

np PORTABLE

ON  
LOCATION  
MIXER





# ELEKTROAKUSTIK AS

np PORTABLE - on location mixer  
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This manual contains the description and technical specification of the portable field unit, np type 5104.2 - Mk IV, and the information presented is divided into the following sections -:

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- 01 -: INTRODUCTION AND DESIGN PHILOSOPHY.
  - 02 -: MECHANICAL DESIGN AND LAY-OUT.
  - 03 -: ELECTRONICS DESIGN AND LAY-OUT.
  - 04 -: GENERAL ELECTRICAL SPECIFICATION.
  - 05 -: INTERNAL SIGNAL LEVEL DIAGRAMME.
  - 06 -: DESCRIPTION OF ELECTRICAL CIRCUITRY.
  - 07 -: RECOMMENDED CALIBRATION PROCEDURES.
  - 08 -: SCHEMATICS AND PCB/COMPONENT LAY-OUT.
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ENCLOSURE -: SPECIFIC PERFORMANCE PROTOCOL  
FOR SERIAL NUMBER -: (18 PAGES)

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The np PORTABLE - on location mixer is engineered and manufactured by -:  
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# ELEKTROAKUSTIK AS

The np PORTABLE - on location mixer  
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## 01 -: INTRODUCTION AND DESIGN PHILOSOPHY =====

Over the last years outside recording and direct broadcasting have earned an increased reputation in several fields, supported by the design and availability of new and handy equipment for video and sound.

As digital sound recording has been refined and easily portable equipment introduced, it became possible to tape hours of music or drama on the same reel - the only limitation being the batteries - and as quality of the stored programme was improved accordingly by the new recording-technique, the portable mixer definitely became the weak link.

Portable mixing units may be many things - ranging from small boxes with just two rotary potentiometers to more 'technical' solutions, but despite efforts and claims, many portables seem to be less professional in terms of electrical specifications and mechanical construction.

Serious discussions with potential clients indicate, that a fully grown, but still easily portable sound mixer is urgently needed in order to provide the professional user with a well designed tool for the off-studio job.

Some years ago the Danish Radio and Television asked us to solve a similar problem, and our original np PORTABLE was developed and built in a number of fifty for the DR's reporter- and film teams.

Since delivery took place these units have been used daily with an IS- or 4S-NAGRA, and having coped perfectly with every possible on-location job under polar temperatures in Greenland or in South American heat and humidity as well, the np PORTABLE already has the reputation of being a highly valued field instrument.

As mentioned above, the units were originally designed to handle a potential client's specific wishes, but despite the fact, that these are more or less similar to the demands of a larger forum, we found it valuable to incorporate several new facilities and extend some of the former.

However - making comprehensive, totally modular electronics fit into tiny boxes will always be a continual fight against millimeters, and in order to cope a complete re-design was carried out.

The main goal was - shortly told - the design and marketing of a portable sound mixer, which would match the NAGRA-level in respect of versatility, electrical performance, mechanical standard and overall finish.

It was considered, that more than ninety percent of all practical field-operations are carried out with four microphones or less, and hence the unit was laid out accordingly, but to cover the last few percent of situations where more microphones are needed, two units may be easily linked.



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For many reasons modular plug-in electronics were chosen and the only wires inside the units are those coming out of the P & G faders.

In production and testing a modular construction is advantageous, but of even greater importance is the free access to separate plug-in units, if field-service is occasionally needed.

As low-noise design takes current and a high signal-to-noise ratio means voltage we accepted the weight of 16 IEC/R14 alkaline batteries, generating sufficient power for an operational period of 30 hours or more - all depending on the way of use.

Except for the 48 volt phantom feeding of condenser-microphones we desisted from using a dc/dc-converter, as this will reduce efficiency of the hand-borne - power with about 15 - 20 %.

The portable unit is developed and built with our large studio consoles in mind, and because of the ultimate overall-specifications the final result has been what we aimed at, namely an uncompromised portable to fill out the need for perfection outside the studio.

Used in a concert hall with a large symphony orchestra or during big-band recording the unit will produce a perfectly distinct and clear sound in a highly defined stereo-format and when used for the recording of voices in interviews, on the film stage or in the theatre a comprehensive equalizer section will cope with almost everything.

Tonal balance, levels and stereo image are monitored via phones, PPM- and phase instruments and also a talk-back-/conference system is incorporated in the monitor unit.

Signals from two separate tone generators are available, namely a 1 or 10 kHz sine reference for level- and tape head corrections and a 15 Hz cue-tone for editing purposes etc.

A unique and hard tested mechanical construction maintains permanent stability of the modular plug-in electronics and ensures total protection of surface and terminal panel during transportation.

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In order to convey an impression of our achievements a detailed description with photos, technical specifications, schematic diagrams and component lay-outs are presented in the following sections.

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## 02 -: MECHANICAL DESIGN AND LAY-OUT

The mechanical construction is a very important part of a portable field-unit and utmost care has been taken in order to obtain the highest grade of stability and protection, combined with a moderate weight.

The np PORTABLE - on location mixer - shown on photos # 1 through 4, is a fully self-contained, portable stereo mixer having the total operational weight of 5,5 kgs and the size of 325 x 262 x 90 mm (W x D x H) with it's cover closed.

Power is supplied from internal batteries or from a centertapped, external source of 24 volt.

The portable mixer is built into a metal cabinet, where a main-board interconnects the following 12 modular plug-in units -:

A) 4 universal input channels	type	IK	34	B
B) 1 monitor- and communications unit	type	MON	56	
C) 1 instrument- and control unit	type	LV	28	
D) 1 power unit with batteries and voltage monitor	type	ID	04	
E) 2 output modules with limiters	type	UK	17	
F) 1 reference- and cue-tone generator module	type	TG	09	
G) 1 DC-converter for phantom feeding of microphones	type	SF	17	
H) 1 terminal panel with interface electronics etc.	type	CP	06	

Photo # 5 is a top view of the open unit showing (from left) four input channels and the monitor-/conference cassette with the instrument cassette installed above them.

Batteries, voltage monitor and power switch, INT/OFF/EXT, are situated in the far right side of the cabinet, and the remaining electronics - output amplifiers and -limiters, tonegenerators and DC-converter - are housed in separate modules, plugged into the main board below each input channel.

The terminal panel is equipped with eight XLR-connectors for the in- and outputs, two 1/4''-headphone-sockets and one interface multi-connector.

### Cabinet, cover and terminal-panel.

The cabinet and the cover are light alloy constructions, where front- and back panels are special extrusions, while the side panels are milled 5 mm standard profiles - all anodised.

The thickness of the profiles is partially reduced to a permissible minimum and hence more than 50 % of the original weight is eliminated without affecting strenght.

The cabinet bottom and the top of the cover are PVC-coated, 1,5 mm aluminium panels, embedded on all sides in the cabinet- and cover profiles.

The rear cover profile is extended to protect the terminal panel during transportation but the construction also serves as a very convenient tilting stand, when the unit is in operation.



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In the closed condition cover and cabinet are linked with pin-type hinges in the lower edge of the rear cover profile but the two parts are easily separated when the cover is opened.

The cover locking mechanism is built into the front profile of the cabinet where a sliding lever with two positions unlocks or locks the cover by means of two spring loaded locking pins.

When the closed unit has been placed on site, the unlocked cover may be opened and swung back 180° - during which the carrying handle should be lifted slightly.

In this manner the unit will be ready for operation at an angle of 20°, but if space is limited the cover may just be turned and the unit placed on top of it, thus accomplishing a tilt of about 15°.

Photos # 1 through 4 demonstrate the above mentioned conditions.

In case instant field service is needed, a normal screwdriver for internal adjustments and an Allen key for mechanical actions are installed in the cover.

Batteries are switched off automatically by a small wedge in the cover, when the unit is closed.

The rear profile of the cabinet forms the terminal panel where 8 NEUTRIK-XLR-connectors are used for the main in- and outputs to the unit.

Interface connections are made via a 37-pole DELTA receptacle and monitor outputs for two headphones in parallel are available on two 1/4-inch stereo headphone sockets - photo # 6.

Signal levels from tape may be adjusted directly above the corresponding NEUTRIK-receptacles, and to obtain a safe catch of the screwdriver guide tubes lead directly to the potentiometers.

All connectors are terminated in a PC-board parallel to the terminal panel, and this board also contains the RF-filters for the inputs, the interface buffer-/inverter-amplifiers and the BERG-pin-connectors for the plug-in connections to the main-board.

When looking at the terminal panel one may notice, that all of the normal locking devices are removed from the armatures and this is done deliberately, because it is much more pleasant to accept a moment's interruption of a microphone, than it is to pick up the entire unit from the floor, if somebody accidentally stumbles over a cable.

Despite the terminal panel seems to be an integrated part of the cabinet, this is also a modular building block, but unless it is urgently necessary any attempt of further dismantling should be avoided.

However, the terminal panel itself is fastened by five 3 mm screws, two of which are found in each side profile of the cabinet and the last one in the upper part of the support rail next to the power section.



After removal of these screws the complete terminal panel may be drawn 10 to 12 mm backwards out of the cabinet - carefully and absolutely straight without twisting - until the BERG-connectors are disengaged, and this action will leave you with the complete terminal panel in your hand.

The bottom panel is held in place by the four profiles of the cabinet and it also forms a sandwich construction with the main-board.

When two screws connecting the main-board and the bottom-plate are taken out, the safer way of further dismantling is to take off one of the side-profiles also before the bottom-plate is removed from the cabinet.

## Input cassettes.

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The units are open cassette-types of which the main board and a parallel shielding plate form a rigid sandwich construction firmly fastened to the front panel and a board carrying switches and potentiometers - photo # 7.

The separate building blocks are held together by a few screws and stud-welded, threaded spacers on the back of the front panel, and the electric intercabling takes place via board-to-board mini-connectors.

The same type of connectors, gold plated Du Pont / Berg, are used between main board and cassettes, and a precision guiding mechanism is needed to protect the small pins - an action carried out by 6 mm tapered stand-offs firmly positioned in the main PCB and fitting sideways precisely between the cassette board and the shielding plate, back and forth on the outer sides of the spacers between them.

The stand-offs have reached their full diameter before the male connector of the main board and the counterpart of the plug-in unit go together and as individual positions of stand-offs, connectors and spacers are all exclusively referred to the PCB-lay-outs, the relative positioning of the cassette and the main board is kept within 0,2 mm in all directions, when the unit is plugged into the cabinet.

Front panel size of an input cassette is 200 x 52 mm with knobs for input selector and potentiometers arranged in the upper part of the area, while a 65 mm P & G - fader and three toggle switches occupy the lower.

Potential users found it important, that the fader area was kept as clean as possible, for which reason the 65 mm fader was chosen instead of a 100 mm type with knobs placed beside it.

The input-cassettes are fastened with two hex-screws, one in each end of the front panel, and when both screws have been removed the units may be lifted straight up - carefully - by the rotary knobs.

A preferred type of microphone may be chosen individually for each input channel by a three-position sliding-type-selector in the shielding-plate, where an 18-pole BERG-jumper sets the input circuit for dynamic- or condenser- microphones of 12 V line-feed- or 48 V phantom-types.



## Monitor cassette.

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The only mechanical differences between this unit and the input cassettes are the reduced front panel width of 30 mm and a firm interconnection between potentiometers, switches and main board - photo # 8.

## Instrument cassette.

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This cassette has front panel measures of 240 x 50 mm and is situated in the rear part of the cabinet where it also forms the mechanical basis for the monitor- and input-cassettes - photo # 9.

Four screws are used to secure the unit in the cabinet - two in the rear edge of the front panel, one in the left hand side profile of the cabinet and the last one in the support rail next to the power section.

The mechanical construction is similar to the input units except for the instrument housing, which is another separate plug-in unit, fastened by screws from the back of the front plate.

## Power unit.

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This is a closed plug-in cassette with a 250 x 75 mm front panel, housing the battery compartment and the electronics section for monitoring of the instant power status.

The unit may be lifted up by the handle, when battery cover and batteries are removed and two hex screws in the front panel taken out - as shown on photo # 10.

Batteries are loaded through a removable cover in the terminal panel and piled into four acid-proof glass fibre tubes each holding four IEC/R14 - type batteries.

The power unit is provided with precision guides and connectors like the other cassettes, and the spring loaded battery terminals are all coded to prevent any damage if batteries should be incorrectly loaded.

Two plug-in fuses are used for protection against reversed polarity of an external power source only, and the fuses are found in the right hand side of the main board below the monitor cassette.

## Sub-modules.

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The generator unit, two individual output-/limiter amplifiers and the 48-volt converter for the microphone phantom power are four closed modules, one of which is situated below each input channel 1 to 4 respectively and retained in position by these.

The modules are mechanically identical - photo # 11 - with measures 165 x 32 x 21,5 mm (L x W x H) and electrical connection is established via two 7-pole BERG-contacts, one in each end.



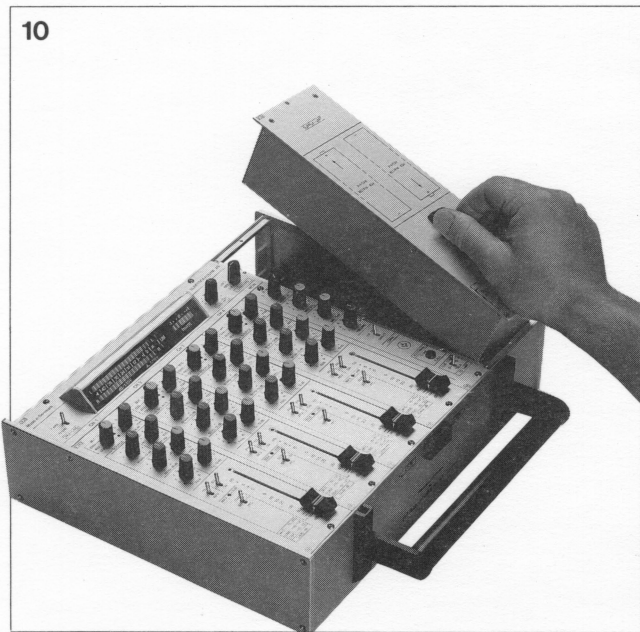
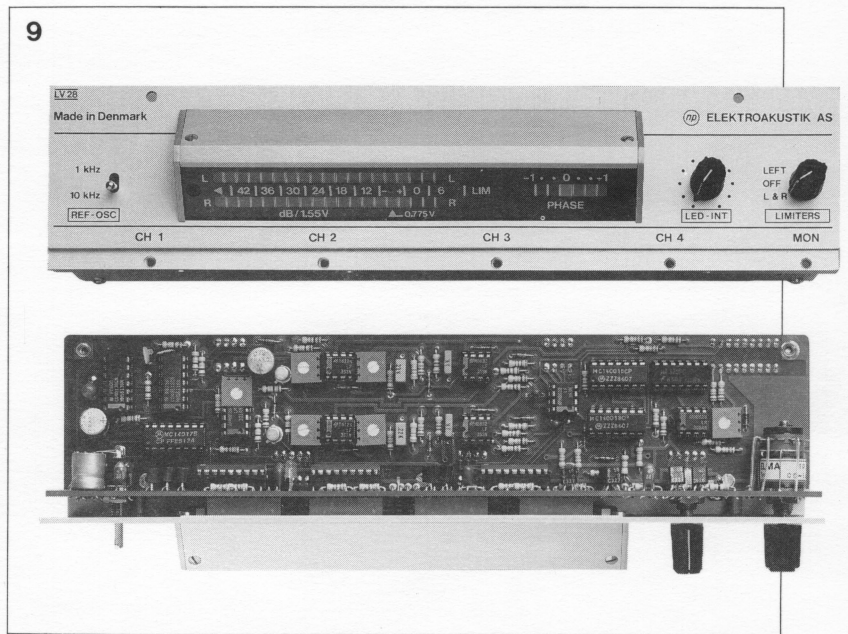
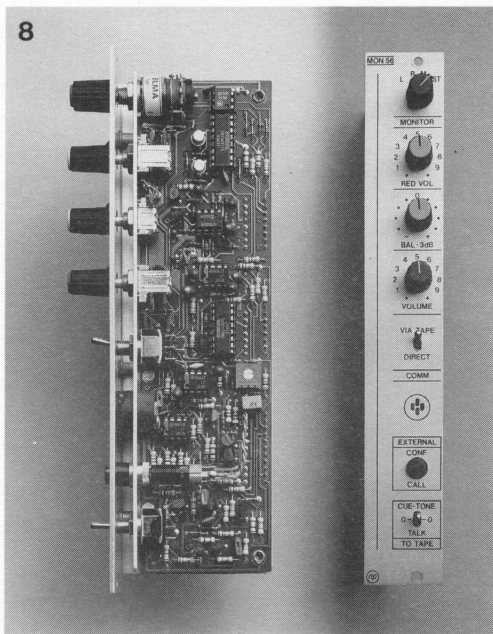
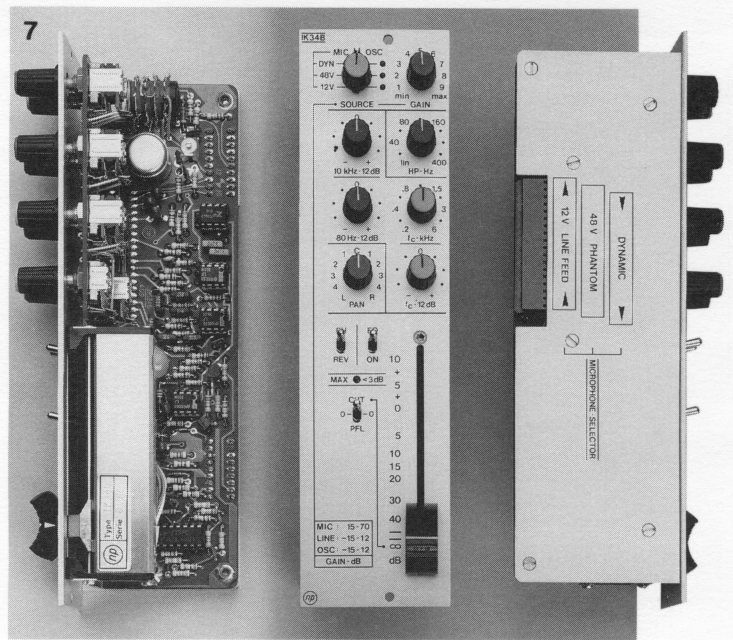
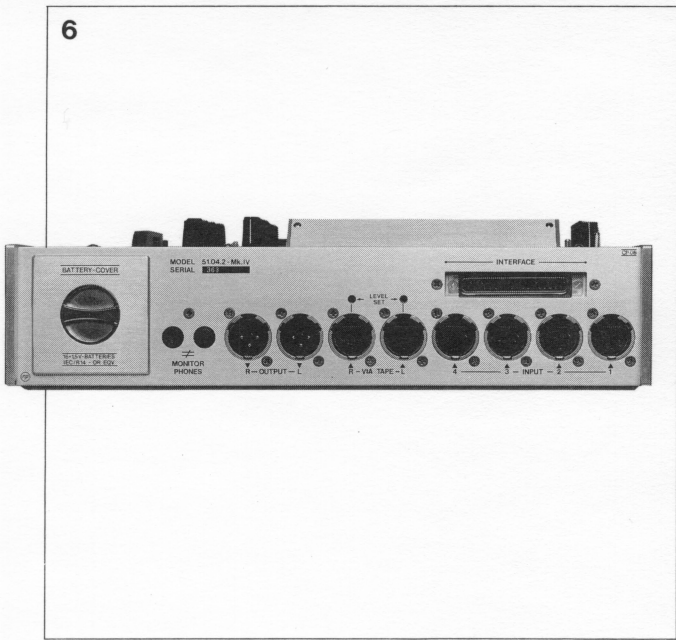
Mechanical guidance during insertion is obtained by two 4 mm POM - stand-offs in the main board, differently positioned for the three types of modules and hence acting as an efficient coding which means, that position and installation are correct if a module may be inserted at all.

