GOTHAM AUDIO CORPORATION

NEW PRODUCTS FROM

1966

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INNOVATION in AUDIO ENGINEERING TECHNOLOGY from EUROPE

- and Introducing -

1. NEUMANN Condenser Microphones with Silicon FET Amplifiers
2. NEUMANN U-79t Correlation Meter
3. NEUMANN AM-66 Computer Controlled Mastering Lathe
4. Studer A-62 Solid State Tape Recorder
5. Beyer DT-48's Professional Dynamic Headphones
6. Gotham ME-101/102 Solid State Wow and Flutter Meters
7. EMT-160 Polarity Tester
8. Eltro Information Rate Changer Mark II
9. EMT-159 FM Stereo Broadcasting Fault Alarm
10. NEUMANN Console Modules with Silicon Technology

THE GRAND SALON, LE TRIANON ROOM
THE ST. MORITZ HOTEL
CENTRAL PARK SOUTH AND AVENUE OF THE AMERICAS
NEW YORK CITY

GOTHAM
AUDIO CORPORATION
2 WEST 46 STREET. NEW YORK 36. N. Y. • 212-CO 5-4111

NEW NEUMANN CONDENSER MICROPHONES WITH SILICON FET AMPLIFIERS

1

MINIATURE CONDENSER MICROPHONE KM-74: The KM-74 Microphone exhibits a cardioid directional characteristic, extremely flat at all angles of incidence to ±135° off axis. The high front-to-back ratio of approximately 30dB allows higher PA system levels before feedback threshhold. The condenser element is gold steamed on a Mylar base material, and is identical to that used in the popular U-64 condenser microphone.

The microphone amplifier is solid state employing silicon planar transistors exclusively. Field effect transistor (FET) technology eliminates the complexity of high frequency circuitry, and provides a long life expectancy. Power is fed through the audio output circuit either from AC powered supplies, from 12 or 24 Vdc sources via supply junction units, or from batteries by means of adaptors. A variation in supply voltage from 7.5 to 14 volts can be tolerated. Hence, both standard 9 volt and 12 volt batteries can be used.

TECHNICAL DATA:

Frequency response................	40 ... 16 000 Hz
Directional characteristic........	cardioid
Output level......................	approx. 3 mV/ub
Noise voltage.....................	$\leqq$ 26 dB re 2 x 10^{-4} ub

MINIATURE CONDENSER MICROPHONE KM-76: This microphone employs a newly developed double condenser element, permitting switchable selection of cardioid, figure 8 and omni-directional characteristics. The condenser elements are pre-stretched moisture resistant Mylar with a gold steamed surface. The microphone is addressed at right angles to its axis, as is the well known KM-56.

The microphone amplifier is all silicon technology, and the first stage utilizes a field effect transistor (FET). The FET technology permits considerable simplification of circuitry over other transistor methods.

TECHNICAL DATA:

Frequency response................	40 ... 16 000 Hz
Directional characteristic........	omni-directional, cardioid, and figure 8
Output level......................	approx. 2 mV/ub
Noise voltage.....................	$\leqq$ 26 dB re 2 x 10^{-4} ub

LAVALIER CONDENSER MICROPHONE KML: Lavalier microphones have been in general use for some time, but have been of the dynamic type directed upward. The quality of such microphones has left a great deal to be desired. The KML employs a special cardioid capsule of gold steamed Mylar directed forward. The linear admittance character of the capsule assures the same frequency response from the wearer's voice as from frontal pickup.

The microphone amplifier is scarcely larger than the capsule itself, and employs FET silicon circuitry. 18 Volts DC is required which can be fed on the audio circuit from a wireless transmitter, or can be supplied by the miniature BSL 45 volt battery unit.

TECHNICAL DATA:

Frequency response....................	40 ... 16 000 Hz
Directional characteristic............	cardioid
Output level..........................	approx. 0,5mV/ub (18 V)
	approx. 1,8mV/ub (45 V)
Noise voltage.........................	< or = 36 dB (18 V) re 2 x 10^{-4} ub
	< or = 27 dB (45 V) re 2 x 10^{-4} ub

CONDENSER MICROPHONE U-77: This microphone is a solid state version of the universally popular U-67. The condenser elements are gold steamed Mylar providing cardioid, figure 8, and omni-directional patterns. Switches are located in the condenser head for proximity effect correction and for overload protection.

The microphone amplifier utilizes silicon FET circuitry and can be powered from an internal battery or through the audio leads as in other NEUMANN microphones with transistor amplifiers.

POWER SUPPLIES N-9 AND N-92: These 9 volt supplies operate from 117 volts AC and measure 1.8" x 3.2" x 4.5". The N-9 and N-92 supplies power one and two microphones respectively. The two power circuits of the N-92 are sufficiently well isolated to allow a short circuit on one side with unimpaired microphone operation on the other.

SUPPLY JUNCTION UNIT SW-1224 AND POWER SUPPLY N-24: The SW-1224 junction units permit distribution of power to NEUMANN condenser microphones from 12 or 24 Vdc sources. The N-24 is a plug-in rack-mounted supply furnishing 24 Vdc. One N-24 and 10 SW-1224 units can power 10 NEUMANN condenser microphones with transistor amplifiers.

BATTERY SUPPLY UNITS BS-9 AND BS-9/200: The BS-9 battery supply utilizes a locally available 9 Volt battery to power one NEUMANN transistor microphone. Battery life is 20 hours. It is light in weight and measures only 1.3" x 2.0" x 3.0". Feed is through the audio circuit. The BS-9/200 operates from six locally available dry cells with supply life of 200 hours continuously or 300 hours intermittently. The BS-9/200 measures 1.1" x 3.3" x 4.6".

BATTERY SUPPLY BSL: The KML Lavalier condenser microphone can be powered from a wireless transmitter furnishing 18 Vdc. It can also be powered from the miniature BSL battery supply furnishing 45·volts. The supply measures 1.3" x 2.0 x 3.0", and it is extremely light in weight to be highly applicable to "fish pole" use.

CORRELATION METER U-79t:

2 The U-79t correlation meter is a new and special instrument which permits the studio engineer to evaluate the mono content of a stereo signal. In essence, the unit indicates the phase relationship of stereo signals. Indication of the correlation coefficient is by means of the deviation of a center position meter.

For properly polarized stereo signals, the meter will indicate between 0 and +1. With phase reversal, the meter will deflect from 0 toward -1. Unrelated signals or the absence of one channel produces a 0 meter reading. The better the mono compatability of the stereo source, the higher the coefficient of correlation, that is, a greater deflection from 0 toward +1. The U-79t correlation meter together with associated solid state circuitry is supplied in a portable case measuring 5.3" x 5.7" x 8.3".

AM-66 COMPUTER CONTROLLED DISK CUTTING LATHE:

3 In keeping with this, the age of solid state technology, the most recent re-design of the NEUMANN AM-32b disk mastering lathe has brought it sharply into the present age and makes it the matchless leader in the disk cutting field. Gone are all the relays which, even though they had not led to complaints in the past, have always produced some minute clicks in the modulation. All this is a thing of the past; the lathe is operated entirely by more than 16 cards equipped with silicon transistor logic technology which provide all of the functions as well as automatic pitch and depth control with a refinement of reaction time never before possible in disk recording.

THE ADVANTAGES:

1) The Pitch Drive: The same box still stands to the right of the lathe proper. It has been restyled as far as the control elements are concerned and is entirely empty except for the two drive motors. All electronic components are rack mounted below in the SA-66 unit. The drive motors no longer heat up as their speed is controlled by means of tach generators mounted on each shaft transmitting feedback information to solid state circuitry, driving the motor by means of pulse width modulation. Motor heat rise is barely above room temperature.

2) The Controls: Five brightly lighted push buttons provide function control, STOP, START and SPIRAL are as before. New is the TIME button which produces timed spirals between cuts on the disk as preset on the lower left-hand selector knob. Also new is a FAST button which is used both for rapid lead-in to the record diameter and for lead-out. The lower right control is a small potentiometer which controls the lines per inch setting. The push button like controls down the right side of the pitch drive unit are lamps displaying the rpm of the turntable drive motor. The three lamps under the LPI meter show the record diameter selected (7", 10", 12").

3) The Depth Control Panel has changed: Also included now is the heated stylus control and meter, as well as the microscope lamp switch. Increased depth of cut is now a differential control on which the amount of deepening is adjusted. This differential will then be added to any depth of cut setting.

4) The Carriage Program Functions: A cam on the back of the carriage slide operates miniature microswitches to perform all outside diameter end-stop and even diameter compensation functions (where desired). Another set of micro switches allows connection of the AM-66 directly to the soon-to-be-released tracing simulator (T.S.) system, a solid state device which reduces tracing distortion on stereo disks to the vanishing point.

5) The Master Programming Plugs: All twenty functions of the AM-66 have been brought out to a connector bank right under the front console border. This connector accepts a plug-in program unit labeled with the record diameter and rpm; i.e. 7" 33-1/3, 12" 33-1/3, 7" 45 etc. In this program plug are all of the controls which determine motor speed, lead-in, spiraling and lead-out speeds, cutter lift delay, beginning and ending diameter switches, pitch and depth levels, storage delays and preview timings. In other words, every function associated with a specific rpm and diameter. Storage space is provided for two programming plugs other than the one in use. Plugging such programmer into the receptacle, lights the appropriate diameter and rpm indicators on the pitch control unit.

6) The Pitch and Depth Preview System: This technology has undergone the most radical change of all. For the very first time a control system is available which differentiates between left and right signals, resulting in unprecedented space savings. Both signals from the stereo preview head and both playback head signals furnish all the intelligence necessary for the system. Right hand excess excursions (outside flank of the groove) require preview pitch information but it need not be remembered. Left hand excess excursion (inside flank of the groove) requires no pitch preview, but must be remembered for one revolution. Increased depth of cut requires room on both sides, as do center sound impressions. The AM-66 treats each of these cases separately. One of the newest innovations is the system of memory circuits to store the pitch and depth information. In previous systems such memory was dependent on the time constant of RC circuits, with resulting slow leak off and space waste. The AM-66 has a novel system of counting the turntable rpm to provide exact charge-up and sharp cancellation of these memory flip-flops. A light source illuminates the underside of the turntable which is painted in alternating black and white quadrants which in turn reflect the light into a light dependent resistor. Proper operation of this unit is indicated by alternately flashing lamps on the front of the SV-66 Control Unit. These alternations produce 1/4 rpm counts and serve to time the charging and cancelling of the memory circuits holding pitch and depth information.

The efficiency of the pitch depth system can add as much as six minutes of playing time to a 12" LP record side without change in recorded level. It makes the AM-66 the most up to date system yet devised for disk mastering. The AM-66 may be used with any of today's disk cutting heads including the NEUMANN SX-45 stereo, ES-59 mono, Westrex 3-D stereo or 2-B mono (actually floated without the use of an advance ball!), Grampian etc. etc.

THE STUDER A-62 SOLID STATE TAPE RECORDER

4

After more than 3 years of field testing and gathering of experience from protypes in use in various countries, the A-62 is now a reality. It is the first completely self contained stereo machine in a 14" x 19" rack mounted configuration.

The electronics: Silicon transistor, plug-in units provide all of the machine's functions. There is a plug-in power supply, tape tension control unit and two record-playback channels. All adjustment controls are grouped together in one field and are blocked by slides covering the holes of access. A special key is needed to expose these controls to authorized personnel only. As in the larger C-37 unit separate level, equalization and bias controls are available for its two speeds,7.5 and 15 ips.

Hold back tension control: STUDER has long been a leader among those realizing the key importance of constant hold back tension. The C-37 has its famous tension balance control. The A-62 has the most sophisticated system yet devised. Mounted on the rewind motor shaft is a magnetic drum on which a permanent high level AC signal has been recorded. A playback head mounted nearby (but not touching) scans this tone recording. Since hold back tension is directly related to the rpm of the rewind motor (higher rpm indicates a smaller tape diameter which means higher tension), the frequency read by this head can be translated via integrating and magnetic amplifier circuitry to control holdback tension within small tolerances. You may use from 3" to 10.5" reels without the need to switch tension! (except to switch between 7" and 10.5" take up reel sizes).

The tape shut-off control: Here too is found a brand new system in the A-62. The left hand idler has a toothed magnetic wheel mounted on its shaft beneath the top plate. As long as this wheel passes a magnetic pickup the machine runs. When the idler stops, the machine stops. No spring loaded tension arms are needed to do the job, and here's another plus of this system: Until the wheel stops, you cannot restart the tape machine after a fast speed wind. You simply keep your finger on the start button; the machine will start the instant the tape has come to a stop.

Threading ease and editing: The A-62 features two rollers which pull in on "start". One is the capstan pressure roller on the right, and the other is a flutter roller on the left of the plug-in head assembly. The result is a straight-across unhampered tape path when the machine is at rest. Push start and both rollers pull in, pressing the tape against the heads. You want to hear the tape slightly as you wind? Simply push the "C" (cutting) button and the tape moves half way in, just grazing the heads.

The tape timer: As usual with European tape machines, a precision tape timer is provided to read in minutes and seconds directly at 15 ips, with an accuracy of ± 3 seconds in half an hour fast winding or playing. Push button reset.

The capstan drive: As in the STUDER C-37, this capstan is an in-line hysteresis synchronous motor with a heavy flywheel and rubber isolation between motor and capstan. The result is a better than ±0.05% flutter content at 15 ips.

Metering and mounting: Machine has line input and output capability with a maximum output of +22 dBm. A metering panel is available as well as a mixer capable of bringing several microphone inputs to two channels of line level. The machine can be mounted vertically or horizontally in a standard EIA 19" rack and has a 14" panel height.

THE NEW BEYER DT-48s DYNAMIC PROFESSIONAL HEADPHONES

5

Making a strong reappearance is the new BEYER DT-48s. Nothing has been changed in its unparalleled quality which has remained untouched for over 30 years, but the comfort to the wearer is in an entirely new class. A completely new headband has been designed and allows adjustment of phone comfort for different size heads while cushioning the pressure on top of the head with soft foam rubber. The cushions are brand new ear conformal foam making the DT-48s at last not only tops in quality but also tops in comfort. Previous owners of the older DT-48 may obtain both the headbands and cushions to re-equip their phones. The cables from each phone separately terminate in a standard tip-ring-sleeve phone plug but may be split into separate connectors for each phone for professional use. The TR-48s transformer permits operation of the DT-48s with 600 Ohm professional equipment. A new UG-8 switching box permits switch-over between loudspeakers and DT-48s with identical loudness. Screw terminals on the back of the UG-8 permit wiring of the monitor amplifier outputs, and speaker inputs, while jacks on the front of the box accommodate two pairs of DT-48s simultaneously. A slide switch connects either phones or speakers to the stereo amplifier.

GOTHAM ME-101/102 SOLID STATE WOW AND FLUTTER METERS

6

The ME-101 and ME-102 at last bring the meaningful measurement of Wow and Flutter within the reach of all recording studios, broadcast operations and service activities. These are the only W & F meters in this price category which incorporate two separate meters, one to measure drift, the other ±% W & F. Further, these instruments conform to the CCIR standard which specifies ± quasi-peak indication and the weighting curve. Linear measurements can also be made, and meter ballistics can be damped to simulate action of the VU meter. The instrument is extremely stable and is self calibrating.. An internal signal generator delivers the standardized 3150Hz test signal. The linear range of the discriminator permits de-tuning to accept 3000Hz. The full scale measuring ranges are:

ME-101 ±0.5% and ±2.5%
ME-102 ±0.15% and ±0.75%

7

EMT-160 POLARITY TESTER

Another example of special test equipment from EMT. To our knowledge, the EMT-160 is the first generally available instrumentation permitting a rapid and positive check of microphone phasing in any multi-microphone situation. The unit has application to both single and multi-channel systems.

The device consists of a sender and an indicating unit. Both sections are solid state, battery operated and light in weight. The sender is both "Acoustical Pistol" and electrical pulse generator. A pulse of such definite polarity is developed that no degree of distortion in amplifying systems under test can falsify readings. The pulse can be fed into a system or element thereof at a selectable level ranging from 1.0mV to 9.0 Volts, or the acoustical output may be fed directly into microphones. The indicator bridges line level and displays correct or reverse polarity by the illumination of green or red tally lamps. The "Acoustical Pistol"/pulse generator stores in a well in the indicator unit and permits testing functions and thereby battery condition as well.

8

ELTRO INFORMATION RATE CHANGER MARK II

This device continues to remain unique throughout the world. Yet the Mark II features many engineering advances over the preceding model. In operation the device allows a change in playing time of 1/4" 15 ips tapes from 53% to 200% of nominal time without change of pitch and with excellent fidelity other than expected sampling rate interference at range extremes.

IMPROVEMENTS IN THE ELTRO MARK II

1) More attractive and functional top plate.
2) Newly designed rotating head with gaps of only 0.2 mils extending freq. response flat to 15kHz.
3) New head brush assembly assures long noise free life without lubrication.
4) Remote start of associated tape machine.
5) Automatic engagement of Studer type pressure roller.
6) Two additional idlers in close proximity to head for low Wow and Flutter content, and easy adjustment of azimuth and wrap angle. All idlers easily adjusted in height.

7) Top supported sturdy capstan. Removable capstan sleeve for 7-1/2 ips.
8) Head assembly removable for substitution of 7-1/2 ips head assembly when available.
9) New control knob permits rapid tempo changes and provides read out in percentage of original time.
10) Internal solid state amplifier delivers line level output. Includes gain and tone controls.

9

EMT-159 STEREO FAULT INDICATOR

In spite of the number of devices available to make stereo transmission foolproof, there is a definite need for an alarm system which will indicate serious faults which can occur during programming, where sufficient critical aural supervision is not possible.

The EMT-159 is a simple solid state computer which will give indication of the following faults:

1) Total absence of audio signal.
2) Left channel missing.
3) Right channel missing.
4) Difference (L-R) channel missing (mono only).
5) L or R channel reversed in polarity.

Input impedance is high and balanced and gain permits bridging line level. Tally signal feeds can be remoted to operate visual and aural alarms.

10

NEUMANN SOLID STATE CONSOLE MODULES WITH SILICON TECHNOLOGY

Experience gained by the NEUMANN firm from the engineering design and manufacture of many consoles in various configurations has led to the development of a new line of console modules employing Silicon transistor technology.

The new console components again emphasize reliability and feature high output level (+25.5 dBm @ 0.5% THD @ 200 Ohm load). Operating temperatures as high as 60°C can be tolerated easily. All inputs and outputs are balanced and floating.

A separate booklet is available at the GOTHAM AUDIO SHOW describing in detail the ten modules in the new series. The well established and time tested Germanium console modules will continue available as well.

AND OF COURSE ....

Unmatched products still available from GOTHAM - The EMT-140 Reverberation Unit, The Gotham EQ-1000, The NEUMANN AM-131 Lathe, The EMT-930/940 Broadcast Turntables, The Lyrec TIM-4A Timer, The Gotham KW Attenuators, LBV Light Beam VU Meter, EMT 420A/421A Wow and Flutter Meter/Octave Band Filter, EMT-137 NoisEx, Beyer Transformers, and Preh Potentiometers, and others.

... THANK YOU ...

FOR ATTENDING OUR EXHIBIT. IT WAS OUR PLEASURE TO INTRODUCE TO YOU INNOVATIONAL AUDIO TECHNOLOGY FROM EUROPE. FURTHER INFORMATION PERTAINING TO ANY OF THE NEW OR STANDARD PRODUCTS IN OUR LINES CAN BE FURNISHED AT YOUR REQUEST.

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M 49 - M 50 -910-02-01

APPLICATIONS

The Neumann condenser microphones M 49c and M 50c are studio microphones with excellent characteristics. The microphone M 49 c is used in broadcasting, films, disc recording, in post-sync studios and everywhere, where good results have to be achieved in spite of difficult accoustic conditions. As this microphone can be switched to three different directional characteristics, it can cope with almost any situation likely to arise.

1164254
Printed in Germany

The microphone M 50 c is a pressure transducer with omni-directional characteristic only and has proved very suitable for recording orchestras in accoustically favourable conditions by the one-mike technique.

BASIC CHARACTERISTICS

Both microphones are of the same shape and size and belong to the standard range of Neumann microphones. They can be distinguished from one another by means of a red or white dot respectively above the NEUMANN sign. They are both of exceptionally robust construction.

The microphone capsule is underneath the removable wire mesh housing and is rubber mounted on a perspex cover. Underneath this cover is the microphone amplifier which, in turn, is mounted on a rubber plate. Due to this construction, the microphones are insensitive to low frequency disturbances such as floor vibration due to walking.

The supply voltage input and the signal output connections to the microphones M 49 c and M 50 c are made via an 8-pin Tuchel plug with bayonet locking. They are supplied under the numbers M 249c and M 250c with RF proof 7-pole standard couplings (Tuchel contact system).

TECHNICAL DETAILS

The microphone amplifiers are equipped with the low noise Telefunken triode AC 701k and work as anode amplifiers feeding a transformer which is astatically wound to avoid hum pick-up.

The capsule of the microphone M 49c consists of two sections, each with a vacuum gold plated plastic diaphragm. Each half of the capsule works as a pressure gradient transducer with a cardioid characteristic. By means of suitable polarising voltages, the two cardioids can be added to give the three characteristics of omni-directional, cardioid and figure-of-8. The change-over is effected by means of a continuously variable potentiometer in the power supply unit enabling the recording engineer to select any intermediate position between omni-directional, cardioid and figure-of-8 from the control room.

The capsule of the microphone M 50c is equipped with a metal diaphragm and mounted inside a perspex sphere. It operates as an omni - directional pressure transducer. The frequency response of the microphone rises at the high frequency end evenly by about 5 dB without becoming too directional. With a diffuse sound field, this results in an almost linear frequency response from 40 to 15 000 cps.

The internal resistance of the output of the amplifiers is normally 200 Ω. By changing two links on the output transformer, the amplifiers can be easily changed to 50 Ω whereby the output voltage falls by 6 dB.

Microphones which have been connected for 50 Ω before leaving the factory are marked with a red dot on the identification plate.

A calibrating voltage for checking the amplifier can be fed via the centre pin of the 8-pin microphone connector.

A C C E S S O R I E S

Power Supply Unit NN 48b

The microphones can be fed from the mains by means of the portable power supply unit NN 48 b. The plate and heater voltages are stabilized and therefore independent of mains voltage fluctuations. The low frequency output voltage appears on a three pin socket and the unit is equipped with a mains-socket.

Power Supply Unit N 52t

The circuit of this power supply unit is transistorized and printed. The N 52t is made as a plug-in unit enabling up to ten units to be mounted side by side by means of a rack mounting frame type S 167.

Battery Unit BB 50

When there are no mains available, the microphones can be fed by means of the battery supply unit BB 50. The unit BB 50 has the same dimensions and technical data as the power supply unit NN 48 b but it is equipped with a 4-cell gas tight accumulator and a transistor DC converter.

NN 48b

Mains voltage.............	117/127/220/240 V ± 10 % 50/60 cps
Fuses.....................	80 mA for 117/127 V 50 mA for 220/240 V DIN 41 571
Valves....................	1 neon stabilizer 150 B 2 (VALVO)
Power consumption.........	11 watts
DC output voltages........	4 V 100 mA 114 V 0.79 mA 0 ... 120 V
Hum voltages..............	≦ 10 µV and ≦ 8 µV respectively
Neon pilot lamp...........	Rafi 110 V No. 2855
Dimensions................	220 x 100 x 100 mm
Weight....................	2.2 kg

T E C H N I C A L   D E T A I L S

M 49c

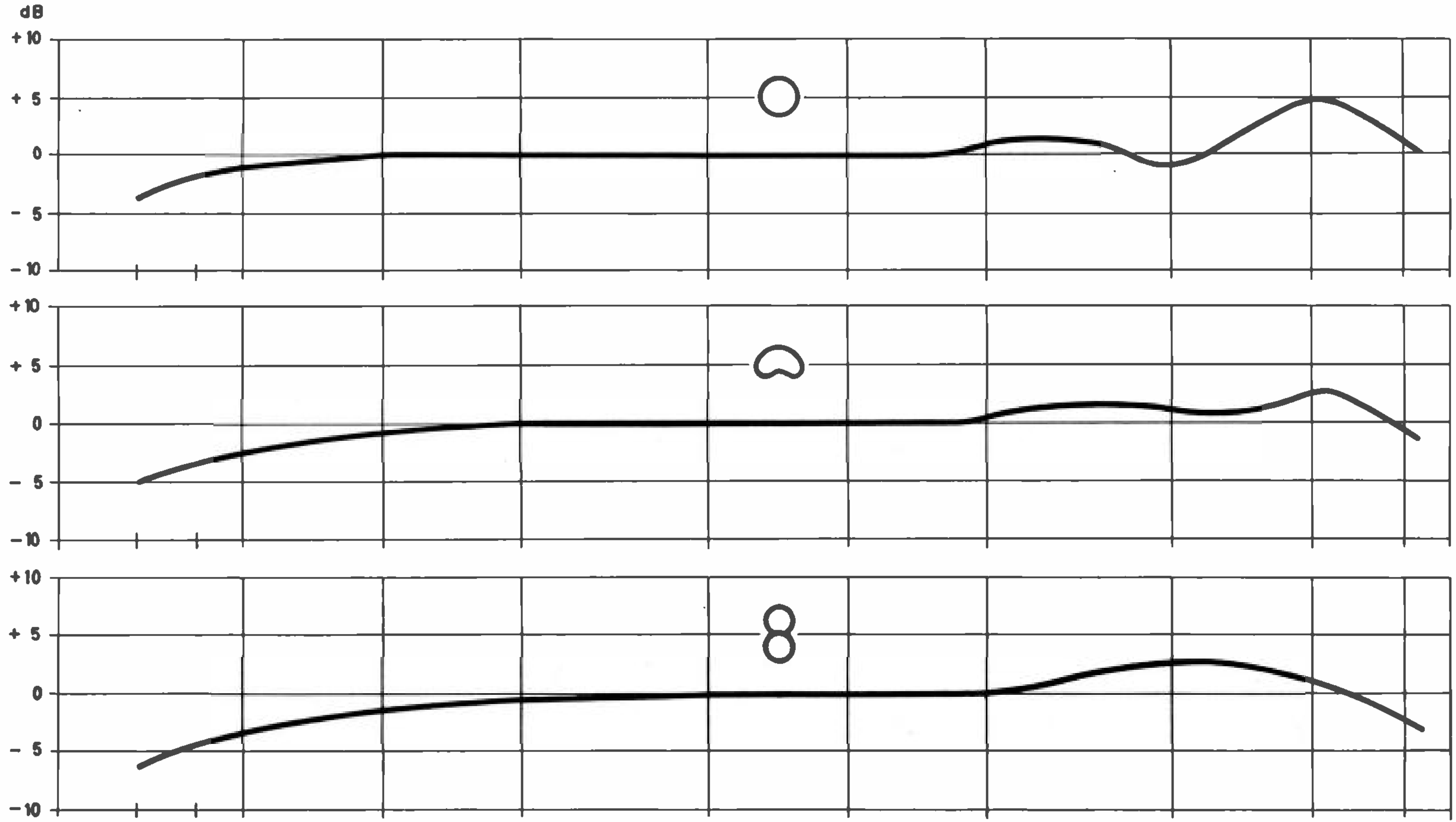
Accoustical operation...	Combination of two pressure gradient transducers which can be electrically switched to omnidirectional, cardioid and figure-of-8 characteristic
Frequency response......	40 ... 16 000 cps
Output levels...........	Omni :0.45 mV/µb into 1000 Ω Cardioid :0.6 mV/µb into 1000 Ω Figure-of-8 :0.8 mV/µb into 1000 Ω
Electrical load resistance..............	$\geqq$ 1 000 (250) Ω
Electrical source resistance..............	200 (50) Ω $\pm$ 20 %
Capacitance of capsule.................	app. 2 x 80 pF
Total noise voltage.....	$\leqq$ 14 µV
Weighted noise voltage (DIN 45 405)...........	$\leqq$ 4 µV
Equivalent loudness.....	$\leqq$ 32 dB (dB above $2 \cdot 10^{-4}$ µb) (DIN 45 405)
Max. sound pressure for .5% distortion at 40 cps, 1 kcps and 5 kcps.......	$\geqq$ 125 dB (dB above 2 10^{-4} µb)
Gain of microphone amplifier at 1 kc/s.....	0 dB
Valves..................	1 x AC 701 k (Telefunken)
Dimensions..............	length 163 mm diameter 80 mm
Weight..................	.8 kg

M 50c

Accoustical operation.......	Pressure transducer
Frequency response..........	40 ... 16 000 cps
Output level................	1.5 mV/µb into 1000 Ω
Electrical load resistance..................	≧ 1 000 (250) Ω
Electrical source resistance..................	200 (50) Ω ± 20 %
Capacitance of capsule......	app. 75 pF
Total noise voltage.........	≦ 16 µV
Weighted noise voltage (DIN 45 405)................	≦ 7 µV
Equivalent loudness.........	≦ 28 dB (dB above $2 \cdot 10^{-4}$ µb) (DIN 45 405)
Max. sound pressure for .5% distortion at 40 cps, 1 kcps and 5 kcps...........	≧ 126 dB (dB above $2 \cdot 10^{-4}$ µb)
Gain of microphone amplifier at 1 kc/s.........	app. - 1 dB
Valves......................	1 x AC 701 k (Telefunken)
Dimensions..................	length 163 mm diameter 80 mm
Weight......................	.8 kg

CABLES C 26 and C 28s

Standard length.............	10 m
Diameter....................	app. 7 mm
Weight......................	1.0 ... 1.3 kg
Thread on stand connector...................	1/2" and 5/8" - 27



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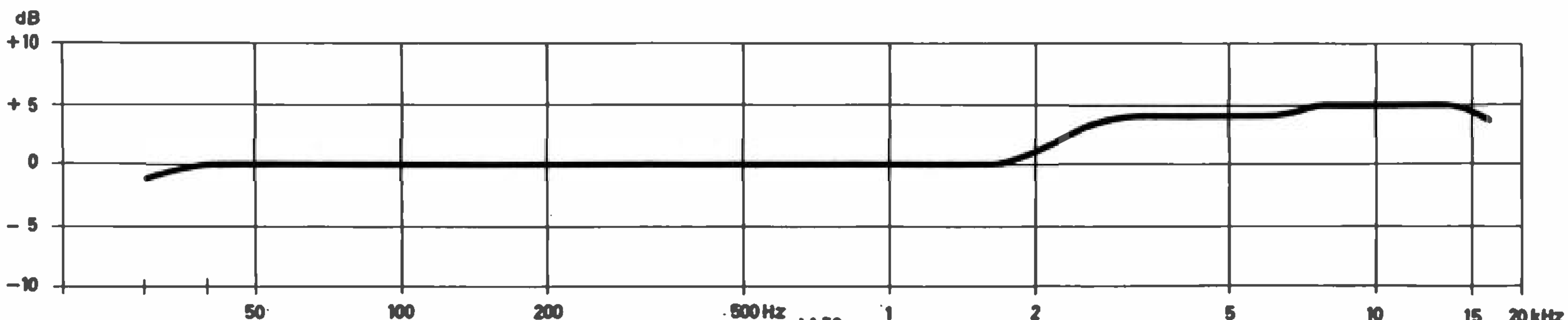

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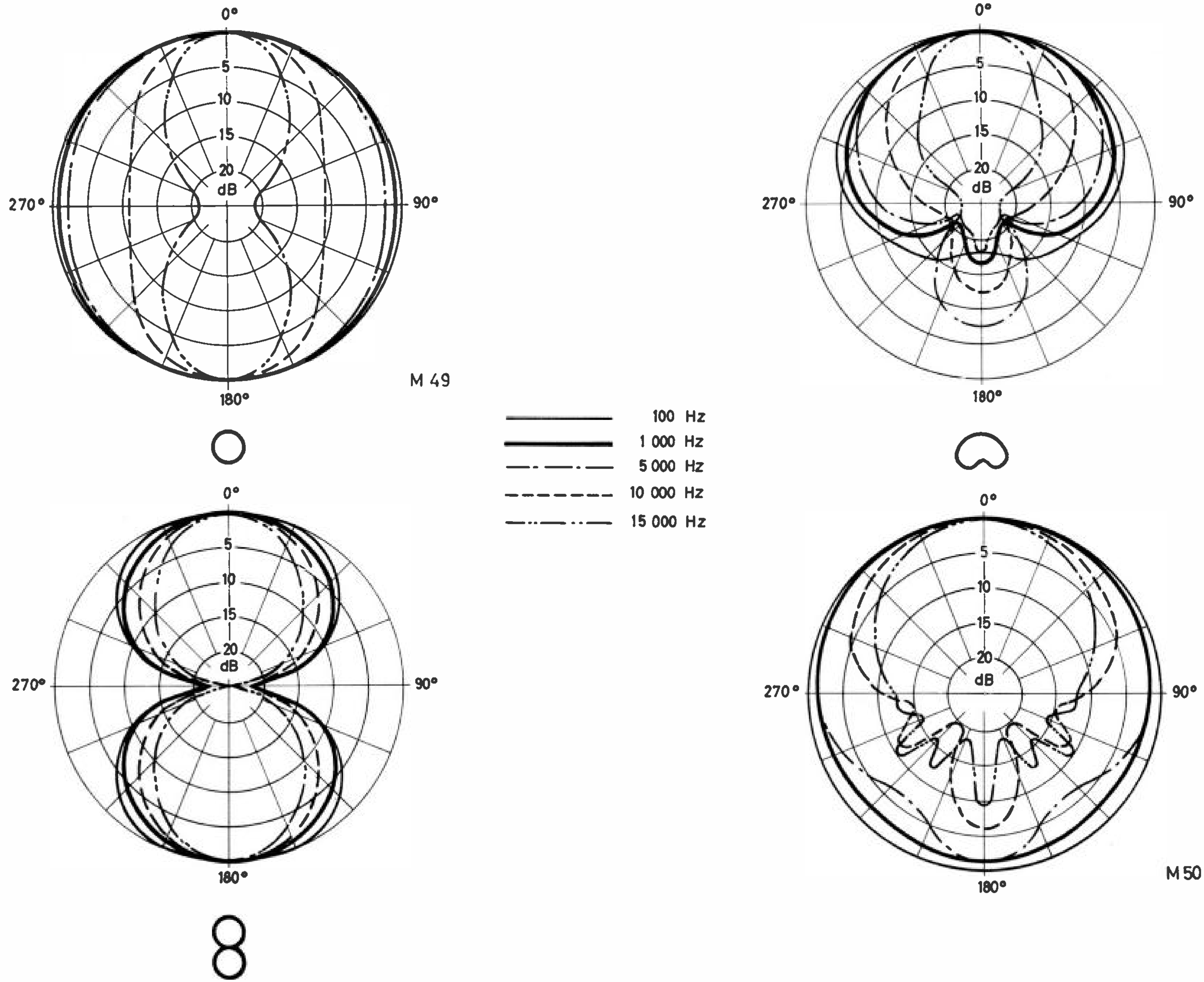

M 49
M 50
100 Hz
1 000 Hz
5 000 Hz
10 000 Hz
15 000 Hz

NEUMANN
U64

linear admittance cardioid

CONDENSER MICROPHONE PROGRAM

®

THE NEUMANN *linear admittance cardioid* CONDENSER MICROPHONE PROGRAM

THE CONDENSER MICROPHONE SYSTEM — ITS REAL MEANING

Condenser Microphones, for all their vaunted quality characteristics, such as wide and smooth frequency response and low distortion, are fundamentally simple devices in spite of their seeming complexity.

The various component parts which make up a condenser microphone system are detailed in this brochure as they pertain to the NEUMANN U-64. What is of utmost importance to a prospective condenser microphone purchaser is this fact: The condenser **element** is the microphone! Regardless of the shape, size or electronic techniques employed, all else is hardware. Even though cleverly engineered and precisely assembled, there is still relatively little genius needed in the design of the hardware aspect of condenser microphones. The expert knowledge, art and experience, particularly as practiced by the scientists and craftsmen of the Neumann Laboratories, lie in the province of the condenser element, the **intrinsic microphone** itself.

Certain microphone buyers erroneously have come to assume that the word "condenser" associated with a microphone is synonomous with ultimate quality. This rather widespread misconception has caused some buyers to be led astray, with resultant disappointment, from the real objective of quality by an intriguing use of hardware.

That NEUMANN does continue to develop new electronics is exemplified, in only one case, by the universally acclaimed U-67. However, the NEUMANN design philosophy will not permit microphone performance to be sacrificed, no matter how slight the degree, by associated electronic circuitry or components.

The NEUMANN U-64 condenser element depicted, and presently to be discussed in detail, is a particular example of NEUMANN dedication. During recent years there has been a marked obsession with **hardware** on the part of many condenser microphone manufacturers. Conversely NEUMANN directed itself toward effecting a major quality improvement where it REALLY counts — at the source. The result, the U-64 condenser element, gains NEUMANN a substantial lead in the field, so that you can expect to see this same condenser element for many years to come; indeed it may appear in different hardware dress, but only when such electronic developments can be accommodated without any sacrifice in quality.

WHERE IT ALL BEGINS — THE CONDENSER ELEMENT, AN INNOVATION

The U-64 condenser element basically consists of an extremely thin Mylar base material something less than ¾" in diameter with a conducting surface of pure gold only molecules thick. The gold is applied by a vacuum evaporating process reflecting nearly 40 years of research, development and experience by the NEUMANN Organization. Immediately behind this membrane assembly is an acoustical phase shift network. The condenser element and acoustical delay network are so accurately engineered as to produce the true cosine function which mathematically defines the cardioid curve. Figure 1 displays the directional characteristic of the NEUMANN U-64 microphone at different frequencies. The cardinal direction of sound pickup is in line with the axis of the microphone, i.e. it is pointed at the source of sound. The frequency response of the U-64 microphone is charted in figure 2. The unusually flat frequency response to sounds at angles as much as 135 degrees from the front of the microphone, inspired the name **LINEAR ADMITTANCE CARDIOID.**

Figure 1 180°

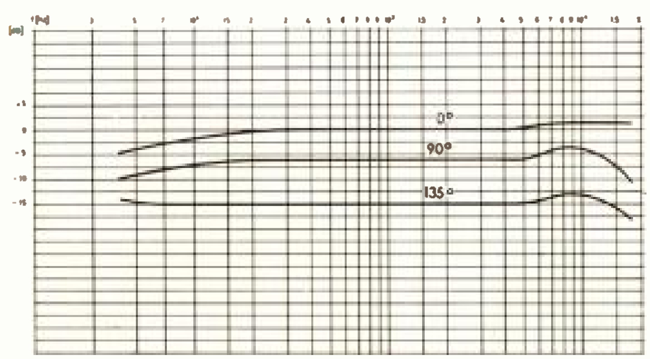


Figure 2

THE AMPLIFIER

TO PROVIDE A USABLE OUTPUT SIGNAL FROM THE CONDENSER ELEMENT

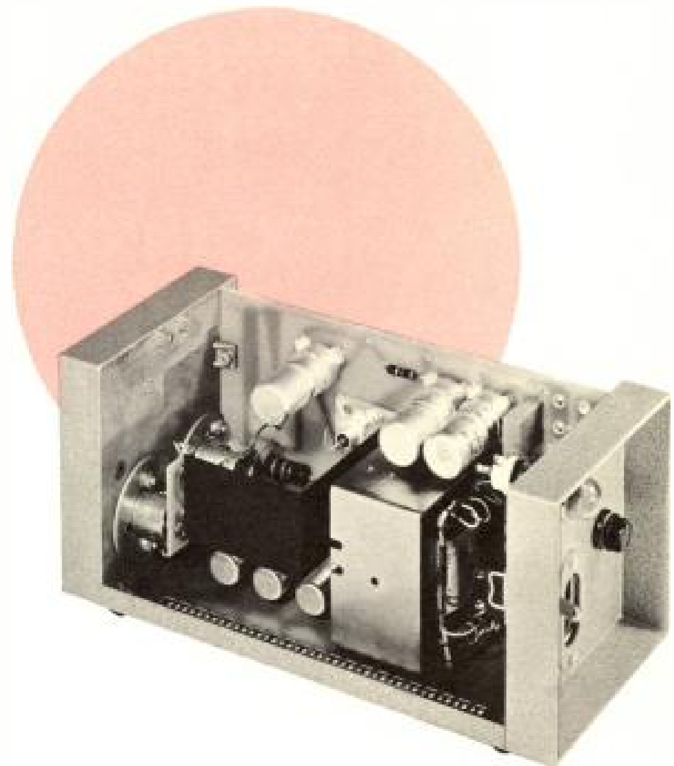
The principle function of the internal amplifier stage is to convert the extremely high impedance of the condenser element to a usable standard low impedance. This amplifier has little or no gain, as the actual signal output of the condenser element is many times the output of dynamic or ribbon type microphones.

The amplifier circuit uses the low noise type 7586 plug-in Nuvistor. As in all NEUMANN microphones, the audio circuit is complete within the microphone housing itself, including the astatically wound output transformer to avoid hum pickup. Operation of the microphone is permitted as much as 400 feet from the associated regulated power supply with no deterioration of quality.

A polarizing voltage of 51 volts D.C. is applied to the condenser element. This exceptionally low polarizing voltage, characteristic of all NEUMANN microphones, obviates many operational difficulties, such as humidity sensitivity, encountered with condenser microphones utilizing polarizing voltages of up to 200 volts.

Additional circuitry is furnished to provide a switchable 10 dB reduction in the signal level delivered by the condenser element prior to input to the impedance converting amplifier. This convenience permits close miking applications of the NEUMANN U-64 in high intensity sound fields without fear of microphone amplifier overload.

Interconnection between the microphone and its power supply is made by means of a 5 conductor shielded cable equipped with 5 pin XLR type connectors.



THE POWER SUPPLY

A NECESSARY ELEMENT IN ALL CONDENSER MICROPHONE SYSTEMS

The NEUMANN U-64 microphone program is the most versatile in adaptation to a wide range of varying microphone applications on the world market today. Four different power supplies are available, allowing multiple microphone users to gain a serious economic advantage over single system costs. The N6u is the basic power supply for one U-64, the N62u powers a pair of U-64's and the N66u powers six U-64's. The N66k power supply for six U-64's is a 19 inch rack mountable unit for permanently wired installations.

NEUMANN power supplies are all solid state, stabilized, and of etched circuit board construction. They are equipped with AC connectors, fuses, audio output and interconnecting cable connectors all of American types. Audio output impedance permits operation into American preamplifiers of from 30 to 250 Ohms input impedance.

The multiple power supplies have the additional safety feature that even in the event of a short circuit on one supply section, the operation of the other microphones is virtually unaffected; cross talk between supply circuits is more than 120 db down.

PARTICULAR ADVANTAGES AND APPLICABILITY

The U-64 is an especially useful microphone for a large variety of applications, from moderately close miking conditions using the acoustic foam anti-pop screen furnished, to distant miking applications such as suspending microphones in auditoriums.

The frequency response of the NEUMANN U-64 microphone exhibits a slight drop below 40 cps to compensate for the normal bass rise at close range typical of all pressure gradient microphones. At the high frequency end, the response shows a rise of approximately 2.0 dB to overcome the usual deficiency in high frequencies due to room absorption of the distant (or reverberant) sound field.

SPECIAL ACCESSORIES AVAILABLE

A wide variety of accessories for special operating conditions are available in the NEUMANN U-64 microphone program as listed below.		
Z-118	Wind screen — for outdoor application	$17.50
EA-21	Elastic suspension — for effective isolation from mechanical shock	15.00
Z-68	Auditorium cable hanger — permits a microphone to be adjusted to any angle while suspended by its own cable	9.50
ST-100	Small collapsible tripod plastic table stand, with swivel	5.00
UC-11	Extension cables — available in standard lengths of 25, 50, or 100 feet	
	UC 11/25 . . . . . 25 ft extension cable	16.00
	UC 11/50 . . . . . 50 ft extension cable	21.00
	UC 11/100 . . . 100 ft extension cable	31.00
	Bulk cable	at 20¢/ft
	For cables of special lengths, compute at 20¢/ft above basic price UC-11/25	

WARRANTY

NEUMANN U-64 microphone systems are warranted unconditionally for 2 years against part failure or defects in workmanship.

HOW TO ORDER

U-64 systems can be ordered complete as such or in various combinations. Multiple systems may be purchased with fewer than the full microphone complement for later completion. The following table lists all individual U-64 components and prices.

TABLE OF U-64 COMPONENTS

DESCRIPTION:	U-64 Microphone & Acoustic Foam Anti-Pop Screen	Swivel Stand Mount ⅝" — 27 Thread	N6u Power Supply Power Cable XLR Mating Conn.	N62u Power Supply Power Cable 2 XLR Mating Conn.	N66u Power Supply Power Cable 6 XLR Mating Conn.	N66k Power Supply	UC 11/25 25 ft Interconnect Cable
INDIVIDUAL PRICE	$195.00	$4.50	$150.00	$195.00	$390.00	$270.00	$16.00
SYSTEMS							
U-64 $ 360.00	1	1	1	—	—	—	1
U-64-2 $ 620.00	2	2	—	1	—	—	2
U-64-6P $1680.00	6	6	—	—	1	—	6
U-64-6R $1560.00	6	6	—	—	—	1	6

In ordering multiple systems less certain components, add the individual prices of only those components wanted to arrive at total price.

Prices are professional net delivered and are available through franchised NEUMANN microphone dealers in the United States and Canada. Multiple systems are available to recognized sound contractors under special conditions.

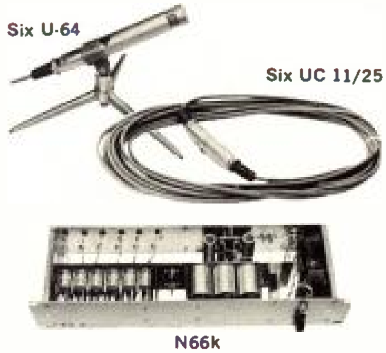
RELATED NEUMANN MICROPHONES EMPLOYING THE SAME CONDENSER ELEMENT

The KM-64 miniature condenser microphone is nearly identical in appearance and operation to the U-64. It differs in that it employs the low filament current AC-701k triode rather than the type 7586 Nuvistor used in the U-64. The connector is the familiar 6 pin miniature Tuchel permitting use with existing NEUMANN NKM power supplies and Nagra tape recorders. Furthermore the KM-64 can be powered from the NEUMANN BB-12 miniature transistorized battery supply. The KM-64 is available as microphone only or as a complete system. The cost is moderately higher than the U-64.

The SRM-64 stage microphone is an execution of the KM-64 with amplifier located in the base of a specially constructed stand. The condenser element is connected to the amplifier input by a long coaxial tube sliding within the stand to permit a wide range of height adjustment. Overload protection is not provided. Price is somewhat higher than KM-64.

A VARIETY OF PRACTICAL APPLICATIONS

Illustrated are certain specific applications of NEUMANN U-64 systems together with key components supplied with each system.



THEATRES, AUDITORIUMS AND MUSIC CENTERS

For theatre footlight usage or for suspension in large halls where quality sound and numerous microphones are the requirement, the U-64-6R system for permanent wiring including the N66k power supply illustrated and six U-64 microphones provide a perfect solution at an attractive per microphone cost figure. If permanent wiring is not contemplated, the U-64-6P with 6 way portable power supply N66u is equally applicable and provides portability as well.

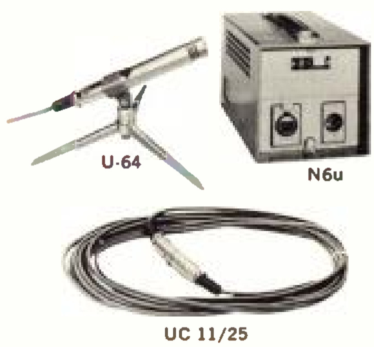
SPEAKER'S ROSTRUM AND STEREO

For ultimate reliability necessitating a pair of microphones for sound pickup, or to provide two separate feeds, the NEUMANN U-64-2 dual microphone system with N62u power supply illustrated and two U-64 microphones is ideal at a considerable saving over the cost of two single systems.

The stereo possibility with the U-64-2 system is obvious. However, it may not be so apparent to the prospective buyer of his first pair of NEUMANN microphones that precision manufacture and rigorous quality control obviate the need for matching of microphones. All microphones of one type manufactured by NEUMANN are matched most accurately as a matter of production technique.

RADIO STATION ANNOUNCE POSITIONS

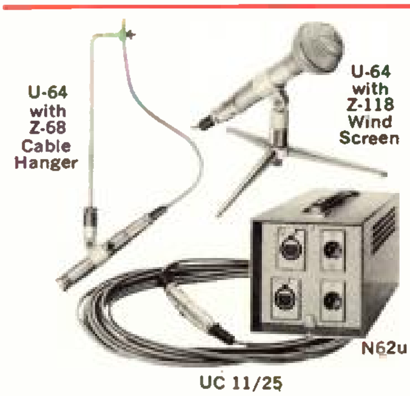
Using the acoustic foam anti-pop screen supplied, a single NEUMANN U-64 system furnishes a station with improved quality where it counts most — where it begins. Announcers need no longer sound less live than the recorded material. Employing only the necessary cardioid pattern, the U-64 completely satisfies announce requirements, and the on-air quality improvement is noticeable to listeners with even the smallest of receivers. Illustrated is the single U-64 system with included acoustic foam anti-pop screen.



RECORDING AND TELEVISION STUDIOS

Used on boom or stand for medium distance as well as more distant microphone placement, the U-64 microphone program finds versatile application.

The name NEUMANN is a household word in the Phonograph Record Industry and among independent recording studios and TV networks. NEUMANN represents a standard of quality by which other microphones and recording equipment are measured. A single U-64 system is illustrated.



PLACES OF WORSHIP

At the pulpit or suspended, the NEUMANN U-64, in single or multiple systems, can provide the quality sound now demanded in these usually troublesome sound applications. In fact, many churches faced with difficult acoustical problems, appearing to require a completely new sound installation throughout, have found a simpler and more economical solution in replacing their old microphones with NEUMANN microphones. The rest of the equipment then proved perfectly satisfactory. Illustrated is the N62u dual power supply with one U-64 on the Z-68 accessory auditorium hanger and one U-64 with Z-118 wind screen.

NEUMANN U64 TECHNICAL DATA

MICROPHONES

Acoustic Operation:	Pressure gradient transducer
Frequency Response:	40 — 18,000 Hz
Condenser Element:	
Construction:	Vacuum gold deposited Mylar with acoustical delay network
Capacitance:	App. 35 pF
Directional Characteristic:	Cardioid
Overload Protection:	Built-in 10 dB switchable
Total harmonic distortion:	⩽0.5% to 120 dB SPL, to 130 dB SPL with overload protection
Effective Output Level:	—43 dBm re 10 dyne/cm² or —137 dBm EIA Gm rating
Unweighted Noise:	⩽17 MicroVolts across 1KOhm
Weighted Noise: (Din 45405)	⩽29 Phon (Equivalent to 91 dB S/N ratio re 0.5% THD)
Output Impedance:	150/250 Ohms nominal, 30/50 Ohms at slight reduction in output level
Gain of Amplifier at 1 KC:	App. —2.5 dB
Amplifier Complement:	1 Type 7586 Nuvistor, available locally
Connector:	5-pin XLR type
Included Accessories:	Acoustic foam anti-pop screen All systems include ⅝"—27 swivel stand mount for each U-64
Finish:	Matte satin chrome, or non-reflecting dark gray (special order)
Size and Weight:	⅞" diameter x 4½" long, 3½ oz.

POWER SUPPLIES

Audio Output Connector:	3 Pin XLR type (not on N66k)
Mains Voltages:	117/127/220/240 Volts — 50/60 Hz
Fuse:	100mA at 117/127 Volts
Power Consumption:	N6u — 10 VA
DC Voltages:	6.2V (135 mA), 120V (0.9 mA)
Noise Voltages:	⩽0.4 mV, ⩽0.7, respectively
Included Accessories:	AC power cord, Mating XLR connectors (except with N66k)
Finish:	Hammertone gray
Size and Weight:	N6u Single U-64 Portable Size 8¾" x 4" x 4", 4 lbs. N62u Dual U-64 Portable Size 8¾" x 4" x 4", 4 lbs. 7 oz. N66u six U-64 Portable Size 19½" x 3½" x 7", 9 lbs. N66k six U-64 Rack Mount Supply Size 19½" x 3½" x 6⅛", 7 lbs. 3 oz.

GOTHAM
AUDIO CORPORATION
2 west 46 street
new york, n. y. 10036
(212) columbus 5-4111

LITHO IN U.S.A.

NEUMANN CONDENSER MICROPHONES

The latest addition to the NEUMANN line of Studio Microphones is the U-67 Universal Condenser Microphone — designed for utmost flexibility — ideal in any application. Among its features are: all three directional characteristics; "Voice-Music" bass cut-off switch; overload protection switch for extremely close pickups; standard EF-86 tube; instantly accessible amplifier; MYLAR double-membrane capsule; printed circuits used throughout.

Non-linear distortion:	less than 0.3% entire range	Output impedance:	50/200 ohms switchable
Output level:	— 53 dBm re 10 dyne/cm²	Field pattern:	non-directional, figure-8, cardioid
Microphone Dimension:	7⅞" x 2¼"	Front-to-back rejection (cardioid):	23 dB
Weight:	1 lb.	Front-to-side rejection (Figure-8):	greater than 30 dB

GOTHAM AUDIO CORPORATION
2 WEST 46 STREET, NEW YORK 36, N.Y. · · · COLUMBUS 5-4111

Gen. Cat.

DISCONTINUED

NEUMANN
U-47a and U-48a
Microphone Systems

These microphones have become the standard of the American Recording, Broadcasting, and Film industries, and are the only condenser microphones in their price range featuring a switchable directional characteristic.

Frequency Range:	30-20,000 cps
Output Impedance:	50/200 ohms switchable
Field Pattern:	Switchable non-directional or cardioid
Non-linear distortion:	Less than 0.9% entire range to 110 dB absolute
Effective Output Level:	— 52 dBm Minimum
Dimensions:	Microphone: 2½" dia.; 8" length. Power supply: 8½" x 4" x 4¾"
Weight:	Microphone: 1½ lbs. Power supply: 4 lbs.

Complete microphone systems consist of microphone, power supply (type NG) inter-connect cable (type UC-3), Z-37 full elastic suspension, AC power cable. U. S. fuse holder, pilot light, power connector and XLR output receptacle with mating cable connector. See price schedule for system makeups available.

U-48a System Same as above but Field Pattern selectable either **bi-directional** or cardioid.

NEUMANN
M-49b
Remote Control Directional-Pattern Microphone System

Fast becoming a favorite in international broadcasting, this microphone permits remote control of the directional characteristic. A smooth, continuous fader-control selects any of the basic directional patterns (non-directional, bi-directional, and cardioid) and any intermediate pattern. Wide frequency response with extremely low distortion, a slight roll-off of frequencies below 40 cycles to prevent shock-noise interference, and extreme ruggedness, make the M-49b the ideal "work-horse" for studio and remote recording, as well as single-mike pick ups from concert halls.

Frequency range:	30-20,000 cps
Output impedance:	50/200 switchable
Field pattern:	Continuously variable through all characteristics—omni-directional, cardioid and figure-eight and all intermediate pattern configurations
Non-linear distortion:	less than 0.3% entire range to 110 dB absolute.
Effective Output Level:	— 46 dBm
Dimensions:	Microphone: 3" dia.; 6¼" length. Power supply: 8½" x 4" x 4¾"
Weight:	Microphone: 1¾ lbs. Power supply: 5½ lbs.

Complete microphone system consists of microphone, power supply (type NN-48), interconnect cable (type C-26), AC power cable, XLR output connector, and MZ-49 swivel mounting harness.

NEUMANN
U-47a and U-48a Microphone Systems

These microphones have become the standard of the American Recording, Broadcasting, and Film industries, and are the only condenser microphones in their price range featuring a switchable directional characteristic.

Frequency Range:	30-20,000 cps
Output Impedance:	50/200 ohms switchable
Field Pattern:	Switchable non-directional or cardioid
Non-linear distortion:	Less than 0.9% entire range to 110 dB absolute
Effective Output Level:	— 52 dBm Minimum
Dimensions:	Microphone: 2½" dia.; 8" length. Power supply: 8½" x 4" x 4¾"
Weight:	Microphone: 1½ lbs. Power supply: 4 lbs.

Complete microphone systems consist of microphone, power supply (type NG) inter-connect cable (type UC-3), Z-37 full elastic suspension, AC power cable. U. S. fuse holder, pilot light, power connector and XLR output receptacle with mating cable connector. See price schedule for system makeups available.

U-48a System Same as above but Field Pattern selectable either **bi-directional** or cardioid.

NEUMANN
M-49b Remote Control Directional-Pattern Microphone System

Fast becoming a favorite in international broadcasting, this microphone permits remote control of the directional characteristic. A smooth, continuous fader-control selects any of the basic directional patterns (non-directional, bi-directional, and cardioid) and any intermediate pattern. Wide frequency response with extremely low distortion, a slight roll-off of frequencies below 40 cycles to prevent shock-noise interference, and extreme ruggedness, make the M-49b the ideal "work-horse" for studio and remote recording, as well as single-mike pick ups from concert halls.

Frequency range:	30-20,000 cps
Output impedance:	50/200 switchable
Field pattern:	Continuously variable through all characteristics—omni-directional, cardioid and figure-eight and all intermediate pattern configurations
Non-linear distortion:	less than 0.3% entire range to 110 dB absolute.
Effective Output Level:	— 46 dBm
Dimensions:	Microphone: 3" dia.; 6¼" length. Power supply: 8½" x 4" x 4¾"
Weight:	Microphone: 1¾ lbs. Power supply: 5½ lbs.

Complete microphone system consists of microphone, power supply (type NN-48), interconnect cable (type C-26), AC power cable, XLR output connector, and MZ-49 swivel mounting harness.


NEUMANN


NEUMANN

NEUMANN

KM-54a & KM-56
Miniature Condenser Microphone System
(KM-56—shown at left)

This miniature condenser microphone is one of the latest to join the precision Neumann line. Its quality is in every way similar to the U-47 series, but its dimensions are amazingly miniaturized.

Frequency Range:	30-20,000 cps
Output Impedance:	50/200 ohms (must be
Field Pattern:	Switchable on microphone: non-directional, bi-directional, and cardioid
Non-linear distortion:	Less than 0.4% entire range to 110 db absolute
Effective Output Level:	—45 dBm Minimum
Dimensions:	Microphone: 7/8" dia. 6" length. Power supply: 8½" x 4" x 4¾"
Weight:	Microphone: 4 oz. Power supply: 5 lbs.

Complete microphone system consists of microphone, power supply (type NKM), interconnect cable (type KC-1), SG-5 Microphone Stand Coupling, AC power cable. U. S. fuse holder, pilot light, power connector, and XLR output receptacle, with mating cable connector.

KM-54a Miniature Condenser Microphone System (shown at right) Same as above but ultra cardioid directional pattern only. Length: 4¾".

NEUMANN

Type SM-2 Miniature
Stereo Double Microphone System

This latest addition to the condenser microphone field comprises two separate and complete condenser microphones and their respective preamplifiers in the same miniature housing. The two condenser capsules are mounted one above the other, the top one being rotatable to achieve the M-S Stereo Recording Technique (also known as **intensity stereo**). Each of the two microphone systems can be separately switched to any pattern (non-directional, bi-directional, and cardioid) or any one of six intermediate patterns. Specifications identical to KM-56 microphone with addition of extreme balance between systems, and numerous intermediate directional patterns.

Dimensions:	Microphone: 1⅛" dia.; 8" length. Power supply: 8½" x 4" x 4¾"
Weight:	Microphone: 9½ oz. Power supply: 5 lbs.

Both Sections:

Frequency range:	30-20,000 cps
Output Impedance:	50/200
Directional pattern:	Remote control switchable to omni-directional, cardioid, and figure-8 as well as six intermediate patterns
Harmonic Distortion:	less than 0.4% entire range to 110 dB absolute.
Effective Output Level:	—43 dBm

Complete microphone system consists of microphone, power supply (type NSM), inter-connect cable (type SC-1) with Z-42 full elastic suspension, AC power cable, XLR Cannon output connectors. U. S. standard pilot light, fuse holder, AC receptacle.

MICROPHONE ACCESSORIES

FOR U-67 MICROPHONE:

UC-5	Interconnect extension cables in 25, 50, and 100 ft. lengths or to special order.
EF-86	Microphone amplifier tube.
NUK	Plug-in power supply for permanent control room installation.

FOR U-47 or U-48 MICROPHONES:

UC-3	Interconnect extension cables in 25, 50, and 100 ft. lengths or to order.
Z-37	Full elastic suspension for elimination of mechanical interference.
Z-18a	Wind and close talking screen.
VF-14M	Microphone amplifier tube (specially selected for low noise).
NG-2	Double power supply for (2) U-47 or U-48 microphones.
NGK	Plug-in power supply for permanent control room installation.

FOR KM54a and KM-56 MICROPHONES:

KC-1	Interconnect extension cables in 25, 50, and 100 ft. lengths or to order.
Z-38	Full elastic suspension for elimination of mechanical interference.
Z-118	Wind and close talking screen.
M-31b	Extendable floor stand with internal shock mounting and goose neck.
Z-29	High intensity overload protection switch for KM-54a.
Z-64	Humidity-proof jeweler's case for KM-series microphones for tropical use.
AC-701k	Microphone amplifier tube (specially selected for low noise).
SG-5	Swivel stand adapter with 5/8-27 thread.
Z-19	Flexible goose-neck extension (wired).

FOR M-49b and M-50b MICROPHONES:

C-26	Interconnect extension cables in 25, 50, and 100 ft. lengths or to order.
AC-701k	Microphone amplifier tube (specially selected for low noise).
N-52a	Plug-in power supply for permanent control room installation. Also usable for KM-series microphones.

FOR SM-2 STEREO MICROPHONE:

SC-1a	Interconnect extension cables in 33 ft. length or to special order.
Z-42	Full elastic suspension for elimination of mechanical interference.
Z-43	Wind and close talking screen.
Z-140	Sum and difference (matrixing) transformers for converting "M-S" to "A-B" or "X-Y" or vice versa.
SG-6	Swivel stand adapter with 5/8-27 thread.
NSK	Plug-in power supply for permanent control room installation.

GENERAL ACCESSORIES:

Z-30	Portable power supply protective enclosure for field use.
CF-3	Studio "Stand-by" and "Go-ahead" signal; attaches to microphone stand.
BB-9k	Battery operated power supply for KM-series, M-49b, and M-50b microphones complete with rechargeable nickel-cadmium batteries and charger.

Other accessories and replacement parts quoted on request. All accessories stocked in New York, N. Y. Price lists published on October 1 and April 1 of each year.

For further information, contact Gotham Audio Corporation or your Neumann dealer!

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NEUMANN FET CONDENSER MICROPHONES

A SUPERIOR INSTRUMENT DESIGNED FOR THE PROFESSIONAL

THE STORY OF NEUMANN CONDENSER MICROPHONES - OLD AND NEW

When you consider the purchase of a NEUMANN microphone, you are close to joining a tradition in manufacturing which dates back over 40 years. The NEUMANN line of condenser microphones represents both the oldest and the newest in microphones. It is the oldest continuously manufactured line of such microphones in the world, and it is the leader among condenser microphones at this very moment. Georg Neumann started building condenser microphones (and nothing but condensers) back in 1926 and the Neumann Company is still delivering spare parts for units in service since 1930.

NEUMANN microphones were sold prior to World War II in the U.S.A., but it wasn't until the U-47 (also known by the name of its then distributor, Telefunken) made its appearance in 1948 that NEUMANN became the indispensable transducer with the most profound influence on the change in recording fashion in the post World War II period.

The U-47 model was practically the only one sold in the United States up to 1959, when Gotham Audio Corporation became the exclusive U. S. and Canadian distributor for NEUMANN, and introduced such other models as the M-49, the KM series of miniature microphones, the SM-2 and SM-69 M-S stereo models, and most recently, the U-64 linear admittance cardioid. All of these models have one important thing in common: NEUMANN quality. The differences between them are strictly in their applications and never really one of frequency response, distortion or noise level. Then, as now, the only thing which dictates which model you should use is the use to which you are going to put the microphone. Alas, all of the above models have one other thing in common: They are all quite expensive when compared to even the most expensive dynamic and ribbon types, but almost 6000 units sold in the U.S.A. in the past ten years testify to their indispensable service to recording, broadcasting, public address, scientific use, etc., etc.

Now the field effect transistor, one of the components of the space age, has brought about a most sensational revolution in the world of condenser microphones. It permits the NEUMANN acoustical quality to continue undiminished, while simultaneously reducing the price to the consumer by as much as 30%. The FET-80 series now makes it possible for everyone to "step up" to NEUMANN condenser quality.

The FET-80 series, introduced to the U. S. market in January 1968, after almost two years of testing on the European market, is the world's only transistor electronics microphone devoid of RF circuitry, RF DC converters, and batteries. Its straightforward design is as much the result of waiting for the right moment in electronics history as it is the product of great laboratory ingenuity. NEUMANN had its first transistor equipped microphone ready and operating as early as every other manufacturer, but was convinced that these early -- these too early designs would result in no benefits to the consumer but would rather lead to early obsolescence. And that is something which NEUMANN users have long respected about the NEUMANN Company: Model changes are made only when breakthroughs are at hand; never for commercial reasons alone.

HOW CAN I TELL A FET-80 MICROPHONE FROM A TUBE TYPE?

There are only two distinguishing features which allow you to tell a FET-80 microphone from its corresponding tube model. The NEUMANN insignia mounted on the FET-80 series is a purple color while the tube types have black insignia. All FET-80 microphones are equipped with Cannon type 3-pole male output connectors while their tube counterparts have at least 5-pole connectors and are for the most part supplied with German Tuchel plugs.

WHAT MAKES THE FET-80 SERIES UNIQUE?

First and most important is the fact that the introduction of transistor electronics to NEUMANN tube microphones in no way alters their acoustical quality. The principal advantage of FET-80 lies in the service-free nature of silicone planar field effect transistors on the one hand, and the tremendous simplification of the power source needed on the other. The latter brings with it significant savings in dollars and cents as well as the ability to provide "compatible central powering," easy battery operation, and the use of standard shielded single pair microphone extensions throughout.

HOW DID THE FET MAKE ALL THIS POSSIBLE?

Of all the types of transistors in use today, the FET more closely resembles the vacuum tube in terms of input impedance than any other. Since it has no "heater," it requires very little power and produces no heat. In fact, a single voltage source suffices to provide both polarizing voltage for the condenser element and operating voltage for the FET itself.

WHAT SORT OF POWER IS NEEDED FOR THE FET-80 MICROPHONES?

While the tube equipped NEUMANN microphones all needed heater power of as much as 850 mW and plate power of another 60-110 mW, the FET-80 series needs only about 19 mW of total power to operate; roughly 1/50th the former power requirement. The nominal powering voltage is 48 Volts DC at 0.4 mA, but the microphones will perform satisfactorily between 42 - 52 Volts. This permits the use of two 22½ V. batteries in series, with a battery life of over 200 hours, or less than 1¢ per operating hour! The use of batteries, however, is never recommended for fixed studio operation where adequate AC supply is available. A variety of AC supplies is available from NEUMANN for such purposes.

HOW DOES THE VOLTAGE REACH THE FET-80 MICROPHONE?

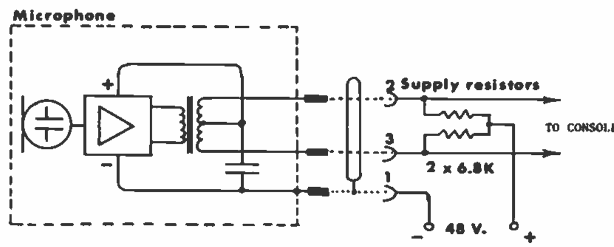
In designing the FET-80 series, the NEUMANN research engineers and scientists were intent on providing the following advantages and features:

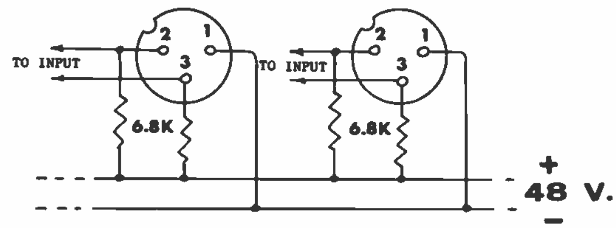
1) Use of standard 2-conductor shielded microphone cables everywhere.

2) Use of power supplies requiring only routine regulation and filtering.

3) Ability to power a large number of FET-80 units from a single centrally located supply, using standard 3-pole studio microphone outlets which could then be used interchangeably for any other microphones.

The FET-80 series has fully realized all of these advantages through the use of "compatible central powering," also referred to as M-S or "phantom" powering. In this technique the DC voltage is applied (or simplexed) between the centertap of the audio circuit and the ground line or shield connection as shown in the schematic below.





CENTRAL POWERING OF TWO MICROPHONE OUTLETS

This unique kind of powering totally separates the DC from the audio voltage signal on the cable. Any other microphone plugged into an outlet so powered would see only +48 Volts DC on BOTH of the audio leads. Since this could not lead to any current flow, it would have no influence on the proper operation of any such other microphones. At the FET-80 microphone, the center tap of the output transformer carries this positive (+) potential with respect to the cable shield. Equal current flow through both halves of the output transformer secondary cancels any saturating effect on the transformer itself. Each FET-80 microphone contains its own stabilizing and filtering network, thus making critical filtering of the supply itself unnecessary.

WHAT SORT OF AC SUPPLIES ARE AVAILABLE?

NEUMANN currently produces three types of AC supplies:

1) N-452 Dual Supply: Provides sufficient power for one or two FET-80 microphones. Equipped with Cannon type connectors and attached 3-wire AC cable.

2) GW-2448 DC Converter: A plug-in card permitting an existing 24 Volt DC supply which is available to be raised to the necessary 48 Volts. Supplies up to 20 microphones.

3) NK-48 Central Supply: A plug-in printed circuit card and frame for installation anywhere. Will power up to 40 studio outlets for FET-80 microphones. Two NK-48 cards may be so interconnected as to provide back-up protection in case one card should fail in service. The other NK-48 will take over unaffected by any short circuits in the defective supply.

WHAT SORT OF BATTERY SUPPLIES ARE AVAILABLE?

1) BS-45 Supply Box: This miniature metal box contains two 22½ Volt batteries and will power one FET-80 microphone for over 200 hours.

MAY I SUPPLY MY OWN 48 VOLTS FROM OTHER SUPPLIES?

The answer obviously is "yes;" BUT experience has shown that there are many supplies made which are not manufactured using sufficiently dependable components. The results may be instant burn-out of the FET transistor, should voltage ever exceed 55 Volts, even for a microsecond. The supply must have good load regulation and must have less than 5 mV ripple. The safest way is to use the supplies built by NEUMANN for the purpose.

WHAT ABOUT NOISE AND OVERLOAD IN THE FET-80 SERIES?

This is one question which is asked more frequently than any other. There have been numerous transistor equipped microphones built in recent years, most of which have left much to be desired from the standpoint of both noise and distortion. NEUMANN would never have considered superseding the long-standing tube models had their specs not been equaled or bettered by the FET-80 series. Noise in all FET-80 types is LOWER than in any tube type they supersede. Overload characteristics of the FET-80 series exceed those of their tube counterparts. NEUMANN considers a 65 dB signal-to-noise ratio from 10 microbar (94 dB SPL) the minimum. FET-80 microphones all measure 68 dB or better.

WHAT IS THE POLARITY (PHASING) OF THE NEW FET-80 SERIES?

There is unfortunately no standard as yet established for the proper phasing of microphones equipped with anything but the "UA" type connector. Since all NEUMANN microphones, both tube type and FET-80 series, are equipped with XLR type connectors, the proper phasing must be stated.

> "ALL NEUMANN microphones equipped with Cannon or Switchcraft XLR type audio connectors have their positive (+) pole on PIN 2. This means that an increasing sound pressure on the membrane directed toward the sound source will produce a POSITIVE voltage surge on PIN 2 of the XLR type connector. For those NEUMANN microphones equipped with European connectors, the positive (+) pin is PIN 1. All XLR type connectors have PIN 1 grounded. European connectors have PIN 3 grounded."

(NOTE: There is one exception to the above: M-49b or c, and M-50b or c models have the reverse polarity. This does NOT apply to M-249 or M-250 units which are equipped with screw ring connectors but solely to M-49/M-50 units using large size bayonet cable connectors.)

NEUMANN ALSO MAKES THE FET-70 SERIES. WHAT IS THAT?

The FET-70 series was designed for use with existing equipment, mostly battery operated, in which supplies of between 7 and 24 Volts are available. These microphones DO NOT use compatible powering and are NOT recommended for any use but such battery operation. The most usual application is the NAGRA portable tape recorder, to which any FET-70 microphone may be connected, using the NEUMANN SVN extension cable which serves both to connect the NAGRA supply to the microphone and the microphone's output to the line input of the NAGRA, leaving its microphone input free for other microphones. FET-70 microphones are equipped with Tuchel connectors and use the KT-1 type extension cables.

neumann transistor condenser microphones fet 80

GOTHAM

AUDIO CORPORATION

2 WEST 46 STREET, NEW YORK 36, N. Y. • 212-CO 5-4111

Table of contents

NEUMANN condenser microphones have been around for over 40 years. The Georg Neumann Company has been building them since 1928 and was the very first company ever to manufacture condenser microphones of their own design. The well known „Neumann bottle", the prototype of condenser microphones of that era, aside from achieving the highest respect of earlier professionals, became the firm foundation of quality on which their reputation is founded throughout the world. Through continuing close cooperation between Neumann and the users of their microphones, a series of models have evolved which serve many special purposes in the broadcasting and recording industry. The well known models U 47 and M 49, featuring switchable directional characteristics, appeared at the end of the forties and have remained much in demand throughout the world to this day. The miniature microphone series was introduced during the early days of television in Europe, and in 1960 there followed the model U 67 which has become the standard of the industry, featuring switchable directional characteristics, overload protection and low frequency reduction to permit the widest adaptability to difficult situations.

A new type of directional condenser capsule bowed in 1964 and has since become standard on nearly all of the miniature microphone types. Its primary advantage lies in the fact that not only direct incident sound, but sound impinging at a wide angle from the front, as well as diffused sound, is reproduced linearly up to the highest frequencies. This development has made it possible for the engineer to select his microphone positions solely on the basis of reverberation and acoustical balance, since the resulting microphone frequency response no longer undergoes change with changing distance from the sound source.

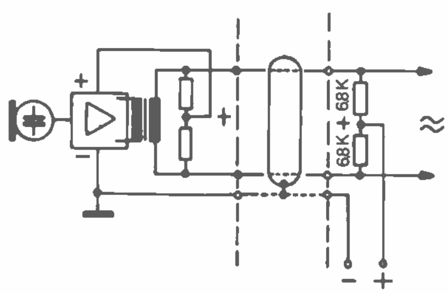
In spite of a growing tendency for the transistorizing of electronic devices, the condenser microphone seemed destined to remain in the domain of tube technology. The conventional circuitry involved in condenser microphones requires an amplifier with extremely high input impedance, something which transistors appeared not to be able to offer. The advent of the Field Effect Transistor (FET) with its high input impedance has created a component which has made conventional transistor amplifiers possible.

The Georg Neumann Company today builds two separate series of transistor condenser microphones whose models are acoustically in every way identical, but which differ with respect to powering and connection. Microphones for both the 12 Volt modulation lead powering (FET-70 series), and compatible phantom powering (FET-80 series) are manufactured. This catalog is designed to show you the microphones of the FET-80 series, intended for 48 Volt phantom powering. A separate catalogue covers the microphones in the FET-70 series.

PHANTOM POWERING

This form of powering feeds the positive pole of the powering source, via the electrical middle of the two modulation leads, to the microphone. This is accomplished by either connecting it to the center tap of the following input transformer or by feeding it to both modulation leads through two identical value resistors.

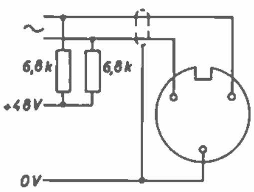
The powering DC voltage (including any superimposed noise voltage) is, in effect, fully decoupled from the microphone output signal. The return (minus pole) goes through the shield. The 48 Volt operating voltage was selected because of the convenience and stability of circuitry involved.



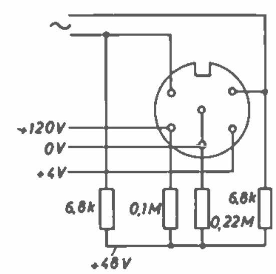
COMPATIBILITY OF THE 48 V PHANTOM POWERING WITH OTHER TYPES OF MICROPHONES

Although new constructions of studios and remote facilities will generally be equipped solely with transistor condenser microphones, it is highly likely that tube microphones as well as dynamic and ribbon types will also find employment.

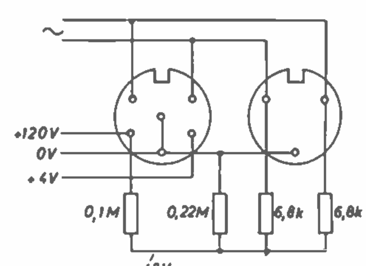
The 48 V Phantom Powering System provides a fully compatible installation since no voltage differential appears between the two modulation leads. Studio outlets so powered may be connected to any other type of microphone without causing the slightest noise interference or deterioration of signal.



Consoles already equipped for the powering of tube condenser microphones may be altered by the simple installation of four 1/10 Watt resistors to permit connection of either the tube types or FET-80 Series microphones. The 48 Volts needed for powering is obtained from a voltage divider across the 120 Volt plate supply and is then applied via two 6.8 kΩ resistors to the two modulation leads.



One can then decide whether the studio is to be equipped with 3-pole connectors for the FET-80 types, each associated with a 6- or 7-pole tube type connector, or whether all microphones used,regardless of which type, are to be converted to 6- or 7-pole configuration to permit a single type of connector for all microphones employed.



TRANSISTOR CONDENSER MICROPHONE U 87 i

The Condenser Microphone U 87 i is a studio microphone of highest quality whose principal advantages are excellent pickup characteristics, a number of special features, and a most attractive form.

The U 87 i is eminently suited to numerous applications in recording, broadcasting, TV, and film. It corresponds in both its acoustical characteristics and dimensions to the well known and popular forerunner model U 67. Its amplifier, however, is equipped with a field effect transistor in place of a tube.

The condenser elements have evaporated gold coated polyester membranes, which are both heat insensitive and age resisting. Below the microphone element there are three separate switches, controlling directional characteristic, frequency response and sensitivity, thereby permitting the flexibility to make the U 87 i useful in a great variety of applications. The high frequency response in both the „cardioid" and „figure-8" positions is linear even for direct incident sound fields. This allows the U 87 i to be used at extremely close distance to the sound source without an accompanying peaked and sibillant sound quality. A frequency dependent feedback ciruit reduces response below 30 Hz at the gate of the FET, while keeping response above 40 Hz linear. The much feared „blocking" of the amplifier caused by wind, mechanical shock, etc. resulting in excessive membrane excursion, is thereby virtually eliminated. This cannot be achieved by the more usual, but ineffective, low frequency filtering a f t e r the microphone preamplifier.

