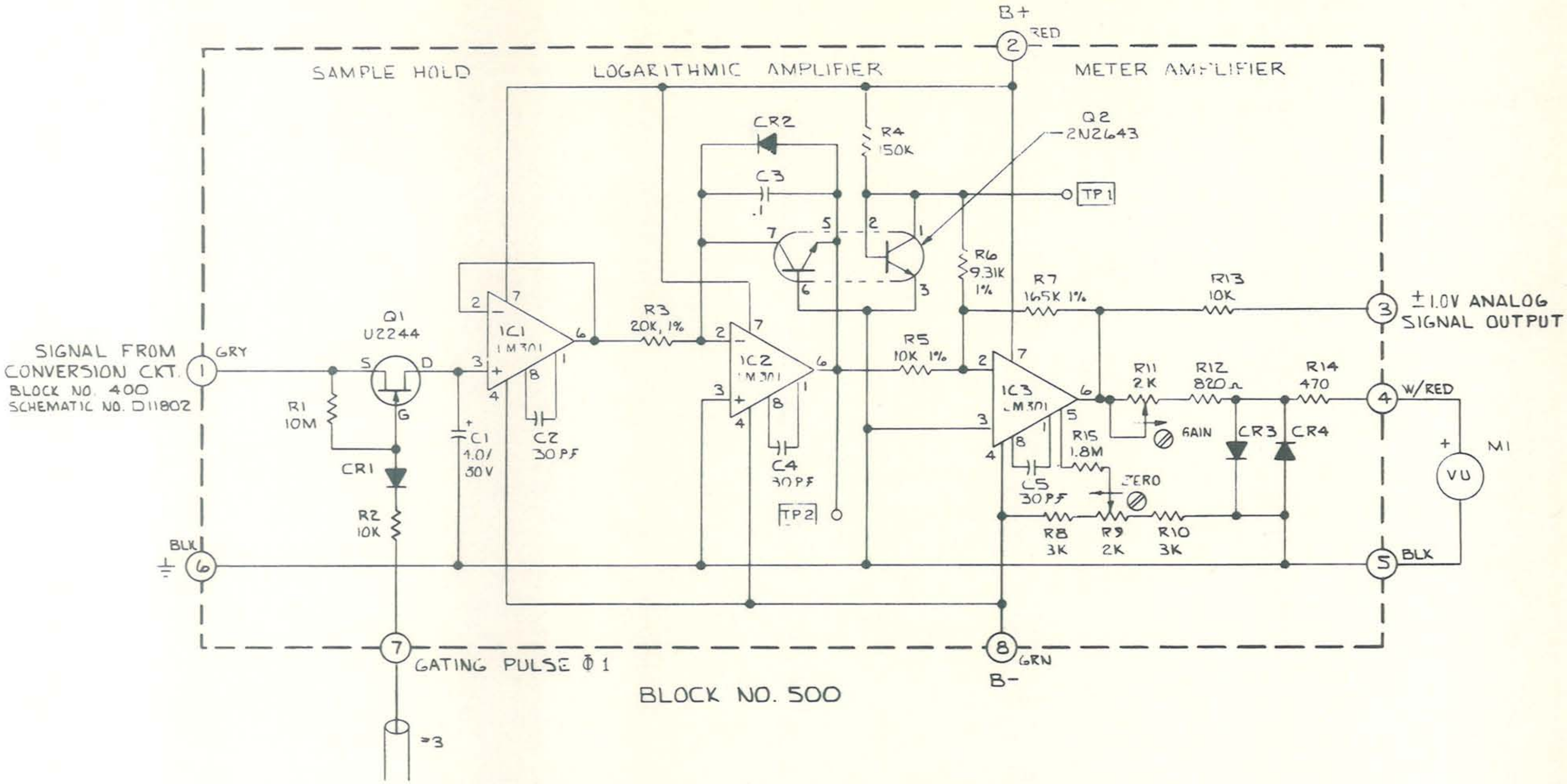
DATE 6-22-72

LOS ANGELES CALIFORNIA

CHECKED L.E.