The four sub-modules may be lifted up directly from their coded positions and it may be mentioned, that the output-/limiter units are identical and hence interchangeable.

Some internal adjustments may be reached from the outside through holes in the cover, but no attempt should be made unless by trained personnel.

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## 03 --: ELECTRONICS DESIGN AND LAY-OUT

### General considerations

As mentioned earlier, low-noise design takes current and a high signal to noise ratio means voltage, but also the auxiliaries such as the 48 volt-converter, monitor- and instrumentation systems, reference- and cue-tone-generators need power and hence a fair number of batteries with a proper capacity are needed in order to obtain a reasonable amount of operational time.

We deal with the same design rules as are valid for the stationary studio equipment, but due to the limitation of physical size and available, portable power the different weight of some parameters must be considered.

The resulting sound must be clear, distinct and transparent and the major requirement to achieve this is a wide, well defined bandwidth independent of any random slew rate limitation.

Another important factor in high fidelity processing of sound is, that if tonal balance must be corrected at all by equalization, only soft-curve filters should be used in a very careful way because of the phase response problems related to frequency dependent circuitry.

Different group delays may deteriorate sound transparency and clearness severely but when stereo is involved - M/S- or A/B-techniques - a total disaster may be the result, because the stereo image and localisation of sources are based on amplitude- and phase-equality in both channels.

According to the above mentioned points the upper transfer frequency is limited only in the front-end of the chain, and the overall roll-off response of the equipment is -6 and -20 dB at 100 and 200 kHz respectively.

Within this frequency range great care has been taken to avoid any kind of slew rate limitation caused by op-amp compensation etc, and the phase response within the audio range is practically only influenced by equalizer- and filter settings, for which reason the equalizer-circuitry has a limited effect of 12 dB per decade and the high-pass filter-slope is set to 12 dB per octave.

To obtain a symmetrical input, correct impedance matching and a good common-mode rejection input transformers are used for microphone- and line inputs, but we do not see any need for output transformers in a battery-powered equipment this kind, and hence the output signals are asymmetrical and referred to 0-volt and chassis.

The connection of this unit to another battery-powered one or to a mains operated system - recorder or console - will not cause any transfer- or reference problems at all.



The demand for a symmetrical output mainly occurs, when long signal lines must be run through noisy environments, but none of our many users have ever seen this as a practical problem, so we found it unreasonable to incorporate the extra space and weight of suitable output transformers just to cope, because the remedy in such few cases is simple - an external line transformer.

Depending on the way of use the operational period of time may be longer than thirty hours with alkaline batteries, but some facilities have been incorporated to bring down the idling power.

The PFL-switch of each input channel has a third position named CUT, and when the fader contact closes this function means, that battery supply is taken away from all pre-fader amplifiers - without any click or noise because of the closed fader - and thus reduces channel current by 75 %.

LED instruments, two PPMs and a phase indicator, are incorporated to make monitoring as effective as possible but, again, to reduce power consumption, we preferred a multiplexed LED system with a flying spot presentation instead of a bar graph display.

Intensity of LED instruments is made variable to cope with the difference of ambient light and the PPM-scale is provided with constantly lit marker LEDs in the bottom end and at the 0,775 volts (-6 dB) reference.

A conference system is built into the monitor in case there is a need for verbal contact with a distant director or stage technician, but the push-button for this purpose may also control remote start or light-signals.

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# ELEKTROAKUSTIK AS

## 04 -: GENERAL ELECTRICAL SPECIFICATION

The values given here are general average data as measured on a number of np PORTABLES, but for exact information of specific unit data, please refer to the individual, numbered performance protocol.

Four input channels -:		MIC*)	LINE
Input impedance		2k ohm	18k ohm
Amplitude vs frequency response	40 Hz-20 kHz	0 dB	0 dB
Amplitude vs frequency response	20 Hz-70 kHz	-3 dB	-3 dB
Common mode rejection ratio	15 kHz	>60 dB	>50 dB
Min gain at fader setting 0 dB		15 dB	-15 dB
Max gain at fader setting 0 dB		70 dB	12 dB
Output noise (RMS / gain 15 dB)	22 Hz-22 kHz	-92 dBu	
Output noise (RMS / gain 70 dB)	22 Hz-22 kHz	-55 dBu	
Output noise (RMS / gain 0 dB)	22 Hz-22 kHz		-87 dBu
THD+N (input level 6 dBu, gain 0 dB)	1 kHz		0,02 %
THD+N (input level 6 dBu, gain 0 dB)	40 Hz-20 kHz		<0,05 %
THD+N (input level -24 dBu, gain 30 dB)	1 kHz	0,01 %	
THD+N (input level -24 dBu, gain 30 dB)	40 Hz-20 kHz	<0,30 %	
Max single channel output - pre or post fader		----	18 dBu ----
Output impedance		----	<50 ohm ----
Output load impedance - min		----	2k ohm ----
Fader buffer gain		----	10 dB ----
High-pass filter - 12 dB/oct. / f <sub>1</sub> =	20 Hz-400 Hz	----	-3 dB ----
Equalizer - shelving	80 Hz	-- +/-	-12 dB ----
Equalizer - shelving	10 kHz	-- +/-	-12 dB ----
Equalizer - bell - Q <sub>max</sub> app. 1 / f <sub>c</sub> =	200 Hz - 6 kHz	-- +/-	-12 dB ----

\*) M/S-microphones for stereo recordings may be used immediately without any extra accessories, because the necessary matrix circuitry is built into the np PORTABLE already.

Two output-/limiter channels -:		
Max output level		18 dBu
Nom output level - (PPM reads 0 dB for 1,55 V)		6 dBu
Output impedance		<10 ohm
Output load impedance		600 ohm
Limiter input level for 1 dB limitation		7 dBu
Limiter tracking error between channels		<0,3 dB
Limiter ratio, nominal		50:1
Limiter attack time		1,5 ms
Limiter release time		1,5 S/10 dB
Limiter THD+N (input level 18 dBu)	1 kHz	<0,1 %
Limiter THD+N (input level 18 dBu)	40 Hz-20 kHz	<0,2 %
Output noise (faders closed, lim off)	22 Hz-22 kHz/RMS	-98 dBu
Output noise (faders closed, lim on)	22 Hz-22 kHz/RMS	-92 dBu



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Monitorsystem -:

Max output level - phones	18	dBu
Output impedance	<10	ohm
Output load impedance - min	600	ohm
Output noise	22 Hz-22 kHz/RMS	-98 dBu
THD+N via monitor	<0,01	%
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Input level via tape - adj (min / max)	-14 / 6	dBu
Input impedance - via tape	10k	ohm
-----		
Max output level - via talk back limiter	8	dBu
Output impedance	<150	ohm
Output load impedance - min	2k	ohm

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Monitor instruments -:

PPM-measuring range ref 1,55 V - in steps of 3 dB	-45 dB - +6 dB
PPM-frequency response (+/-1 dB)	31,5 Hz - 20 kHz
PPM-fall-back speed	10 dB / S

PPM-attack response at 5 kHz sine-burst with zero-cross switching -:

Pause	Burst	Meter reads*)	NORMAL N 9**)
-74 dBu --- 5 S	6 dBu --- 10 mS	-3 dB	-1 dB
-74 dBu --- 5 S	6 dBu --- 5 mS	-3 dB	-2 dB
-74 dBu --- 5 S	6 dBu --- 3 mS	-6 dB	-4 dB
-74 dBu --- 5 S	6 dBu --- 0,4 mS	-15 dB	-15 dB

\*) PPM -: Readings are all within the N 9 NORMAL regarding 3 dB steps.

\*\*\*) N 9 -: NORDIC RECOMMENDATION FOR BROADCAST EQUIPMENT - PPMs.

Phase meter reading vs phase difference -:	Diff	Reading
2 signals present - in-phase	0°	+1
1 signal missing or 90° out of phase	90°	0
2 signals present - one phase reversed	180°	-1
-----		
Phase meter sensitivity	-:	-36 dBu
Phase meter frequency range	-:	31,5 Hz - 20 kHz

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Oscillators -:

Reference oscillator, 1 kHz or 10 kHz, sine	6	dBu
Cue tone oscillator, 15 Hz composite output	6	dBu



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Power supply -: 48 V microphone phantom voltage.  
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Supply current / 48 V positions not selected	< 2 mA
Supply current / 48 V - 0 mA / $V_B = +/-12$ V / 48 V <sub>out</sub>	< 8 mA
Supply current / 48 V - 1 mA / $V_B = +/-12$ V / 48 V <sub>out</sub>	< 12 mA
Supply current / 48 V - 20 mA / $V_B = +/-12$ V / 48 V <sub>out</sub>	< 73 mA
Supply current / 48 V - 20 mA / $V_B = +/- 8$ V / 48 V <sub>out</sub>	<100 mA

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Power supply -:  $V_B = +/-12$  volt.  
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Supply current, min / excl. 48 V generator -:	150 mA
Supply current, max / incl. 48 V generator -:	250 mA
Voltage monitor LED - constant light -:	$V_B > 8,5$ V
Voltage monitor LED - flashing light -:	$8,5$ V > $V_B > 8,0$ V
Voltage monitor LED - off -:	$8,0$ V > $V_B$

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The graphics presentation over the next pages is carried out and recorded by automatic equipment from AUDIO PRECISION, and the following specifications are listed -:  
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Fig. 01 -: Microphone input frequency response.

Fig. 02 -: Microphone input noise versus gain.

Fig. 03 -: Microphone input gain and THD+N versus frequency.

Fig. 04 -: Microphone input common mode rejection versus frequency.

Fig. 05 -: Line input frequency response incl. variable high-pass filter.

Fig. 06 -: Line input noise versus gain.

Fig. 07 -: Line input gain and THD+N versus frequency.

Fig. 08 -: Line input common mode rejection versus frequency.

Fig. 09 -: Equalizer shelving response, 80 Hz and 10 kHz.

Fig. 10 -: Equalizer bell response at 200 Hz, 1kHz and 6 kHz.

Fig. 11 -: Limiter gain versus input level.

Fig. 12 -: Limiter tracking error versus input level.

Fig. 13 -: Fader attenuation versus frequency.  
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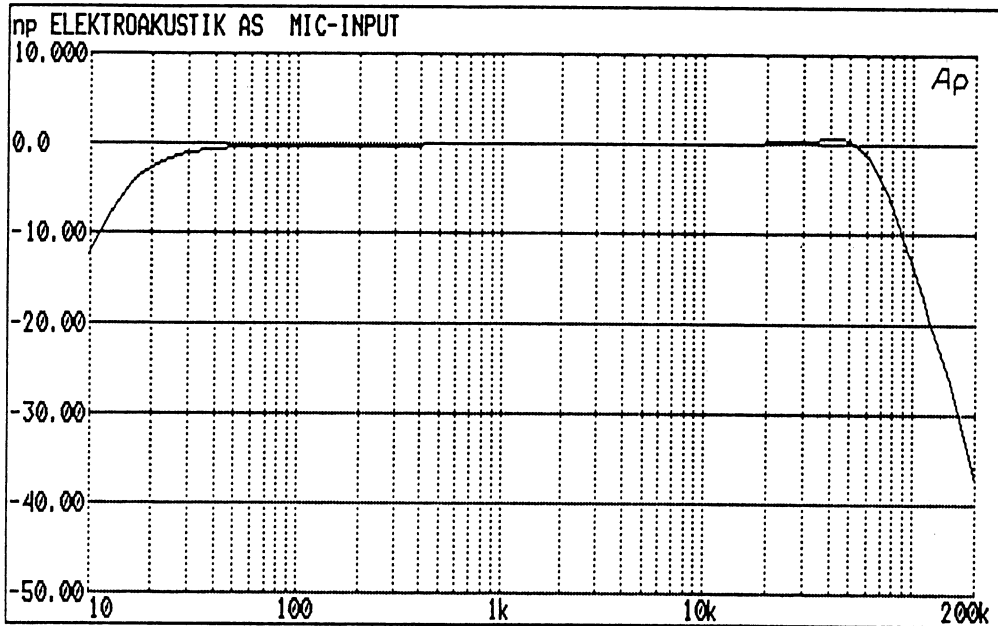
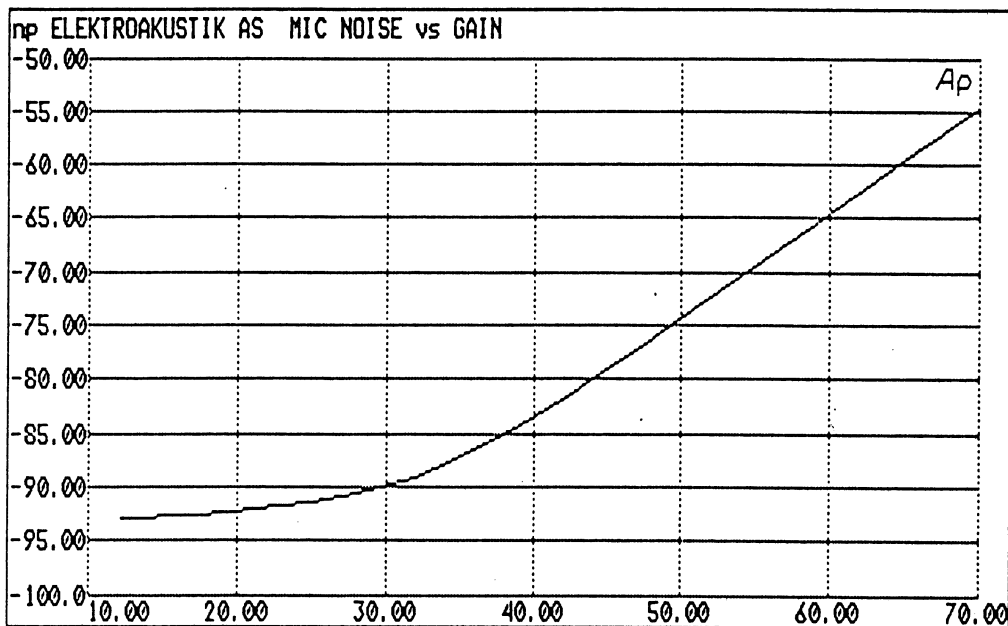


Fig. 01 --: Microphone input frequency response -  
input level -60 dBu / gain 60 dB.

Fig. 02 --: Microphone input noise versus gain -  
measured and plotted in 10 dB steps.



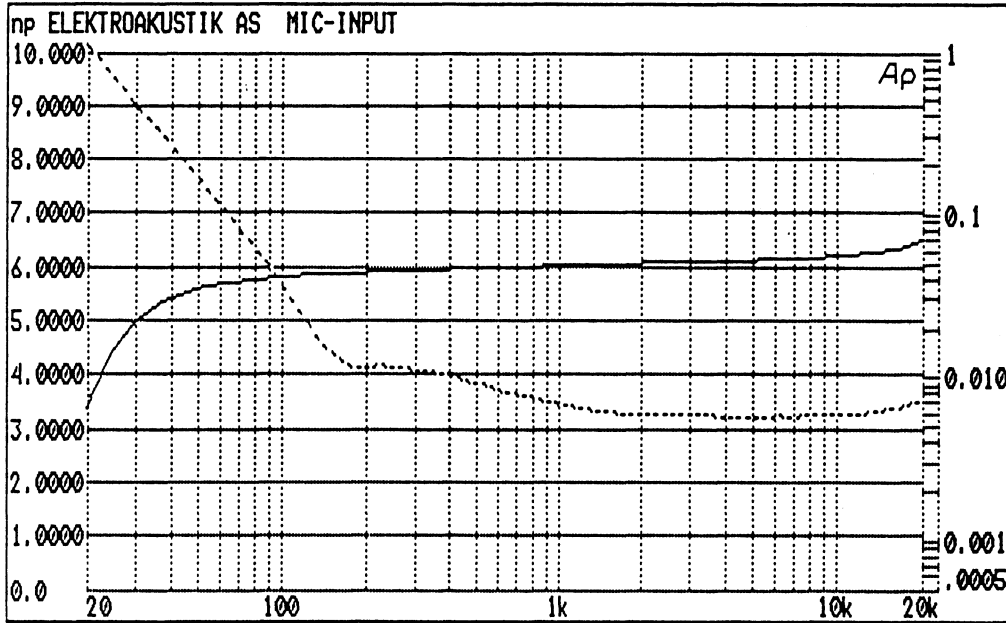
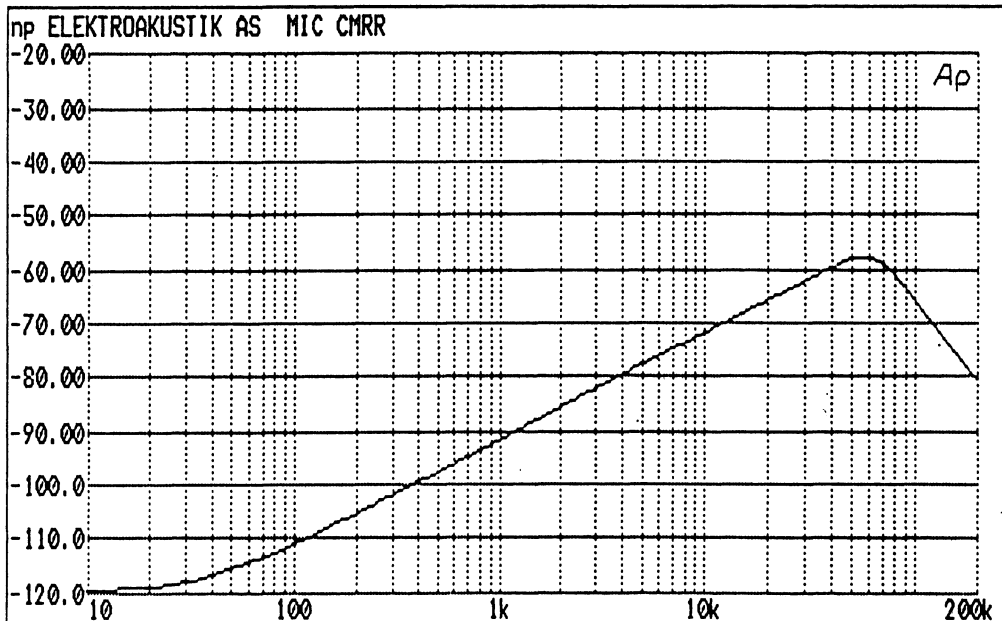


Fig. 03 --: Microphone input gain and THD+N versus frequency / THD+N right hand scale.