The 3 dB point of the low frequency roll-off can be raised to 200 Hz by means of a switch on the microphone itself. This is especially useful for close pickup of vocalists, speakers, or for elimination of room noise in TV studios. This switch will also restore low frequency linearity at the amplifier input, whenever the proximity of the sound source gets down to a foot or less, causing low end boost, as is the case in all pressure gradient microphones (cardioid or figure-8). A second switch below the condenser element reduces sensitivity by 10 dB ahead of the FET, making the U 87 i virtually impervious to the highest sound pressure levels imaginable. This allows the microphone to be brought within inches of a piano, vocalist or brass instrument.

The microphone is easily opened by unscrewing the ring at its base. This provides access for the purpose of installing two 22½ Volt batteries, making the U 87 i independent of any external power source. Batteries of the IEC 15 F 15 type are used which will power the microphone for more than 150 hours.

O ∞ 8

U 87 i

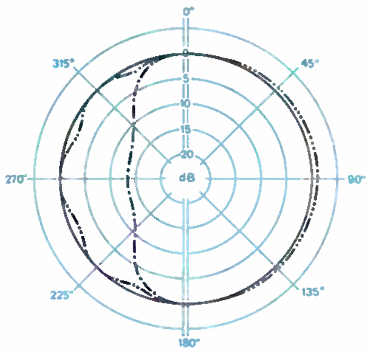
U 87, opened

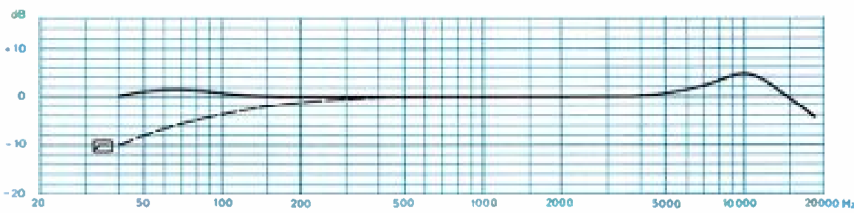
Type „i" for international applications
Cannon connector equipped

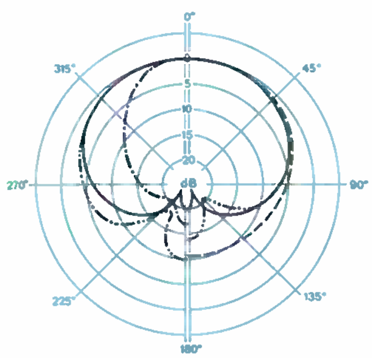
Technical specifications U 87 i

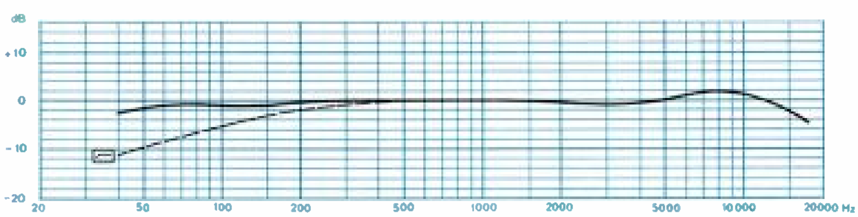
Acoustical operating principle	pressure gradient
Directional characteristics	omni, cardioid, figure-8
Frequency range	40 - 16,000 Hz
Sensitivity	approx. 1 mV/μbar (across 1000 Ω) - 38 dBm effective output level re 10 μbar
EIA sensitivy (G_m)	- 137 dBm
Matching impedance	250/1000 Ω
Source impedance	switchable 50/200 Ω
Capsule capacity	2 × approx. 53 pF
Self noise level (DIN 45 405)	approx. 26 dB (re 2 × 10^{-4} μbar)

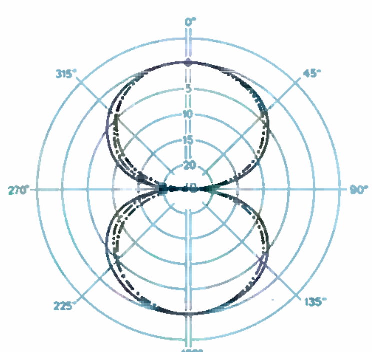
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz	≧ 200 μbar ≙ 120 dB
with sensitivity reduct.	≧ 650 μbar ≙ 130 dB
Operating voltage	48 VDC + 6 V - 8 V
Current consumption	0.4 mA
Battery life of external battery supply	≧ 200 hours
Weight	550 g (20 oz.)
Dimensions	2¼" dia. × 8" long





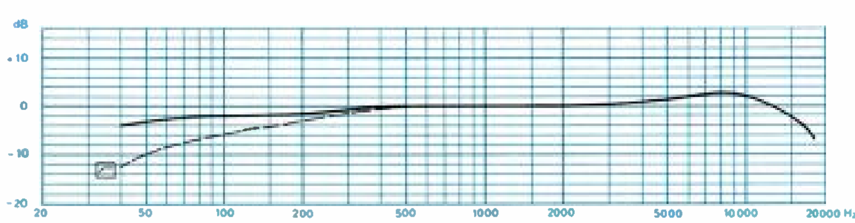






125 Hz
1 kHz
4 kHz
8 kHz
12.5 kHz

Curves not shown are identical to the 1 kHz curve



Power Supplies:

The U 87 i microphone may be powered by the following power supplies: (* equipped with Cannon-type audio connectors)

*N 451 i An AC power supply (4½" × 3¼" × 1¼") for operating one microphone from power lines. The power transformer may be selected to operate from between 100 V and 240 V.

*N 452i An AC power supply (4½" × 3¼" × 1¼"") for operating two microphones from power lines. The power transformer may be selected to operate from between 100 V and 240 V.

N 45 k A casette (plug-in) supply of unit size 1 (same as N 52) fitting the standard plug-in shelf. This supply can power up to 40 microphones. It delivers 48 Volts which is fed to the microphone outlets via two selected precision 6.8 k Ω resistors. These resistors may be ordered matched in pairs.

NK 48 An AC power supply mounted on a 31-pin printed circuit card mating with the Siemens C 42 334-A 11-A 1 frame. This supply is intended for installation in consoles and is electrically equivalent to the N 45 k supply. It is connected to the microphone outlets identically to the N 45 k supply above.

GW 2448 A supply likewise mounted on a 31-pin print card, and intended for use where 24 Volt DC is available; such as in outside mobile units and portable consoles. The GW 2448 is a transformer-less DC converter which uses a multivibrator to step up the available 24 VDC to the required 48 VDC. As is the case with both the N 45 k and NK 48 supplies, the GW 2448 can supply up to 40 microphones and is similarly connected to the microphone outlets.

*SW 45 i A powering branch-off which mounts besides the Cannon-type microphone outlet the two 6.8 k Ω resistors required for powering microphones from the N 45 k, NK 48, and GW 2448 supplies.

*BS 45i Even though the U 87 i microphone can accommodate two batteries within it, it may sometimes be desirable to power it from an external battery supply. The BS 45 i battery box is provided for that purpose and provides space for two IEC 15 F 20 22½ Volt batteries and will provide power for more than 200 hours. The BS 45 i dimensions are 3½" × 2" × 1¼".

U 87 i

MICROPHONE CABLES:

The U 87 i microphone is equipped with a Cannon type 3-pin male connector, and the power supplies fitting it carry the Cannon type 3-pin female chassis connector. The following cables are available:

IC 3 A 10 m (33 ft.) cable without stand mount swivel for connecting the microphone to its power supply (for example when using the Z 48 elastic suspension). It may also be used as an extension cable.

SG 367 A stand mount swivel which permits the U 87 i microphone connected to an IC 3 cable to be mounted on a microphone stand or boom.

Tuchel connector version

The U 87 i microphone is also available equipped with the Tuchel NT 3470 7-pin male base connector, while the appropriate power supplies are equipped with the Tuchel T 3263 3-pole connector. The microphone then carries the designation U 87. The following cables serve to connect the U 87 microphone with its power supplies:

UC 73 A 10 m (33 ft.) cable with stand mount swivel to connect the microphone with its power supply.

KC 73 A 10 m (33 ft.) cable without stand mount swivel for connecting the microphone to its power supplies (for example when used with the Z 48 elastic suspension).

KT 1 A 10 m (33 ft.) extension cable that may be inserted between either the UC 73 or KC 73 cables and the power supply. The KT 1 cable may NOT be connected directly to the U 87 microphone.

UC 073 A 10 m (33 ft.) cable with stand mount swivel for connecting the microphone to a microphone outlet of the T 3082 type. For battery or central powering uses.

MC 3 A 10 m (33 ft.) extension cable which may be inserted between the UC 073 and a microphone outlet of the T 3082 type. The MC 3 cable may NOT be connected directly to the U 87 microphone.

U 87 I, SG 367, IC 3

POWER SUPPLIES FOR THE U 87 MICROPHONE:

N 451 An AC power supply to power one U 87.

N 452 An AC power supply to power one or two U 87 microphones.

N 45 k
NK 48
GW 2448
These central supplies may power up to 40 U 87 microphones.

SW 45 A powering branch off which mounts, besides the microphone outlet T 3263, the two 6.8 kΩ resistors required for powering microphones from the N 45 k, NK 48, and GW 2448 supplies.

BS 45 Battery supply to power the U 87 for more than 200 hours

KM 83 i

KM 83 i, MTS 21, IC 3

MINIATURE TRANSISTOR CONDENSER MICROPHONE KM 83 i

The miniature microphone KM 83 i is a pressure transducer with an evaporated gold coated polyester membrane and an omni-directional characteristic. Its frequency is linear except for a purposeful slight rise at the high frequency end. The microphone is therefore most often applied to the over all pickup of large orchestras. The KM 83 i has an amplifier section identical to that of the model KM 84 i and therefore its omni-directional condenser element may easily be replaced by the cardioid element of the KM 84 i. In order to prevent overload of the microphone preamplifier resulting from the pickup of high level solo instruments, a 10 dB switch on the body of the microphone reduces sensitivity at the gate of the FET. This allows even the highest sound levels encountered in practice to be reproduced without distortion.

Technical specifications KM 83 i	
Acoustical operating principle	pressure transducer
Directional characteristic	omni directional
Frequency range	20 ... 20 000 Hz
Sensitivity	approx. 1 mV/µbar (across 1000 Ω) -38 dBm effective output level re 10 µbar
EIA sensivity (G_m)	-137 dBm
Matching impedance	≧ 1000 Ω/250 Ω
Source impedance	200 Ω/50 Ω
Capsule capacity	approx. 43 pF
Self noise level (DIN 45 405)	approx. 25 dB (re 2×10^{-4} µbar)
Weighted noise level (DIN 45 405)	≦ 4 µV peak value
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz with sensitivity reduct.	≧ 200 µbar ≙ 120 dB ≧ 650 µbar ≙ 130 dB
Operating voltage	48 VDC +6 V −8 V
Current consumption	0.4 mA
Battery life	≧ 200 hours
Weight	80 g (3 ozs.)
Dimensions	7/8" dia. × 4 3/8" long

Microphone cables for the KM 83 i

IC 3 — 10 m (33 ft.) extension cable connects the microphone with its power supply. The SG 21 clamp is used as the microphone stand mount.

Power supplies for the KM 83 i

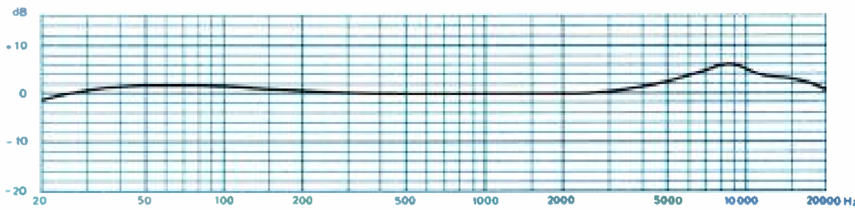
N 451 i — AC power supply to power one microphone

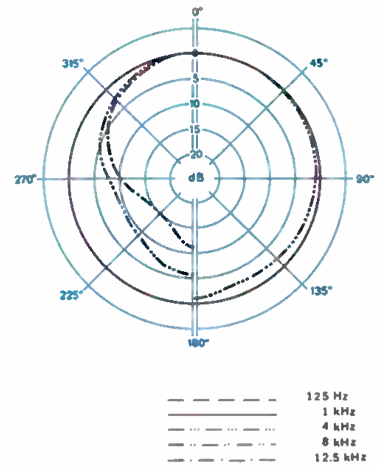
N 452 i — AC power supply to power one or two microphones

N 45 k, NK 48, GW 2448 — These central supplies may power up to 40 microphones

SW 45 i — Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45 i — Battery supply to power one microphone for more than 200 hours





Curves not shown are identical to the 1 kHz curve

SW 45

Type „i" for international applications Cannon connector equipped

BS 45 i

Tuchel connector version

The KM 83 i is also available equipped with Tuchel connector at its base. It then carries the designation KM 83.

Microphon cables for KM 83

KT 1 10 m (33 ft) extension cable for connecting the microphone with its power supply.

KT 2 10 m (33 ft) cable with stand mount swivel for connecting the microphone with its power supply.

Power supplies for KM 83

N 451 AC power supply to power one microphone

N 452 AC power supply to power one or two microphones

N 45 k
NK 48
GW 2448 These central supplies may power up to 40 microphones

SW 45 Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45 Battery supply to power one microphone for more than 200 hours.

KM 84 i

KM 84, KT 2, MF 1

Type „i" for international applications
Cannon connector equipped

MINIATURE TRANSISTOR CONDENSER MICROPHONE KM 84 i

The miniature condenser microphone KM 84 i features a cardioid characteristic. Its special advantages are a virtually frequency independent directional characteristic and extended high frequency response. Its miniature size makes it especially suited to applications in TV and film studios.
The microphone has a condenser element designed as a pressure gradient with acoustical delay network. The membrane is coated with evaporated gold over a polyester foil. Response extends from 40 Hz to 20 kHz and a gradual roll-off below 40 Hz compensates for the low frequency boost inevitable in all pressure gradient devices for sound sources close to the microphone. In its upper response range, the KM 84 i has a rise of about 2 dB.
Development of the KM 84 capsule had as one of its principal demands, the creation of a directional characteristic as nearly frequency independent as possible. Its resulting response at an angle of ± 135 degrees from front is nearly parallel to the 0° on axis curve. Attenuation at 135° is about 15 dB between 100 Hz and 18 kHz. The result of this unique feature is that a sound source impinging on the microphone anywhere within a ¾ arc around it will have the same reproduction sound quality, albeit at different levels. This happens, for example, if a vocalist is on-mike while the band is off-mike, or while a number of people in an interview all are at varying angles to the microphone, or an actor turns his head or changes his position in a dramatic presentation. A further application is in the post-sync studio where sound quality can change significantly as an actor changes his head position between reading of his script and looking at the projection screen.

In order to prevent overload of the microphone preamplifier resulting from the pickup of high level solo instruments, a 10 dB switch on the body of the microphone reduces sensitivity at the gate of the FET. This allows even the highest sound levels encountered in practice to be reproduced without distortion.

Technical specifications KM 84 i

Acoustical operating principle	pressure gradient
Directional characteristic	cardioid
Frequency range	40 ... 20 000 Hz
Sensitivity	approx. 1 mV/µbar (across 1000 Ω) -38 dBm effective output level re 10 µbar
EIA sensitivity (G_m)	-137 dBm
Matching impedance	≧ 1000 Ω / 250 Ω
Source impedance	200 Ω / 50 Ω
Capsule capacity	approx. 34 pF
Self noise level (DIN 45 405)	approx. 25 dB (re 2×10^{-4} µbar)
Weighted noise level (DIN 45 405)	≦ 4 µV peak value
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz	≧ 200 µbar ≙ 120 dB
with sensitivity reduct.	≧ 650 µbar ≙ 130 dB
Operating voltage	48 VDC $^{+6\,V}_{-8\,V}$
Current consumption	0.4 mA
Battery life	≧ 200 hours
Weight	80 g (3 ozs.)
Dimensions	$^7/_8$" dia × $4^3/_8$" long

Microphone cables for the KM 84i

IC 3 — 10 m (33 ft.) extension cable connects the microphone with its power supply. The SG 21 clamp is used as the microphone stand mount.

Power supplies for the KM 84i

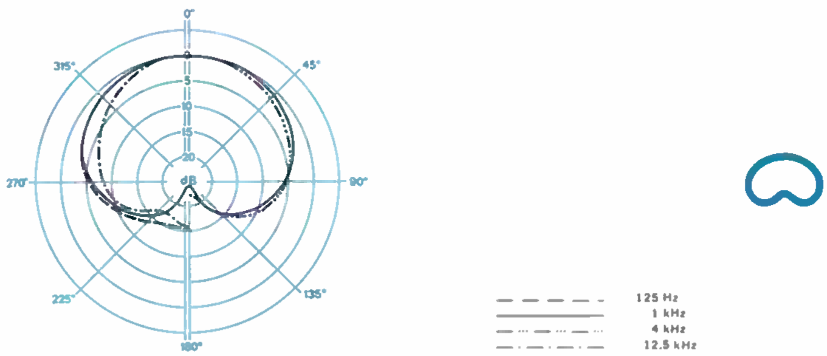
N 451i — AC power supply to power one microphone

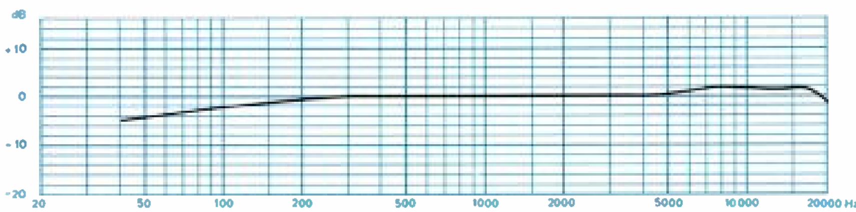
N 452i — AC power supply to power one or two microphones

N 45 k, NK 48, GW 2448 — These central supplies may power up to 40 microphones.

SW 45i — Powering branch-off for powering microphones from the N 45k, NK 48 and GW 2448 supplies

BS 45i — Battery supply to power one microphone for more than 200 hours





Tuchel connector version

The KM 84i is also available equipped with Tuchel connector at its base. It then carries the designation KM 84.

Microphone cables for KM 84

KT 1 — 10 m (33 ft) extension cable for connecting the microphone with its power supply.

KT 2 — 10 m (33 ft) cable with stand mount swivel for connecting the microphone with its power supply.

Power supplies for KM 84

N 451 — AC power supply to power one microphone

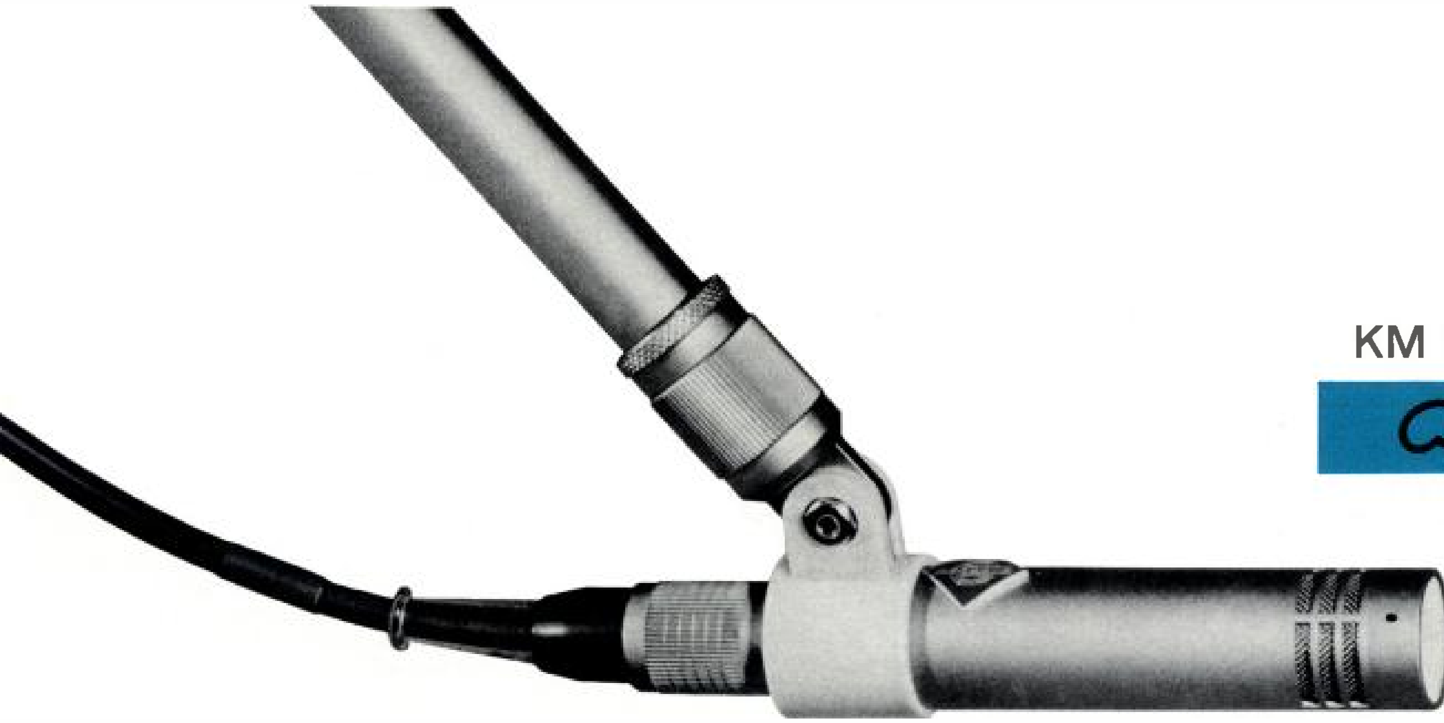
N 452 — AC power supply to power one or two microphones

N 45 k, NK 48, GW 2448 — These central supplies may power up to 40 microphones.

SW 45 — Powering branch-off for powering microphones from the N 45k, NK 48 and GW 2448 supplies

BS 45 — Battery supply to power one microphone for more than 200 hours.

BS 45 opened



KM 85 i

KM 85, SG 21, KT 1

Type „i" for international applications
Cannon connector equipped

MINIATURE TRANSISTOR CONDENSER MICROPHONE KM 85 i

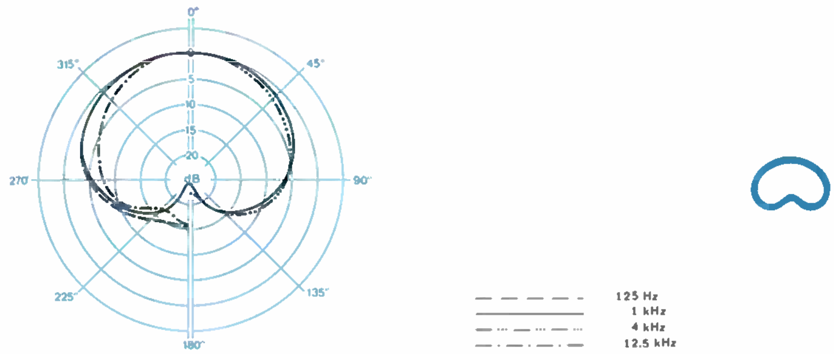
The miniature microphone model KM 85 i is a special version of the KM 84 i. Its frequency response is rolled off at the low frequency end reaching an attenuation of 12 dB at 50 Hz. As a result the microphone is largely insensitive to low frequency interference. It is therefore widely used in public address applications such as theater footlights, close talking, recording outdoors, and for use on booms in film and TV studios where rapid motion is anticipated.

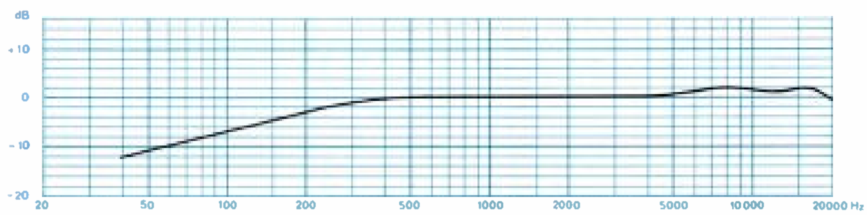
In order to prevent overload of the microphone preamplifier resulting from the pickup of high level solo instruments, a 10 dB switch on the body of the microphone reduces sensitivity at the gate of the FET. This allows even the highest sound levels encountered in practice to be reproduced without distortion.

The condenser elements of the KM 83, KM 84 and KM 85 microphones are mechanically identical and may all be readily mounted on one and the same amplifier. The designations of the condenser elements are KK 83, KK 84 and KK 85. To identify the particular element, the KK 83 has an engraved ring around its perimiter, the KK 85 has a black dot on its edge, while the KK 84 is devoid of any marking.

Technical specifications KM 85 i

Acoustical operating principle	pressure gradient
Directional characteristic	cardioid
Frequency range	40 ... 20 000 Hz
Sensitivity	approx. 1 mV/μbar (across 1000 Ω) -38 dBm effective output level re 10 μbar
EIA sensitivity (G_m)	-137 dBm
Matching impedance	≧ 1000 Ω / 250 Ω
Source impedance	200 Ω / 50 Ω
Capsule capacity	approx. 37 pF
Self noise level (DIN 45 405)	approx. 25 dB (re 2×10^{-4} μbar)
Weighted noise level (DIN 45 405)	≦ 4 μV peak value
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz	≧ 200 μbar ≙ 120 dB
with sensitivity reduct.	≧ 650 μbar ≙ 130 dB
Operating voltage	48 VDC + 6 V - 8 V
Current consumption	0.4 mA
Battery life	≧ 200 hours
Weight	80 g (3 ozs.)
Dimensions	7/8" dia × 4 3/8" long





Microphone cables for the KM 85 i

IC 3 10 m (33 ft.) extension cable connects the microphone with its power supply. The SG 21 clamp is used as the microphone stand mount.

Power supplies for the KM 85 i

N 451 i AC power supply to power one microphone

N 452 i AC power supply to power one or two microphones

N 45 k
NK 48
GW 2448 These central supplies may power up to 40 microphones.

SW 45i Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45 i Battery supply to power one microphone for more than 200 hours

Tuchel connector version

The KM 85 i is also available equipped with Tuchel connector at its base. It then carries the designation KM 85.

Microphon cables for KM 85

KT 1 10 m (33 ft) extension cable for connecting the microphone with its power supply.

KT 2 10 m (33 ft) cable with stand mount swivel for connecting the microphone with its power supply.

Power supplies for KM 85

N 451 AC power supply to power one microphone

N 452 AC power supply to power one or two microphones

N 45 k
NK 48
GW 2448 These central supplies may power up to 40 microphones.

SW 45 Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45 Battery supply to power one microphone for more than 200 hours.

N 451 i

TRANSISTORIZED CONDENSER MICROPHONE KM 86 i

The Condenser Microphone KM 86 i is equipped with a head assembly containing two separate cardioid elements which may be set by means of a switch located below the head, to the three directional characteristics: omni, cardioid and figure-8. In the „cardioid" position, the KM 86 i delivers the excellent characteristics of the KM 84 i unit, while its other directional patterns produce unfalsified low frequency response even for distant pickups, something of which previous dual-element microphones have not been capable. The axis of maximum sensitivity is at the side, at right angle to the microphone's amplifier body.

In order to prevent overload of the microphone preamplifier resulting from the pick-up of high level solo instruments, a 10 dB switch on the body of the microphone reduces sensitivity at the gate of the FET. This allows even the highest sound levels encountered in practice to be reproduced without distortion.

KM 86 i

KM 86, EA 2124, KT 1

Type „i" for International applications
Cannon connector equipped

Technical specifications KM 86 i

Acoustical operating principle	pressure gradient
Directional characteristics	omni, cardioid, figure-8
Frequency range	40 ... 20 000 Hz
Sensitivity	approx. 0.7 mV/μbar (across 1000 Ω) -41 dBm effective output level re 10 μbar
EIA sensitivity (G_m)	-140 dBm
Matching impedance	≧ 1000 Ω/250 Ω
Source impedance	200 Ω/50 Ω
Capsule capacity	2 × approx. 34 pF
Self noise level (DIN 45 405)	approx. 28 dB (re 2×10^{-4} μbar)
Weighted noise level (DIN 45 405)	≦ 4 μV peak value
SPL limit for 0.5% THD at 40 Hz, 1 kHz and 5 kHz with sensitivity reduct.	≧ 200 μbar ≙ 120 dB ≧ 650 μbar ≙ 130 dB
Operating voltage	48 VDC $^{+6\,V}_{-8\,V}$
Current consumption	0.4 mA
Battery life	≧ 200 hours
Weight	200 g (7 1/8 ozs.)
Dimensions	7/8" and 1¾" dia × 7¼" long

Microphone cables for the KM 86 i

IC 3 10 m (33 ft.) extension cable connects the microphone with its power supply. The SG 21 clamp is used as the microphone stand mount.

Power supplies for the KM 86 i

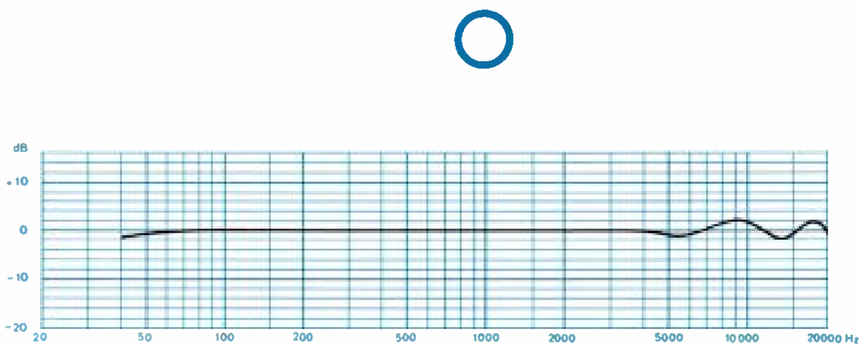
N 451 i AC power supply to power one microphone

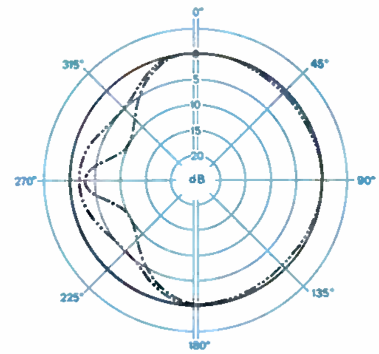
N 452 i AC power supply to power one or two microphones

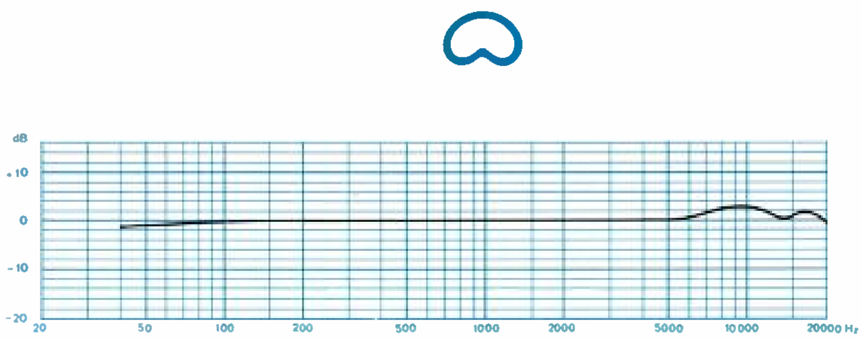
N 45 k, NK 48, GW 2448 These central supplies may power up to 40 microphones.

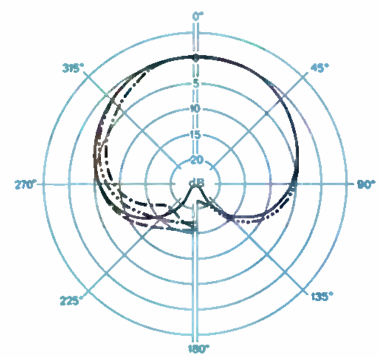
SW 45 i Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

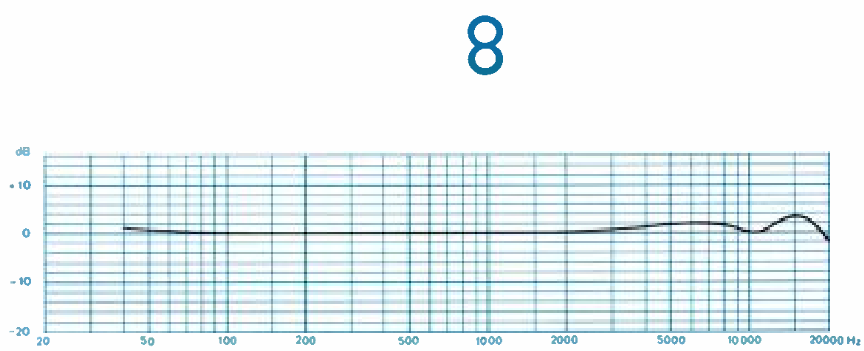
BS 45 i Battery supply to power one microphone for more than 200 hours

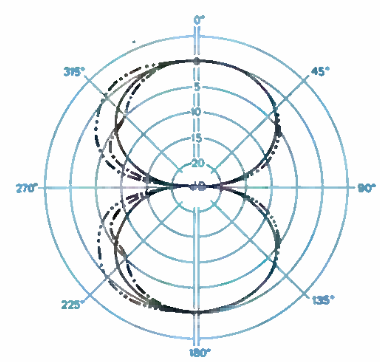












Curves not shown are identical to the 1 kHz curve

Tuchel connector version

The KM 86 i is also available equipped with Tuchel connector at its base. It then carries the designation KM 86.

Microphon cables for KM 86

KT 1 10 m (33 ft) extension cable for connecting the microphone with its power supply.

KT 2 10 m (33 ft) cable with stand mount swivel for connecting the microphone with its power supply.

Power supplies for KM 86

N 451 AC power supply to power one microphone

N 452 AC power supply to power one or two microphones

N 45 k
NK 48
GW 2448 These central supplies may power up to 40 microphones.

SW 45 Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45 Battery supply to power one microphone for more than 200 hours.

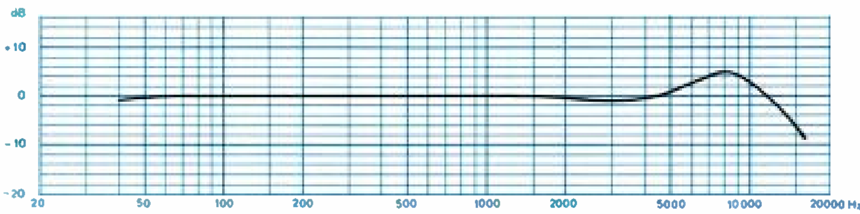
KM 88 i

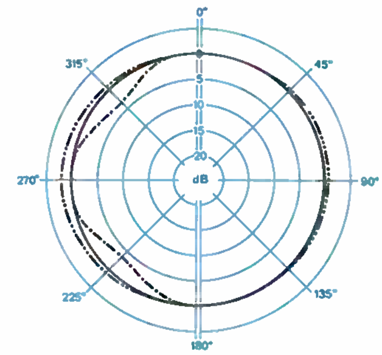
KM 88 Type "i" for international applications
Cannon connector equipped

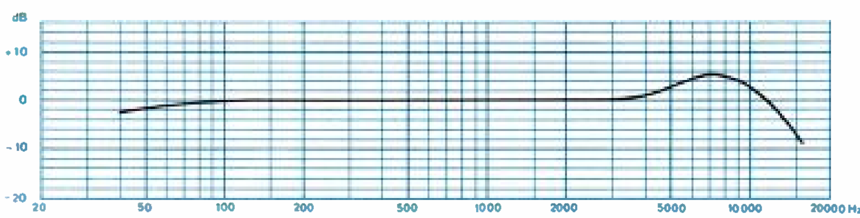
MINIATURE TRANSISTOR CONDENSER MICROPHONE KM 88 i

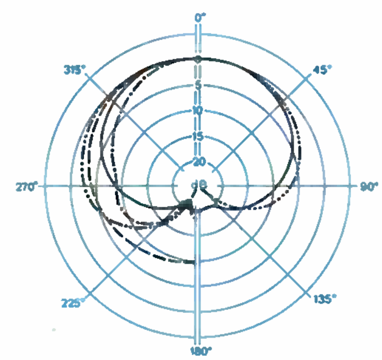
The Miniature Microphone KM 88 i is a successor to the well known KM 56 tube model. It is switchable to omni and cardioid patterns. Axis of maximum sensitivity is at the side at right angles to the microphone body. It is equipped with two nickel membrane capsules and therefore features the acoustical quality of its forerunner, the KM 56 model, which had won many friends in the recording, broadcasting, TV and film industries. It satisfies the oftimes voiced requirement for a vocal microphone in public address work which does not point at the soloist, but stands vertically, and which is less obtrusive than the KM 86 i model with its considerably larger head assembly. In its omni-directional position it is particularly well suited to round table discussions, while its cardioid characteristic suppresses unwanted sounds in film and TV studios, and often permits satisfactory recordings in acoustically unfavorable rooms and halls. A switch to reduce sensitivity at the amplifier input is also provided in this model.

Technical specifications KM 88 i	
Acoustical operating principle	pressure gradient
Directional characteristic	omni, cardioid
Frequency range	40 ... 16 000 Hz
Sensitivity	approx. 0.7 mV/µbar (across 1000 Ω) -41 dBm effective output level re 10 µbar
EIA sensitivity (G_m)	-140 dBm
Matching impedance	≧ 1000 Ω/250 Ω
Source impedance	200 Ω/50 Ω
Capsule capacity	2 × approx. 30 pF
Self noise level (DIN 45 405)	approx. 28 dB (re 2×10^{-4} µbar)
Weighted noise level (DIN 45 405)	≦ 4 µV peak value
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz with sensitivity reduct.	≧ 200 µbar ≙ 120 dB ≧ 650 µbar ≙ 130 dB
Operating voltage	48 VDC +6 V / -8 V
Current consumption	0.4 mA
Battery life	≧ 200 hours
Weight	100 g (3½ ozs)
Dimensions	7/8" dia. × 5 7/8" long









Curves not shown are identical to the 1 kHz curve

Microphone cables for the KM 88 i

IC 3 — 10 m (33 ft.) extension cable connects the microphone with its power supply. The SG 21 clamp is used as the microphone stand mount.

Power supplies for the KM 88i

N 451i — AC power supply to power one microphone

N 452i — AC power supply to power one or two microphones

N 45 k, NK 48, GW 2448 — These central supplies may power up to 40 microphones.

SW 45i — Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45i — Battery supply to power one microphone for more than 200 hours

Tuchel connector version

The KM 88i is also available equipped with Tuchel connector at its base. It then carries the designation KM 88.

Microphon cables for KM 88

KT 1 — 10 m (33 ft) extension cable for connecting the microphone with its power supply.

KT 2 — 10 m (33 ft) cable with stand mount swivel for connecting the microphone with its power supply.

Power supplies for KM 88

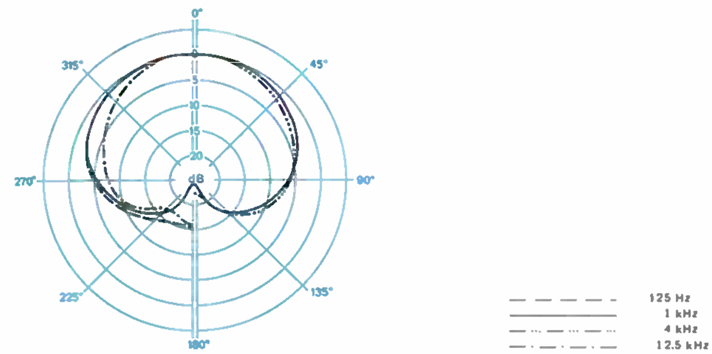
N 451 — AC power supply to power one microphone

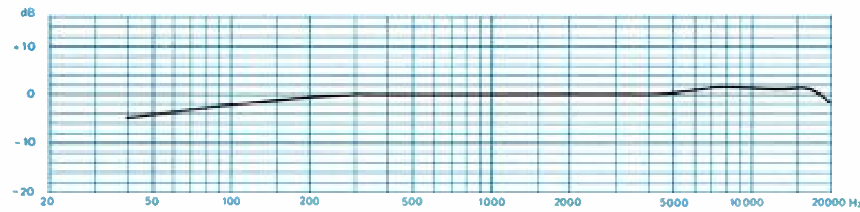
N 452 — AC power supply to power one or two microphones

N 45 k, NK 48, GW 2448 — These central supplies may power up to 40 microphones.

SW 45 — Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

BS 45 — Battery supply to power one microphone for more than 200 hours.






SRM 84 i

SRM 84 i

TRANSISTOR FLOOR STAND MICROPHONE SRM 84 i

The Floor Stand Microphone SRM 84 i is usable in all situations where the use of any of the other microphone models would obstruct the view of an audience or camera. This is especially true in film and TV studios. A further application for the SRM 84 i is as a vocal or speaker microphone on a stage or rostrum. The microphone is made up of the condenser element, a thin extension tube, the microphone amplifier, the guide tube and the microphone stand base. The condenser element is connected to the amplifier by the long tube. The amplifier is free to slide within the guide tube which is mounted atop the stand's base. A clamp nut allows the amplifier to be adjusted and fastened over a wide range of heights within the guide tube, so that the condenser element may be placed anywhere between 120 and 150 cm (48 - 60 inches) above the floor. The bend in the connecting tube directs the condenser element towards the sound source. Its directional characteristic is cardioid and the directivity is virtually independent of frequency. The frequency response curves within an angle of ± 135° from front are nearly parallel.

This microphone can also be supplied as an SRM 83 i or SRM 85 i. The frequency response and directional diagrams for these models may be found on pages 6 and 10.

Microphone cables for the SRM 84 i

IC 3 — 10 m (33 ft.) extension cable connects the microphone with its power supply.

Power supplies for the SRM 84 i

N 451 i — AC power supply to power one microphone

N 452 i — AC power supply to power one or two microphones

N 45 k, NK 48, GW 2448 — These central supplies may power up to 40 microphones.

SW 45 i — Powering branch-off for powering microphones from the N 45 k, NK 48 and GW 2448 supplies

Technical specifications SRM 84 i

Acoustical operating principle	pressure gradient
Directional characteristic	cardioid
Frequency range	40 ... 20 000 Hz
Sensitivity	approx. 0.8 mV/µbar (across 1000 Ω) -40 dBm effective output level re 2×10^{-4} µbar
EIA sensitivity (G_m)	-139 dBm
Matching impedance	≧ 1000 Ω/250 Ω
Source impedance	200 Ω/50 Ω
Capsule capacity	approx. 34 pF
Self noise level (DIN 45 405)	approx. 28 dB re 2×10^{-4} µbar
Weighted noise level (DIN 45 405)	≦ 4 µV peak value
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz	≧ 200 µbar ≙ 120 dB
Operating voltage	48 VDC $^{+6\,V}_{-8\,V}$
Current consumption	0.4 mA
Battery life	≧ 200 hours
Weight	7 ½ lbs
Dimensions	3/8" and 7/8" dia × 48" ... 60" long

SPECIAL VERSIONS

Microphones with dark color option

For applications where the avoidance of light reflection is of utmost importance, all Neumann condenser microphones, as well as their cables and stand mount swivels, are available with a matte dark color surface treatment. The color tone is dark umber grey similar to RAL 7032. The designation to be used in ordering is (mt) placed after the model number of the unit in question; i.e. U 87 i (mt).

Microphones with 7-pin universal connector

For purposes of standardization, many studios have all microphones, power supplies and cables equipped with the 7-pin universal connector T 3470. All of the transistor condenser microphones of the FET-80 series may be so equipped. Quotations are available on special application.

Microphones for unbalanced microphone inputs

To permit the connection of the transistor condenser microphones of the FET-80 series to unbalanced console inputs, they are also available in an unbalanced version. These microphones use 2-conductor cables for interconnection. The 48 VDC supply appears on one of these conductors, while the modulation uses the other and shield. The acoustical characteristics of the unbalanced microphones are identical to their balanced counterparts. To differentiate these types, the unbalanced versions carry an additional number „8" ahead of the model number.

The following microphone models and power supplies are available:

Microphones:	U 887 i
	KM 883 i
	KM 884 i
	KM 885 i
	KM 886 i
	KM 888 i
Power supplies:	N 851 i
	N 852 i
	BS 845 i

The central power supplies N 45 k, NK 48 and GW 2448 are identically applicable to both the balanced and unbalanced models.

NOTE: The Model U 87 i, when powered by internal batteries, may be used both with balanced and unbalanced preamplifier and console inputs.

Unbalanced microphones with Tuchel connectors

The microphones and power supplies of the unbalanced series are available also equipped with Tuchel connectors. All cables are identically applicable to both the balanced and unbalanced models.

Special versions intended for connection to amplifiers, whose inputs are intended for use with dynamic microphones

The output levels of condenser microphones are approximately 10 - 20 dB higher than the same sound pressure will produce from dynamic microphones (150 - 250 Ω). Many amplifier inputs are intended for use with low level dynamics and can easily be overloaded when fed from condenser microphones. Aside from that, microphone inputs in many foreign countries, are equipped with input transformers which will only work correctly when fed from a 150/250 Ω source (principally USA technology).

All Neumann condenser microphones of the FET-80 series may be readily altered, by opening two wire links to conform with this requirement. Restrapping the output transformer reduces the output level, while severing two wire links activates two 47 Ω resistors, one in each modulation lead. This raises the 50 Ω source impedance to the required min. 150 Ω.

Microphones which have been so connected at the factory bear the designation „p" after the model number; i.e. U 87 p (Tuchel connector) or U 87 ip (Cannon type connector), and in unbalanced versions KM 884 p or KM 884 ip.

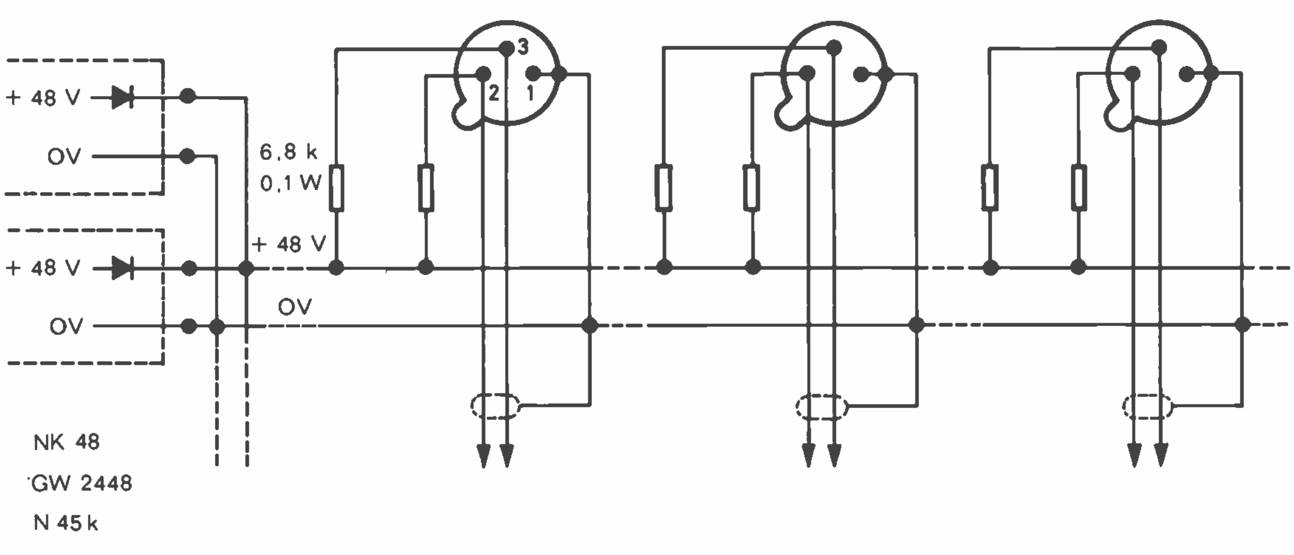
NOTE: All microphones sold in the USA and Canada are automatically connected as indicated above and therefore are of the „p" type.

CENTRAL POWERING

The central power supplies N 45k, NK 48, and GW 2448 may be used to power up to 40 microphones of the FET-80 series. They are intended primarily for installation within mixing consoles. Besides the power supply, two 6.8kΩ resistors selected to be within 0.5% are to be connected at some convenient place, to each microphone circuit. Connection is made according to the schematic shown below. When using other than the microphone connectors pictured here, the connection is to be undertaken in a like manner; i.e. each modulation lead is to be connected via one of the 6.8 kΩ resistors with + 48 VDC, while the minus (-) pole is to be connected to cable shield or chassis. When only 10 microphones are connected to a power supply, two modulation leads; i.e. one modulation pair, may be shorted to ground without affecting the performance of the rest of the connected microphones.

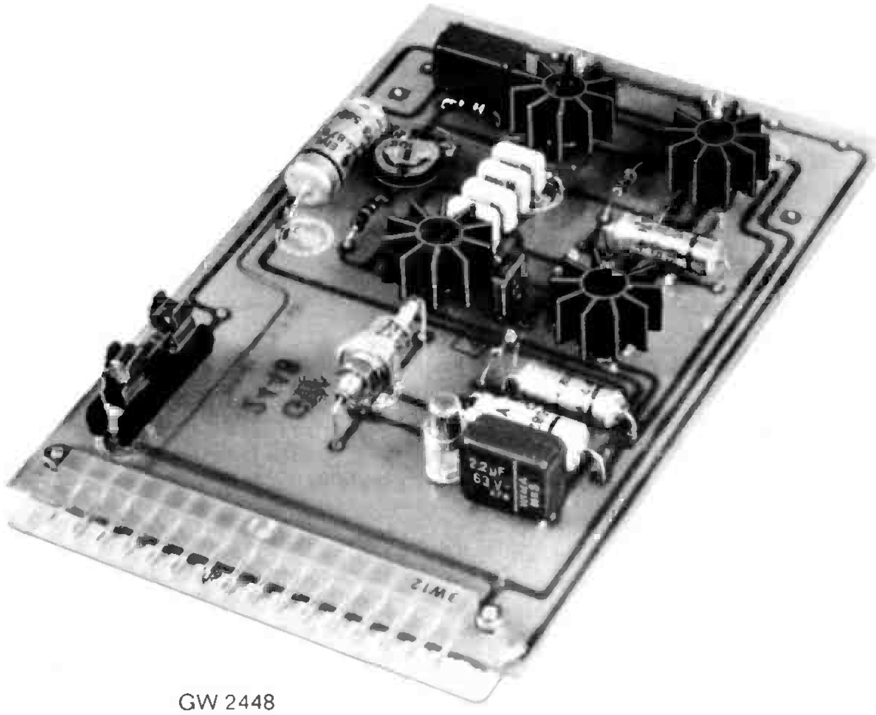
The outputs of two or more central supplies may be connected in parallel. Should one supply fail, the other supplies will take over. By virtue of the fact that the supplied DC voltage passes through an isolating silicon diode located within each supply, the shorting of one supply will not short out the remaining supplies.

N 45k



Technical Specifications N 45 k and NK 48

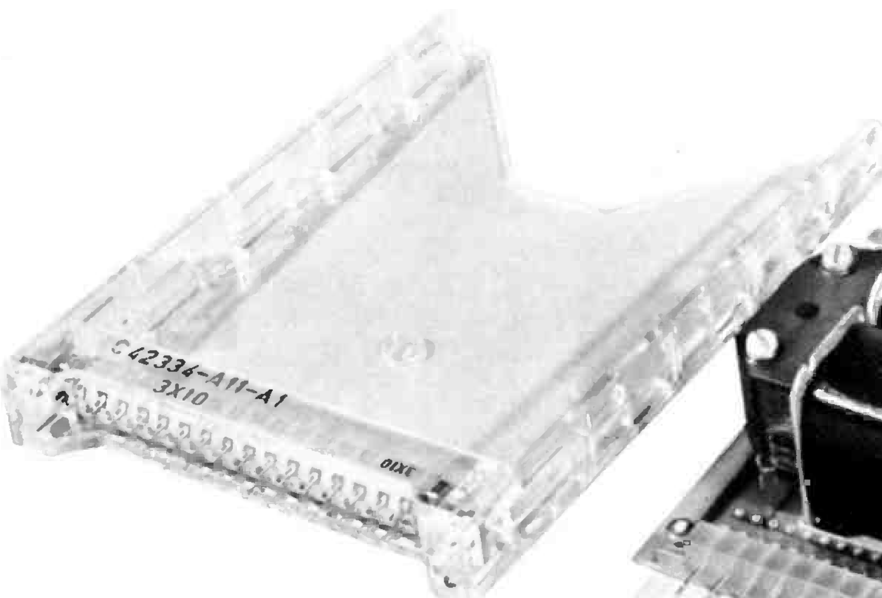
Power consumption	≦8 VA
Output voltage	+ 48 ... + 54 VDC
Available output power	max. 18 mA
Ripple	≦10 mV
Fuse for 220 Volts	0.1 A slo-blo
Fuse for 110 Volts	0.2 A slo-blo



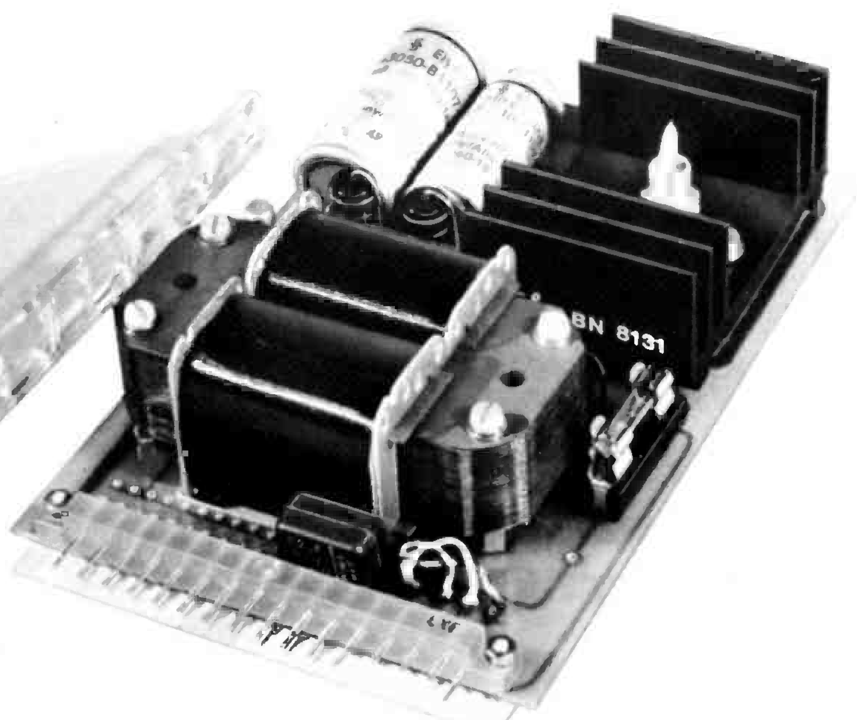
GW 2448

GW 2448

Input voltage	+ 24 VDC ± 3 V.
Current consumption (at 24 V)	approx. 50 mA
Output voltage	+ 48 ... + 53 VDC
Available output power	max. 16 mA
Ripple	≦1 mV
Fuse	0.2 A slo-blo



Plug-in frame C 42 334-A 11-A 1



NK 48

TRANSISTOR LAVALIER MICROPHONE KML

The Condenser Microphone KML was developed to improve pickup conditions of interviewers, quizmasters, masters of ceremony, and all other applications, where a speaker wearing a lavalier wishes to interview another speaker not equipped with a microphone. Aside from that, it is oftimes used when a regular pickup microphone as might be used in TV or film should be as unobtrusive as possible. When suspended from a „fish-pole" it provides ultimate quality at lowest possible weight, making the boom handler's job significantly easier.

The microphone is equipped with a cardioid condenser element whose principal axis of pickup does not point directly upward at the mouth of the speaker, but rather straight ahead. Since this microphone has a linear frequency response 6 dB down at an angle of 90° from front, there is no falsification of sound quality of the wearer. On the other hand, persons standing opposite the wearer, on axis, are likely to be at a distance which reduces their sound level also by about the same amount, thus providing proper balance. As a result, the interviewer no longer has to hold his microphone towards the person being interviewed.

The KML microphone has both a lanyard for hanging around the neck, and a tie clasp which allows it to be fastened to a shirt, tie or blouse. The KML is equipped with a connector prepared so as to allow the microphone to be plugged into wireless microphone transmitters designed for 18 Volt operation. It thus deviates from the usual connector standard of the other miniature condenser microphones. The approx. 1 mA current consumption of the microphone is so low, that it hardly decreases the 18 V. battery life at all.

Microphone cable KL

The microphone KML is normally equipped with a 1 m (3 ft.) cable. This may be extended by 10 m (33 ft.) using the extension cable KL.

Power supplies

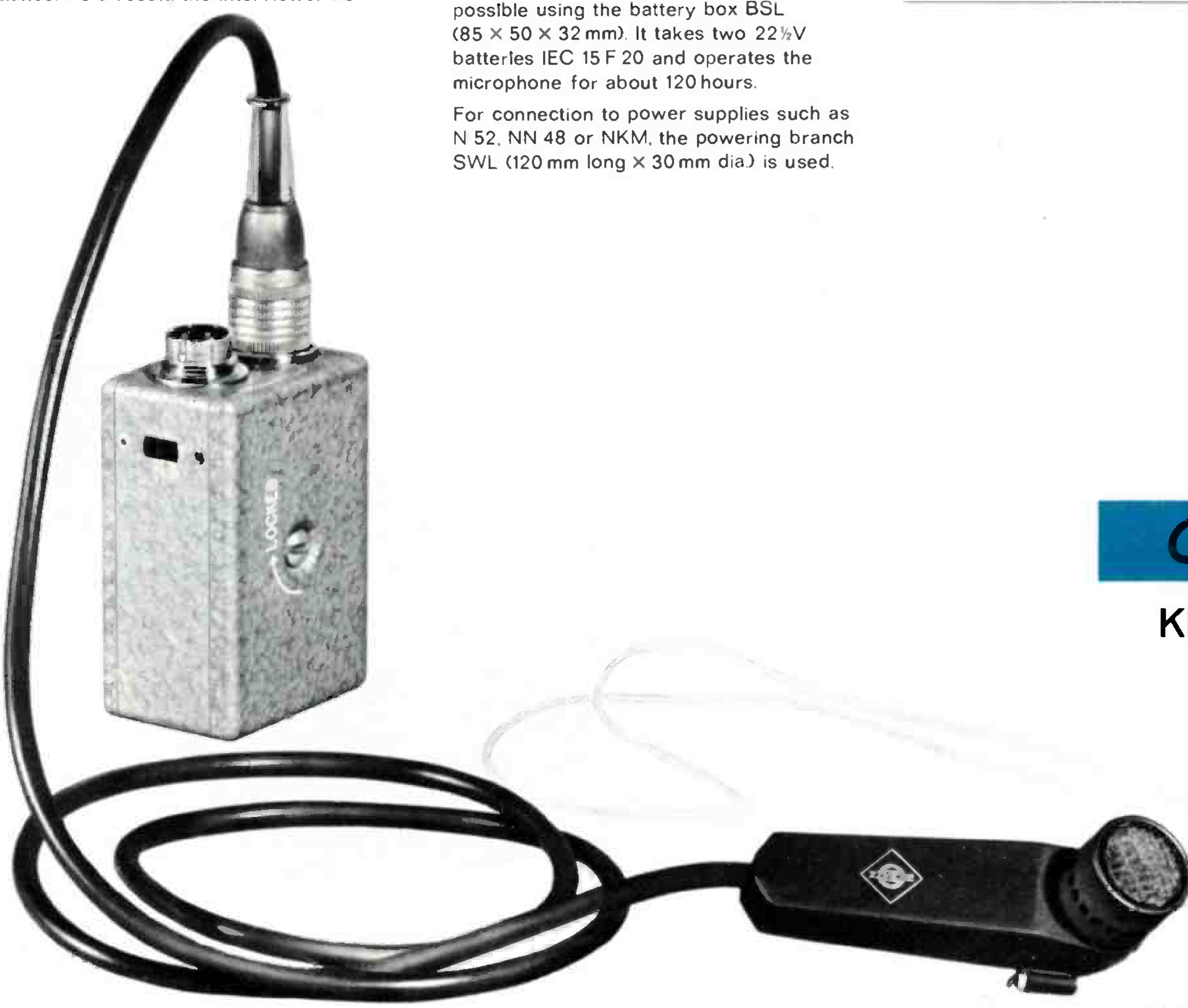
The microphone KML may be powered from battery supplies or from power supplies normally used with tube amplified microphones. Battery operation is possible using the battery box BSL (85 × 50 × 32 mm). It takes two 22½ V batteries IEC 15 F 20 and operates the microphone for about 120 hours.

For connection to power supplies such as N 52, NN 48 or NKM, the powering branch SWL (120 mm long × 30 mm dia.) is used.

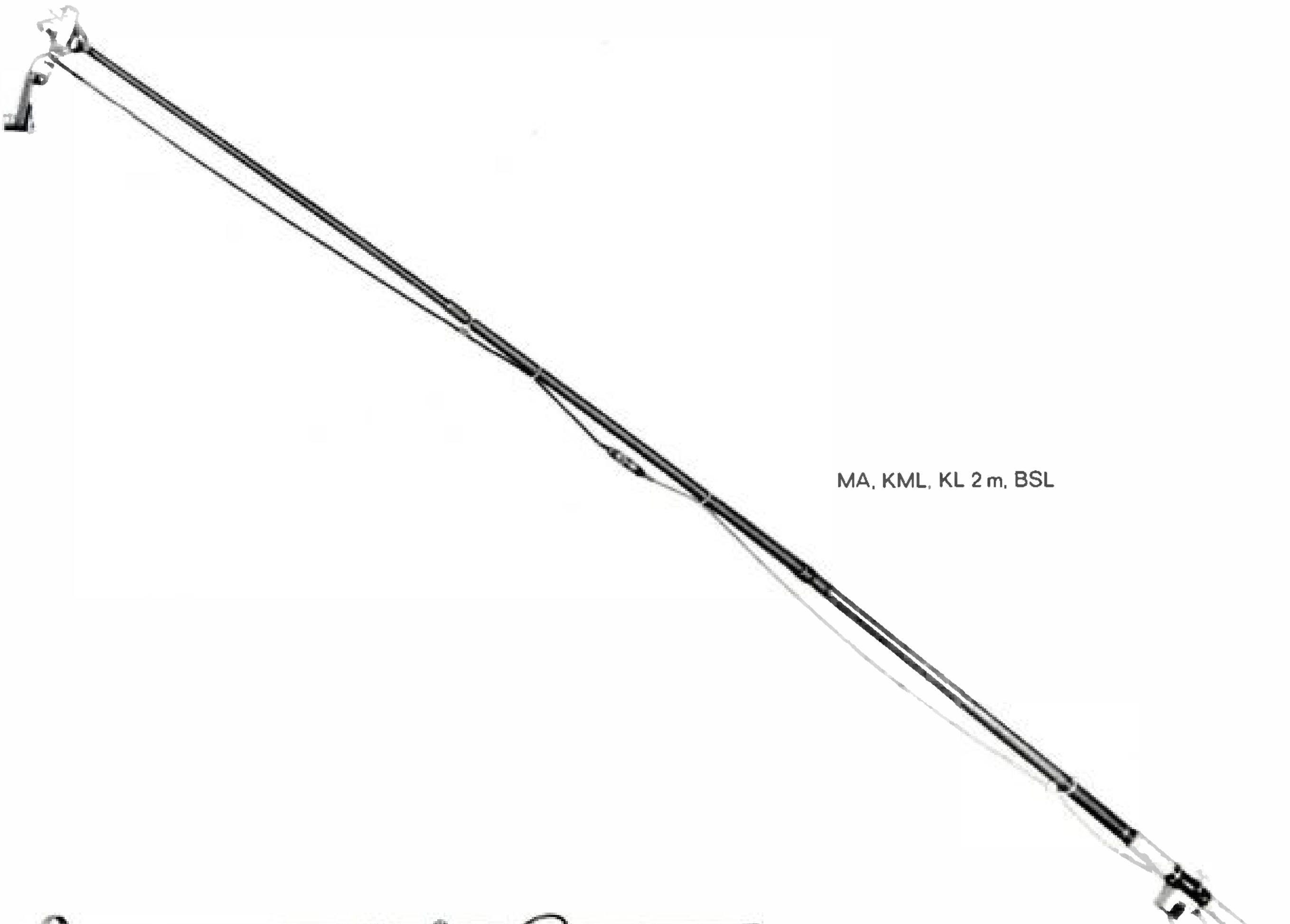
Technical specifications KML

Acoustical operating principle	pressure gradient
Directional characteristic	cardioid
Frequency range	40 ... 20 000 Hz
Sensitivity	approx. 0.9 mV/μbar -39 dBm effective output level re 10 μbar
EIA sensitivity (G_m)	-138 dBm
Matching impedance	≧ 250 Ω
Source impedance	50 Ω
Capsule capacity	approx. 37 pF
Self noise level (DIN 45 405)	approx. 26 dB re 2×10^{-4} μbar
Weighted noise level (DIN 45 405)	≦ 4 μV peak value
SPL limit for 0.5 % THD at 40 Hz, 1 kHz and 5 kHz	≧ 200 μbar ≙ 120 dB
Battery life	≧ 120 hours
Weight	1¾ ozs.
Dimensions	7/8" dia × 3 1/8" long

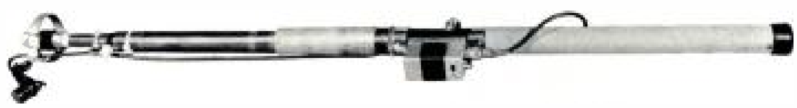
KML



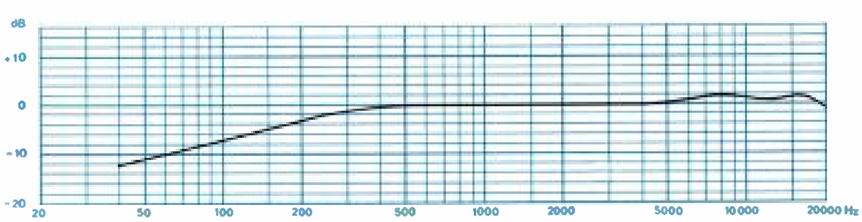
KML, BSL



MA, KML, KL 2 m, BSL



MA, telescoped



"FISH-POLE" MA

The Lavalier microphone KML is especially suited, by virtue of its weight (approx. 50 g [2 oz.] without cable), for mounting at the end of the "Fish-pole" MA. For this reason a new and extremely light weight pole has been developed. The "Fish-pole" consists of four telescoping tubes and only weighs about 600 g (1¼ lbs). The working length may be set for 3.5 or 2.5 meters (12 ft. or 9½ ft.). The pole may be collapsed to only 94 cm (40").

At the front end of the pole there is a threaded stud, mounting an elastic suspension. A rotatable fork is attached to this and the KML microphone fastens to this fork.

The battery box BSL or the powering branch SWL may be mounted between the two cork covered handles at the other end. The Lavalier microphone is connected to the power source using the 2 m (6 ft.) long special KL 2 m cable.

A complete „Fish-pole" microphone consists of:

Lavalier microphone	KML
2 m. special cable	KL 2 m
Battery box	BSL
or powering branch	SWL
Microphone „Fish-pole complete	MA

TRANSISTOR STEREO CONDENSER MICROPHONE SM 69 FET

The stereo condenser microphone SM 69 FET is a highest quality studio microphone for the compatible stereo recording technique. It was developed to serve the various forms of intensity stereo (M-S). Since the SM 69 FET is equipped with two completely separate microphones combined within a single unit, it may be applied to all uses where two microphones of like or differing characteristics are to be set up in the same spot. The microphone consists of a plug-in head containing two condenser elements in close proximity, and an amplifier part containing two completely separate microphone preamps. Each condenser element consists of two solidly joined perforated fixed electrodes, and two steamed gold covered polyester membranes. Each of the two condenser element halves has a cardioid characteristic. By applying specific polarizing voltages to the fixed center electrode and the two membranes, the three basic directional patterns cardioid, omni, and figure-8, and several intermediate patterns, are created. These may be remotely selected from the power supply for each condenser element in nine discrete steps. The upper capsule may be rotated against the lower through an arc of 270 degrees. The principal sensitivity axis is at right angles to the microphone body. It is indicated for the lower element by the Neumann insignia and for the upper by a mark on the head grille.

SM 69 FET

SM 69-FET,SC 6

Technical specifications SM 69 FET

Acoustical operating principle	pressure gradient
Directional characteristics	omni, cardioid, figure-8
Frequency range	40 ... 16 000 Hz
Sensitivity	approx. 2 mV/µbar across 1000 Ω -32 dBm effective output level re 10 µbar
EIA sensitivity (G_m)	-131 dBm
Matching impedance	≧ 1000 Ω / 250 Ω
Source impedance	200 Ω / 50 Ω
Capsule capacity	4 × approx. 53 pF
Self noise level (DIN 45 405)	≦ 22 dB re 2×10^{-4} µbar
Weighted noise level (DIN 45 405)	≦ 5 µV peak value
SPL limit for 0.5% THD at 40 Hz, 1 kHz and 5 kHz	≧ 200 µbar ≙ 120 dB
Weight	1 lb
Dimensions	$1^{3}/_{16}$" and $1^{7}/_{8}$" dia × 10" long

Microphone cables

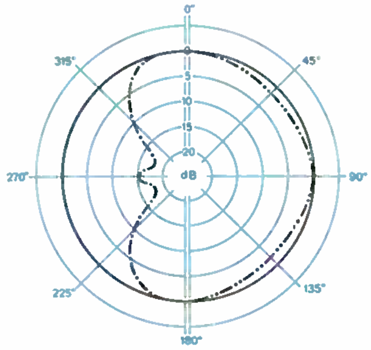
The stereo microphone is equipped with the 12-pole connector NT 3617. The following cables are available:

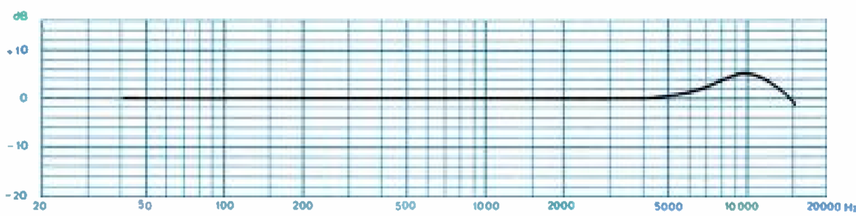
SC 1 A 10 m (33 ft.) cable without stand mount swivel.

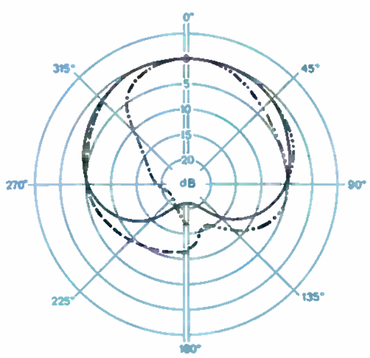
SC 6 A 10 m (33 ft.) cable with rotatable and tiltable stand mount swivel.

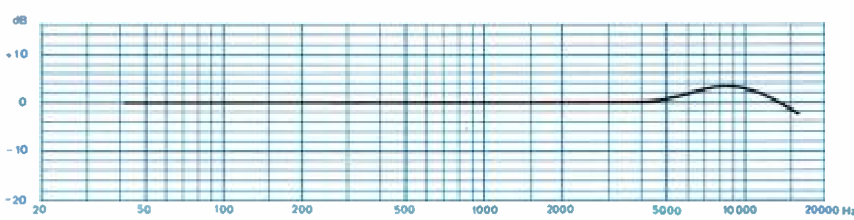
Power supply NS 69

For powering the stereo microphone, the NS 69 power supply is available. The directional characteristics are selected by two 9-position switches on it.









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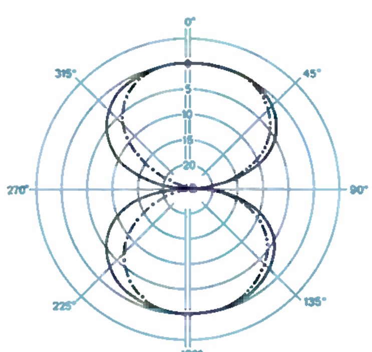
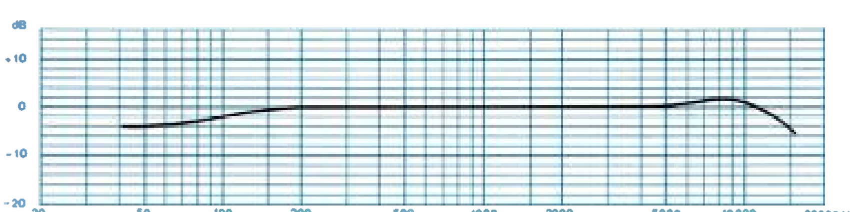


Curves not shown are identical to the 1 kHz curve



Z 26

Elastic mount Z 26

To prevent the transmission of shock noise interference between the microphone stand and the microphone, the Z 26 elastic mount may be interposed between stand and swivel. It is equipped with ½" thread.

SG 17, SG 21

Stand mount swivels SG 17 and SG 21

The stand mount swivel SG 17 permits the clamping of miniature Tuchel connectors and their mounting on standard microphone stand threads.

The slotted clamp of this swivel is tiltable; its base fits ³/₈", ½" and ⁵/₈"-27 threads. The SG 21 serves a similar function but fits the amplifier body of all miniature microphones.

KM 84, KT 2, Z 73, MF 1

Dual microphone mount Z 73

The dual microphone mount is screwed onto any microphone stand, boom or hanger, and permits the mounting of two microphones side by side on 5/8"-27 threads.

M 272

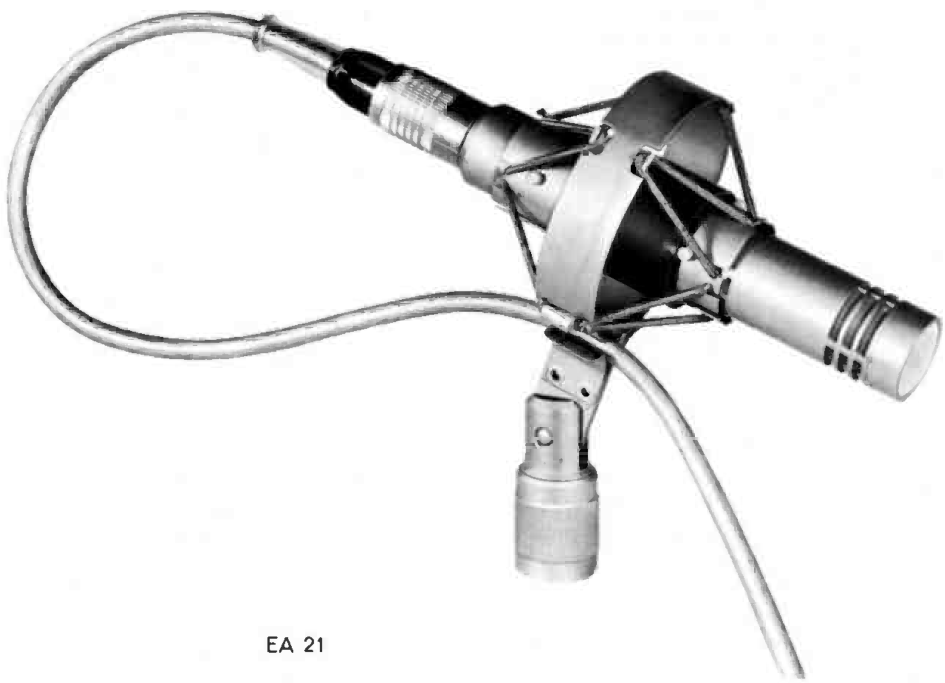
Table stand M 270

The table stand M 270 was designed specifically for the electrical and mechanical connection of a U 87 microphone. A connector mounted at its base permits connection of a KC 73 cable.

CF 3, CC 2, Z 24

Signaling lights CF 3 and CF 35

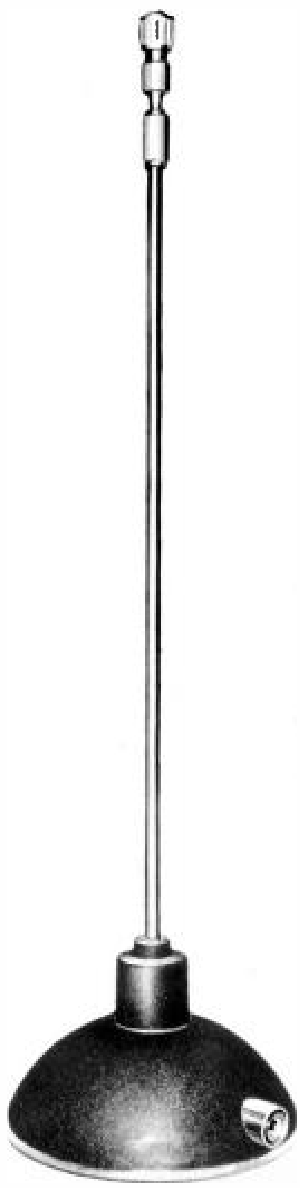
This accessory provides two-color light signals, often used as go-ahead signals in recording and broadcasting. The signaling light may be fastened to any microphone stand using the Z 24 clamp. The cable CC 2 is used to connect the signaling light. The CC 2a cable provides tandem connection of two such signaling lights (4 colors). The acknowledgement button on the CF 3 interrupts the current path, while on the CF 35 it can send back an additional signal. The CF 3 and CF 35 are available with Cannon-type connectors as CF 3i and CF 35I.



EA 21

Elastic microphone suspensions EA 21, EA 2124, EA 30 and Z 48

To suppress mechanical shock interference which travels along the microphone stand, stand mount swivel and suspension to the microphone itself, the use of an elastic suspension is recommended. For suspending the miniature microphones, the EA 21 suspensions (21 mm) are to be used or EA 2124 (21 and 24 mm). The same function is served by the EA 30 for the SM 69 FET, while the Z 48 suspension is used with the U 871 microphone. The elastic suspensions are equipped with swivels which will fit 3/8", ½" and 5/8"-27 threads.



M 272-10

MFS 3

M 32

Microphone floor stands M 272 and M 1272

The microphone floor stand M 272 is used with the U 87 microphone. It may be extended from 120 to 200 cm (50" to 80"). The electrical connection to the microphone is by means of an elastic cable within the microphone stand tube, thus presenting an elegant microphone mounting without unsightly exposed cable. The upright pipe is mounted in a heavy base which rests on a rubber shock mounting ring.

A slightly modified version, the microphone floor stand M 1272 is intended for the SM 69 FET microphone.

Microphone floor stand MFS 3

The microphone floor stand MFS 3 mounts all miniature microphones (with Tuchel connector). The model MFS 3i serves the same microphones equipped with Cannon-type connectors. The stand has a goose-neck with the 3-pole connector at its head. The upright tube is rubber mounted in the cast iron base. The stand may be extended between 1 m and 1.8 m (3 ft. and 5 ft.). The cable is threaded through the upright tube and goose-neck, and is 10 m (33 ft.) long. The MFS 73i stand is supplied for the U 87i microphone. The cable is 10 m (33 ft.) long.