D11801B

31



5. ALL DIODES UREI 13-0135

4. REFERENCE TEST PROCEDURE TP11813

3. REFERENCE METER P.C. ASSY B11813

2. RESISTOR VALUES ARE IN OHMS, ±5%, 1/2W

1. CAPACITOR VALUES ARE IN MICROFARADS

NOTES: UNLESS OTHERWISE SPECIFIED

BLOCK 500

UNITED RECORDING ELECTRONICS INDUSTRIES	METER CIRCUIT SCHEMATIC SONIPULSE	
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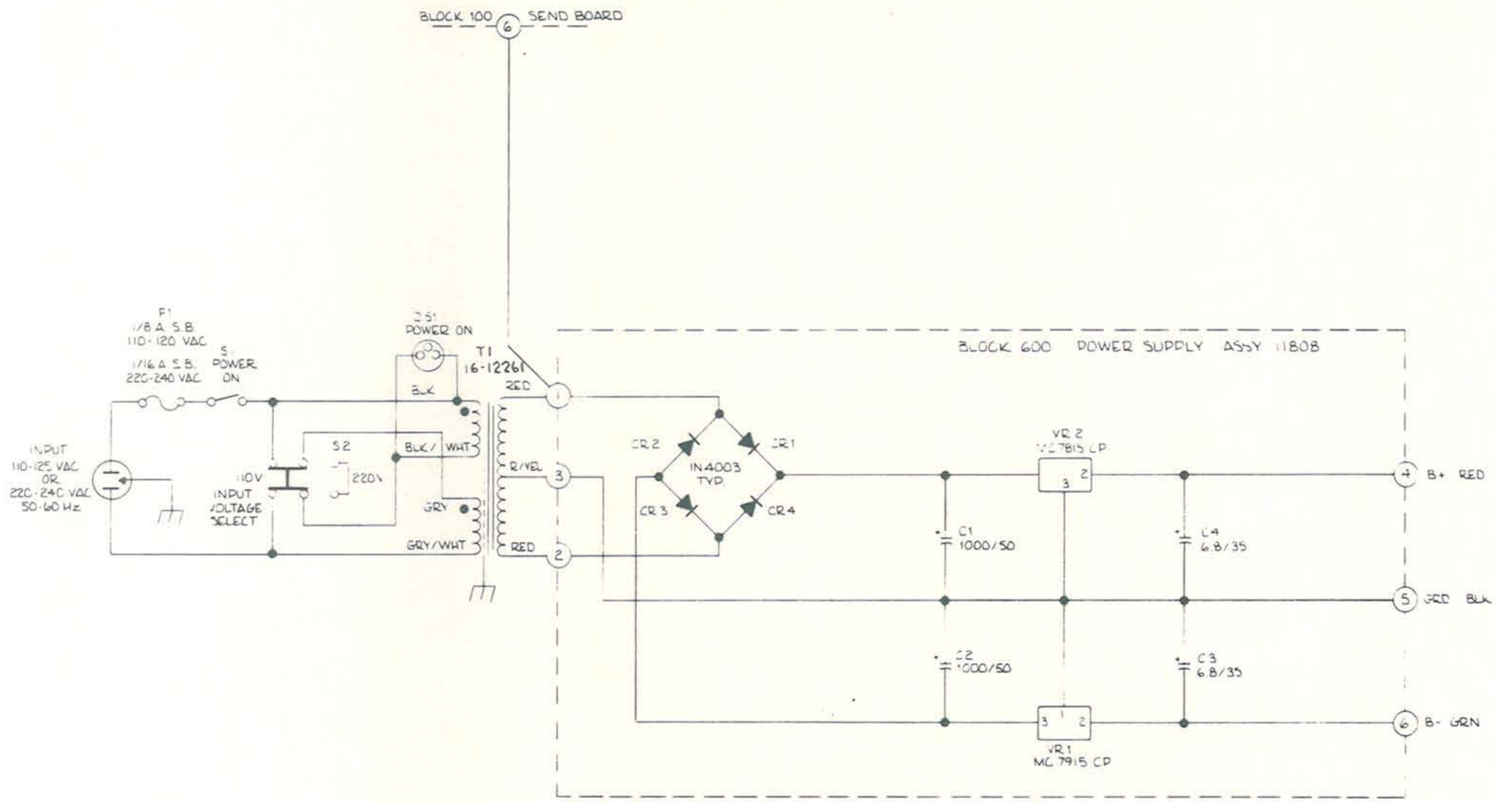

2 REFERENCE POWER SUPPLY PC ASSY 11808

1 CAPACITOR VALUES ARE IN MICROFARADS

NOTES:

REVISIONS				
ZONE	LTR	DESCRIPTION	DATE	APPROVED

NEXT ASSY	USED ON	UNLESS OTHERWISE SPECIFIED DIMENSIONS ARE IN INCHES TOLERANCES ARE FRACTIONS DECIMALS ANGLES XX XXX	CONTRACT NO	UNITED RECORDING ELECTRONICS INDUSTRIES
APPLICATION		MATERIAL	APPROVALS / DATE DRAWN 3DM 8-6-75 CHECKED	SCHEMATIC SONIPULSE POWER SUPPLY
		FINISH		SIZE D · MODEL NO 100A · DRAWING NO 11809 A
		DO NOT SCALE DRAWING		SCALE · SHEET OF