Fig. 04 --: Microphone input CMRR versus frequency.



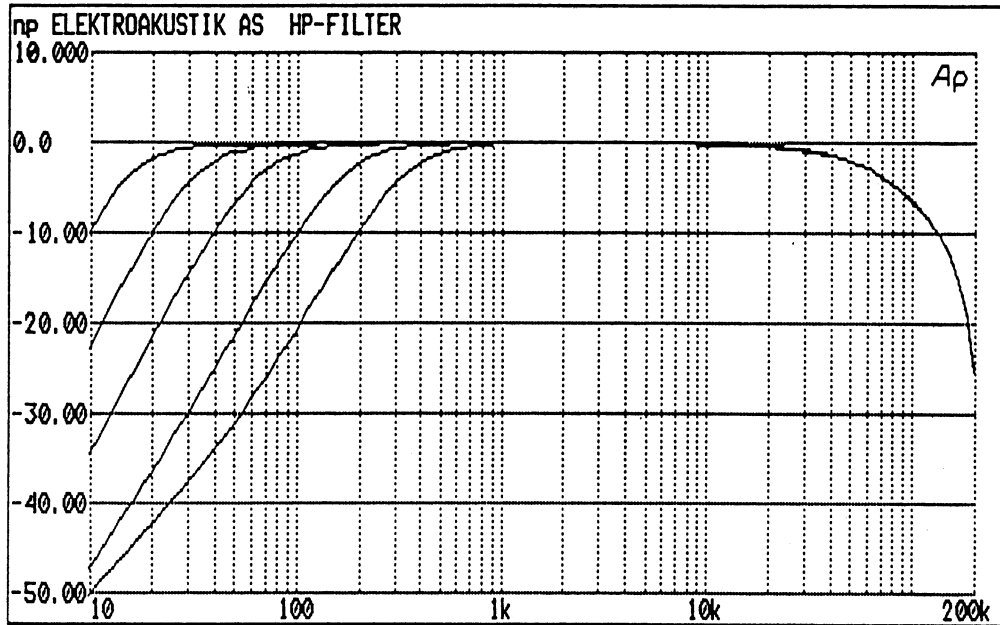
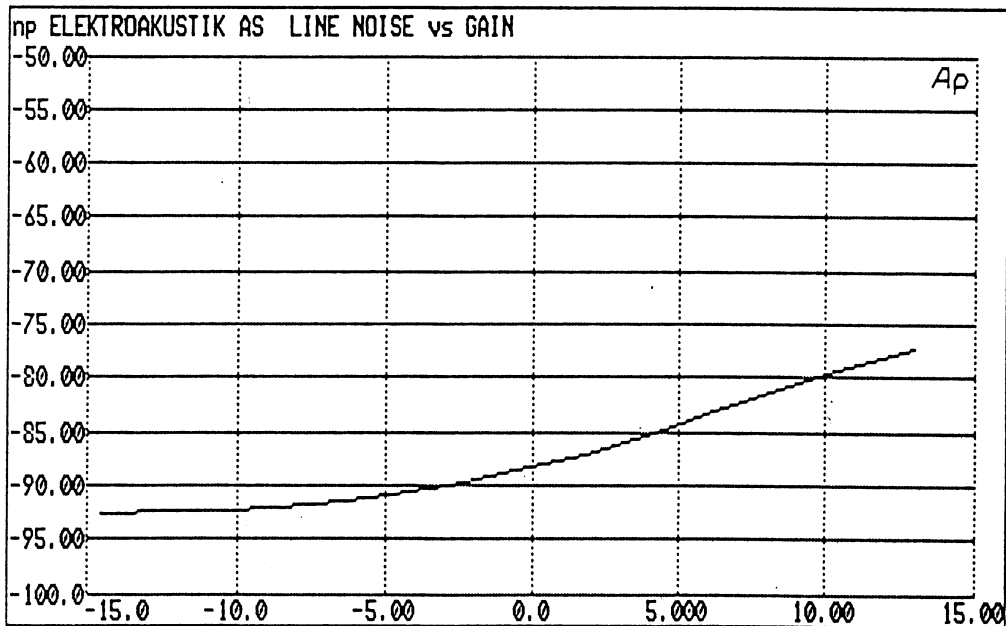


Fig. 05 --: Line input frequency response with HP-filter in pos --: 11n, 40, 80, 160 and 400 Hz.

Fig. 06 --: Line input noise versus gain - measured and plotted in 5 dB steps from min to max gain.



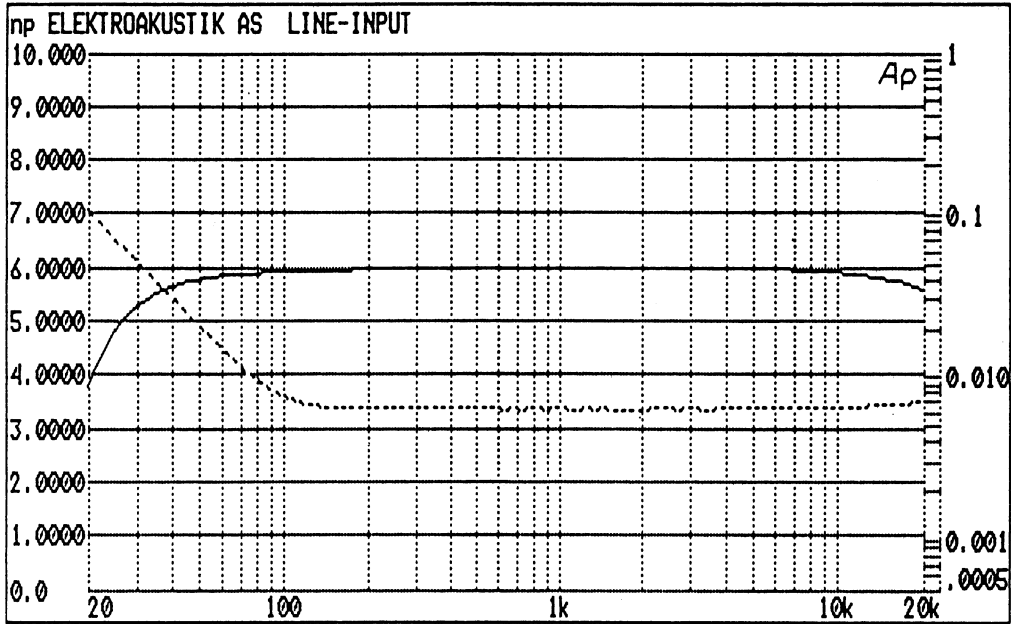
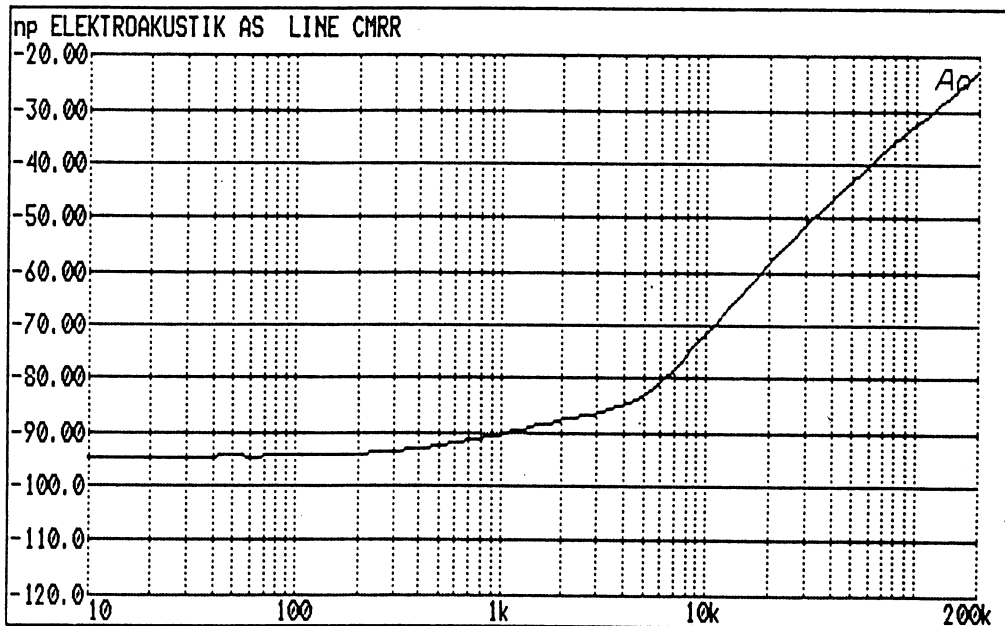


Fig. 07 --: Line input gain and THD+N versus frequency - THD+N right hand scale.

Fig. 08 --: Line input CMRR versus frequency



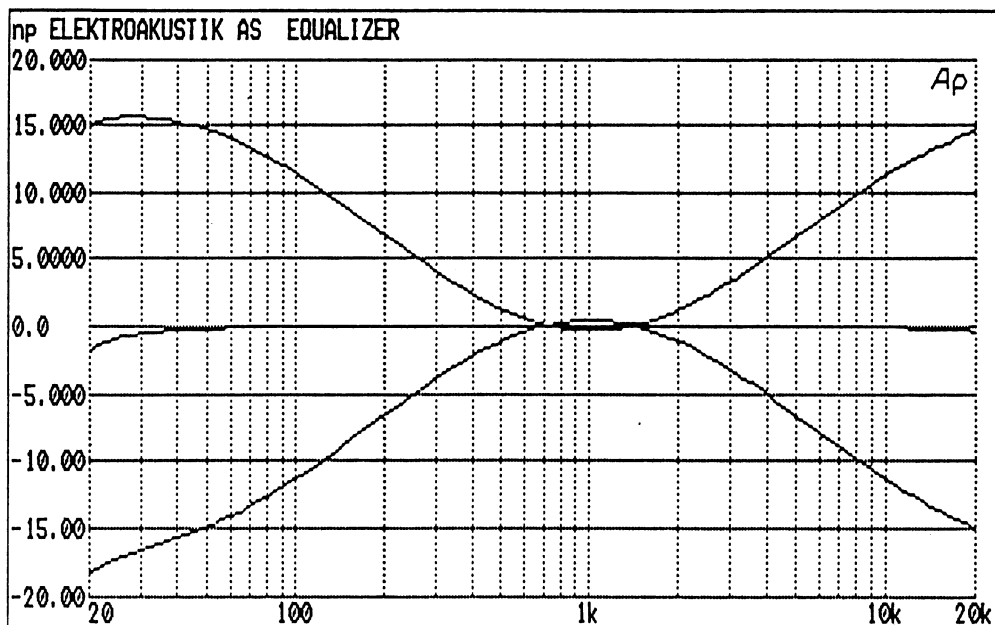
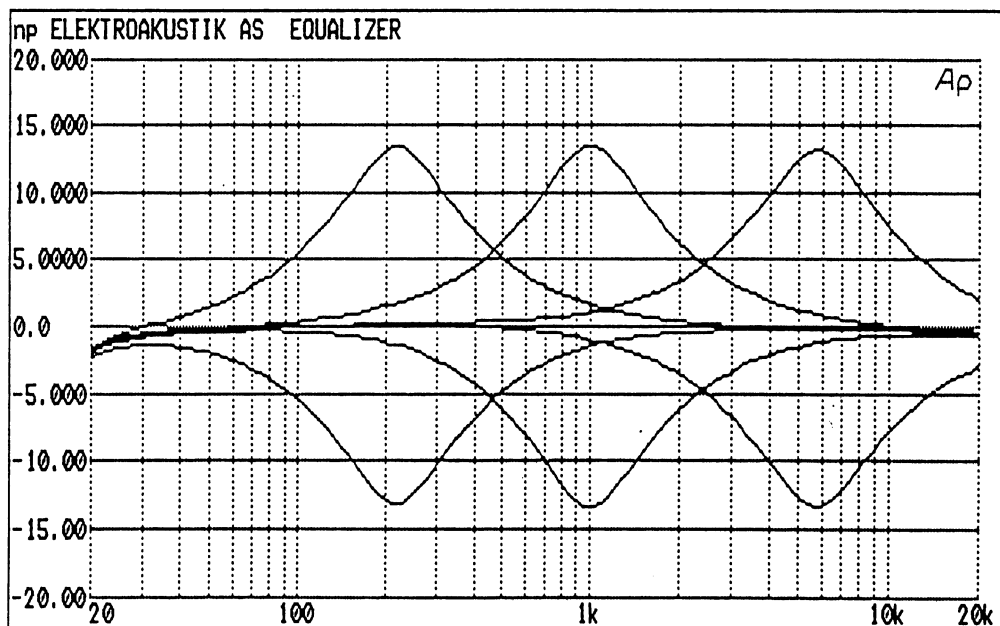


Fig. 09 --: Equalizer shelving response, 80 Hz and 10 kHz controls at max and min settings.

Fig. 10 --: Equalizer bell response, 200 Hz, 1 kHz and 6 kHz at max and min settings.



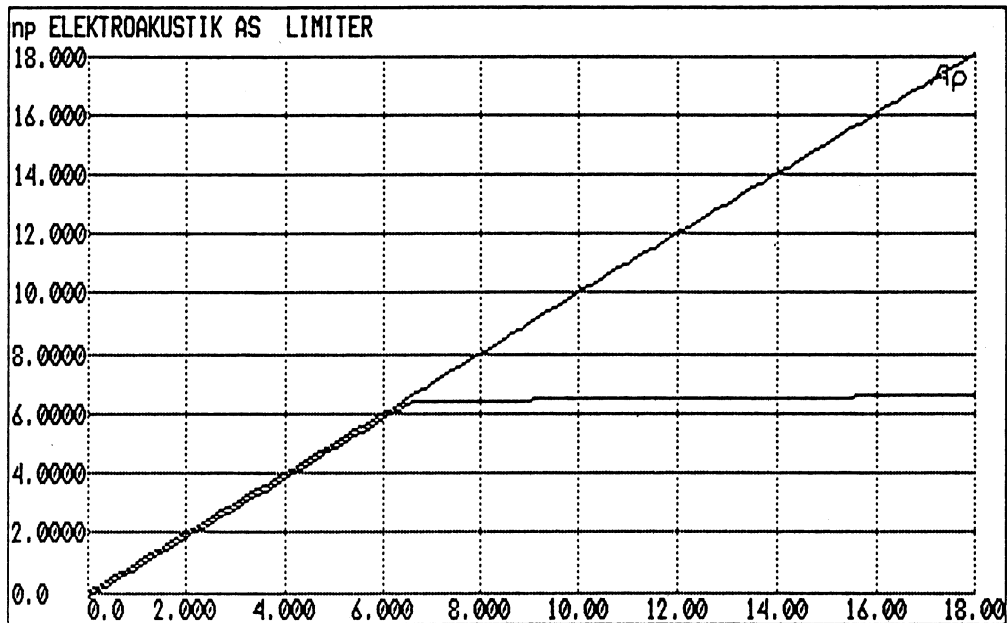
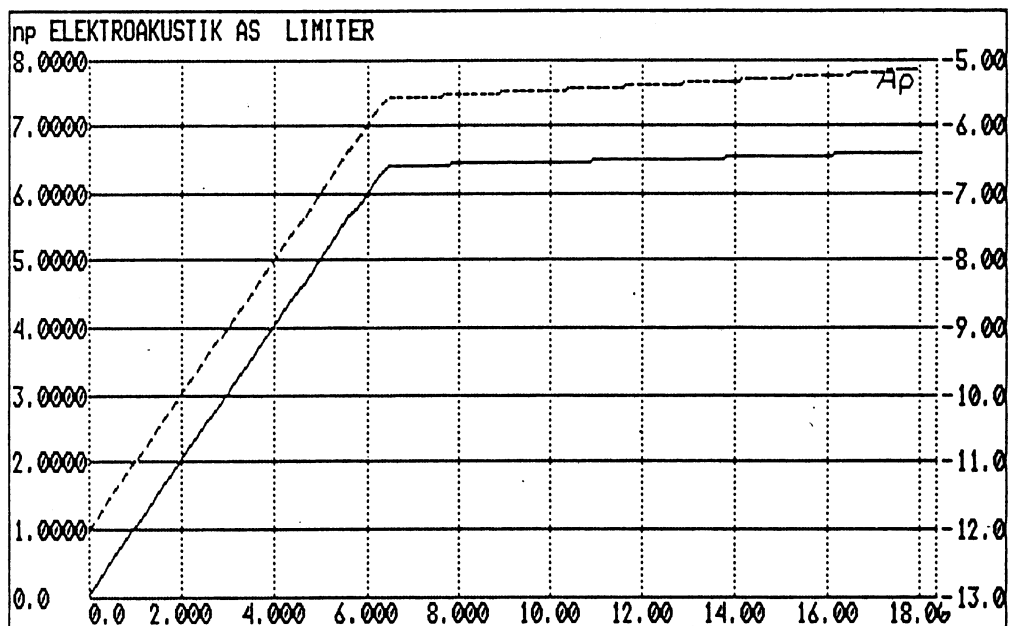


Fig. 11 -- Limiter on and off - versus input level.