Floor stand M 31

This tripod floor stand is mechanically similar to the MFS 3, but it has a 3/8" and ½" threaded stud which permits mounting a cable equipped with swivel.

Folding stand M 32

The M 32 stand is collapsible and may be extended between 1.15 m and 1.8 m (4 and 5 ft.). The upright tube has a ½" threaded stud for fastening microphone cables and swivels.

Floor stand with boom attachement M 210

This stand can mount a boom attachment with 75 cm (30") reach. The boom attachment has a threaded stud for fastening microphone cables and swivels.

Studio boom M 184

Movable on heavy duty casters. Height adjustable from 1.8 to 2.5 m (5 ft. to 9 ft.). Boom reach 1.2 to 2.9 m (4 ft. to 10 ft.). Height when set at an angle: approx. 4.5 m (14 ft.). Weight: 60 kg (132 lbs).

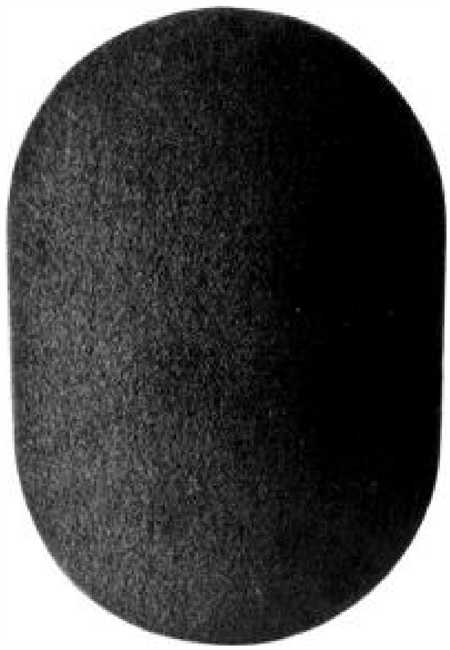
Floor stand M 35 with boom attachment G 35

The floor stand M 35 may be extended to a height of 5 m (17 ft.). It may furthermore mount a G 35 boom attachment with a reach of another 2.5 m (8 ft.). A swivelable threaded stud (½" plus adapters for other sizes) allows the mounting of microphone cables, swivels and suspensions.

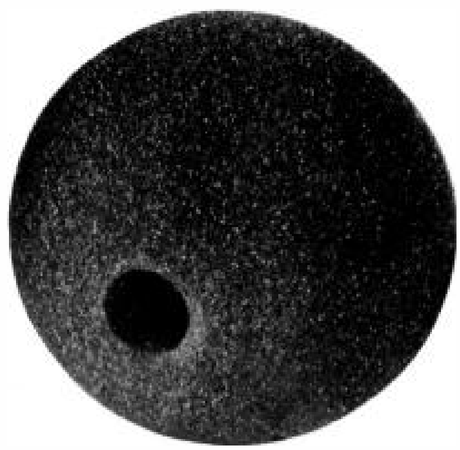
WIND AND CLOSE TALKING GUARDS

To avoid distortion resulting from close talking or the influence of wind there are wind and close talking guards made of open pored poly-urethane foam. These guards do not create undesired resonances and virtually influence the frequency response of the microphone only insignificantly (WNS 21 at 10 kHz approx.-1 dB). The following wind and close talking guards are available:

WNS 21 and WS 21	for KM 83 i, KM 84 i, KM 85 i and KM 88 i
WNSL	for KML
WS 86	for KM 86 i
WS 67	for U 87 i
WS 69	for SM 69-FET
WS 9	for SRM 84i



WS 67



WS 21

WNS 21

Table stand M 181

The table stand M 181 is a heavier version of the MF 1 and permits the setting up of larger size microphones. It has a ½" and ⁵/₈"-27 thread for the attachment of microphone cables and swivels.

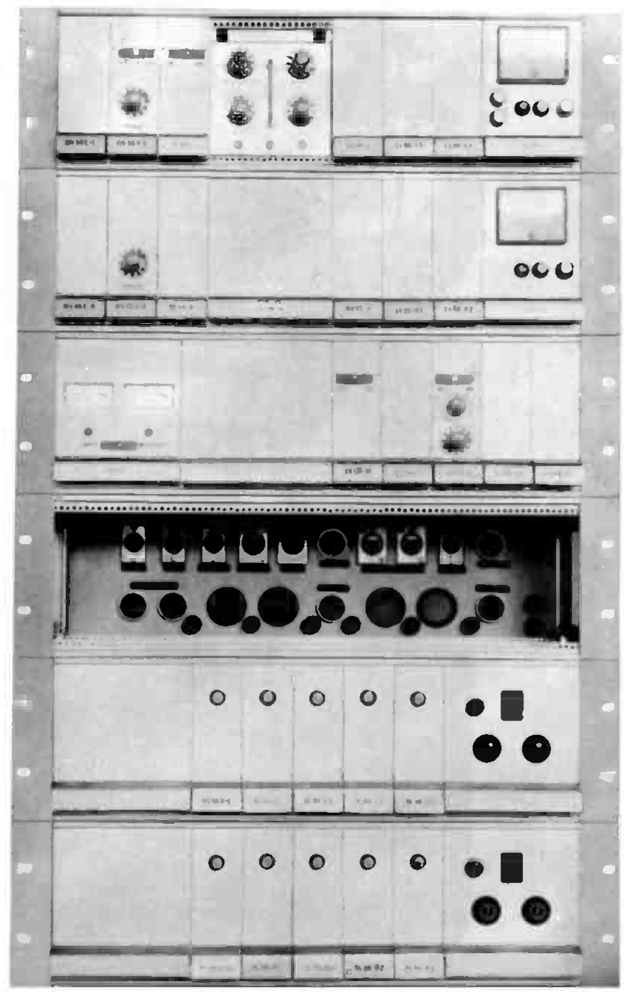
Table stand MF 1 and MTS 21

These microphone stands are useful for setting up miniature microphones on conference tables, speakers rostrums, pulpits etc. The MF 1 stand has a ½" and ⁵/₈"-27 thread for attaching microphone cables and swivels.

The three-legged stand MTS 21 has a clamp with 21 mm inside diameter. This clamp will fit all Neumann miniature microphones.

KM 84, KT 1, MTS 21

Z 68, UC 73, U 87



VG 66 S mono-stereo cutting system

Auditorium hanger Z 68

This accessory permits any microphone suspended from its own cable and swivel to be brought into any desired angular position.

DISK CUTTING LATHES

MONO CUTTERHEADS

STEREO CUTTERHEADS

AMPLIFIERS

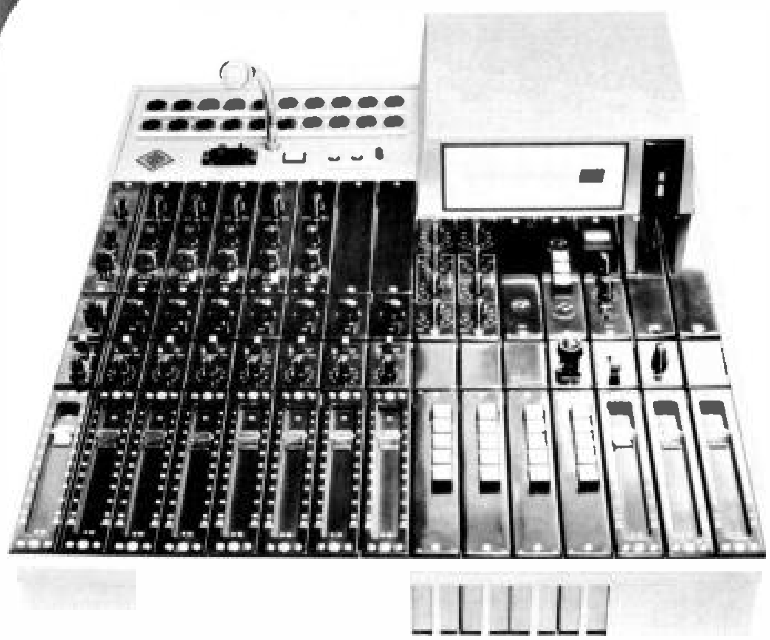
Our machines may be augmented up to completely automatic tape-to-disk transfer systems.

We will gladly send you brochures on our diverse products.

Neumann transistor studio amplifiers

A large number of control consoles in silicon planar technology has been designed and built by Neumann engineers over the past years.

Utilize this experience. Neumann engineers will be delighted to plan your consoles according to your individual wishes.



Neumann fet 80 Condenser Microphones

GOTHAM

Neumann fet 80 Condenser

KM 83/84/85

KM 86

KM 88

These three microphones all have identical electronics sections, permitting the capsules to be readily field interchanged. The KM 84's cardioid capsule provides a significant breakthrough in cardioid capsule design and is justifiably called the "Linear Admittance" capsule. It displays unprecedented linearity of response not only for on axis impinging sound, but also over an off-axis angle of some ±135°. This permits sound sources located far off axis to be picked up without coloration, and in large enclosures provides for a total sound field reproduction without the all too common bass-boost quality. The KM 83 omnidirectional pressure transducer is useful for extremely close voice pick-up, due to its insensitivity to popping and its lack of proximity effect bass rise. The KM 85 is a cardioid like the KM 84, but with a gradual low frequency roll-off, which reaches about −12 dB at 50 Hz, built into the capsule itself. It is most suitable for sound reinforcement applications. A −10 dB overload protection switch is provided on all of these microphones.

This microphone consists of two KM 84 capsules mounted back-to-back within its screen head assembly, however the subjective quality it achieves is surprisingly different from that of the KM 84, due to the size and shape of that screen enclosure. The use of two capsules provides directional pattern switchability: omni, cardioid, figure-8. A −10 dB overload protection switch is provided. The KM 86 is especially noteworthy for its linear low frequency response for all three directional patterns, even at a great distance from the sound source. The microphone, therefore, finds its greatest application in the medium and distant pick-up of instrument sections. It is not recommended for close up use. Its small size makes it ideal as an orchestra microphone in television.

Although it is a dual membrane, three-pattern switchable microphone (omni, cardioid, figure-8), the KM 88 is notably small in its outside dimensions. The capsule's dual membranes are made of nickel, the only such used on any fet 80® microphones, and give the KM 88 its characteristic crisp, brilliant sound. Its axis of maximum sensitivity is at right angles to the microphone body. Many studios find this the ideal microphone for string pick-up. The −10 dB overload switch is recessed and thereby protected against inadvertent operation.

Microphones for 48 V Phantom Powering

U 89

U 87

U 47 fet

A new microphone model from the ground up! Similar *only* in looks to, but about 20% smaller than the U 87, equipped with a completely new capsule (the right one pictured on our cover), and an electronics package containing eleven transistors vs the two in the U 87. And yet it most definitely is *not* in any sense a successor to, or replacement for the U 87, which continues its preeminent position in the industry. The capsule is unique in that all of its exposed surfaces are at ground potential, making it highly unlikely that the usual combination of dirt and humidity will cause capsule failure. Two new directional characteristics—hyper-cardioid and wide-angle cardioid—make the U 89 the most versatile studio unit available today. Its maximum SPL capability of 134 dB (140 dB with overload switch) and low-frequency roll off selectable to a boundary frequency of 80 Hz or 160 Hz, add even more flexibility.

Although similar in appearance to the U 87, it will likely be used more often in medium distance pick-up applications, concert halls and those places where previously the tube equipped M 49 model reigned supreme.

The model U 87 is the best known and most widely used of the fet 80® series. Its dual membrane capsule (the left one pictured on the cover of this brochure) uses evaporated gold on polyester film which has proven to be the most heat and aging resistant material. Three switches are provided beneath the capsule itself: for selecting the three directional patterns, frequency response and sensitivity. Its high frequency response is practically linear even in its cardioid and figure-8 positions even close-up. The response below 40 Hz is purposely rolled off to prevent low frequency blocking. This roll-off may be switched to 200 Hz to allow compensation for the bass rise common to all good directional microphones when used at close range. The U 87 is specifically designed for close miking studio applications. No microphone in Neumann's history has had as long and distinguished a career as the U 87. The venerable, tube equipped U 47 was manufactured for only 12 years, while the U 87's twentieth anniversary is already history! An enviable track record.

The U 47 fet continues the tradition of the world famous Model U 47, built from 1947-1960, which rightfully is credited with revolutionizing the world's recording and broadcasting industries. Its exterior strongly resembles its predecessor, but its technical properties represent the state-of-the-art today. It is protected against wind and pop interference; its capsule is elastically mounted to isolate it against mechanical shock disturbances; it features both a 10 dB overload protection switch at the input of its internal electronics and a 6 dB switchable output pad to permit matching to highly sensitive microphone input circuits. A low-frequency roll-off of 12 dB at 50 Hz is provided by a third switch. The result is a versatile unit which will take most microphone applications in stride. The dual membrane capsule is a pressure-gradient transducer with cardioid characteristic.

USM 69 (SM 69 fet/ QM 69)

KMS 84

KMR 82

Neumann manufactures two similarly shaped microphones for stereo recording, one for quadraphonic. The SM 69fet and the new USM 69 (pictured). The USM 69's electronics have been upgraded and it may simply be plugged directly into any two phantom powered outlets. These stereo microphones consist of two completely separate and independent microphone capsule systems mounted one above the other. The upper element may be rotated 270° with respect to the fixed lower one. This enables the user to apply the various intensity stereo recording techniques —such as M-S or X-Y—without the danger of arrival time (phase) differences between the systems. It is the only method which guarantees mono compatibility, while providing unprecedented three-dimensional localizability. Both microphone systems may be switched to nine different directional patterns, the SM 69fet from its NS 69 ac supply or CU 48 phantom powered controller; the USM 69 on the microphone itself.

The QM 69 is a quadraphonic unit, featuring four, 90° spaced, cardioids with four separate electronics in a single envelope.

A microphone for vocal and instrumental soloists must meet special performance criteria. It must be insensitive to explosive sounds ("popping"), must handle enormous sound pressure levels, and may not reproduce finger noise from handling in hand-held applications. The KMS 84 was specifically developed for this use. A multi-stage acoustical filter in front of its capsule combined with an extremely linear operational amplifier prevent overloading caused by the sub-audio parts of sibilants and speech explosives. All this is accomplished without in any way detracting from the traditional brightness typical of condenser microphones. The wire mesh grille is easily unscrewed and is available in red, yellow, green, blue, dark matte and satin finish to allow ready identification when used in sound reinforcement. GOTHAM also sells extremely supple microphone cable in these same, and some additional colors.

NEUMANN is certainly not the first to enter the shot-gun field—but it is the best! Great advancement in the design of such microphones have given the KMR 82 the most frequency independent directional pattern of any shot-gun microphone available today. The result is a low-frequency directional pattern that is virtually as narrow as the high—something never before achieved. Add to that the low, 12 dB equivalent loudness level, its relatively short 15½" (395 mm) length, its light 250 g weight, its low 0.7 mA power consumption, and its convenient accessories, and you have a major break-through in ultra-directional microphone design. A 120 Hz low-end roll-off and a high frequency reducing switch for close-up work are provided. A most convenient and unique accessory is the battery powered handle which obviates any need for outside powering. The KMR 82 is normally supplied in dark matte color.

Neumann fet 80 Condenser Microphones for 48 V Phantom Powering

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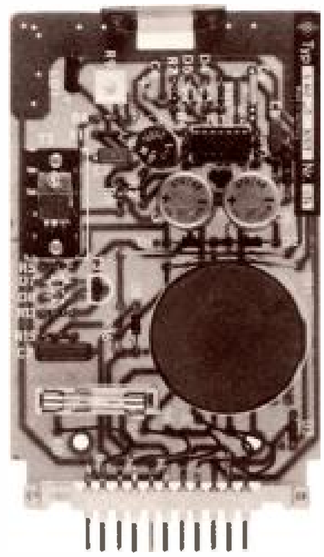
The N 80G DUAL POWER SUPPLY is a small, compact and economical way to power one or two fet 80® microphones. Simply plug it into an ac outlet, insert one or both of the Switchcraft XLR adapters into microphone inputs, the cable from the microphones into the adapters, and that's all that's necessary. Of course for fixed installations, it's preferable to have your microphone outlets centrally powered. In such cases the model N 80G5 may be wired into small consoles of up to five inputs.



The N 448 CENTRAL PHANTOM SUPPLY CARD may be hidden in your console, rack, or main frame and will supply up to one hundred fet 80® microphones with power. This supply may actually be connected to any number of outlets in any number of locations, as long as you don't plug in more than 100 microphones at a time. You may connect two supplies in parallel, and obtain back-up protection.

The KMA LAVALIER MICROPHONE is a high sensitivity, frequency compensated lapel unit which is powered either by the BS 18 battery supply (shown) or the SWA phantom powering adapter. Special mounts are available to mount the KMA to a violin or cello for extremely close pick-up. It is available only in unobtrusive dark matte color.

The MA "Fishpole" consists of five telescoping fiberglass sections, which extend to 12 ft 4 in and retract to a compact 4 ft length. The 16 oz weight is remarkable and helps fight operator fatigue. An elastic suspension at the head end accommodates any KM series microphone, including the KMR 82 "shot gun". Accessories include a battery supply holder and a swivel clamp to permit the MA to be used as a far reaching boom mounted atop any microphone stand.

The KM 83/4/5 Miniature Microphones may have their capsules located at a distance from the amplifier using the KV straight and curved extension tubes, available in lengths from 8" to 24". They are ideally suited to use on speakers' rostrums, in churches, for TV interviews and conference tables; any situation in which unobtrusiveness is a must.

WIND AND POP SCREENS are available for all Neumann studio microphones. Some are meant to be on the microphone at all times when such units are used for close talking or singing, while others are in use only outdoors or in environments where air currents, such as air conditioning system anemostats produce low frequency interference. The overall response and directional characteristics of the microphones is virtually unaffected by their use.

MISCELLANEOUS ACCESSORIES:

Neumann provides a large number of excellent accessories for its microphones. Elastic suspensions, wind and pop screens, auditorium cable hangers, intensity stereo mounts and matrixing transformers, twin mike mounts and many more. A complete catalog of these accessories is available.

Specifications

Type	KM83	KM84 KM85	KM86	KM88	U 89	U 87	U 47 fet	USM 69	SM 69fet	KMS 84	KMR 82	KMA
Directional patterns								2x	remote contr.			
Acoustic operating principle	pressure transducer	p r e s s u r e g r a d i e n t t r a n s d u c e r										pressure transducer
Frequency range	40-20,000 Hz	40-20,000 Hz	40-20,000 Hz	40-16,000 Hz	40-18,000 Hz	40-16,000 Hz	40-16,000 Hz	40-16,000 Hz	40-16,000 Hz	40-16,000 Hz	40-20,000 Hz	40-16,000 Hz
Eff. output level ref 1 Pa[1]	-41 dBm	-38 dBm -39 dBm	38 dBm	-42 dBm	-40 dBm	-40 dBm	-39 dBm	-38 dBm	-32 dBm	-44 dBm	-31 dBm	5mV open circuit
Source impedance	150 ohms/bal.	150 ohms/bal.	150 ohms/bal.	150 ohms/bal.	150	150	150	150	150	150	150 ohms/bal.	800 ohms/unbal.
Equivalent loudness level due to inherent noise (IEC 179)	20 dB-A	17 dB-A 18 dB-A	19 dB-A	19 dB-A	17 dB-A	18 dB-A	18 dB-A	13 dB-A	13 dB-A	18 dB-A	12 dB-A	24 dB-A
S/N ratio (A weighted) re 1 Pa at 1 kHz	74 dB	77 dB 76 dB-A	75 dB	75 dB	77 dB	76 dB	76 dB	81 dB-A	81 dB	78 dB	82 dB	70 dB
Max. SPL for less than 0.5% THD[2]	133 dB	133 dB	133 dB	134 dB	134 dB	132 dB	137 dB	133 dB	123 dB	138 dB	128 dB	113 dB
Total dynamic range of the microphone amplifier[3]	113 dB	116 dB 115 dB	114 dB	115 dB	117 dB	114 dB	119 dB	120 dB	110 dB	120 dB	116 dB	89 dB
Power supply +48 ±4 Vdc	0.4 mA	0.4 mA	0.4 mA	0.45 mA	0.8 mA	0.4 mA	0.5 mA	2 x 0.7 mA	0.8 mA	0.5 mA	0.7 mA	+16 to +24 Vdc 0.33 mA
Weight	80 g	80 g	210 g	130 g	400 g	500 g	710 g	510 mm	465 mm	210 g	250 g	30 g
Dimensions: dia length	21 mm 110 mm	21 mm 110 mm	21/47 mm 185 mm	21 mm 170 mm	46 mm 185 mm	56 mm 200 mm	63 mm 160/219 mm	30/48 mm 292.5 mm	30/48 mm 260 mm	21/40 mm 177 mm	21 mm 395 mm	33 × 18 × 15 mm 1 × w × h

[1] 1 Pa = 94 dB SPL = 10 dyne/cm²
[2] THD of the microphone amplifier when an input level equivalent to the capsule output at the specified SPL is applied.
[3] Referred to IEC 179 weighted equivalent loudness level.

CORRECTIONS:	U 89	U 47 fet	KMS 84
S/N ratio (A weighted) re 1 Pa at 1 kHz			76dB
Max. SPL for less than 0.5% THD[2]	140dB	147dB	
Total dynamic range of the microphone amplifier[3]	123dB	129dB	

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CONDENSER MICROPHONES

by

STEPHEN F. TEMMER, Pres.

Gotham Audio Corporation

(Reprinted from *Sound Merchandising / May, 1963*)

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THE ENGINEERING DEPARTMENT

OF

GOTHAM AUDIO CORPORATION

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Today's general trend of demand for ever better reproduction of sound has created great advancements in the art of condenser microphone construction, development of which goes back to the early 1920's. This article is intended to point out the basic qualities, advantages, disadvantages, and applications of condenser microphones in the high-fidelity conscious world of today.

When we speak of transducers, such as loudspeakers, playback cartridges, and microphones, the objective-measurement criteria that we rely on so heavily in the evaluation of electronic equipment assume only a limited role. It is generally accepted that the objective and subjective evaluation of such devices do not at all times agree; i.e., it is quite common to find two transducers of identical measured performance but of vastly differing subjective impression. With acoustical transducers such as speakers and microphones this is due, in large part, to the fact that performance is so highly dependent on the acoustical environment in which they are used, whereas they are usually measured in anechoic chambers or dead rooms. But, unfortunately, even the so-called "technical data" published about microphones is so often either "doctored up" or written in such meaningless terms that they fail to provide any significant help to someone attempting to choose a microphone for a particular application. The only answer is to try different makes and types; not just for a quick "whoof, one, two, three" check, but under actual operating conditions.

Neumann U-67 Condenser Microphone — shown disassembled into its amplifier, capsule head plug-in unit, housing and fastening ring. The M-269 Model version of this same unit is equipped with remote controlled directional characteristic.

Advantages of the Condenser Microphones

The size and weight of condenser microphones can be minimized by virtue of the fact that they require no magnet structure. Condenser microphones do not generate electricity (as is the case with dynamic or ribbon types), but only modulate an existing voltage. (We shall not discuss crystal or ceramic types for obvious reasons of restricted quality.) Dynamic or moving coil microphones have relatively heavy moving systems, a stiff membrane with a coil of wire attached. The ribbon microphone has a very lightweight aluminum ribbon as a moving system, but this requires a very large and heavy magnet to produce even the minimum output level needed and, due to its extremely low impedence, requires a built-in step-up transformer with ratios as high as 1:50—something that is difficult and expensive to produce. The condenser microphone, on the other hand, uses one of two possible moving systems: a plastic membrane as thin as ⅓ mil with a steamed-on layer of gold so thin you can see through it, or a membrane made of pure nickel which is even thinner. In either case there is a moving element much lighter than any other microphone type and, as a result, a frequency response range virtually free of resonant peaks and dips (See Figure 1). With microphones, just as with pick-up cartridges, the quest is for lighter and lighter moving systems.

To recap, condenser microphones have no magnet system and, therefore, no buky mass to disturb the sound field around the membrane. This makes for better directional characteristics of condenser microphones intended to be directional, and more uniform and nondirectional response for pure pressure transducers. In today's world of both recording and sound re-inforcement, it is the highly directional microphone which is most sought after, and in this respect the condenser microphone stands out the most. "Directionality" is sometimes referred to as the "front-to-back" ratio or, in other words, the relative sensitivity of a directional microphone's pick-up from the front and the back. This is generally given in db, but most manufacturers sidestep the most important issue here: How does the directionality vary with frequency? It is quite easy to make a microphone with a high front-to-back ratio for a narrow band of frequencies, but very difficult and expensive to make this rejection broadband, as is necessary (See Figure 2). Here again, the condenser microphone is tops for its size. The actual size of the *largest* condenser microphone capsule is about 1¼" in diameter by ⅞" thick. Capsules used in acoustical measurement work can be as small as ⅛" in diameter and have a linear response to as high as 100 Kc.

Probably one of the best-known advantages of condenser microphones is their ability to change their directional characteristic by means of a switch on the microphone, which can select omni-directional, figure-8, or cardioid pick-up patterns. It is interesting to note that no United States manufacturer of condenser microphones produces such a switchable di-

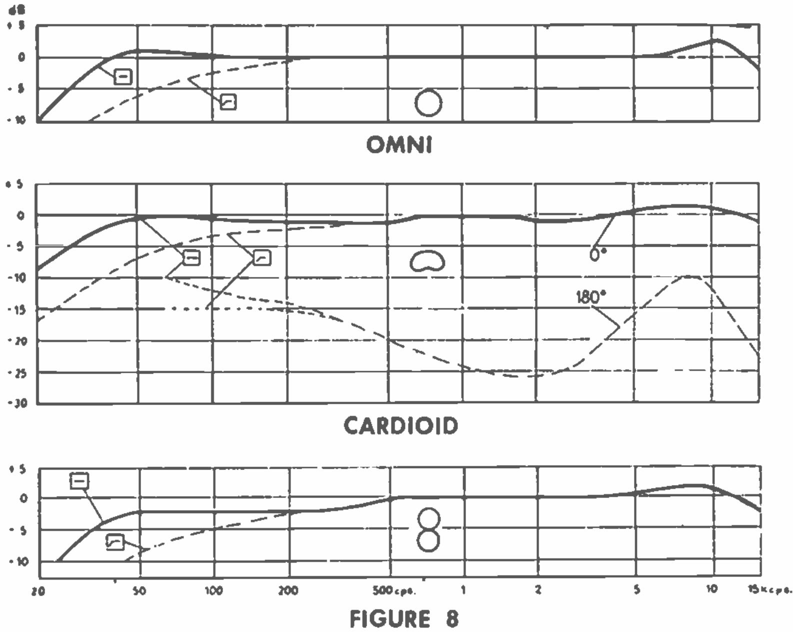


FIGURE 1. Response versus frequency for the Neumann U67 microphone. Dashed line is for low-frequency roll-off filter.

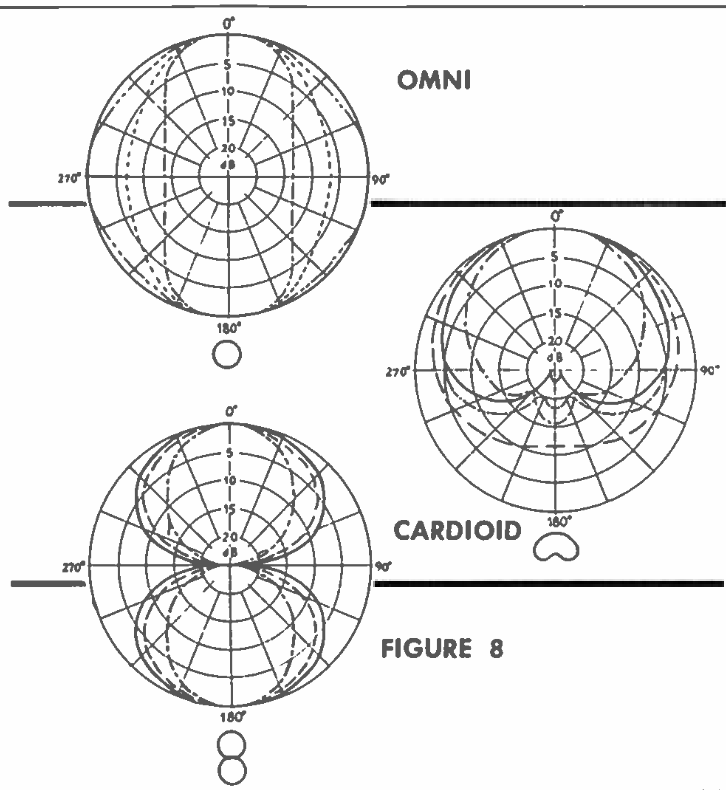


FIGURE 2. Polar patterns for several frequencies of the Neumann U67 microphone.

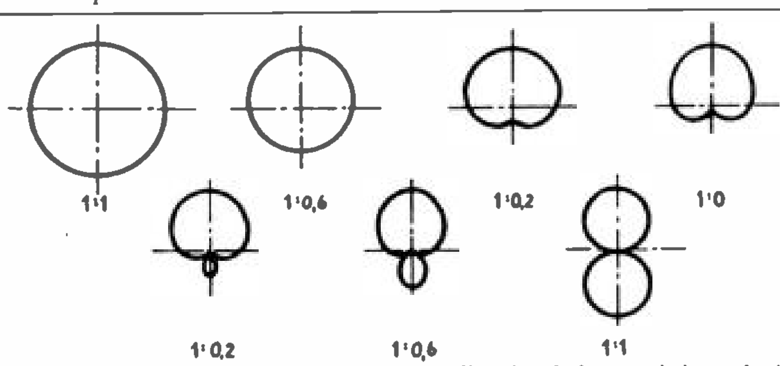


FIGURE 3. Seven of the infinite number of directional characteristics of both the Neumann M-49b and M-269 Condenser Microphones which are selectable by remote control from as much as several hundred feet away from the microphone.

rectional pattern unit. They are all manufactured overseas, as are the vast majority of condenser microphones. Two of these manufacturers offer an additional feature that is one of the most helpful in the entire microphone field—the remotely controlled directional characteristic. As the name implies, these microphones can change their directionality by operation of a control from as far as 500 feet away. Even during use this can be done without the introduction of any noise. This is especially helpful when recording in a large hall where the microphone is suspended from the ceiling out of reach. In such a situation, it may well turn out that the acoustics of the hall during rehearsal (when it is empty) may require a highly directional pattern, while at the performance with a full audience, the resulting "deadness" will demand more all-around reverb pick up (see Figure 3).

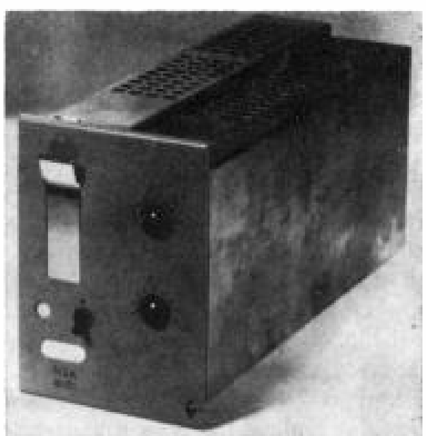
Drawbacks of the Condenser Microphone

Almost all of the problems that were encountered in the early days of condenser microphone manufacture have been solved in recent years with the remaining few about to disappear. The most bothersome problem, sensitivity to humidity, remains only in less expensive condenser microphones. Any old-time radio engineer will recall how they had to bake these microphones in an oven to stop them from crackling. The reason for this is that the air between the diaphragm and back plate ionized when exposed to high humidity and caused the polarizing voltage to arc over. One of the great advantages of the plastic membrane is that it is insulated against such arc-overs. In modern condenser microphones the polarizing voltage is greatly reduced (as low as 60 volts), which further reduces the danger of arc-over. Today's high-quality condenser microphones will tolerate a relative humidity of 95% with no adverse effect.

The most obvious drawback of condenser microphones is their need for a self-contained amplifier stage and its accompanying power supply. The impedance of a condenser capsule is well beyond 100 Megohms and therefore needs an impedance converting amplifier which will bring it down to a professional microphone impedance of 250 ohms or less. Contrary to common assumption, this is not because of the condenser element's low output—condensers are very high output transducers considerably more sensitive than dynamics or ribbons. The Neumann U-67 condenser microphone has a sensitivity of –53 dbm in the figure–8 and non-directional positions, and –48 dbm in the cardiod position for a sound pressure level of 10 dyne/cm^2. Included in these figures is a 12 dbm pad, as the unpadded sensitivity of 41 and 36 dbm, respectively, would cause overloading in preamps not geared for the microphone. These figures are for a 150/250 ohm output.

It is in the quality of the amplifier that much of the comparative merit of condenser microphones lies. Some are quite noisy whereas others will produce distortion or overload readily. Here again, you get what you pay for. It is obvious that amplifiers, can be built at many different price levels and so can their power supplies. High quality units will operate continuously for five years or more without any maintenance or even tube replacement.

Development of a transistorized condenser microphone amplifier is progressing rapidly and the day when tubes and power supplies on the floor will be a thing of the past, is just around the corner. Such transistorized



Neumann NSK Plug-in Power Supply unit for the Model SM-2 Stereo Microphone showing the two separate remote controls for changing the directional pattern of each of its two microphone capsules.

units will be highly miniaturized, self-contained, battery powered, and may even operate with balanced output without an output transformer. When this becomes a reality, condenser microphones of tremendous sensitivity and output level will be available in a miniature size never before dreamed of and with virtually unlimited service-free operation.

Comparison Between Condensers and Other Mikes

Just as there are dynamic microphones for $3.00 and for $500.00, there is also a wide range of prices in condenser microphones. They range in price from less than $200 to more than $800 for dual stereo units. Roughly speaking, though, the following can be said in comparing the average dynamic with the average condenser microphone: A wider frequency response range with greatly reduced peaks and dips within the range, and a more accurately maintained directional characteristic over the frequency range, again without major peaks or dips in the response of the rejected direction signal—qualities highly desirable in recording and sound work. These factors should be considered when choosing a condenser microphone: Amount of self-noise, overload distortion, output level, output impedance of the microphone unit itself. Many condenser microphones have their output transformer in the power supply unit and as a result you must run an unbalanced line, which, for long runs, may cause induced interference from stray fields. Many of these types of microphones use special expensive extension cables which can run the cost up considerably.

In servicing microphones you may experience quite a surprise by comparison to dynamic or ribbon units. Many manufacturers offer repair/exchange deals for servicing dynamic units at a fixed charge, which sometimes is as high as 25% of the initial cost of the microphone. Ribbons, in particular, actually need periodic replacement because they stretch with use, while such problems as loosening of the moving coil form, displacement in the magnet gap, and hardening of the plastic may slowly change the quality of a dynamic microphone over a period of time. Our twelve year experience with the repair of condenser microphones shows that from the standpoint of the condenser element itself, such units are what is called "go-no go" devices; i.e., they either function normally and properly, or they fail altogether, in which case a simple plug-in replacement of the capsule is required. Find out the cost of such a capsule replacement before choosing a condenser mike. Costs of such replacements vary widely, and so does the amount of time it will take to do the work and the time that you will be without your unit. As for the electronics, most people repair those problems themselves, or have repairs taken care of by the distributor at very nominal charges.

Applications of Condenser Microphones

Condenser microphones can be used in every application in which microphones are used at all. There is no situation in which it might be said that a condenser should definitely not be used. Many existing marginal P.A. installations can be converted into first-rate systems without replacing a single loudspeaker or amplifier, simply by making use of the greater intelligibility level obtained from condenser microphones. Many times better audibility can be had in auditoriums by *decreasing* the actual power fed to the loudspeakers and extending the frequency range, with less chance for positive feedback howls if the smooth response of a condenser microphone is utilized. Many examples can be given in which condenser microphones have made the critical difference in the performance of sound systems. The use of two condenser microphones on the main speaker's rostrum of the United Nations General Assembly, the more than 100 condenser microphones in the footlights of the Broadway stage, and the estimated 2,000 condenser microphones in the recording industry that have revolutionized the sound of records are all applications that speak for the upswing in their use. The Riverside Church, the Cathedral of St. John the Divine in New York, and literally hundreds of churches around this country have invested their limited available funds in expensive condenser microphones and have *saved* a great deal of money as a result of having to buy far fewer units than the dynamics they would have needed. Acoustical consultants such as Bolt Beranek & Newman specify condenser microphones in an increasing number of auditorium constructions, schools, concert halls, and speakers' rostrums. ◀

GOTHAM AUDIO CORPORATION

TRANSISTOR CONSOLE MODULES

TMV-60a TRVa TEVa TSVTa TSVRa TLVa TTMV TPG-5 TKV TVV TNGL TNG-1 TVa TV-34b

NEUMANN TRANSISTOR CONSOLE COMPONENTS DESCRIBED IN THIS CATALOGUE

(FRONT COVER CONSOLE: 18 inputs (any level from line to microphone); 4 Channel; 4 Echo Channel; All inputs with equalizers; Overdub cue channel; Mono mix output; Fully self-contained; Operation from 117 VAC or 12 VDC; Full patching facilities for limiters etc.)

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GENERAL:

In an effort to satisfy the demands born of modern studio console design, NEUMANN has developed a series of modular components which can be simply combined to produce mixing consoles for a vast variety of applications. The basic composition of consoles more often than not depends on the use to which they are to be put. The requirement may be for a fixed multi-channel studio unit, or one built into a remote truck, or carried for field or remote work, or it may be used to combine high level outputs in a re-record or film mixing room. For reasons of flexibility these units were so constructed as to allow the systems engineer, or the person responsible for the console's physical layout, the widest possible latitude in arranging their relative positions. Every unit has the same top plate dimensions: 40 x 190 mm (except for power supplies which are double width) and therefore they may be arranged laterally or vertically in any order which will optimize operational convenience for the purpose at hand. At any rate, it becomes unnecessary to fabricate large and complicated panels with complex mounting holes and engraving; panels which soon are obsolete because of necessary layout changes which must be made in time.

In spite of the wide variety of console applications, basically they all must perform the following functions:

1) Amplification of the microphone output level to a value which will permit use of attenuators and equalizers.
2) Combination of several microphone channels into a group or buss.
3) Subsequent amplification, gain control, and perhaps equalization of the group or buss formed through combination of several mikes.
4) Observation of levels at various points within the console through use of a volume indicator.

THE NEUMANN APPROACH:

The microphone channel consists of three basic units: the Transistor Microphone Amplifier TMV-60, the Transistor Equalizer Amplifier TEV, and the Transistor Attenuator Amplifier TRV. The gain of the TMV-60 is adjustable in 6 dB steps between 0 dB and 60 dB. An isolated echo feed may be branched off as can an isolated overdub cue feed.

The second requirement may be solved in a conventional way through a combining network consisting of decoupling resistors. The loss incurred in this network must be compensated by an ensuing Transistor Amplifier TV. Or use may be made of the Transistor Combining Amplifier TSVT. The TSVT allows the combination of up to six channels without loss by means of its six input isolation amplifiers.

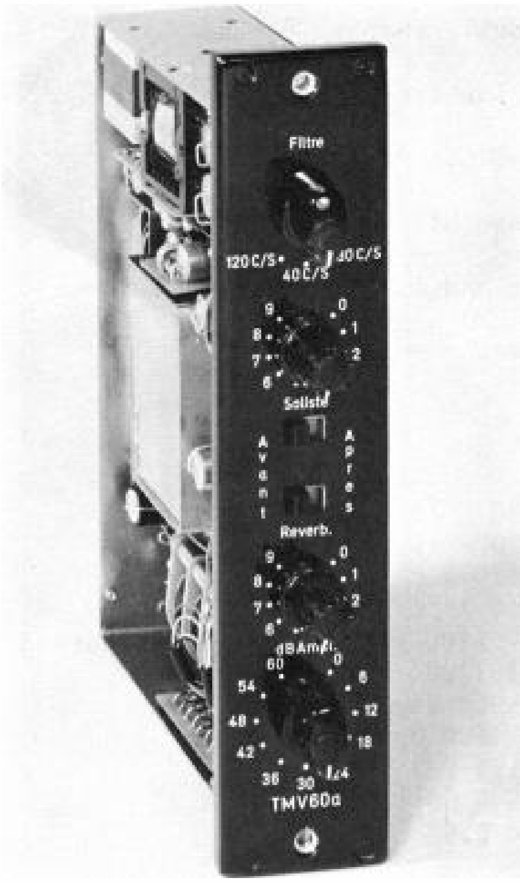
The group or program channel may start with a TV Amplifier to further amplify the combined signal, followed by a Transistor Equalizer Amplifier TEV and a Transistor Attenuator Amplifier TRV as a master gain control. The program channel ends in the Transistor Line Amplifier TLTV which brings the signal to standard output line level. A further TSVT Combining Amplifier may be used to combine program channels into any number of combined outputs.

Program level may be observed either by means of a standard VU meter or the Transistor Peak Indicating Amplifier TTMV in combination with its light beam indicator. A selector switch allows observation of existing levels at numerous circuit points within the console.

Simpler mixing consoles may be constructed by eliminating the TEV Equalizer, and when mixing higher levels, even the TMV-60 or TV-60 Amplifiers need not be used.

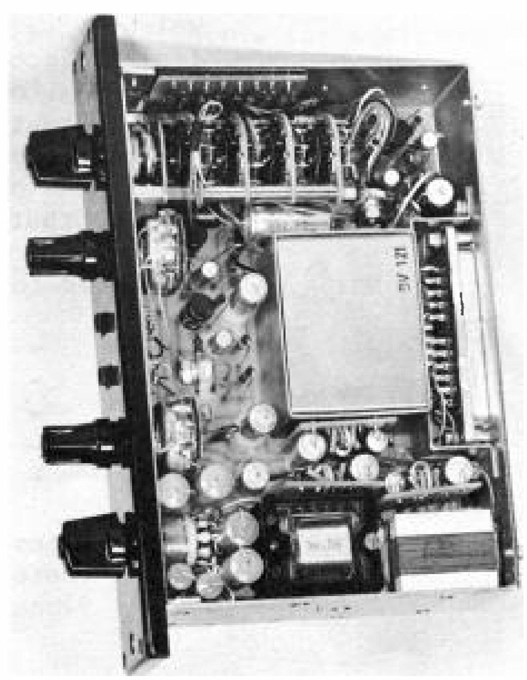
Standard modular unit enclosures with blank panels are available to permit installation of individual circuit components such as talk-back, monitor switching, studio signals, etc.

TRANSISTOR MICROPHONE AMPLIFIER TMV-60a



The TMV-60a evolved last in this chain of equipment and combines several functional as well as gain components within a single modular unit. It is intended as an input unit for consoles using microphones as a signal source.

This unit is fully transistorized and operates on 12 Volts DC. Its input is balanced and floating and the input impedance is $>$ 1200 Ohms. It may be fed from any source impedance up to 600 Ohms. The output is unbalanced and has an output source impedance of $<$ 60 Ohms. Two additional outputs, isolated from the main output by two collector-base amplifiers, serve echo feed and overdub cue. A switch selects the echo feed to tap off ahead of or after the attenuator which follows this unit. An echo and a cue fader are provided to permit separate control of both functions. The overdub cue may also be selectively taken either before or after the fader following. All three outputs are capable of +12 dB output levels.



Maximum amplification for the TMV-60a is 60 dB. A sensitivity switch located on the unit allows the gain to be reduced to zero in 6 dB steps. To accomplish this while maintaining at each step of this switch the optimal noise level referred to input, a complex switching circuit is used which simultaneously changes feedback, attenuation, and gain stages used. A built in low frequency cut off filter is selectable to 40, 80, and 120 cps 3 dB points for use in TV and film, while a low pass filter rolls the response off beyond 15,000 cps. The TMV-60 is housed in a standard modular unit and its controls have been arranged on the top panel in a functional manner.

TECHNICAL DATA:

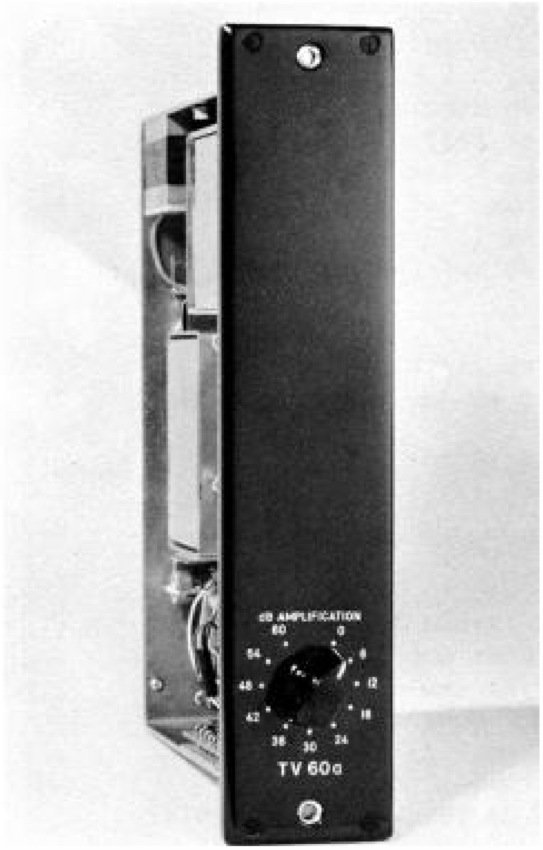
Frequency range:	40 - 15,000 cps.
Frequency response:	50 - 15,000 cps ± 0.5 dB at 40 cps -2 dB at 20 cps -12 dB
Input impedance:	> 1200 Ohms balanced & floating
Max. input level: (for gain = 0)	3.1 Volts (+12 dB)
Output source impedance:	< 60 Ohms unbalanced
Load impedance:	> 600 Ohms
Max. output level:	3.1 Volts (+12 dB)
Gain:	0 - 60 dB switchable
Total rms distortion:	< 0.5%
Noise level referred to input for gain = 60 dB:	< -122 dB (Input 200 Ohms)
Low frequency filter:	40, 80, 120 cps cut off.
Slope of filter curves:	> 12 dB per octave.
Operating power:	12 VDC @ 100 mA.
Operating temperature:	0 - 45° C.
Transistors used:	9 x AC-151r
Depth behind panel:	134 mm (5 1/2")
Weight:	4 lbs.
Isolated echo output-	
Max. output level:	3.1 Volts (+12 dB)
Output source impedance:	< 60 Ohms unbalanced
Isolated overdub cue output-	
Max. output level:	3.1 Volts (+12 dB)
Output source impedance:	< 60 Ohms unbalanced

Noise level referred to input in various positions of sensitivity selector:

54 dB gain	-	< -119.5 dB	24 dB gain	-	< -119.5 dB
48 dB gain	-	< -120.0 dB	18 dB gain	-	< -119.5 dB
42 dB gain	-	< -120.5 dB	12 dB gain	-	< -112.0 dB
36 dB gain	-	< -120.5 dB	6 dB gain	-	< -106.0 dB
30 dB gain	-	< -120.5 dB	0 dB gain	-	< -102.0 dB

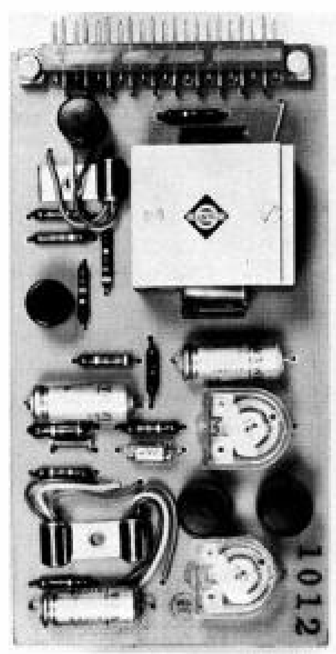
N.B. The last three positions (0, 6, and 12 dB) would be used for input levels so high that these noise levels are more than sufficient not to affect the over all signal to noise ratio in any way.

BASIC TRANSISTOR AMPLIFIERS TV, TV-60a



The TV-60a is a 60 dB voltage amplifier mounted in a standard modular unit for console mounting. With a 12 VDC 50 mA power consumption, this amplifier produces up to 60 dB gain with an output of +12 dB in the range from 40 - 15 KC with less than 0.5% THD. Its input and output are balanced and floating. The 60 dB maximum gain can be reduced in 6 dB steps to zero gain by means of a step switch which is panel mounted. To accomplish this NEUMANN selected a complex switching circuit which simultaneously changes feedback, changes attenuation pads, and removes stages of gain so that for every position of the sensitivity switch the optimum value of noise level referred to input may be obtained while maintaining the +12 dB output capability.

The TVa unit is an auxiliary, printed circuit board mounted, 30 dB voltage amplifier which may be mounted anywhere within the console for additionally required gain. This two stage amplifier contains two feedback loops of which one is fixed and is used to reduce non linear distortion, while the other is brought out to terminals and may be varied by changing external resistors. The value of the resistors used determines the gain, which may be set for any value between 18 and 30 dB.



TECHNICAL DATA:

TV-60a:

Frequency range:	40 - 15,000 cps.
Frequency response:	± 0.5 dB.
Input Impedance:	> 1200 Ohms balanced & floating.
Max. input level (gain = 0 dB):	3.1 Volts (+12 dB).
Output source impedance:	< 80 Ohms balanced & floating.
Load impedance:	> 600 Ohms.
Max. output level:	3.1 Volts (+12 dB).
Gain:	0 - 60 dB switchable.
Total rms distortion:	< 0.5%.
Noise level referred to input for gain = 60 dB:	< -122 dB (Input 200 Ohms).
Operating power:	12 VDC @ 40 mA.
Operating temperature:	0 - 45° C.
Transistors used:	6 x AC-151r.
Depth behind panel:	134 mm (5 1/2").
Weight:	2.5 lbs.

TVa:

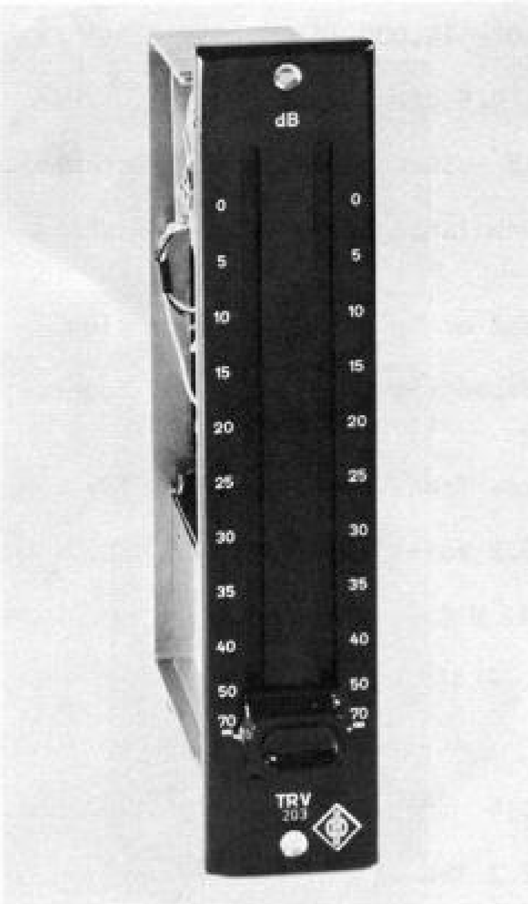
Frequency range & response:	same as TV-60a above.
Input impedance:	> 1200 Ohms unbalanced.
Max. input level:	0.39 Volts (-6 dB).
Output source impedance:	< 60 Ohms unbalanced.
Load impedance:	> 600 Ohms.
Max. output level:	3.1 Volts (+12 dB).
Gain:	18 - 30 dB feedback selectable.
Total rms distortion:	< 0.5%.
Noise level referred to input:	< -122 dB (input 200; output 600 Ohms).
Operating power:	12 VDC @ 25 mA.
Operating temperature:	0 - 45° C.
Transistors used:	3 x AC-151r.

TV-34b:

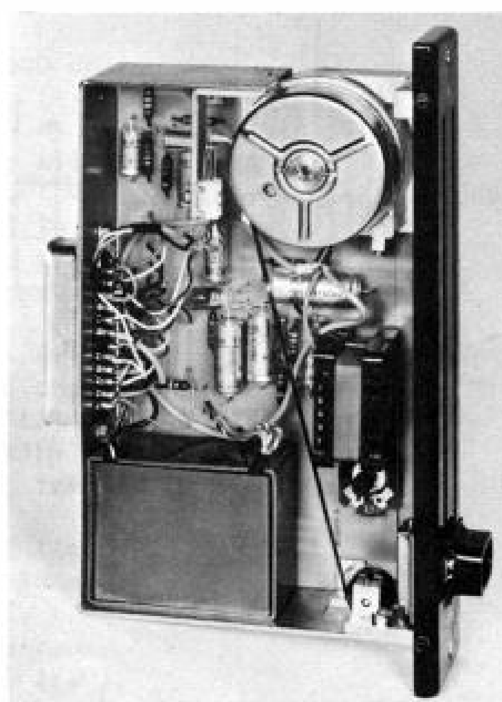
A 34 dB fixed gain amplifier version of the TVa unit is available with balanced input and output. See Matching and Level data sheets.

Input impedance:	> 3000 Ohms. balanced & floating.
Max. input level:	0.06 Volts (-22 dB).
Output source impedance:	< 40 Ohms balanced & floating.
Load impedance:	> 600 Ohms.
Max. output level:	3.9 Volts (+14 dBm).
Gain:	34 dB fixed.
Noise level referred to input:	< -122 dB (input 200; output 600 Ohms).
Operating power:	12 VDC @ 40 mA.
Transistors used:	6 x AC-151r.
Mounting:	Printed circuit card with receptacle.

TRANSISTOR ATTENUATOR AMPLIFIER TRVa

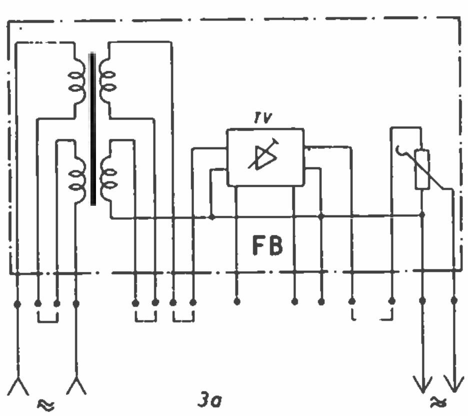


The TRV serves as an active attenuating unit. Within this module are a transformer, a variable attenuator, and a Transistor Amplifier TVa. The attenuator is a ladder with a 600 Ohm input and a 340 Ohm output impedance which can be raised to 600 Ohms by the addition of a 240 Ohm resistor in its output. Attenuation range extends to a maximum of 75 dB before infinity, while its isolation when closed is > 110 dB. It is a precision 57 step unit with 0.85 dB steps in the "working range" from 0 - 35 dB attenuation, after which it tapers to infinity. The attenuator actuates a transfer (C) contact in its infinity position which may be used for cue or signaling purposes. Furthermore the unit contains a Transistor Amplifier TVa and a well shielded transformer with a split and balanced primary. All of the inputs and outputs of these units as well as the signal contacts and feedback connection for the TV Amplifier and supply voltage leads are brought out separately to a 23-pole connector. As a result this unit may be used in any configuration since the circuit sequence within it is a matter of connector strapping.

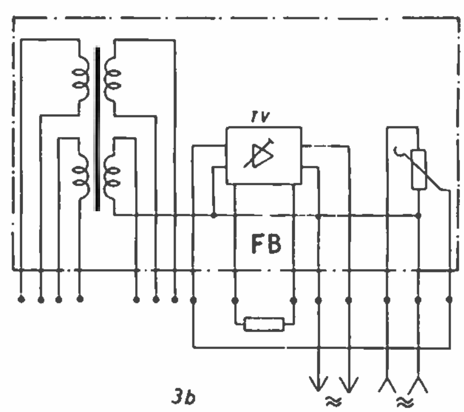


TECHNICAL DATA:

Frequency range:	40 - 15,000 cps.
Frequency response:	± 0.5 dB.
Transistor amplifier gain:	18 - 30 dB (feedback selectable).
Linear motion attenuator:	unbalanced ladder; 0 - 75 dB attenuation.
Transformer ratios:	1:2 or 1:1 selectable on plug.
Max. input and output levels:	dependent on sequence of connection of components.
Impedances (input, output, load):	see Transistor Amplifier TV.
Transformer max. input level:	6.2 Volts (+18 dB).
Operating power:	12 VDC @ 25 mA.
Operating temperature:	0 - 45° C.
Transistors used:	3 x AC-151r.
Depth behind panel:	134 mm (5 1/2").
Weight:	2.2 lbs.

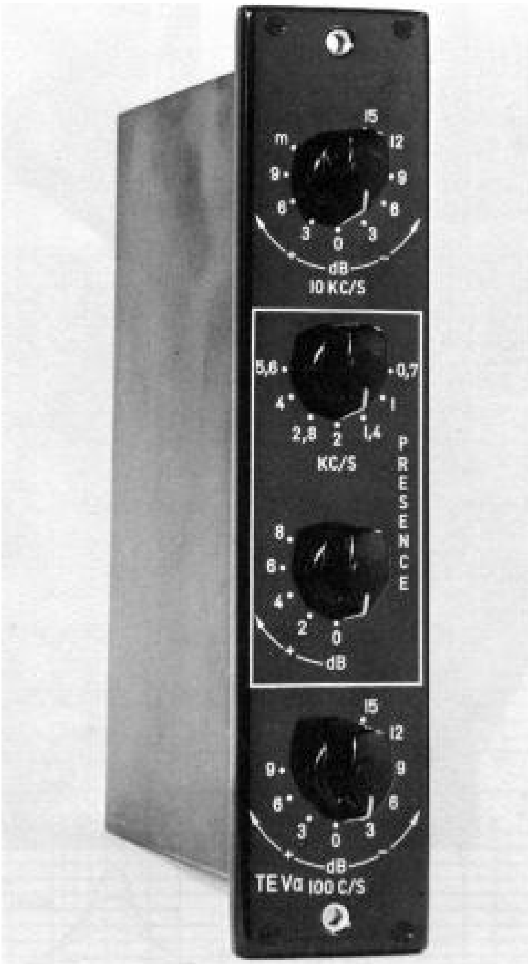


a) MODEL TRV FADER-AMPLIFIER UNIT CONNECTED AS MICROPHONE INPUT PREAMPLIFIER.



b) MODEL TRV FADER-AMPLIFIER UNIT CONNECTED AS MASTER GAIN WITHOUT TRANSFORMER.

TRANSISTOR EQUALIZER AMPLIFIER TEVa



This modular unit contains a high frequency, a low frequency, and a presence equalization section as well as a booster amplifier TVa. Maximum boost at 100 cps is 9 dB and at 10 KC it is 11 dB.

The presence equalization section consists of parallel resonant sections whose effectiveness is varied by shunting them with damping resistors of various values.

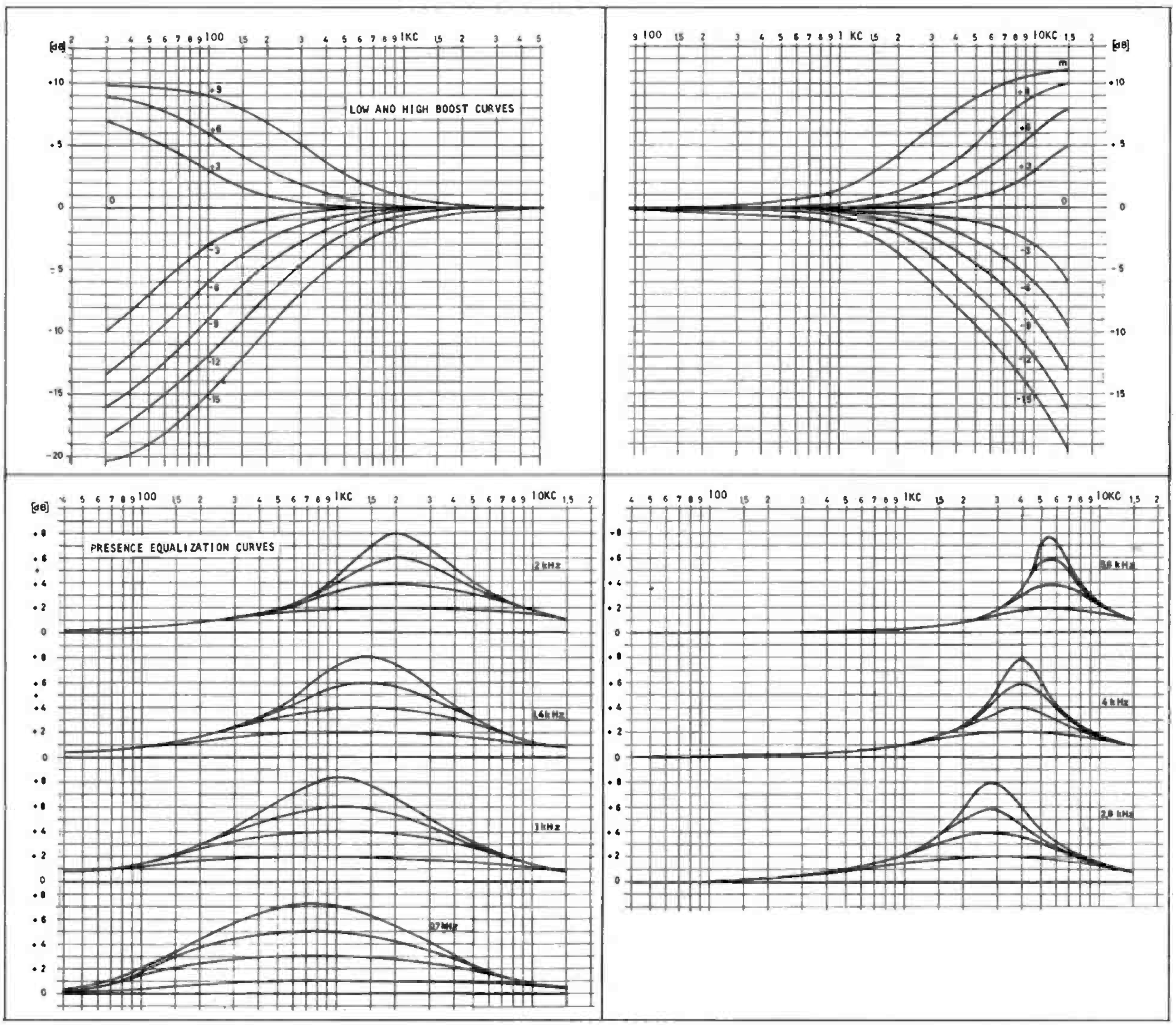
A collector-base amplifier section using a single transistor assures for the equalizer section the proper high impedance termination it requires.

For a gain setting of 30 dB of the TVa Amplifier, the entire TEVa unit produces an overall gain of 18 dB. The input and output connections of the individual sections are brought out separately to a 23-pole connector, permitting connection in any internal sequence.

TEVa:

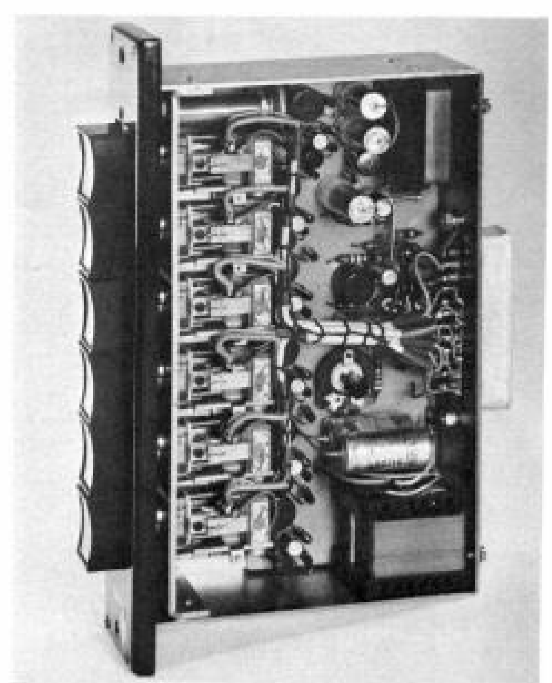
Frequency range:	40 - 15,000 cps.
Frequency response:	± 0.5 dB. (All controls at zero).
Equalization at 100 cps:	-15 dB to +9 dB in 3 dB steps.
Equalization at 10 KC:	-15 dB to +11 dB in 3 dB steps.
Presence equalization:	700, 1KC, 1.4KC, 2KC, 2.8KC, 4KC, 5.6KC.
	0 - +8 dB in 2 dB steps.
High frequency cutoff filter:	effective above 15 KC.
Input impedance:	> 1200 Ohms unbalanced.
Max. driving impedance:	70 Ohms.
Max. input voltage:	3.1 Volts (+12 dB).
Output source impedance:	< 60 Ohms unbalanced.
Load impedance:	> 600 Ohms.
Max. output level:	3.1 Volts (+12 dB).
Gain:	0 - 18 dB feedback selectable.
Total rms distortion:	< 0.5%.
Noise level referred to input:	< -101 dB (input 200; output 600).
Operating power:	12 VDC @ 30 mA.
Operating temperature:	0 - 45° C.
Transistors used:	4 x AC-151r.
Depth behind panel:	134 mm (5 1/2").
Weight:	2.9 lbs.

FREQUENCY CURVES ACHIEVED WITH THE TEVa TRANSISTOR EQUALIZER MODULE



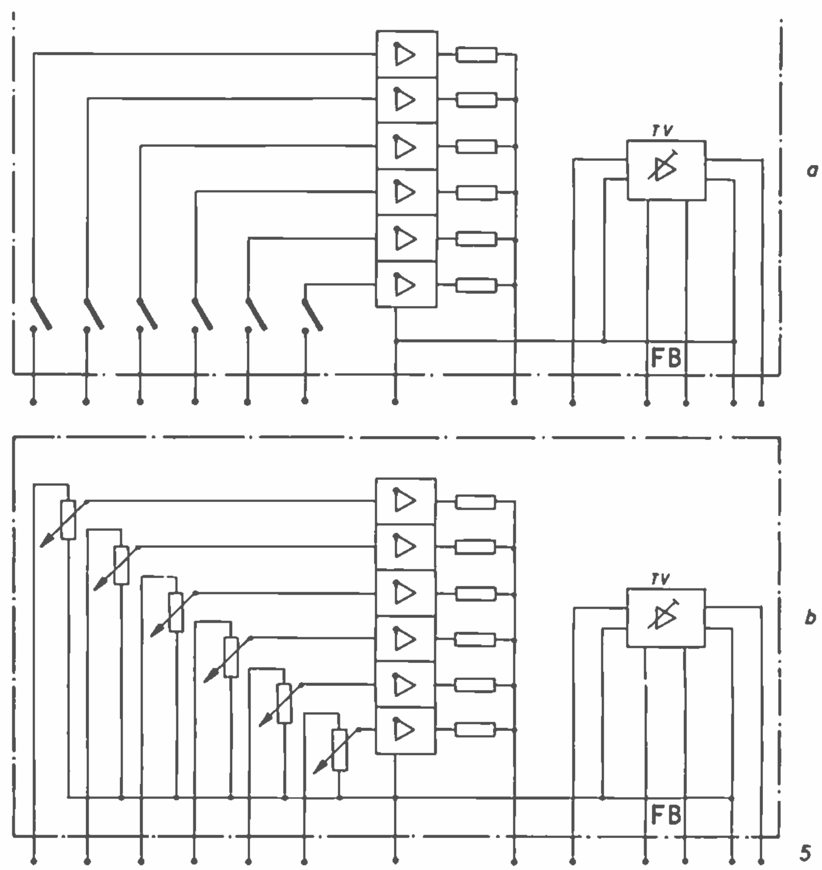
TRANSISTOR COMBINING AMPLIFIERS TSVTa

These amplifiers are intended for combining numerous microphone channels, or for that matter program channels, into a single output. Each of six unbalanced inputs feed into a collector-base amplifier (KBV) followed by a resistive combining network which is so chosen as to produce a constant loss regardless of the number of actual inputs connected to the unit. The TSVT employs push buttons to connect the inputs to the KBV Amplifiers. In the "off" position these buttons short the KBV inputs assuring that noise will be kept at a minimum level. The TSVR differs in that potentiometers are used at the six inputs, permitting the use of various input levels or, as in its use as an echo mix unit, the feeding of various proportions of the six input signals to the common output. The combining network output, as well as the input and output of the TV Amplifier are brought to a 23-pole connector. The overall gain of the units is adjustable between 6 - 12 dB by means of a self contained potentiometer. In order to make possible the series connection of several of these units, their output is balanced and floating. The TSVT buttons light when depressed.



TECHNICAL DATA: (Applies to both TSVTa & TSVRa unless indicated)

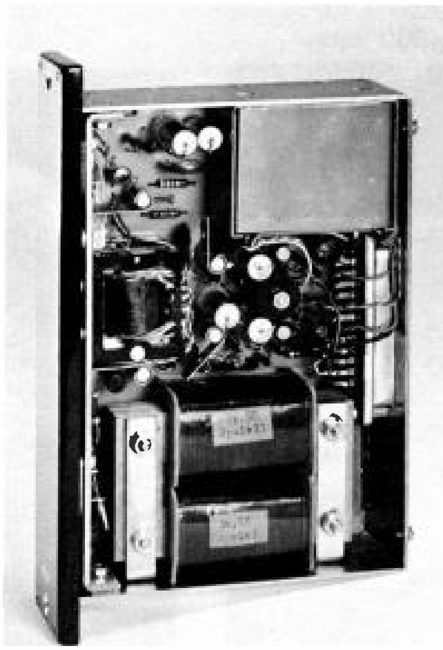
Frequency range:	40 - 15,000 cps.
Frequency response:	± 0.5 dB.
Input impedance: TSVTa:	> 8000 Ohms unbalanced.
TSVRa:	> 2500 Ohms unbalanced.
Max. input level:	0.775 Volts (± 0 dB).
Output source impedance:	< 80 Ohms balanced & floating.
Load impedance:	600 Ohms or more.
Max. output level:	3.1 Volts (+12 dB).
Gain:	6 - 12 dB variable on units.
Noise level referred to input:	< -100 dB (input 200; output 600 Ohms).
Total rms distortion:	< 0.5%.
Operating power:	12 VDC @ 50 mA.
Operating temperature:	0 - 45° C.
Transistors used:	9 x AC-151r.
Depth behind panel:	134 mm (5 1/2").
Weight:	TSVTa: 1.5 lbs. TSVRa: 2.2 lbs.



a) MODEL TSVTa PUSH-BUTTON JUNCTION AMPLIFIER
b) MODEL TSVRa POTENTIOMETER JUNCTION AMPLIFIER

TRANSISTOR LINE AMPLIFIER:

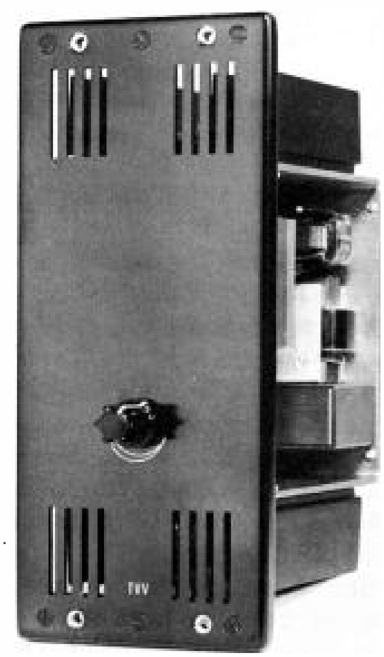
TLVa
TVV



The TLVa unit is a completely push-pull line amplifier module capable of a balanced and floating line output level of +22 dBm at a source impedance of < 60 Ohms. It will feed a 600 Ohm termination with less than 0.5% THD. The TLVa has a high impedance balanced input and a gain of 30 dB. It is intended for use wherever a standard buss or line is to be established with a level of up to +12 VU. It occupies the standard single module space. All connections are brought to a 23-pole connector at its back.

CUE SPEAKER AND HEADPHONE MONITOR AMPLIFIER:

The TVV Cue Amplifier was created to provide a 2 watt output at 5 Ohms to power a small cue speaker from low level program sources. Its high input impedance of 3000 Ohms (bal), permits switching by means of selector or push buttons or attenuator cue contacts, across any point in a console layout for instant check of circuit conditions. It may also be used to power headphones for overdubbing. The TVV is contained in a dual width module and has a volume control on its front panel.



TECHNICAL DATA:

TLVa:

Frequency range:	40 - 15,000 cps.
Frequency response:	± 0.5 dB.
Input impedance:	> 1000 Ohms balanced & floating.
Max. input level:	- 8 dB.
Output source impedance:	< 60 Ohms.
Load impedance:	> 600 Ohms.
Max. output level:	+ 22 dBm.
Gain:	30 dB.
Total rms distortion:	< 0.5%. at +22 dBm.
Noise level referred to input:	< 100 dB (input 200; output 600 Ohms).
Operating power:	12 VDC @ 350 mA.
Operating temperature:	0 - 45° C.
Transistors used:	4 x AC-151r; 2 x AC-153; 2 x AD-150.
Depth behind panel:	134 mm (5 1/2").
Weight:	2.6 lbs.

TVV:

Frequency range:	60 - 10,000 cps.
Frequency response:	± 1 dB.
Input impedance:	> 3000 Ohms balanced & floating.
Min. input level:	12 mV (- 36 dB).
Output load impedance:	5 Ohm speaker.
Max. output power:	2 watts.
Gain:	adjustable to 69 dB.
Total rms distortion:	< 3% at 2 watts.
Operating power:	12 VDC @ 700 mA.
Operating temperature:	0 - 45° C.
Transistors used:	3 x AC-151r; 2 x AC-153; 2 x AD-150.
Depth behind panel:	134 mm (5 1/2").
Width:	80 mm (3 1/4") Dual Module.

TTMV: PEAK INDICATING VOLUME METER AMPLIFIER.

Most of the European broadcasting and recording systems do not use the VU meter prevalent in the United States, but make use of a so-called "Peak Volume Indicator" comprising a logarithmic amplifier whose DC output is fed to a light beam type meter calibrated to read directly from -40 to +5 dB.

The TTMV amplifier has a 10 ms attack time and a 1.5 sec. return time (compared to 300 ms for both time constants on the VU meter). The result is a much more accurate indication of actual levels; i.e. peak levels. The advantage of the VU meter lies in the fact that in measuring "volume" it comes closer to the subjective indication of "loudness" as perceived by the human ear and is therefore a better indicator of levels when the program content is mixed; i.e. contains both speech and music as is usual in broadcasting. Gotham will be pleased to supply additional information on request.

TKV TPG 5

TKV TRANSISTOR TALKBACK AMPLIFIER

TPG-5 TRANSISTOR TEST OSCILLATOR

The TKV is a specialized transistor talk-back amplifier capable of bringing the output of a dynamic talkback microphone directly to line level while incorporating a 20 dB limiter which prevents undesirable talk-back speaker overload when excessive level is applied to the talk-back microphone. The unit occupies a single module housing and draws but 75 mA of current from the supply.

The TPG-5 is a Wien bridge RC oscillator having five fixed frequencies. Its output level meter provides direct output level control. Its output level can be continuously varied in two output level ranges 60 dB apart. Each of its outputs is available either balanced and floating, or unbalanced. The oscillator unit is mounted in a single module with meter and controls mounted on its top panel.

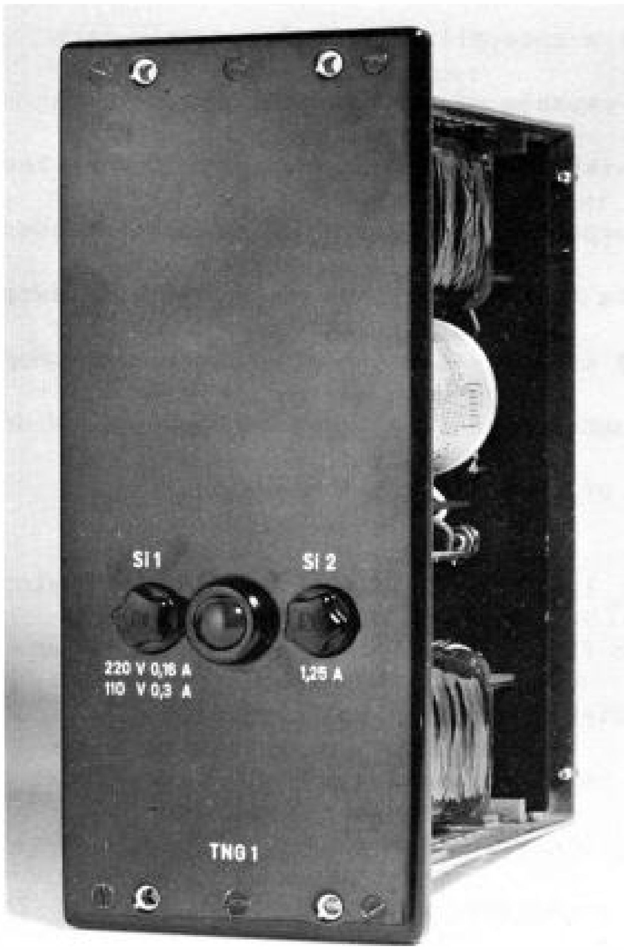
TKV TALKBACK AMPLIFIER DATA:

Frequency range:	100 - 10KC.
Input impedance:	> 1000 Ohms bal.
Input level requ.:	> -70 dB.
Gain:	76 dB.
Total rms dist.:	< 1%.
Output impedance:	< 60 Ohms bal.
Output level:	3.1 V (+12 dB).
Limiting, max:	20 dB.
Operating power:	12 VDC @ 75 mA.
Transistors used:	6xAC-151r;2xAC-153.

TPG-5 TEST OSCILLATOR DATA:

RC Oscillator freq:	60, 1KC, 7KC, 10KC, 14KC.
Max. output level:	1.8 Volts.
Cont. variable level:	0 - 1.8 Volts.
	0 - 0.18 Volts.
Output impedance:	< 60 Ohms; bal. & floating.
Load impedance:	> 200 Ohms.
Total rms dist.:	< 0.5%.
Operating power:	12 VDC @ 50 mA.
Transistors used:	2xAC-151r; 1xAC-153.

TNG
TNGL



POWER SUPPLIES:

The various transistor components are powered from 12 VDC supplies while the lamps in the TSVTa Push Button Combining Unit are powered from a well regulated 6 Volt lamp supply.

The 12 VDC current is available from one of two supply models: the TNG supplies two times 400 mA in separate supply sections, while the TNG-1 (illustrated) has a single output of 1000 mA.

The proper supply will be determined on the basis of how many modules are to be powered from a single supply for minimum interaction.

The TNGL Lamp Supply provides highly stabilized voltage for lamp and relay operation while the TNGLR gives the additional facility of a dimmer on its front panel to regulate lamp brightness.