Fig. 12 -- Limiter tracking error versus input level.  
Lim # 1 (solid line, left scale) controls  
Lim # 2 (dotted line, right scale), which  
is inactive at a 12 dB lower level.



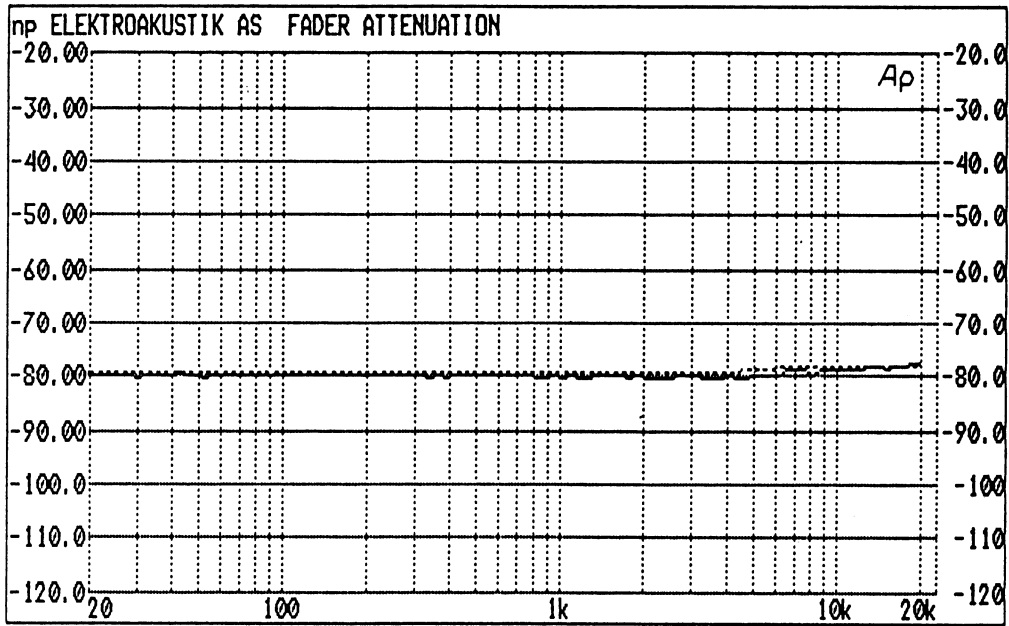
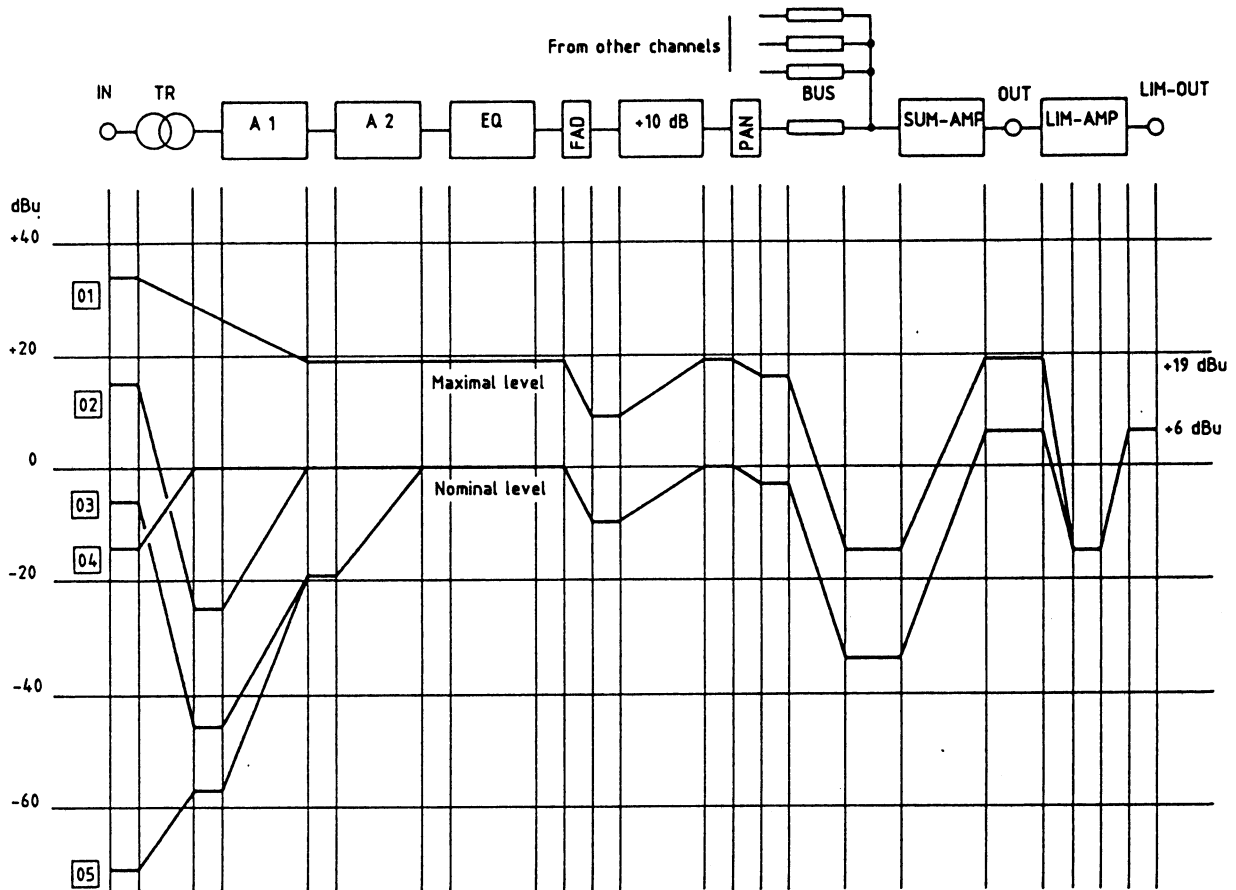


Fig. 13 --: Fader attenuation and cross talk between channels versus frequency.

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**05 --: INTERNAL SIGNAL LEVEL DIAGRAMME**

This drawing demonstrates the internal signal levels along the amplifier chain of the PORTABLE, and the maximal values depend on the available operational voltage, +/-  $V_B$ .

Curve 01 --: Here the maximal levels are shown for LINE-input at minimum gain and with +/-12 V at disposal.  
 If  $V_B$  drops to +/-5 V the unit may still be working, but the maximal levels will be about 10 dB lower and no LEDs are active any longer.

Curve 02 --: Nominal levels - LINE-input at minimum gain.

Curve 03 --: Nominal levels - LINE-input at maximum gain.

Curve 04 --: Nominal levels - MICR-input at minimum gain.

Curve 05 --: Nominal levels - MICR-input at maximum gain.

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06 --: DESCRIPTION OF ELECTRICAL CIRCUITRY  
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06.01 - 5104.2/IV  
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Drawing # 88032761 is an overall schematic of the portable field unit, np type 5104.2./IV, showing the building blocks with their interconnections, and the intention is to convey a quick and convenient general view of the equipment.

In the left hand side of the drawing input channel # 1 is partially shown followed by three identical boxes to illustrate channels 2, 3 and 4.

Four sub-modules are next --: reference- and cuetone generator, two identical output amplifiers and the 48 volt phantom power supply.

The right hand side of the drawing features the monitor unit and battery power supply, and the upper part of the drawing is occupied by the terminal panel and the instrument unit.

Detailed information of each component will, however, be referred to specific diagrams, but the overall schematic may be valuable in order to obtain a general view of the internal relations.

06.02 - IK 34 B  
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Drawing # 85030511 displays the electronics of an input channel, IK 34 B, and to the far left the microphone selector, XX35 PCB, is seen.

This three-position jumper assembly is installed in the shielding plate of the input cassette and changes some internal connections via the main board, when the cassette is inserted.

Jumpers on pins 1/2/3/4/5 are used to indicate selected type of microphone via LEDs D9, D10 and D11 - named DYN, 48 V and 12 V.

In position 48 V a jumper on pins 7/8 connects the 48 volt generator output to the microphone line and the jumper on pins 10/11 lets the cassette enable the generator.

In position 12 V LINE FEED jumpers on pins 13/14 and 17/18 apply 0 volt and +12 volt to microphone input pins # 2 and 3 respectively via two 180 ohm resistors - placed in the main board as R 1/2 through 7/8 for input channels 1 to 4.

An electrolytic capacitor to prevent DC loading of the input transformer is also placed in the main board and named C 1 through 4 for respective input channels, but in positions DYN and 48 V the capacitor is shortened by the jumper on pins 15/16 and 16/17.



# ELEKTROAKUSTIK AS

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An eight section rotary switch, O 1, selects MIC, LINE or OSC as a source and it also controls some auxilliary functions -:

In position MIC the R 40 / C 22 complex brings Q 4 into conduction for a short period during which time LEDs, D 9, 10 or 11, will indicate the selected type of microphone - a way of indication chosen in order to reduce battery current.

Starting the 48 volt generator and the oscillator is also accomplished by this switch in positions MIC and OSC, but it's main purpose is to connect the transformer in different modes and set the amplification range of input amplifier, A 1.

In MIC-position A 1 acts as a non-inverting amplifier with variable gain from 0 to 34 dB and with an additional 14 dB step-up of transformer, T 1, the total gain between MIC-input and the output of A 1 may be varied from 14 to 48 dB by the potentiometer, P 4-II.

The second input stage, A 2, which incorporates the second order HP-filter, is regulated by the same potentiometer, P 4-I, and whether MIC, LI or OSC is selected, the A 2 maintains a variable gain of 0 to 22 dB.

Added to the output from A 1 a total gain of 14 to 70 dB is available before the signal enters the equalizer at a nominal level of 1,55 volts.

In the LI- and OSC-positions, however, T 1 is used as a current converter with it's secondary connected to A 1's inverted input via C 5, and in order to maintain the correct phase as related to the input, opposite polarity is applied from the secondary winding.

LI-input to the primary of T 1 is supplied through symmetrical resistors, R 2 and R 3 and the gain range of A 1 is adjusted to LI-level by disconnecting R 6 from 0 V.

By this action A 1 (incl T 1) obtains a reduced, variable sensitivity of -15 to -10 dB, and followed by the A 2 variation of 0 to 22 dB the total LI-input gain may be varied from -15 to 12 dB.

OSC-input is asymmetrical and arrives through R 1 to one side of the primary of T 1 - the other being grounded - and the oscillator is started if one of four input channels is set to OSC.

As mentioned earlier the upper transfer frequency of the system is limited only in the front end of the amplifier chain, where R 12 / C 11 etc. act as a passive low-pass filter between pre-amplifiers A 1 and A 2.

Depending on the position of equalizer switch, O 2, the output from A 2 is available either direct or via the equalizer circuit, built around a dual FET-amplifier, Q 3.

First section of Q 3 has two functions as it - 1) operates the Baxandall equalizer at 80 Hz / 10 kHz and - 2) combines this circuit with the presence-/absence equalizer of the second section, in which up to 12 dB cut or boost may be applied to a variable  $f_c$  between 200 Hz and 6 kHz.



The equalizer circuit inverts the signal and in order to maintain the same phase with or without the equalizer O 2 works in conjunction with O 3, which is the phase inverter switch.

O 3 controls the inverter action of Q 6, and as the signal is not interrupted during the phase shift, this operation is practically noiseless.

From the output terminal of Q 6 three separate signals are obtained - one for the P & G fader, another one for the PFL-function via FET-switch Q 7, and the third one as a direct channel output via S 3.

S 3 is an internal jumper switch on the input channel's PCB, and the idea is to place a direct signal at disposal - pre or after fader and independent of the mixed signal - intended for a PA-unit or other purposes.

Via the P & G fader, P 8, the signal passes a 10 dB, low noise amplifier, Q 9, in order to obtain a suitable reserve, and from here it goes directly into the pan-pot assembly and on to the left and right summing busses.

The pan-pot is designed to maintain an equal stereo sound pressure - independent of panned position - and a 3 dB loss is introduced in the center position.

The PFL-signal is taken to the PFL-summing amplifier via FET-switch Q 7, and when PFL/CUT-switch, O 4, is in position PFL, Q 7 conducts and a control voltage is - via D 1 - sent to the monitor unit, where both monitor channels are shifted to the PFL-output bus, hence presenting one or more PFL-signals as a monitor output - including both PPMs.

If PFL/CUT-switch, O 4, is placed in CUT-position, the  $\pm V_B$  control circuit, Q 10, 11 and 12, will turn off the supply voltage to all circuitry pre fader, when the fader switch closes.

This is done in order to reduce battery current by about 75 %, when an input channel is closed, and as the fader stands in the bottom position in the moment of switching, no noise problems occur when the fader is activated, but if, however, a direct signal is taken out pre fader via S 3, the CUT-function should - for obvious reasons - be avoided.

The last thing to mention is the MAX-indicator LED, D 12, and the belonging control circuitry, built around Q 8, a quad comparator.

The outputs from A 1, A 2 and the equalizer are monitored constantly via peak rectifiers, D 3 to D 8, and if one of these signals tends to exceed a value 3 dB below the clipping level, the fast MAX-LED will show it.

This indication is just a warning before anything happens, but the maximum signal level depends on the present battery voltage which means, that the clipping limit will vary as well.

However, the comparator's reference is derived from the battery voltage, and hence the indication is maintained at a constant 3 dB distance to the critical point - independent of the present battery status.



06.03 - MON 56  
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Drawing # 86110961 shows the circuitry of the stereo monitor and the conference-/talk-back systems, while drawing # 83083111 features the details around the talk-back microphone with its limiter amplifier.

The various building blocks are mainly interconnected via the main PCB - drawing # 88032761 - and it may be a good idea to consult this drawing as well in order to obtain a clear view of the system.

Inputs to the monitor are selected by the DIRECT/VIA TAPE-switch, O 3, in the front panel of the unit, but the selected signal may be overridden by the PFL-signal, if pre-listening is effected in one or more of the input channels.

In the DIRECT-position left and right output signals arrive directly from the output amplifiers, with or without limiter function, into pins 13 and 15, and via O 3 and FET-switches Q 23 these signals are taken out on pins 12 and 16, named MON L and MON R.

The function of Q 23 is controlled by the PFL-switches in the input channels, and Q 23 applies the PFL-bus signal to both monitor channels when a PFL-switch is activated and sends a control signal to pin 9.

This means, that the normal monitor signal, DIRECT or VIA TAPE, is interrupted by the PFL-signal, which is now presented in both monitor channels - including the two PPM-instruments.

In the VIA TAPE-position signals to the monitor system come from the XLR-female connectors in the terminal panel, where correct level and balance may be set by two rotary potentiometers, before the signals are taken into the monitor on pins 17 and 18, from where they are applied to the buffer amplifiers, Q 24, each having a gain of 20 dB.

VIA TAPE-signals from Q 24 may thus be equalized to the DIRECT-level in order to compare the results at an even basis before and after recording.

MON L and -R signals are connected to the monitor amplifier assembly on pins 25 and 26 respectively, and furthermore they are led to the instrument unit, where they are monitored by two PPMs and a phase instrument.

The rotary switch, O 4, has three sections of four positions each, named ST, M, R and L, where the R and L-positions send R or L-signals to both of the monitor amplifiers in order to allow the correct adjustment of the monitor balance.

In the M-position R and L-signals are summed via two 11 K resistors, R 70 and R 71, and the composite M-signal is sent to both monitor amplifiers.

The passive mixing across R 70 and R 71 keeps the mixed level down by 6 dB, if the R and L-signals are identical and by about 3 dB, when stereo signals are present - the intention of which is to obtain a fairly equal listening level, when a stereo programme is checked in mono.



O 4's ST-position is the normal setting and connects the R and L-signals to the R and L-sections of the monitor amplifier respectively.

Inputs to the R and L-sections of the monitor amplifiers are taken from a dual volume regulator, P 2, the input of which may be muted more or less by the RED-VOL potentiometer, P 3, acting like a variable voltage divider with R 70 and R 71 when switches, Q 12 and Q 13, are activated.

The arms of P 3 are separately grounded by Q 12 and Q 13, when muting takes place, and the idea is to avoid acoustical feed-back if monitor speakers are incorporated instead of head phones.

Control signals for the muting system are derived from both switches, O 1 and O 2, being the CONF/CALL-pushbutton and the CUE-TONE/TALK-switch, situated in the monitor's front panel.

Q 25 is a dual amplifier, that buffers the monitor signal and sends it to two paralleled 1/4'' stereo phone sockets on the terminal panel - intended for the immediate use with head phones and/or as line signals for a stereo amplifier, if loudspeakers are preferred.

It may be noticed, that P 2, named BAL-3dB, is incorporated in the feed-back path of the right channel despite this is a stereo monitor circuit, where no separate balance control should ever be found.

It is evident, however, that sensitivity and equality may vary slightly with different types of head phones - or users - and for this reason P 2 is there in order to allow a modest +/-3 dB gain variation in one channel only - ok ?

The balancing should be carried out by sending one signal only into both channels as mentioned above and it should be noticed, that this operation has no influence at all concerning the internal balance of the signals to the metering circuit.

We will go back to the input circuit again, where Q 26 is placed in order to shift the right monitor channel from the normal input - via O 4 - to a conference signal, CONF INP, that may be received from outside.

An external signal, CONF ON, controls Q 26 via the opto-coupler, Q 27, by which the connection is made potential-free, and the diodes, D 11-14, make the polarity of the incoming signal indifferent.

CONF INP arrives on pin 21 via the interface connector's pin 13, and if a stage technician or a conductor is equipped with a small conference set a two-way communication may be established between the distant partner and the np PORTABLE operator.