TECHNICAL DATA:

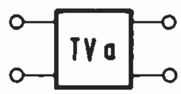
	TNG	TNG-1	TNGL (TNGLR)
AC Power:	117/220 VAC 50-60 cps all supplies. - - - - - - - - - - - - - - - - - -		
Output power:	2 x 12 VDC @ 400 mA.	12 VDC @ 1000 mA.	6.5 - 6.9 VDC @ 1 Amp.
Ripple, full load:	< 50 microVolts.	< 50 MicroVolts	< 1 mV.
Temperature range:	0 - 45^{o} C. all units -		
Transistors used:	2xAC-151r;2xTF-78/30.	2xAC-153;1xAD-130.	AC-151r;AC-153;2xAD-130 1 x TF-80/30.
Depth behind panel:	151 mm (6").	151 mm (6").	151 mm (6").
Width:	80 mm (3 1/4").	80 mm (3 1/4").	80 mm (3 1/4").
Weight:	5 lbs.	5 lbs.	4.4 lbs.

MIXING-CONSOLE-UNITS

Matching and level data

Input imped. ≧ 1,2 kΩ unbal.

Input voltage max. .39 V (-6 dB)



Source imped. ≦ 60 Ω unbal.

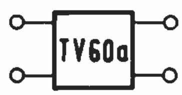
Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dB)

Ampl.=18...30 dB Current drain: 25 mA.

Input imped. ≧ 1,2 kΩ bal.

Input voltage max. 3.1 V (+12 dB) at 0 dB ampl.



Source imped. ≦ 80 Ω bal.

Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dB)

Ampl.=0...60 dB Current Drain: 40 mA.

Input imped. ≧ 1.2 kΩ bal.

Input voltage max. 3.1 V (+12 dB) at 0 dB ampl.

Source imped. ≦ 60 Ω unbal.(3 outputs)

Load imped. ≧ 600 Ω

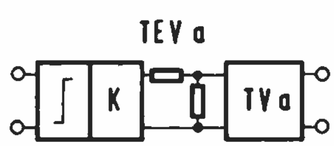
Output voltage max. 3.1 V (+12 dB)

Ampl.=0...60 dB Current Drain: 100 mA.

Input imped. ≦ 1.2 kΩ unbal.

Source imped. max. 70 Ω

Input voltage max. 3.1 V (+12 dB)



Source imped. ≦ 60 Ω unbal.

Load imped. ≧ 600 Ω

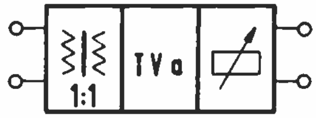
Output voltage max. 3.1 V (+12 dB)

-12 dB -6 dB +18 dB Current Drain: 30 mA.

Input imped. ≧ 1.2 kΩ bal.

Input voltage max. .1 V (-18 dB) at 30 dB ampl.

TRVa (circuit a)



Source imped. ≦ 60 Ω unbal.

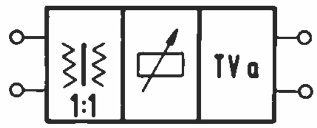
Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dB)

Input imped. ≧ .6 kΩ bal.

Input voltage max. 3.1 V (+12 dB) at 18 dB ampl. and 18 dB att.

TRVa (circuit b)



Source imped. ≦ 60 Ω unbal.

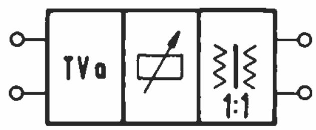
Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dB)

Input imped. ≧ 1.2 kΩ unbal.

Input voltage max. .1 V (-18 dB) at 30 dB ampl.

TRVa (circuit c)



Source imped. ≦ 300 Ω bal.

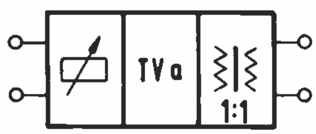
Load imped. ≧ 1 kΩ

Output voltage max. 3.1 V (+12 dB)

Input imped. ≧ .6 kΩ unbal.

Input voltage max. .1 V (-18 dB) at 30 dB ampl. and 0 dB att.

TRVa (circuit d)



Source imped. ≦ 80 Ω bal.

Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dB)

Current Drain: 25 mA.

Input imped. ≧ 8 kΩ unbal.

Input voltage max. 0.775 V (0 dB)

Ampl.=0...12 dB

Source imped. ≦ 80 Ω bal.

Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dB)

Current Drain: 50 mA.

Input imped. ≧ 2.5 kΩ unbal.

Input voltage max. 0.775 V (0 dB)

Ampl.=0...12 dB

Source imped. ≦ 80 Ω bal.

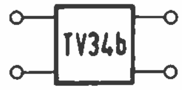
Load imped. ≧ 600 Ω

Output voltage max. 3.1 V (+12 dL

Current Drain: 50 mA.

Input imped. ≧ 3000 Ohms bal.

Input voltage max. .06 V (-22 dB)



Ampl. = 34 dB

Source imped. ≦ 40 Ω bal.

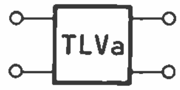
Load imped. ≧ 600 Ω

Output voltage max. 3.9 V (+14 dBm)

Current drain: 40 mA.

Input imped. ≧ 1000 Ohms bal.

Input voltage max. - 8 dB



Ampl. = 30 dB

Source imped. ≦60 Ohms.

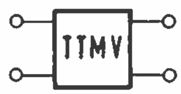
Load imped. ≧ 600 Ohms.

Output voltage max. 10 V. (+22 dBm).

Current Drain: 350 mA.

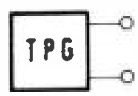
Input imped. ≧ 3 kΩ bal.

Input voltage min. .5 V for 100%



Output for J 47b

Current Drain: 300 mA.



Source imped. ≦ 60 Ω bal./unbal.

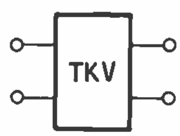
Load imped. ≧ 600 Ω

Output voltage max. 1.8/.018 V

Current Drain: 50 mA.

Input imped.: > 1000 Ohms

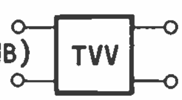
Input voltage: > -70 dB



Source imped.: < 60 Ohms.
Load imped.: > 600 Ohms.
Output voltage max. 3.1 V (+12 dB).
Current Drain: 75 mA.

Input imped.: > 3KOhms

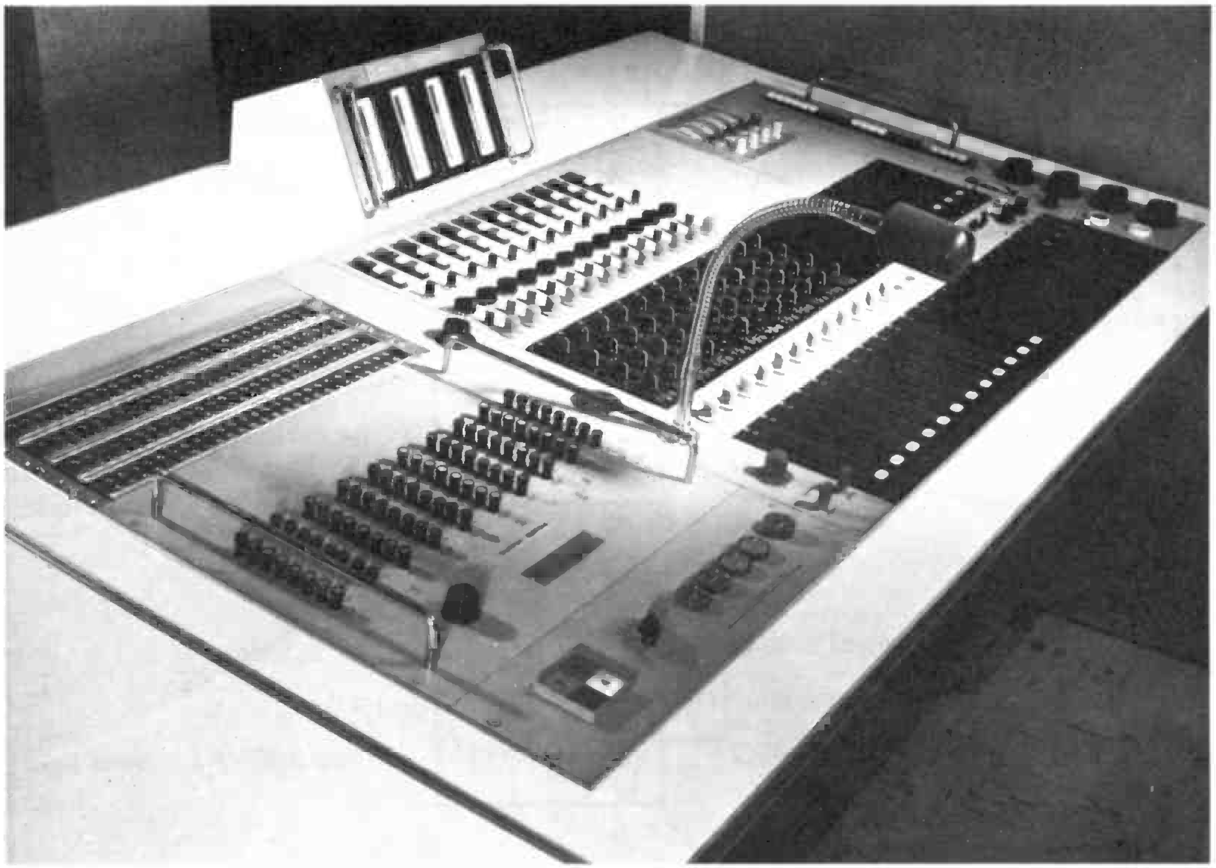
Min. input voltage: 12 mV (-36 dB)



Load imped.: 5 Ohm speaker voice coil.

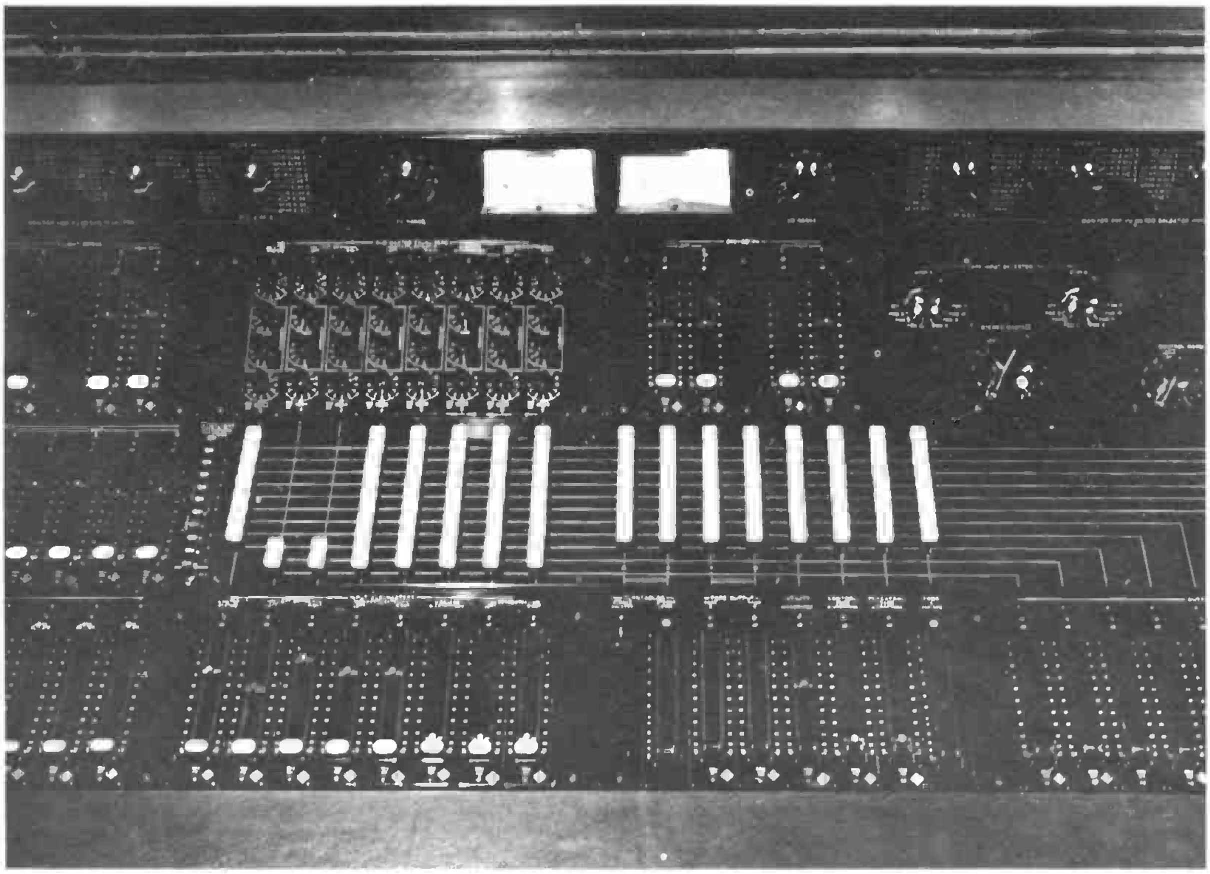
Output power: approx. 2 watts

Current Drain: 700 mA.



12 input x 4 channel Recording Console; Golden World Records, Detroit, Michigan
(built by Kenneth R. Hamann, Cleveland Recording Corp.)

Program production console with EMT-930st Turntables built for WDOK, Cleveland, Ohio by Kenneth R. Hamann, Cleveland Recording Corp.



Sound Reinforcement Console for the New York State Theater at Lincoln Center, New York built by Sound Systems Inc. Long Island City, N.Y.

Recording Studio Console at Dick Charles Recording Service, New York City designed by Gotham Audio Corporation; built by Magna-Tech Electronics, N.Y.C.

GOTHAM AUDIO CORPORATION

2 WEST 46 STREET, NEW YORK, N. Y. 10036 – (212) COLUMBUS 5-4111

neumann silicon transistor console components

PM 930 - 02 - 00

PV 46

The line amplifier PV 34 will no longer be manufactured. It is substituted by the line amplifier PV 46 . The technical data of the PV 46 correspond with those of the PV 34. The amplification can, however, be set to any figure between 20 and 46 dB through variation of the negative feedback value.

PV 15

In addition to the isolation stage PTS there is also the isolation amplifier PV 15 available. The amplification can be set to 10, 12 or 15 dB. The amplifier is mainly used for compensation of the channel fader loss.

HTS 600

The high-low pass filter is an L-C- network with balanced input and output which should be operated with ≦ 600 Ω source resistance and 600 Ω load resistance. The filter can be set for the following frequencies: 60, 125, 250, 500 Hz and 8, 10, 12, 14 kHz. The filter steepness is ≧ 12 dB/octave.

PMR

The mixing fader PMR is a balanced fader combination with 0 dB insertion loss. It is mainly used for mixing a compressor or the echo return to a modulation channel. The faders and the operating knob are contained in a console component unit with a front panel size of 40 x 64 mm.

PTMV

The peak level indicator has further been improved. It now contains a built-in oscillator for 0 % and 100 % calibration, and a push button for shifting the operation range by 20 dB for measuring smaller voltages.

PM 930-01-00

PV 46

Der Leitungsverstärker PV 34 wird nicht mehr hergestellt. Er ist durch den Leitungsverstärker PV 46 ersetzt. Die technischen Daten entsprechen denen des PV 34; die Verstärkung kann jedoch durch Änderung des Gegenkopplungsgrades von 20 ... 46 dB eingestellt werden.

PV 15

Neben der Trennstufe PTS steht nun auch ein Trennverstärker PV 15 zur Verfügung. Die Verstärkung ist durch Trafo-Umschaltung auf 10, 12, 15 dB einstellbar. Der Verstärker wird hauptsächlich zum Ausgleich der Reglergrundeinstellung verwendet.

HTS 600

Die Höhen-Tiefensperre HTS 600 ist ein LC-Filter mit symmetrischem Ein- und Ausgang, das mit ≦ 600 Ω Quell- und 600 Ω Abschlusswiderstand betrieben werden soll. Das Filter ist auf folgende Frequenzen schaltbar: 60, 125, 250, 500 Hz und 8, 10, 12, 14 kHz. Die Filtersteilheit ist ≧ 12 dB/Oktave.

PMR

Der Mischregler PMR ist eine symmetrische Reglerkombination mit 0 dB Einschaltdämpfung, die z.B. zum Einmischen eines Kompressors oder Rückführen des Halls in einen Übertragungsweg eingesetzt wird. Das Bedienungsteil ist als Mischpult-Kassette (40 x 64 mm) ausgeführt.

PTMV

Der Tonmesserverstärker PTMV ist weiterentwickelt worden. Er enthält nun einen eingebauten Eichgenerator für die 0 % / 100 % Eichung und eine Taste, mit der der Anzeigebereich zur Messung kleiner Spannungen um 20 dB verschoben werden kann.

In an effort to provide the modern studio technology with instruments of the highest quality, Neumann has delivered a series of components which may be combined into studio consoles for divers applications.

While the construction of a studio console depends in large measure on whether the console is to be installed in a mobile unit, is to be portable, or installed in one particular location, these components were so dimensioned as to leave the technician the widest latitude with regard to the physical layout of these devices. To this end, all of the components containing operating controls were installed in console plug-in units. The panel dimensions of these plug-in units are the usual ones for devices of this kind: 40 x 190 mm (approx. 1,5 x 7,5"). In this way, individual requirements can be met with regard to the juxtaposition of attenuators, equalizers, gain verniers, etc., and by maintaining this standard size it is possible to build these components into standard console panels.

Booster amplifiers, line isolation amplifiers, and isolation amplifiers are devices without operating elements. They were therefore mounted on plug-in cards (108 x 150 x 40 mm and/or 82 x 150 x 40 mm = approx. 4,3 x 5,8 x 1,6 and/or 3,2 x 5,8 x 1,6") in order that they may be mounted on the most favorable position inside the studio console without additionally requiring space on the console front panel.

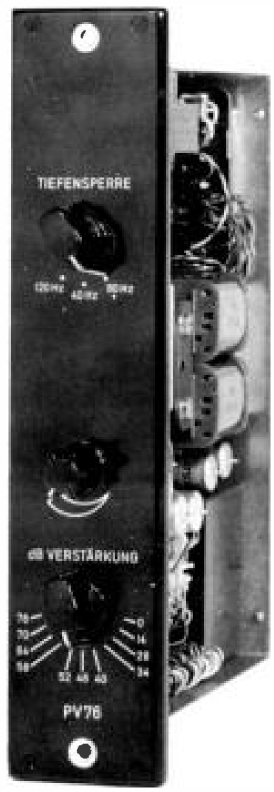
All of these devices are constructed utilizing silicon planar transistors. In view of the extended life expectancy of these semi-conductors, a very high degree of service-free operation over a period of many years can be expected. The heat generation of even largest of mixing consoles is negligible so that pleasant working conditions can be assured even in smallest of control rooms. On the other hand, these transistors permit an ambient temperature of + 60° C (140° Fahrenheit) so that even with the most unfavorable heat conditions undisturbed operation can be expected.

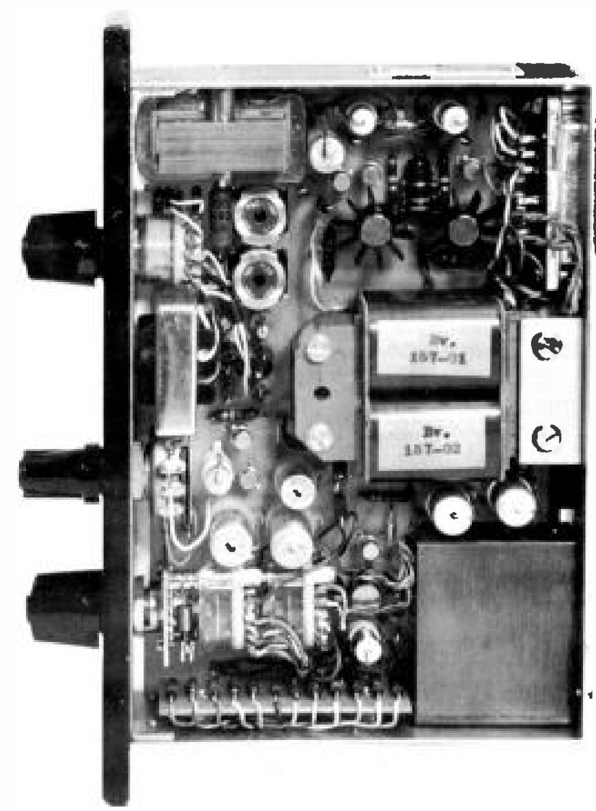
The fact that all of these devices feature balanced transformer inputs and outputs permits them to be integrated into any other studio technology. It is therefore possible to use, for example, the equalizer PEV or any other device from this P-series in a studio console that is equipped with tube amplifiers.

Another one of the important ear-marks of the Neumann silicon transistor console components is the high output level capacity. For an output level of + 21 dB terminated in 200 Ω, the total harmonic distortion for the entire frequency spectrum is $\leqq$ 0,5 %. All of these devices can be switched on their output connector to provide an output level of + 21 dB to a termination of 600 Ω with reduced current consumption. Any one of a number of well known attenuators may be utilized for mixing purposes. Balanced attenuators are to be preferred because of their higher isolation properties.

The Neumann silicon transistor console components are operated from 24 V DC. The ripple may not exceed 1 mV. Printed circuit mounted power supply cards are available supplying 24 V DC at 500 mA each.

MICROPHONE PRE-AMPLIFIER PV 76 (plug-in unit)





The microphone pre-amplifier PV 76 is enclosed in a plug-in console unit. It utilizes silicon planar transistors exclusively. Input and output are balanced and floating.

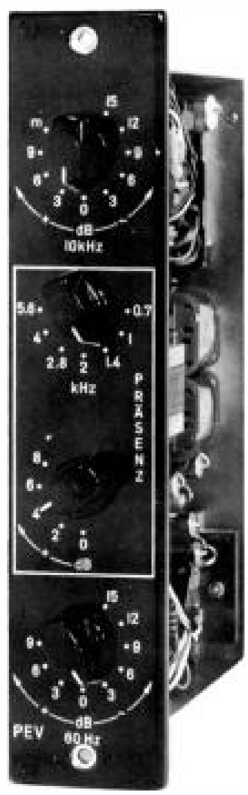
The PV 76 is an input amplifier for purposes of raising the microphone level to the standard + 6 dB level used in the console, which then permits mixing and equalizing of the signal. Maximum gain of the unit is 76 dB, adjustable in discreet steps and furthermore continuously adjustable between steps. A 3-frequency switchable built-in low frequency cut-off filter is provided.

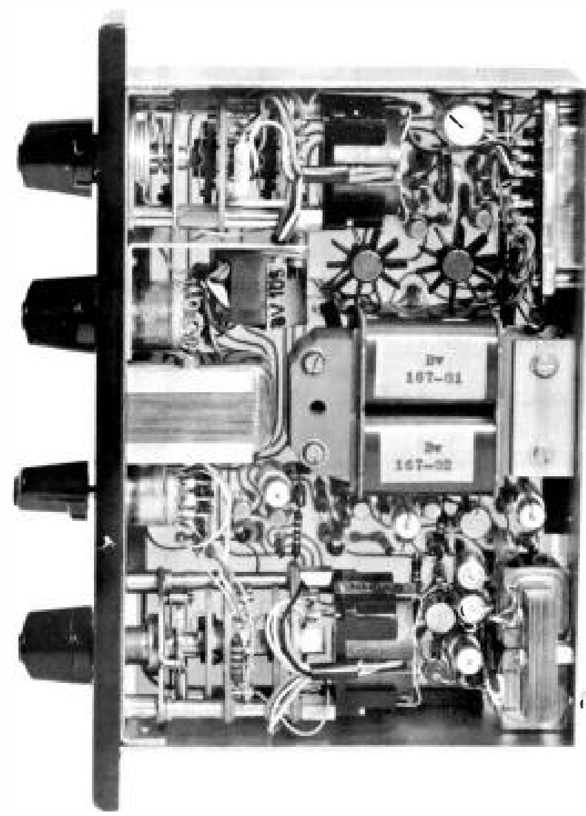
Maximum output level is + 21 dB to a termination of 600 Ω. Re-strapping permits this same level to be terminated into 200 Ω with commensurately higher current drain. All electrical connections are made via a 13-pole Tuchel rack connector.

Technical Specifications PV 76

Frequency range	40 ... 15 000 Hz
Frequency response 60 ... 15 000 Hz	$\leqq \pm 0,5$ dB
roll-off at 40 Hz	$\leqq$ 2 dB
roll-off at 20 Hz	$\geqq$ 12 dB
Low frequency cut-off filter with the frequencies	40 - 80 - 120 Hz
Filter steepness	$\geqq$ 12 dB per octave
above 15 kHz steady roll-off	
Input impedance 40 ... 15 000 Hz	$\geqq$ 1 kΩ balanced
Output source impedance	$\leqq$ 25 Ω balanced
Terminating impedance	$\geqq$ 600 Ω or $\geqq$ 200 Ω
Maximum output level	+ 21 dB
Gain switchable	0 ... 76 dB
in steps of 6 dB	28 ... 76 dB
below 28 dB adjustable to	14 and 0 dB
In-between values adjustable by potentiometer within the limits of	0 ... -6 dB
Total harmonic distortion in frequency range at output level of + 21 dB	
term = 600 Ω	$\leqq$ 0,5 %
term = 200 Ω	$\leqq$ 0,5 %
Weighted and unweighted noise level relative to input, for gain = 76 dB	weighted = -121 dB (DIN 45 405) unweighted = -123 dB
Operating voltage	24 V DC
Current consumption	
for term 600 Ω	90 mA
for term 200 Ω	160 mA
Maximum ambient temperature	+ 60° C (140°Fahrenheit)
Dimensions of front panel	190 x 40 mm (7.5 x 1.6")
Depth behind panel including mating connector	117 mm (4.6")
Weight	approx. 1.5 kg
Connector	T 2706

HIGH-LOW MID FREQUENCY EQUALIZER PEV (Plug-in unit)





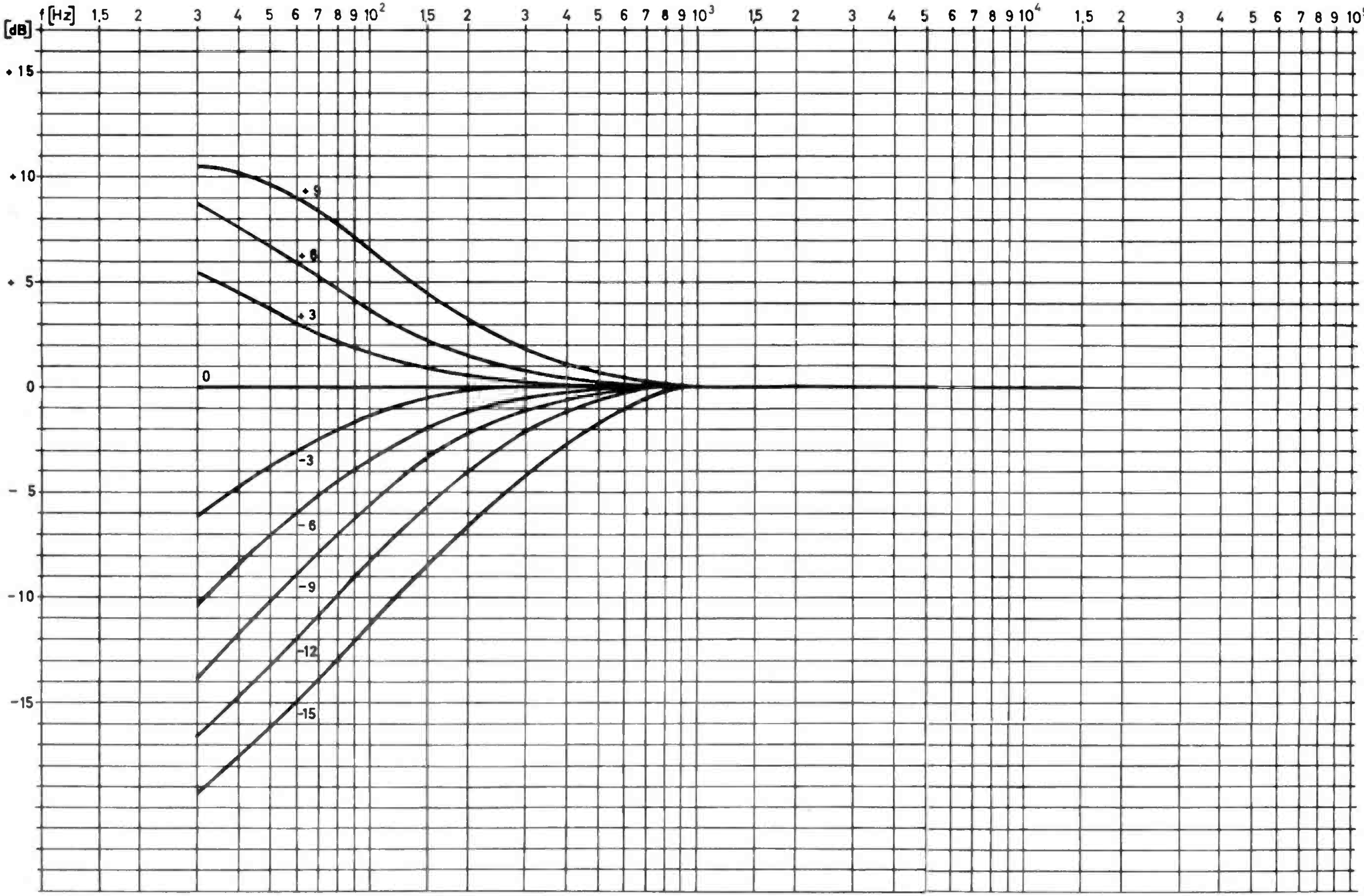
The equalizer PEV is enclosed in a plug-in console unit. A low and high frequency equalizer together with a mid-range band pass and an amplifier are combined to effect an active equalizer system. Input and output are balanced and floating.

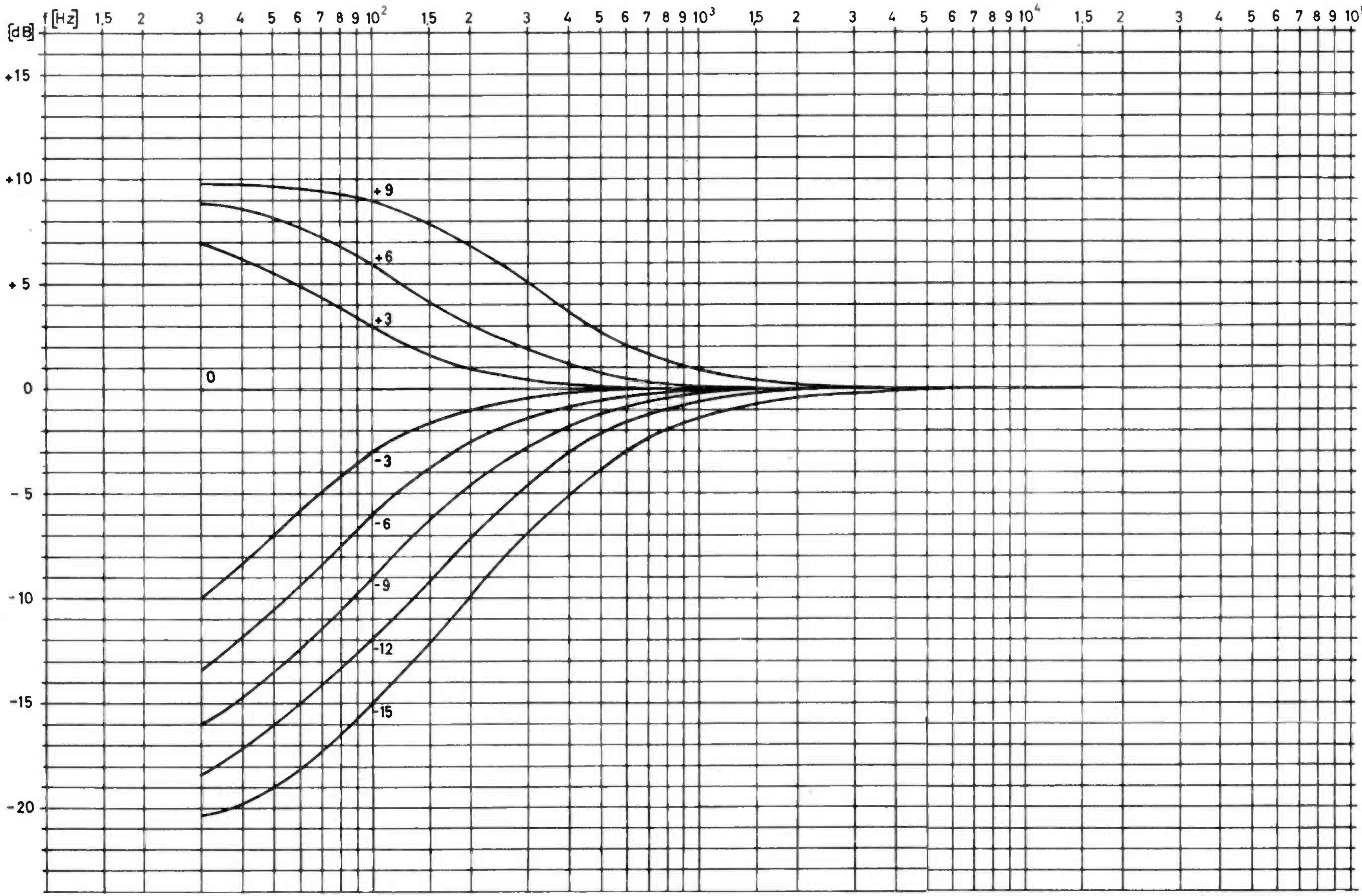
The high and low frequency equalization is accomplished using R-L components while the mid-range band pass utilizes L-C-R tuned circuits. The built-in amplifier is balanced and restores the gain lost by the equalizer network. The amplifier is equipped with silicon planar transistors. The equalizer's high input impedance permits connection to relatively high impedance sources without affecting the equalization curves. This permits the device to be utilized in many applications.

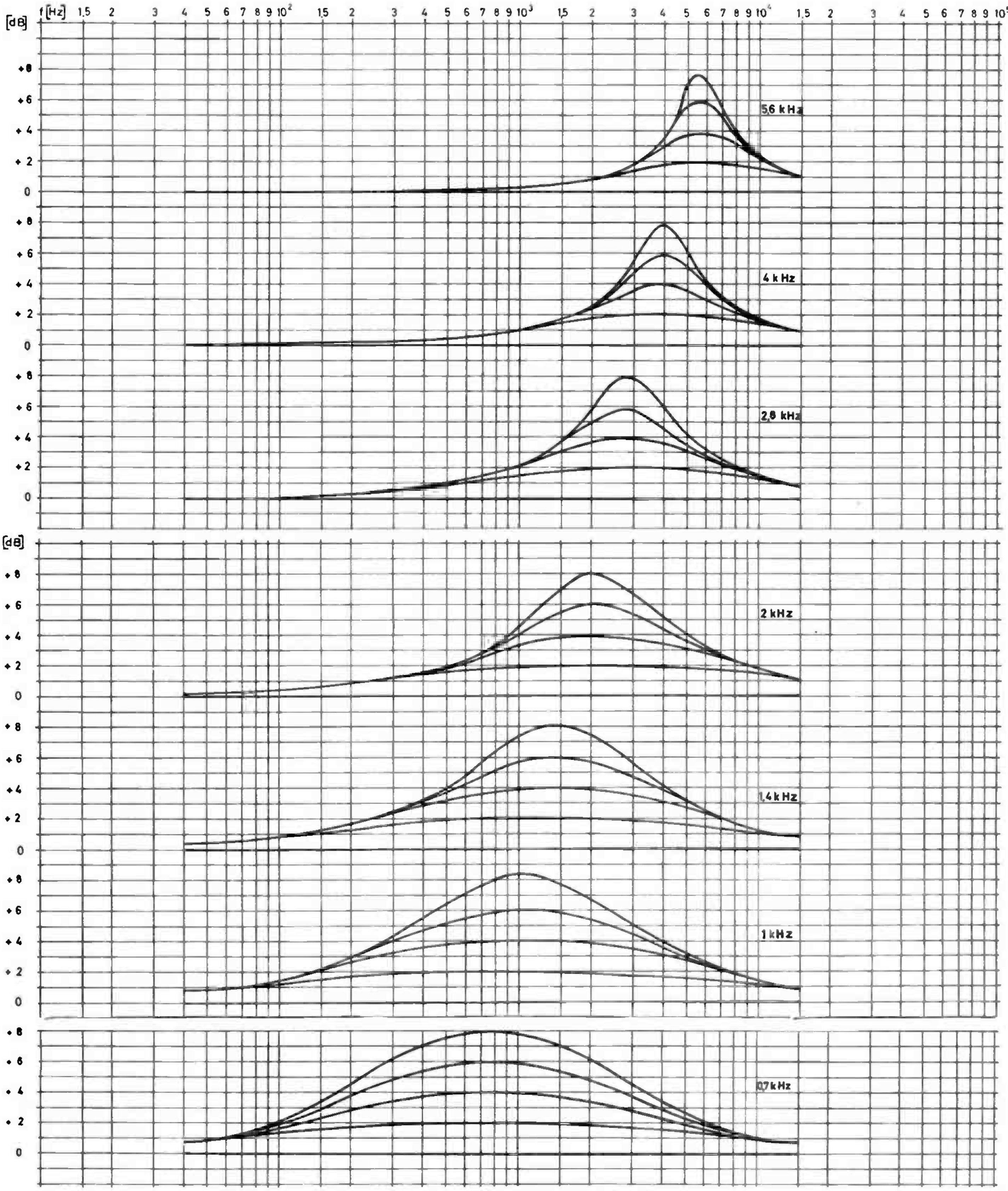
Maximum output level is + 21 dB to a termination of 600 Ω. Re-strapping permits this same level to be terminated into 200 Ω with commensurately higher current drain. All electrical connections are made via a 13-pole Tuchel rack connector.

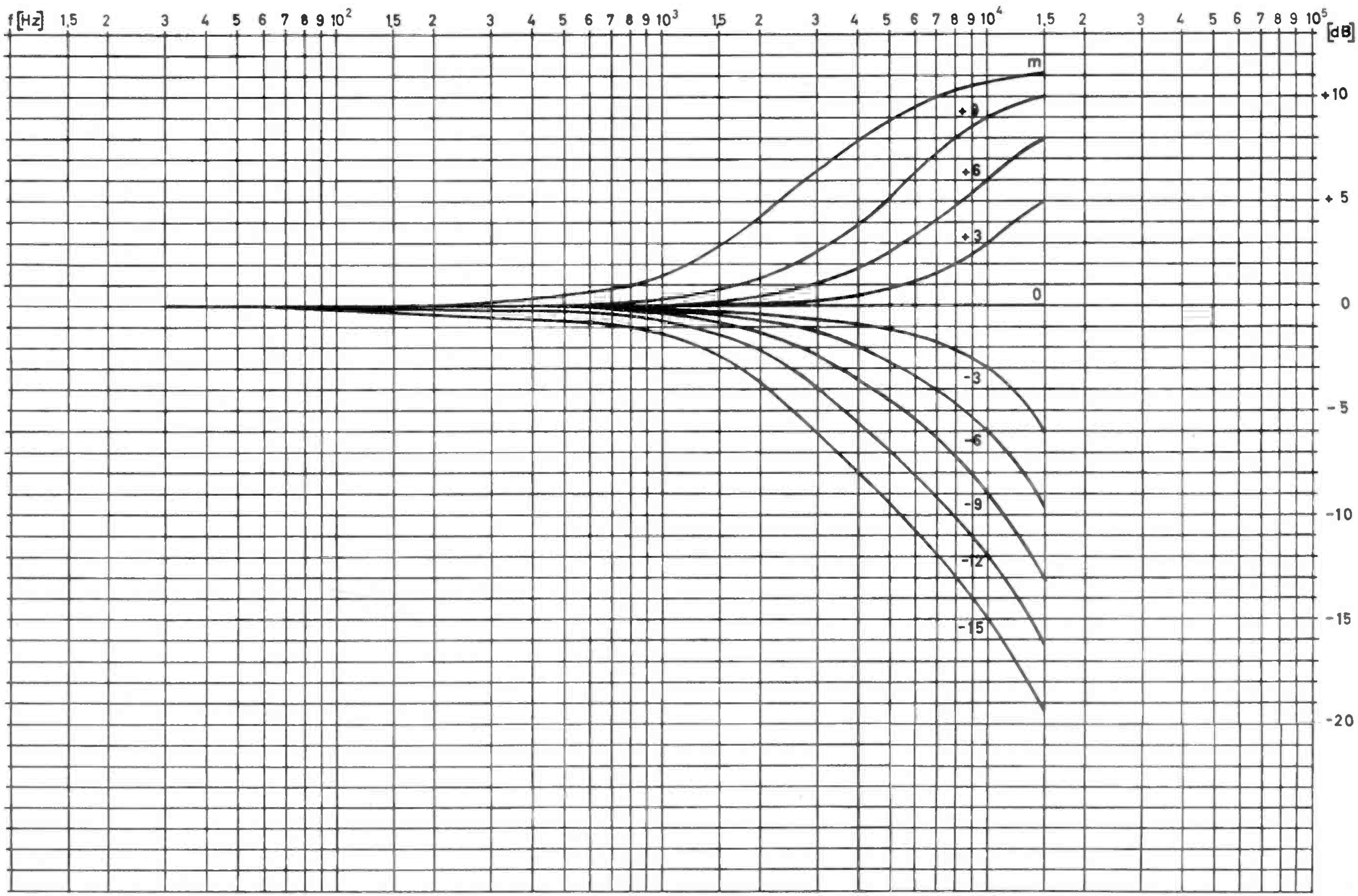
Technical Specifications PEV

Frequency range	40 ... 15 000 Hz
Frequency response, all switches in 0 position	$\leqq \pm$ 0,5 dB
Equalizing at 60 Hz or 100 Hz	-15 ... + 9 dB
Equalizing at 10 kHz	-15 ... +11 dB
adjustable in steps of 3 dB	$\pm$ 0,5 dB
Mid-range boost (presence)	0,7 - 1 - 1,4 - 2 - 2,8 - 4 - 5,6 kHz
adjustable in steps of 2 dB	0 ...+8 dB, $\pm$ 0,5 dB
Input impedance	$\geqq$ 4 kΩ balanced
Recommended source impedance	$\leqq$ 600 Ω
Output source impedance	$\leqq$ 30 Ω balanced
Terminating impedance, selectable	$\geqq$ 600 Ω, $\geqq$ 200 Ω
Maximum output level	+ 21 dB
Gain	0 dB, / $\pm$ 0,5 dB
Total harmonic distortion in frequency range at output level of + 21 dB,	
term = 600 Ω	$\leqq$ 0,5 %
term = 200 Ω	$\leqq$ 0,5 %
Weighted and unweighted noise level	weighted =-89 dB(DIN 45 405) unweighted = -89 dB
Operation voltage	24 V DC
Current consumption switchable	
for term = $\geqq$ 600 Ω	65 mA
for term = $\geqq$ 200 Ω	140 mA
Maximum ambient temperature	+ 60° C (140° F)
Dimensions of front panel	190 x 40 mm (7.5 x 1.6")
Depth behind panel including mating connector	117 mm (4.6")
Weight	approx. 1,8 kg
Conncector	T 2706

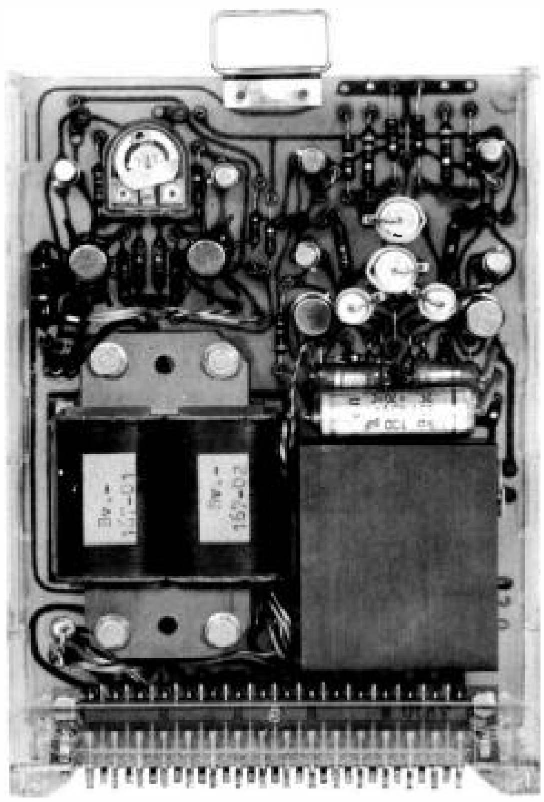

[dB]
f [Hz]
+15
+10
+5
0
-5
-10
-15
+9
+6
+3
0
-3
-6
-9
-12
-15


f [Hz]
1,5
2
3
4
5
6
7
8
9
10^2
1,5
2
3
4
5
6
7
8
9
10^3
1,5
2
3
4
5
6
7
8
9
10^4
1,5
2
3
4
5
6
7
8
9
10^5
[dB]
+15
+10
+5
0
-5
-10
-15
-20
+9
+6
+3
0
-3
-6
-9
-12
-15


f [Hz]
[dB]
56 kHz
4 kHz
2,8 kHz
2 kHz
1,4 kHz
1 kHz
0,7 kHz


f [Hz]
1,5
2
3
4
5
6
7
8
9
10^2
10^3
10^4
10^5
[dB]
+10
+5
0
-5
-10
-15
-20
m
+9
+6
+3
0
-3
-6
-9
-12
-15

LINE AMPLIFIER PV 34 (plug-in card)



The amplifier PV 34 is mounted on a plug-in printed circuit card. It utilizes silicon planar transistors exclusively. Input and output are balanced and floating. The gain is selectable by straps on the output circuit to 34, 40, or 46 dB.

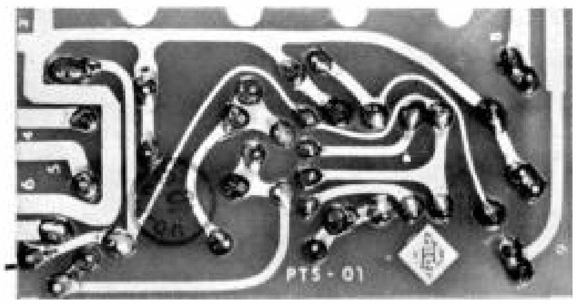
The amplifier is to be used as a line amplifier for console channel outputs. It can, however, also be utilized as a microphone preamplifier for restoring gain of such loss devices as attenuators and equalizers and combining networks as used when several microphone inputs are combined into one channel.

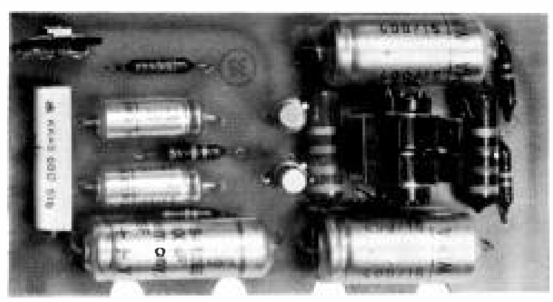
Maximum output level is + 21 dB to a termination of 600 Ω. Re-strapping permits this same level to be terminated into 200 Ω with commensurately higher current drain.

Technical Specifications PV 34

Frequency range	40 ... 15 000 Hz
Frequency response	$\pm$ 0,5 dB
Input impedance	$\geqq$ 2,5 kΩ balanced
Output source impedance	$\leqq$ 30 Ω balanced
Terminating impedance, selectable	$\geqq$ 600 Ω, $\geqq$ 200 Ω
Maximum output level	+ 21 dB
Gain, selectable	34 - 40 - 46 dB
Total harmonic distortion in frequency range at	
term = 600 Ω	$\leqq$ 0,5 %
term = 200 Ω	$\leqq$ 0,5 %
Weighted and unweighted noise level relative to input	
for gain = 46 dB	weighted = -120 dB(DIN 45 405) unweighted = -122 dB
for gain = 34 dB	weighted = -118 dB(DIN 45 405) unweighted = -120 dB
Operation voltage	24 V DC
Current consumption switchable	
for term = 600 Ω	65 mA
for term = 200 Ω	140 mA
Maximum ambient temperature	+ 60° C (140° F)
Dimensions	plug-in printed circuit card 108 x 150 x40 mm (4.3 x 5.8 x 1.6")
Weight	0,7 kg

ISOLATION AMPLIFIER STAGE PTS (plug-in card)





The isolation amplifier stage PTS is constructed in a balanced manner on a printed circuit board equipped with silicon planar transistors. Input and output are balanced. In special cases when a DC isolation is required, transformers may be installed additionally.

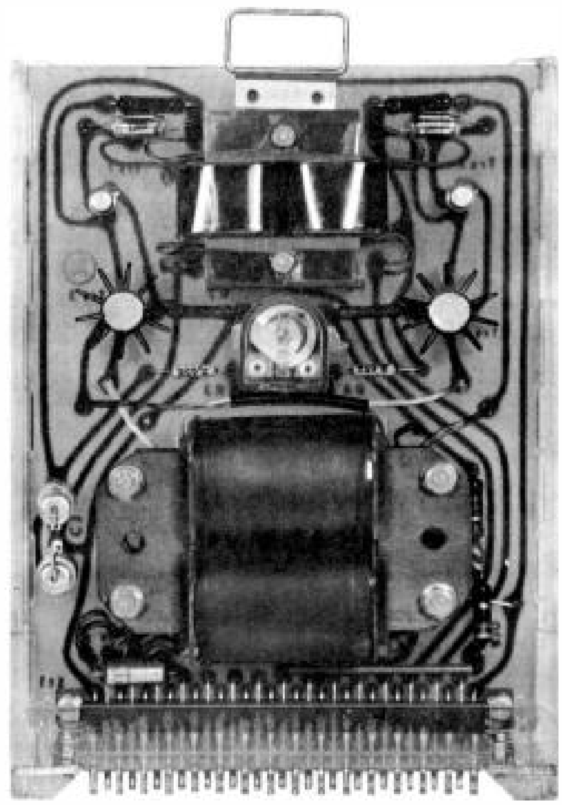
The isolation amplifier stage will find application in a console where especially stringent requirements with respect to isolation or channel separation are indicated. Through the use of such an isolation amplifier stage and by selecting the proper source and terminating impedances, isolation in excess of 80 dB can be achieved.

Since in actual practice numerous such isolation amplifier stages may be utilized, two such units are available on a 21-pole plug-in card and three such units on a 31-pole plug-in card.

Technical Specifications PTS

Frequency range	40 ... 15 000 Hz
Frequency response	≦ ± 0,5 dB
Input impedance	≧ 6 kΩ balanced
Output source impedance	≦ 4 Ω balanced
Terminating impedance	≧ 1 kΩ
Maximum output level	+ 21 dB
Gain	0 dB, -0,5 dB
Total harmonic distortion in frequency range at term = 1 kΩ and + 21 dB	≦ 0,5 %
Isolation	≧ 60 dB
Operation voltage	24 V DC
Current consumption	100 mA
Maximum ambient temperature	+ 60° C (140° F)
Weight	approx. 150 gr.
Dimensions	50 x 95 mm (2 x 3.7") (single printed board)

LINE ISOLATION AMPLIFIER PLTV (plug-in card)



The line isolation amplifier PLTV is constructed in a balanced manner on a plug-in card and is equipped with silicon planar transistors. Input and output are balanced and floating.

The PLTV is an output amplifier for mixing consoles. Its outputs are adjustable to conform either to the post office level of + 15 dB or + 9 dB, or the broadcast level of + 6 dB. The output source impedance is $\leqq$ 30 Ω. A + 6 dB monitor and control output, transformer isolated, is available beside the main line output. Devices connected to this control output via 2 x 20 Ω series resistors will not influence the main output level by more than 1 dB for the case of a short circuit.

The isolation afforded by the PLTV output to input is $\geqq$ 90 dB. Any disturbance appearing at the output of this amplifier will therefore not influence the input. Maximum output level is + 21 dB to a termination of 600 Ω. Re-strapping permits this same level to be terminated into 200 Ω with commensurately higher current drain.

Technical Specifications PLTV

Frequency range	40 ... 15 000 Hz
Frequency response	± 0,5 dB
Input impedance	≧ 2,5 kΩ balanced
Output source impedance, output I + II	≦ 30 Ω balanced
Nominal output level, output I	+ 15 dB (4,4 V) max. +21 dB (8,8 V)
and/or	+ 9 dB (2,2V) max. +15 dB (4,4 V)
output II	+ 6 dB (1,55 V) max. +13 dB (3,5 V)
Terminating impedance, selectable, output I	≧ 600 Ω, ≧ 200 Ω
selectable, output II	≧ 600 Ω, ≧ 200 Ω
Gain, output I	9 dB, ± 0,5 dB
output II	0 dB, ± 0,5 dB
Isolation in the frequency range	≧ 90 dB
Total harmonic distortion in frequency range	
at + 21 dB, term = 600 Ω	≦ 0,5 %
at + 21 dB, term = 200 Ω	≦ 0,5 %
Weighted and unweighted noise level relative to input	weighted = -100 dB (DIN 45 405) unweighted = -100 dB
Operation voltage	24 V DC
Current consumption for term = 600 Ω	65 mA
for term = 200 Ω	135 mA
Maximum ambient temperature	+ 60° C (140° F)
Dimensions	plug-in printed circuit card (108 x 150 x 40 mm) (4.3 x 5.8 x 1,6")
Weight	approx. 0,75 kg

"PAN-POT" PR

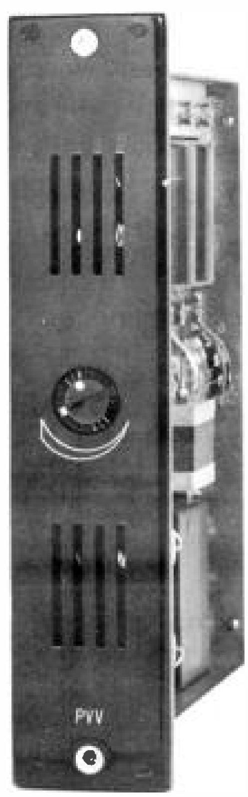
The "Pan-Pot" PR consists of two balanced attenuators with contrary motion which are mounted on one shaft and are contained within a console component unit with a front panel dimension of 40 x 64 mm (1.6 x 2.5 ") plus two isolation amplifier stages PTS, mounted on one plug-in card.

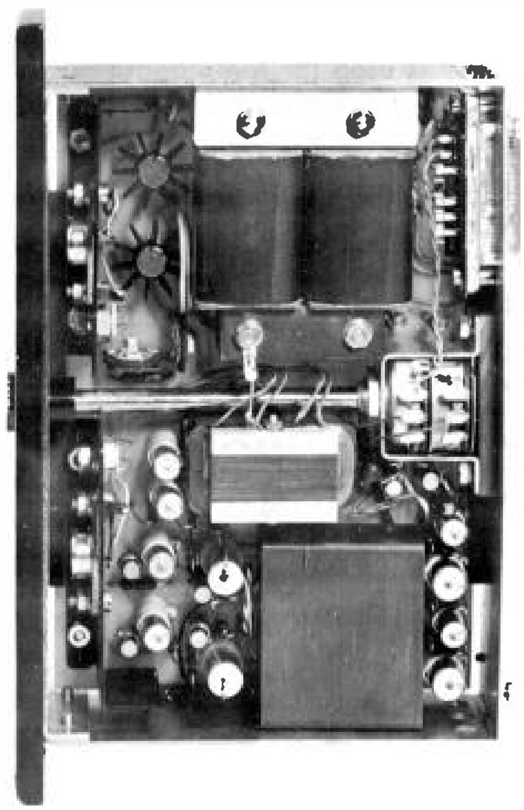
The "Pan-Pot" is used for the purpose of distributing a signal between two channels with perfect isolation. The signal may be moved from one side to the other in 15 detented attenuator positions. The gain curves were so selected that the acoustical impression appears to move in front of the listener along a straight line and not along a convex semi-circle. This attenuator maintains a constant power sum of the two channels in any position.

Technical specifications PR

Frequency range	40 ... 15 000 Hz
Frequency response	$\pm$ 0,5 dB
Input impedance	$\geqq$ 600 Ω balanced
Output source impedance	2 x $\leqq$ 4 Ω balanced
Terminating impedance	$\geqq$ 1 kΩ
Maximum output level	+ 21 dB
Recommended source impedance	$\leqq$ 120 Ω
Minimum loss	$\leqq$ 1,5 dB
Total harmonic distortion in frequency range at +21 dB, term 1 kΩ	$\leqq$ 0,5 %
Operation voltage	24 V DC
Current consumption	approx. 200 mA
Maximum ambient temperature	+ 60° C (140° F)
Arc of rotation	180°
Switch position	15
Dimensions of the front panel	40 x 64 mm (1.6 x 2.5 ")
Connector	T 2706
Dimensions of the card Length of the plug-in card reduced to 105 mm	plug-in printed circuit card 82 x 150 x 40 mm (3.2 x 5.8 x 1.6")
Weight	approx. 400 gr. + 100 gr.

PREVIEW AMPLIFIER   PVV   (plug-in unit)





The preview amplifier PVV is enclosed in a plug-in console unit. It utilizes silicon planar transistors exclusively. Input and output are balanced and floating.

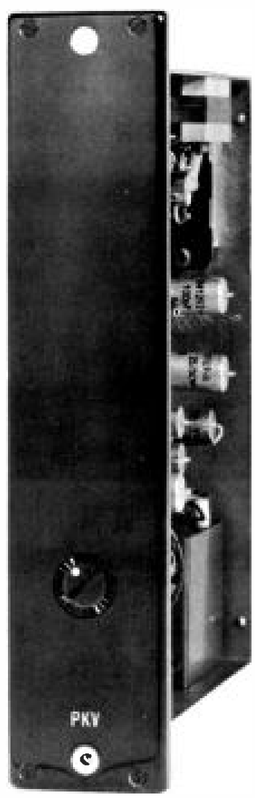
The amplifier is principally intended for the operation of a small loudspeaker mounted in the studio console. Its full output power of 5 W is reached with an input level of only 50 mV. This device may also be used as a monitor amplifier for small consoles.

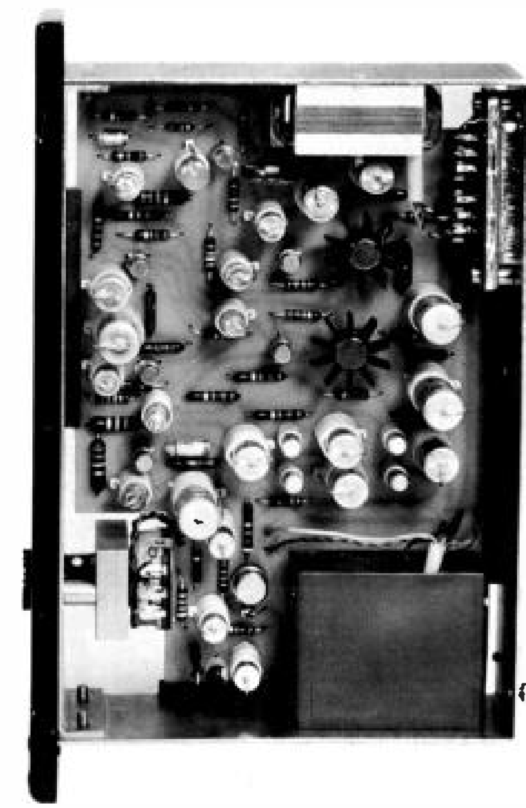
All electrical connections are made via a 13-pole Tuchel rack connector.

Technical Specifications PVV

Frequency range	60 ... 15 000 Hz
Frequency response	± 0,5 dB
Input impedance	≧ 5 kΩ balanced
Output source impedance	5 Ω
Output power at 5 Ω in the range of 100 ... 15 000 Hz	5 W
Input voltage for 5 W Input level adjustable	50 mV (-24 db)
Harmonic distortion at 5 W, 5 Ω	≦ 2 %
Harmonic distortion at 50 mW, 5 Ω	≦ 0,5 %
Signal-to-noise ratio at 50 mW	≧ 62 dB
at 5 W	≧ 82 dB
Operation voltage	24 V DC
Current consumption	
Idling current	100 mA
Maximum current at 5 W	400 mA
Maximum ambient temperature	+ 60° C (140° F)
Dimensions of front panel	190 x 40 mm (7.5 x 1.6 ")
Depth behind panel including mating connector	117 mm (4.6 ")
Weight	approx. 1,8 kg
Connector	T 2706

TALK-BACK AND LIMITER AMPLIFIER PKV (plug-in unit)





The talk-back and limiter amplifier PKV is enclosed in a plug-in console unit. It utilizes silicon planar transistors exclusively. Input and output are balanced and floating.

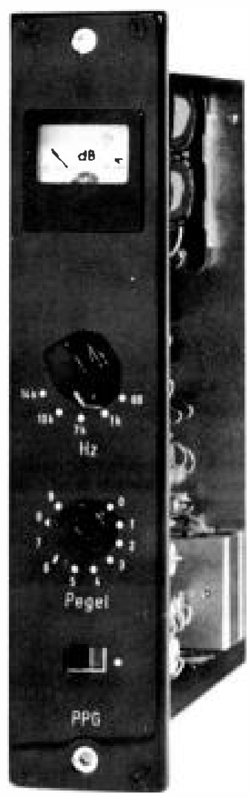
The amplifier is intended for the amplification of the output voltage of a dynamic talk-bac microphone to the standard studio level of + 6 dB. The output level remains constant for ar overload of up to 20 dB. The gain is continually adjustable.

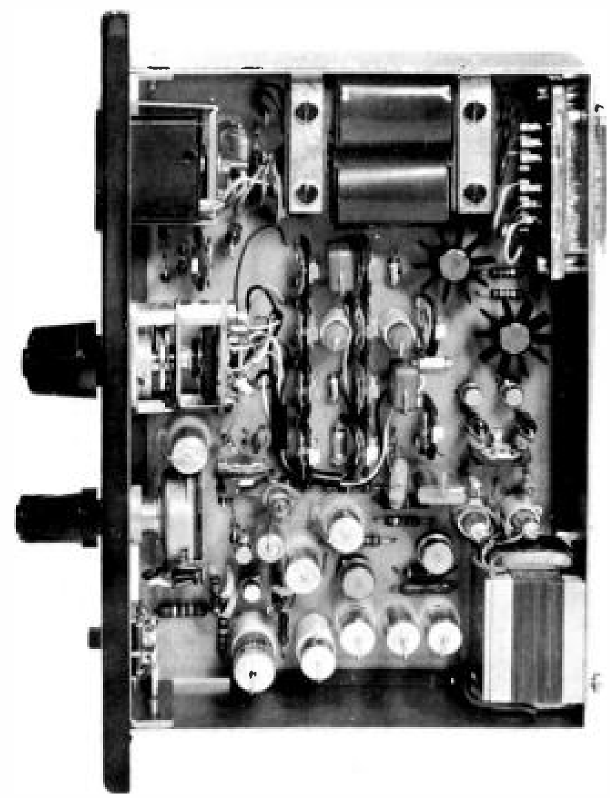
All electrical connections are made via a 13-pole Tuchel rack connector.

Technical Specifications PKV

Frequency range	300 ... 6 000 Hz ± 1 dB
below 300 Hz and above 6 kHz	steady roll-off
Input impedance	≧ 1 kΩ balanced
Output source impedance	≦ 25 Ω balanced
Terminating impedance	≧ 600 Ω
Gain continually adjustable from	76 ... 56 dB
Limiter action: maximum overload	20 dB
The limitation occurs at an output level of	+ 3 ... + 5 dB
Output level at max. overload of 20 dB	+ 6 ... + 7 dB
Limiter attack time	approx. 5 ms
Recovery time	approx. 2 s
Distortion for maximum gain below limiter threshold within the frequency range ..	≦ 0,5 %
for maximum limitation	≦ 3 %
Weighted and unweighted noise level relative to input for gain = 76 dB	weighted =-119 dB(DIN 45405) unweighted = -121 dB
Operation voltage	24 V DC
Current consumption	55 mA
Maximum ambient temperature	+ 60° C (140° F)
Weight	approx. 1,1 kg
Dimensions of the front panel	190 x 40 mm (7.5 x 1.66")
Depth behind panel including mating connector	117 mm (4.6")
Connector	T 2706

TEST OSCILLATOR PPG (plug-in unit)





The test oscillator PPG is enclosed in a plug-in console unit. It utilizes silicon planar transistors exclusively. The output is balanced and floating.

The PPG is a wien-bridge R-C oscillator. Five frequencies are selectable, and its output level can be adjusted in two ranges. The operating elements and a voltmeter are mounted on the front panel.

All electrical connections are made via a 13-pole Tuchel rack connector.

Technical Specifications PPG

RC oscillator with 5 frequencies	60 / 1000 / 7000 / 10 000 / 14 000 Hz
Maximum deviation between individual frequencies	$\leqq \pm 0,5$ dB
Frequency deviation	$\leqq 3$ %
Nominal output level for output I continually adjustable from	+ 6 dB -40 ... +10 dB
Output source impedance I	$\leqq 20\ \Omega$ balanced
Terminating impedance	$\geqq 200\ \Omega$
Nominal output level for output II continually adjustable from	-54 dB -100 ... - 50 dB
Output source impedance II	$\leqq 10\ \Omega$ balanced
Terminating impedance	$\geqq 200\ \Omega$
Total harmonic distortion at term = $200\ \Omega$ and output level of + 10 dB	$\leqq 0,5$ %
Operation voltage	24 V DC
Current consumption	60 mA
Maximum ambient temperature	+ 60° C (140° F)
Dimensions of front panel	190 x 40 mm (7.5 x 1.6")
Depth behind panel including mating connector	117 mm (4.6")
Weight	approx. 1 kg
Connector	T 2706

PEAK LEVEL INDICATOR PTMV

The peak level indication amplifier PTMV is used for the measuring of the console outputs. Both the light beam indication instrument J 47b as well as other instruments with like characteristics may be connected to it.

The balanced input of this transistorized amplifier is followed by a low pass filter which restricts the transmission range above 15 kHz. Generation of the logarithmic function of the amplifier is accomplished through the use of semi-conductors. The peak level indication amplifier PTMV is contained within a plug-in console unit. The connections are made via a 13-pole Tuchel rack connector T 2706. Two potentiometers on its front plate are provided for the separate adjustment of 0 % to 100 % pointer indication. The alignment level for 100 % may be obtained from the transistor oscillator PPG.

Technical Specifications PTMV

Frequency range	40 ... 15 000 Hz
Dynamic indicating range	-40 ... +5 dB
Input impedance	≧ 10 kΩ balanced
Minimum input voltage for 100 %	500 mV
Indicating error for steady 1 kHz, 0% and 100% calibrated in the range from:	
-40 ... -5 dB	± 2 dB
- 5 ... +5 dB	± 1 dB
Frequency dependency of the indication at 100 %	± 1,5 dB
Rise time for 90% of steady indication	approx. 10 ms
Overshoot of the pointer for 100% steady tone	≦ 1,5 dB
Recovery time of 100% to 10%	1,5 s
Operation voltage	24 V DC
Current consumption	approx. 200 mA
Maximum ambient temperature	+ 60° C (140° F)
Dimensions of front panel	190 x 40 mm (7.5 x 1.6")
Depth behind panel including mating connector	117 mm (4.6")
Weight	approx. 1,5 kg
Connector	T 2706

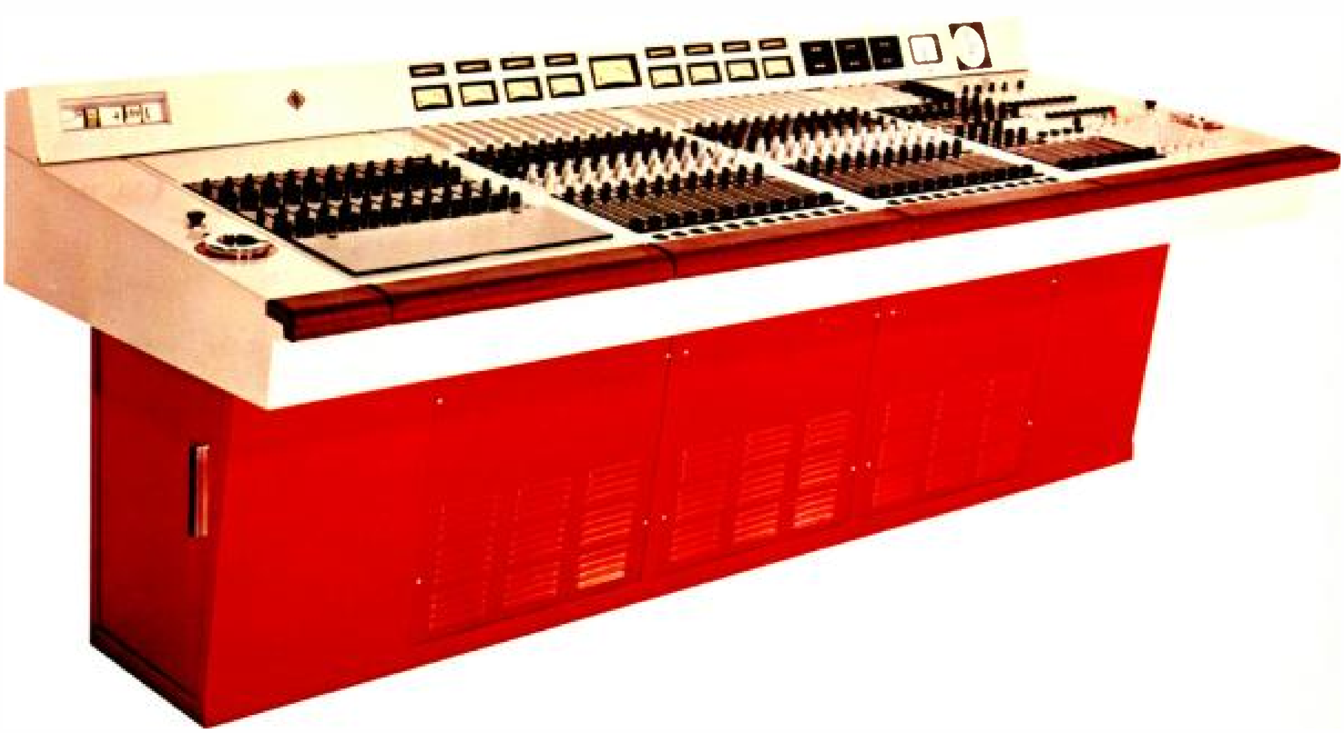
GEORG NEUMANN GMBH ELECTROACUSTIC · 1 BERLIN 61 CHARLOTTENSTR. 3 · TEL. (03 11) 61 48 92 · TELEX 01 84 595

GOTHAM
AUDIO CORPORATION
2 WEST 46 STREET, NEW YORK, N.Y., 10036 • 212-CO-5-4111

6-e

NEUMANN 8-CHANNEL 24-INPUT STUDIO CONSOLE

A SUPERIOR INSTRUMENT DESIGNED FOR THE PROFESSIONAL



THE GOTHAM-NEUMANN MODEL 24/8 CUSTOM CONSOLE

THE CONCEPT:

The ever rising prices for custom built consoles coupled with the many problems encountered by both builders and clients alike, has prompted GOTHAM to engage in a "first" in the industry. Neumann is building for us five identical consoles featuring 24 inputs and eight output channels, expandable to 16. The engineering investment prior to the start of construction was some $ 20,000, which will now be amortized over five units rather than being spent on a single console, as is usually the case. Never before has any company undertaken the "stocking" of consoles of such magnitude. These consoles will become available at the rate of about one per month beginning November 1, 1968.

OVER-ALL CONSIDERATIONS:

The 24/8 consoles will be equipped with NEUMANN console components of the "P" series featuring silicon transistor technology throughout. The console desk will be 102" in length and will be fully enclosed on all sides, the wiring being entered through the base pedestal. Rear flush doors provide access to all plug-in card frames as well as the terminal blocks for connection to the outside. A complete built-in patch bay using 312 "Bantam" ADC tip-ring-sleeve jacks will provide access to all inputs, outputs, switching junctions and facilities which might be involved in cross-connection requirements. The patch bay is recessed at the left end of the desk and is covered by a hinged glass panel.

THE CONSOLE INPUTS:

The console is equipped to handle 24 inputs of any level, between -80 dBm and +22 dBm. The inputs are balanced and floating and are provided with vernier controls which permit optimum input gain settings for any microphone sensitivity or distance from the sound source while maintaining constant input noise. The vernier is adjustable in 6 dB steps with an additional smooth control between steps. A high pass filter may be switched to 40, 80 and 120 Hz cut-off frequencies. An additional 24 line level inputs by-pass the 76 dB gain input unit (PV-76) and may be switch selected in place of the microphone inputs. The last eight line inputs on the console (17-24) are normaled from the eight playback lines of the 8-track recorder, but may be patched to accept any line level feeds.
Two NK-48 power supplies provide all 24 inputs with NEUMANN FET-80 transistor microphone compatible power, and any outlets in the studio connected to the console will autmomatically provide all FET-80 NEUMANN microphones with power, without in any way disturbing any other type of microphone employed. The 24 PV-76 input preamp units are grouped at the left side of the console, clearly numbered, so as not to clutter the operating area in front of the studio engineer.

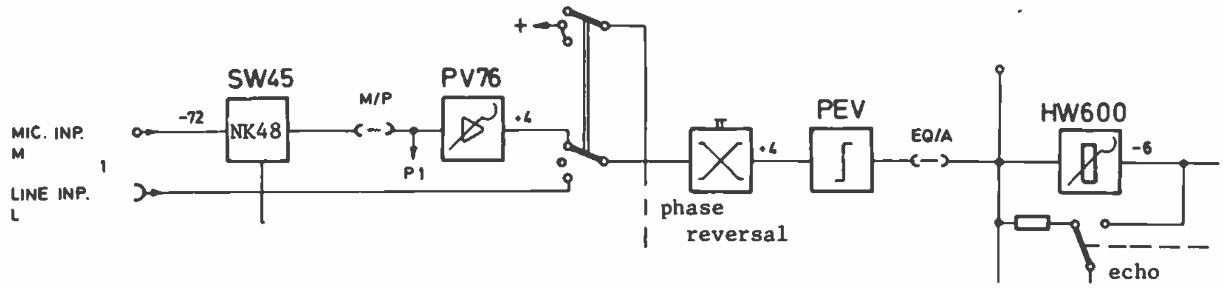
There are four high level inputs in addition to the above, which handle the ECHO RETURN lines.

INPUT EQUALIZATION:

Each of the 24 inputs is equipped with a PEV no loss low/mid/high frequency equalizer permitting low and high frequency boosts to +12 dB and droops to -15 dB, while the mid-range band-pass provides seven selectable frequencies between 700 and 5600 Hz and a boost of up to +8 dB at any one of these. These equalizers are effective in addition to the 3-frequency low end cut-off filters which are part of the PV-76 input units. The PEV equalizer outputs are capable of levels to +22 dBm and are brought out separately to solder terminals from which a 24-track tape machine may be fed directly at line level. A phase reversing switch provides reversal preceding the PEV equalizers.

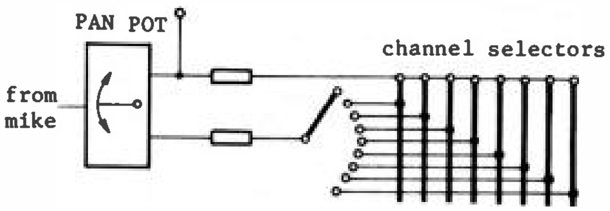
INPUT ATTENUATORS:

There is only one way in which to insure maximum channel separation in console inputs: that is to use meticulously balanced circuits. This is simply basic physics regardless of the present day vogue for eliminating transformers. To maintain the input circuits balanced, the attenuators used must be balanced ladders. The 24/8 console uses GOTHAM linear motion balanced ladder attenuators using 56-step precision elements, fully silicone encapsulated and guaranteed against noise for a full five years. The attenuators have 85 dB attenuation before infinity, providing for smooth fades. (HW-600). Separation both between input circuits and output lines is greater than 65 dB at 10 kHz and more than 75 dB in the mid range.



PAN POTS:

Each of the 24 input circuits is provided with a pan pot. Each of these consists of an attenuator element on the console panel and a plug-in amplifier card in a card carrier at the rear. The PAN POTS permit panning between any combination of push-button selected channels on the one hand, and any single channel selected using the pan pot selector located below each linear attenuator.

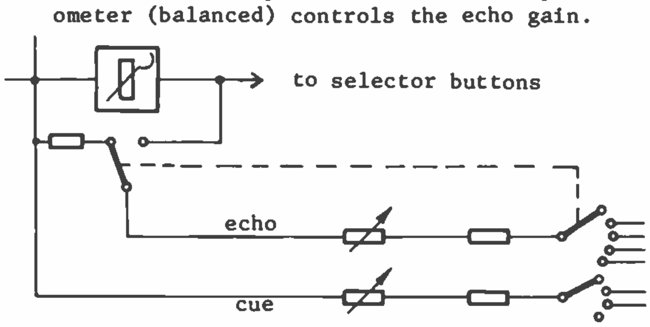


CHANNEL SELECTION MEANS:

Each of the 24 inputs is provided with a row of eight lighted push buttons. These connect the particular input to any one, any combination, or all eight output channels without interaction, level loss, or crosstalk. Each button is of the push-push type. The four ECHO RETURN circuits each have a nine button row of the same type. The ninth button permits Echo to be returned to the MONO MIX channel. Depressing this ninth button automatically releases any of the other eight that may have been depressed.

ECHO CIRCUITS:

Each of the 24 inputs has a concentric echo buss selector switch and echo potentiometer. The echo buss selector has 11 positions: four each to select the four ECHO SEND busses either before or after the linear attenuator, and three "off" positions. The echo potentiometer (balanced) controls the echo gain.



Each of the four ECHO SEND BUSSES is equipped with a 46 dB gain booster amplifier (PV-46) followed by four multi-frequency cut-off filter units (HTS-600), followed by four HW-600 balanced ladder linear attenuators (see "Input Attenuators"). Four PV-15 line amplifiers capable of +22 dBm output levels feed special jacks in the patch bay and terminals in the rear of the console to which the new EMT-970 Audio Tape Delay System may be connected. The EMT-970 units may be instantly by-passed using the four push-push buttons provided. The output levels are easily supervised on the four "magic eye" type cathode ray indicators.(N-840).

CUE BUSS SYSTEM:

A second selector/potentiometer system similar to that used in the ECHO SEND system described above, feeds CUE for use in either headphones or cue speakers to one of two CUE BUSSES from BEFORE the input attenuator. The two CUE BUSSES are equipped with CUE MASTER potentiometers and "magic eye" indicators (N-840). Also selectable to the two CUE BUSSES is the output of a nine position mixer using miniature slide potentiometers. This CUE MIXER has at its inputs the eight playback channels of the 8-track machine plus the mono machine output. (Also see further facility under "Monitor System").

CONSOLE OUTPUT CHANNELS: (MULTI)

The eight combining busses feed via eight 46 dB booster amplifiers to externally connectable limiting amplifiers. These may be instantly by-passed using the by-pass buttons on the console desk. A dual element ladder linear attenuator (KW-600/2) is provided for every two output channels, making four master attenuators in all. (NOTE: for special requirements these may be replaced by dual attenuators with separate slide actuators, giving individual control of all eight channels). Eight PV-46 Line Amplifiers deliver up to +22 dBm output level to the eight main console OUTPUT CHANNELS. Eight API VU meters are provided for level indication.