When a call arrives a light signal will occur in pushbutton, O 2, and the message will be present in the right monitor channel only as the left one stays on the programme.

The operator answers a call by pushing the button, O 2, which enables the V<sub>B</sub>-control circuit, Q 2, Q 3 and Q 6, for the talk-back microphone ampli-



fier and also activates the FET switch, Q 26, through which the talk-back signal is taken out to the interface connector on the terminal panel.

In addition a control signal, TALKB ON, is present on pin 8, CONF LAMP on pins 10/11 and the incoming control signal for the switch, Q 27, on pins 22/23 - which are all available on the interface connector as well.

The way to benefit from these facilities must be determined by the user, as it highly depends on the situation, but later, when the terminal panel is described, different possibilities will be exposed.

By the way - the above mentioned  $V_B$ -control circuit is another effort taken in order to bring down the power consumption where possible.

For editing purposes it is possible to place comments on tape and also a cue-tone for the easy re-location of these comments may be recorded - an action carried out by the three position toggle switch, O 1.

A spring loaded position enables the cue-tone generator from which a low frequency signal of about 15 Hz is injected into the L and R-summing busses at a level resulting in a 0-dB reading on the PPM-scale (+6 dBu).

In the editing process, when the tape is winded back and forth at a higher speed, the low frequency signal will be transposed accordingly and recognized as a distinct 'bleep', thus helping to find the marked places.

The other active position of O 1 is named TALK, in which three different things happen -:

- 1) Via D 1, Q 12 and Q 13 the muting function is established, and P 3 may be used to adjust the rate of muting if speakers are active.
- 2) Also via D 1 the  $V_B$ -control circuit, Q 2, Q 3 and Q 6, is enabled and power applied to the talk-back microphone amplifier.
- 3) Via D 2 a control signal is sent to the switches, Q 7 and Q 8, through which the output signal from the talk-back microphone amplifier reaches both summing busses, L and R, on pins 5 and 6.

Recording of the cue-tone may take place before or after the editing comment or it may be used in itself as a pointer or a flag - it should just be noticed, that IF O1 IS EVER ACTIVATED EITHER A CUE-TONE OR THE SIGNAL FROM AN OPEN TALK-BACK MICROPHONE AMPLIFIER WILL BE PRESENT AND MIXED INTO THE MAIN PROGRAMME MATERIAL.

## 06.03 - TB MIC AMP

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The talk-back microphone amplifier with it's limiter circuit is shown separately on drawing # 83083111.



A small electret capsule from SENNHEISER or AKG is built into the front panel of the monitor unit, all embedded in soft foam rubber to minimize a transfer of mechanical noise from switches etc.

The dual FET OP AMP, Q 12, is used as a microphone amplifier and limiter with a total gain of 50 dB - including 40 dB of regulation which means, that the gain may vary from 10 dB to 50 dB.

R 33 and Q 9 form the regulating element, the control voltage for which is produced by the second dual FET OP AMP, Q 11, where one part acts as a full wave rectifier for the control circuit, formed by the other part.

Q 9 has an AC-feed-back path set for minimum distortion by the trimmer, P 4, and the limiter time constants are determined mainly by R 17 / C 9 and R 10 / C 9, of which R 17 sets the attack time and R 10 the decay.

Q 1 is a constant current source producing the reference for the control circuit but as the amplifier is powered only if one of the TALK-switches is activated, a 'wake-up' circuit is needed to ensure, that gain is present immediately, despite the suddenly applied  $V_B$  will cause the regulator to reduce the momentary gain to a minimum.

Q 4 takes care of this action as it, via R 12, discharges C 9 over a period, determined by the time constant of R 11 / C 2, but as soon as C 2 has charged, Q 4 will stop conducting and the decay time is controlled by R 10 only.

With about 35 db of gain in the control amplifier as related to the regulating characteristics of Q 9, this speech-controlled talk-back microphone amplifier will generate a nominal output within a very wide range of input signal, and hence the use is quite uncritical.

## 06.04 - LV 28

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The instrument unit, LV 28, is shown on drawing # 86110861, and the main content are two PPM-instruments and a phase correlation meter, but also a rotary switch for the limiter control and a toggle switch to select the 1 kHz or 10 kHz output from the reference oscillator are situated here.

At the very left side of the digramme stereo input signals from left and right monitor channels, MON L and MON R, are buffered by the dual FET OP AMP, Q 25, before they are applied to both PPM- and phase meter circuits.

Two separate peak level detectors for the left and the right PPM-channel comprise the precision rectifiers with normalized time constants for the attack- and fall-back-times of the instruments, and both output signals are constantly present on resistors R 57 and R 58.

The buffered L and R-input signals are also connected to the comparator, Q 22, and Q 21, R 32 / D 3 and P 2 determine the minimum reference level, -36 dBu, at which the phase meter shall operate.



Q 23 and Q 24 form the phase detector, from which the DC-signal is taken via resistor R 70 and on to the LED-driver, Q 18.

The PPM-instrument has two rows of each 18 LEDs, covering the range from +6 dB to -45 dB in steps of 3 dB, and with 1,55 volt as the 0 dB reference, the indicated voltage range is from 3,1 V to about 8 mV.

Only one LED at a time - in each row - will be used to indicate the peak signal value (flying spot), and for this reason two fixed reference marks are provided on the scale with constantly lit LEDs - namely the lower end next to the two -45 dB LEDs and the -6 dB value, corresponding to 0,775 volt.

Each row has a green LED, D 16 and D 36 - like the instrument LEDs - in the bottom of the scale, and in the center of the scale a small, yellow LED, D 65, is placed between the two -6 dB instrument LEDs, the idea being to maintain a visible reference - even if the unit is operated in a dark theatre with no signal present.

Red LEDs are used for 0, +3 and +6 dB, and also the limiter LEDs, D 35 and D 55, are red, while the phase instrument has red LEDs for negative values, green LEDs for positive and a yellow LED in the middle.

In order to keep the power consumption at a minimum, the entire LED-indicator system is a single serial chain, multiplexed at a frequency of about 100 Hz, and the available, total current to the LED-complex may also be varied manually to adjust the intensity in accordance with ambient light.

$I_{LED}$  is derived from  $+V_B$  in the supply unit, where one or two serial LEDs are the first ones in the chain, and regulated by P 1, it goes on to the common emitters of Q 3 to 9 in the phase meter circuit, where the signal to pin 5 of the LED-driver determines which of the BC 327s shall be conducting in order to light the LED in series with it.

As the total, switched LED-current passes the diodes of the phase meter, a filter condenser, C 1, is incorporated to eliminate flicker.

Via A-A the current is distributed in the PPM-circuitry, where one or another serial path way will be opened by the multiplexed switches, Q 11 to Q 15, and without looking at the timing, the following possibilities are open -:

- 1) Via the common anodes of D 16, D 36 and D 65/66 in parallel, the latter to reduce the rather sharp light of reference LED, D 65, through the switches Q 15 or Q 14 or Q 13 respectively, in series with -:
- 2) Q 11 or Q 12, connected to the common anodes of PPM-LEDs for the right and left channel respectively before ending up in the last part of the serial chain, the LED-driver, that allows a single diode of the PPM to be lit, when a signal is present.
- 3) In case no input signal is present, diodes D 63 and D64 come into the picture, as they substitute the way through the PPM-LED and the driver in order to keep the serial path closed and thus allow the prior part of the chain to remain active.



A 500 Hz oscillator followed by two counter circuits form the multiplexer unit, which is there to control the switches of the serial LED-chain and also to select the correct PPM-input signal accordingly.

Two sections of Q 32 form a 500 Hz timing oscillator of which both output signals are used - one for a decimal counter, Q 31, and the other for the flip-flops, Q 33, used as a 4 : 1 counter.

As Q 31 uses the 1-, 3- and 5-state to switch Q 15, 14 and 13, each will conduct shortly about 100 times every second, and to maintain the impression of a constant light, filtering is established by C 18 and C 3.

From Q 33 and the last two sections of Q 32, a 125 Hz timing signal controls switches Q 11 and Q 12 and also - via Q 16 and Q 17 - selects the correct signal from the peak level detectors accordingly.

Both signals are present simultaneously on resistors R 57 and R 58, from where one or the other is shortened to ground by Q 16 and Q 17, and via R 61 or R 62 the non-shortened signal is applied to a LED-driver assembly, consisting of Q 19, Q 20 and Q 30.

The two LED-drivers are identical units with 3 dB stepped log comparators and each having 10 outputs, so to handle a total of 18 LEDs the units are cascaded at a 'distance' of 30 dB, accomplished by Q 30.

The 'high-end' driver, Q 19, takes care of the LEDs from -15 dB and up on basis of the signal from R 61/62, while the 'low-end' driver, Q 20, has a -45 dB LED at the lowest and having the same reference this means, that Q 20 needs a boost of 30 dB to cope.

For the phase meter a similar LED-driver, Q 18, is used, but here the internal comparators operate with linear steps, and the variable DC-signal from R 70 - centered around  $1/2 \times -V_B$  - will light one diode at a time in accordance with the fixed time constant of R 38/39 - C 5.

## 06.05 - UK 17

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One output amplifier and limiter unit, UK 17, is used in each of the output channels, and the schematic is shown on drawing # 86110561.

Two low noise OP AMPs, Q 1 and Q 2, form the amplifier / limiter circuit, and the first unit also acts as the summing amplifier, in which the input signals from four channels and an external signal from the interface are mixed together on pins 2 and 3.

If the limiter function is not used, the mixed output signal is taken directly from Q 1 and sent to the terminal panel via selector switch, O 1, situated in the instrument unit.

It may be noticed, that both amplifiers are DC-coupled, and this is the reason for the presence of P 1 and P 2, used to eliminate the DC-offset.



The output from Q 1 is also used as an input to Q 2 via the regulating element formed by R 6 and Q 3, for which P 3 sets the correct amount of AC-feed-back in order to minimize distortion.

Limiter output is available on pin 4, from where it is routed to the terminal panel like the direct output, but it is also connected to the precision rectifier, Q 6, the output of which is sent to the amplifier, Q 7.

Q 7 is a dual FET-OP-AMP, of which one section acts as a comparator, that determines the limiter's operating level and the other section is used to detect, if the operating level has been reached or passed, an information needed to control the decay of the limiter.

The reference circuit, controlled by the constant current source, Q 5, is providing the operating reference levels for Q 7 and Q 3, the latter set by P 5 to obtain a limiter gain of 0 dB - within the specified tolerances of J 230.

From Q 7, pin 1, the amplified control signal passes the timing circuit on its way to the output buffer, from where a part of it, set by P 4, is returned to the regulator, Q 3, in order to obtain a ratio of 3 V per 10 dB attenuation.

The output on pin 13, LIM CTR, is constantly connected to the same pin of the other limiter to obtain tracking between the channels, and the harder working limiter will thus control the other in order to avoid problems of sound image location.

The limiter's attack time is determined by R 28 and C 9 only, but the decay time is controlled by R 29/30, Q 8 and the detector, Q 7, in order to eliminate 'pumping' and other unwanted behavior.

A negative output occurs on Q 7, pin 7, when the limiter is working, and this will prevent Q 8 from conducting which means, that no discharge way is available for C 9.

The charge of C 9 depends on the output from the comparator, Q 7, pin 1, and also on the duration of the limiting state, and hence the discharge time will vary accordingly from that moment the limiting state is left and Q 8 starts conducting with a delay of 50 milliseconds, caused by the time constant of R 32 / C 11.

The momentary limiter gain is always determined by the highest signal level and if, by the way, the limiter has been exposed to a signal, resulting in some degree of limitation, the limiter gain will stay at the corresponding value, until the signal level falls beyond the limiter level for more than 50 milliseconds - or a higher signal arrives.

The delay is introduced to ensure a soft shift between the limiting and non-limiting states and also to achieve, that signals below the highest one - but still within the limiter range - will be treated at the same gain setting and hence maintain the original dynamic range.



An active limiter function is indicated by a LED in the instrument unit, driven by the comparator, Q 10, where a part of the limiter control signal via voltage divider, R 35/36, activates the LED if the limitation exceeds 5 dB - a value found more interesting than information about, that the limiter is just working.

If only one limiter is active the other one has no positive DC-level on pin 7 via the limiter-LED, because the LED is turned off by the limiter selector switch in the instrument unit, and this will cause Q 14 to conduct.

When that happens the control signal on R 28 is shortened to ground and hence the inactive limiter - though it works - is prevented from having any part of control over the other one via the LIM CTR - link.

## 06.06 - TG 09

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The generator module is shown on drawing # 86100461 and comprises a sine reference oscillator of 1 or 10 kHz, a cue-tone generator with a triangular waveform and - for practical reasons - the summing amplifier for the PFL-function.

The latter mixes the PFL-signals from four input channels and an external input from the interface function, if this is used, and the mixed output signal is sent to the monitor unit's pin 14.

A 15 Hz cue signal is produced by the cue-tone generator as long as the spring loaded toggle switch in the monitor unit is activated.

A positive control signal applied to pin 3, CUE ON, starts the generator and closes both switches, Q 10 and Q 11, when Q 8 and Q 9 starts to conduct, and the output signal is sent to both summing busses, L and R, at a level of about 3,5 V<sub>pp</sub> - set by P 3.

The sine reference generator consists of a Wien-bridge circuit with a separate gain control regulator, all built around a dual FET-OP-AMP, Q 21.

Start of the oscillator takes place from each of the four input channels, if one of the rotary input mode switches is set to OSC, by which a positive control signal is sent to pin 7, OSC ON, making Q 3, 4 and 5 conduct and hence apply +/-V<sub>B</sub> to the circuit.

The oscillator is an ordinary Wien bridge type, remotely tuned to 1 or 10 kHz by the switchable RC-network via FET-switches Q 6 and Q 7.

With no control signal applied on pin 1, Q 6 will conduct and Q 7 be open and vice versa, when a positive control signal arrives from the frequency selector switch in the instrument unit, where 1 or 10 kHz may be set.

When Q 6 conducts and shortens C 5 etc, the oscillator is tuned to 1 kHz but with Q 7 in conduction and Q 6 open the frequency will be 10 kHz.

Q 1 is the amplitude regulator, controlled by the reference- and regula-



tor circuit formed by the other part of Q 21, and the output level is set to 6 dBu on pin 4 by P 2.

To obtain a quick start and a steady level, D 1 and D 2 immediately limit the maximum output amplitude at a level just above the normal, until the regulating circuit has settled and gained the normal balance to take over control via Q 1, to which the AC-feed-back is set for minimum distortion by P 1.

## 06.07 - SF 17

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The regulated 48 volt phantom power supply is seen on drawing # 86110361, consisting of an/off- and input filtering circuit, the regulated fly-back converter, Q 5, and the output regulator and filter, Q 1, Q 2 and Q 3.

When one or more input channels are set to MIC/48 V, a control signal is present on pin 13, by which  $\pm V_B$  is applied to the power supply via the switches Q 6 and Q 7.

Output from the fly-back converter is set to 53 V by P 2 and adjusted to a final, constant output of 48 V in the regulator circuit by P 1 - both voltages referred to 0 V.

To keep any residue of the 100 kHz operating frequency away from the surroundings, chokes L 1, L 2 and L 3 are installed in the supply lines and in the output.

## 06.08 - ID 04 B

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The battery power supply unit has a  $V_B$ -status monitor built into it, the circuitry of which is shown on drawing # 86110863.

The operating voltage may come from the batteries or from an external power source via the interface connector in the terminal panel, and selected by the power switch, O 1,  $\pm$  pass the fuses I 1 and I 2 before entering the equipment as  $\pm V_B$ .

All battery connectors are coded in order to prevent any damage, if batteries are incorrectly loaded, but in case an external power source has a wrong polarity, diodes D 1 and D 2 will blow the fuses immediately.

The  $V_B$ -status monitor informs about the present battery status in three ways as the LED, D 10, will be constantly on, flash or go out, when the available voltage changes from nominal over medium to low.

The nominal voltage is specified within the range of  $\pm 12$  V to  $\pm 8,5$  V, medium voltage from  $\pm 8,5$  V to  $\pm 8,0$  volt and low voltage means values less than  $\pm 8$  V.

The reference circuit, Q 1, Q 2, Q 3 etc, controls the LED-driver, Q 8,



in accordance with the present  $V_B$ -value, and within the nominal range Q5 and Q 8 will conduct and keep Q 2 off, which makes D 10 light constantly.

When the  $V_B$ -value gets into the medium range, Q 5 will turn off and make Q 8 act as a multivibrator, by which Q 2 is switched on and off at a frequency set by the time constant of R 20, R 22 / C 3 - and hence D 10 will flash accordingly.

In the low-status, the voltage on Q 8, pin 3, goes down and prevents Q 8 from oscillating, and as the output voltage will no longer keep Q 2 off, LED-indicator, D 10, is shortened and no light is present.

D 11 is the 48 volt indicator, normally shortened by Q 6 until a 48 volt signal arrives on pin 1, turns Q 6 off and makes D 11 active, and it may be mentioned, that D 10 and D 11 are the first two LEDs in the total, serial chain, of which the last link is the PPM's LED-driver.

## 06.09 - CP 06

Finally we come to the terminal panel, drawing # 88032762, where all connections to and from the unit are established via eight 3-pole XLRs, two 1/4'' jack sockets and a 37-pole multi armature, and in order to stay informed of the internal relations between the various parts of the unit, it may be advisable to have the overall drawing, # 88032761, at hand.

Four input signals, LI or MIC, arrive at F 1 to F 4, and via RF filters - 3 dB down at 1 Mhz, when microphones are used - the symmetrical signals are sent to the four input channels.

F 5 and F 6 are two external inputs, normally via tape and pre-set by P 1 and P 2, before they are sent to the monitor system.

S 1 and S 2 are male XLRs for the left- and right channel outputs and F 8 and F 9 are 1/4'' jack sockets in parallel, from both of which the stereo monitor signals are available.

F 7 is a 37-pole DELTA-armature, named INTERFACE, and several connections and functions are possible with an appropriate wiring.

The main purpose is to establish an easy link with another identical unit in order to raise the number of available input channels to eight without loosing any facilities such as input channels, PFL-control etc, but also the direct outputs from each input channel, the connections for the talk-back-/conference system and an external power supply are terminated here.