MONITOR AND MONO OUTPUT MIXER:

The console is equipped with a third slide potentiometer mixer, with eight inputs and a master. This mixer is fed from the switch arms of eight relays which allow for switching between console output and tape output. The mixer output feeds two separate facilities: first, a combining network forming a MONO MIX BUSS; second, the bank of 4 X 8 push buttons serving the monitor speaker selector bank. The mixer permits selecting the relative levels at which the eight channels - either console or 8-track tape - are to be heard on the speakers or fed to the mono output channel.

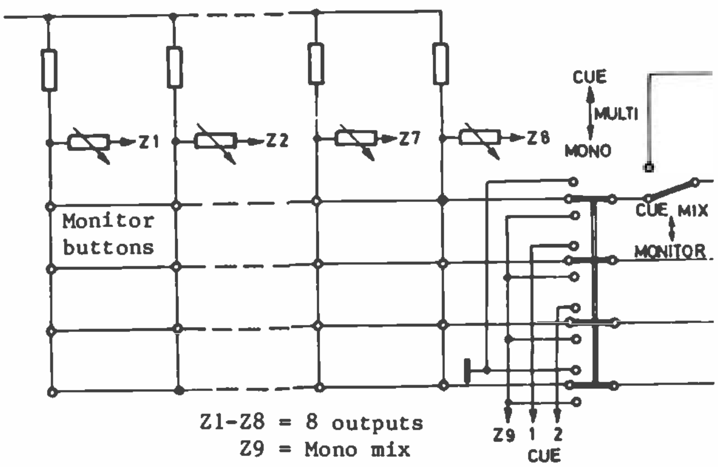
MONOPHONIC COMBINED OUTPUT:

The combined output of the MONO MIX BUSS described above is fed via a PV-46 46 dB booster amplifier and a master slider control through a PV-46 Amplifier to the MONO OUTPUT CHANNEL. A centrally located large API VU meter indicates MONO level. It is into this

MONO OUTPUT CHANNEL that the four ECHO RETURN channels are returned using the ninth push-button in the 9-button row. (see MEANS FOR CHANNEL SELECTION).

MONITOR SELECTOR SYSTEM:

Although GOTHAM feels that two control room monitor speakers should not be exceeded, four sets of selectors have been provided. A bank of 8 X 4 buttons allows the eight outputs of the MONITOR AND MONO OUTPUT MIXER to be selected to any of four control room speaker circuits in any combination without cross talk. The buttons are of the push-push type and may be added or cancelled individually. A ninth button actuates a master solenoid which releases all depressed buttons in that row. Two further switches are concerned with the monitor system: the MONO-MULTI-CUE switch provides simultaneous switching of all four monitor output channels to either the MONO OUTPUT CHANNEL, (on all 4 speakers at once), the four selector banks, or CUE BUSSES 1 & 2 on speakers 2 & 3, while speakers 1 & 4 are silenced. The second switch permits speaker channel one to be switched to the output of the CUE MIXER (see CUE BUSS SYSTEM).

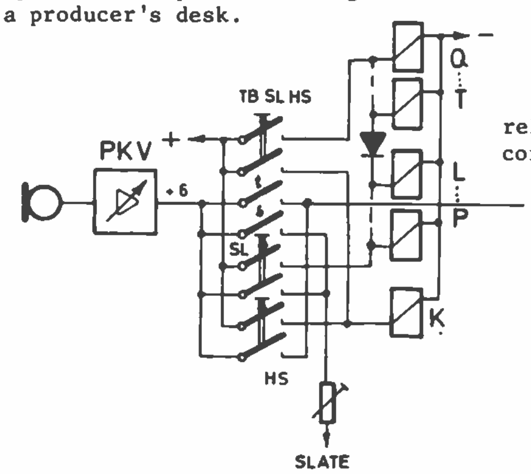


CONTROL ROOM AND STUDIO SPEAKER OUTPUTS:

There are line level connections for eight loudspeakers: speakers 1 - 4 to the control room and speakers 3-8 in the studio. The studio speakers are connected in parallel to their respective control room speaker circuits. A relay system prevents feedback through the studio speakers. Eight separate speaker controls are provided.

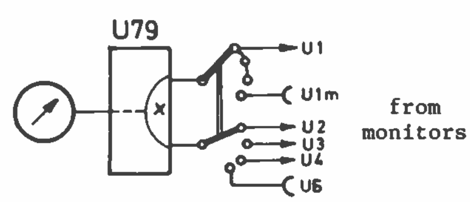
THE TALK BACK SYSTEM:

The console is provided with a built-in dynamic talk back microphone and talk back limiting preamplifier (PKV) whose line level output may be applied to various points by means of relays. 20 dB of talk back limiting prevents speaker blast-out. Three push buttons provide the following announce functions: 1) TB-HS-SL: this provides announce function simultaneously into the studio (via speaker 6), into both CUE CHANNELS (and therefore into all headphones), and into all 8 output channels and the MONO MIX channel. 2) SL: provides slating in the 8 output channels and MONO MIX; and 3) provides announce function in the CUE BUSSES only. Operating these buttons, simultaneously operates relays which mute the four control room speakers to a level preventing feedback but permitting two-way conversation without letting go of the button. Remote connections are provided to permit multing these functions at a producer's desk.



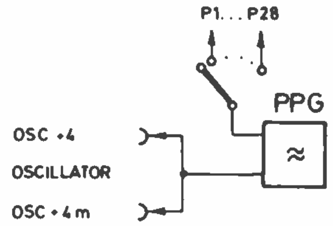
CORRELATION METERING:

One very important consideration in stereo or multi-track recording is the degree of "correlation" of the signals. The "correlation" is a significant indicator of compatibility in the final recording. The NEUMANN U-79 Correlation Meter is permanently mounted on the console turret and its amplifier card is mounted in a card carrier below. One of the two inputs of the U-79 is permanently connected to Monitor Channel 1, while the second input may be selected by means of a rotary switch to either Monitor Channel 2, 3, or 4, and to a patchable jack in the patch bay. The indications of the U-79 are entirely independent of relative levels in those monitor circuits!



REVERBERATION CONTROLS:

Four complete sets of remote controls for the EMT-140FB Reverberation Unit are provided on the console. These include ON/OFF switching for the units, push button for increasing and decreasing reverberation decay time, and meters calibrated in seconds to indicate decay time selected.

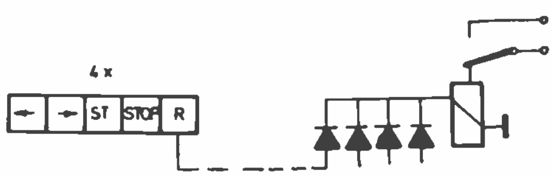


TEST OSCILLATOR CIRCUITRY:

As is usual in all NEUMANN consoles a five frequency precision low distortion RC oscillator is provided (PPG-5) with its own output level meter. A 25-position switch selects the -70 dB output of the oscillator to any of the 24 inputs to the console. (plus OFF position). Four more positions feed +4 dB output to the ECHO RETURN inputs of the console, while two +4 dB jacks in the patch bay allow other testing uses. The oscillator frequencies are: 60 Hz, 1, 7, 10 and 14 kHz at less than 0.5% THD.

TAPE REMOTE CONTROLS:

Four sets of tape recorder remote control buttons are provided with indicator lights to show both play and record functions. Each record button is connected through a diode to a muting relay which turns on the SILENCE sign outside the studio whenever any of the tape machines is in its RECORD function.



STUDIO SPEAKER LINE MUTING:

To prevent the studio speaker feeds from entering the microphone circuitry and causing feedback, the input switches which select either microphone or line level inputs (or OFF) are equipped with contacts which operate studio speaker muting relays such that the studio loudspeakers can only be on whenever all microphone switches are in either their OFF or LINE positions. This is standard broadcast practice and has been shown to be the only "safe" way to operate.

TIE LINES AND MULTS:

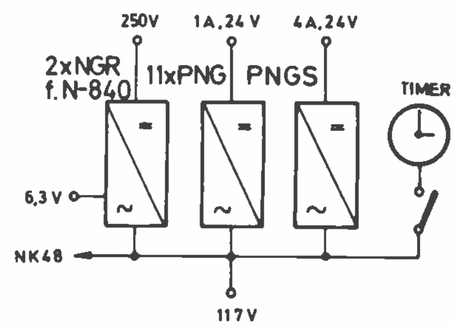
Twenty (20) TIE LINES from the patch bay to the terminal blocks are provided to permit remote equipment such as limiters, filters, special equalizers, etc. etc. to be accessible on the console. Four sets of 4-jack mult strips are provided.

POWER SUPPLY BANKS:

The console operates from 117 Volts 60 Hz lines. It is equipped with 11 type PNG supplies delivering a total of 24 amps at 24 VDC. a PNGS 24 VDC supply delivers 4 Amps of current for operation of relays and tally lamps, while two NGR supplies provide both 6.3 VAC and 250 VDC to the N-840 "magic eye" output meters. Two NK-48 supplies for phantom powering of NEUMANN FET-80 Microphones have been described in the first section of this description.

STOP TIMER:

A 4" electric STOP TIMER is mounted on the turret panel. A START/STOP and a RESET button remotely operate this timer. Parallel connections may be made to a producer's desk.



LIMITER FACILITIES:

As described earlier (see CONSOLE OUTPUT CHANNELS - MULTI), the console is equipped to operate eight external limiting amplifiers. Edgewise Meters above each of the eight channel output meters provide gain reduction metering, while by-pass push buttons allow the limiters to be circumvented. Enough rack space is provided for mounting such limiters at the back of the console. Small limiters of dimensions similar to console attenuators may be mounted alongside the VU meters in the console turret.

EXPANSION TO SIXTEEN (16) OUTPUT CHANNELS:

Should it ever prove expedient to augment the output capabilities of this console to 16 or even more channels, a sub-consolette would be placed alongside the engineer. A cable from it would plug directly into rack connectors in the side of the 24/8 console and pick up the 24 input channel feeds just before their push-button selectors. Such a sub-consolette could then be equipped with any number of additional output channels capable of accepting these 24 feeds one at a time or in any combination. The eight gain reduction VU meters provided on the main console could be re-connected to meter the additional channels.

PRE-FADE CUE SYSTEM:

A unique system is provided on the 24/8 console which permits instant identification of individual microphone outputs. Each of the 24 inputs features a momentary push button. Depressing this cuts off Control Room Speakers 1, 2 & 3, and switches speaker 4 to a special monitor buss fed from the points following each input attenuator. Therefore any button pushed will "display" that input channel only on the monitor system without disturbing any recording in progress. These buttons may be freely pushed during recording.

MECHANICAL CONSTRUCTION:

The console desk itself is all steel welded construction with a baked on chip resistant lacquer. The console top raises up in four separate bays and may be propped up with four support bars provided inside. All sections are reinforced steel angle welded structure. All plug-in component units may be removed from the top while all connectors are fixed mounted in recessed pans.

ADDITIONAL CONVENIENCE FEATURES:

1. Flush mounted ash trays located at each end of the console desk.
2. A digital clock with date and day-of-week readout.
3. A padded front edge trim for the sake of comfortable operation.
4. Electric auto-type cigarette lighters at each end of console desk.

COMBINED FACILITIES AND TECHNICAL SPECIFICATIONS

Microphone inputs:	24
Line level inputs:	24
Main output channels:	8
Secondary output channels:	
Mono combine channel:	1
Echo send channels:	4
Cue busses:	2
Monitor outputs:	8
Echo return inputs:	4
Pan pots:	24
Program equalizers (PEV)	28
Cut-off filters (HTS-600)	4
VU meters- level control:	9
gain reduction:	8
Peak indicators:	6
Correlation meters:	1
Linear attenuators:	50
Jacks in patch bay:	312
Push buttons:	302
Separate amplifiers:	85
Max. gain - microphone inputs:	96 dB
line inputs:	20 dB
Input level - microphone inputs:	-72...+21 dBm
line inputs:	-16...+21 dBm

Input sources:	30...600 Ohms
Output levels, maximum:	+21 dBm
Output terminating impedances:	
Channels & busses:	≥600 Ohms
Echo sends:	≥600 Ohms
Monitor outputs:	≥5 kOhms
Output source impedances:	
Channels & busses:	≤25 Ohms
Echo sends:	≤10 Ohms
Monitor outputs:	≤25 Ohms
Frequency response:	40...15,000 Hz ±1 dB
Tot. harm. distortion, entire range	
at nominal levels:	≤0.5%
at maximum levels:	≤1 %
Noise level (wtd) rms:	
@ -54 dB input & +4 dBm out:	-73 dB
@ -37 dB input & +21 dBm out:	-90 dB
(equivalent input noise ≤ -127 dB)	
Channel separation:	≥75 dB @ 1 kHz ≥65 dB @10 kHz
Max. ambient temperature:	+50° C.
Power requirement:	117 V +10% -15% 50/60 Hz

TERMS: As agreed. FOB JFK International Airport.
DELIVERY: Nov.1968; Jan.1969; Mar.1969; May 1969; Jul.1969.

PRICES: Equipped with 20 inputs $________

Equipped with 24 inputs $________

GOTHAM
AUDIO CORPORATION
2 WEST 46 STREET, NEW YORK 36, N. Y. • 212-CO 5-4111

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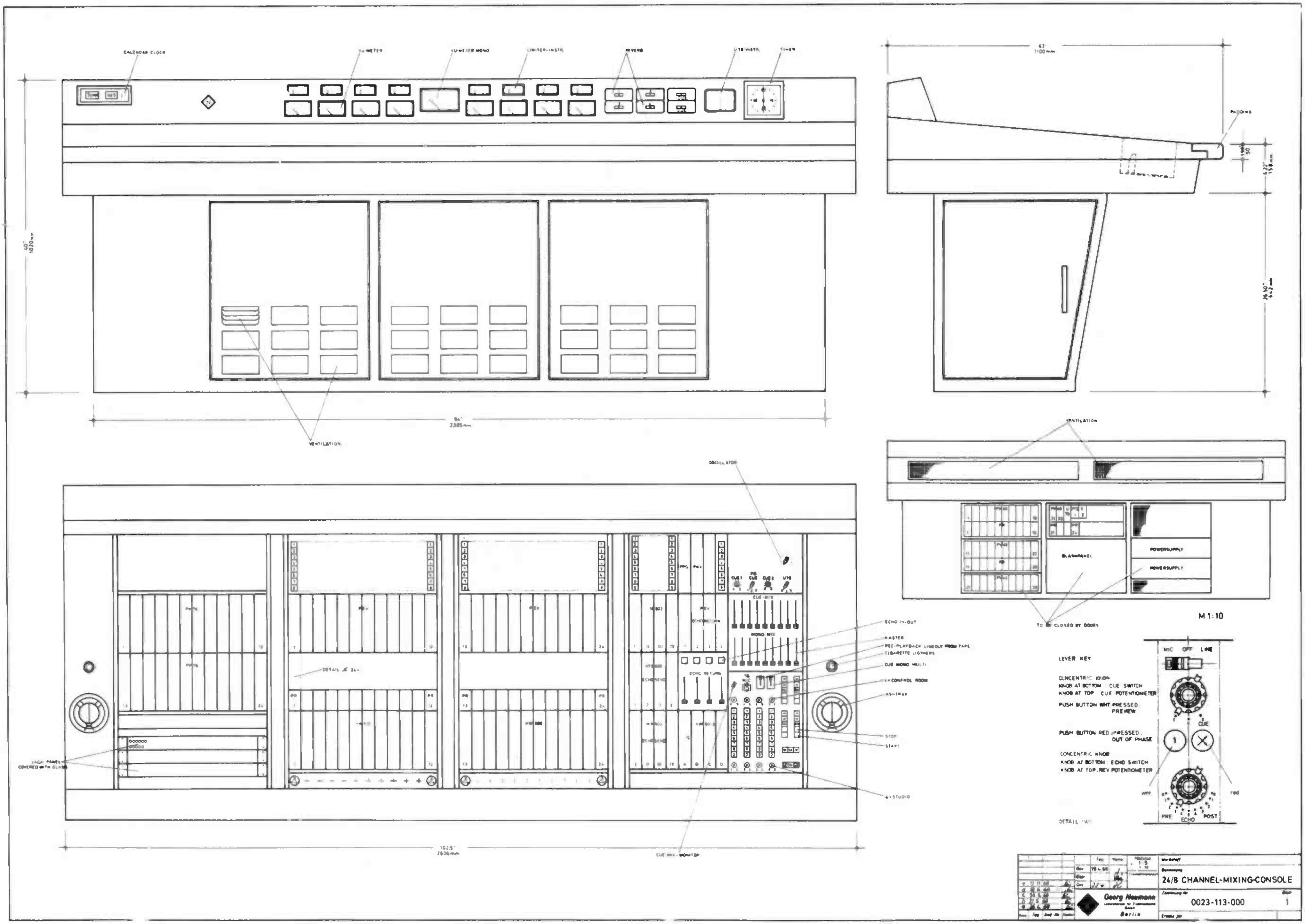
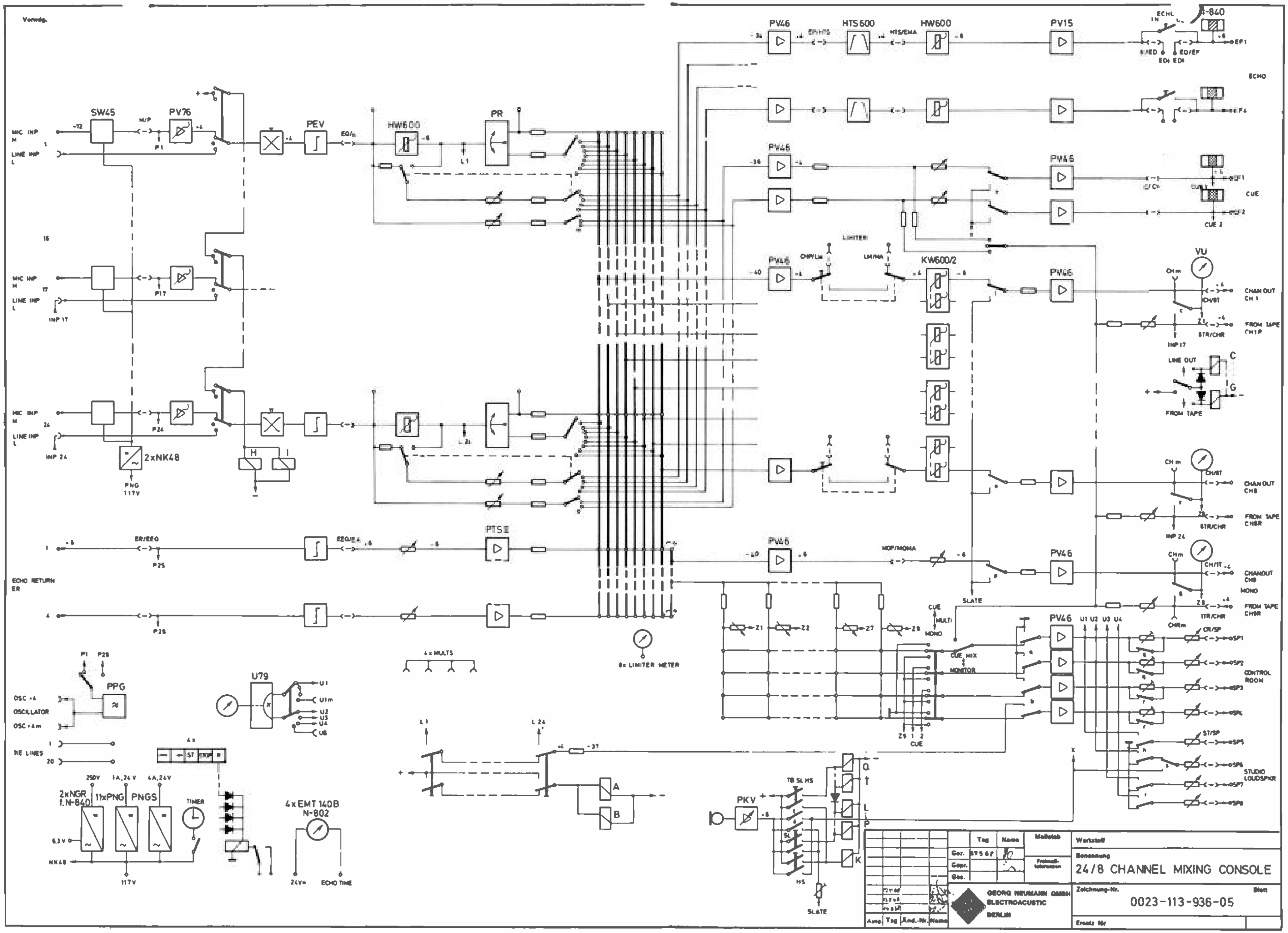

CALENDAR CLOCK
VU-METER
VU-METER MONO
LIMITER-INSTR.
REVERB
TIMER
VENTILATION
94"
2385 mm
40"
1020 mm
41"
1100 mm
PADDING
26.50"
642 mm
6.22"
158 mm
OSCILLATOR
ECHO IN-OUT
MASTER
REC-PLAYBACK LINEOUT FROM TAPE
CIGARETTE LIGHTERS
CUE MONO MULTI
CONTROL ROOM
STOP
START
STUDIO
JACK PANEL COVERED WITH GLASS
ECHO RETURN
CUE-MIX
MONO MIX
CUE 1
CUE 2
U 10
DETAIL
102.5"
2606 mm
POWERSUPPLY
BLANKPANEL
TO BE CLOSED BY DOORS
M 1:10
LEVER KEY
CONCENTRIC KNOB
KNOB AT BOTTOM : CUE SWITCH
KNOB AT TOP : CUE POTENTIOMETER
PUSH BUTTON WHT PRESSED: PREVIEW
PUSH BUTTON RED (PRESSED): OUT OF PHASE
CONCENTRIC KNOB
KNOB AT BOTTOM : ECHO SWITCH
KNOB AT TOP : REV POTENTIOMETER
MIC
OFF
LINE
CUE
PRE
ECHO
POST
red
1 : 5
1 : 10
24/8 CHANNEL-MIXING-CONSOLE
0023-113-000
1
Georg Neumann
Berlin



Verwdg.
SW45
PV76
PEV
HW600
PR
PV46
HTS600
HW600
PV15
KW600/2
PTS II
VU
MIC INP M
LINE INP L
INP 17
INP 24
2xNK48
PNG 117V
ECHO RETURN ER
ER/EEQ
EEQ/EA
CHAN OUT CH 1
FROM TAPE CH1P
LINE OUT
FROM TAPE
CHAN OUT CH8
FROM TAPE CH8R
CHAN OUT CH9
MONO
FROM TAPE CH9R
ECHO
CUE
CUE 2
LIMITER
MOP/MOMA
SLATE
CUE MIX
MONITOR
CONTROL ROOM
STUDIO LOUDSPKR
8x LIMITER METER
4x MULTS
PPG
OSC +4
OSCILLATOR
OSC +4m
TIE LINES
U79
2xNGR f.N-840
11xPNG
PNGS
TIMER
4xEMT 140B N-802
24V=
ECHO TIME
PKV
6,3 V
NK48
117V

Werkstoff
Benennung
24/8 CHANNEL MIXING CONSOLE
Zeichnung-Nr.
0023-113-936-05
Blatt
Ersatz für
GEORG NEUMANN GMBH ELECTROACUSTIC BERLIN
Tag
Name
Maßstab
Gez.
Gepr.
Ges.
Ausg.
Änd.-Nr.

NEUMANN-Mischpulte in Transistortechnik

NEUMANN Mixing Consoles in Transistor Technique

0023-910-01+02-01

4 Mikrophoneingänge
4 Hochpegeleingänge
1 Summenkanal

4 Microphone inputs
4 Line inputs
1 Line output

18 Mikrophoneingänge
4 Summenkanäle
4 Hallsummenkanäle

18 Microphone inputs
4 Line outputs
4 Echo busses

GEORG NEUMANN GMBH · ELECTROACUSTIC · 1 BERLIN 61 (WEST)

345 Okt. 66
Printed in Germany

10 Mikrophoneingänge
4 Gruppenkanäle
2 Summenkanäle
2 Hallsummenkanäle

10 Microphone inputs
4 Group channels
2 Line outputs
2 Echo busses

4 Mikrophoneingänge
1 Summenkanal
1 Hallsummenkanal

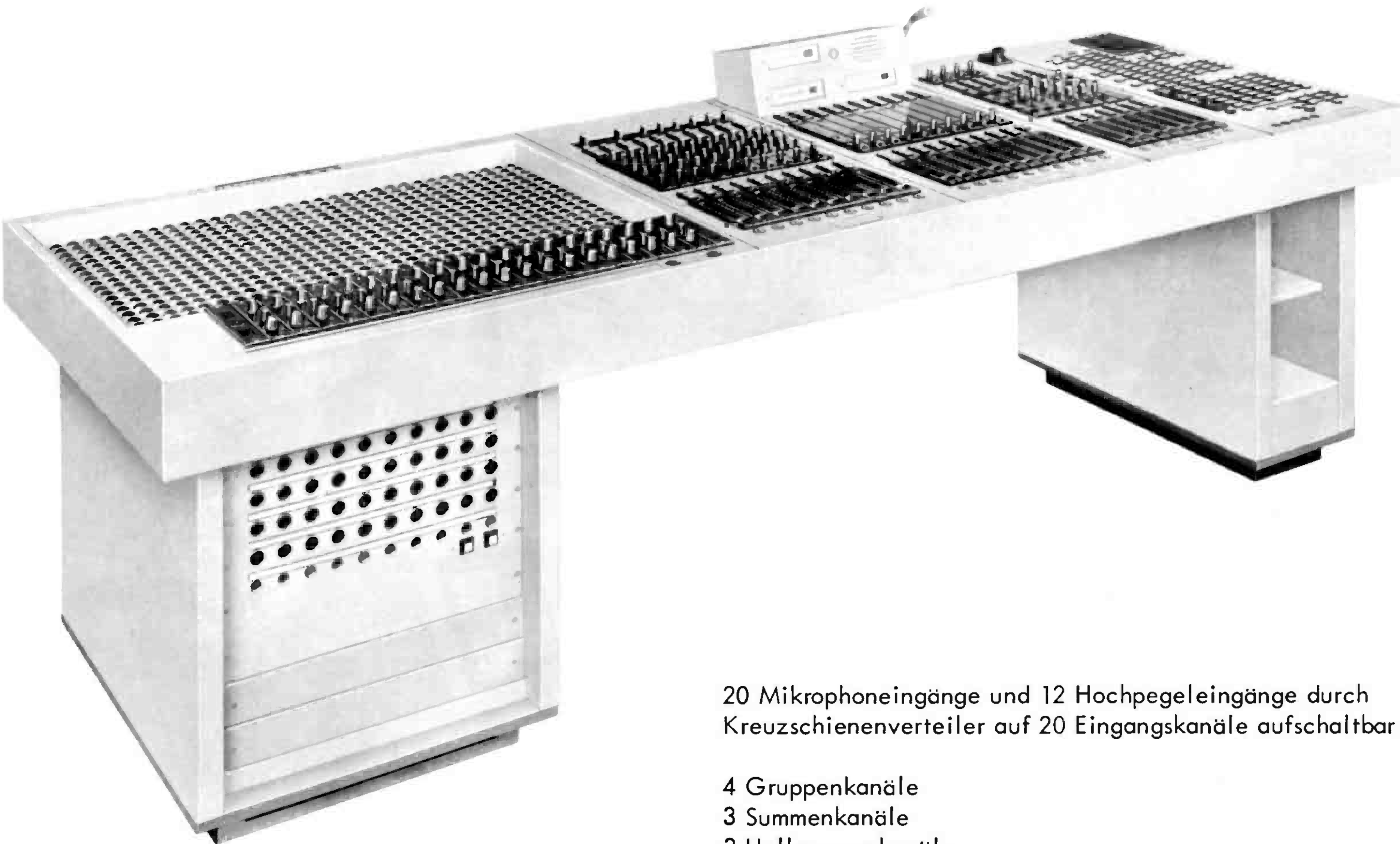
4 Microphone inputs
1 Line output
1 Echo bus

8 Mikrophoneingänge
4 Hochpegeleingänge
4 Gruppenkanäle
2 Leitungsausgänge
2 Hallsummenkanäle

8 Microphone inputs
4 Line inputs
4 Group channels
2 Line outputs
2 Echo busses

6 Mikrophoneingänge
2 Summenkanäle

6 Microphone inputs
2 Line outputs



20 Mikrophoneingänge und 12 Hochpegeleingänge durch Kreuzschienenverteiler auf 20 Eingangskanäle aufschaltbar.

4 Gruppenkanäle
3 Summenkanäle
3 Hallsummenkanäle

20 microphone inputs and 12 line inputs to be distributed through crossed bus bars to 20 input channels.

4 Group channels
3 Line outputs
3 Echo busses

NEUMANN-Mischpulte in Transistortechnik

NEUMANN Mixing Consoles in Transistor Technique

0023-910-01+02-02

0023 - 70

6 Mikrophon-Leitungseingänge
1 Summenkanal
1 Hallsummenkanal

6 Microphone line inputs
1 Line output
1 Echo bus

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24 Mikrophoneingänge
4 Summenkanäle
4 Hallsummenkanäle

24 Microphone inputs
4 Line outputs
4 Echo busses

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385 Juli 67

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0023 - 48

8 Mikrophoneingänge
1 Summenkanal

8 Microphone inputs
1 Line output

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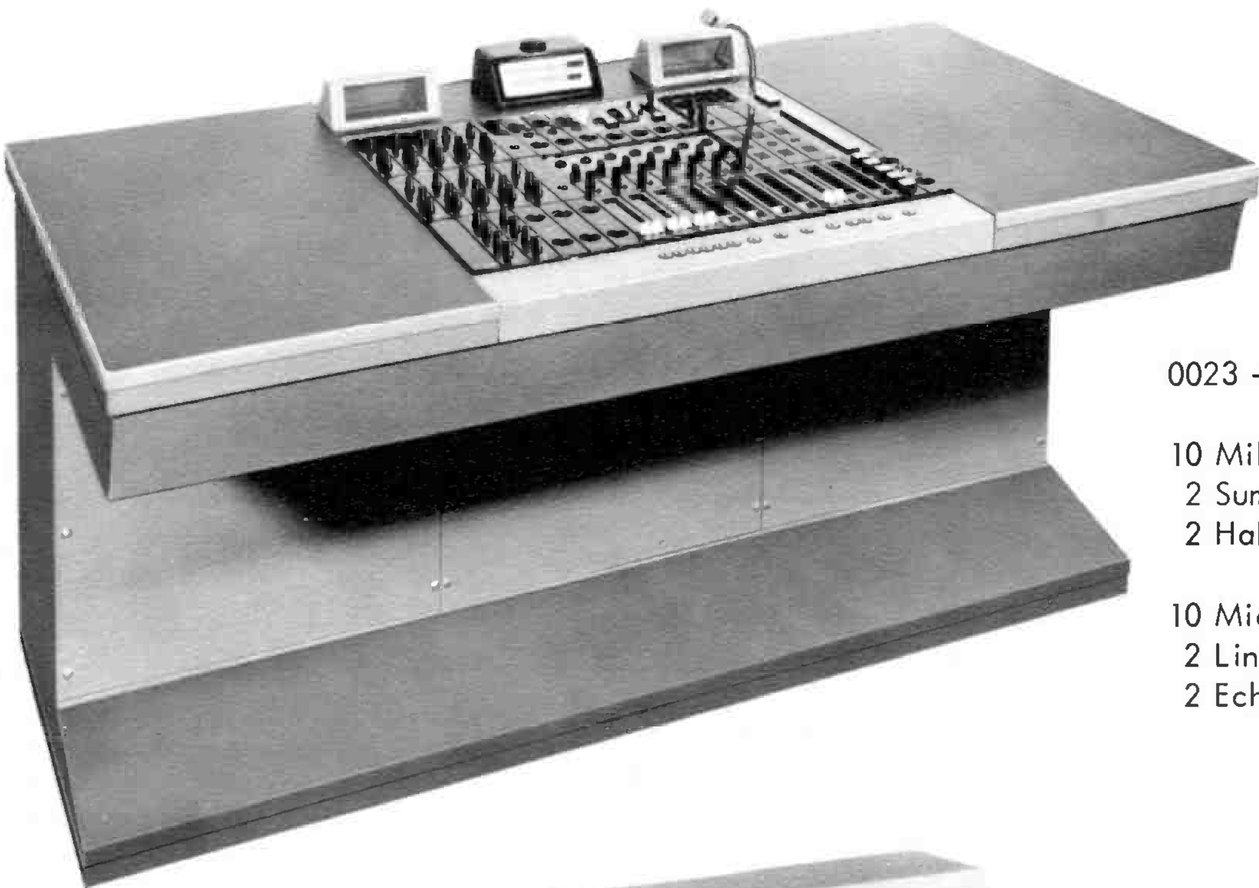
9 Mikrophoneingänge
1 Leitungseingang
4 Summenkanäle
1 Hallsummenkanal

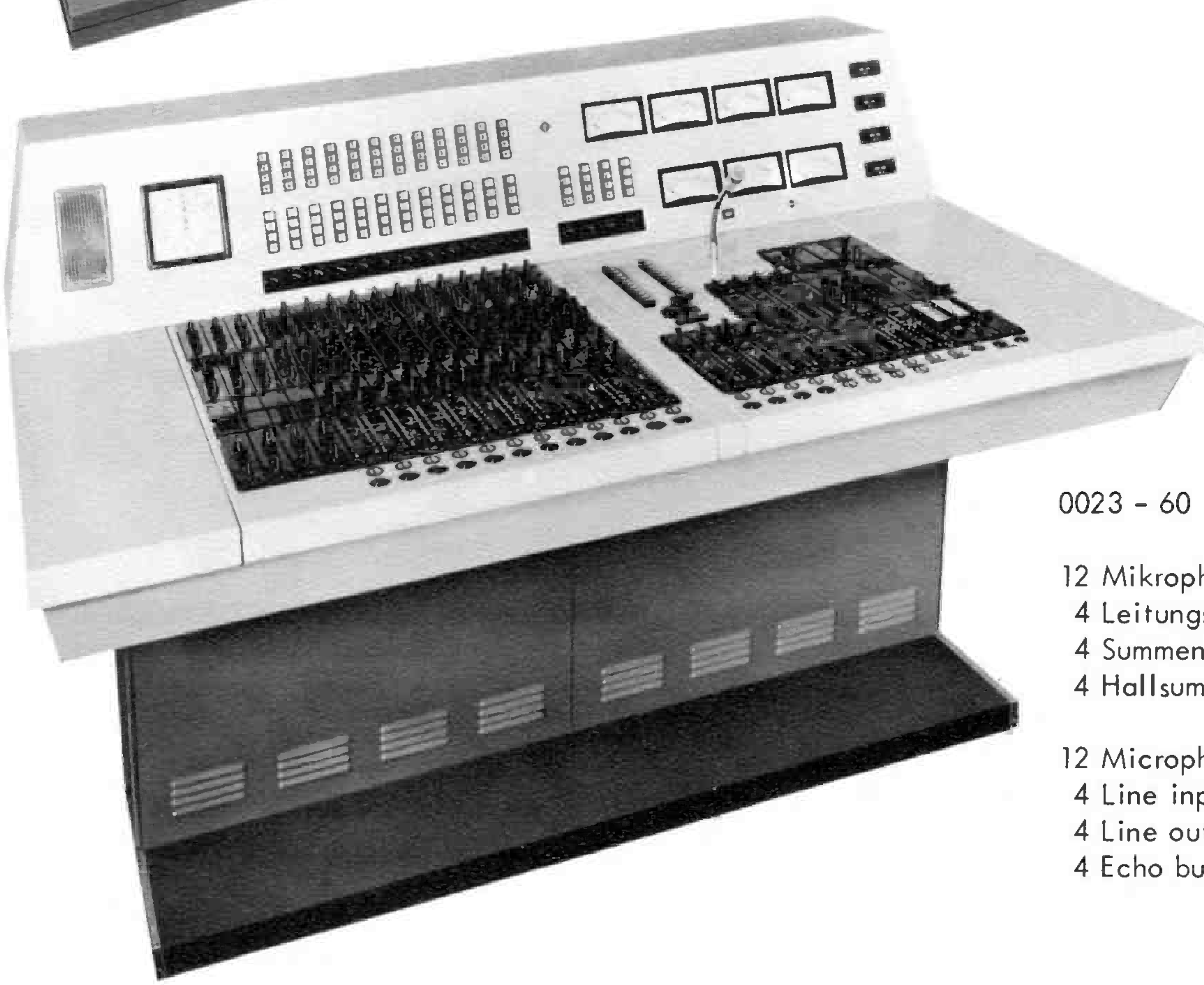
9 Microphone inputs
1 Line input
4 Line outputs
1 Echo bus



0023 - 50

10 Mikrophoneingänge
2 Summenkanäle
2 Hallsummenkanäle

10 Microphone inputs
2 Line outputs
2 Echo busses



0023 - 60

12 Mikrophoneingänge
4 Leitungseingänge
4 Summenkanäle
4 Hallsummenkanäle

12 Microphone inputs
4 Line inputs
4 Line outputs
4 Echo busses

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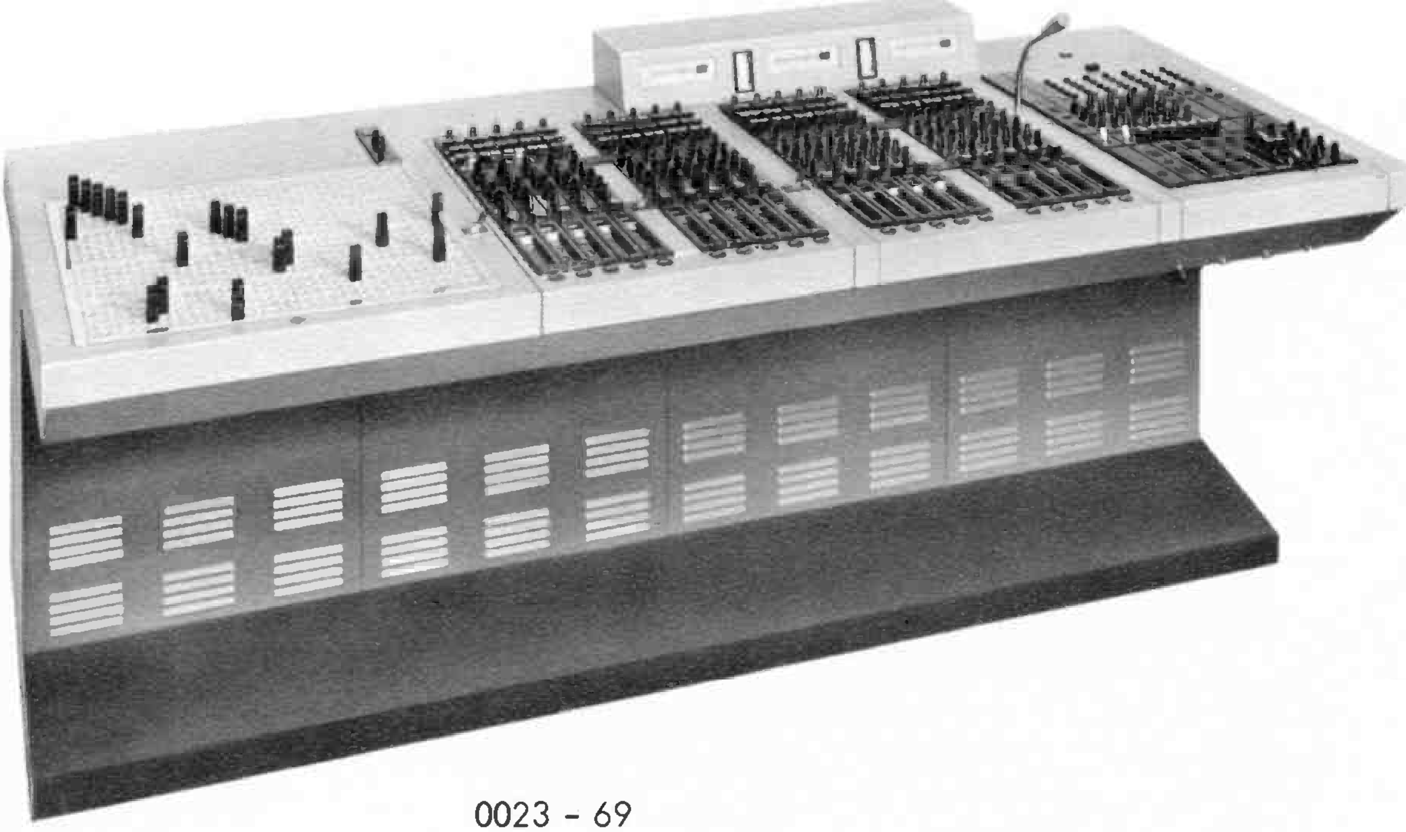
4 Mikrophoneingänge
1 Leitungseingang
2 Summenkanäle
1 Hallsummenkanal

4 Microphone inputs
1 Line input
2 Line outputs
1 Echo bus



0023 - 69

34/20 Leitungseingänge	34/20 Line inputs
4 Summenkanäle	4 Line outputs
2 Hallsummenkanäle	2 Echo busses

NEUMANN-Mischpulte in Transistortechnik
NEUMANN Mixing Consoles in Transistor Technique

0023-910-01+02-03

0023-71

18 Mikrophoneingänge
2 Leitungseingänge
3 Gruppenkanäle
2 Summenkanäle
2 Hallsummenkanäle

18 Microphone inputs
2 Line inputs
3 Group channels
2 Line outputs
2 Echo busses

0023-68

4 Mikrophoneingänge
1 Summenkanal

4 Microphone inputs
1 Line output

GEORG NEUMANN GMBH · ELECTROACUSTIC · 1 BERLIN 61 (WEST)

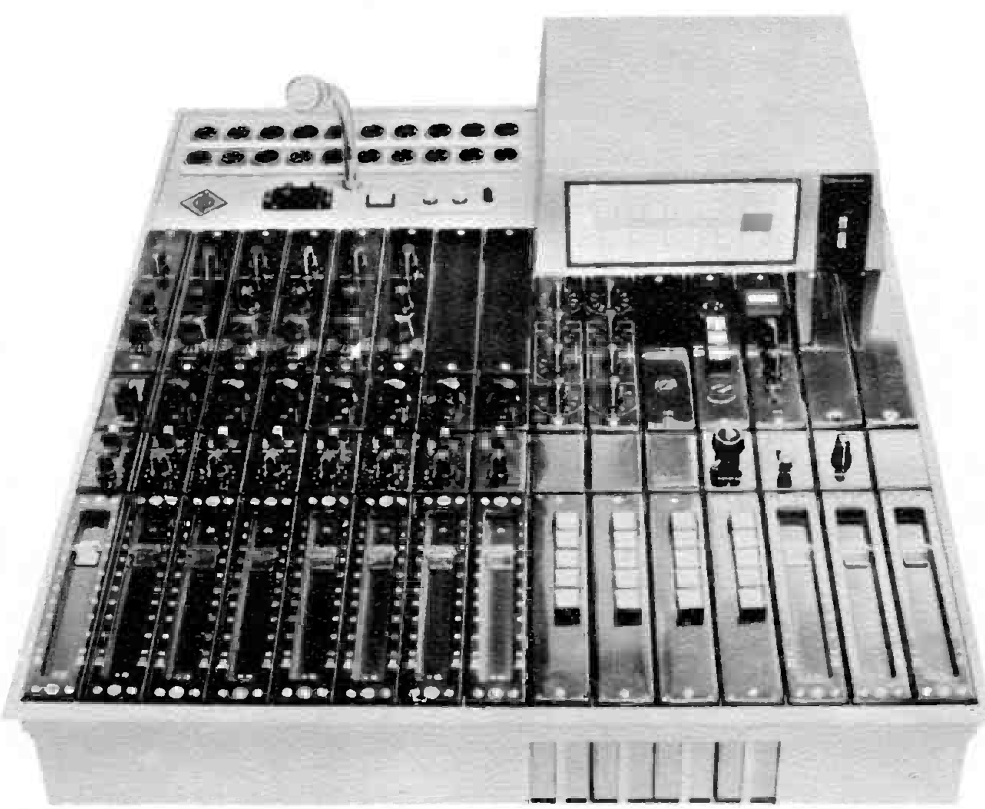
402 Jan.68

Printed in Germany

0023-80

7 Mikrophoneingänge
3 Leitungseingänge
1 Summenkanal
1 Hallsummenkanal

7 Microphone inputs
3 Line inputs
1 Line output
1 Echo bus



0023-81

6 Mikrophoneingänge
2 Leitungseingänge

2 Summenkanäle
1 Hallsummenkanal

6 Microphone inputs
2 Line inputs
2 Line outputs
1 Echo bus

0023-74

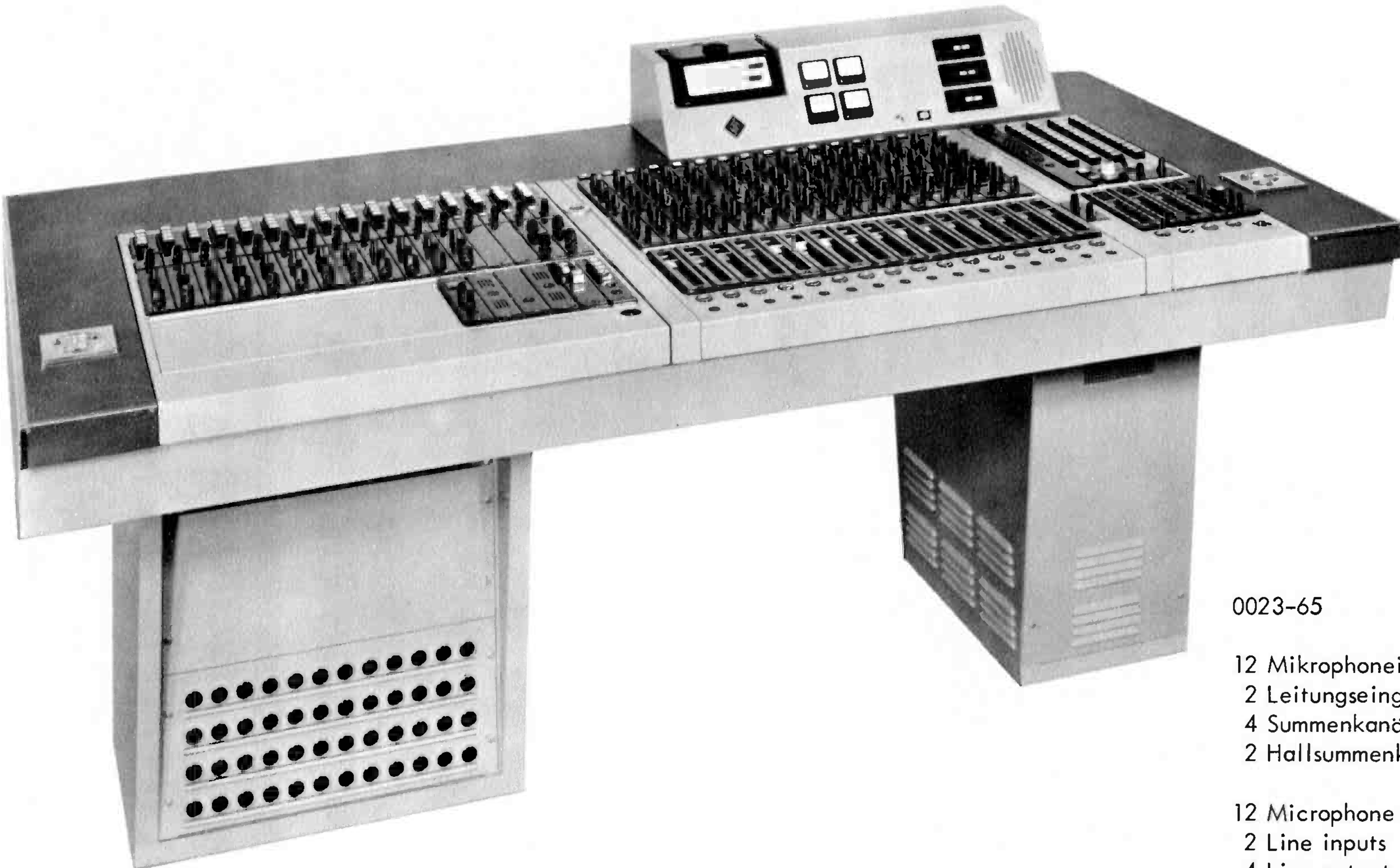
2 Mikrophoneingänge
16 Leitungseingänge
4 Summenkanäle
2 Echosummenkanäle

2 Microphone inputs
16 Line inputs
4 Line outputs
2 Echo busses

0023-72

7 Mikrophoneingänge
1 Summenkanal
2 Hallsummenkanäle

7 Microphone inputs
1 Line output
2 Echo busses



0023-65

12 Mikrophoneingänge
2 Leitungseingänge
4 Summenkanäle
2 Hallsummenkanäle

12 Microphone inputs
2 Line inputs
4 Line outputs
2 Echo busses

CONTROL CONSOLE SP 66 S MONO - STEREO

SP66S-910-02-00

Modern methods of cutting discs must fulfil a number of new and modern requirements. Commercial reasons demand that it should be possible to cut both STEREO and MONO discs from the same tape. In addition, STEREO discs must be compatible for use with MONO playback equipment. As a result of international agreements, many companies are forced to process tape recordings from countries with deviating standards. A guarantee must therefore be provided that when tape-to-disc transfer of such recordings is repeated, all variables can be reproduced with precision.

In recent years as a result of close cooperation with our customers a range of equipment has been developed which meets the above requirements. All this equipment together with a tape recorder incorporating a tape spacing arrangement for controlled cutting of grooves is conveniently grouped together in the SP 66 S control console. The console also contains all the equipment required for taking measurements and supervising the whole of the tape-to-disc transfer equipment.

GEORG NEUMANN GMBH · ELECTROACUSTIC · 1 BERLIN 61 (WEST)

411 Feb.68

Printed in Germany

Playback Amplifier TW 20

The voltage generated in the playback heads is amplified in the TW 20 playback amplifiers. Each playback amplifier is associated with two WE 66 playback equalizers for NAB and CCIR. The relevant card is cut in by operation of a press button. Operation of the tape speed switch (19/38 cm/s) causes related automatic switch-over of the playback equalization system. The PEV equalizer amplifier contains high-pass, low-pass and presence filters. These permit corrections to be made during the tape-to-disc transfer.

Rotary Attenuator RR 66

The RR 66 rotary attenuator is designed as a two-channel system. The pre-listening channel and the modulation channel are controlled together. Adjustments can be made in steps of 0.5 dB within a range of ± 10 dB referred to the nominal level. Further steps are -15 dB, -20 dB and -∞.

On occasions it is desirable to adjust the level of the pre-listening channel at variance to the level of the modulation channel. A second control in RR 66 caters for this eventuality. Referred to the level of the modulation channel, a variant deviation of from -2 dB to +1 dB for the pre-listening channel can be set in steps of 0.5 dB. From +1 dB to +6 dB the deviation can be set in steps of 1 dB. Settings where the levels of the pre-listening and modualtion channels deviate from one another are indicated by a red lamp in RR 66.

Compatibel Cutting Equalizer EE 66

An EE 66 compatibel cutting equalizer is included in both modulation channels and both pre-listening channels. Here the vertical component is derived from the information of the left-hand and right-hand channels and processed as a function of the frequency. The frequency response of the vertical component referred to 1000 Hz is -3 dB at 150 Hz. Beneath 150 Hz, the frequency response drops with 6 dB / octave. This equipment can be employed for cutting "compatible" stereo discs, i.e. discs which can also be played back with MONO pickups. Restriction of the vertical amplitude at low frequencies results in a lengthening of the maximum playing time due to space saving. The influence of the compatibel cutting equalizers can be switched off.

High-Low Pass Filter HT 66

HT 66 high-low pass filters are also incorporated for additional corrections. These too are designed as two-channel systems. Pre-listening and modulation channels are controlled together. The frequency response can be cut off with a gradient of 12 dB/octave below the transition frequencies 8, 11.2 and 15 kHz. The influence of the HT 66 systems can be switched off.

The outputs of the control console are formed by PV 46 line amplifiers.

Level Indicator

The level is indicated via a J 55 double light beam instrument with two PTMV peak volume indicator amplifiers. Two VU-meters are operated in parallel to this system via VU 66 VU-meter amplifiers. The technical limits of the level control system as well as the subjective impression of the volume of the disc being cut can be supervised in this manner. A sensitivity switch-over system is provided for matching the instruments to the particular nominal level.

Correlation Coefficient Meter U 79

The U 79 correlation coefficient meter gives a reading of the degree of correlation of the phase state of the two stereo channels.

Test Oscillator

A PPG test oscillator is incorporated for testing purposes. A test voltage can be fed in at all inputs via cut off jacks.

Monitor Outputs

Two variable 0-dB-outputs are available for monitoring tests. These can be employed for monitoring a series of test points and the outputs of the control console as well as the outputs of the playback amplifiers of the connected amplifier equipment for the cutterheads. Two AV 66 loudspeaker amplifiers can be incorporated in the control console on request.

All amplifiers and instruments contained in the control console are equipped with silicon transistors. They are designed either as card amplifiers on printed circuits or when equipped with controls as mixing console modules.

The following equipment and instruments are contained in the control panel:

4 PEV Equalizer Amplifiers
2 RR 66 Rotary Attenuators (two-channel)
2 HT 66 High-Low Pass Filters (two-channel)
2 PTMV Peak Volume Indicator Amplifiers
1 J 55 Double Light Beam Instrument
2 VU-meters
1 Correlation Coefficient Meter
1 Press Button Assembly for operational switching
1 Press Button Assembly for switching test and monitoring systems
Switch for test circuit sensitivity
AR 66 Monitor Attenuator
1 Telefunken Magnetophon M 10A with special tape spacing arrangement

The control console in its standard form is equipped for playback of MONO and STEREO tapes. The "Magnetophon" is therefore equipped with special interchangeable head assemblies for MONO and STEREO.

The power supply is accomodated on the right hand side of the console frame. In the standard design, two component-assembly supports with the following plug-in units are situated on the left hand side:

4 TW 20 Playback Amplifiers
8 WE 66 Playback Equalizers (NAB/CCIR) with 19/38 cm/s change-over arrangements
4 PV 46 Line Amplifiers
2 EE 66 Compatible Cutting Equalizers
2 VU 66 VU-meter Amplifiers
1 U 79t-100 Correlation Coefficient Meter
1 UE 66 Switching Spark Suppressor

If the "Magnetophon" incorporated in the SP 66 S control console is also intended for recording purposes, a further component-assembly support can be accommodated in the console frame to take the plug-in units required for recording. The following amplifiers are required for producing STEREO recordings:

2 TA 20 Record Amplifiers
1 TL 20 Erase Oscillator

In addition the special head assembly must be replaced by a standard head assembly.

The SP 66 M MONO control console is fully designed for STEREO purposes but is only equipped for MONO usage. The following instruments, required for STEREO operation are absent:

2 TW 20 Playback Amplifiers
4 WE 66 Playback Equalizers
2 PEV Equalizer Amplifiers
1 MR 66 Rotary Attenuator
2 EE 66 Compatibel Cutting Equalizers
1 HT 66 High-Low Pass Filter
2 PV 46 Line Amplifiers
1 U 79 Correlation Coefficient Meter
1 VU 66 VU-meter Amplifier
1 VU-meter
1 PTMV Peak Volume Indicator Amplifier

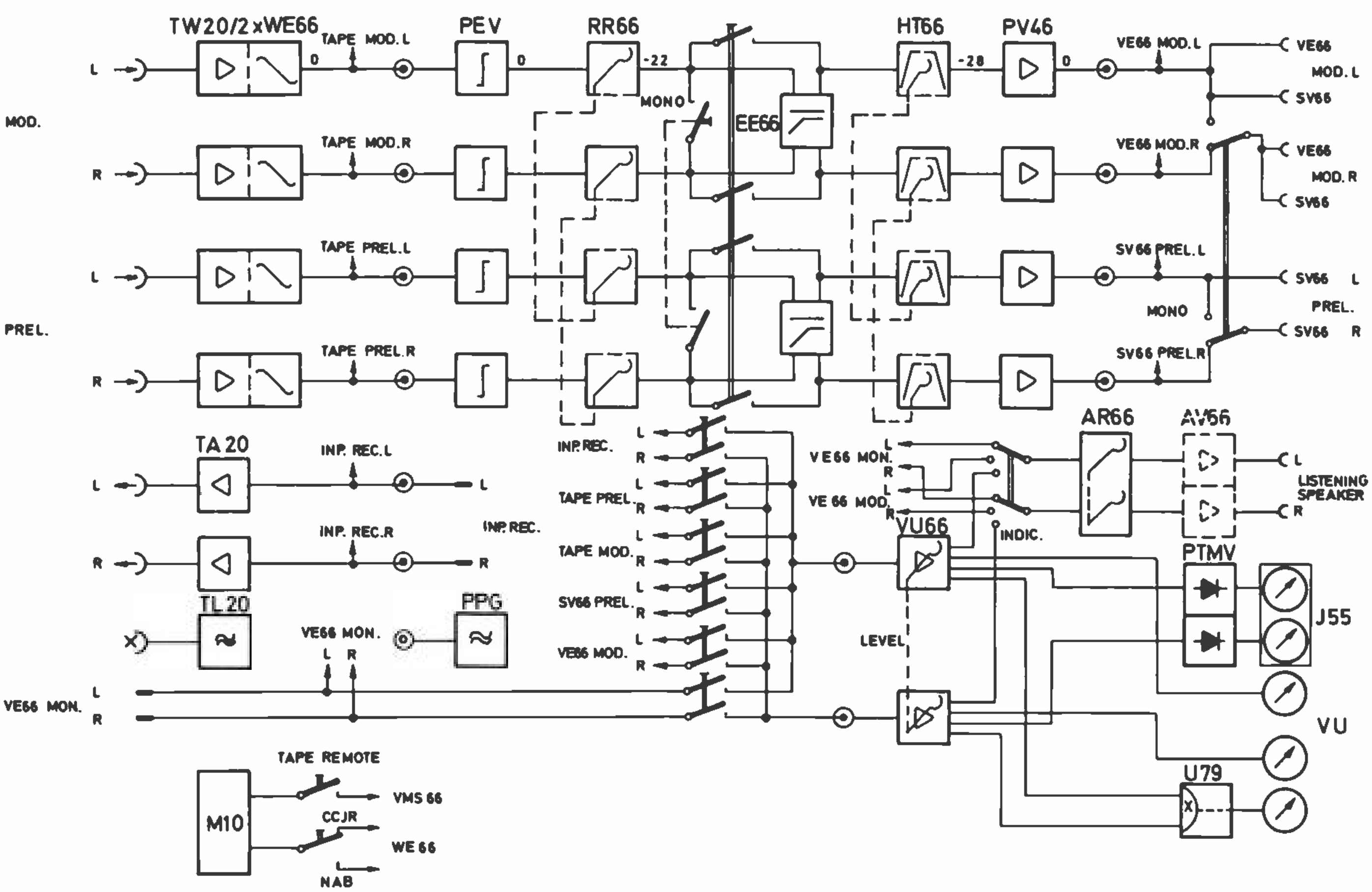
The J 55 double light beam instrument is also employed in the MONO control console to avoid unnecessary extra cost on conversion of the equipment to STEREO. The "Magnetophon" is equipped with a special MONO head assembly. The record amplifier and erase oscillator can be incorporated on request. In addition the special head assembly must be replaced by a standard head assembly.

The tape spacing arrangement is adjustable. The tape length between the heads is therefore variable. The following table provides information on the relationship between the turntable speed, the tape speed and the tape length between the heads:

	33 1/3 rpm	45 rpm
19 cm/s	21 cm	16 cm
38 cm/s	42 cm	32 cm

Technical Data

Tape speed	38.1 cm/s and 19.05 cm/s
Tape width	6.25 mm
Head assembly	Special head assembly, MONO or STEREO for variable groove pitch control
Equalization	Switch-over facilities for two different tape flux standards CCIR - NAB
Output	0 dB ≙ 775 mV
Maximum output level	+16 dB
Signal-to-noise ratio	
unweighted	≧ 65 dB
weighted	≧ 70 dB (DIN 45 405)
Input for recording	balanced floating
Input resistance	≧ 2000 Ω
Input level	+ 6 dB
Remote control facilities	Start, Stop
Output level of complete system	0 dB across 600 Ω
Maximum output level	+ 21 dB
Output level (monitoring)	0 dB across approx. 2.7 kΩ
Power supply	Variable 110 ... 240 V, 50/60 Hz approx. 400 W
Dimensions	1,410 mm wide, 600 mm deep, 1,150 mm high
Weight	Approx. 300 kg



DISK CUTTING LATHE AM-131

for those

who require the ultimate . . .

Front view of the compact AM-131 Disk Recording Lathe with "Spencer" 50-power Microscope

GOTHAM AUDIO CORPORATION

2 WEST 46 STREET, NEW YORK 36, N.Y. · · · COLUMBUS 5-4111

The NEUMANN AM-131 DISK CUTTING LATHE meets the recording industry's demands for a medium priced, high quality lathe for cutting of instantaneous acetates and masters for the production of phonograph records and transcriptions. Its technical specifications are in every respect equal to the famous AM-32b Stereo Disk Mastering Lathe. The economies are effected solely through elimination of automation features, while the principal operating components such as the drive motor, fluid turntable coupling, vacuum chuck turntable, gliding carriage, etc. are identical to the full-automatic lathe.

The design of the AM-131 is based on three decades of experience. It is executed by West Germany's finest machinists, and it is installed, serviced and guaranteed by Gotham Audio Corporation, New York and Hollywood.

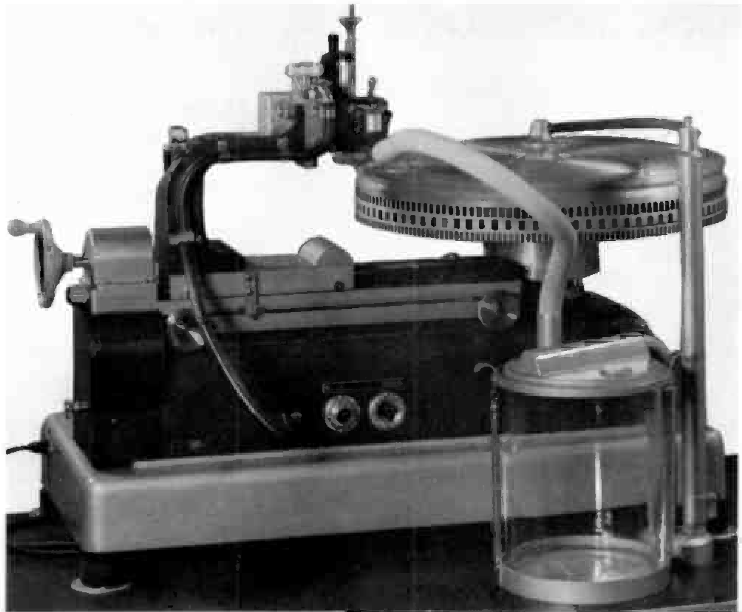
NEUMANN AM-131 Disk Recording Lathe Features which produce lacquers of unsurpassed quality:

1. Three-speed *synchronous* motor drive *without belts, gears, or chains:* 16-2/3 rpm electronic converter available as accessory.
2. The 65 lb. turntable flywheel smoothes out all wow and flutter. Oil coupled to the drive shaft; no solid connection between motor and turntable.
3. The 16" vacuum hold-down turntable has switchable air valve; prevents hissing through unused holes when using small acetate blanks; quiet operation.
4. Step-gear pitch change and interchangeable gear trains provide pitch from 84 to 447 lines per inch.
5. Cutterhead suspension complete with dash-pot with variable damping adjustment.
6. Automatic cutter lift at correct pre-determined inner end-groove diameter adjustable for the three RIAA end diameters.
7. Automatic cutter lift when lead screw is disengaged. Positively prevents destruction of stylus due to cutting into aluminum.
8. Heated stylus metering on lathe unit; calibrated in amperes.
9. DC Heated stylus heating built in and automatically switched when cutter lifts.
10. Illuminated microscope moves with cutterhead transport giving "standing still" image of grooves.
11. Lead-in and lead-out grooves may de deepened easily.
12. Full vacuum collection and tubing provided. Chip jar; vacuum distribution valve; chip removal pipe; vacuum hold-down connection; all tubing; all you do is connect your vacuum pump.
13. New standard American "Spencer" Microscope (delivered with *all* U.S. manufactured lathes) with 50-power magnification and calibrated reticle. Mounted on heavy-duty base and arm; swings over turntable; bright illuminating attachment.

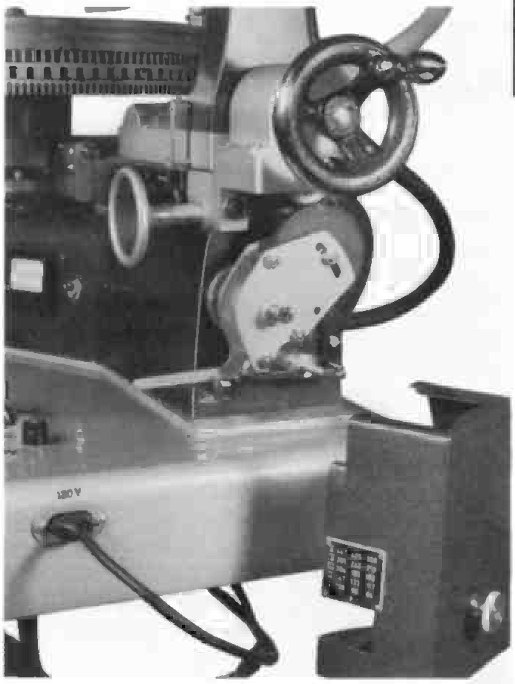
All NEUMANN Lathes are unconditionally guaranteed by Gotham Audio Corporation for 1 year from date of delivery.

A PARTIAL LIST OF USERS OF THE NEUMANN AM-131 LATHE:

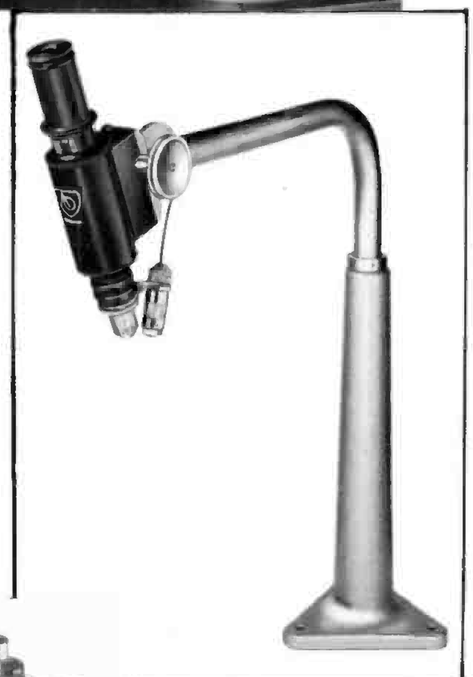
Dick Charles Recording Service Inc.
New York, New York (2 units)
Quality Recording Service
Stockton, California
A & R Recording, Inc.
New York, New York
Pepper Records, Inc.
Memphis, Tennessee
Recordings, Inc.
Baltimore, Maryland
Garrison Recording Studios
Long Beach, California
Mico Record Mfg. Co.
Manila, Philippines
Federal Record Mfg. Co.
Kingston, Jamaica
Regent Sound Studios Inc.
New York, New York
Gaity Recording Studio
Minneapolis, Minnesota
Tom H. Jones Recording
Rochester, Minnesota
Audio Recorders of Arizona
Phoenix, Arizona
Cleveland Recording Company
Cleveland, Ohio
Whitney Recording Studio
Glendale, California
Radio Station WEMP
Milwaukee, Wisconsin
John P. White Recording Co.
Baton Rouge, Louisiana
Sound Makers
New York, New York
Liberty Custom Recorders
Hollywood, California
Bud Pressner Music & Recording
Service, Gary, Indiana
Sunset Sound Recorders
Hollywood, California
Neophon Recording Corp.
Hollywood, California
A.M.S. Sound Recording Inc.
Philadelphia, Pennsylvania
Edgewood, Inc.
Washington, D.C.
Cadet Records, Inc.
Culver City, California
Findlay Recording Company
Findlay, Ohio



Back view of the AM-131
— illustrating all-new
vacuum collection.



Side Section (with cover removed)
—illustrating Pitch Change Gear Trains.



American Optical "Spencer"
50-power Microscope

LYREC SM 8/3–A
3-speed SYNCHRONOUS MOTOR

Disassembled LYREC SM 8/3–A
3-speed SYNCHRONOUS MOTOR

TECHNICAL SPECIFICATIONS:

Turntable speeds:	33⅓, 45, and 78 rpm synchronous speeds from LYREC SM-8/3A three-speed motor. Electrical half speed converter permits 16⅔ rpm cutting.
Turntable diameter:	16"; accommodates 17¼" master blanks.
Wow and Flutter content:	Less than ± 0.035% RMS.
Rumble content (all frequencies from 0-250 cps):	More than 55 dB down.
Pitch available:	84 to 447 LPI in fifteen equally spaced steps by means of five changeable gear trains.
Cutterhead suspension:	Semi-automatic type complete with dash-pot, automatic lift, groove deepening arrangement, connections for cutter drive, feedback, and DC stylus heat terminals.
Cutterhead mounting:	Universal for NEUMANN MS-52H, ES-59, SX-45 Stereo, Grampian, Fairchild, Presto, Westrex, etc.
Lead-in and lead-out grooves:	Standard RIAA dimensions; carriage advanced by handwheel; grooves may be deepened as desired.
Automatic features:	Cutter lift at concentric circle; stylus heat cutoff.
Microscope:	American Optical "Spencer" 50-power; calibrated in mils; illuminated; mounted on swivel arm attached to lathe table.
Dimensions of lathe:	14" deep x 28" wide x 23" overall height.
Dimensions of motor:	11" x 11" x 12" high.
Weight of lathe unit alone:	244 lbs.
Weight of turntable:	65 lbs.
Weight of LYREC motor:	110 lbs.
Power requirements:	117 volts 60 cycles; 110 watts total. (220 volts and 50 cps available for export).
Delivery information:	Shipped in three wooden crates. Complete with: Vacuum chip collecting jar, vacuum turntable attachment, and rubber hoses; motor connecting rod, two types of oil and oiling can, Pitch gear No. II, complete instruction manual and check-out report.

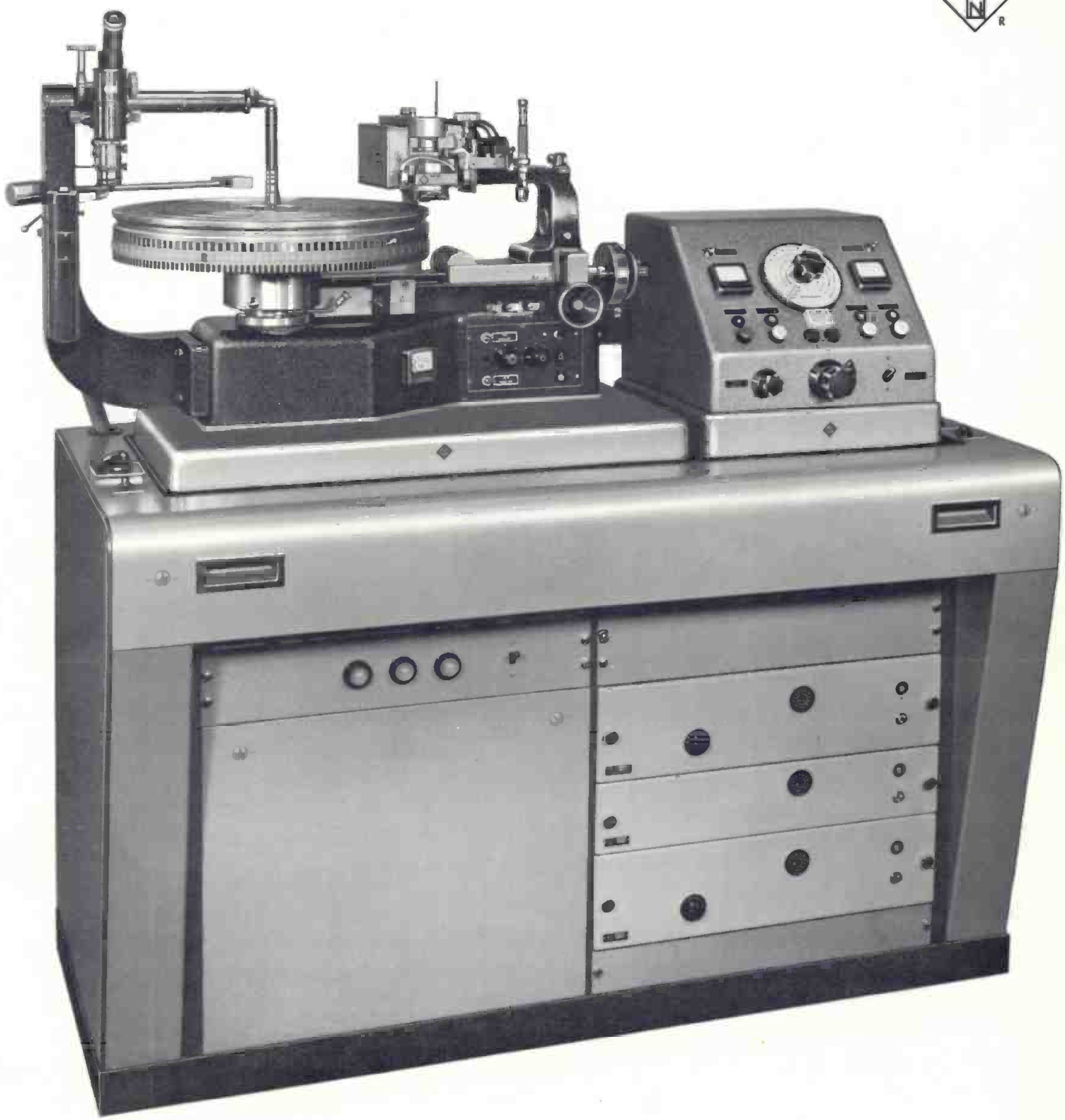
OPTIONAL EQUIPMENT:

Gear Trains	I, III, IV, V.
Half-speed converter for 16⅔ rpm:	Type ZA-33
Hold-down knob for small acetates:	Type ZA-22
Table stand with pickup arm (16") for Std. or LP records:	Type ZA-25
Buchmann-Meyer pattern observation lamp and transformer:	Type ZA-21
Disk Cutting Systems:	GOTHAM-GRAMPIAN Monophonic magnetic. NEUMANN ES-59 Dynamic Feedback System (MONO). NEUMANN SX-45 Dynamic Feedback System (STEREO).
Magnetic Disk Recording Cutter:	NEUMANN MS-52H—may be driven from any 30 watt amplifier.

For further information, contact Gotham Audio Corporation or your Neumann dealer:

PRINTED IN U.S.A.

NEUMANN MASTER DISK RECORDING LATHE AM-32b



GOTHAM AUDIO CORPORATION

2 WEST 46 STREET, NEW YORK 36, N.Y. · · · COLUMBUS 5-4111

It has been some time since the design of an entirely new disk lathe has come to the fore. Now with stereo disk placing even greater demands on the manufacturer — demands for quality and quantity — it is time to re-evaluate the entire line of equipment used in the production and reproduction of stereophonic records. One important link to date has been ignored: the disk cutting lathe itself. Reason for placing renewed emphasis on the lathe is the advent (or should we say "re-advent") of vertical compliance in cartridges, which demands of the disk even more rumble-free performance than heretofore. Secondly, the complex excursions of the groove have made it necessary to reduce the amount of time possible on a disk, making an investigation into the possibilities of *disk space conservation vital*. Furthermore, since the hazards of Stereo are considerably more than double those of monophonic recording, further automation of the actual cutting process has become essential.

A. MAIN LATHE ASSEMBLY — AM-32b:

The basic lathe bed is built up on a heavy cast iron base, standing on four rubber shock mountings. The turntable itself is of heavy cast iron, weighing 65 lbs. Its edge is cast with three stroboscopic rings—one for each speed—which are illuminated by a small neon spotlight bulb for easy observation. The turntable is completely isolated from the drive below by means of an oil-filled coupling, preventing rumble and flutter from being transmitted from the drive. The turntable is driven solely by a film of oil between two walls of two concentric cylinders. The lathe bed is of the slide type with two ball bearings riding the top of the bed to relieve the strain placed on the sled by the weight of the cutter suspension and cutterhead.

Directly beneath the lathe bed is a calibrated diameter scale on which are mounted the starting cams and end groove stop. Three cams, for 7", 10", and 12" disks are provided, the one to be used being raised into operating position. The end groove stop is adjustable to the three standard RIAA end groove diameters, and causes the cutterhead either to lift immediately (for eccentric grooves), or with an adjustable delay to provide a locked groove (now RIAA standard for LP as well). The lead-screw engaging lever is interlocked in such a way that the cutter will lift at any time that it is not being driven by the lead-screw, making stylus catastrophies impossible The newly installed ZA-19 braking assembly prevents coasting of the lead-screw after the end stop has been reached. The result is a perfect, noise-free concentric end groove.

1) VACUUM CHUCK TURNTABLE — ZA-3:

This unit fits on top of the regular cast iron turntable of the AM-32b and provides vacuum hold-down for all size blanks from 10" to 17¼". A disabling valve provides for vacuum shut-off to those holes not covered by a particular size acetate, thus preventing air escape and hiss. The Vacuum Chuch Turntable is machined in such a way as to prevent the usually heavier coated edge of acetates from causing surface "dishing". Acetate rejection rate diminishes impressively as usually rejected blanks become perfectly leveled by the vacuum.

2) CUTTERHEAD SUSPENSION — SA-32:

All of the cutterhead connections and functions are accomplished by a rectangular device mounted on the lathe's transport sled. The cutter is plugged into this unit by means of a small metal plug attached to it, permitting exchange of cutters from monophonic to stereo without misalignment. The connections are made by means of twa 6-prong plugs an the suspensian. These pravide the twa drive signals, twa feedback signals, and DC stylus heat. Both the heating current and the twa input signals are switched off when the cutter is raised. A release salenoid lifts the cutterhead whenever the "stop" button is depressed, when the sled hits the end graave stap, or when the lead-screw is disengaged.

A dash-pat at the front af the suspension is equipped with a perforated piston. An adjustable shield aver these perfarations allows a wide range of damping effects without changing viscosity of fluid.

The wiring from the cutter is led concentrically through the pivot ball bearings to prevent any stiffness ar resistance to free motion. The tilting mechanism to which the cutter is mounted is connected to a large moving coil system which, together with the depth-of-cut control provides electronic depth variation.