S 5 to S 8 each carry the input- and output signals from input channels 4 to 1 respectively, but in addition S 5 is wired with  $+/-V_B$  for the interface amplifiers, Q 1 and Q 2, S6 and S7 each has a connection for the external inputs to the left and the right summing busses, and finally S 8 passes the PFL-signals in and out as well as the PFL-ON-control function.

Three interface buffer-/inverters - dual device Q 1 and Q 2 - are incor-



porated in order to match the external stereo- and PFL-signals to the polarity and level of the busses, but Q 1 and Q 2 are not active until the indicated bridges are established in the counterpart of F 7.

$\pm V_B$  is applied only, if pin 27 is connected to pin 8 and pin 28 to pin 9, and also the buffer outputs must be linked on to the correct summing bus via pins 1/3, 10/34 and 29/37 - for the reason of minimized power and summing noise, if the interface connection is not used.

An external 24 V centertapped DC-voltage may be applied on pins 11/30/31, named 0 V / +12 V / -12 V respectively - with the correct polarity carefully observed.

The L/R-stereo output from the PORTABLE is available on pins 16/18, and the individual outputs from channel 1 to 4 are found on pins 23, 24, 25 and 26 - in that order and referred to 0 V.

Pins 12/13/14/15 and 32/33 are occupied by the conference-/talk-back-facilities, intended as a duplex conference channel with a distant director or stage technician or when just some editing comments are to be addressed via a PA-unit via speakers or head phones.

The outgoing conference is controlled by the lighted push button in the monitor unit, by which a microphone signal and a  $+V_B$  signal are sent out on pins 32 and 12 respectively.

The  $+V_B$  signal may be used solely for controlling a start/stop function, a red light or as a muting signal for a distant loudspeaker system etc.

Ingoing conference consists of a speech signal and two DC-signals, used to control the monitor function and/or the CALL-lamp in push button, O 2.

The external speech signal is connected to pin 13 and referred to 0 V on pin 11, but the DC-control signals are floating, and with an indifferent polarity they have one common connection on pin 15, CONF COMM, where one side of the lamp in the CONF/CALL-push button, O 2, is tied to one side of the opto-coupler, Q 27, and a DC-level between 8 and 12 volt will activate the CALL-lamp via pin 33 and the opto-coupler via pin 14, both referred to pin 15.

Shortly recapitulated the speech signal from the distant partner will override the right channel monitor signal, when a control signal for the opto-coupler is received, and talk-back is possible by pushing the CONF/ CALL-button, O 2.

If the duplex conference possibility is frequently needed it is advisable to obtain a small conference unit, comprising a head phone- and a microphone-amplifier like those built into the monitor - with own batteries or supplied from the PORTABLE's interface connector, pins 27/28.

An external connection box may also be practical if some of the interface facilities are regularly used -: external power supply, channel outputs, access to the summing busses, a talk-back signal for a loudspeaker system or just a control signal for a recorder or a red light, but also if a set



of output transformers are needed they might be installed in such a box.

In order to obtain more than four input channels two PORTABLEs may be interconnected by a multicore cable from one unit's interface connector to the other's, and unit # 1 should be connected to unit # 2 - considered as the main unit - the following way -:

The stereo output from unit # 1 is taken from pins 16/18 and sent to pins 35/19 in unit # 2 via screened wires (L>L and R>R), and the PFL-signal is treated the same way, taken from pin 21 in unit # 1 and sent to pin # 20 in unit # 2.

The PFL-control function may be used separately in each unit or linked in order to shift both monitor systems to PFL, when one or more PFL-switches are activated in either unit, the latter being more practical because all PFL-signals from both units will occur in the same monitor system, when a wire connects pin 22 in unit # 1 with pin 22 in unit # 2.

The cable part, connecting to unit # 2, of which the interface inverters are engaged, must have links on pins 8/27, 9/28, 29/37, 10/34 and 1/3 in order to supply +/-V<sub>B</sub> and the three outputs to the summing busses.

Also ground, or 0 V, should be interconnected via pins 2, 7 or 36 in both units.

If two interconnected units are in operation, the mixed inputs from eight channels are at disposal as a stereo output from unit # 2, and the monitor here will act as a monitor for both units, but still the stereo output from unit # 1 is available and may be considered as a separate group of four, of which the local monitoring facilities are fully maintained.

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Addendum -: Using M/S-microphones directly with the np PORTABLE.

One or two M/S-microphones may be used very easily, and the M/S >> L/R - conversion is established by a few bridges in the interface connection.

The M/S >> L/R - relation is expressed by -:  $L = M+S$  and  $R = M+(-S)$  which means, that the S-signals, directly and inversed, added to the M-signal form the left and right stereo signals respectively, and by means of the interface buffer/inverters this phase-dependent addition takes place on the L- and R-summing busses.

Each M/S-stereo microphone occupies two input channels and hence two microphones this type may be operated simultaneously by the np PORTABLE.

For practical reasons input channels 1 and 3 should be reserved for the M-signals and their neighbours - input channels 2 and 4 - for the S-signals respectively.

The M-signal is treated like a normal mono signal, which it is in fact, and with the pan-pot of input channel 1 placed in the center position,



this signal is distributed equally to the left and the right summing buses.

To make the S-signal in input channel 2 act in accordance with the above mentioned equation the pan-pot of channel 2 must be turned all the way to the LEFT, which adds the in-phase S-signal to the M-signal on the left summing bus and thus fulfills the first requirement -:  $L = M+S$ .

Now we just need to obtain the inverted S-signal and add it to the M-signal, which is present already on the right summing bus, and this is done the following way -:

The S-signal is taken out from input channel 2 via jumper switch, S 3, in position AFTER FADER, and this signal is present on pin 24 of the interface connector.

A jumper is connected from pin 24 to pin 19, which is the input terminal of the buffer/inverter, Q 1, for the right summing bus.

Access to the bus is made by another jumper between pins 29 and 37, and in order to power the buffers,  $\pm V_B$  must be applied by connecting pin 8 to pin 27 and pin 9 to pin 28 - please refer to drawing # 88032762.

The result is, that the inverted S-signal is added to the M-signal on the right summing bus and this complies with the second part of the equation, where the right signal should be -:  $R = M+(-S)$ .

All of the buffer/inverters have a 0-dB gain, and as the AFTER FADER output level is identical to the level of the L- and R-positions of the pan pot, the two parts of the S-signal are of equal amplitude accordingly.

It was mentioned earlier, that the pan pot has an attenuation of 3 dB in the center position in order to maintain a nearly constant sound pressure during panning.

As the M-signals for the left and right summing busses are taken from the pan pot center position in channel 1, the S-signals are at a 3-dB higher level, if channels 1 and 2 operate at equal gain settings - the effect of which will be discussed later.

If channels 3 and 4 are also used for an M/S-microphone, there is a minor difference from the above mentioned method, because each summing bus has one external input only and hence the addition of the M-signal and the inverted S-signal must take place on the left bus.

The S-signal is taken out from input channel 4 via jumper switch, S 3, in position AFTER FADER, and this signal is present on pin 26 of the interface connector.

A jumper is connected from pin 26 to pin 35, which is the input terminal of the buffer/inverter, Q 1, for the left summing bus, and another jumper between pins 10 and 34 connects the buffer/inverter output to the bus.

Power has already been applied when channels 1 and 2 were discussed, and



now we will look into the proper settings of channels 3 and 4 in order to obtain the correct M/S >> L/R conversion.

Channel 3 is the M-channel and the pan pot should be placed in the center position in order to apply the M-signal equally to both busses - L and R.

Channel 4 is the S-channel, panned all way to the RIGHT, which adds the S-signal and the M-signal on the right bus, but in phase as  $R = M+S$ , contradicting the equation for the R-relation -:  $R = M+(-S)$ .

The inversed S-signal arrives on the left summing bus and added to the M-signal this result is also wrong -:  $L = M+(-S)$  - instead of -:  $L = M+S$ .

In both cases the S-signal arrives in an opposite phase, so in order to cope the PHASE REVERSAL SWITCH in channel 4 must be used.

When the internal jumpers are placed in the counterpart of the interface connector as described above, the following few points are important -:

- 1) M/S-microphone # 1 on input channels 1 (M) and 2 (S).
- 2) M/S-microphone # 2 on input channels 3 (M) and 4 (S).
- 3) M-channel 1 and M-channel 3 are panned to CENTER.
- 4) S-channel 2 is panned all way to the LEFT.
- 5) S-channel 4 is panned all way to the RIGHT with PHASE REVERSED.

Users being familiar with M/S-stereo techniques need not read any further but for fair reasons - just a few comments -:

M/S-stereo technique is totally compatible with L/R-stereo and it is not difficult to handle with a suitable microphone and a little bit of confidence.

Two signals are needed, of course, but here left and right are created as a result of a phase-/intensity related summing of both.

The main signal, called M, is a normal mono signal from an omnidirectional microphone capsule - facing the middle of the signal source - and the assistant S-signal comes from another capsule with a figure-of-eight characteristic, orientated sideways - or  $90^\circ$  - as related to the M-system.

Both microphone systems are built closely together - normally in the same house and arranged in a vertical axis, the upper system being the S.

One may imagine, that a sound from the exact middle of the source is not registered by the S-system because of the  $90^\circ$  orientation, but fully received by the M-system - as a mono signal.

It is also possible to accept, that a sound coming directly from the left or the right side of the M/S-microphone system will be registered by both units simultaneously, and now there is only one more fact to consider before the action of the M/S-microphone is clarified, namely the influence of the phase- and intensity relations between the M- and the S-signals.



A positive air- or sound pressure normally accomplishes a negative pressure somewhere else in order to maintain status quo, and if we define the positive pressure to move the membrane of the microphone inwards and create a positive going signal amplitude on the signal core it is evidently so, that a negative going pressure will have the opposite effect.

Now it may be possible to accept, that a positive sound pressure from the left side of the S-capsule will do the same thing to the membrane as a negative one from the right side which means, that the output signal from the S-capsule goes positive, if a positive sound pressure occurs from the left but goes negative, if it happens from the right.

The omnidirectional capsule will produce a positive going output whenever a positive sound pressure is present anywhere, but the figure-of-eight unit will detect if this positive pressure comes from the left or from the right side - or seems not to occur at all, if it originates in the middle, from which direction the figure-of-eight unit is almost 'deaf'.

It is obvious, however, that when a sound signal is moved off the centerline, the S-system will produce a signal, that grows larger the more the sound comes from the direction, in which the S-system is most sensitive.

The conclusion is, that detection of left or right is purely determined by phase relation of the M- and S-signals, but the impression of how far out left or right depends on the momentary amplitude relation and hence the M/S-stereo method is referred to as phase/intensity stereo.

The M-signal contains the full tonal information and when the phase- and amplitude related S-signal is added to the M-signal, a highly defined directional stereo image may be obtained.

The pan pot function is fully active in the M-channel, and the directional impression may be varied by this control as usual, but the S-channel acts a little different.

Here the pan pot is fixed in the left or the right side - due to the conversion M/S >> L/R as explained above - and the fader determines how much S-signal and hence the stereo basis or width - all the way from pure mono and into a stereo image with a larger basis, than the original.

Filter settings of the M- and S-channels should be done with great care as the M/S >> L/R conversion is only true in fact, when the original phase- and amplitude relations are fully maintained.

This means, that the M- and S-channels should be lined up identically and with moderate equalizer settings only, but everything depends on the situation and it should be mentioned, that very special effects may be obtained by different equalizing of the S-channel.

Finally it should be noted, that if two np PORTABLES are linked together it is still possible to have the special jumpers for the M/S-microphones installed in the first unit, because the buffer-/inverters here are not used for the interface function.

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07 -: RECOMMENDED CALIBRATION PROCEDURES  
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If any calibration or adjustment is needed, the following procedures are recommended, and in order to find the locations of trimmers or other adjustments the PCB/component lay-out drawings should be consulted.

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Calibration procedures are outlined for the following units -:  
-----

IK 34 B	-	PCB/component lay-out drawing #	88032561
MON 56	-	-----	# 86110962
LV 28	-	-----	# 86110862
UK 17	-	-----	# 86110562
TG 09	-	-----	# 86100462
SF 17	-	-----	# 86110362

-----



IK 34 B -: LINE INPUT CMRR  
=====

Pin #	Pin name	Test connection
1	Channel output	AC-voltmeter ref. GND
2	0 V	GND
3	-V <sub>B</sub>	-12 V
4	+V <sub>B</sub>	+12 V
7	CH	GND
18	INP+	Common mode generator
19	INP-	ref GND - R <sub>g</sub> = 2*25 E

Front panel settings -: SOURCE >>>>> LI  
GAIN >>>>>> max  
HP FILTER >> lin  
EQUALIZER >> OFF

Channel output jumper -: Pre fader

Generator -: 15 kHz / 6 dBu

Select C<sub>x</sub> and adjust C 2 for minimum reading on voltmeter

NB -: The shielding panel must be mounted on IK 34 B during adjustment.



**MON 56 -: TALK-BACK MIC AMP DIST ADJ**

=====

Pin #	Pin name	Test connection
2	0 V	GND
3	-V <sub>B</sub>	-12 V
4	+V <sub>B</sub>	+12 V
7	CH	GND
20	TALK-BACK	Audio analyzer ref GND

- 
- 1) Disconnect the microphone capsule from the PCB of MON 56.
  - 2) Connect a generator between microphone input (C 103) and GND.
  - 3) Adjust the generator to 1 kHz / -30 dBu.
  - 4) Press the CONF/CALL-pushbutton and -
  - 5) Adjust P 104 to minimum distortion on audio analyzer.
  - 6) Reconnect the microphone capsule to the PCB.
-



## LV 28 --: PHASE METER- & PPM-ADJUSTMENTS

Phase meter --: Minimum detector level adjustment

PPM - instrument --: Rectifier off-set adjustment  
Rectifier gain adjustment  
LED-reference adjustment

Pin #	Pin name	Test connection
7	I <sub>LED</sub>	+12 V
8 + 9	L + R-METER INPUTS	Generator ref GND
10	CH	GND
14	+V <sub>B</sub>	+12 V
15	0 V	GND
16	-V <sub>B</sub>	-12 V

Generator setting --: 1 kHz / -36 dBu

1) Phase meter --: Set P 2 to the point, where the phase meter reading goes from '+' to '0' - that is all.

2) PPM-instrument --: Adjust P 5 and P 6 until the PPM-readings just change from -45 dB to -42 dB

Generator setting --: 1 kHz / 6 dBu

3) PPM-instrument --: Adjust P 3 and P 4 until the PPM-readings just change from -3 dB to 0 dB

Generator setting --: 1 kHz / -12 dBu

4) PPM-instrument --: Adjust P 7 until the PPM-readings just change from -21 dB to -18 dB

5) Repeat step 2) at the generator setting --: 1 kHz / -36 dBu

6) Repeat step 3) at the generator setting --: 1 kHz / 6 dBu

7) Repeat step 4) at the generator setting --: 1 kHz / -12 dBu

8) For the sake of good order steps 2, 3 and 4 may be run through for a last time just for a final check, but normally steps 1 - 7 will do.



## UK 17 -: OUTPUT AMP & LIMITER ADJUSTMENTS

Summing amplifier -: off-set adjustment

Limiter amplifier -: off-set adjustment  
gain adjustment  
distortion adjustment  
control voltage adjustment

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Pin #	Pin name	Test connection
2	IF- INP	Generator ref GND
4	LIM-OUT	
6	SUM-AMP-OUT	
7	LIM-LED	+12 V
9	-V <sub>B</sub>	-12 V
11	0 V	GND
12	+V <sub>B</sub>	+12 V
13	LIM-CTR	
14	CH	GND

---

Generator setting -: off

- 1) Adjust P 1 for minimum off-set (< 0,1 mV) on pin 6
- 2) Adjust P 2 for minimum off-set (< 0,1 mV) on pin 4

Generator setting -: 1 kHz / 0 dBu

- 3) Measure the output level on pin 6 and -
  - 4) Adjust P 5 to obtain the same output on pin 4 (LIM OUT)
  - 5) Adjust P 3 for minimum distortion on pin 4 (LIM OUT)
  - 6) Repeat step # 4)
  - 7) Apply +3,00 volt ref GND to pin 13 (LIM CTR)
  - 8) Adjust P 4 until the level on pin 4 has dropped 10,0 dB
-





**SF 17 -: PHANTOM VOLTAGE ADJUSTMENTS**  
=====

Fly-back converter -: voltage adjustment

Output regulator -: output voltage adjustment

-----

Pin #	Pin name	Test connection
4	48 V	
9	-V <sub>B</sub>	-12 V
11	0 V	GND
12	+V <sub>B</sub>	+12 V
13	48 V ON	+12 V
14	CH	GND

-----

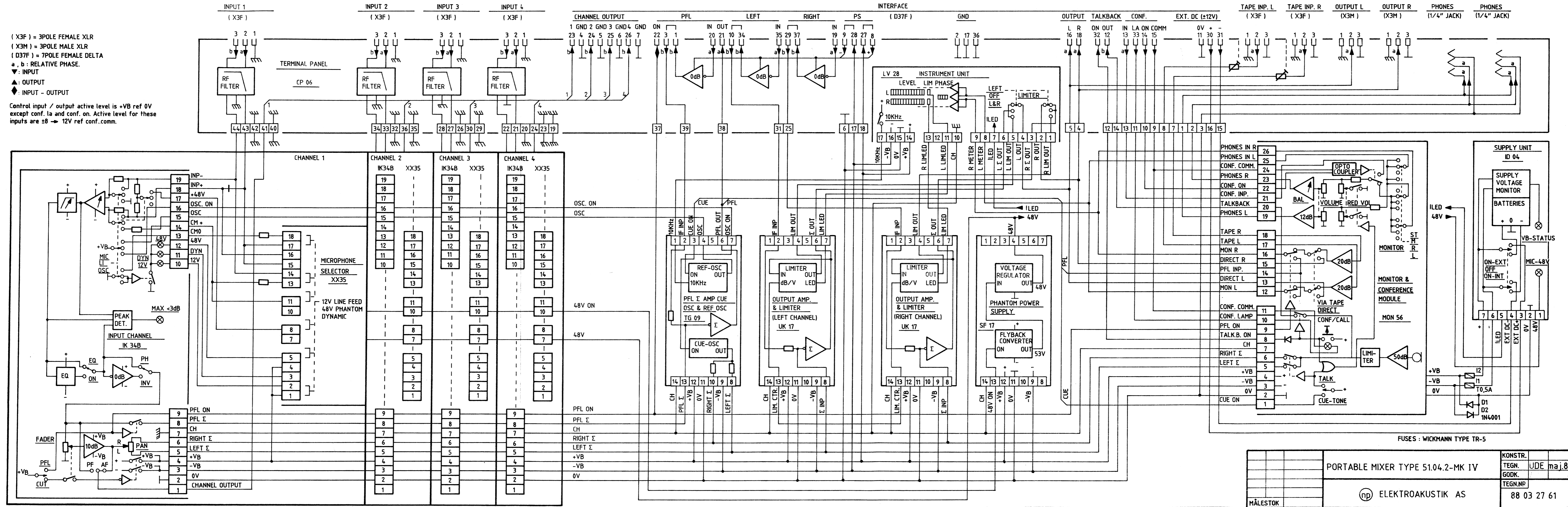
- 1) Connect DC-voltmeter between cathode of D 2 and GND.
  - 2) Adjust P 3 until the DC-voltmeter reads 53 V
  - 3) Disconnect voltmeter from D 2 and connect to pin 4 (maintain GND)
  - 4) Adjust P 2 until the DC-voltmeter reads 48 V
-



SCHEMATICS AND PCB/COMPONENT LAY-OUT

Schematics		Drawing number
01	-: 5104.2/IV - overall block schematic	88032761
02	-: IK 34 B - input channel	85030511
03.1	-: MON 56 - monitor- and conference module	86110961
03.2	-: MON 56 - talk-back microphone amplifier	83083111
04	-: LV 28 - instrument unit	86110861
05	-: UK 17 - output amplifier and limiter	86110561
06	-: TG 09 - reference- and cuetone generator	86100461
07	-: SF 17 - phantom power supply	86110361
08	-: ID 04 B - battery power supply and -monitor	86110863
09	-: CP 06 - terminal panel	88032762

PCB/component lay-out		Drawing number
11	-: HP 66 - main board	84051611
12	-: IK 34 B - input channel	88032561
13	-: MON 56 - monitor- and conference module	86110962
14	-: LV 28 - instrument unit	86110862
15	-: UK 17 - output amplifier and limiter	86110562
16	-: TG 09 - reference- and cuetone generator	86100462
17	-: SF 17 - phantom power supply	86110362
18	-: ID 04 B - battery power supply and -monitor	86110864
19	-: CP 06 - terminal panel	83100513

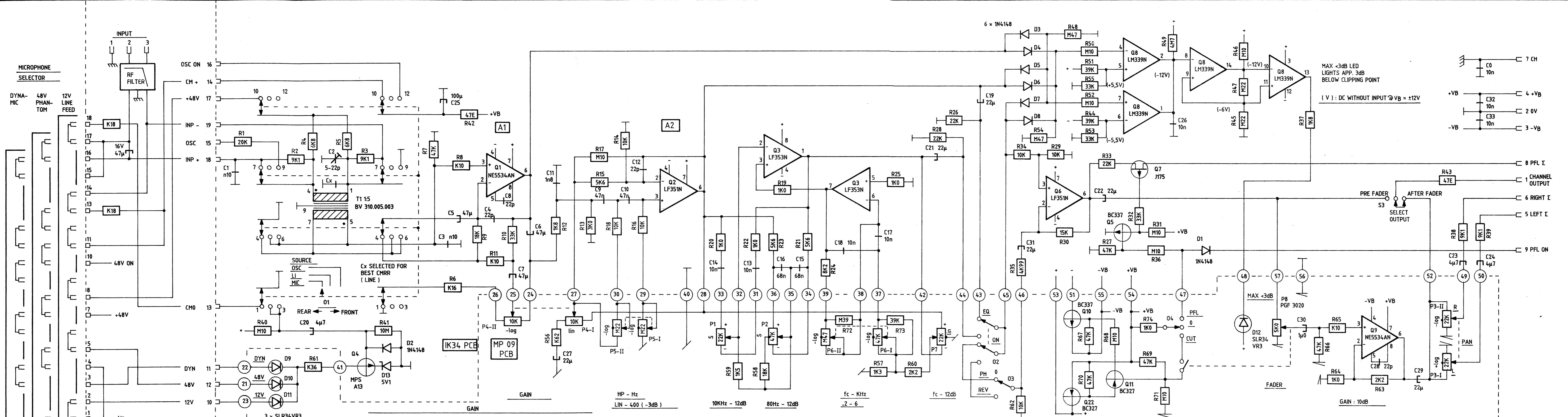


PORTABLE MIXER TYPE 51.04.2-MK IV

Ⓝ ELEKTROAKUSTIK AS

KONSTR.	
TEGN.	UDE maj.88
GODK.	
TEGN.NP	88 03 27 61

MÅLESTOK

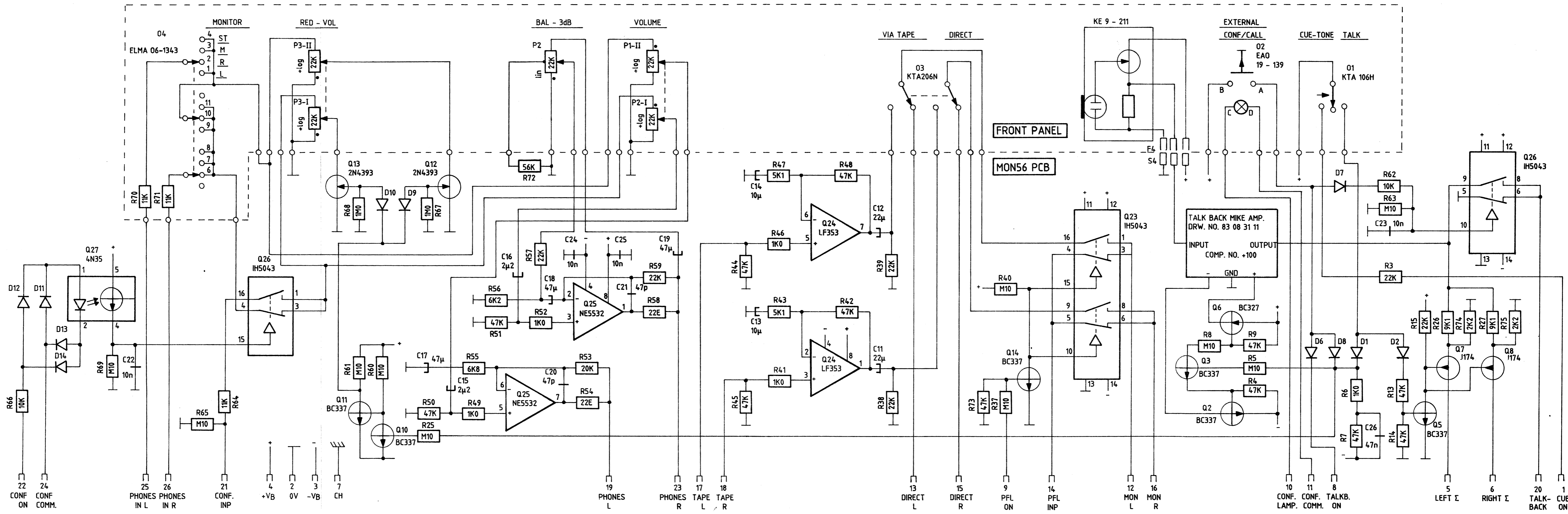


	T1	A1	A2	TOTAL
MIC min	14dB	0dB	15dB	15dB
MIC max	14dB	34dB	22dB	70dB
LI min		-15dB	0dB	-15dB
LI max		22dB	22dB	12dB
OSC min		-15dB	0dB	-15dB
OSC max		-10dB	22dB	12dB

GAIN  
 LIN - 400 (-3dB)  
 10KHz - 12dB  
 80Hz - 12dB  
 fc - KHz  
 .2 - 6  
 fc - 12dB

MAX -3dB LED  
 LIGHTS APP. 3dB  
 BELOW CLIPPING POINT  
 (V) : DC WITHOUT INPUT @ V<sub>B</sub> = ±12V

8805 PP R51, R61, C29	INPUT CHANNEL	IK 34B	KONSTR.	
8807 PP D1, D13, D7			TEGN.	UDE apr.88
PP 8837 C20, T7, Q7			GODK.	
			TEGN.NR.	
MALESTOK	ELEKTROAKUSTIK AS		85 03 05 11	

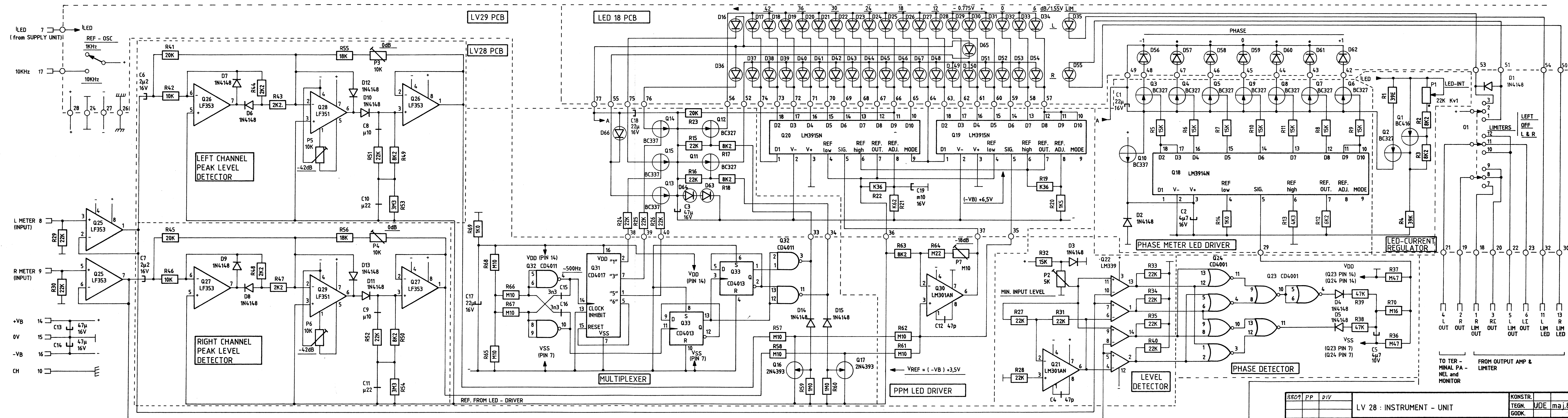


PP	B810	R72, C26, S4, F4	KONSTR.	
			TEGN.	UDE apr.88
			GODK.	
			TEGN.NR.	
MÅLESTOK	ELEKTROAKUSTIK AS			86 11 09 61

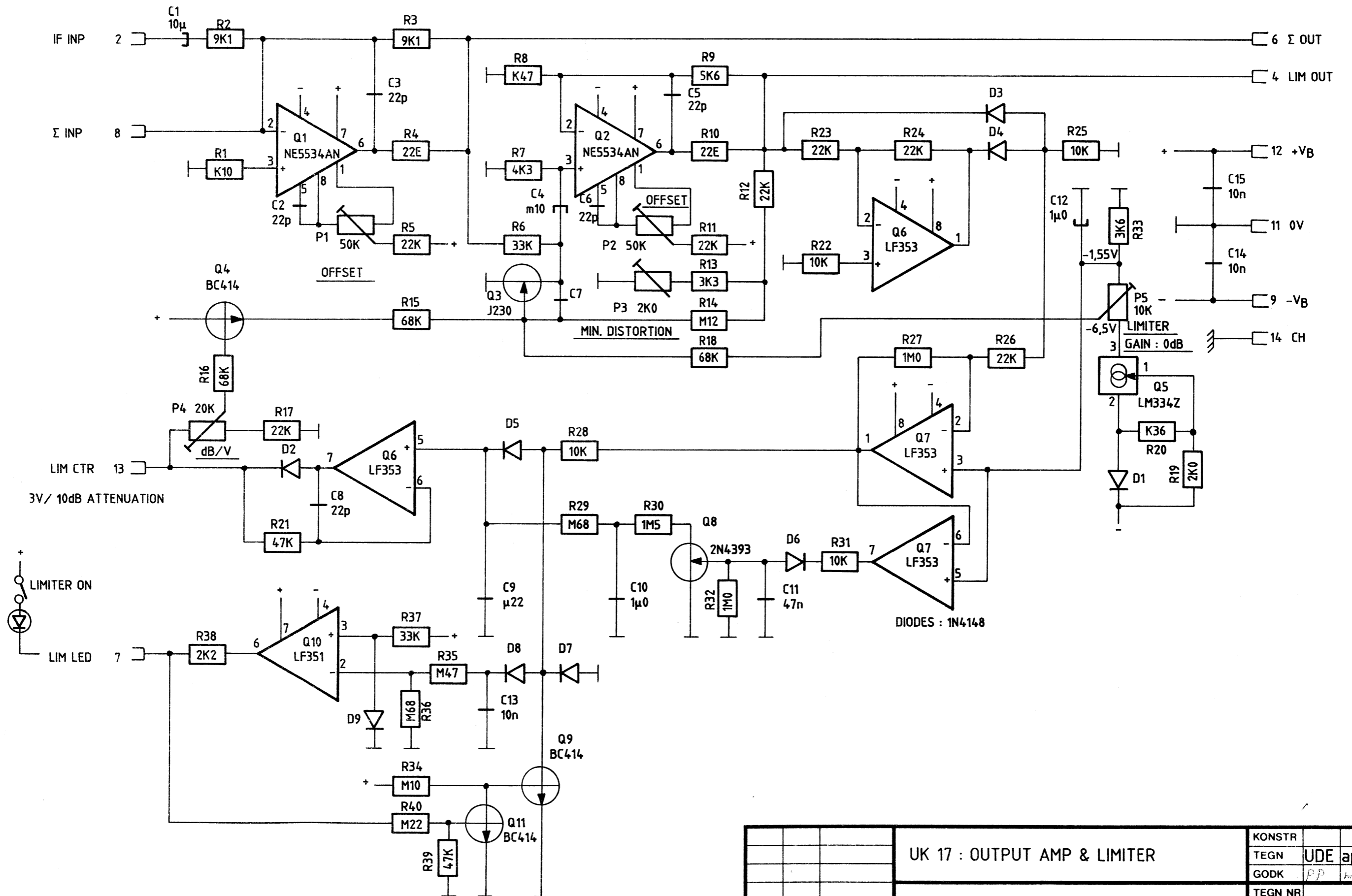
MON 56 : MONITOR & CONFERENCE MODULE

Ⓟ ELEKTROAKUSTIK AS



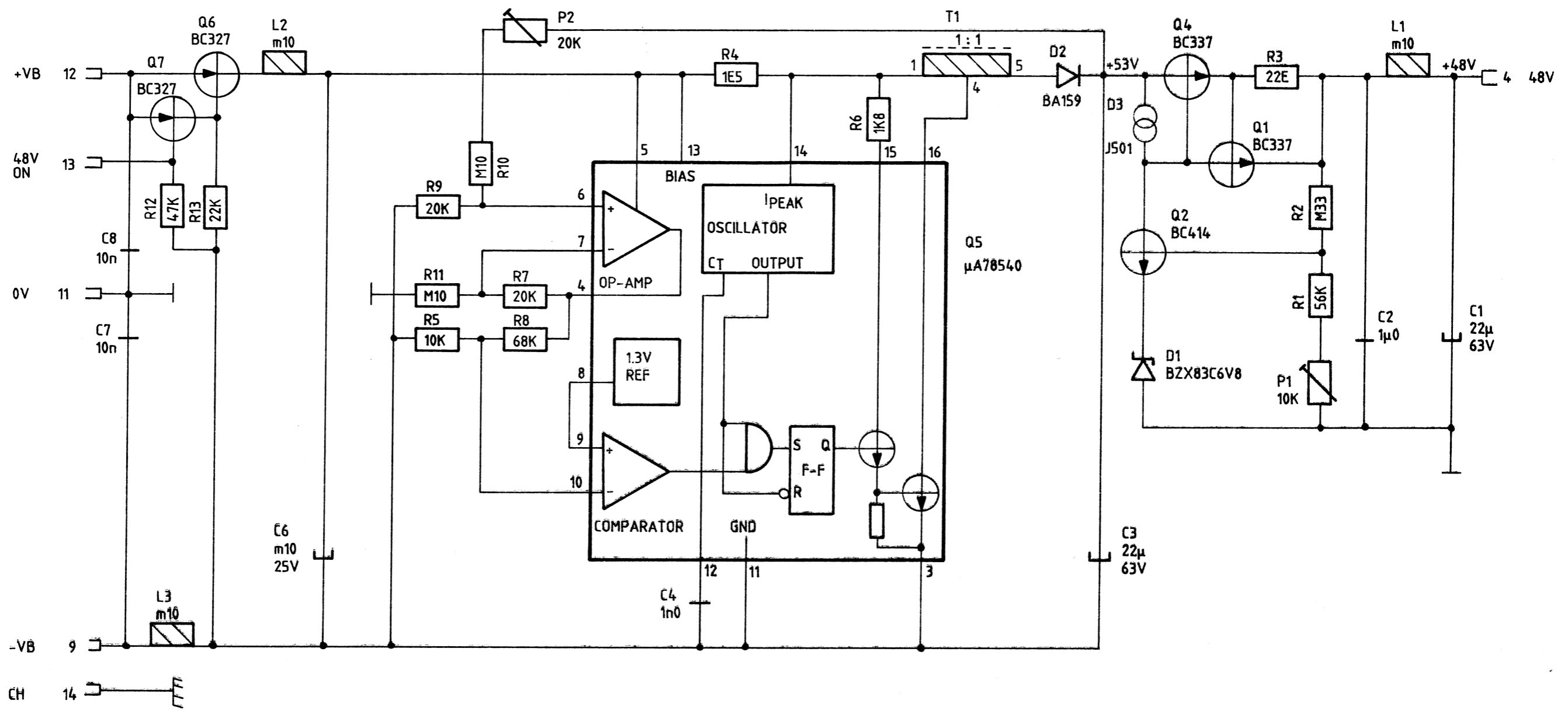


3809	PP	DIV			
			LV 28 : INSTRUMENT - UNIT		
KONSTR.		TEGN. UDE		maj.88	
GODK.		TEGN.NR.			
MÅLESTOK			ELEKTROAKUSTIK AS		
			86 11 08 61		