The SA-32 Cutterhead Suspension permits the mounting of either the Westrex or the Fairchild feedback systems without necessity for advance ball, and provides perfect electronic depth control with the cutterhead floating. Resultant reduction in rumble, acetate rejection due to smear, and surface space economies are impressive.

3) DEPTH-OF-CUT CONTROL PANEL—TE-32b:

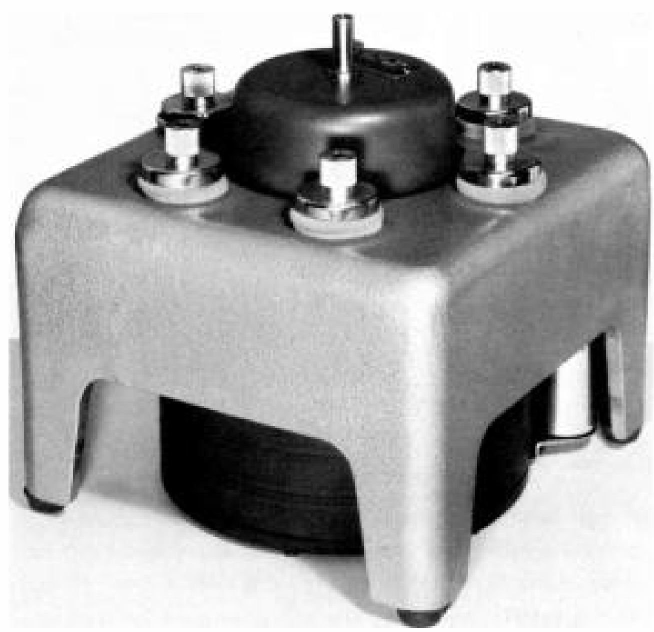
Located in the base of the lathe bed is the electronic depth-of-cut control panel. It supplies a DC current to the moving coil system in the cutter suspension which relieves cutterhead pressure on the disk. A simple potentiometer provides full range of cutting depth, which can be observed in the microscope while adjustment is made. A second, similar control pre-sets the increased depth generally used in lead in, lead out, and spiraling grooves. A test button allows for proper adjustment prior to commencement of cutting operation. Two toggle switches permit automatic deepening of the grooves during spiraling, lead in or lead out, or disable it far any one of these functions. Increased depth operates with a time delay to prevent cross-cutting.

B. PITCH CONTROL UNIT — VA-32 f:

The pitch control and general control consolette is a separate piece of equipment, and is situated next to the lathe on its right. NO power or drive is taken from the turntable motor for any drive of the lead-screw; the pitch drive is entirely self-cantained, and is coupled to the lead-screw by a four-way shock isolated coupling. The console mounts the following controls: motor on-and-off; motor speed control selector; heated stylus control (DC); — stop; start; lead in; spiraling; lead aut buttons and tally lights; heated stylus ammeter; microscape light switch; cutterhead pick up delay switch far cancentric end graove; main pitch selection dial; and pitch cantrol current meter. The depth of cut contral current meter is mounted on the base of the lathe itself.

The pitch matar is cannected by means af a belt and through an oil filled gear train and the flexible caupling to the lead-screw. A capper disk on the shaft af this mator runs aver the si pales of an electramagnet, in which a DC current produces a braking action which stabilizes the pitch motor's RPM. A secend, identical motor is likewise belt connected to an overdrive in the gear train, and serves for speed-up of pitch for lead-in, spiraling, and lead-out. It likewise has a stabilizing brake by means of which the lead in, spiraling and lead out pitch can be separately and accurately adjusted at each speed. The braking current on the pitch motor is supplied from an internal rectifier. The main pitch selector control which is calibrated from 54 to 475 lines/inch, is a powerstat controlling the AC voltage fed to the pitch drive motor. Complete remote start and stop facilities for a tape machine are also provided in this unit.

C. DRIVE MOTOR UNIT — LYREC SM 8/3-A



This motor, manufactured by LYREC of Copenhagen, Denmark, is unique in its field. It is of the synchronous type with motor RPM equal to disk RPM. A gear-like wheel, about

10 inches in diameter, rotates inside a similar inside gear in which, by means of a winding, a rotating magnetc field is set up. The motor must be started (for which a starting motor is mounted atop the unit) and requires a flywheel (the turntable) to continue "hunting" from one gear-tooth to the next. For each speed, 33, 45, and 78, there are two such gears and they are all arranged coaxially on the drive shaft. Each speed therefore has its own motor. Wow and flutter of the lathe is below ± 0.035 RMS total, at ALL speeds. The motor is connected to the turntable by a connecting rod with two simple rubber coupling disks. Alignment of the connecting rod is not critical. For those interested in the reverse cutting of disks, incidentally, the motor can rotate in either direction in synchronism, depending on the direction in which it is initially started.

Disassembled LYREC SM-8/3 A 3-speed SYNCHRONOUS MOTOR

D. AUTOMATIC PITCH CONTROL AMPLIFIER — SV - 32 SR:

In the ZT-32S lathe console cabinet there is provided space for the mounting of either the SV-32SR Pitch Control Amplifier alone, or all three control amplifiers necessary for complete stereo pitch-depth control. All wiring necessary for maximum stereo control is prepared in all such console cabinets. All components are plug-in for easy installation of future aquisitions.

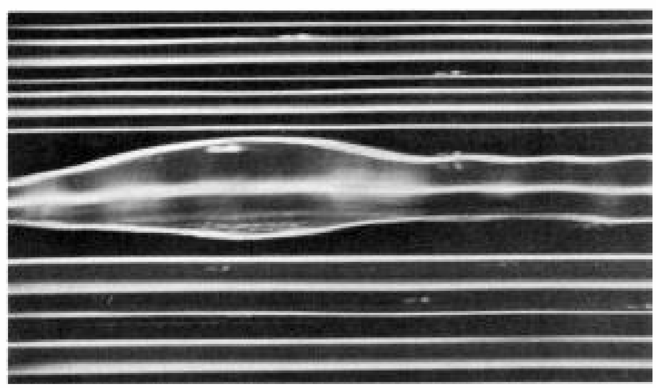
A preview head assembly is required to be mounted on the playback tape machine so as to provide these control amplifiers with prior knowledge of the modulation to be fed to the cutting system. A fulltrack preview head serves for mono recording while a stereo preview head is necessary for stereo control. The outputs of the playback amplifiers associated with these preview heads are fed to the inputs of the control amplifiers.

In monophonic recording the depth of cut is kept constant while the pitch is varied as a function of the preview information. In stereophonic recording the pitch control is actuated by the SUM of the left and right channel signal, while depth-of-cut is varied according to the DIFFERENCE signal obtained from the stereo preview head.

It is the function of the Pitch Control Amplifier to translate the previewed signal through a specially designed equalizing curve into variations in braking current which, in turn, are applied to the pitch control motor (see section B) to vary its speed and with it the lines per inch. Since control is electronic it is instantaneous and not subject to delays caused by mechanically acting controls. Relays operated from the drive motor speed switch automatically adjust the recovery time of the braking current to the proper speed of revolution used.

E. AUTOMATIC DEPTH CONTROL AMPLIFIER — SV - 32 VR:

The Depth Control Amplifier SV-32VR is in every detail identical to the pitch control unit. Its output, however, is fed to the solenoid in the cutter suspension, and produces there the varying relief of cutter pressure, acting against the weight and counterbalancing spring of the cutter.



Micro Photo of Groove showing action of Both Pitch and Depth Controls

F. INTEGRATION AMPLIFIER UNIT — SV - 32 T:

In lateral and vertical modulation, increased depth requires increased pitch, so that any deepening of the groove allowed by the depth control amplifier, must be translated in turn into increased pitch. The SV-32T integration amplifier does just that, by adding to the pitch control current whenever increased depth is called for. It has been shown over the past years that this form of independent pitch and depth control can add up to 6 minutes to the playing time of one side of a 12" LP recording. Of course, as is always the case, recordings of a high dynamic range benefit most from such controls, while highly limited or compressed material produces but slight space saving.

Each of the control amplifiers is equipped with a plug-in equalizer for RIAA equalization, which translates the RIAA pre-emphasized velocity curve into its counterpart in stylus excursion. As such, both depth and pitch control are most active when low frequency signals are received, while high frequency sound of even the highest intensity produces practically no increased depth or pitch.

G. PRECISION INSPECTION MICROSCOPE — ZA - 36:

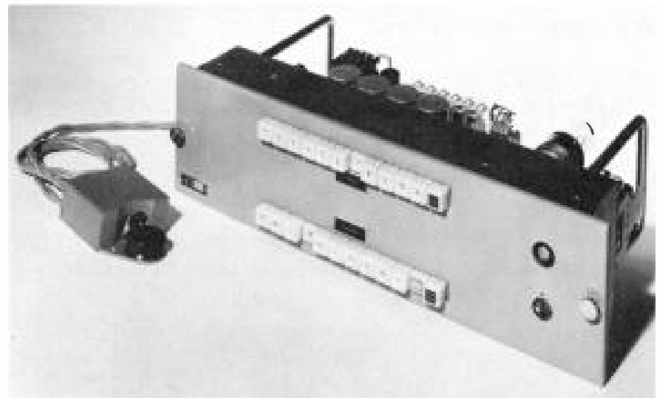
A 156-power precision "LEITZ" Microscope with concentric illumination and micrometer focusing is accurately rolled along an overhead track. The groove is brightly lit while the space between grooves stays dark. This is in sharp contrast to most such microscopes in which the brightest illumination is wasted on that part of the disk not under observation. The microscope is at right angles to the disk surface and moves along its radius. The microscope support arm also serves as the vacuum conductor for the vacuum chuch turntable. On it is further mounted the stroboscopic neon illumination and the ZA-29 pickup arm, if used.

The microscope is accurately calibrated in mils (.001") with a separately focusable eye piece.

H. LATHE CONSOLE CABINET — ZT - 32 S:

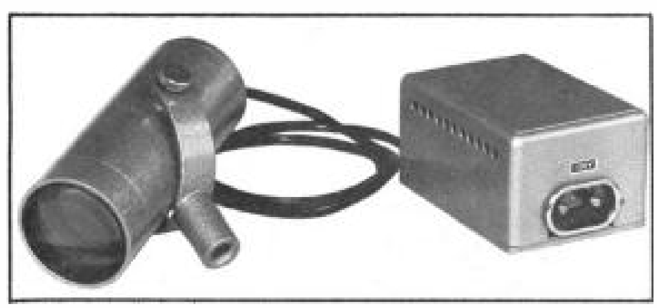
The entire complement of equipment described (except for the ZA-34b) is contained in a single steel frame console cabinet. The sides of the console are removable with the aid of a coin. A wiring harness connects all components both with power and audio leads. Controls mounted on this cabinet include main power switch and circuit breaker, vacuum pump switch, 16-2/3 rpm switch, remote start cutoff, and another associated with the photocell automation unit. The chip collection jar and all distribution valves and tubing are also neatly mounted inside this cabinet, which rests on shock mounts. All connections to the lathe are made to concealed terminal strips and connectors. The console always comes fully wired for the full stereo system even when only a monophonic complement of equipment is purchased initially.

I. PHOTO-CELL OPERATED AUTOMATIC SPIRALING AND LEAD-OUT UNIT — ZA - 34 b:



The AM-32b Stereo Disk Mastering Lathe can come equipped with a unit for the complete automation of the disk cutting process. A photo-cell placed on the tape playback machine directly after the capstan drive senses leader tape (or a silver mark) and actuates the ZA-34b unit. The unit itself has two rows of 12 push buttons. One row serves to pre-select the number of cuts or bands to be put on the disk, while the other chooses the number of seconds of spiraling between cuts. Once these are set the unit will automatically reproduce the identical spiral space for each leader, at the same time lighting up the button corresponding to the band being cut. When the pre-selected final cut has been reached a lead-out warning light lights and the last leader starts the lead-out program. When the cutterhead lifts, the automation unit returns to its "home" position, ready for the next disk.

J. BUCHMANN-MEYER PATTERN INSPECTION LIGHT — ZA - 21:



An indispensable aid to proper calibration of both monophonic and stereophonic disk systems is the Buchmann-Meyer pattern or "Christmas Tree". The ZA-21 light source and transformer provides the only absolute method of calibrating the frequency response of disk cutting systems independently from playback. The light pattern furthermore provides an instantaneous proof of recording levels and, to the experienced eye, a clue to the frequency response of recorded material. The ZA-21 is mounted on the ceiling above the lathe itself.

K. PICKUP ARM — ZA - 29:

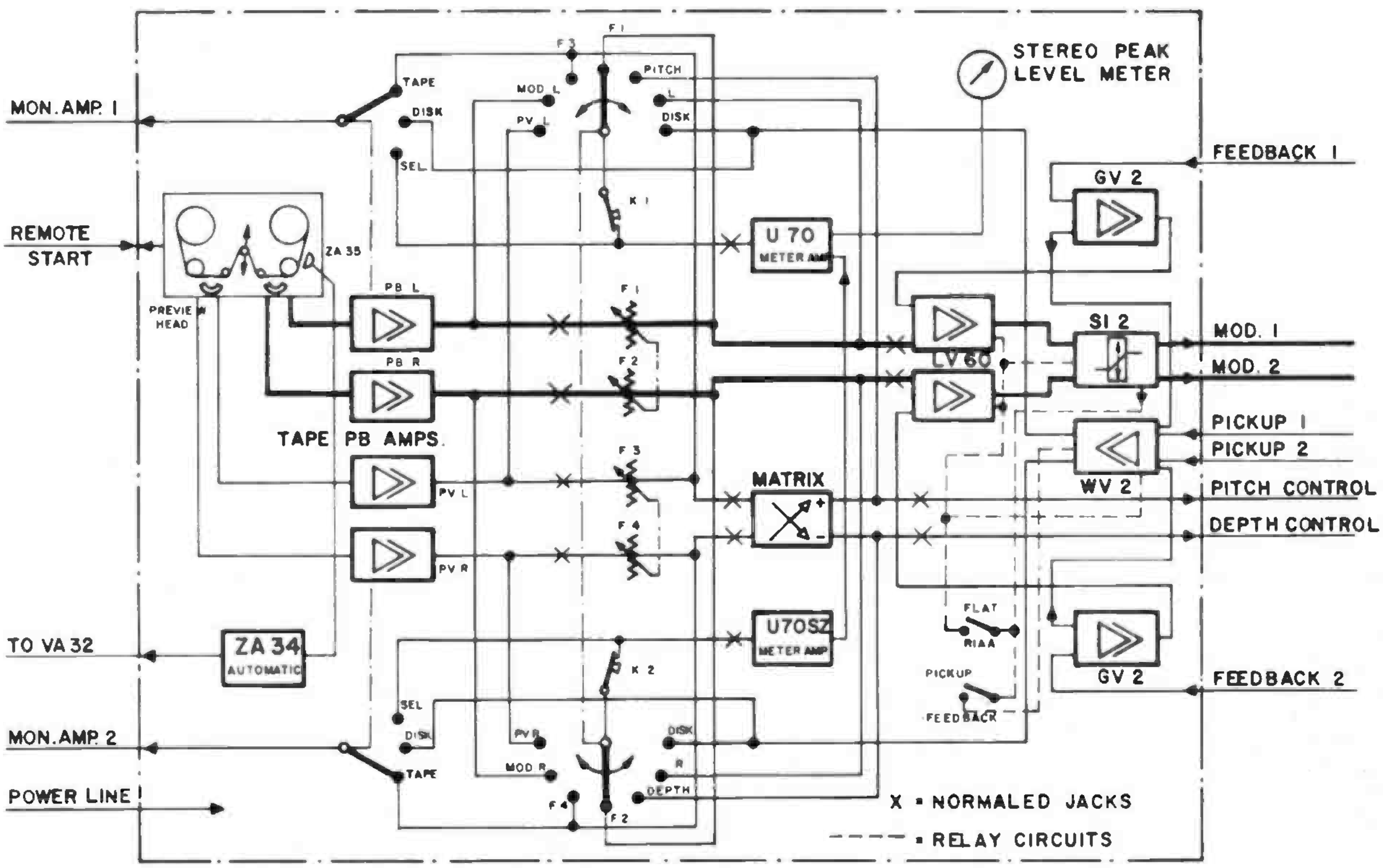
A pickup arm is very useful on a disk cutting lathe both for measurement of rumble and wow and flutter, and for instantaneous playback of recorded material as an "A-B" comparison with the feedback monitor output taken from the cutterhead. By simultaneously recording and playing the same groove, it is possible to get a direct comparison similar to that provided by professional 3-head tape recorders. The NEUMANN DST-62 double-dynamic stereo cartridge is supplied for utmost performance.

L. ELECTRONIC HALF-SPEED CONVERTER FOR 16-2/3 RPM — ZA-33R:

This small 4" x 4" x 5" unit containing a transformer and two selenium rectifiers, reduces the 33-1/3 RPM speed motor to 16-2/3 by doubling the ripple of the rotating magnetic field. For experimental uses, this unit will also produce the half-speeds of 45 or 78 RPM, useful for cutting frequency response disks and the like.

NEUMANN MONOPHONIC AND STEREOPHONIC RECORDING INSTALLATION

WFAA BROADCASTING CENTER • DALLAS, TEXAS



STEREO TAPE TO DISK TRANSFER

TECHNICAL SPECIFICATIONS:

Turntable speeds:	16⅔*, 33⅓, 45, 78 rpm (*with ZA-33R half-speed converter).
Turntable diameter:	16"; accommodates 17¼" blanks.
Vacuum Chuck Turntable:	Vacuum area adjustable for 10", 12", 13⅓" and 16" diameter. Hold-down knob (ZA-22) for holding 7" and 8" blanks.
Wow and flutter content:	Less than ± .035 RMS.
Rumble attenuation:	Greater than 55 dB.
Grouping:	None visible at 300 ipi (no modulation).
Cutterhead Suspension — SA-32:	Full-automatic with cutter lift, heated stylus switching, and electronically controlled depth-of-cut; dash pot with variable damping adjustment.
Cutterhead Mounting:	For NEUMANN Type ES-59 (mono) and SX-45 (stereo), and all standard cutters like Grampian, Westrex, Fairchild, Presto, Olsen, etc.
Starting cams for lead-in grooves:	Set for 7", 10" and 12" records (RIAA Standard).
Finishing grooves:	Concentric automatically; set for RIAA standard.
Groove pitch range, continuously variable:	
for 16⅔ and 33⅓ rpm:	90 to 475 ipi
for 45 rpm:	80 to 375 ipi
for 78 rpm:	60 to 350 ipi
Lead-in and spiraling pitch:	Adjustable 1.5 to 36 ipi individually for all three speeds; automatically switched.
Lead-out pitch:	Adjustable 0.9 to 36 ipi individually for all three speeds; automatically switched.
Lead-in spiraling program:	Automatically produces preferred "catch groove" which prevents stamper break-down.
Stylus heating current:	Adjustable up to 1.3 Amperes.
Microscopes:	150-power LEITZ and 80-power follower scope; both illuminated, and with calibrated reticle.
Power requirements:	208-240 Volts, 50-60 cycles, single phase; approx. 650 VA.
Dimensions of complete system:	51½" wide x 24" deep x 57" high.
Weight of complete system:	approx. 770 lbs. net.
Gross shipping weight:	approx. 1190 lbs. in six crates.

For further information, contact Gotham Audio Corporation

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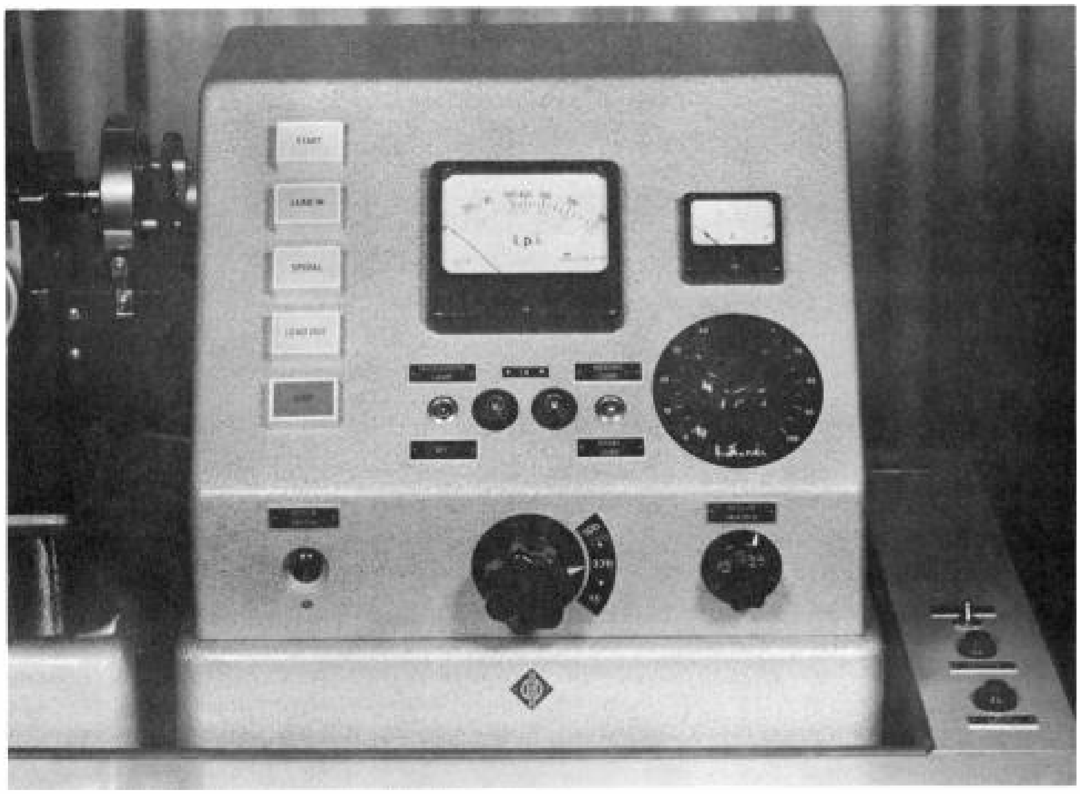


IMPROVED MODEL VA-62g PITCH CONTROL UNIT FOR NEUMANN AM-32b DISK MASTERING LATHE

Effective October 1962 the model VA-32d pitch control unit described in bulletin 3-a will be replaced by the improved model VA-62g, pictured below. Due to the lack of demand for recording at 78 RPM, this speed has been eliminated from the pitch drive unit's main functions, although the turntable motor speed remains available if desired.

The main dial calibrated in lines per inch (LPI) has been replaced by an electrical meter coupled to a generator which is tightly coupled to the lead screw drive mechanism and which accurately reads lines per inch directly at all times, even when cutting with automatically previewed pitch control.

A single current meter replaces the two previously used. A momentary switch permits the reading of either stylus heat current or pitch drive braking current (now no longer of interest during cutting since pitch control is indicated by LPI meter).

Lighted push buttons have replaced the tally lamps previously used. A fifth button permits the cutting of a special lead-in program as follows: When test grooves have been completed, the LEAD-IN button is pushed; the carriage advances at lead-out speed until it reaches the lead-in cam; a single revolution is cut at fine line pitch (so-called catch groove to prevent playback arms on changers from slipping outward); the regular lead-in is cut (tape machine starts); regular cutting begins.

The cutterhead lift delay switch (concentric-eccentric) previously on the front operating panel has been moved to the rear of the unit, since hardly anyone cuts anything but concentric end grooves at this time. The VA-62g pitch control unit permits selection of 16-2/3, 33-1/3, and 45 RPM on the main speed switch, while the accessory ZA-33R half-speed converter permits selection of 22-1/2 RPM. The 16-2/3 and 22-1/2 RPM speeds serve for high level cutting of disks at half speed. All dimensions and other operational functions of the VA-62g are identical to the previous VA-32 model.

Present owners of NEUMANN AM-32b lathes incorporating the VA-32 series of pitch drive units may convert these to the newer VA-62 model or may purchase separately the lines per inch indicating generator and calibrated meter for external mounting on their present equipment.

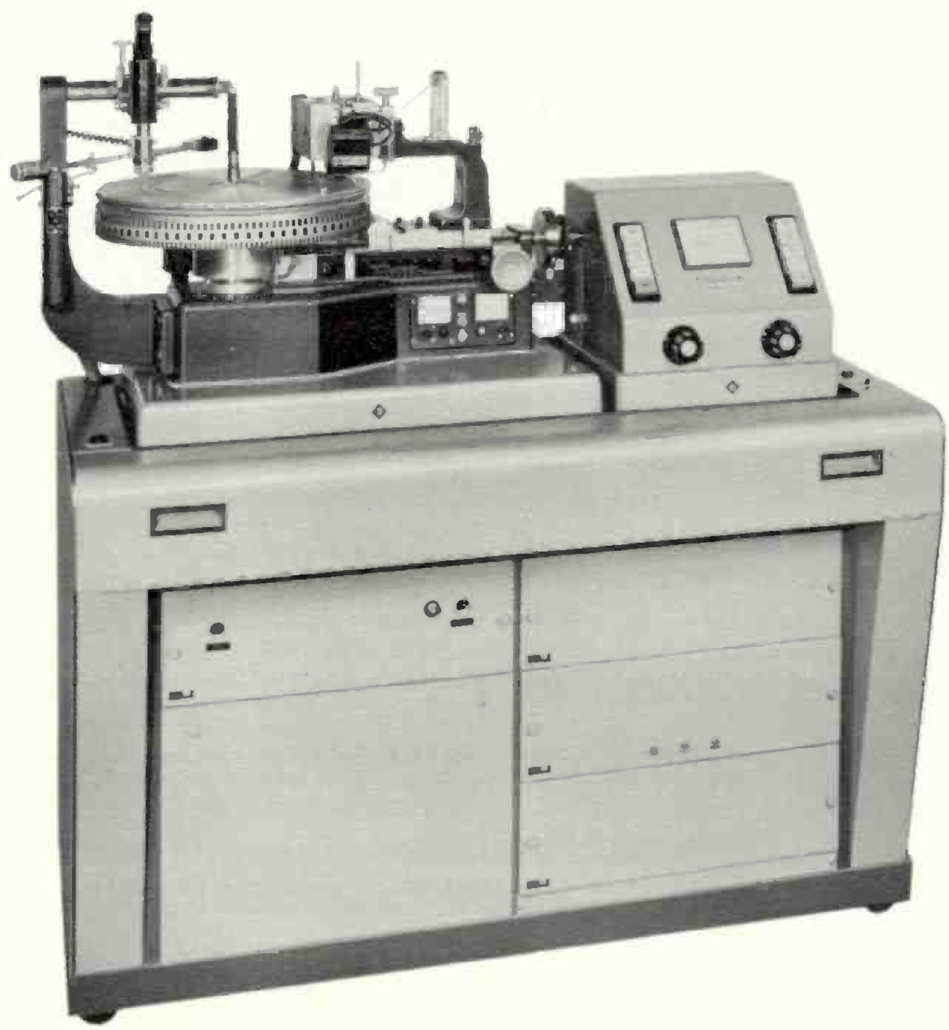
1062149

GOTHAM AUDIO CORPORATION
2 WEST 46 STREET. NEW YORK 36. N.Y. · · · (212) COLUMBUS 5-4111

VMS 66 - 910 - 02 - 00

The disk cutting lathe system VMS 66 is the latest addition to the world-famous line of disk mastering lathes made by a company with the "know how" that comes from nearly 40 years of experience in the design, construction and manufacture of disk recording machines and associated equipment.

The machine system is designed for cutting Mono and Stereo records of all standard diameters and revolutions per minute. Most modern control systems make the optimum utilization of the available disk surface possible. The cutting process is partly automatic. Start and stop of the tape deck, lowering and lifting of the cutter, and cutting of lead-in and final groove are performed automatically. All means for full-automatic disk cutting are provided.



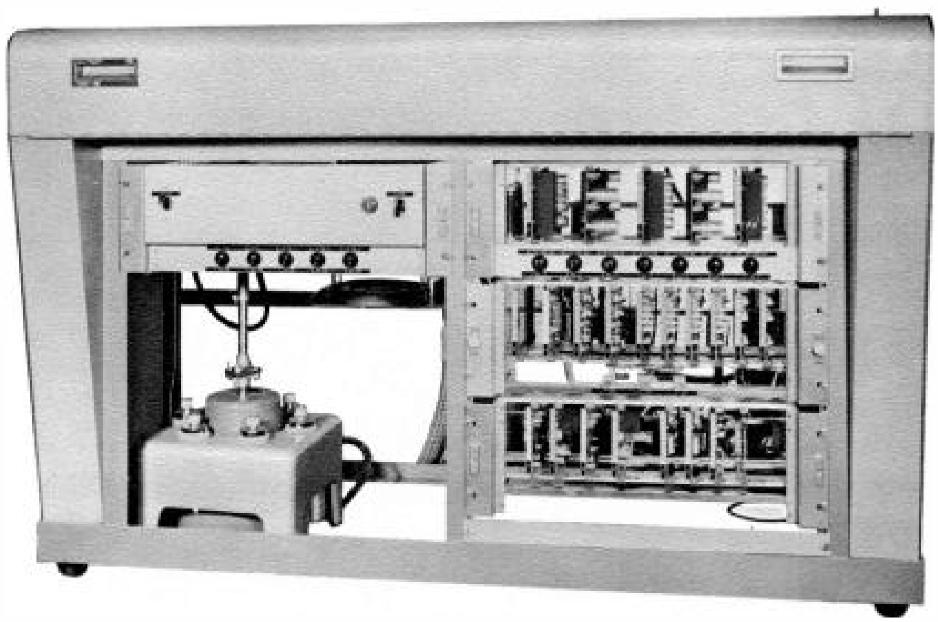
Features

The disk cutting lathe system VMS 66 incorporates the following main constituents:

Disk cutting lathe	AM 66
Leadscrew drive unit	VA 66
Drive control device	AS 66
Control amplifier	SV 66
Power supply	NA 66
DC supply	NG 66
Lathe console	ZT 66

All control units of the machine system are equipped with silicon planar transistors. No relays are included in the system; the switching operations are done by means of electronic switches.

The principal item of the machine system is the drive control system AS 66. It contains the control elements for all fundamental operations. The components are mounted on printed circuit boards plug-in type. The AS 66 is designed as a rack-mounted unit. Both the control amplifier SV 66 for amplitude controlled cutting and the central power supply NA 66 / NG 66 for alimentation of the machine system are designed in the same way. These units are incorporated in the lathe console ZT 66.



The leadscrew drive unit VA 66 contains the motors, the electronic operating knobs, as well as the instruments for indication of the groove pitch, and illuminated coloured plates indicating the function at work.

The revolutions per minute of the leadscrew drive motors is regulated by a servo mechanism. The range of revolutions per minute to be utilized is thus being increased and the gear can be made single stage. The axle of the servo generator is directly connected with the axle of the leadscrew drive motor. By this means all external load variations are controlled.

The disk cutting lathe AM 66 contains a new depth control unit TE 66 and a new cutterhead suspension type SA 66. A number of micro switches is located at the rear of the carriage support which deliver the pulses for the lead-in and the final grooves and for radius compensation when operating with the tracing simulator TS 66.

The turntable speed is set by plugging in the respective program plug. This program plug is located behind the front cover of the ZT 66. With this program plug the following operations are switched and controlled, respectively:

Selection of one of the speeds 16 2/3, 22 1/2, 33 1/3, 45, 78 rpm,

Conversion of the standard diameters 17, 25, or 30 cm and of the respective final groove,

Signalling of these functions to the leadscrew drive unit VA 66,

Limitation of the basic groove pitch to 3 ... 20 grooves/mm,

Conversion of the indicating instrument for the groove pitch (LPI) so that only one scale is necessary for all revolutions per minute,

Conversion of the pitch of the lead-in grooves, spirals and lead-out grooves under consideration of the turntable speed,

Conversion of the cutter lift delay of the cutterhead at the end of the disk,

Conversion of the duration of the safety groove to approx. 1,2 turntable revolutions. This safety groove function can be switched off.

Selection of the start order for the tape-playback console to the disk diameter necessary.

For amplitude controlled cutting also the variable pitch amplification is adjusted to the standardized level for the selected speed and the delay for the depth of cut is converted.

The groove cut with variable pitch and depth dependent on the amplitude is controlled by the control amplifier SV 66. A new method is being applied. The control amplifier equalizes the modulation to the space needed, dependent on the amplitude and frequency under consideration of the cutting characteristic. It also distinguishes between vertical modulation, outer groove wall, and inner groove wall. The control signals for the vertical modulation are taken from both the prelistening channels, the one for the outer groove wall from the right prelistening channel and the one for the inner groove wall from the left modulation channel. The control system requires an interval of half a turntable revolution between prelistening head and playback head on the tape deck.

The control signals are stored in four groups of four stores in every quarter of a revolution and are kept for half a revolution.

The first group stores the vertical signal for vertical pitch control.
The second group stores the vertical signal for lateral pitch control.
The third group stores the right signal for lateral pitch control.
The fourth group stores the left signal for lateral pitch control.

The three components for lateral pitch control change the speed of the lead screw drive, the component for the vertical pitch control changes the depth of cut.

The effect of this new control system is a much greater exactitude of control and thus a better utilization of space. There is only a control for the groove wall that needs space. The interrogated stores are erased as soon as they are not allowed to take any more influence on the control procedures. The storing and erasing functions are controlled by a pulse-generator from the turntable.

Further constituents of the VMS 66 sytem are:

Three program plugs	
Cutterhead suspension	SA 66
Depth of cut control panel	TE 66
Tonearm	ZA 39
Pickup	DST 62
Vacuum chuck turntable	ZA 3
Turntable drive motor	SM 8/3-A
Vacuum pump	ZA 1
Hold down knob	ZA 22

The disk cutting lathe system VMM 66 - MONO differs from the system VMS 66 - MONO-STEREO in so far that four printed circuit boards which are necessary for stereo operation are not included.

For optional augmentation the following accessories are available:

Buchmann-Meyer light for lightband width operation	ZA 21
Strobotron for checking of turntable speed	ZA 42
Tonearm with lowering mechanism	ZA 29
Monophonic feedback cutter complete with associated amplifier system	ES 59
Stereophonic cutter complete with associated amplifier system	SX 15

Technical Data

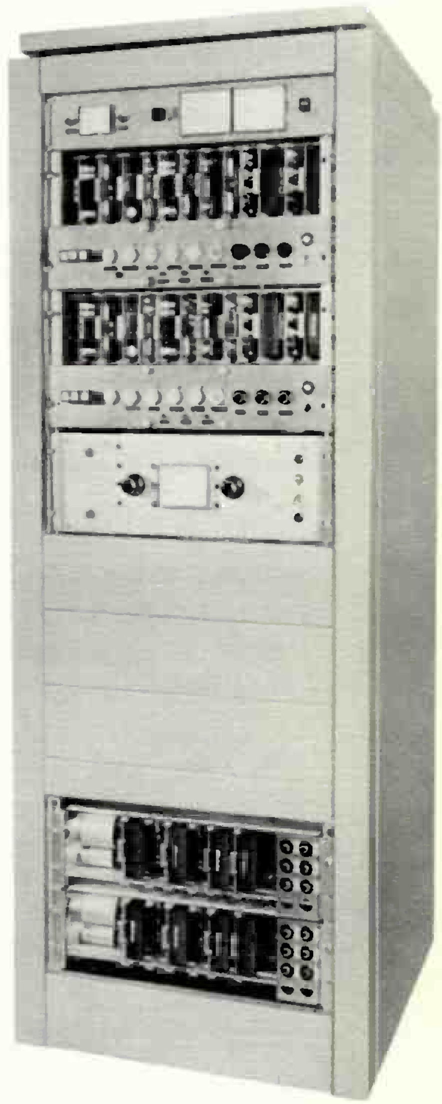
Turntable speed	78, 45, 33 1/3, 22 1/2, 16 2/3 RPM
Turntable diameter	16", accommodates 17 1/4" blanks
Vacuum-chuck turntable	vacuum area adjustable for 10", 12", 13 1/4" and 16" diameter
for small size blanks	hold down knob
Wow and flutter content	below .05 % peak-to-peak
Rumble below 10 cm/s 1000 Hz	≧ 70 dB (DIN 45 539)
Grouping	non-visible at 300 LPI (no modulation)

Cutterhead suspension	full automatic with cutter lift, heated stylus switching and electrically controlled depth-of-cut, dash pot with variable damping adjustment
Cutterhead mounting	for Neumann Type ES 59 (mono) and SX 15 (stereo), and all standard cutters like Ortofon, Westrex, Grampian, Fairchild, Presto
Automatic deepening	for lead-in and final groove
Automatic lead-in grooves	set for 7", 10" and 12" records, IEC standards
Final groove	concentric automatically, adjustable to IEC and other standards

Groove pitch range, continuously variable	for 78 RPM from 76 to 306 LPI for 45 RPM from 76 to 382 LPI for 33 1/3 RPM from 114 to 510 LPI for 22 1/2 RPM from 76 to 382 LPI for 16 2/3 RMP from 114 to 510 LPI
Lead-in and spiraling pitch, adjustable	0 ... 1/16" per revolution

Lead-out pitch, adjustable	for 78 RPM 0 ... 0,16" per revolution for 45 RPM 0 ... 0,27" per revolution for 33 1/3 RPM 0 ... 0,35" per revolution for 22 1/2 RPM 0 ... 0,35" per revolution for 16 2/3 RPM 0 ... 0,35" per revolution
Stylus heating current	variable up to 1.5 ampères
Microscope	202 power microscope with calibrated reticle
Power requirements	220 VAC, 50 or 60 Hz, approx. 500 watts
Dimensions of VMS 66 system complete	51 1/2" wide 24" deep 57" high, measured to top of microscope
Weight of VMS 66 system complete	approx. 680 lbs.
Gross shipping weight	approx. 1100 lbs.

VE 66 S -910-02-00



The amplifier equipment VE 66 S MONO-STEREO contains all units which are necessary for operation of feedback stereo cutterheads. All modules are executed in silicone planar technology. They are mounted on printed boards with front panel. All controls necessary for the set-up procedure of a cutterhead are contained in the cutter equalizer CE 66. This is a plug-in type unit which can easily be exchanged when another cutterhead is used. The following modules are mounted in several component assembly supports:

2 Recording equalizers	SE 66
2 Cutter equalizers	CE 66
1 Elliptical cutting equalizer	EE 66
2 Power amplifiers	LV 66
2 Feedback amplifiers	GV 66
2 Circuit breaker units	Si 66
2 Playback amplifiers	WV 66
2 Loudspeaker amplifiers	AV 66
2 Power supplies	NB 66

The circuit breaker unit Si 66 protects the cutterhead from thermal overload through disconnecting when the temperature of the driving coils grows too high. When a short-circuit occurs, the power supplies are disconnected automatically.

The amplifier equipment VE 66 M - MONO is fully STEREO-prepared. One each of the above mentioned units and the compatible cutting equalizer EE 66, however, is omitted.

TECHNICAL DATA

Amplifier equipment VE 66S

Input impedance	≧ 10 kΩ bal.
Nominal input level	0 dB (775 mV)
max. input level	+ 18 dB
after restrapping input transformer	+ 21 dB
Cutting characteristic (SE 66)	75/318/3180 µs corresponding to the standards DIN 45 537, IEC Publ. 98, RIAA

Power Amplifier LV 66

Frequency range	20 ... 20 000 Hz
Input level	approx. 0 dB
Input impedance	≧ 1 kΩ unbal.
Output power	≧ 100 Watts across 12 Ω
Distortion	≦ 1 % (40 ... 20 000 Hz, 100 Watts)

Feedback amplifier GV 66

Frequency range	20 ... 200 000 Hz
Input impedance	≧ 1.5 kΩ
Output impedance	≦ 30 Ω
Amplification	62 dB max.
Distortion	≦ 0.5 % (40 ... 20 000 Hz)

Playback Amplifier WV 66

Frequency range	20 ... 20 000 Hz
Input impedance	47 kΩ for magnetic pick-ups
2. input	for feedback monitoring
Output level	0 dB ... +6 dB adjustable + 21 dB max.
Distortion	≦ 0.5 % (40 ... 20 000 Hz)
Output impedance	≦ 30 Ω
Load impedance	≧ 200 Ω
Input sensitivity for +6 dB output	1 mV/cm/s velocity
Max. input voltage	60 mV
Reequalization of cutting characteristic	

Loudspeaker Amplifier AV 66

Frequency range	20 ... 20 000 Hz
Input impedance	≧ 1 kΩ
Input voltage	500 mV
Output power	15 Watts across 5 Ω
Distortion	≦ 0.5 % (40 ... 20 000 Hz, 15 Watts)

Power consumption VE 66	100 ... 240 V 50/60 Hz, approx. 400 Watts

DISK CUTTING MACHINE...... COMPUTER CONTROLLED

(Reprinted from RADIO MENTOR Electronic Magazine, October 1966 issue with the kind permission of the publishers)

by DIETER BRASCHOSS, Georg Neumann GmbH
Electroacustic, Berlin
translation by STEPHEN F. TEMMER
Gotham Audio Corporation, New York

Following a brief historical introduction, this article delves into the precision which characterizes the present state of the art in disk recording technology, and from them deduces the performance criteria of a modern disk recording system. The most meaningful solution of these requirements is to be found in the use of semi-conductor technology; i.e., the replacement of all relays by electronically operating flip-flops. The control of the pitch drive motors is electronically solved. The transistor functions and circuits are described in detail.

History:

More than a century has elapsed since the first preliminary efforts were made in the mechanical sound recording field. In the year 1859, an Englishman named L. Scott successfully recorded speech sounds via a membrane and needle on ıp black coated paper. Twenty years . .er, Edison invented the first actual recording process by substituting a tin cylinder for the lamp black paper, enabling the playback of the sound via needle and membrane. The phonograph record in its present form goes back to Emile Berliner (1887), who substituted the lateral cutting method used to this day, in place of Edison's vertical or "hill and dale" recording.

State of the Art:

In spite of the 87 year old tradition of of disk recording, there are still decisive improvements possible both with respect to technical matters, such as stereophony and tracing correction, and manufacturing. The following facts will serve to underline today's demand placed on disk recording precision. The excursion of a groove for a peak recording level at 1 kHz of 8 cm/sec. is ±15.9 micron (approx. ±0.7 mils). Disk reproduction of a signal of -60 dB is still audible. That means that the cutting stylus records an amplitude of ±0.0007 mils or 7 x 10 -7 inches. This excursion equals the wavelength of soft x-rays! The grooves on a disk have an average width of 2 mils and a spacing of .5 mils. The sum of these equals the thickness of a human hair. The pitch control, therefore, must be accurate to a fraction of a hair to prevent overcutting to the neighboring groove.

These two magnitudes are characteristic for the required precision in the industrial manufacture of phonograph records. The pitch control system for cutterheads must, therefore, be highly insensitive to disturbances of both an electrical and mechanical nature. The lead screw drive system must be continuously and reproducibly variable, and the amplifier system for the cutterhead must be devoid of any extraneous impulses.

New Developments Using Solid State Technology:

The new Disk Cutting Lathe, VMS-66, was developed in concert with the TELDEC Company and the Albrecht Company, both of Berlin. On the one hand it conforms to the most critical demands for precision; on the other, it is simple and automated. Since the cutting of acetate masters requires a large number of switching functions, and relay contacts are a constant source of electrical disturbance, the VMS-66 replaces all relays with silicon transistor flip-flops. The entirely altered operating principles of electronic switching also required a new control system offering greater flexibility and simplicity in operation. The high life-expectancy of the silicon transistors used can likewise be regarded as an important improvement.

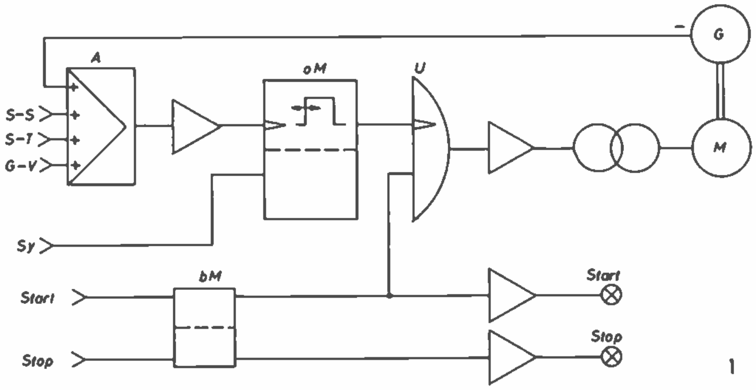
The most cogent operation simplification is represented by the programmer plug with which the entire disk cutting system is converted from one type of disk to another in a matter of seconds. Every desired combination of rotational speed (16-2/3, 22-1/2, 33-1/3, 45 and 78 rpm)* with any diameter (7", 10" or 12") may be selected. The required 20 function conversions are all effected through each programmer plug, which contains a number of passive circuit elements: 4 switching modes, 7 level changes, 9 time delay adjustments.

The Motor Control:

This simplification was made possible in the main through the use of an electronic speed control of the pitch drive motor. The amplitude dependent pitch drive requires rpm change of 1:3 within 0.1 seconds. Since such acceleration can only be obtained from motors with a minimal rotor mass, the VMS-66 is pitch driven, using Ferraris motors. Taking into consideration turntable speeds between 16-2/3 and 78 and the amplitude dependent pitch control, the motors must be capable of a control range of 1:22. The Ferraris principle is actually unsuited to such a range if a usable amount of power is to result, and it was only the advent of completely electronic control which made gear switching between turntable speeds unnecessary. Figure 1 shows the function block schematic of the new motor control with its operational elements. A power line

.pl. – Ing. Dieter Braschoss studied at the Technical University, Berlin and has been at the Georg Neumann Company since 1963, assigned to development research in the Disk Recording Division.

*) 16-2/3 rpm and 22-1/2 rpm turntable speeds are used for the cutting of 33-1/3 and 45 rpm records respectively when special quality demands are to be fulfilled.



1

Function schematic of the new motor control system with adder stage A, for the control voltages, the power line synchronized mono stable multivibrator aM, a bi-stable multivibrator bM and the AND-Gate U for control of the motor M.

synchronized multivibrator (aM) stabilizes the rpm of the pitch motor (M) through use of pulse width modulation, controlled by the generator (G) running on the motor shaft. A summing amplifier (A) combines the voltages for the basic pitch (G-V) as well as the amplitude dependent variable pitch (S-S) and variable depth (S-T). The START command for the motor is given via the bi-stable multivibrator (bM) and the AND-gate (U).

Figure 2 shows a schematic of the control circuit. T-4 and T-5 form a multivibrator which is power line synchronized via transistor 3. The pulse width is determined by means of the network consisting of T-1 and T-2 AND the re sistors R-1, R-2 and R-3. The operating point of T-1 and T-2 is determined by the control inputs. The voltage at G-V determines the basic pitch (LPI). Input S-S feeds the amplitude depend-dent voltage for the space needed to accommodate the lateral groove modulation, while input S-T feeds information regarding increased groove space required as a result of increased depth cutting in stereo recording.

The rpm dependent voltage from the generator (G) is fed to R-4. The pulse width controlled square wave is obtained from the collector of T-4 and is fed to the switching transistors T-8 and T-9 after amplification through T-6 and T-7. Since the impulse rate is power line synchronized, one of the switching transistors is bridged by a diode during each half wave, while the other is conductive for a time equal to the width of the pulse.

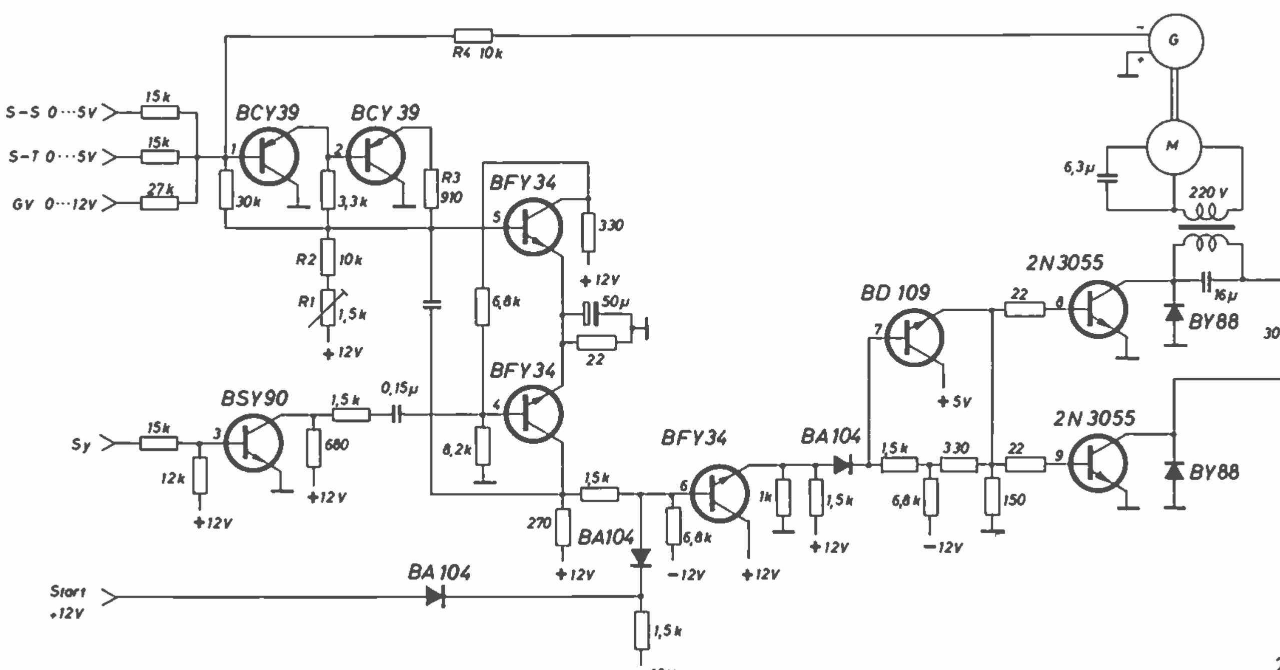
T-6 is blocked prior to the start by a negative bias of -12 V. Only after the START command connects +12 V are the square wave impulses connected to the switching transistors, and the motor runs. Any changes in the mechanical loading of the motor would cause a drop in rpm, which in turn would be compensated by the regulating loop through broadening of the square wave pulses.

A change in rpm through external influence is effected by changing the magnitude of one of the input signals G-V, S-S or S-T. This change in effect indicates to the regulating circuit a change in generator voltage, for which it tries to compensate by changing the the motor rpm, until the generator voltage compensates for the change in input voltage. Using this electronic regulating method, it is possible to obtain a stable 1:22 range in rpm.

The Pitch Control:

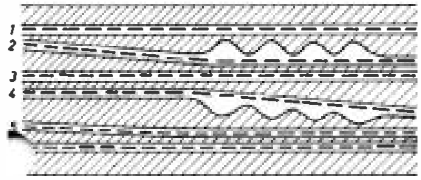
Together with the conversion of the pitch drive functions to semi-conductor circuitry, the amplitude dependent pitch and depth control amplifiers were like-were transistorized. A new control system was developed which differentiates between the various time constants associated with the left and right ex-excursions of the groove in stereo disk cutting.

Figure 3 shows a series of grooves schematically. Groove 1 has no modulation. In groove 2 a right channel sig appears. You can see that space has be provided even before the start of modulation. With the start of modulation the pitch drive can run more slowly again. Groove 3 has no modulation. Groove 4 shows the left channel modulation. In this case the basic fine line pitch can continue right to the start of



2

Complete schematic of the motor control system.



Cross section of 6 record grooves showing varying degrees of space required for outside and inside groove flanks.

modulation, when increased pitch is called for to prevent overcutting of groove 5, which follows. The following empty groove 6 shows by its spacing from groove 5 the operation of the control system.

The control signals needed for variable pitch and depth control are produced in a newly devised storage system. Every quarter revolution of the turntable, a memory storage unit is charged to a voltage proportional to the maximum modulation during that part of the revolution. This voltage is stored for two more quarter revolutions, while three other memory units are charged ¼ revolution apart. The maximum value of three successive memories produces the control signal for either pitch or depth control. After ¾ revolution of the turntable, each memory is cancelled in turn, so that after a full revolution the memory is available again and the cycle can begin anew. The independent treatment of the left and right channels requires four groups of four memories each. The memory group for the control of the left flank is obtained from the playback signal, while the right flank control signal and the groove depth control, both of which take place at the same time, are obtained ½ revolution before by means of a preview head. The charging, interrogation and cancellation in proper synchronism with the turntable revolution is effected by means of a counting circuit, which gets its impulses photoelectrically from the turntable itself. As a result of the time displaced control signal starts for the left and right flanks of the groove, as well as the precise cancellation after ¾ turntable revolution, a much more efficient utilization of the playing surface of the disk results.

TROUBLE SHOOTING IN DISK CUTTING SYSTEMS

by

STEPHEN F. TEMMER

Probably no other single piece of equipment in a recording studio is as misunderstood as the disk recording cutterhead or cutting system. It is without doubt the "moodiest" part of the technical setup; and judging from the many phone calls we get about such problems, it is a mystery to a vast majority of recording engineers. The following hints, although directly aimed at the GOTHAM-GRAMPIAN Disk Cutting System, apply equally to any disk cutting installation.

TYPES OF CUTTING EQUIPMENT:

Professional cutting heads are classified like any other transducer. (Transducers convert electrical into mechanical energy or vice versa):

MAGNETIC: Those which have a fixed coil and a moving iron armature. These include the Grampian, Fairchild, Presto, Olson, Lipps, NEUMANN MS-52H, and many others in the "less than $500" price class. One of these cutters, the Grampian, has a feedback coil wound right over the drive coil which acts as a magnetic field correction at frequencies below 1000 cps only. Except for the Grampian, these cutters have a mechanical resonance peak in the 4KC-7KC range which is damped either through the use of rubber, or other plastic solids, or with silicone fluid (Grampian) or silicone grease (NEUMANN MS-52H). If plastic solids are used, these deteriorate with time and must be replaced by returning the unit to the manufacturer, although some engineers manage to replace this themselves. The Grampian must have fluid added from time to time while the NEUMANN MS-52H is permanently sealed and contains silicone grease which lasts for the life of the cutterhead.

Magnetic cutters are fairly rugged, but they do display a relatively high distortion level of no less than 2% and as much as 4%. Except for the Grampian they are somewhat sensitive to acetate loading which means that the response is affected by the work which the stylus must do going through the acetate material. With decreasing groove speed at the inside of the disk this loading increases and the cutterhead looses some high frequency response. The Grampian cutters (both feedback and non-feedback models) have a resonance near 10 kHz achieved by greatly increasing the stiffness of the armature motion. The result is greatly reduced sensitivity, resulting in power requirements up to 150 Watts for driving, while other magnetic cutters may be driven from as little as 30 Watts. The higher resonant frequency requires corrective feedback at the low frequency end, and it is therefore recommended that the Grampian never be used without feedback. The advantage of this higher Grampian resonance is independence from acetate loading, extended high frequency response, and greater ruggedness. The GOTHAM PFB-150WA 150 Watt Power Amplifier, especially developed by GOTHAM to drive the Grampian Feedback Cutter, has become a standard work horse in the world's recording studios for over 12 years.

DYNAMIC: These cutterheads, like dynamic microphones, phonograph pickups, or loud-speakers have a coil attached to the armature which moves in a magnetic field. It is this class of cutterhead which is used by every major record producer in the world and which represents the highest available cutting quality. Such cutterheads are more sensitive and have their primary resonance in the 900 - 1200 Hz range. This resonance peak is removed by the use of negative "motional feedback". What this actually means is that a coil is attached somewhere near the stylus itself, magnetically shielded from the driving coil, which actually feeds back a signal directly proportional to the motion of the stylus. This feedback signal is amplified and fed back to the input of the driving amplifier out of phase, thereby correcting changes in loading, response, distortion, etc. Harmonic distortion levels achievable with such dynamic cutting systems are significantly less than 1%. Dynamic cutting systems are the NEUMANN ES-59, Westrex 2-B, and Ortofon. These are all MONO cutterheads. The same companies plus Fairchild also produce dynamic stereo cutterheads which together comprise the only dynamic moving coil cutters made in the world today. It naturally follows that dynamic cutting systems are considerably more expensive than the magnetic ones, and yet no major record producer uses anything but dynamic cutterheads for the mastering process.

HOW DO I CHECK THE LEVEL ALIGNMENT OF MY CUTTING SYSTEM ?

First of all you must have a standard way of aligning your system to a level and response performance which you can check at any given time. We recommend the use of the newly issued NAB 12" Test Record which, for the first time, marks the issuance of a standard disk by a standards association. The disk contains a 7 cm/sec alignment tone as well as the complete NAB (RIAA) frequency runs (MONO on one side and STEREO on the other). The disk is available through GOTHAM at $3.50 plus 50¢ for packing and postage. (For those of you who previously used the RCA 12-5-49 disk, you will note that this new NAB disk is 2 dB higher in its alignment level, since the 12-5-49 disk, even though many people thought otherwise, has an alignment tone level of 5.5 cm/sec.) The new NAB Test Record belongs in every disk cutting room! Your level alignment should be made at the 7 cm/sec NAB level. Common 45 rpm disk levels in the industry reach levels 4-7 dB <u>above</u> this NAB test level, while LP recordings are made up to 4 dB above this level. Bear in mind that the level as shown on your VU meter for program material has only a vague relationship to actual groove excursion on the recording, and it is this groove excursion and its curvature which determine whether or not the recording can be played back successfully with an average pickup cartridge. The amount of equalization used, limiting, compression; all these have a critical influence on disk playback performance. The only answer is <u>experimentation</u>. But get in the habit of identifying the level you do use by relating it to the RCA test record; i.e. "4 dB above 12-5-49 zero" (or the NAB Test Record; i.e. "2 dB above NAB Test Record zero"). Then we at GOTHAM will know what levels you are talking about. The quickest way to set up your cutting level standard after you have arrived at a setting of the volume control on your driving amplifier which produces the record loudness you want, is to then cut a 1000 cps test tone at the same zero level to which you adjusted your tape modulation level. Now dim the lights in the cutting room and shine a flashlight at a 45 degree angle at the disk you have cut, holding the light some 3 feet away. Measure the width of the shiny light beam which your lKC tone left and note <u>this measurement</u>. The NAB test tone has a light band width of about 40 mm (1 9/16"). You can, of course, check the level on your playback turntable as well.

<u>HOW DO I CHECK THE FREQUENCY RESPONSE OF MY CUTTING SYSTEM?</u>

This is one of most misunderstood and troublesome parts of the alignment. FIRST: There is absolutely no way you can check the entire response of your cutting system by cutting an acetate and playing it back on an aligned turntable! Acetates are and remain soft and the playback stylus deforms the groove and falsifies the playback levels compared to a pressing of the same test acetate. You must there- ore use another proven method: The BUCHMANN-MEYER LIGHT PATTERN or "Christmas Tree Pattern" as it is known. This is the same pattern described earlier for level alignment and is to be used only for frequencies above 1000 cps. Variations in recorded levels at any frequency result in a change in width of the light pattern and these relative widths are easily converted into dB. LIGHT BEAM WIDTH RELATES AS VOLTAGE. That means double the light band width means 6 dB higher level; half the width means 6 dB lower level. How to proceed:

1) Take a fresh 12" acetate and set the lathe for 78 rpm or, if you don't have that speed, 45 rpm. Pitch should be coarse; say between 80 and 120 lines/inch.

2) Set the equalization switch on your GOTHAM amplifier to "FLAT" position.

3) Feed an oscillator into your system so that you obtain a zero VU reading at 1000 cps on your program VU meter. You must continually observe this level to remain constant for all subsequent tests.

4) Start cutting this 1000 cps tone; cut for 15 seconds by the clock. Pause 5 seconds; switch to 15KC; cut 15KC for 10 seconds; pause 5 seconds; cut 14KC for 10 seconds; pause 5 seconds; etc., down to 1000 cps again. Make sure all frequencies read ZERO on your VU meter. Make sure you do it from high end to low end as indicated above.

5) Dim the cutting room light and, using a flashlight at 3 feet from the stationary disk, shine the beam at a 45 degree angle to it (angle not very critical for mono test).

6) What you should see for a properly operating system is all light bands parallel and of equal width. Don't forget the allowable tolerance. ±2 dB, for example, would mean the width may vary from 0.8 to 1.25 times the width of the 1000 cps test tone. ±3 dB would mean from 0.7 to 1.4; ±4 dB a change from 0.63 to 1.6; ±5 dB a change from 0.56 to 1.8; ±6 dB from 1/2 to double.

For the frequencies below 1000 cps use your turntable as follows: Play the NAB test record for those frequencies and note the exact error in your playback system at each frequency played as it relates to the 1000 cps tone in the middle of the frequency test portion of the test record. Do not use the alignment tone you have been using until now. Now record those same frequencies on an acetate without changing the former settings. Play the acetate back on the same turntable and compare the readings. The difference in the two errors is your actual cutting response on the disk.

HOW DO I DIAGNOSE MY PROBLEM?

The first thing to look for in the high frequency light band test is the condition around the resonant frequency of the cutterhead. For a Grampian, this is between 9 and 11 KC. If you see a peak in this area, it most likely means you need a few more drops of silicone fluid. Use the 100 cs viscosity. Some cutters are harder to damp than others and these need the 200 cs viscosity fluid. Add about 10 drops.

Adding any more will have no further effect and will only cause the cutter to leak. Let the system rest for an hour and then check just the 1KC light band and the band which had the peak in it side by side. It is this latter test which should be made once each week to determine if more damping fluid is needed. NOTE: Do not confuse the viscosity rating of the fluid (50,100,200 centi-stokes/second) with the type offluid which unfortunately happens to be "200 fluid"!

If your light pattern falls off slightly above the 10KC frequency, you may correct this condition by raising the screwdriver control at the "FLAT" position of the equalizer. Do not add too much of this compensation for it will make your recording even more sibilant. A slight roll off at the extreme high end is often helpful and desirable.

Any serious peaks or dips in the 3 - 7 KC range must be taken as an indication of malfunction of equipment. Check amplifier alone (see below) and cutterhead resistances as shown under "Other possible problem signs".

HOW DO I CHECK THE AMPLIFIER'S PERFORMANCE ALONE?

If your cutterhead did not behave well at the high frequency end, then you must check the response of the amplifier itself without cutting. Leave the settings as before and connect a Vacuum Tube Voltmeter between one side of the cutterhead and ground on your GOTHAM Amplifier output. DO NOT connect between output leads! On other amplifiers you may connect the VTVM directly across the cutterhead drive winding itself.

Now repeat the response run without cutting and observe the relative output levels. A properly functioning amplifier in its FLAT position should be flat ±2 dB from 1000 cps to beyond 17KC. NOTE: Since the screwdriver control at the "FLAT" position of the amplifier is effective at all times, you may find a rise above 10KC peaking at about 17-18KC depending on the setting of this control. You may note the setting of this control and then briefly turn it back to the left end to check the amplifier response thoroughly.

Once you have determined that the amplifier is flat in its "FLAT" setting, you may wish to check the RIAA pre-emphasis curve. To do this reduce the amplifier gain control by exactly 16 dB (8 steps on the GOTHAM Amplifier gain control). Set the equalizer switch to 33 or 45 position and again run through the frequencies beginning with 1000 cps and working up the line covering the frequencies listed on the NAB test record album liner. Your VTVM should now show the dB readings shown on the album; i.e. at 10 KC the output should be up 13.75 dB and at 15 KC 17.5 dB above the 1000 cps level. You may adjust both the control at the equalizer position as well as the FLAT control to bring the curve into line.

WHERE ELSE SHOULD I LOOK IF ALL THAT CHECKS OK?

1.) Feedback polarity: The Grampian feedback coil must be connected correctly. Reversing the polarity will produce a very odd response and distortion. Refer to your instruction manual.

2.) Stylus type used: A stylus with too great a burnishing edge will make you lose highs. Check the manual for our recommendations which are "B.B type MRS (Master Recording Stylus, recognized by its black aluminum shank), 87-90 degrees, wired for heat". Styli are available from Capps & Co. or other suppliers.

3.) Stylus heat setting: Do not use heat greater than 0.5 Amps. DO NOT adhere to the old and long since abandoned rule of thumb referring to "turning it up until the wires glow and then back off some". This is far too much heat! If you have an old stylus heat supply which uses a voltmeter (such as early Fairchild units) change the meter to an Ammeter showing 0 - 1 Amp. AC or DC may be used for heating. We suggest the use of the GA-4 Grampian Heated Stylus Clip for fast exchangeability.

4.) Angle of the cutterhead: The stylus should make a nearly vertical angle with the acetate disk. Check by lowering cutterhead on to the stationary acetate and look from the side. The stylus reflection and the stylus should make an angle of slightly less than 180 degrees at the cutterhead rear. In other words the stylus should slightly stroke the disk. This should be kept to a minimum.

5.) Line along which stylus moves: The stylus should move along a radius of the acetate. You may check this by carefully lowering the cutter to the stationary acetate and while lowered moving the cutterhead carriage 1/4" to the side so as to make a scratch. Now repeat at the inside of the disk without moving the disk. Lay a ruler along these two scratch marks and see if it goes through the disk's center or slightly forward of it. This line may come tangent with the front of the center pin but may not be at all behind the acetate's center.

OTHER POSSIBLE PROBLEM SIGNS:

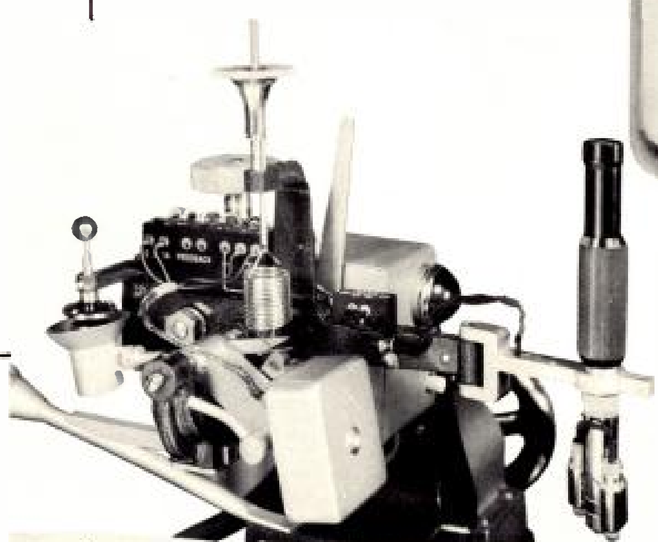
1.) Plate current excessive during cutting. For highest level disks this current should not ever exceed 450 mA readings. If it does and your levels are proper, check the DC resistance of the Grampian Cutter. Drive winding should be 3.7 OHMS and feedback winding 23 Ohms. Significantly lower drive ohmage may point to burned coil. Problem may be burned-out output transformer. Replacement available from Gotham Audio Corporation.

2.) Excessive silicone leak. Cutterhead should be returned for re-sealing. Grampian cutters are not serviced by Gotham Audio Corporation but by their importer:

Reeves Equipment Corp.
8 Third Avenue, Pelham, N.Y. att: Mr. Bowman
(212) TY 2-7010

3.) "Spitting" or "Sibilant" sound: This is almost invariably connected with improper damping resulting in high frequency boost or with excessive high end response of the tape itself or equalizer setting used in the transfer. Reducing high end will, in most cases, actually result in more apparent high end response due to better tracking.

MAGNETIC DISK CUTTING HEAD MS-52 H



As illustrated:
the MS-52H mounted
on the AM-131 Lathe

The NEUMANN MS-52H Magnetic Disk Cutting Head was developed most recently to fill a great gap in the disk recording equipment field. Presently there are available two basic disk recording cutterheads: the GOTHAM-GRAMPIAN Magnetic Feedback Cutting System and several even higher priced Moving Coil Dynamic Feedback Systems. To satisfy those recording studios and broadcasters who would like a cutterhead of excellent quality specifications able to be driven to high velocities by medium power speaker amplifiers, the MS-52H was developed as a sensitive device without necessity for damping fluids or solid damping materials which need to be periodically replaced.

GOTHAM AUDIO CORPORATION
2 WEST 46 STREET. NEW YORK 36. N.Y. · COLUMBUS 5-4111

OPERATING PRINCIPLES AND CONSTRUCTION:

The MS-52H Magnetic Cutterhead uses a brand new suspension system for the armature. While the Grampian cutter uses a torsion bar which produces tremendous stiffness and with it the need for power in excess of 100 watts for driving, previous magnetic systems such as Presto, Fairchild, Olson, etc. used solid damping materials and pivots which were prone to dislocation and breakage. The MS-52H utilizes a metal diaphragm mounted armature which produces high efficiency, sideways stiffness, and ruggedness. Attached to the armature above the diaphragm is a vibrating reed which is sealed into a tube filled with silicone grease for damping. This silicone grease need never be replaced or added to. It is permanent. The elimination of a stylus set screw reduces the mass of the system and extends the frequency response.

The cutterhead has built-in clamps to secure the heating wires from the stylus. Leads for drive coil and heating wires are brought out in one cable. The back of the cutterhead mounts a vertical rib which fits all NEUMANN lathes while an adapter supplied with the cutter fits all American lathe mountings. The stylus sapphire is mounted in a brass tapered shank. It is made by Frank L. Capps & Co. in New York. It is simply inserted into the cutter or removed from it by means of a tool which is provided.

TECHNICAL SPECIFICATIONS:

Frequency range:	40 — 12,000 cps
Impedance of drive coil:	16 Ohms.
Maximum power requirement:	8-10 VA (peak power not exceeding 30 watts).
Maximum allowable recorded velocity:	25 cm/second.
Total Harmonic Distortion: 12 cm/sec. at 1 KC:	less than 2% (played back with dynamic cartridge).
Maximum stylus heating current:	1.2 Amps.
Dimensions:	2" h. x 1¾" w. x ¾" d.
Weight:	10.5 ozs.

ACCESSORIES:

EQ-52H—RIAA recording pre-emphasis equalizer 600/600 Ohms.
Standard and Microgroove recording styli with heating wires, (in brass shank).

For further information, contact Gotham Audio Corporation or your Neumann dealer

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ES 59 - 910-02-01

APPLICATION

More than 25 years experience in the field of research, development and manufacture of cutterheads has led to the design of the E S 5 9 electro-dynamic monophonic cutterhead, which meets the most exacting demands of present-day professional users.

The E S 5 9 cutterhead can be used in conjunction with all commercial disk cutting machines for lateral micro and standard groove recordings. It is suitable for the cutting of lacquer masters, as well as for the cutting of other recording blanks for direct playback purposes.

FEATURES

The E S 5 9 single channel cutterhead is a moving coil feedback type for operation with hot styli. The construction of the armature together with a novel suspension results in an oscillating system whose first secondary resonance does not occur until approximately 55,000 c/s. This permits such a great amount of feedback to be applied around the cutterhead that it becomes linear over a frequency range from 30 to 24,000 c/s without any additional electrical equalization. This feedback even applies to the very low frequencies where distortion might otherwise occur.

TECHNICAL DETAILS

The armature of the E S 5 9 is made in the shape of a hollow cone, the tip of which forms the stylus holder. Also near the tip of the cone are the two feedback coils, while the drive coil is mounted at the upper end of the armature right-angled to the plane of the feedback coils. This ensures that the feedback voltage obtained is proportional to the stylus movement.

On the back of the cutterhead is a fixing rail of the same type as on the well-known cutterheads MS52H (NEUMANN) and DS522 (ORTOFON). An adapter which slides up and down this fixing rail is supplied with the cutterhead. This adapter is used for mounting the cutterhead on the suspension of the NEUMANN A M 3 2 b disk cutting lathe.

The electrical connections to the drive and feedback coils, as well as to the stylus heater, are brought out at the front of the cutterhead on a 10-conductor TUCHEL miniature connector.

The sapphire is held in a hollow shank which in turn is taper-fitted to the stylus holder in the cutterhead.

A power amplifier and a feedback amplifier are needed to operate the cutterhead. Also available are a circuit breaker unit to protect the cutterhead against thermal overload, and a playback amplifier for simultaneous monitoring and for the playback of records after cutting. These items can be mounted together in a separate swing-out amplifier rack, or they may be built into the console of the cutting machine.

ACCESSORIES

Power Amplifier L V 6 0

The L V 6 0 power amplifier is push-pull throughout and has a class AB output stage. Exchangeable plug-in equalizers are used to give the usual cutting characteristics of RIAA, DIN or others.

Feedback Amplifier G V 2 a

The G V 2 a feedback amplifier is free of phase shift and is used to amplify the feedback voltage obtained from the cutterhead.

One-channel Playback Amplifier W V 1

The W V 1 one-channel playback amplifier is a four-stage voltage amplifier with several feedback loops. It can likewise be used to amplify and equalize both the output of the pick-up or that of the feedback coil in the cutterhead.

Circuit Breaker Unit S i 1

The function of the S i 1 circuit breaker unit is to protect the cutterhead from thermal overload. The breaker causes the cutterhead circuit to open within a certain voltage range after a defined time delay. The delay is so chosen that it corresponds with the thermal inertia of the drive coil in the cutterhead, and thus interrupting the circuit only shortly before the critical temperature is reached.

An ammeter, which may also be fitted separately from the disk cutting machine at any convenient location, permits continuous control of the actual cutterhead current.

The circuit breaker unit also provides the DC supply necessary for actuation of the various control relays in the other amplifiers.

A l l amplifiers have a separate built-in mains supply.

They are available for rack mounting according to choice either for DIN 41490 or 19" rack width.

S P E C I F I C A T I O N S

Cutterhead E S 5 9

Construction	moving coil feedback cutterhead without mechanical damping
Frequency range	30 to 24,000 cps
Impedance	
drive coil	6 ohms
feedback coil	150 ohms
Impedance characteristics ..	see graphs
Max. velocity	
for normal cutting over the entire frequency range	30 cm/sec. or 170 mm light band width at 33 1/3 rpm
at mid-frequencies	30 cm/sec. or 170 mm light band width at 33 1/3 rpm
at high frequencies	10 cm/sec. or 57 mm light band width at 33 1/3 rpm
Max. continuous load	500 ma
Max. amplitude of stylus (peak-to-peak)	0.3 mm (.012")
Stylus mounting	taper fit 1 in 20
Dimensions	
height	51 mm
width	44 mm
depth	21 mm
Weight	.5 lbs.

Power Amplifier L V 6 0

Frequency range	30 to 15,000 cps
Input level	app. 0 dB (775 mV)
Input impedance	greater than 1000 ohms
Output power	60 watts
Noise level	less than 5 millivolts
Distortion	less than 1%, 40 to 10,000 cps
Power source	110/127/220 volts, 50/60 cps
Power input	200 VA
Tubes	3 x EF804s, 1 x E80CC, 2 x EL156
Mounting	DIN 41490 or 19" rack
Dimensions	
width	520 mm
height	200 mm
depth	250 mm
Weight	42 lbs.

Feedback Amplifier G V 2 a

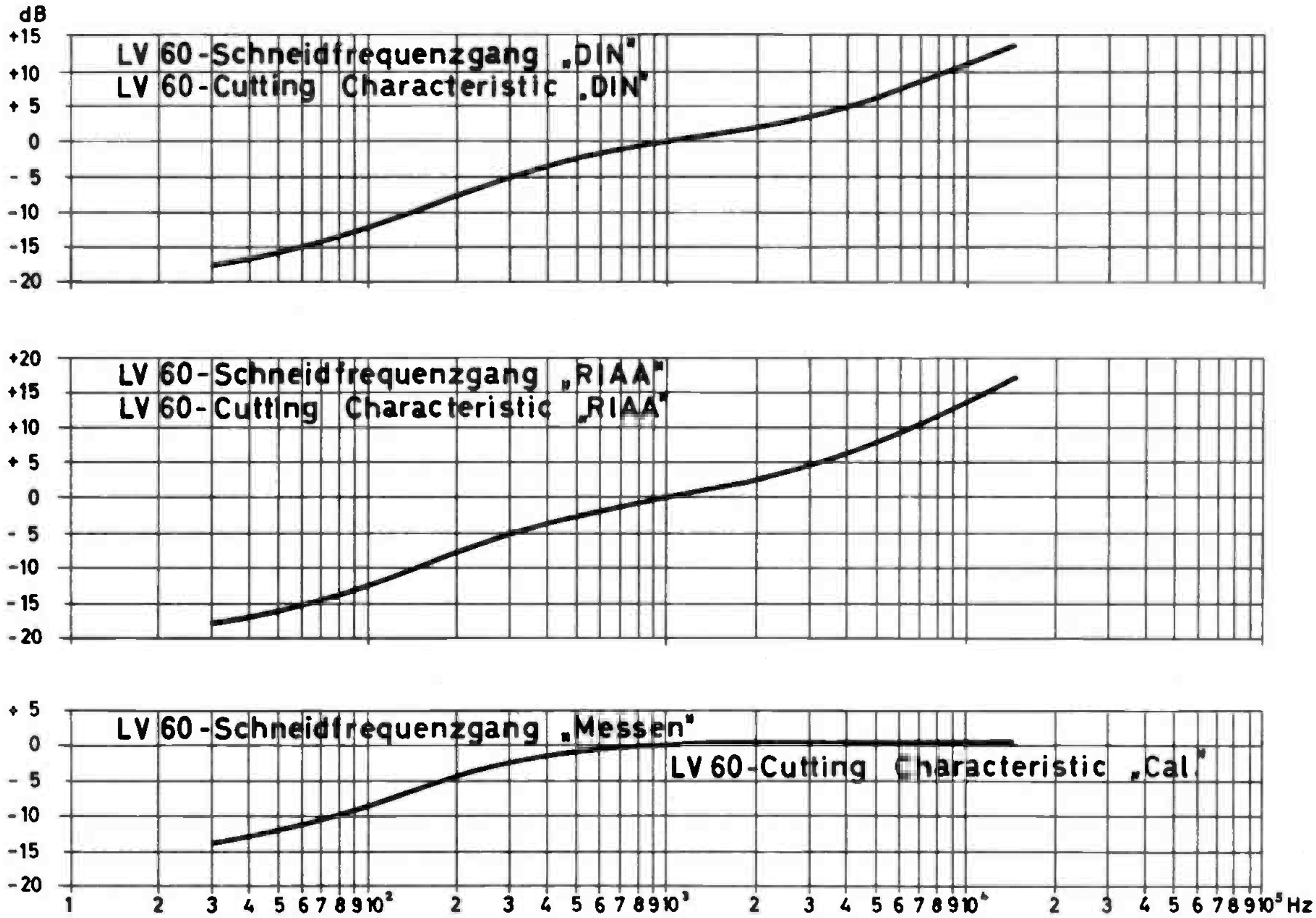
Frequency range	20 cps to 200 kcps
Input matching	for cutterhead ES59
Output matching	1. balanced for LV 60 2. unbalanced for WV 1
Gain	52 dB maximum
Noise level	less than .5 millivolts
Distortion	less than .8%, 40 to 10,000 cps
Power source	110/127/220 volts, 50/60 cps
Power input	22 VA
Tubes	2 x E80CC, 1 x EF80
Mounting	DIN 41490 or 19" rack
Dimensions	
width	520 mm
height	70 mm
depth	240 mm
Weight	11.3 lbs

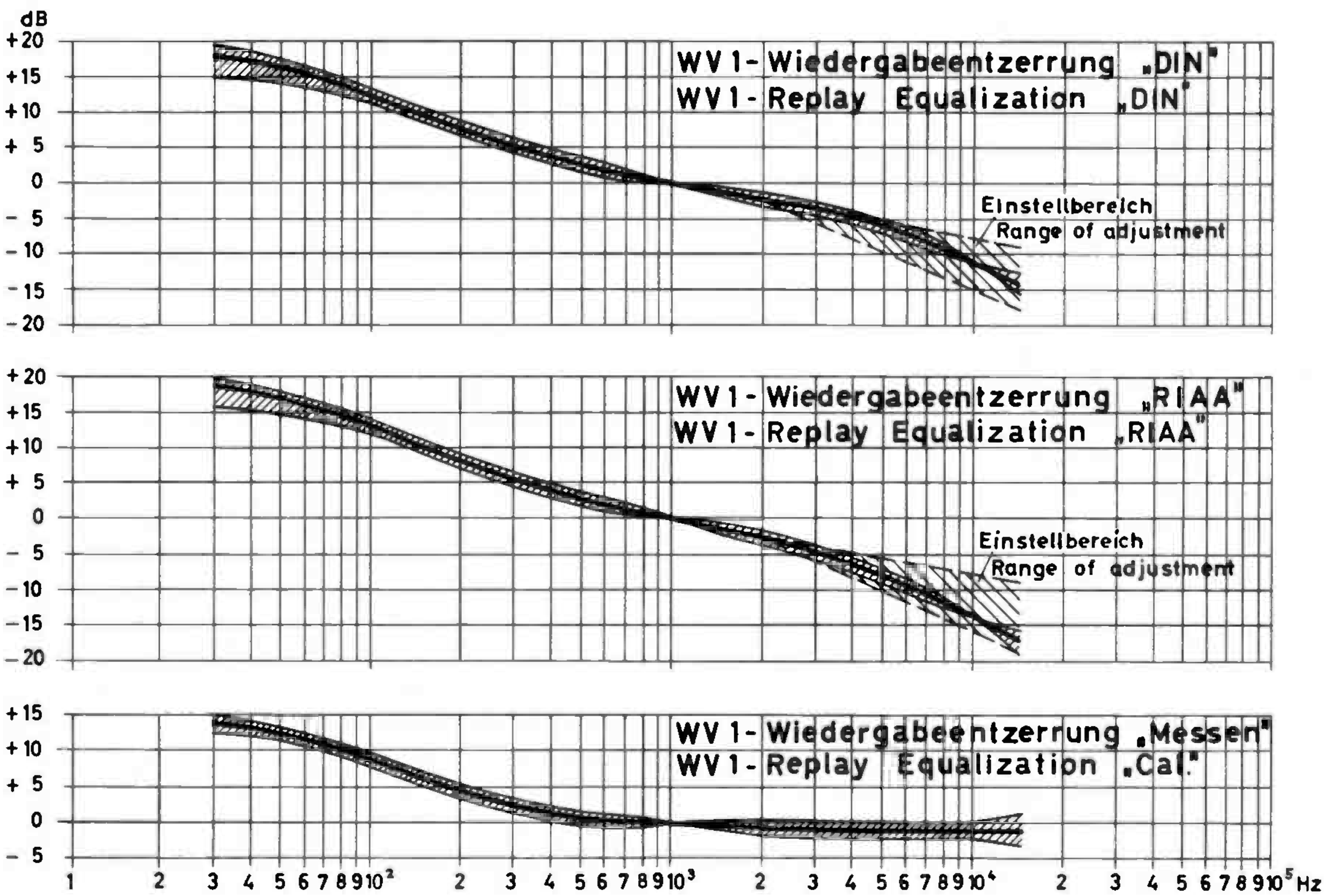
One-channel Playback Amplifier W V 1

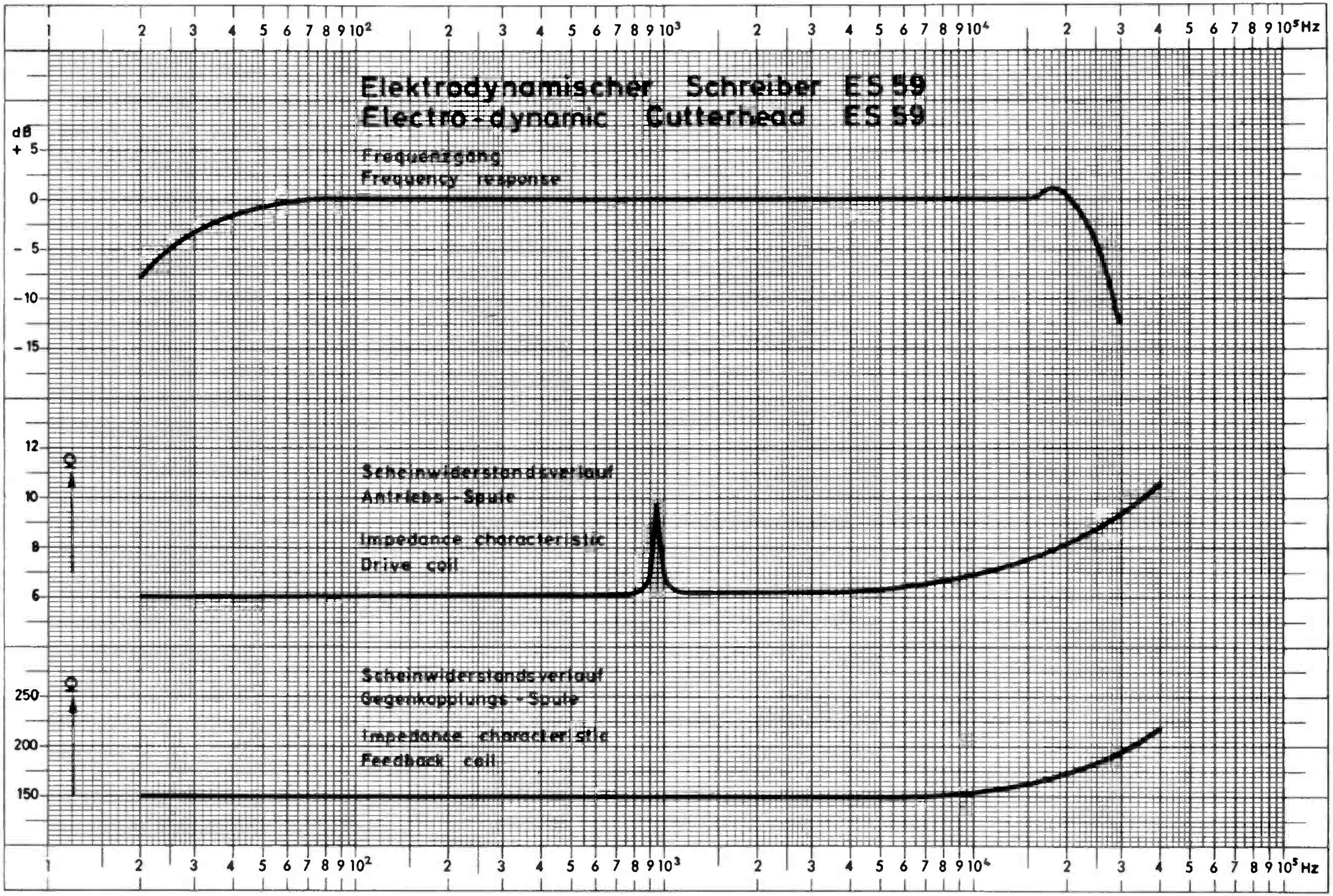
Frequency range	30 to 15,000 cps
Input matching	1. for moving-coil pickups with impedances from 1 to 40 ohms 2. for feedback amplifier GV2a
Output level and Output impedance	+6 dB (1.55 volts),less than 30 ohms
Output level and Output impedance on VU meter output	+18 dB (6.2 volts),approx.350 ohms
Terminating impedance	greater than 200 ohms,resp.6 kilohms
Gain at 1 kc	50 to 60 dB, depending on pickup impedance
Signal-to-noise ratio	better than 60 dB
Distortion	less than .4%, 40 to 10,000 cps
Power source	110/127/220 volts, 50/60 cps
Power input	16 VA
Tubes	2 x E 80 CC
Mounting	DIN 41490 or 19" rack
Dimensions	
width	520 mm
height	70 mm
depth	165 mm
Weight	6.5 lbs.

Circuit Breaker Unit S i 1

Frequency range of the cutterhead current meter ...	30 to 50,000 cps
Voltage available for relay operation	approx. 24 volts, .4 amps
Power source	110/127/220 volts, 50/60 cps
Power input	10 VA
Mounting	DIN 41490 or 19" rack
Dimensions	
width	520 mm
height	70 mm
depth	200 mm
Weight	8.8 lbs.


LV 60-Schneidfrequenzgang „DIN"
LV 60-Cutting Characteristic „DIN"
LV 60-Schneidfrequenzgang „RIAA"
LV 60-Cutting Characteristic „RIAA"
LV 60-Schneidfrequenzgang „Messen"
LV 60-Cutting Characteristic „Cal."
dB
Hz


WV 1-Wiedergabeentzerrung „DIN"
WV 1-Replay Equalization „DIN"
Einstellbereich
Range of adjustment
WV 1-Wiedergabeentzerrung „RIAA"
WV 1-Replay Equalization „RIAA"
Einstellbereich
Range of adjustment
WV 1-Wiedergabeentzerrung „Messen"
WV 1-Replay Equalization „Cal."
dB
Hz


Elektrodynamischer Schreiber ES 59
Electro-dynamic Cutterhead ES 59
Frequenzgang
Frequency response
Scheinwiderstandsverlauf
Antriebs - Spule
Impedance characteristic
Drive coil
Scheinwiderstandsverlauf
Gegenkopplungs - Spule
Impedance characteristic
Feedback coil
dB
+ 5
0
- 5
-10
-15
12
10
8
6
250
200
150
Ω
1 2 3 4 5 6 7 8 9 10²
2 3 4 5 6 7 8 9 10³
2 3 4 5 6 7 8 9 10⁴
2 3 4 5 6 7 8 9 10⁵ Hz

GOTHAM
AUDIO CORPORATION
2 WEST 46 STREET, NEW YORK, N.Y., 10036 · 212-CO-5-4111

SX-68

4·g

The demands placed upon stereo cutterheads in the last ten years have risen steadily.

Initially the demands were for technical quality such as frequency response, channel separation and lack of distortion. Cutterheads were built using state-of-the-art precision machine shop techniques which allowed feedback to be applied over the entire frequency range and thereby fulfilled these quality demands.

All of the world's major record companies have come to the conclusion over the years that the sales success of phonograph records not only depends on the specified quality parameters, but additionally is a function of the amount of level which is recorded on these records. In many cases even the largest record companies compromised in such a way as to raise the recording velocity at the expense of such quality considerations as frequency response and, more importantly, distortion. In many cases this compromise involved the correcting of frequency response predominantly at the high frequency end, or the use of cutterheads which achieved higher sensitivity at high frequencies by secondary resonances within the audio range.

These measures lead to records with high recorded velocities. The unwanted secondary effect, however, is that distortion is radically increased as a result of a secondary resonance within the audio range. The feedback which normally suppresses distortion cannot be applied in the area of the secondary resonance due to the phase shift at these points, and as a result has to be significantly reduced over the entire range.

To solve this problem, NEUMANN developed the Stereophonic Cutterhead SX-15 which has no secondary resonances within the audio response range. This solved the problem of the undesirable distortion increase. As opposed to the previous models ZS-90/45 and SX-45, the SX-15 has two separated drive coils on two armatures which are solidly linked to form a single moving system. This construction increased sensitivity resulting in a significant increase in recordable level on the record.

Although many large record companies throughout the world have had striking success using the SX-15 stereo cutterhead, the NEUMANN Company nevertheless continued research work to find an even better solution which would satisfy all of the requirements of all recording laboratories everywhere. It might be interesting at this point to explain the name "TELDEC" which has been closely linked with that of the NEUMANN Company over the past twelve years. "TELDEC" stands for Telefunken-Decca, a 50/50 partnership between these two important German and English companies based in Hamburg and Berlin, and the record producing, pressing, and distributing organization for all Telefunken, Decca, London, RCA-Victor and several smaller U.S. record concerns. TELDEC has important basic engineering laboratories and indeed did most of the early development work on stereophonic cutterheads dating as far back as 1953, fully four years before the introduction of the first U.S. stereo cutter. The three stereo cutterheads produced by NEUMANN thus far, the ZS-90/45, SX-45, and SX-15 were all developed jointly with TELDEC and were so marked on their serial number plates. the here described Model SX-68 is the first stereo cutterhead developed entirely by the NEUMANN organization and more specifically by Mr. Neumann himself.

In the design of an "ideal" stereophonic cutterhead, the following points will have to be considered:

1. No secondary resonance in the audio range; if possible it should be far outside the range.

2. High feedback capability (motional feedback available only on dynamic or moving coil systems) over the entire audio range; feedback reserve to prevent oscillation.

3. As a result of 1. and 2. linear frequency response with deviation less than 1 dB over the entire audio range.

4. As a result of 2., extremely low distortion over the entire audio range.

5. High degree of channel separation.

6. Ability to record the highest levels without endangering the cutterhead mechanism.

7. Simple operation; change of cutterhead without time consuming corrections.

8. High degree of mechanical integrity and ruggedness. Simple, non-critical stylus change.

The NEUMANN Company and GOTHAM AUDIO CORPORATION are delighted to be able to report that the solution to these problems has now been found. The principle employed in the model SX-15 has been developed further to provide 2 drive coils on 2 separate armatures rather than on a single armature. All cutterheads known up'til now which utilize this system, have also shown secondary resonances within the audio range. The new NEUMANN Dynamic Stereophonic Cutterhead SX-68 utilizes two separated armatures so linked as to produce <u>no</u> secondary resonances below 75,000 Hz!

The cutterhead furthermore has a vertical tracking angle of 15 degrees and can be used for the cutting of highest level monophonic masters (even 45 rpm rock) with quality performance even superior to that of any current mono cutter.

A new, fully transistor equipped amplifying system, MODEL VG-68, has been developed which complements the excellence of the NEUMANN SX-68 Stereo Cutterhead.

<u>TECHNICAL DATA:</u>

Frequency range:	40 - 16,000 Hz
Frequency response "overall":	± 1 dB
Channel separation over the frequency range:	≥ 35 dB
Feedback capability at 100 Hz:	6 dB
Maximum velocity (lateral motion) at 10 kHz (continuous sine wave):	
without cooling:	15 cm/sec = 1 Amp.
with cooling:	26.5 cm/sec = 1.8 Amp.
at 10 kHz (for 10 seconds):	
without cooling:	22.5 cm/sec = 1.5 Amp.
with cooling:	40 cm/sec = 2.7 Amp.
Drive coil impedance:	5 Ohms
Feedback coil impedance:	80 Ohms
Weight (with cable):	approx. 1 lb.

GOTHAM UNIVERSAL EQUALIZER EQ-1000

The latest studio operating techniques in phonograph recording, broadcasting, film, and TV have made the ability to influence the sound picture through use of equalizers a necessity and an art. Accurate changes in frequency response within the audio spectrum are also needed in the acoustical measurement, psycho-acoustic, electronic music, and communications industries. To meet these requirements, numerous devices have

GOTHAM AUDIO CORPORATION
2 WEST 46 STREET. NEW YORK 36. N.Y. · · · COLUMBUS 5-4111

appeared on the market, each with a specific function of equalization; each one different from the next. Almost without exception these devices have used inductively tuned circuits to achieve frequency discrimination. Furthermore, to achieve complete flexibility, many engineers have found it necessary to use more than one type of equalizer in a single circuit. The resultant phase shift, distortion, noise level, and transient degradation of the signal has been regrettable. It was the search for one universally flexible equalizer capable of the functions of all the different types on the market without the accompanying signal distortion, that has led to the development of the GOTHAM EQ-1000 Universal Equalizer — an equalizer which offers an unusually varied set of possibilites for influencing the frequency response, amplitude, and slope of the response curve within the audio spectrum. Amplitude and frequency changing controls are either pushbuttons or step controls, permitting simple reproduceability of any setting of the equalizer at any time. The resultant accurate maintenance of response curves makes the device eminently suitable for stereophonic work or in the measurement field.

OPERATING PRINCIPLES:

The most unusual feature of the GOTHAM EQ-1000 Universal Equalizer is the fact that it achieves its equalization with not a single inductive component. It is entirely RC operated. The elimination of inductively tuned circuits has the following advantages:

1. Very low harmonic distortion.
2. Minimal intermodulation distortion.
3. Extremely low self-noise level.
4. Low sensitivity to induced magnetic and electrical fields.
5. Excellent square wave response; no overswing; no ringing.
6. Smooth transition between individual filter networks; high accuracy of response settings.

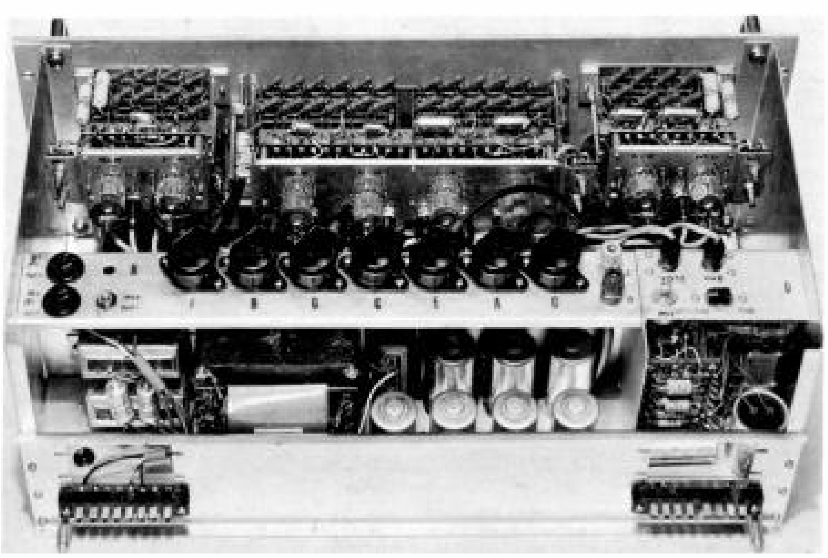
CONSTRUCTION:

The EQ-1000 Universal Equalizer features unitized construction throughout. Each set of push-buttons mounts its own amplifying stages, resistors, and condenser, and gets its power from a power distribution strip atop the power supply which forms the back of the entire unit. There are seven such sub-assemblies in all including the balanced input and output stages of the equalizer. The "building block" units are interconnected with plug-in connection cables. Each of the six building blocks influences a specific part of the frequency spectrum. Each of the building blocks makes it possible to alter the frequency response curve in frequency, amplitude, and slope simultaneously and independently from one another at eight distinct segments of the audible range.

APPLICATION:

Due to its extremely low noise level, the EQ-1000 may be inserted directly in the microphone channel of a studio console, while its high output capability makes it ideally suited to line level operations.

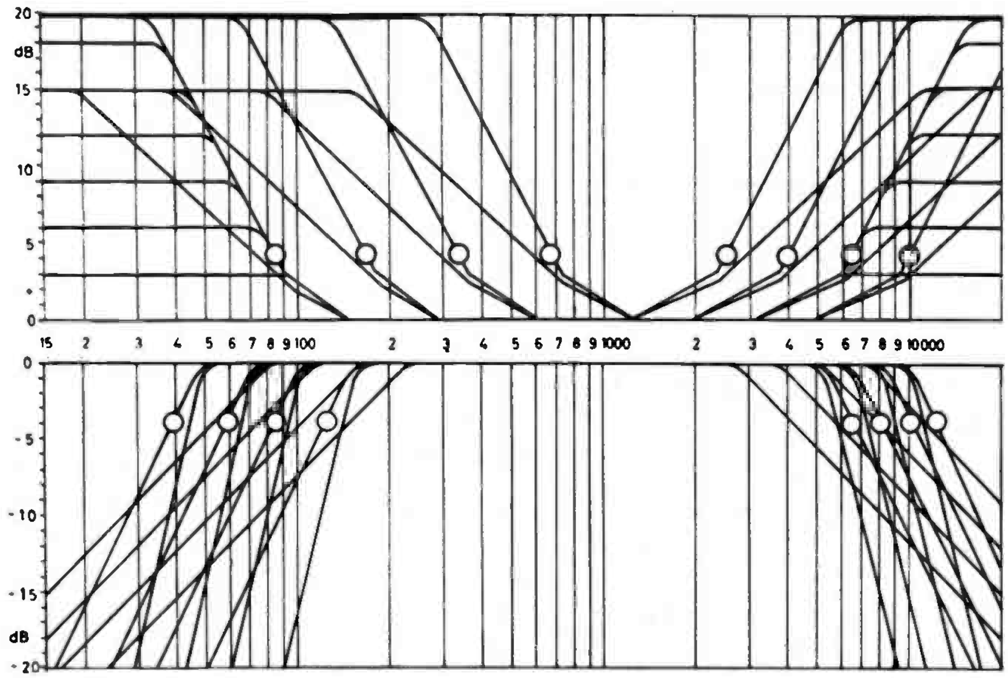
The EQ-1000 may either be used to produce a known curve, to compensate for an unwanted frequency characteristic, or it may be adjusted entirely by ear to suit the subjective taste of recording engineers and artists. It may be used in either or both channels of a stereophonic recording without the danger of phase shift incurred when equalizing both channels independently from one another, using inductively tuned equalizers. Its accuracy of adjustment and freedom from "sweep-type" controls makes it possible to note settings accurately for future reproduction.



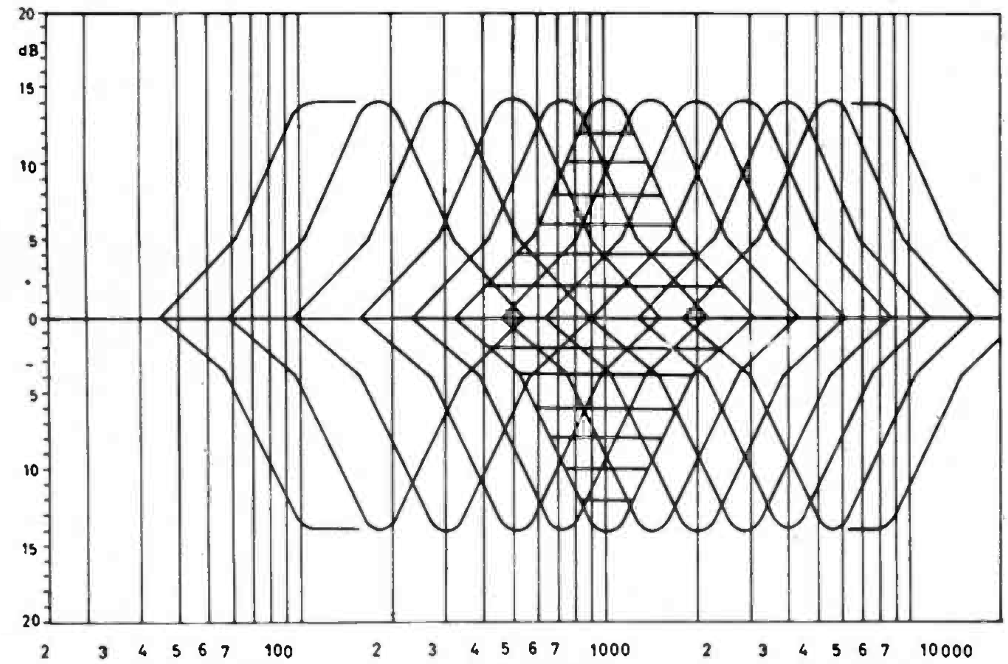
INSIDE VIEW OF EQ-1000

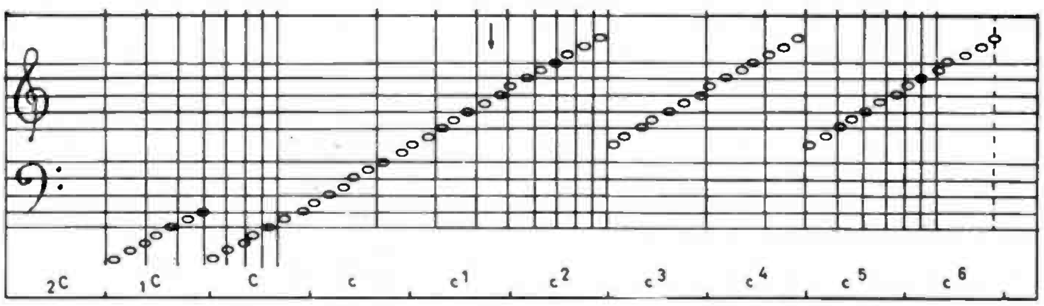
Boost Characteristics

Droop and Cut-off Characteristics

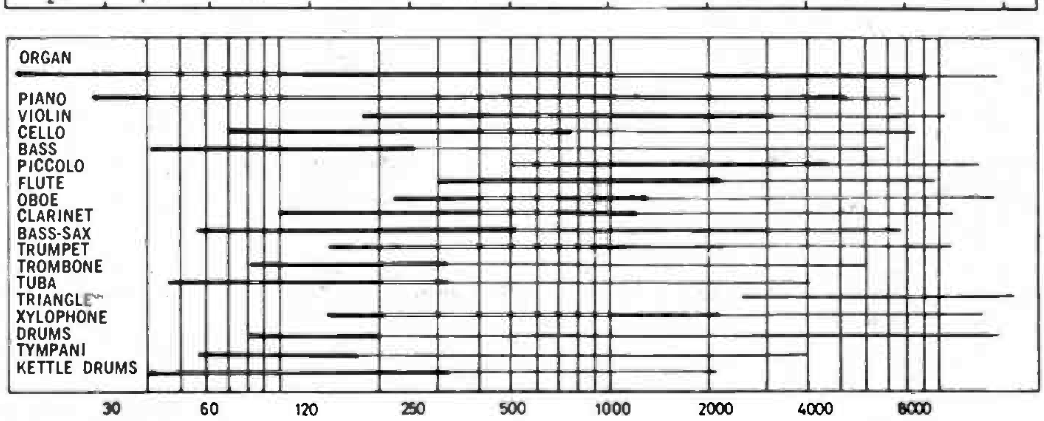


Band-Pass and Reject Characteristics





Tonal Range of Instruments Related to EQ-1000 Equalization Curves



EQUALIZING FREQUENCIES, SLOPES AND CORRECTIONS:

Low frequency BOOST:	freq:	85, 170, 340, 680 cps.
	slope:	6 or 12 dB/octave.
	dB steps:	0, +3, +6, +9, +12, +15, +18, +20 dB.
Low Frequency DROOP:	freq:	60 cps.
	slope:	6 dB/octave.
	dB steps:	0, −3, −6, −9, −12 dB.
Low frequency CUT-OFF:	freq:	40, 60, 85, 125 cps.
	slope:	12 or 24 dB/octave.
	dB steps:	Fixed
Low frequency BAND-PASS: (or band-reject)	$f_{takeoff}$:	60, 110, 180, 300, 420, 600
	f_{center}:	180, 300, 500, 700, 1000
	f_{return}:	300, 500, 850, 1200, 1700, 2400 cps.
High frequency BAND-PASS: (or band-reject)	$f_{takeoff}$:	600, 850, 1200, 1700, 2400, 3200
	f_{center}:	1400, 2000, 2800, 4000, 5600
	f_{return}:	2400, 3200, 4800, 6000, 9000, 13,000 cps.
High frequency CUT-OFF:	freq:	6.5, 8, 10, 12 KC.
	slope:	12 or 24 dB/octave.
	dB steps:	Fixed.
High frequency DROOP:	freq:	10 KC.
	slope:	6 dB/octave.
	dB steps:	0, −3, −6, −9, −12 dB.
High frequency BOOST:	freq:	2.5, 4, 6.5, 10 KC.
	slope:	6 or 12 dB/octave.
	dB steps:	0, +3, +6, +9, +12, +15, +18, +20 dB.

TECHNICAL SPECIFICATIONS:

Frequency response in LINEAR position:	20 – 20,000 cps ± 0.25 dB.
Input impedance:	5000 ohms balanced.
Output impedance:	600 ohms.
Total RMS harmonic distortion:	at + 18 dBm output: 50 cps 120 cps 1 KC 5 KC 10 KC 0.3% 0.22% 0.25% 0.3% 0.38%
Maximum output level (1000 cps)	+ 22 dBm < 0.7% RMS distortion.
Total intermodulation distortion:	less than 1% 50/7000 (4:1) at + 18 dBm.
Noise level (absolute):	Unweighted: − 78 dBm (100 μV). Weighted CCIR: − 93 dBm (20 μV).
Phase distortion:	Low frequencies < 10%; high frequencies < 2%.
Accuracy of discrete frequencies:	± 1 dB.
Accuracy of dB settings:	± 1 dB.
Amplification available:	0 dB or 5 dB gain; switchable.
Insulation resistance:	> 10 Megohm chassis to 0 Volt.
Tube complement:	(9) 12AT7 (4) 12AX7 (1) 6689
Power requirement:	117/220 Volts; 50-60 Cycles; 66 Watts.
Overall dimensions:	21½" w. x 9½" h. x 14" d.
Weight:	48 lbs. net

For further information, contact Gotham Audio Corporation or your dealer:

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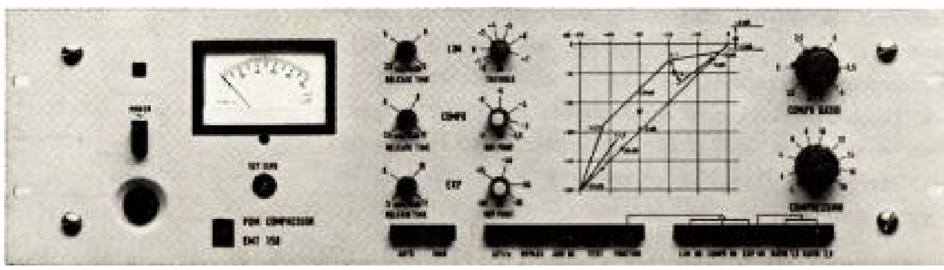
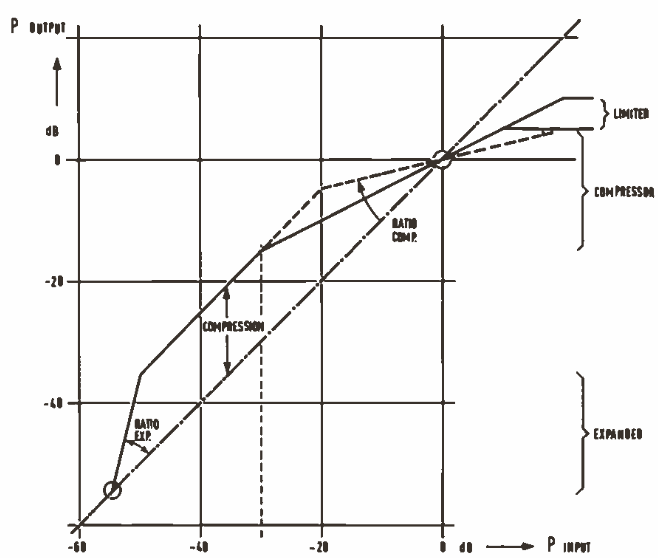
PDM-COMPRESSOR EMT 156



The pulse-duration-modulation (PDM) which is used in the compressor EMT 156 is a new method in this field. It provides an elegant technical solution to the problem of controlling the compressor characteristics. The PDM compressor EMT 156 is therefore particularly adaptable to the varying demands in practical operation. It can according to requirements work as compressor, limiter or a combination of the two. The range of possible variations of its characteristics in the three above mentioned functions is shown in the diagram 'static characteristics'.

Furthermore, the recovery time can be made short, long or a function of the programme material. A further special feature of the PDM compressor is that it avoids the usual rise in background hiss during signal pauses.

FUNCTIONS	VARIABLE CHARACTERISTICS
Limiter	Limiting threshold
Compressor	Threshold of compression Compression ratio Compression Expansion threshold for low levels Expansion ratio for low levels
Limiter and compressor	As under limiter and compressor
External control	The gain becomes a function of an external DC voltage
0 db gain amplifier	——



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TYPICAL APPLICATIONS

LIMITER:

Protection of AM-transmitters and to achieve a higher average modulation of the transmitter during news bulletins (Range). In general with speech to prevent transient overload.

Special effects in radio plays and for protecting lines and amplifier systems.

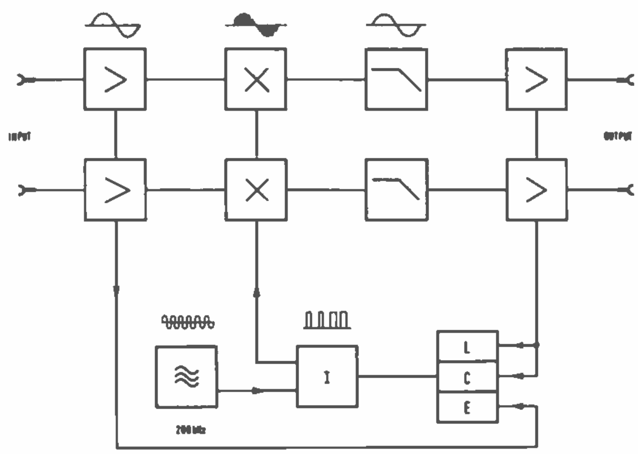
COMPRESSOR:

Equal and simultaneous compression and limitation of both channels in stereo systems.

For archieving a 'dense' sound in particular with dance music. For increasing the range of AM-transmitters and for compensating tone sources with different apparent loudness for special effects.

PRINCIPLE OF OPERATION

The input signal goes via a pre-amplifier to a modulator X where it is cut up by means of a 200 kc/s square wave with variable mark-space ratio. The subsequent low-pass (integrator) re-assembles the signal into its original shape. But its level is now a function of the mark-space ratio of the control signal, in other words it depends on whether the multiplier was cutting wide or narrow sections out of it. The control signal is derived from the output via a logarithmic amplifier and after rectification the time constants are formed as well as the functions: Limiting, Compression, and Expansions. The DC-voltage corresponding to these functions now controls the mark-space ratio in the information unit J (pulse-duration-modulation) of the 200 kc/s pulse signal which cuts the input signal up in the modulator as described above.



VERSIONS

19" rack mounting (ASA) with one audio amplifier system mono, with two audio amplifier systems stereo.

PROVISIONAL TECHNICAL DATA

Input	
Level	+6 db
Impedance	10 k-ohm
Output	
Level	+6/+12/+18 db
Impedance	37.5/150/600 ohms
Limiter	
Threshold	adjustable from +4 ... +12 dbm
Attack time for 10 db over modulation	app. 50 μs
Overload margin on input	+20 dbm
Compressor	
Compression	adjustable from 0 ... 20 db
Compression ratio	adjustable from 1:1 ... 1:3
Attack time	app. 3 ms
Recovery time	short, long or automatic
Amplifier	
Frequency response, 30 c/s ... 15 kc/s	± 1 db
Distortion at 1 kc/s in static condition	max. .5 %
Goin control through external voltage	0 .... —30 db
Suppression of the of the 200 kc/s carrier	min. 70 db
Dimensions	483 x 133 x 310 mm deep

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AUDIO DELAY UNIT

EMT 970

AUDIO DELAY UNIT
EMT 970

As the ideal transmission of sound with regard to volume, high fidelity and dynamics, is nowadays only restricted mainly by obstacles of a commercial nature, successful attempts have been made in the field of further improvement with a view to transmitting also three-dimensional components and directional parameters of the actual tone. This highlights the phenomenon of the effect of sound transmission time on the human ear.

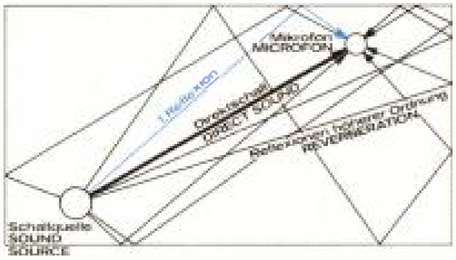
The professional sound recording and reproduction technology now has to be in a position to manipulate transmission times. Audio Delay Unit EMT 970 is an instrument which makes it possible to delay secondary audio-frequency signals by given periods in relation to the primary signal. The transmission time settings have been specially chosen to suit applications in electro-acoustics and studio technology. They range from 25 to 250 milliseconds and thus correspond to the transmission time of sound in sound recording and reproduction methods, and are of significance as regards auralphysiological processes.

The ability of the Audio Delay Unit EMT 970 to produce or compensate sound transmission time opens up a whole series of interesting applications.

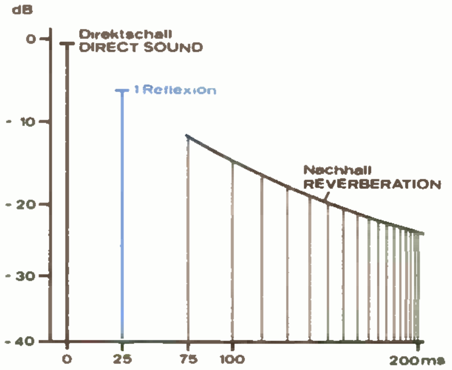
Applications

Delay of Reverberation and introduction of the first reflection

The acoustic impression an observer gains of a room is primarily dependent on the time sequence in which direct sound, first reflection, and reverberation appear. The greater the reverb delay with constant reverb decay time, the larger the acoustic impression of the room. Unfortunately this delay destroys the room's "transparency". Inserting the first reflection between direct sound and reverberation preserves, even for a large room, its acoustic "transparency". This first reflection gives information concerning the geometric dimensions of a room (i.e. long and narrow or very high), and within limits, even the treatment of its walls.



The greater the number of sound wave reflections, the longer is their delay compared to the direct sound. The time intervals between the individual reflections arriving at the microphone become ever shorter; in other words their density increases. At the same time these reflected sound waves lose energy, and this in turn results in an exponential loss of total sound level within the room as a function of time: reverberation results.



The sketch shows how the first reflection and reverberation are delayed compared to the direct signal and thereby forms the acoustical characteristic of a room.
Room reverberation is best created by the well known EMT 140 Reverberation Unit. Delaying its input using the EMT 970 Audio Delay Unit permits simulation of a particular acoustic enclosure.
The first reflection, which is also a determining factor in room character simulation, can likewise be generated by the EMT 970, by delaying the direct signal.

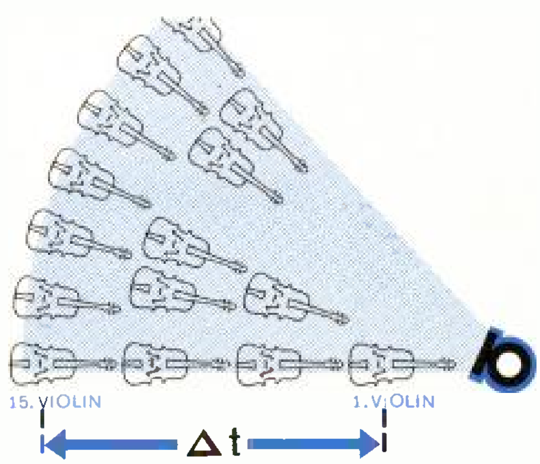
Effects

ECHO EFFECTS

Popular music production and certain effects in radio dramas oftimes require echo effects which may either consist of a single reflection or a number of diminishing echoes. Delay times of 125, 150 or 250 ms produce just such echo effects. To achieve such effects, the output signals of the particular playback amplifiers are fed through level controls which are adjusted to produce an exponential decay of the individual echoes.
To produce multiple repeating echoes, these controlled output signals are returned to the Delay Unit input, with the result that they theoretically decay exponentially indefinitely.

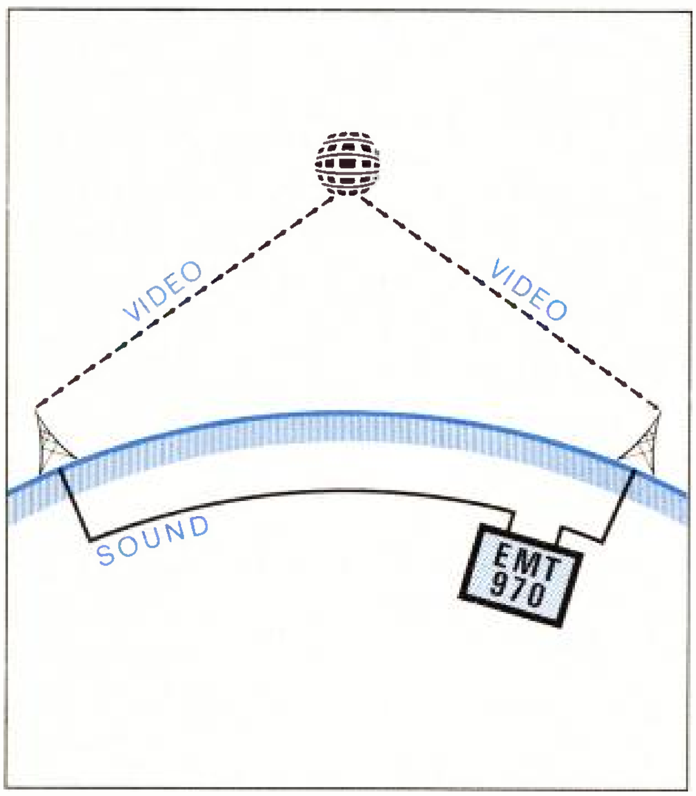
"ENLARGEMENT" OF A SMALL STRING GROUP

The output of the string microphone is fed through the minimum delay time of the EMT 970 and returned with greatly reduced level to the direct signal channel. This produces a certain "lack of precision" which is the typical hallmark of a large string section. When using a delay time of 25 ms there is no danger that the string sound will acoustically fall apart into two distinct signals.



Satellite transmissions

During TV transmissions beween continents, both video and audio signals are sent via satellite and arrive simultaneously at the receiving station. At times, when problems in audio transmission arise, undersea cables are used as an alternate feed. In such cases the audio signal arrives approximately 250 ms ahead of the video signal and must be delayed to bring the program back into synchronism. The EMT 970 Delay Unit is admirably suited to use for such delay purposes.

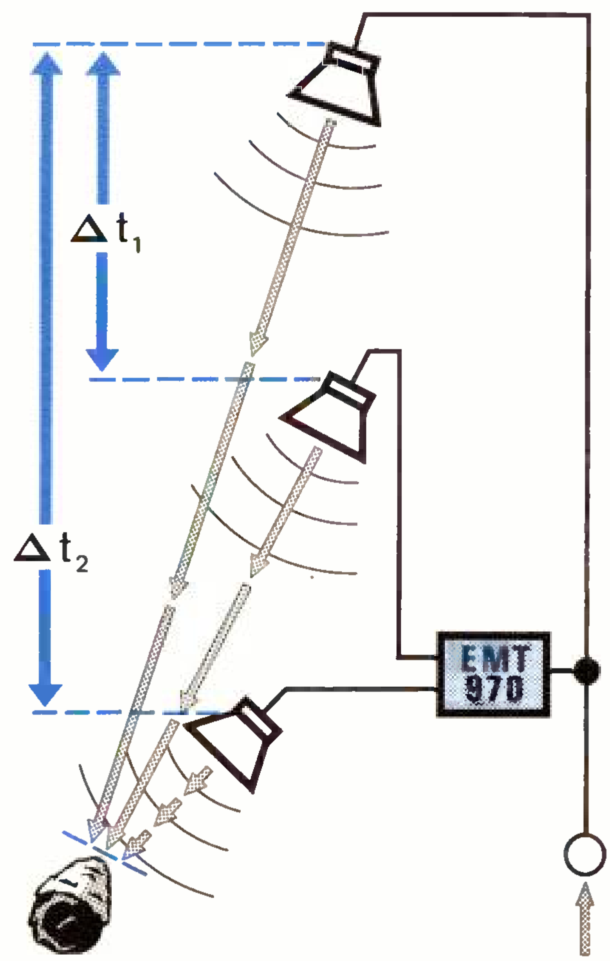


Foreign language dubbing of films

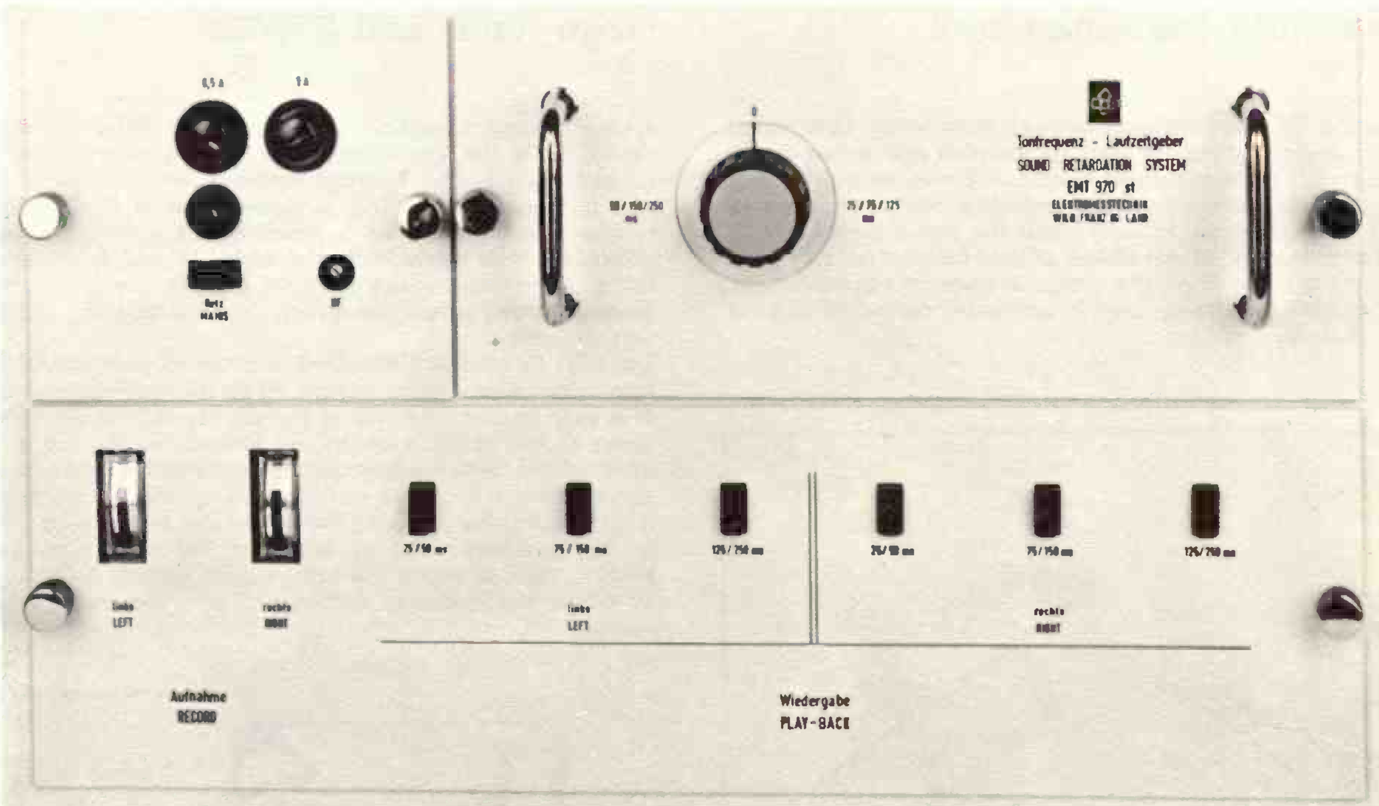
The dubbing of foreign language films produce problems of lip synchronizing the speaker with existing picture. Dependent on individual reaction time, this may produce word start delays of varying lengths which may easily be compensated by delaying the dubbed language via the EMT 970 Audio Delay Unit.
Similar problems are encountered in post-syncing of TV operas.

Loudspeaker installations in large halls and arenas

A very important application for the EMT 970 in electroacoustics is the improvement of intelligibility of loudspeaker and public address installations in large halls and outdoor arenas. The problem here is that sound reaches the listener, as a result of his widely varying distances from loudspeaker groups, at widely varying times. In certain cases multiple echoes are produced which seriously affect the quality and intelligibility of the performance.
The EMT 970 Audio Delay Unit is useful in compensating these disturbing sound delays. In so doing it makes use of a very unique property of human hearing: In the presence of two sound sources, we always localize the one whose wave front reaches our ears first, even if the second arrival, within about 35 ms time, is of higher intensity (Haas effect). By delaying the sound in a meaningful way to the speakers used to reinforce the original sound source, we can make the total sound picture clear and its origin unmistakably coincident with the live sound source.



Technical description



The unit consists of a series of plug-in units contained within a single, stable frame. The stabilized power supply and mechanical plug-in units are in the top half of the unit, while the individual amplifier cards made of glass-epoxy are plugged into the lower half card carrier. All amplifiers are equipped exclusively with silicon planar transistors.

The mechanical system is constructed as follows:

An extremely thin, pliable and tensilized Mylar foil is stretched like a drum head over a rotating turntable-like ring. There are no splice gaps as may be found in endless tape loops. The foil is coated on its underside. Since the magnetic heads ride spring loaded on the uncoated, smooth Mylar surface, they practically do not wear, and above all make excellent and constant contact with the magnetic foil. A stationary anti-static cloth resting on the surface of the foil prevents any accumulation of dirt. The heads themselves are mounted on three points, are alignable and are mu-metal shielded. They are mounted on cast supports which are screwed to the deck of the mechanical unit which, with the main bearing, forms one stable assembly. The disk is rim driven by a friction idler not unlike a turntable, while a two step motor pulley permits a 2:1 change of rotational speed. The motor is fully mu-metal shielded.

SIGNAL PATH

The audio signal at the unit's input is fed via the record amplifier and record head to the magnetic drum. The signal then runs through an arc past three playback heads which results in three separate time delays. These delayed signals are each fed through individual equalized playback amplifiers to push buttons, by means of which they are selected to form an output buss. The playback amplifier outputs are further brought out to a multi-pin connector at the back of the unit to permit a variety of interconnections depending on the application intended. It is at this point that the unbalanced playback amplifier outputs may also be connected individually and unmixed to other units. In the case of the standard unit, these three playback amplifier outputs are combined and fed to the input of a line amplifier from whose line level output the delayed signals may be then fed to other facilities.

STEREO-VERSION EMT 970 st

In the stereo version of the unit there are two complete sets of amplifiers.

Technical specifications

Head complement	1 erase head 1 record head 3 playback heads
Linear track speeds	90 cm/s and 45 cm/s (app. 35 and 17½ ips)
Delay periods at 90 cm/s at 45 cm/s	 25 ms, 75 ms, 125 ms 50 ms, 150 ms, 250 ms
Flutter content (weighted according to CCIR std) at 90 cm/s at 45 cm/s	 max. ± 0.2% max. ± 0.4%
Level in stability	max. 1 dB
Frequency response at 90 cm/s at 45 cm/s	 40 Hz – 16 kHz } +2 dB 40 Hz – 12 kHz } – 3 dB
Tot. Harm. Distortion at 1 kHz at 250 Hz	 max. 3% max. 3%
Input Nominal level Level range Impedance	balanced & floating +4 dBm, +6 dBm, +8 dBm 0.6 V – 9 V (+21 dBm) min. 2000 Ohms
Output Nominal level Max. level Impedance	balanced & floating +4 dBm, +6 dBm, +8 dBm +18 dBm at 600 Ohms approx. 30 Ohms

Stereo Channel Separation at 1 kHz 100 Hz – 10 kHz	 min. 40 dB min. 30 dB
Phase angle error at 10 kHz	 max. 30°
Erase attenuation	min. 70 dB
S/N Ratio (unweighted) at 90 cm/s at 45 cm/s	 min. 47 dB min. 47 dB
S/N Ratio (weighted) at 90 cm/s at 45 cm/s	 min. 56 dB min. 58 dB
Bias oscillator freq.	65 kHz
Bias suppression at output	min. 40 dB
Magnetic field at 2" from unit	max. 15 mGauss
Acoustical noise directly at the unit	max. 40 Phon
Dimensions Rack width Panel height Distance behind panel	 19" 10½" max. 18"
Weight EMT 970 st	 33.3 kg (73 lbs.)

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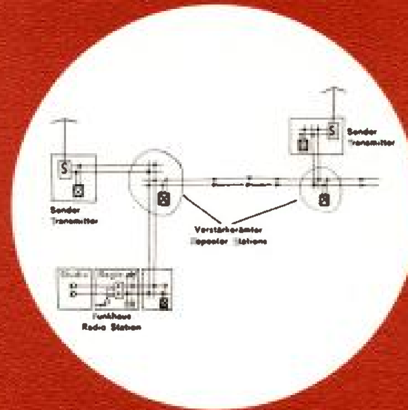
STEREO MONITOR
EMT 159
Pause
INTERVAL IN TRANSMISSION
Rechts fehlt
RIGHT NO RESPONSE
EMT
Links fehlt
LEFT NO RESPONSE
Seite fehlt
NO SIDE RESPONSE
Verpolt
CHANGED POLES

Stereo-Monitor EMT 159

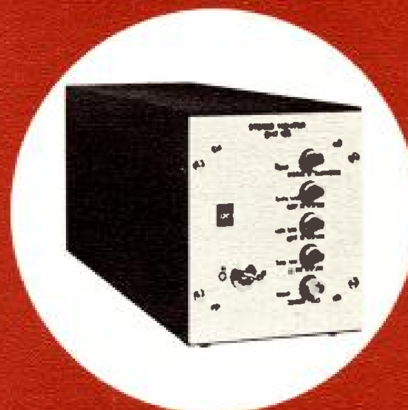
For the operational monitoring of stereo programmes at any point of the programme chain:

IN THE STUDIO · IN REPEATER STATIONS AND SWITCHING CENTRES · AT THE TRANSMITTER



The continuous monitoring of the stereophonic character of programmes with conventional methods is very tedious, costly and furthermore subjective. The decision whether there is a fault in the transmission depends in the end on the reliability and skill of a person, not only with direct listening but also in the case of measuring processes involving meters or cathode ray oscilloscopes. The continuous monitoring of stereo transmissions in this way involves an unreasonable strain and is unreliable since the concentration of any man must falter with time. The stereo monitor EMT 159 overcomes these human failings. In particular this unit enables the personnel in repeater stations, switching centres and at the transmitter to keep a reliable check on the stereo characteristics of the signal without involving additional work. Warning lights will show the following faults in a stereo programme without the need to lift a finger:

	Colour of the lamp
1. No signal	Yellow
2. No signal in right-hand channel	Green
3. No signal in left-hand channel	Red
4. No difference signal	Blue
5. One channel out of phase	White



Since each colour corresponds to a specific fault, the indication is clear and unmistakable and the monitoring, therefore, simple.
The indication can furthermore be repeated on an external light panel and furthermore coupled with an acoustic warning signal. This enables for instance the simultaneous monitoring of several stereo programmes continuously and reliably without anyone having to concentrate on them all the time. The stereo monitor will indicate immediately if the stereo character of a programme has been lost. But it can also monitor two mono signals e. g. as a check at the transmitter and for checking tape copying systems the stereo monitor can be used advantageously.
The unit contains no mechanical contacts. It uses transistors and diodes exclusively.
A delay mechanism on the lamps prevents false alarms due to short pauses or very quiet passages. The delay periods and the triggering threshold have been determined in accordance with statistical sampling of programme material.

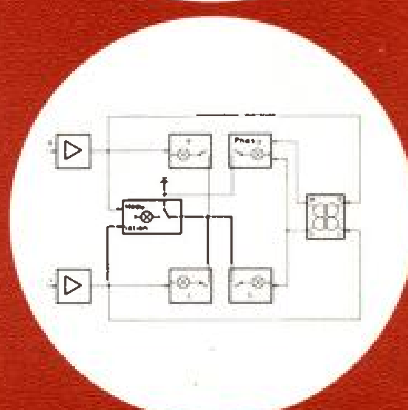
The stereo monitor is available in two versions:

Bench unit EMT 159
with mains switch on the front panel, mains cable (with plug) and plug and socket connection at the back of the unit.

Twin cassette EMT 159 K
Mechanical locking on the front panel, all connections via multi-pin plugs on the back of the unit. Without mains switch.
In one frame for standard racks in accordance with DIN 41 490 up to five twin cassettes can be housed.

Technical data

Inputs	two, balanced. One each for left and right-hand channel, connection to MS channels via external sum-and-difference network.
Input level	+ 6 db (1.55 V) peak programme level
Input impedance between 40 c/s and 10 kc/s	2.5 k-ohm, each channel
Ambient temperature	+ 5° C to + 50° C permissable
External pilot lamp supplies	five, 12 V at max. 100 mA
Mains voltage	switchable 100 . . . 125 V 50/60 c/s 200 . . . 250 V 50/60 c/s
Power consumption	8 W with one pilot lamp on.
Dimensions Bench unit Cassette unit	 140 x 112 x 265 mm cassette size II (134 x 94 x 256 mm)
Weight	app. 2.8 kg (6 lbs)



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6·d

MONITOR SPEAKER MODEL OY

A SUPERIOR INSTRUMENT DESIGNED FOR THE PROFESSIONAL

GOTHAM INTEGRATED STUDIO MONITOR SPEAKER MODEL OY

APPLICATIONS

The new Gotham Monitor Speaker Model OY is a wall-mounted loudspeaker with two built-in 30 Watt Silicon-Transistor-Amplifiers which permits highest-quality monitoring without the need for separate power amplifiers. As a result of its small size, it can be mounted in a most favorable position in control rooms, mix-down areas, editing rooms, disk-cutting laboratories, etc.

The Model OY Loudspeaker may be conveniently mounted next to the picture monitor in TV control rooms so that the sound seems to come from the direction of the picture, and all this without the bother of separate power amplifiers.

Recent advances in loudspeaker enclosure design has made it possible to achieve low-frequency reproduction even with cabinets of such small size. The Gotham Model OY Monitor Speaker at last makes possible quality monitoring for location recording and broadcasting as well as for TV remote vans where space is at a premium. All that the model OY requires at its input is a line level low impedance buss. The stringent requirements with respect to frequency response and distortion which studio monitors pose are completely fulfilled by the Model OY.

DESIGN CONSIDERATIONS

Design of high-quality loudspeakers always poses very fundamental problems. The loudspeaker should reproduce the audible range (40 Hz - 16 kHz) with a linear sound pressure curve (±2 dB). This means the loudspeaker must reproduce a frequency of 100 Hz with a wave length of 3.4 meters with the same sound pressure level as the frequency 10,000 Hz with a wave length of 3.4 centimeters. Loudspeakers that are able to do this with a single system are unlikely. As a result, most quality speaker systems involve at least two loudspeakers. For very high quality reproduction the frequency range is divided into high, mid and low frequency speakers. Dividing the speaker system into three loudspeakers has the advantage of low distortion and minimal intermodulation.

This is all easy to understand if one considers that a membrane of say 12 inches in diameter can hardly be expected to radiate both 40 Hz and 5 kHz simultaneously without forming frequency products which become audible and measurable as distortion. The proper balancing of three such loudspeakers to produce a linear sound pressure curve is not simple, since the loudspeakers each operate at different efficiency levels. Most of the time the mid and high frequency speakers must be attenuated to the level capability of the "woofer." The crossover frequencies of the filter must be so chosen that the resonant frequency of the mid and high-frequency systems falls outside of their respective transmission ranges. To accomplish this, most manufacturers design crossover networks with a filter steepness of some 12 dB/Octave. When such a crossover network is placed between the amplifier output and the loudspeakers, poor rise time is often the result. It stands to reason that a loudspeaker which is driven directly from the output of a transistor amplifier with a source impedance of 0.1 Ohm will produce better impulse distortion characteristics than one which is fed through a crossover network consisting of coils and condensers.

THE NEW LOUDSPEAKER CONCEPT OY

The Gotham Model OY Loudspeaker (manufactured by Klein & Hummel, West Germany) was developed in an effort to avoid the pitfalls of the aforementioned problems. The Model OY has low, mid and high-frequency loudspeakers. The crossover frequencies are 500 Hz for the woofer, 500 Hz and 6 kHz for the mid range speaker, and 6 kHz for the high-frequency horn. The woofer on the one hand, and the mid and high-frequency loudspeakers on the other, are fed from separate transistor power amplifiers, 30 Watts sine wave power each. The crossover between these two amplifiers is accomplished by means of electronic filters prior to the output stages of the power amplifiers (voltage stage only) with 12 dB/Octave roll-off. The low-frequency woofer is connected directly to the transformerless output of the low-frequency amplifier. Frequencies above 500 Hz are filtered out of the low-frequency amplifier while the mid/high-frequency amplifier is not fed any frequencies below 500 Hz. Therefore, intermodulation products can hardly appear. At the output of the mid/high-frequency channel there is a simple crossover network which couples the high-frequency horn to the mid range speaker at the take-over frequency of 6 kHz.

Proper matching of the different efficiencies of the loudspeakers is now a simple matter, as is the equalization of each loudspeaker to produce a linear response curve overall.

The equalization is accomplished at the input to the OY Speaker System using a special pre-amplifier stage. As can be seen from these explanations, the use of a 2-channel amplifier system with electronic crossover brings with it many advantages which heighten the quality of such a speaker system.

DESCRIPTION OF THE MODEL OY LOUDSPEAKER SYSTEM

The Gotham Model OY Studio Monitor System is contained entirely within a housing 19" wide, 12" high and 9" deep. The very solidly-built wooden enclosure is covered with chip-proof Formica. The front of the speaker is covered with a removable metal mesh grille cloth. A switch accessible from the outside permits the low-frequency system to be properly adjusted for the particular mounting place of the OY system.

It is a well-known fact that low-frequency radiation depends to a great extent on the actual positioning of the loudspeaker. High-frequency reproduction, on the other hand, is a function of the reverberant nature of the listening room. For this reason the Model OY has a four-step, high-frequency control in addition to the low-frequency control mentioned earlier.

An input attenuator allows gain control of the entire system. There are two paralleled input connectors which facilitate the interconnection of several such loudspeaker systems into loudspeaker groups.

There are separate trimming potentiometers which are mounted concealed, to permit accurate adjustment of the frequency response of the entire system, but these should only be adjusted when laboratory acoustical measurement facilities are available.

The combination of directly-fed loudspeakers from a source impedance on the order of 0.1 Ohms and the drastic reduction in inductances and capacitances are the cause of the exemplary impulse distortion performance of the OY Loudspeaker System and the practically unmeasurable ringing and overshoot.

OSCILLOSCOPE PHOTOGRAPHS OF THE SQUARE WAVE RESPONSE OF THE GOTHAM MODEL OY STUDIO LOUDSPEAKER SYSTEM

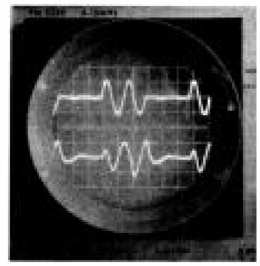
Display using a dual trace Oscilloscope, photographed with Polaroid Camera.

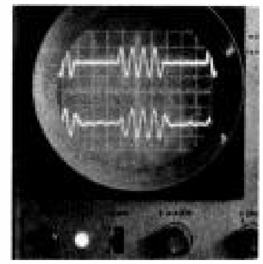
UPPER TRACE Audio signal at the input to the OY

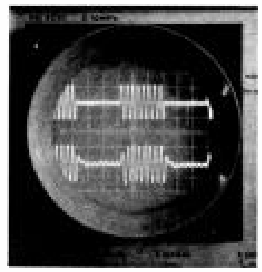
LOWER TRACE Audio output of a measurement microphone located in front of the OY Loudspeaker System

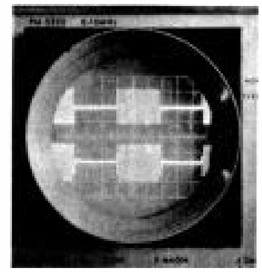
Grid Spacing	=	10 milliseconds
Impulse Duration	=	30 milliseconds
Repetition Rate	=	30 milliseconds

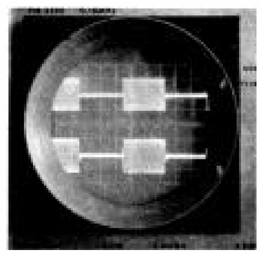
The pictures show modulation frequencies between 60 Hz and 16kHz.

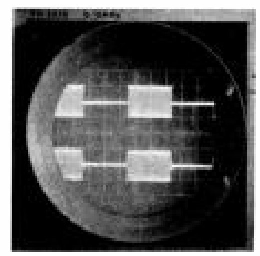

60 Hz 10 ms

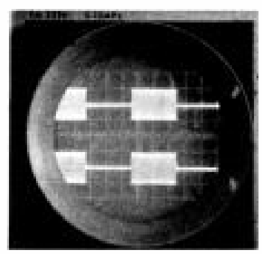

120 Hz 5 ms

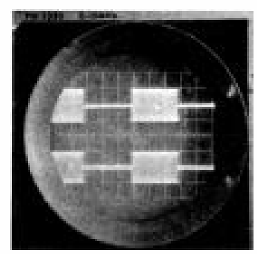

240 Hz 3 ms


480 Hz 2 ms


1 kHz 1 ms


2 kHz 0.5 ms

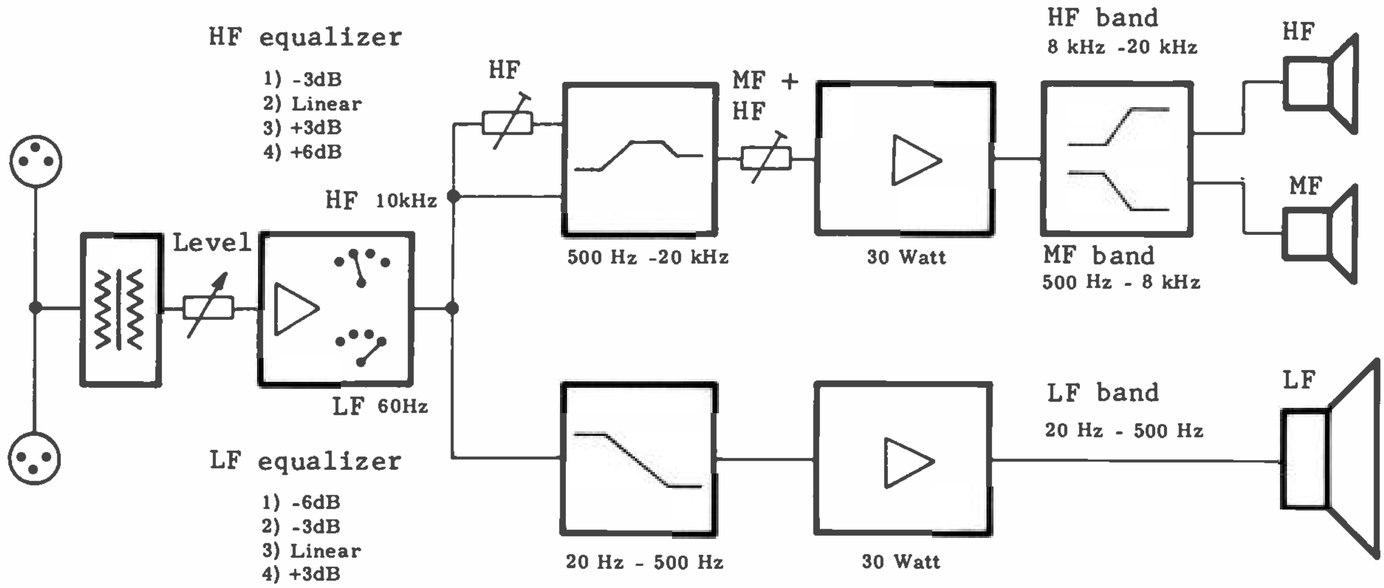

4 kHz 0.25 ms


16 kHz 0.1 ms

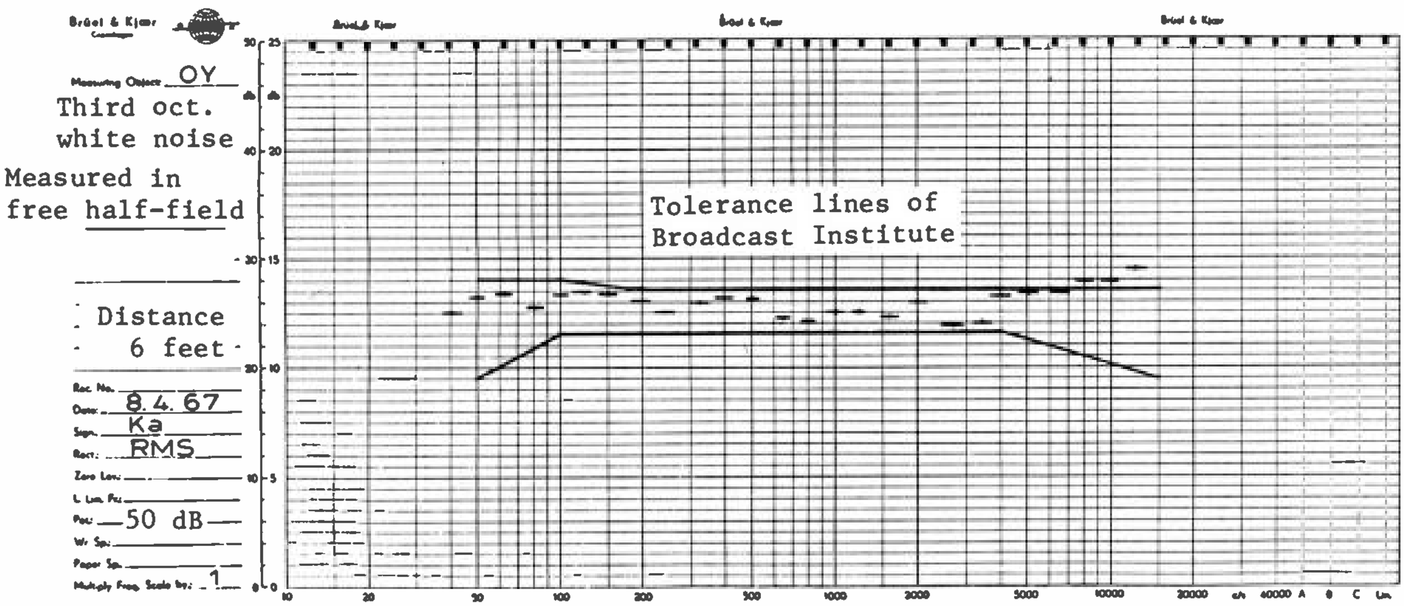
TECHNICAL DATA OF GOTHAM "OY" SPEAKER

Sound pressure level (at 1 meter distance):	107 Phon (curve B)
Freq. range:	30 - 20 kHz
Freq. response (measured with third octave white noise):	40 - 16 kHz ±2 dB
Self-noise level at 3 ft:	10 Phon
Dynamic range:	$>$90 dB

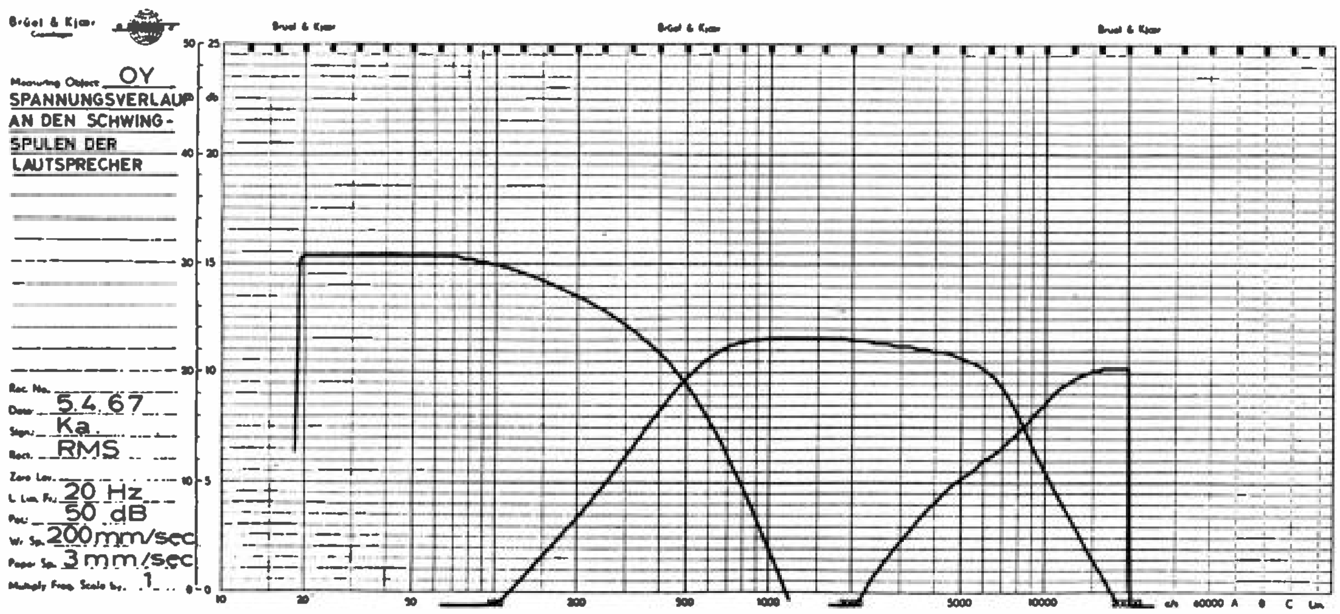
(continued on page 4)



Block schematic of the Model OY Monitor Loudspeaker System



Level recorder diagram of free field response using third octave band white noise.



Voltages measured on each of the three voice coils after system has been aligned.

Dispersion angle:	± 30 degrees
Rise & decay times:	10 ms @ 60 Hz 5 ms from 100 Hz 2 ms from 500 Hz 1 ms from 1 kHz
Tot. harm. distortion:	< 1% (mid-range)
Crossover frequencies:	500/8 kHz (elec) 500/6 kHz (acou)
Total enclosure volume:	34 liters (1.2 cu. ft.)
Dimensions:	19" x 12" x 9"d.
Weight:	44 lbs.

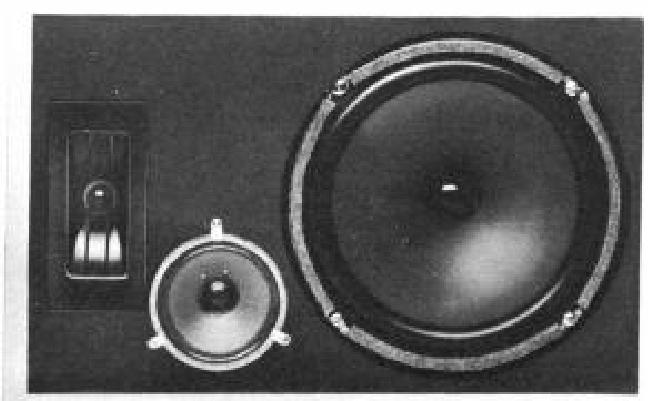
TECHNICAL DATA FOR "OY" AMPLIFIERS

Input level:	-6 dB minimum
Input impedance:	> 4700 Ohms; bal. & floating
Output power per channel:	30 W. sine wave into 4 Ohms
Output source impedance:	approx. 0.1 Ohm both chan.
Frequency range Low freq. channel: Mid-high channel:	 20 - 500 Hz 500 - 20 kHz
Low freq. control:	1. -6 dB 2. -3 dB 3. Linear 4. +3 dB
High freq. control:	1. -3 dB 2. linear 3. +3 dB 4. +6 dB
Steepness of crossover:	approx. 12 dB/ octave
Total harm. distortion:	< 0.25% in both channels at 30 W. into 4 Ohms over entire range
Unweighted noise Voltage:	< 0.7 mV across 4 Ohms
Weighted noise Voltage:	< 0.5 mV across 4 Ohms
Correctional controls for freq. response:	concealed trim pots: a) trimmer: mid/ high adjust relative to low freq. chan. b) trimmer: high freq. channel
Input connections:	two parallel 3-pole XLR type
Transistor complement:	22 silicon transistors 1 silicon rectifier
AC power:	117/220 V 40-60 Hz
Idling power consumption:	approx. 15 W

With the exception of the input transformer, both amplifiers are completely devoid of any interstage or output transformers.

All connectors, AC cords, designations, schematics and instructions are U.S. Standard.

FINISH: Available in either high impact light grey Formica or oiled prime walnut. Price identical for both types.



Model OY Speaker with front grill removed

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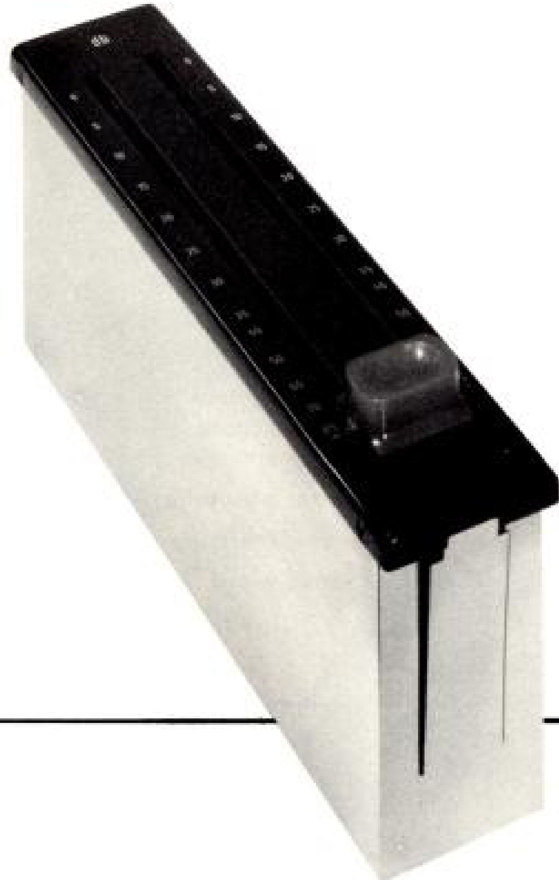
GOTHAM LINEAR ATTENUATORS – TYPE KW

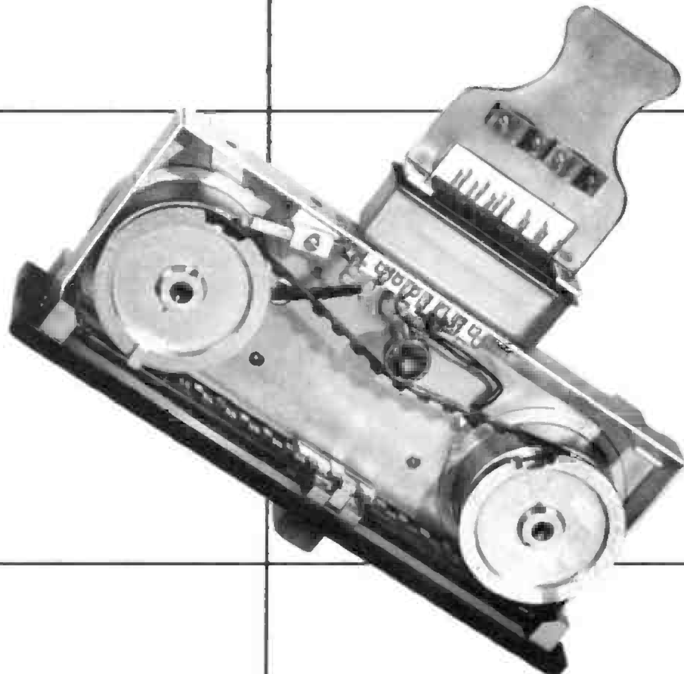


Linear Attenuators (Straight Line Mixers) are a fairly recent addition to the custom studio console field. In the short years of their use here they have found large scale acceptance due to the operating ease they afford the engineer. They have also, at the same time, produced considerable problems due to noisiness, sticking, and space consuming size. Studio consoles on the European Continent have ALWAYS used LINEAR ATTENUATORS as long as studios have been in existence. Over 20 manufacturers make them, and the problems we are only now experiencing have been solved there long ago. One of the very best of these European makes is the one GOTHAM is offering here. Its newly developed silicone encapsulated precision contact path (no wirewound card) and its ability to couple up four elements without additional width are some of the advantages.

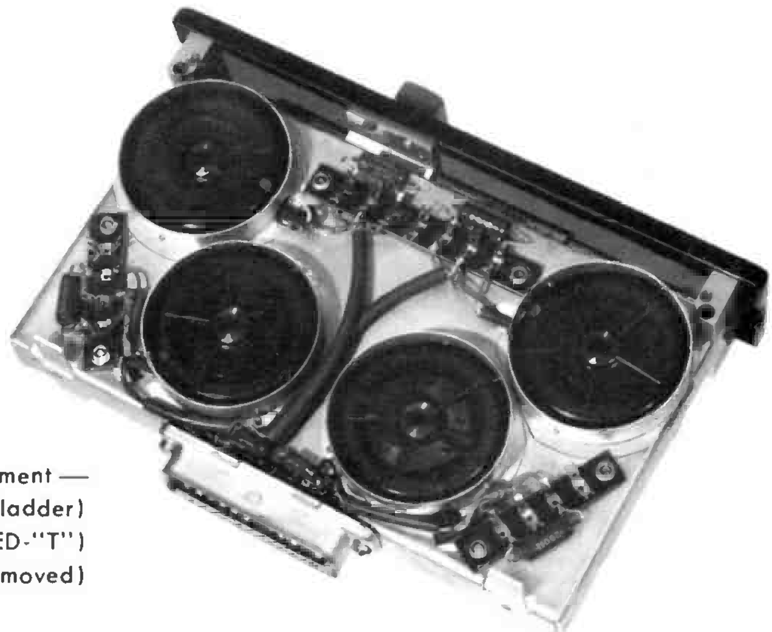
GOTHAM LINEAR ATTENUATOR ADVANTAGES:

1. Complete hermetically-sealed encopsulation of the contact elements in liquid silicone. Never requires servicing or cleaning.
2. LONGER fader travel — full 5¾" — makes for smoother operation.
3. 56-step construction makes it the most finely divided of any step-type attenuator.
4. Upper 60% of ottenuator trovel is considered operating range with 0.5 dB per step within this range. Increasing toper over bolance of travel provides a unique 85 dB of attenuation before infinity. This compares with a maximum of 55 dB before infinity for oll other foders, whether rotary or linear.
5. Constont, slightly viscous operoting force of 180-240 groms (oll types).
6. Avoilobility of one, two, three, and four-element foders WITHOUT odditionol width and only slightly additional depth below front ponel. Foders moy hove ony stondard configurotion—even potentiometers up to 250,000 ohms ovailoble.
7. Plug-in construction with retroctor handle for front occessibility.
8. Knobs color coded for chonnel identificotion, reodily changed.
9. Guoronteed ogoinst noise for five yeors!

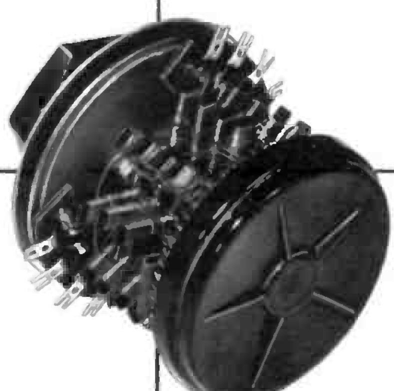
Gothom Single Element MIXER
(KW—lodder)
— cover removed —
with coble connector



Gotham Dual Element —
2-Channel MASTER (KW/2—ladder)
or single channel BRIDGED-"T" (TW/2)
— cover removed —
with cable connector



Gotham Four Element —
4-Channel MASTER (KW/4—ladder)
or 2-Channel MASTER (BRIDGED-"T")
(cover removed)



GOTHAM Silicone encapsulated 56-step attenuators are also available in rotary motion types. These are designated as Model "K" and are stocked as LADDERS and single element only. Other configurations available on special order. Standard mounting and shaft dimentions.

Illustration, above, showing construction.

Illustration, at right, showing sealed case.

TECHNICAL SPECIFICATIONS:

Frequency response:	20 — 20,000 cps ± 0 dB
Maximum input level:	+ 22 dBm
Available impedances: Potentiometers:	100 — 250,000 Ohms
All others:	100 — 1000 Ohms
Slide motion: One and two element:	glide bushing on honed steel rod.
Three and four element:	ball bearing sleeve.
Tracking for multi-element faders:	tightly coupled by toothed timing belt with imbedded steel strands against elasticity.
Finish:	black anodized aluminum with permanently etched lettering scale calibrated in dB. Plastic knob.
Dimensions: One and two element:	1-9/16" x 7-7/16" x 2-11/16" behind panel.
Three and four element:	1-9/16" x 7-7/16" x 4-1/8" behind panel.
Connection: One and two element:	12-pole rack connector. Mating plug with handle.
Three and four element:	23-pole rack connector. Mating plug. Separate transfer contacts standard in off position.

For further information, contact Gotham Audio Corporation or your dealer!

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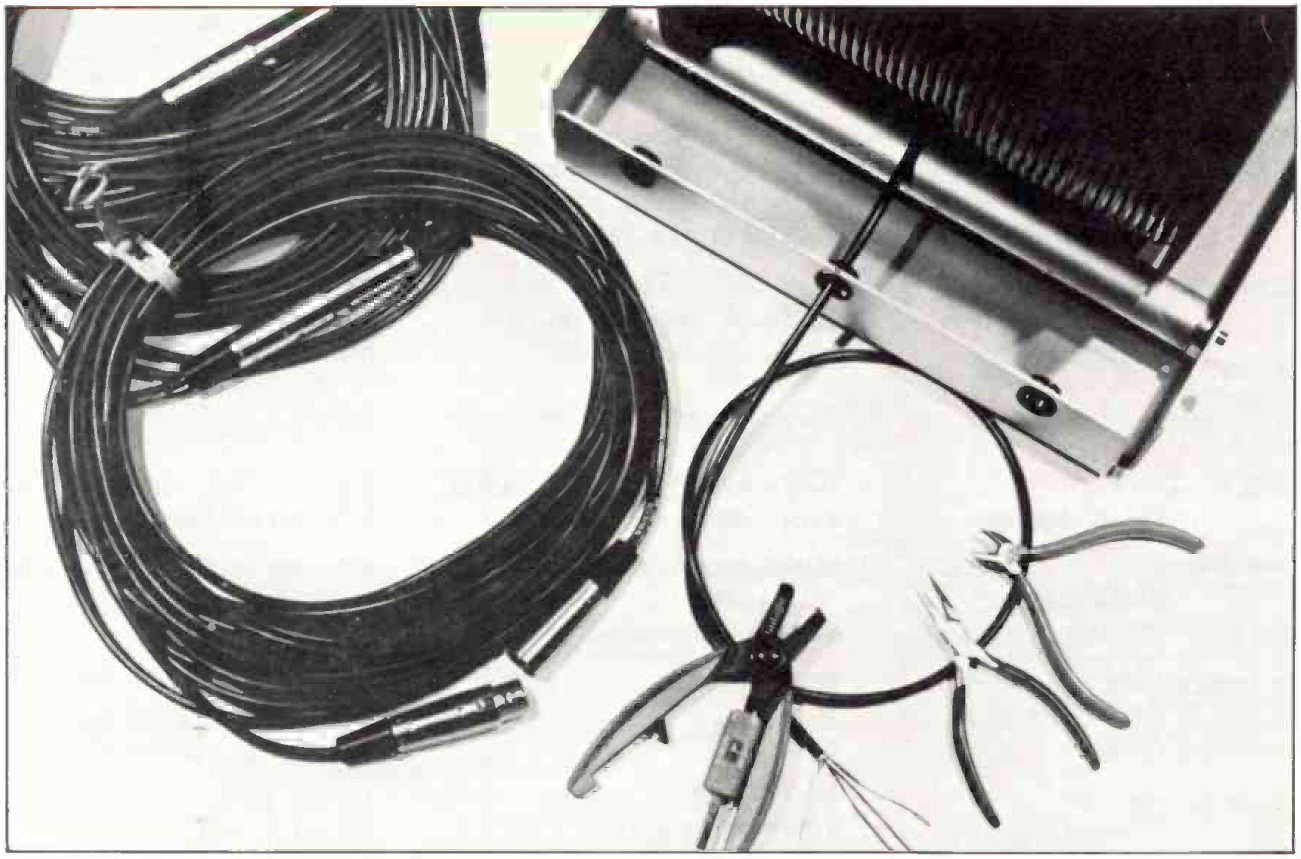

SUPERIOR GOTHAM AUDIO CABLES

Until Gotham became involved in audio cables, this important aspect of recording and broadcasting was left to a few large companies whose sights were aimed at the mass market. Such features as flexibility, resistance to cold weather, ease of stripping and soldering, color brightness but dull, matte finish, shielding to the ever increasing RFI fields, and last but not least, a neutral odor (!), were simply unavailable.

A look at commonly available cables shows that the highest number of individual strands used by any major manufacturer for each conductor is 37, while the 3-conductor cable listed here is made up of 96 strands of split-hair thin 0.05 mm diameter copper! This combined with two non-woven, Reusen layers of shielding provides both a high degree of RF rejection and suppleness.

With today's popular phantom powering, which requires a third conductor to complete the powering circuit, it no longer suffices to use the shield as this conductor. Any microphone cable with a broken shield is fine for use with dynamic microphones, but fails to operate phantom powered condenser microphones. To avoid this danger, a third conductor tied to the ground pin and shield on both ends will assure proper powering under all conditions at a minimal additional cost. Three-conductor cables also produce more uniformly round cables.

Color coded microphone cables, when combined with like colored foam pop screens permit easy identification in sound reinforcement situations using hand held microphones. It's easy to keep track of whose microphone is controlled by which console fader, simply by color coding the faders to match the cable and pop screen colors.

741 Washington Street, New York, NY 10014 • (212) 741-7411
West Coast Sales Office: Hollywood, CA • (213) 874-4444

GOTHAM

CABLE MAKE-UP:

	3-CONDUCTOR SHIELDED LIYSDDY 3x96x0.05mm	10 SHIELDED PAIRS LIYDDYY 10x2x0.19 mm^2
Each conductor:		
Gauge:	25 gauge AWG stranded	same
Strands:	96 x 44 gauge wire (0.05mm)	25 x 38 gauge wire (0.1mm)
Diameter:	0.49 mm (19 mils)	0.49 mm (19 mils)
Insulation:	VDE 207 YJ2	VDE 207 YJ1
PVC diameter:	1.2±0.05 mm (47 mils)	1.15±0.05 mm (45 mils)
Colors:	white, brown, green	red and white
Yarn Cover:	Rayon yarn; left serve (S)	none
Shield No.1:	Reusen: 0.1 mm (38 AWG)	Reusen: 0.1 mm (38 AWG)
Blank copper:	right serve (Z)	left serve (S)
Shield No.2:	Reusen: 0.1 mm (38 AWG)	Reusen: 0.1 mm (38 AWG)
Blank copper:	left serve (S)	right serve (Z)
Shielding cover:	95% minimum	95% minimum
Pair insulation:	N/A	VDE 207 YM2; PVC
PVC diameter:	N/A	3.3±0.1 mm (130 mils)
Colors:	N/A	according to resistance code
Outside cover:	Matte PVC	Matte PVC
Diameter:	4.9±0.15 mm (0.2")	15.2±0.2 mm (0.6")
Colors:	brown, red, yellow, green blue, white, ultra-black*	gray

ELECTRICAL DATA:

Conductor resistance:	max. 95 ohm/km (29 ohm/1000 ft)	
Insulation resistance:	18 Mohm/km (5.5 Mohm/1000 ft)	
Shield resistance:	max. 22 ohm/km (6.1 ohm/1000 ft)	
Operating capacitance: (at 800 Hz)		
cond-to-cond:	max.185 nF/km (56 nF/k ft)	max.200 nF/km (61 nF/k ft)
cond-to-shield:	max.300 nF/km (91 nF/k ft)	max.380 nF/km (116 nF/k ft)
Test Voltage:		
cond-to-cond:	500 V_{rms}	500 V_{rms}
cond-to-shield:	2000 V_{rms}	2000 V_{rms}

* Ultra black outside diameter is slightly larger.

GOTHAM Wow & Flutter Meter ME 101/2

The ME-101 and ME-102 instruments are intended for the investigation and checking of wow and flutter in all kinds of recording and reproducing devices, including tape and disk record/reproduce equipment, film equipment, etc.

They are all solid state, extremely easy and convenient to use and are therefore particularly suited to laboratory investigation, maintenance and service departments, and manufacturing checkout procedures on the assembly line.

A built-in oscillator provides the measurement frequency of 3150 Hz. For static calibration of the wow and flutter meter, this oscillator may be detuned +2%; for dynamic calibration it can be frequency modulated ±0.3% at the line frequency. Both these calibrations are available by means of push buttons on the front panel.

Pitch variations (flutter) between ±0.02 and ± 2.5% may be read unweighted or weighted as quasi peak values according to CCIR and DIN standards. This permits measurement of a wide range of devices from the highest quality studio recorder to a dictation machine. In its "fast" mode, the flutter meter's indications conform to the CCIR and DIN standards.

A second instrument, the "drift" indicator, measures the deviation of the mean value of the test frequency from its theoretical value. Both instruments may be switched to either "fast" or "slow" indication as preferred. Besides the normal input and output connections on the front panel of the instrument (standard instrument jacks), the back is provided with a standard DIN socket for direct connection to European home tape recorders. Other connections are provided for insertion of external filters, or for connection to oscilloscopes and pen recorders.

The two models ME-101 and ME-102 differ only in the measuring ranges they cover, the ME-102 being more sensitive and therefore more applicable to high quality professional equipment. The Model ME-101 reads to ±2.5% and is therefore more applicable for use in the measurement of home equipment and dictating machines.

NET PRICE: ME-101 $ 375.00 ME-102 $ 395.00 fob New York, N.Y.

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SPECIFICATION

(Specifications apply to both models unless otherwise indicated.)

OSCILLATOR SECTION:

Test frequency:	3150 Hz
Frequency stability:	1 x 10^{-3} after brief warm-up period.
Output voltage:	Approx. 0.4 V_{rms} at instrument jack on front panel
	Approx. 20 mV at "diode" cable socket at back of unit.
Calibrating facilities:	+ 2 % detuning for static calibration.
	ME-101 = ± 0.3%; ME-102 = ± 0.1% (60 Hz) for dynamic calibration.

WOW & FLUTTER METER:

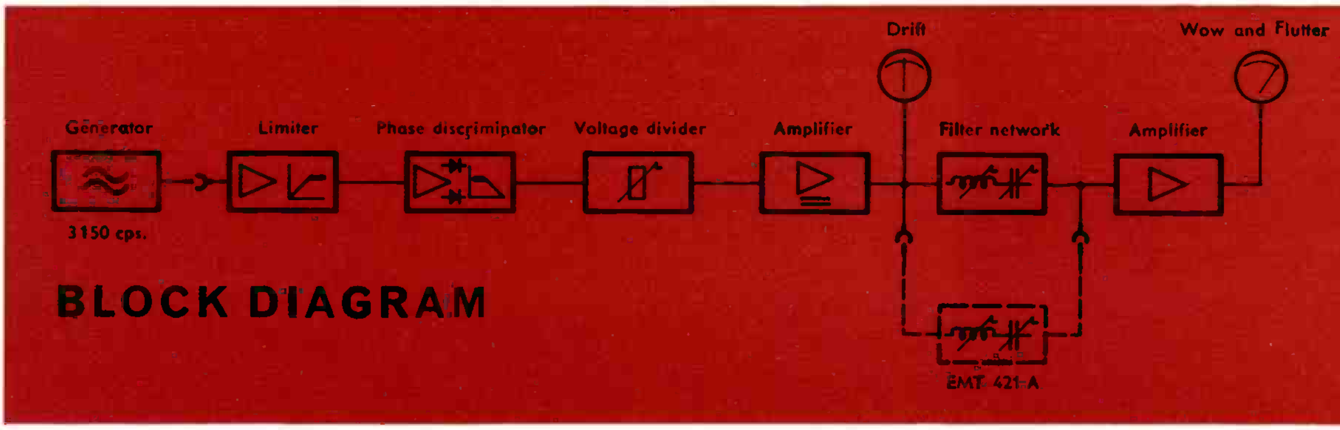
Input voltage:	30 mV to 30 V, 3150 Hz ± 5% without gain adjustment.
Input impedance:	10 kOhms.
Measuring ranges: (full scale) ME-101:	± 0.5% & ± 2.5% (lowest readable flutter: ± 0.02%)
ME-102:	± 0.15% & ± 0.75% (lowest readable flutter: ± 0.01%)
Indicating mode:	Quasi peak value according to CCIR and DIN standards.
Frequency response:	Unweighted: 0.5 to 500 Hz (-3 dB points).
	Weighted: according to DIN; external filter may be used.
Drift indication:	max. ± 4.5%
Output for oscilloscope or high speed pen recorder:	
ME-101:	approx. 20 V P/P (source impedance 22 kOhms).
ME-102:	approx. 10 V P/P (source impedance 22 kOhms).
Controls: Push buttons:	ON-OFF; TEST (connects oscillator directly to input of meter); + 2%; ± 0.3% (± 0.1%); WEIGHTED/UNWEIGHTED.
Control knobs:	Range switch: 2 pos. each "FAST" and "SLOW". Tuning control: zero centering of drift meter.
Transistor complement: ME-101:	10 germanium, 2 silicon, 10 diodes, 2 Zeners.
ME-102:	12 silicon, 10 diodes, 2 Zeners.
Power requirement:	110-125 or 220-240 Volts switchable; 50/60 Hz; 8 VA.
Dimensions:	11 3/4" w. x 8" h. x 7" d.
Weight:	net: 11 lbs. 4 ozs.; shipping weight: 13 lbs. 4 ozs.

All data subject to change without prior notice. Equipment manufactured in W. Germany. Printed in USA.

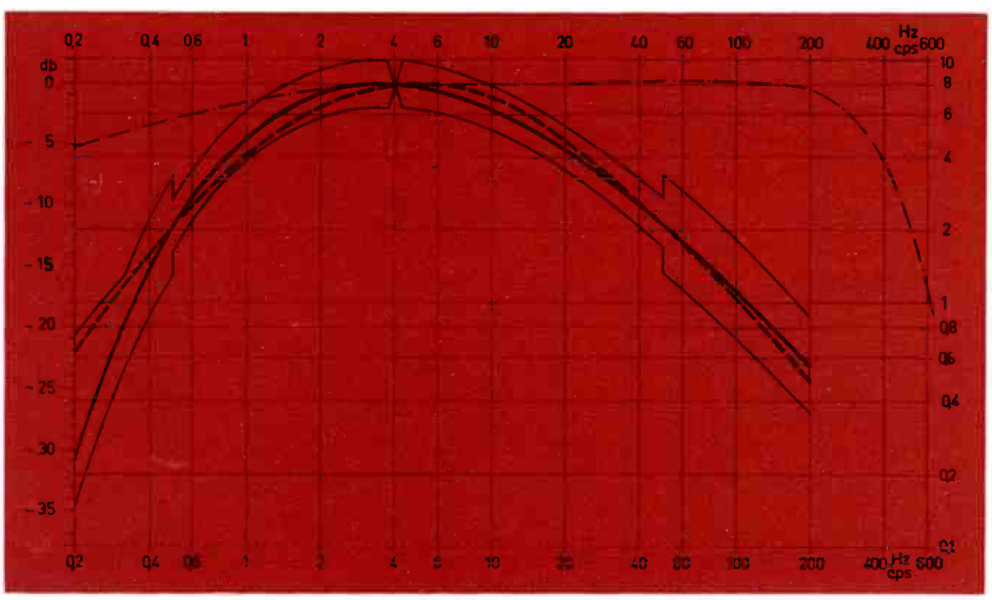
Wow and Flutter Meter

EMT 420

EMT 420 A



EMT 420A WOW AND FLUTTER METER AND ANALYSER



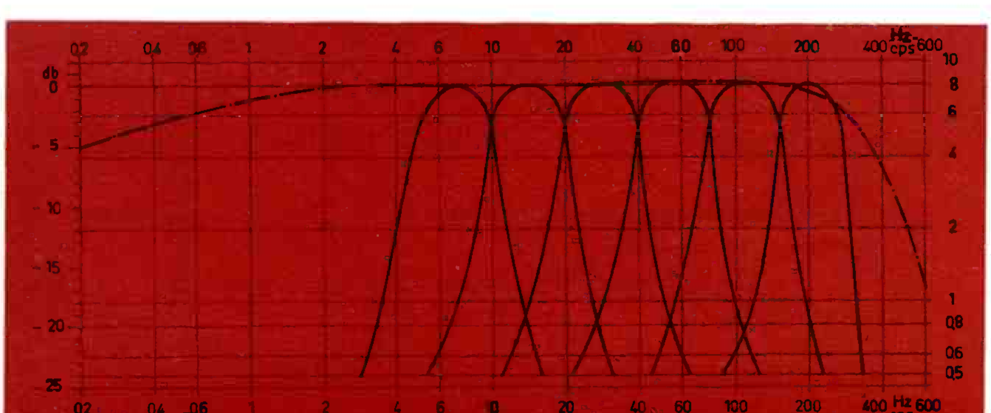
The diagram shows the characteristics of the frequency response.

——— standard characteristic with tolerance specifications weighted according to CCIR 210 Doc 2153 and DIN 45 507

– – – – weighted characteristic indicated on instrument 2 of EMT 420 A

– · – · – unweighted indicated on instrument 2 of EMT 420 A

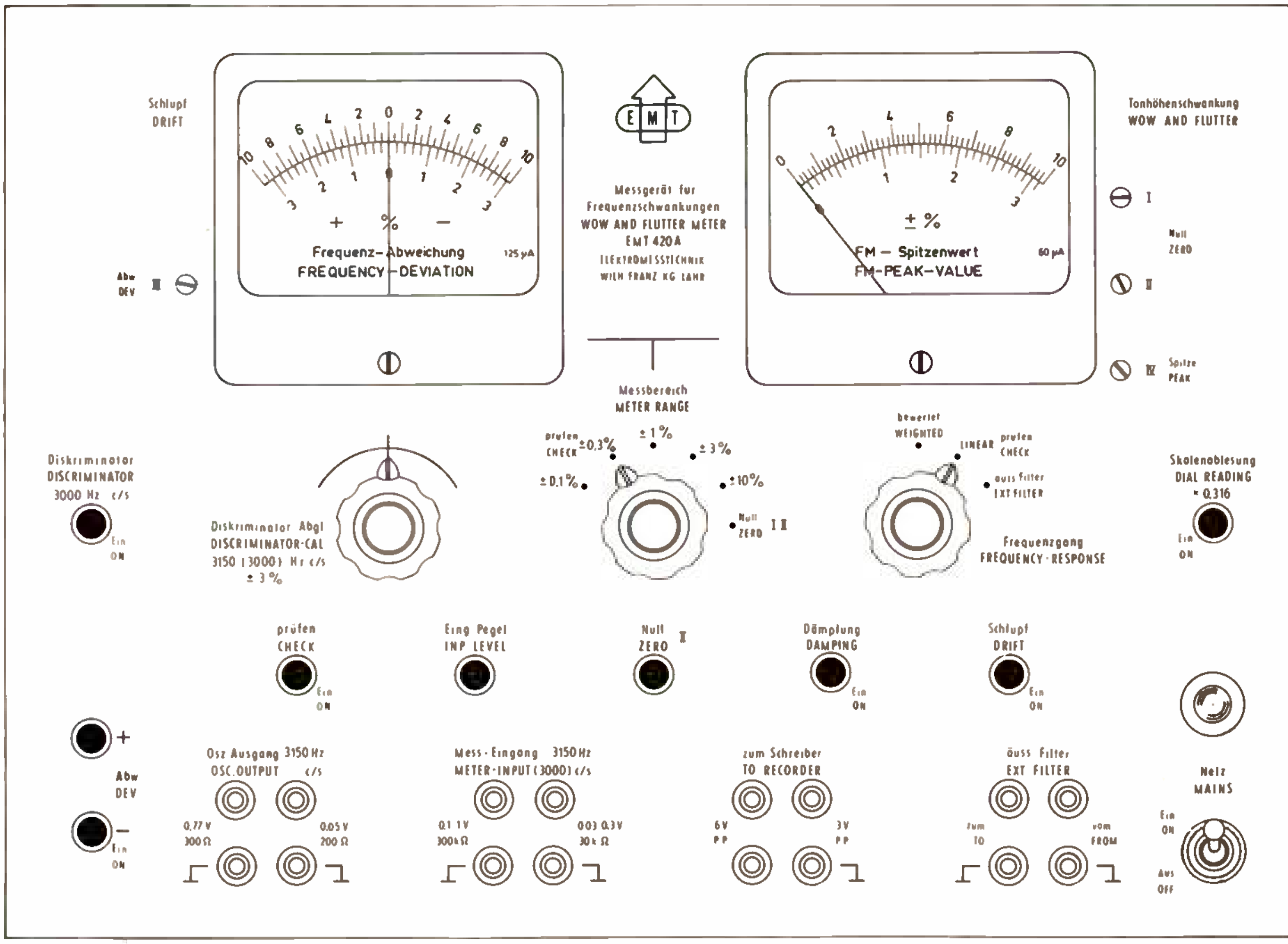
EMT 421A OCTAVE BAND ANALYSIS FILTER UNIT



The unweighted characteristic of EMT 420 A and characteristics of the six switchable

OCTAVE BAND FILTER INTERVALS

5 – 10 cps
10 – 20 cps
20 – 40 cps
40 – 80 cps
80 – 160 cps
160 – 320 cps



EMT-420A WOW AND FLUTTER METER AND ANALYSER

APPLICATION:

The EMT-420A and its companion octave band filter unit, the EMT-421A, are designed for the analysis of the wow and flutter components in sound and instrumentation playback equipment. It is a must in all laboratories working on the design, manufacture, and quality control of tape recorders, disk cutting equipment, turntables, and other sound playback equipment. In addition it is to be used for the maintenance of such equipment in recording, broadcasting, motion pictures, and instrumentation applications.

MEASUREMENT VALUES OBTAINED:

The EMT-420A/421A Wow and Flutter Measuring Equipment measures the frequency deviation from a standard test frequency and expresses this deviation in percentage points. It furthermore indicates the frequency drift, thereby serving as an absolute indication of the true tape, disk, or film speed. Both peak and average readings can be read, and unweighted as well as weighted response curves are built in. The octave filter unit permits analysis of the interfering signal to permit rectification of the causes.

OPERATING PRINCIPLES:

A built in oscillator provides the internationally standardized test frequency of 3150 cps which is to be recorded on the device under test. For testing of disk or optical recorders appropriate test disks and films are available. The subsequently reproduced test frequency is then amplified in the EMT-420A, is limited and then measured by the phase discriminator which has a linear response in the range of 3150 cps ± 400 cps permitting accurate indication of frequency modulation with a deviation of as much as ± 10%. The modulating frequency is then indicated on two meters, one meter for drift (0–0.2 cps) and another for the peak indication of wow and flutter. External level recorders, oscilloscopes, and filter units are easily connected through the jacks provided for that purpose.

TECHNICAL SPECIFICATIONS: EMT-420 A WOW AND FLUTTER METER

Oscillator output for test frequency: 3150 cps; accuracy 2%
Output levels: a) 0.775 Volts @ 300 Ohms
b) 50 mV @ 200 Ohms

Measuring input for test frequency: 3000 & 3150 cps switchable.

Discriminator tunability for test frequency error: ± 3%

Input level capability: a) 0.1 – 1 Volt @ 300 K Ohms
b) 30 – 300 mV @ 30 K Ohms

Test ranges: ± 0.1 – 0.3 – 1 – 3 – 10%
Extra test range: ± 0.03%

Indication of wow and flutter: ± % either peak or RMS value; switchable.

Indication of drift: + or – % frequency deviation within the range of 0–0.2 cps.

Calibration and control facilities:
a) input voltage
b) frequency deviation
c) center frequency
d) peak indication

Frequency response of indicated wow and flutter switchable to:
a) Weighted
according to ISO, CCIR and DIN specifications. Peak at 4 cps; roll off of 6 dB/octave above and 10 dB/octave be-ow that frequency.
b) Linear
0.5 – 300 cps; roll off of 3 dB/octave and 15 dB/octave above that range.
c) Via external filter network with impedance of 6000 Ohms (such as EMT-421 A).

Connection for external oscilloscope or level recorder within range: 0 – 200 cps; 3 or 6 Volts peak-to-peak; Min. load impedance 100 K Ohms.

Power requirement: 110–117–125–220–240 Volts 50–60 cps. approximately 80 VA.

Dimensions: Front panel: 10 5/8" x 14 3/4"
In cabinet: 12 1/4" x 16 3/8" x 9 5/8"

Weight with cabinet: 36.5 lbs. (16,5 kg).

EMT-421 A OCTAVE BAND ANALYSIS FILTER UNIT:

Bandpass ranges, switchable:
5 – 10 cps
10 – 20 cps
20 – 40 cps
40 – 80 cps
80 – 160 cps
160 – 320 cps

Insertion loss of center of pass band: 0 dB

Loss at boundary frequencies: approx. 3 dB

Roll-off beyond boundary frequencies: approx. 18 dB/octave

Input and output impedances: 6000 Ohms (matches EMT-420 A)

Dimensions: Front panel: 10 5/8" x 4 1/2"
In cabinet: 12 1/4" x 5 1/2" x 9 5/8"

Weight in cabinet: 19.5 lbs (8,8 kg).

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A wow and flutter meter for

measuring frequency deviation in recorders

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EMT GERAETEWERK
W. FRANZ, LAHR

Revised and translated
by Stephen F. Temmer

One of the interesting quality considerations in the evaluation of sound recording media such as tape recorders, film recorders, dictating devices, record players, etc. Is its speed constancy, pitch variation, or, in other words, its frequency variation. This is caused by the limited accuracy of drive mechanisms causing slow or rapid changes in the speed of tape, film, wire, or record. These speed variations cause frequency variations which both for slow occurrence (WOW) or rapid change (FLUTTER) make themselves disagreeably noticeable. The New Wow and Flutter Meter EMT 420A was created especially to provide continuous observation of sound reproducing devices as well as for final checkout of production lines engaged in manufacture of such devices. Beyond that this test meter will be very useful in developmental work involving sound recording devices when used in conjunction with selective filters (EMT 421A) and pen recorders.

Devices for the measurement of frequency variations in sound recording devices have been available for the past fifteen years and more. They all are based on the same measuring principle, indicating the frequency deviation of a test tone recording on tape, film, or disk. It is usually required that this deviation measurement have some relationship to the annoyance caused in the human ear. It is in this latter fact that the various available test devices vary greatly as do the numerous regional standards written throughout the world, many of which ignore the problem as indicated. The result is that measurements made with various test devices produce vastly differing readings so that an absolute evaluation or comparison between machines under test is virtually impossible.

It was the German Standards Association (DIN) which first issued a proposed standard (DIN 45507) in February 1961 based on an original proposal dated October 1956, and entitled "Measurement device for frequency deviation". This standard was revised to include all of the latest knowledge of acoustical psychology and the state of the engineering art.

THE NEW STANDARD PUBLICATION

The proposal DIN 45507 February 1961 specifies under paragraph 2 the use of 3150 cps as the preferred test frequency for acoustical measurements, conforming to ISO Recommendation 402.

Paragraph 3 and 4 involve the weighting networks in the unit and the measurement indication of the frequency deviation relating to the subjectively sensed aural disturbing influence.

Paragraph 3 indicates the weighting curve to be used in the range from 0.2–200 cps. This curve is a compromise between several such curves used internationally. According to statistical knowledge the frequency of maximum disturbance is 4 cps, while frequencies above and below this point are less disturbing as shown by the weighting curve selected. It follows from this that all of the disturbing frequencies are to be measured in one measurement.

Paragraph 4 describes the meter ballistics, which are quasi peak indicating for the frequencies involved, as follows:

"When testing the indicating properties with one sided frequency deviations in the form of square wave impulses, the following values are to be obtained if an impulse of 100 ms produces 100% indication:

Impulse time (ms)	10	30	60
Pointer deflection (%)	25 ± 3	59 ± 6	85 ± 6

(Impulse frequency 1 cps)

The decay time is to be measured with impulses of 100 ms at a rate of 1 cps. The pointer is to return to 42±8% between pulses. This measurement is to be undertaken using the weighting curve specified."

From these values the integrating property of the meter movement may be seen which is again designed to agree with the subjective aural sensitivity. The values as given in the standard characterize both the rise and decay time properties of the meter movement when using the weighting network. The standard also specifies the method of rectification as a voltage doubler in order to read peak-to-peak values. The scale markings of the peak indicator follow the usual practice in the RF field in which only one side of the frequency deviation is given, being marked "± %". The value indicated is then equal to one half the peak-to-peak value. Furthermore the indicated values refer to a sine wave frequency modulation of the test frequency. These are the most important points in which this new standard differs from previous standards or other standards used throughout the world.

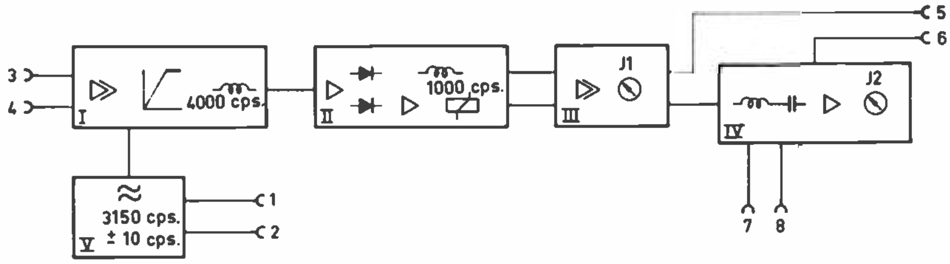


Figure 1
Block schematic of the Wow and Flutter Meter for sound recording media. The letters I to V in the five blocks refer to partial circuits.

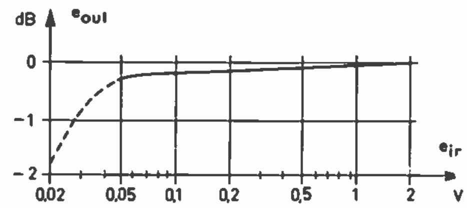
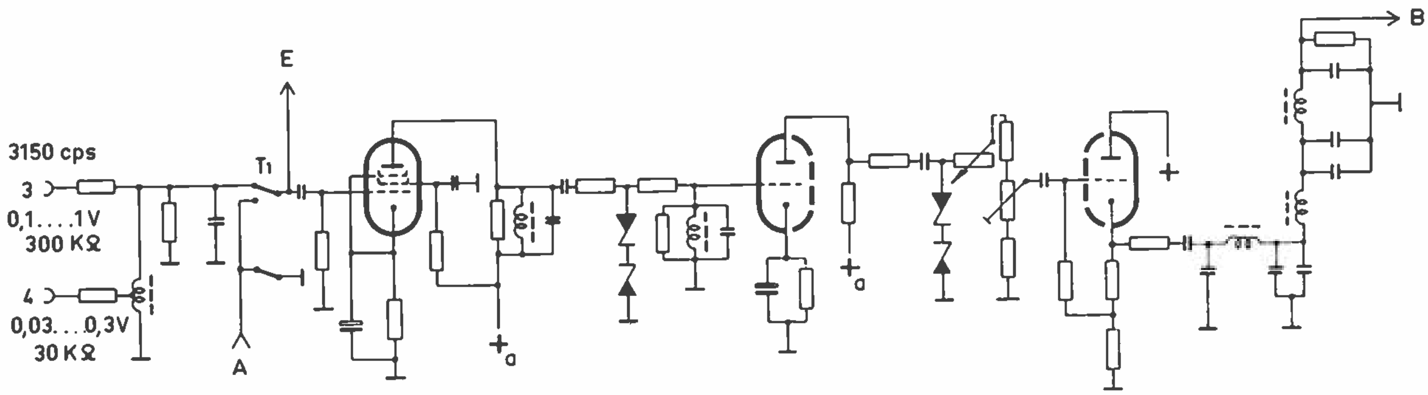


Figure 2
Amplitude limiting. Relative output voltage e_{out} at Part A vs input voltage e_{in} at jack 3 In range e_{in} = 0.05 . . . 2 V an amplitude limiting of about 1 : 40 is obtained.



Circuit I
Pre-amplifier with limiter and low pass filter.

GENERAL CONSTRUCTION

The EMT 420A Wow and Flutter Meter and its associated EMT 421A Filter Network conform in every way to the new DIN standard as well as to ISO Recommendation 402 and the newly adopted CCIR standard of February 1962. Furthermore it features such helpful accessories as an internal oscillator which may be used to feed a tape or disk film record to produce the test frequency. It furthermore contains its own calibrating sources as well as a relay which blocks indication in the absence of signal. The EMT 420A and EMT 421A are each housed in a separate metal housing of identical height and depth so that the two instruments can be placed neatly alongside one another.

AMPLIFIER AND DISCRIMINATOR

Figure 1 shows the block schematic and the circuits I through V show the actual detailed schematics of the EMT 420A. The following description refers to these schematics. The test frequency as reproduced through the playback amplifier of the tape, film, wire, or disk reproducer is fed to either jack 3 or 4 of the unit. In Part I this test frequency is amplified by tubes 1 and 2 and fed through highly damped resonant circuits in order to obtain sufficient level for amplitude limiting. The limiting is accomplished by two pairs of Zener diodes connected in opposition, where one pair is a pre-limiter while the second pair is the main limiter.

The second triode half of tube 2 is followed by a low pass 4 KC filter at whose output the test frequency now appears with a constant amplitude independent of the input level (Fig. 3). Pressing push button T 2 in Part IV allows a measurement on meter J2 to determine if the input level lies between the admissible values required for proper operation. The input level

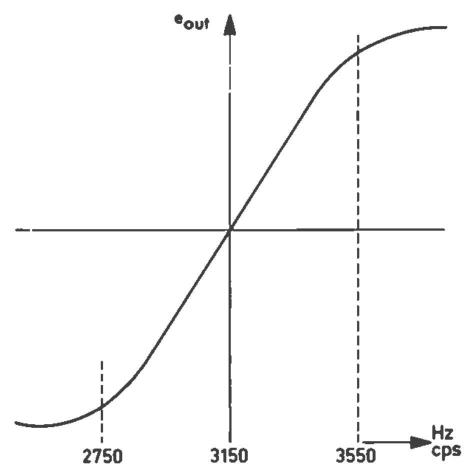
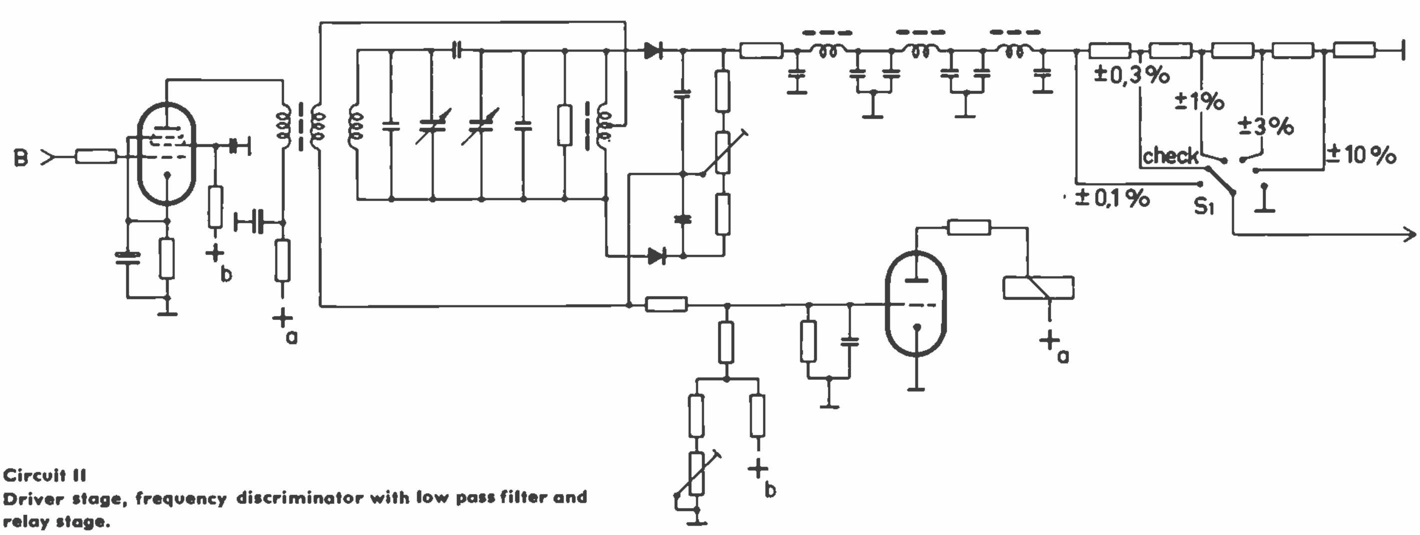
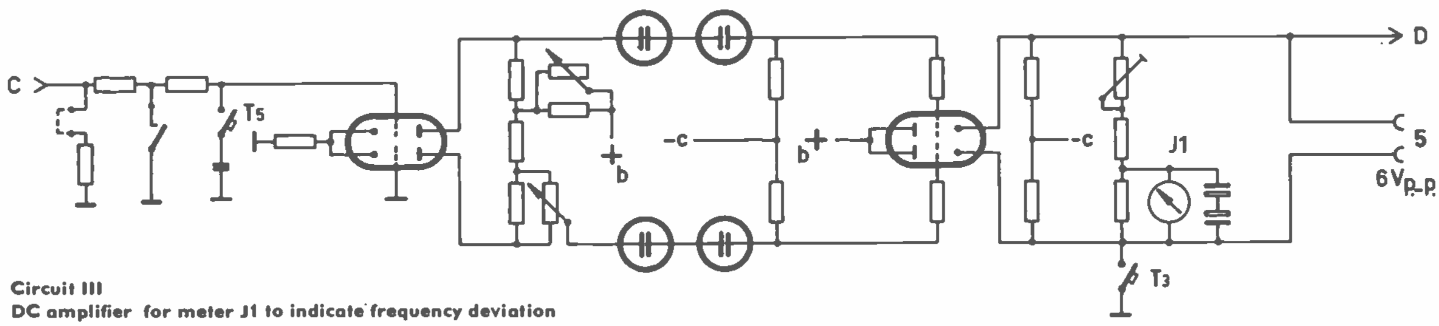


Figure 3
Transfer characteristic of phase discriminator. Output voltage e_{out} at Part II vs input frequency. Range 3150 cps ± 400 cps.



Circuit II
Driver stage, frequency discriminator with low pass filter and relay stage.



Circuit III
DC amplifier for meter J1 to indicate frequency deviation

at jack 3 must lie between 0.1 and 1 Volt while at jack 4 it must have a value of between 30 mV and 300 mV. Input impedance at 3150 cps at jack 3 is about 300 KOhms and at jack 4 about 30 KOhms (the latter is useful for transistor devices).

In Part II the amplitude limited test frequency is fed via tube 3 to a frequency discriminator circuit of usual design with a strongly damped band filter. This produces a transfer characteristic with sufficient linearity in the range of ± 400 cps on both sides of the test frequency of 3150 cps, allowing for measurement of the FM even for deviations of as much as ± 10% (Fig. 4). This also permits use of the 3000 cps test frequency used in the United States for which a special button is to be found on the front panel.

In order to be able to measure reproducers riven by non synchronous drives which may run off speed as a result of power line voltage fluctuations and which in turn will deviate the test tone from the 3150 cps point, it is possible to detune the phase discriminator band filter by more than ± 2% using the ganged tuning condenser (2×485 pf). The discriminator is followed by a low pass IKC filter for the suppression of the carrier frequency.

The switch S1 permits selection of five measurement ranges by means of a voltage divider (± 0.1, 0.3, 1, 3, 10%). A further button increases sensitivity to ± 0,03%. Furthermore a signal voltage is obtained from the phase discriminator which operates a relay via tube 4, whose contacts short the input of the ensuing DC amplifier whenever the input signal is absent.

INDICATION:

Part III is a two stage DC amplifier using tubes 5 and 6. Both tubes obtain their plate voltage from the stabilized supply to maintain constancy of the zero adjustment. The meter J1 which lies in the bridge arm between the cathodes of tube 6, has 3 tasks. Primarily it is used to align the frequency discriminator to the test frequency. econdly, the instrument indicates drift between beginning and end of a tape recording directly in % when push button T5 is depressed which limits the frequency response at the DC amplifier input to 0–0.2 cps. Thirdly, the instrument is used for calibration which will be described later.

The jacks 5 which are parallel to meter J1 permit the connection of a pen recorder for analysis of the frequency spectrum of the deviation. The deviations found at these jacks cover a frequency range of 0–200 cps (–1 dB). The output at the jacks is devoid of plate voltage and their source impedance is about 1 KOhm. Input impedance of pen recorders connected here must be > 100 KOhms, so as not to falsify the peak indications. The available output voltage is about 6 V peak-to-peak. If the pen recorder permits fully balanced input, the balancing properties of the DC amplifier with regard to line voltage fluctuations may be usilised.

Part IV contains the filter and the peak indicating section. Switch S2 is used to insert the desired weighting curves. In position "WEIGHTED" the standard weighting curve (Fig. 5) is used. In "LINEAR" position the indication is linear from 0.4 to 300 cps (–3 dB). In the position "EXT. FILTER" the special filter unit EMT 421A which serves to analyse the interference

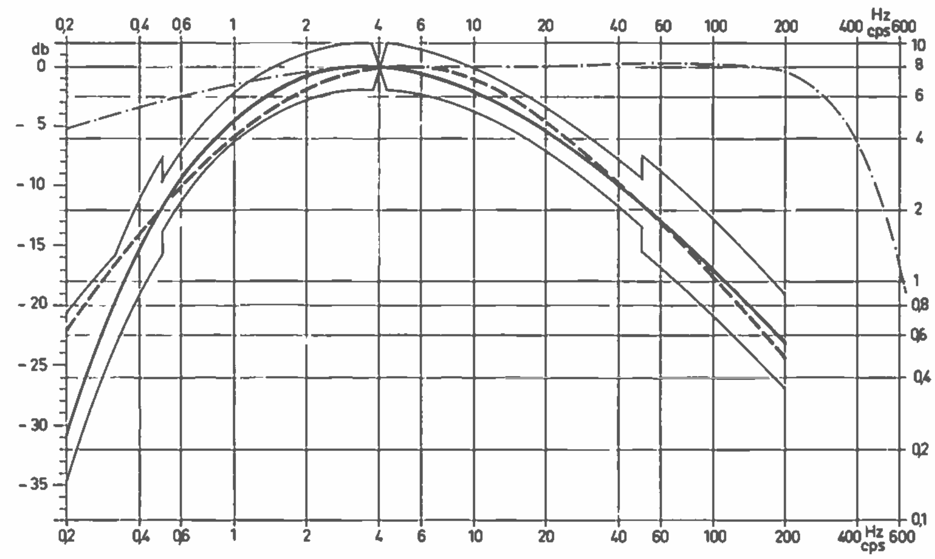


Figure 4
Indication on meter J2 vs deviation frequency, reference value 80 marks of the 100-mark-dial at 4 cps.
——— rated curve with tolerance field, weighted according to CCIR and DIN
– – – – frequency response "weighted"
– · – · – frequency response "unweighted".

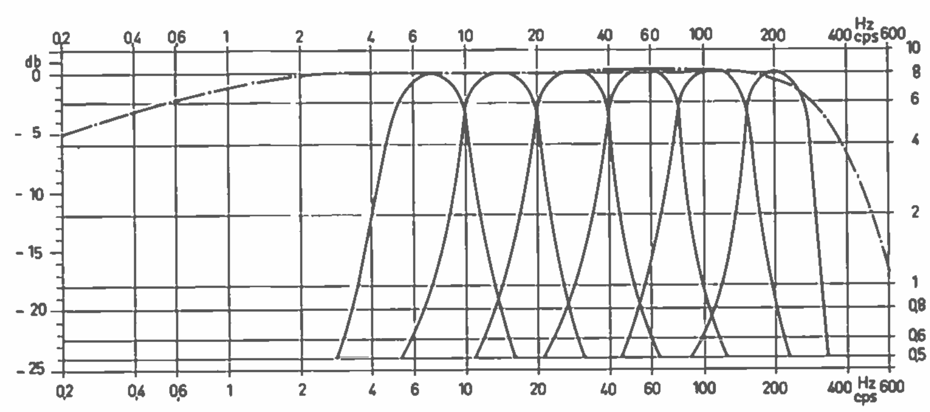
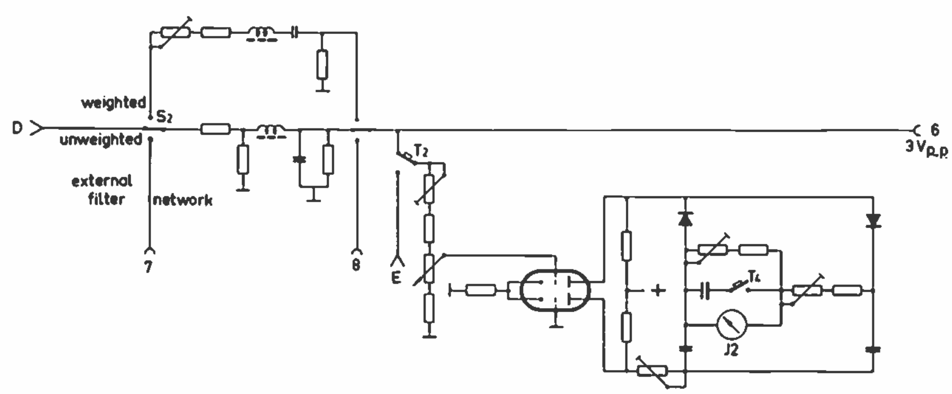


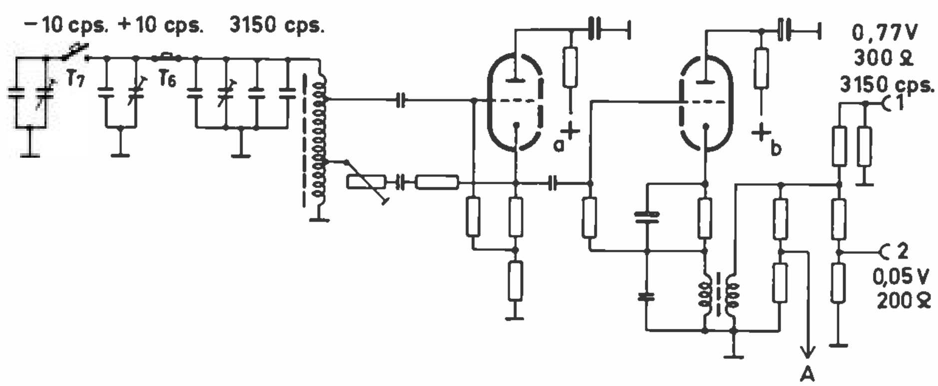
Figure 5 **The unweighted characteristic of EMT 420 A and characteristics of the six switchable octave band filter intervals.**



Circuit IV
Switchable filter unit with DC amplifier for meter 2 to indicate the FM peak value

signal, may be connected between jacks 7 and 8. Available are five high pass and five low pass sections in octave bands with extremely steep flanks (Fig. 5). This EMT 421A Filter Unit, through use of either the high or low pass section at 12.5 cps, permits the separation of the Wow from the Flutter components of the signal.

Because of the fact that (when using the weight-



Circuit V
Generator for 3150 cps.

ing filters) readings on meter J2 may fluctuate excessively, it is possible to damp the meter further by depressing push button T4. It is of course also possible to insert other selective filters across jacks 7 and 8 (impedance 6000 Ohms). Jack 6 allows another connection after the filter, of either a pen recorder or oscilloscope with 3 V peak-to-peak available at this point. The peak indication is connected to the filter via tube 7 and is read on meter J2. Readings according to the standard are made only with the weighting network connected. For fluctuating indications the highest deflection of the meter must be taken. It is important to point out that measurements in the "LINEAR" mode will ordinarily produce readings of two or three times the weighted reading. Such a measurement often presents a better possibility of analysing the cause of the wow and flutter in the device under test.

The practical aspects of recording devices produce interference of an impulse type as well as sine wave type interference of either a periodic or aperiodic nature. This emphasizes the need to standardize on a particular peak indication which will permit comparisons between measurements taken anywhere. The peak indicating requirements as described in paragraph 4 of the standard are fully met with the EMT 420 A.

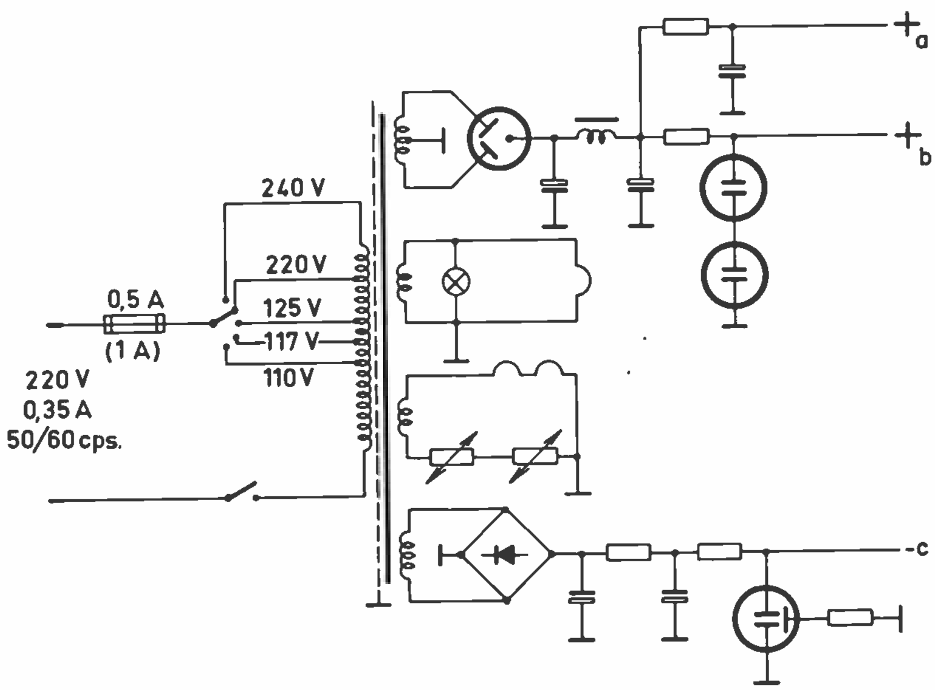
To give some insight into the indications to be expected from wave forms of identical peak value but differing wave forms, we enumerate several here compared to an indication of 100% for a sine wave.

Sine wave	100%
Square wave	110%
Saw tooth	95%
Single impulse	100%

These relative indications are obtained for linear response measurement with a middle interference frequency such as 40 cps. The number values obtained demonstrate rather well the meaning of the standard when it refers to "quasi peak" indication. This same peak indication is then used through the weighting network to perform the standard wow and flutter measurements.

OSCILLATOR:

The oscillator is shown in part V. The test frequency of 3150 cps is obtained at jack 1 with a level of about 0.77 Volts at a source impedance of 300 Ohms to be connected to the record amplifier of the device under test. At jack 2 the same frequency is obtained with a level of 50 mV at 200 Ohms. Frequency accuracy of the test tone is $> \pm 1 \times 10^{-3}$ and is quite sufficient for measurements with meter J2.



Circuit VI
Power supply unit

Short term accuracy over a period of 5 minutes is > ± 1 x 10^{-4} which is sufficient to perform drift measurements using meter J1, if the recordings at the beginning and end of the tape are made quickly one after the other.

The built-in oscillator has the dual purpose of providing the calibrating means for checking the accuracy of the entire instrument. This is accomplished by first pushing push button T1 in section setting switch S1 to the range 1 ± 0,3% Cal. and S2 to "LINEAR CAL.". This connects the oscillator output to the input of the unit and the ganged condenser is used to center meter J1. Operating push buttons T6 and T7 detune the 3150 cps tone by a fixed ± 10 cps. This operation must cause needle deflections of ± 0.318% on meter J1 and these points are marked in red. This also produces calibrating marks at jacks 5 and 6 for a pen recorder input so that the full scale deflections of the needle are recorded on the pen recorder paper. This automatically determines the total meter range at all settings of the range switch. This allows an immediate reading of the frequency deviations on the written record.

An impulse shaped frequency shift is also used to control the proper peak indication of meter J2. This requires that T7 remain depressed while T6 causes a total detuning of 20 cps every ime it is depressed or released. This shift should cause meter J2 to perform an identical one time deflection after which the needle returns to zero. The maximum deflection to be attained is similarly marked on the meter face.

The detection of the interference causing elements of the drive system under test is best undertaken using a pen recorder or external filter networks. The pen recorder permits analysis of the frequency of the interference and its relationship to rotating parts in the device under test while the filter network provides quantitative analysis of the various pass bands.

Literature

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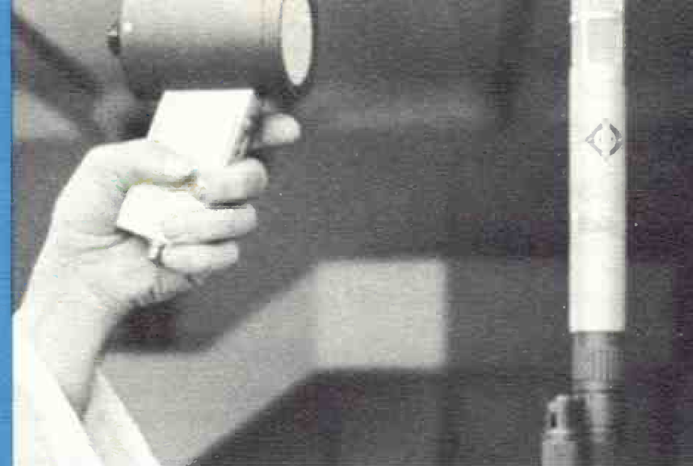
Presented by:

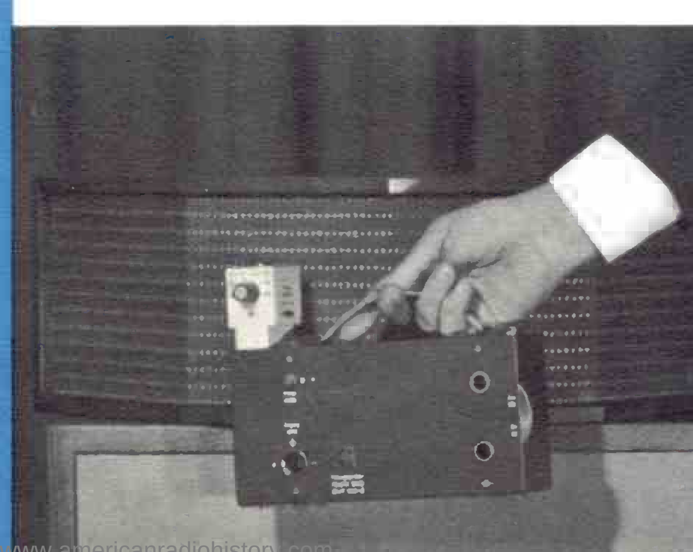
ELEKTROMESSTECHNIK WILHELM FRANZ GMBH

WETTINGEN (AG) SWITZERLAND

New test unit for Studio application and Audio engineering



Polarity Tester EMT 160



The Polarity Tester EMT 160

is the first instrument of its kind which enables polarity testing in electro-acoustical systems **reliably, quickly** and **easily.**

How important is correct polarity?

If an acoustic event is picked up simultaneously by several microphones, their output voltages will be coherent, i. e. correspond with each other as regards wave form and phase relationship. If two or more coherent signals are combined out of phase — i. e. with wrong polarity — they will more or less cancel each other. The question of polarity is therefore at least as important as the right level.

When does right polarity matter?

It matters when two or more such coherent signals are going to be mixed together. It is basicly irrelevant whether the mixing is going to take place electrically or acoustically. In practice this means that every mixing of incorrectly poled microphones will produce certain cancellations.
In stereo recordings the effect of wrong polarities is even more serious because the special effect is destroyed and the stereo character will be lost.

How can a polarity error be traced in operation?

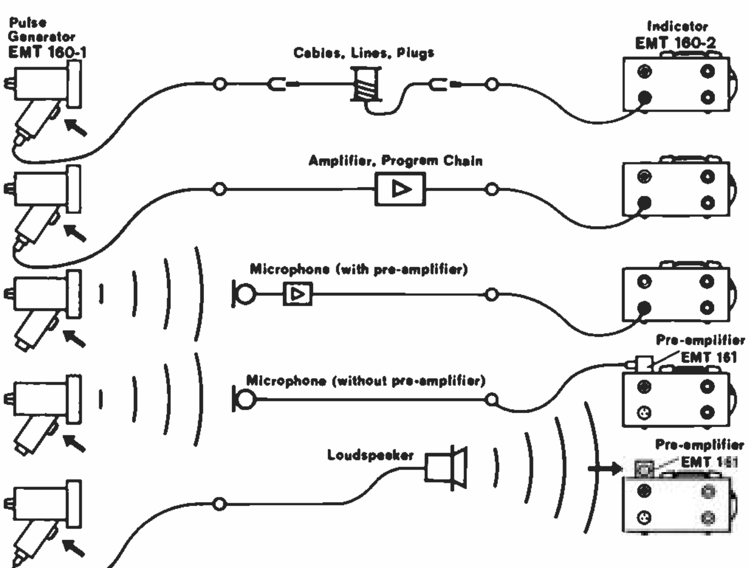
Conventional methods are either too costly, too time consuming or have only limited application. The EMT polarity tester is specially conceived for the requirements of studios and in production testing. The process does not rely on complex scientific considerations and it therefore enables moderately skilled personnel to obtain reliable results. The principle is as follows:
A steep pulse of definite polarity is fed into the system. The digital polarity indication at the output of the system is not affected by the shape of the pulse nor by overload of the system nor the transient response. It depends solely on the polarity of the pulse and is therefore reliable.

What can be effectively tested with the polarity tester EMT 160?

Everything which could be out of phase in an installation: from a simple plug and cable to mixers and the most complicated studio programme chain.
The EMT polarity tester offers its greatest advantages in the testing of microphones and loudspeakers. For the testing of microphones an acoustical pulse is produced, for testing loudspeakers it receives the acoustic pulse via a microphone.
Even immediately before a recording session all the microphones which have already been set up can be checked for their polarity quickly and reliably as a final check without altering their position.

How is the polarity tester EMT 160 used?

The unit consists of two parts. The sender which is held in the hand like a pistol and triggered produces an acoustic or electric pulse at one of four different levels. The indicator which is connected to the output of the system under test indicates the polarity by means of a green or red light. For transport and for testing the unit within itself the sender can be pushed into a recess in the indicator.
The pre-amplifier EMT 161 is plugged into the same recess. It contains a microphone for direct test of loudspeakers and it increases the sensitivity of the indicator unit sufficiently to test dynamic microphones directly without microphone pre-amplifier.



Application Field

Technical details

1. Pulse Generator EMT 160–1	
Acoustic pulse (pressure pulse positive)	300 µbar at 2" distance
Electric pulses	9 V , 1 V , 100 mV , 1 mV
Power supply	9 V Microdyn battery (built in)
2. Indicator EMT 160–2	
Nominal input voltage	1.55 V (+ 6 db)
Minimum input voltage	200 mV RMS
Maximum input voltage	100 V RMS
Input impedance	app. 10 k-ohm
Input	unbalanced, floating
Switching and blocking time	less than 1 sec
Transistors	11, silicon planar transistors in all critical positions.
Diodes	2
Temperature range	–10° C to + 50° C
Green indicator lamp	polarity right
Red indicator lamp	polarity wrong
Power supply	2 flat batteries 4.5 V (built-in)
3. Pre-amplifier EMT 161	
Gain continuously variable	10 db to 50 db
Input unbalanced	for direct connection of dynamic microphones
Input impedance	10 k-ohm
Output impedance	1.8 k-ohm
Power supply	from the indicator
Average sensitivity of the built-in microphone	90 V/µbar.

EMT

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Polarity Tester
EMT 160

by Reinhard Plantz

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1965 / Heft 20

BY REINHARD PLANTZ

Polarity Tester EMT 160

A new test instrument for Studios

The polarity of electro acoustical installations becomes crucial whenever the signals from several transmission lines have to be electrically or acoustically mixed.
This necessarily applies to all stereophonic systems but it also applies to monoraul systems where a number of microphones are being used which have to be mixed to form an electrical sum signal.
To avoid cancellation of signals it is important to be able to ascertain whether and where an intentional or unintentional phase reversal is taking place. It is also important to know, to mention just a few examples, whether a phase reversal is taking place in microphones, amplifiers or loudspeakers and to ascertain whether all the individual loudspeakers in a multiple speaker system are in phase.
The quick routine checking of the polarity of transmission systems during operation always presents problems, particularly where transducers are involved. A method which is frequently used, consists of sending a series of electrical or accoustical pulses of known polarity and assymetrical amplitude through the transducer under test and to ascertain the phase relationship at the output of the transducer by means of an oscilloscope. This method is not very satisfactory because of the complexity of the instruments involved such as the pulse generator and the oscilloscope and it is too cumbersome for a quick check through as final test, often immediately before the start of a transmission or recording. For this reason, the Institut für Rundfunktechnik, Hamburg has developed the equipment which will be described hereafter for testing the polarity of electrical and acoustical transmission systems, with special regard to the requirements of recording and broadcasting studios.

The Operation of the Polarity Tester

The block schematic of fig. 1 shows the operation of the polarity tester when testing a transmission line, consisting of the microphone 2 and the amplifier 3. The polarity tester consists of two separate units, i. e. the pulse generator 1 and the digital indicator 4. The comparatively simple pulse generator 1 produces a single acoustic pulse of definite polarity by discharging the condenser by means of the push button via the moving coil of a small loudspeaker. In our example a short pressure wave followed by a suction wave is produced. The characteristics of the pulse are largely determined by the properties of the loudspeaker used. The shape and the phase relationship of the pulse is shown inside the little circles in fig. 1.
The acoustical pulse is changed into an electrical one by means of the microphone 2, amplified by the amplifier 3 and then taken to the indicator unit 4. The pulse passes first through a symmetrical limiter C. The inputs of the flipflop (electronic switch) stages A and B are in parallel. The circuit A responds only to positive pulses and circuit B only to negative ones. The first positive pulse, i. e. the first positive half of the pulse, will therefore trigger the electronic switch A, and thereby switch on the lamp L1 which is the positive indicator. In this condition the circuit B is simultaneously blocked via the blocking line X1. The circuit B can therefore no longer respond to subsequent negative pulses.
This quasi-stable state will last for several seconds. During this time the positive indication L1 remains switched on and stage B remains blocked. After the lapse of the predetermined time lag stage A will return to its rest position of its own accord and the blocking of the multi vibrator B is thereby also cancelled. The resultant indication is therefore always determined by the phasing (polarity) of the first half of the test pulse. If, for instance, a phase reversal had occurred in the amplifier 3 the switch B would have been triggered first, causing the lamp L2 — negative indication — to light up at the same time blocking switch A via X2.
The pulse generator is equipped with several electrical outputs of varying magnitude to enable amplifiers or loudspeakers under test to be fed directly. When a loudspeaker is to be tested, it is fed with a pulse from the generator. A microphone, the polarity of which has been ascertained previously, is then placed in front of it. The polarity of the loudspeaker can then be determined by connecting the indicator to the microphone output.
To ascertain the phase relationship in multiple speaker systems where all the speakers are triggered simultaneously by the test pulse, the test microphone should be brought as closely as possible in front of the speaker to be tested. As it is always the first in a series of pulses which determines the reaction of the indicator, the polarity of the loudspeaker which is physically closest to the test microphone will be indicated.

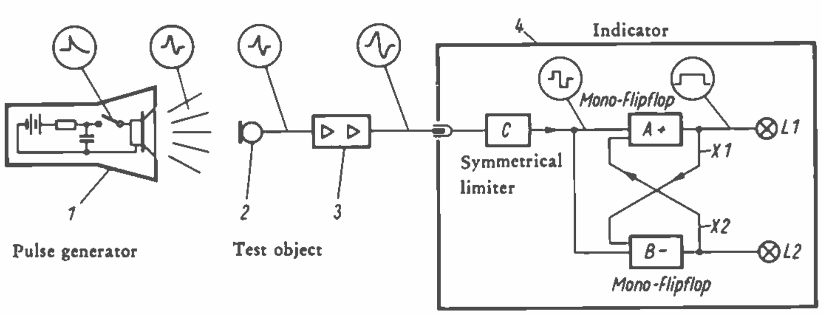


Fig. 1 Block Schematic of Polarity Tester

Circuit and Constructional Details.

Fig. 2 shows the complete circuit diagram of the polarity tester made by Messrs. Elektromeßtechnik W. Franz KG. The pulse generator EMT 160—1 contains a 9 V battery. By depressing the button Ta 1, which op-

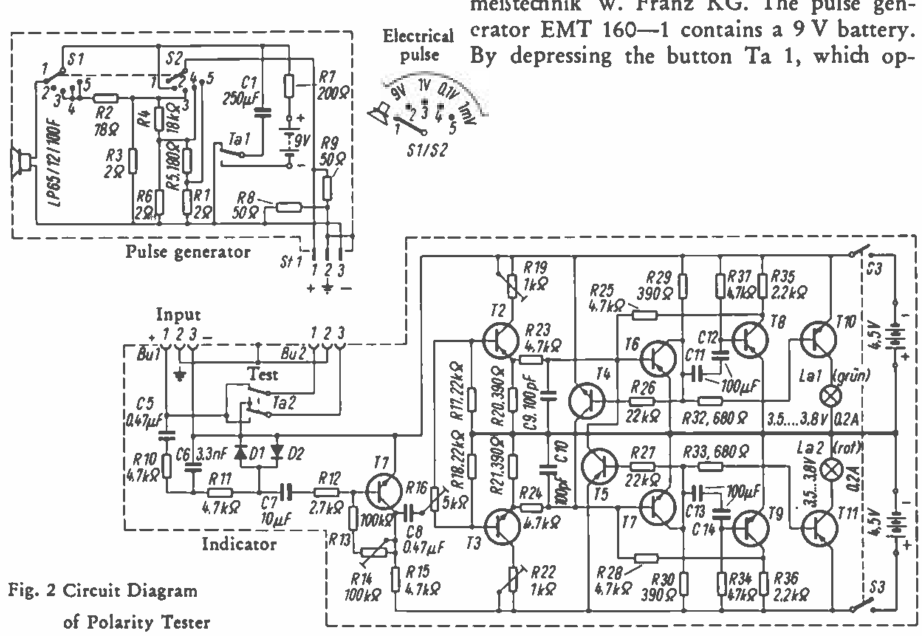


Fig. 2 Circuit Diagram of Polarity Tester

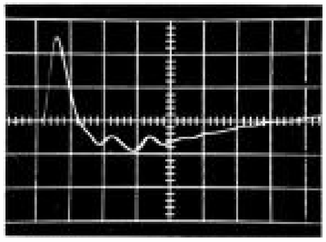
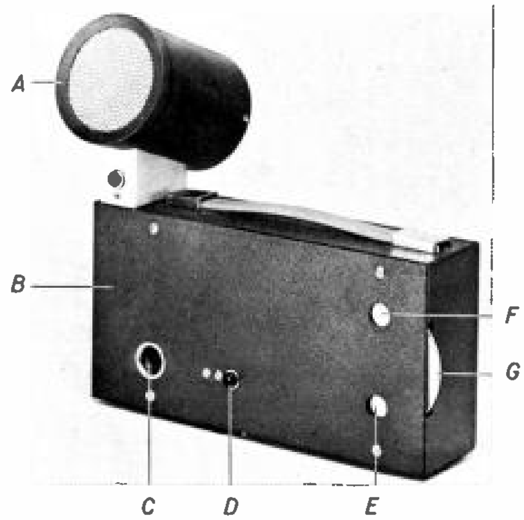


Fig. 3 Oscillogram of the acoustical pulse, total length 2 msec.



Right:

Fig. 4 Complete Polarity Tester
A = Pulse generator
B = Indicator
C = Input socket
D = Test button
E + F = Signal lamps
G = Main switch

erates a changeover switch, the condenser C1 is charged to the battery voltage via the resistor R7. Upon releasing the trigger button (shown in the diagram in its rest position Ta1) the condenser C1 is discharged through the loudspeaker. The resultant acoustical pulse is shown in the oscillogram of fig. 3.

By means of the selector switch S1/S2 the magnitude of electrical pulses for testing amplifiers and loudspeakers can be varied according to the particular requirements. In the switch position 2, the condenser C1 is discharged without any series resistance directly via the instrument under test which is connected to the output St 1. This high level output is suitable for exciting loudspeakers with impedances from 4 to 64 ohms. The potential divider tappings 3 to 5 on the switch S1/S2 provide output pulses of 1 V, 100 mV and 1 mV for testing amplifiers of varying input sensitivity as are commonly met in studios (see technical details).

The indicator EMT 160—2 is transistorised and contains a power supply unit consisting of two 4.5 V batteries in series. The positive or negative pulse at the input is first of all amplified by the pre-amplifying transistor T1 in emmitter configuration. The full battery voltage of 9 V is required to provide sufficient voltage swing. The diodes D1 and D2, which are connected in anti-parallel, act as voltage sensitive resistors thereby preventing the transistor T1 from being overdriven even in the event of very large input voltages up to about 100 V. The indicator section which is constructed symmetrically is fed via the potentiometer R 16. It consists of two independent loops which are equipped with complementary transistors which are drawn above and below the zero volt line in heavy black.

Positive pulses are passed on through the stage T3, negative ones through the stage T2. The collector stage with the transistor T2 acts as amplitude limiter. The degree of limiting can be adjusted by means of the trimming resistor R 19. The limited signal then reaches the base of the typical mono-stable flipflop consisting of the transistors T6 and T8. In the rest condition the transistor T6 is not conducting and the base of the transistor T10 receives the full negative voltage via the resistor R32. The power switch T10 is therefore blocked.

When the base of the transistor T6 is triggered with a negative pulse the flipflop will turn into its conditionally stable state, the duration of which is largely determined by the time constant of the elements C11/C12 and R31 which is approximately 3 seconds. During this time the transistor T6 is conducting, thereby making the base of the transistor C10 positive thus causing current to flow through the indicator lamp La1.

The blocking stage with the transistor T4 lies in parallel with the transistor T10 and is made conducting at the same time. The entire negative battery voltage is therefore brought to the base of the other mono flipflop circuit consisting of the complementary transistors T7 and T9. This stage is thereby safely blocked, especially as the signal is limited to such a level that the limited signal voltage is considerably smaller than the blocking voltage. In the case of positive pulse going through the stage with the transistor T3 the same applies, but with opposite polarity.

The stages T2/T3 not only act as limiters, but also serve for testing the state of the battery in order to give a clear indication of the reliability of the indicator unit. As the magnitude of the limited signal is necessarily dependent on the available battery voltage, the limited amplitude will fall below the threshold of the mono flipflop if one of the battery voltages has fallen below the permissible value. The unit will then suddenly fail to respond. To test the operational reliability of the indicator unit an electrical test pulse is fed from the generator to the test input Bu2, the polarity of which can be reversed by means of the switch Ta2. This will show whether the indicator is ready to give positive and negative indication. The state of the battery in the pulse generator unit can be judged from the loudness of the acoustical pulse.

Construction of the Unit

Fig. 4 shows the small neat polarity tester. During transport and storage the pistol shaped pulse generator A is pushed with its handle into the corresponding recess in the indicator unit B. The dimensions of the indicator (without pulse generator) are approximately 220 mm x 110 mm x 50 mm. C is the three pole input socket (Bu 1). The test socket (Bu2), the contacts of which can be reversed by means of the switch D, is not accessible from the outside. It is mounted at the bottom of the recess. When the pulse generator is

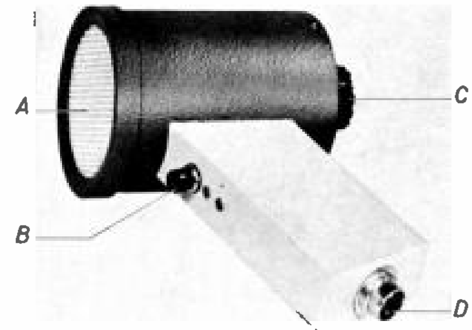


Fig. 5 Pulse generator
A = Loudspeaker
B = Operating button
C = Selector switch for electrical amplitude
D = Output socket

plugged into the indicator unit, contact is made automatically between the output of the pulse generator and the test input of the indicator and in this position, which is shown in fig. 4, the indicator can be tested as described above. E and F in fig. 4 are the two indicator lamps which have different colours. They are covered by the knurled disc G and are uncovered when the unit is switched on. The battery compartment is accessible by removing the back of the indicator unit.

Fig. 5 shows the pulse generator in full. The round section which has a diameter of approx. 70 mm contains the loudspeaker system A for the acoustical pulses and the selector switch C for adjusting the magnitude of the electrical pulses. The 3-pole output socket D for the electrical signals is situated at the bottom of the handle. The handle also contains a 9-V Microdyn battery and the push button B for triggering the pulses.

TECHNICAL DETAILS

Pulse Generator EMT 160-1

Acoustical pulse: 300 μbar at 2“ distance (pressure wave, positive) for testing microphones

Electrical pulses: 9 V_{pz} for the direct testing of loudspeakers and multiple speaker systems with impedances of 4 to 64 ohms; 1 V_{pz} (denoted with red dot) for testing of power amplifiers and for the internal testing of the indicator; 100 m V_{pz} for testing of amplifier chains; 1 mV_{pz} “microphone level“ for testing of complete transmission lines.

Pulse release: by means of push button

Power supply: 9-V Microdyn battery (built in)

Indicator Unit EMT 160-2

Input voltage (nominal): 1.55 V (+ 6db)

Minimum input voltage: 200 mV rms

Maximum input voltage: 100 V rms

Input impedance: approx. 10 000 ohms

Input: unbalanced, floating

Switching and blocking time: less than 1 μsec

Transistors: 11 (silicon planar transistors are used in all critical circuits)

Diodes: 2

Temperature range: —10° C to +50° C

Green pilot lamp: no phase inversion

Red pilot lamp: phase inversion

Power supply: 2 x 4.5 V flat batteries (built in)

Power consumption: 4 mA per battery in quiescent state, 250 mA during indication (duration approx. 3 seconds)