UK 17 : OUTPUT AMP & LIMITER		KONSTR	
		TEGN	UDE apr.88
		GODK	PP maj.88
		TEGN NR	
ELEKTROAKUSTIK AS		86 11 05 61	





ON/OFF CONTROL & INPUT FILTER

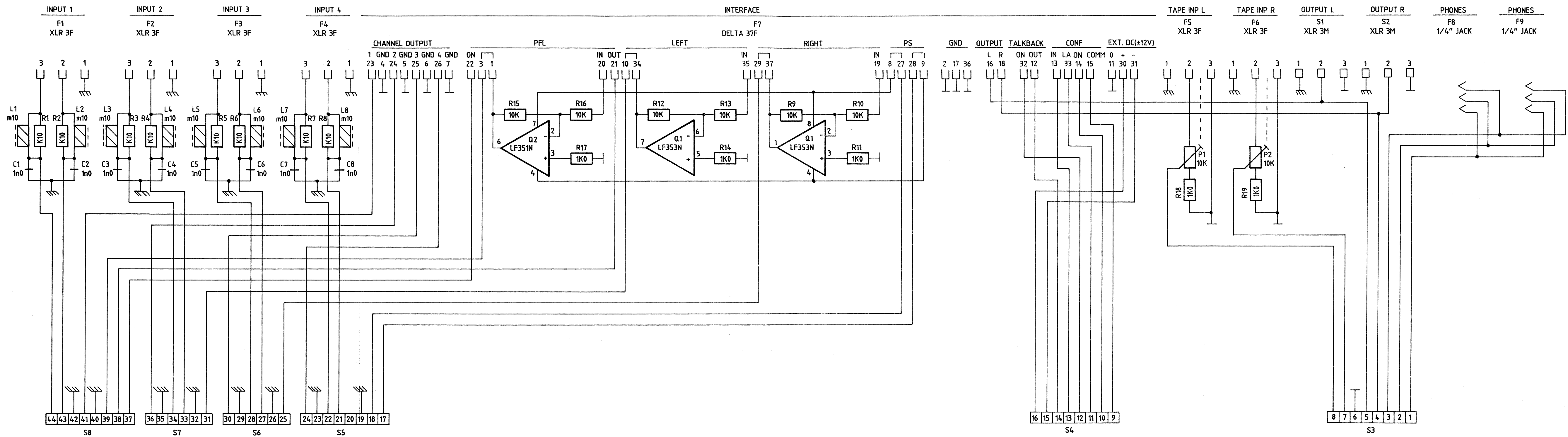
REGULATED FLY - BACK CONVERTER

OUTPUT REGULATOR

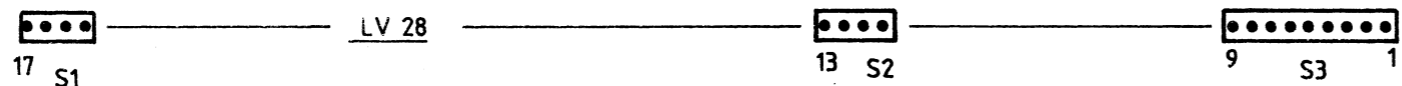
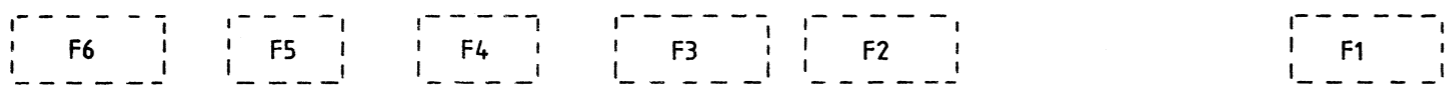
ADJ. P2 TO +53V ( REF. 0V ) ON D2 - CATHODE  
 ADJ. P1 TO +48V ( REF. 0V ) ON OUTPUT

		SF 17: PHANTOM POWER SUPPLY		KONSTR	
				TEGN	UDE maj.88
				GODK	PP maj.88
				TEGN NR	
MÅLESTOK		np ELEKTROAKUSTIK AS		86 11 03 61	

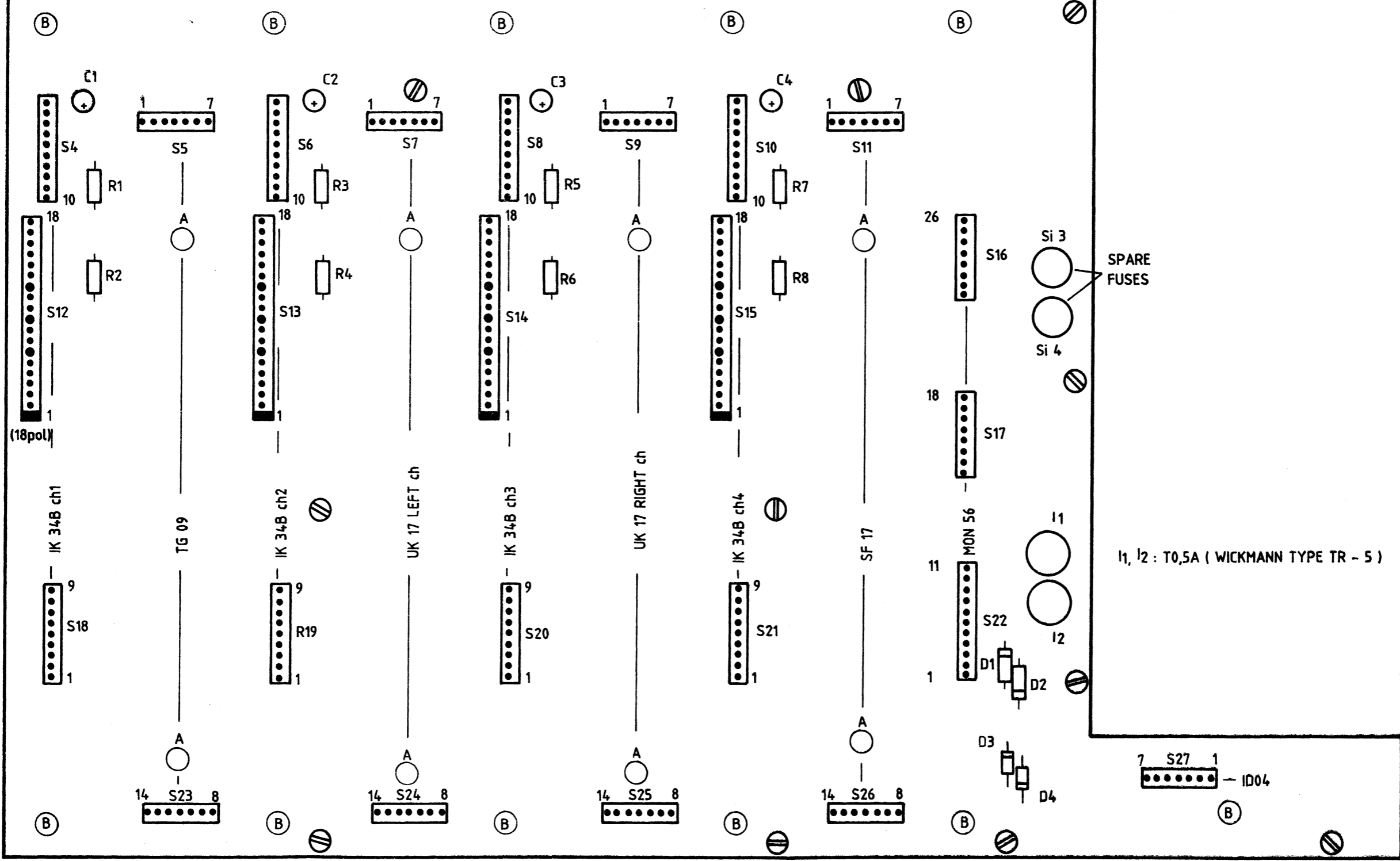




CP06 : TERMINAL PANEL		KONSTR.	
		TEGN.	UDE apr.88
		GODK.	
		TEGN.NR.	
MÅLESTOK	ELEKTROAKUSTIK AS		88 03 27 62



EXTRACTED PINS



SPARE FUSES

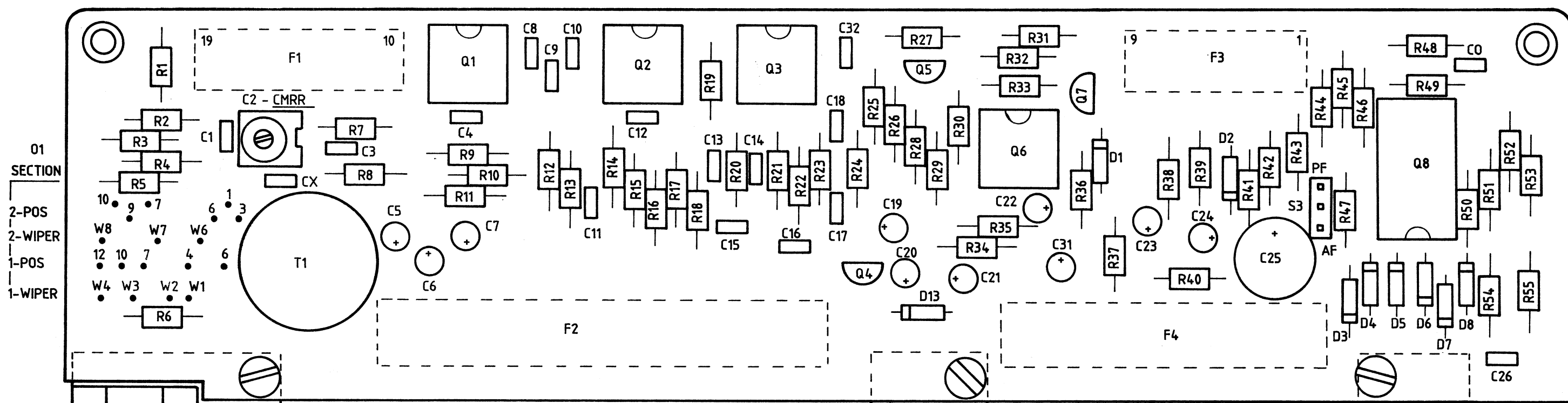
I1, I2 : T0,5A ( WICKMANN TYPE TR - 5 )

		KONSTR	
		TEGN	UDE maj.88
		GODK	
		TEGN NR	
MÅLESTOK		np ELEKTROAKUSTIK AS	
		84 05 16 11	

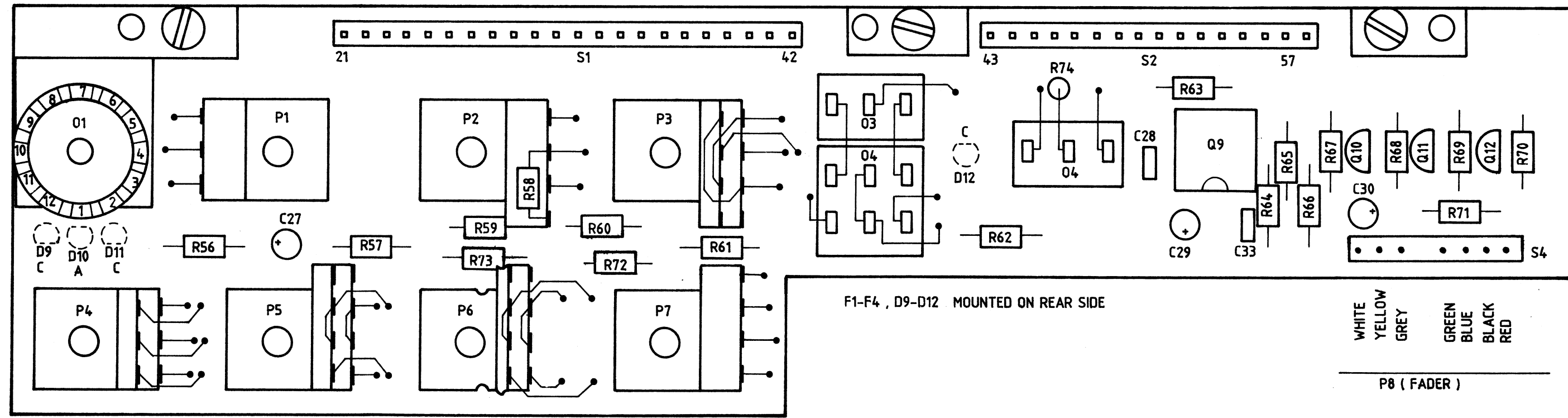
COMP. LAYOUT FOR HP66

np ELEKTROAKUSTIK AS

84 05 16 11



S3 : CHANNEL OUTPUT SELECTOR : PF = PRE FADER  
AF = AFTER FADER



F1-F4, D9-D12 MOUNTED ON REAR SIDE

WHITE  
YELLOW  
GREY  
GREEN  
BLUE  
BLACK  
RED

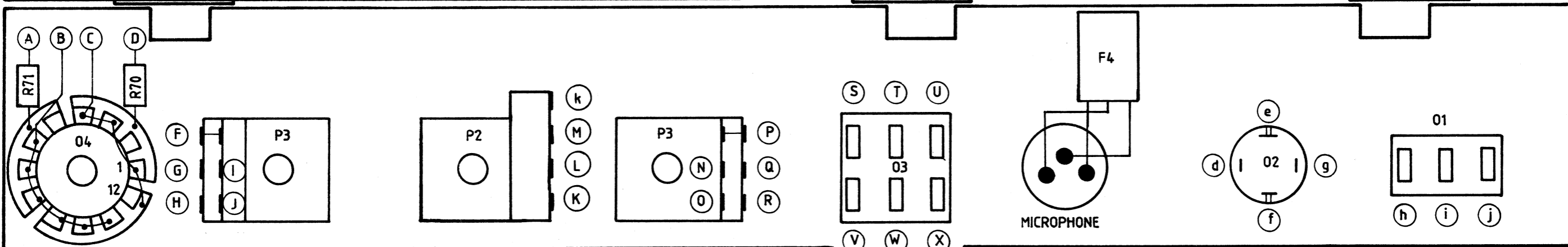
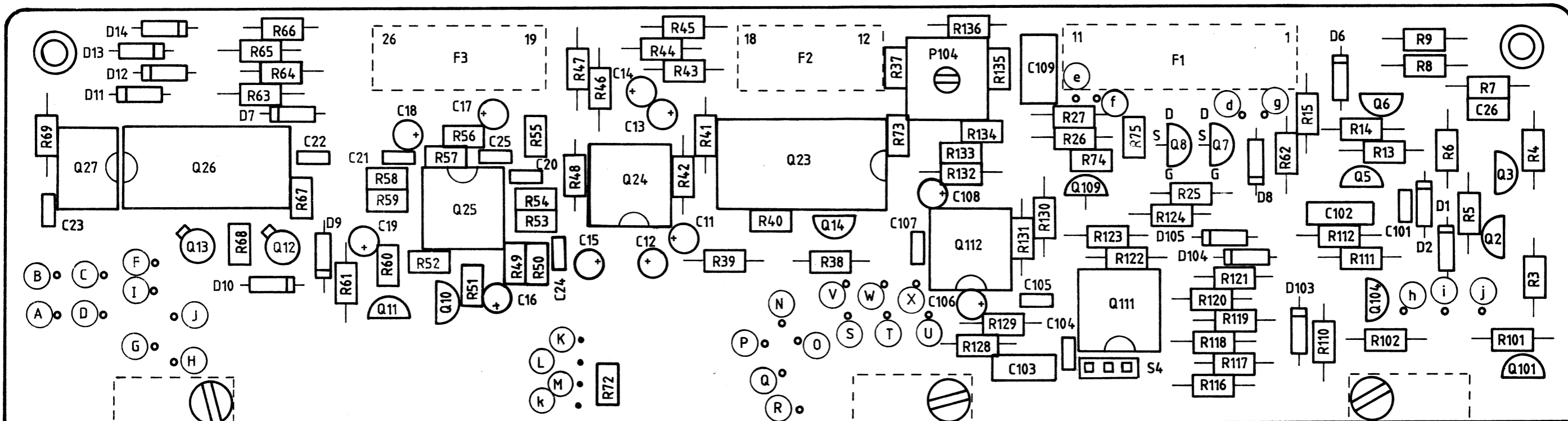
—————  
P8 ( FADER )

8807	B3	A72+73+S4
MÅLESTOK		

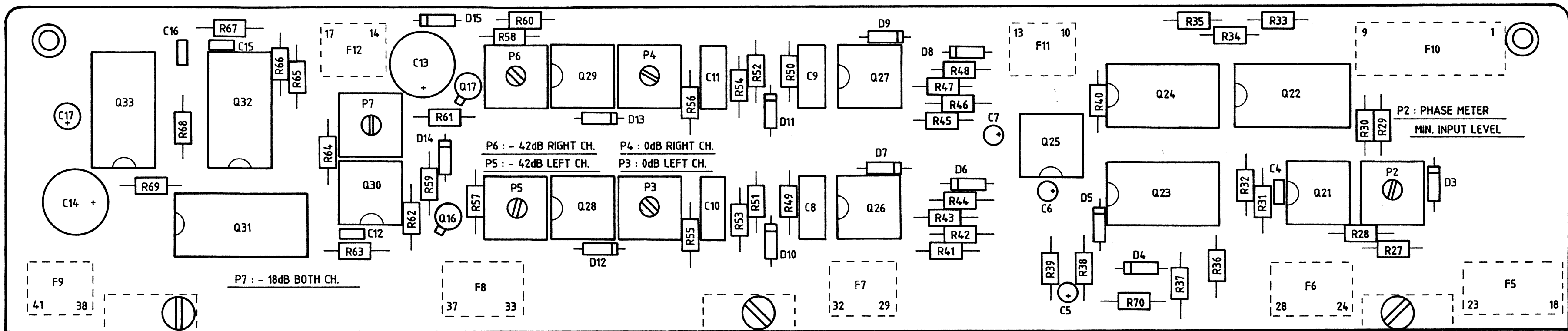
COMP. LAYOUT FOR IK34B & MP09

(np) ELEKTROAKUSTIK AS

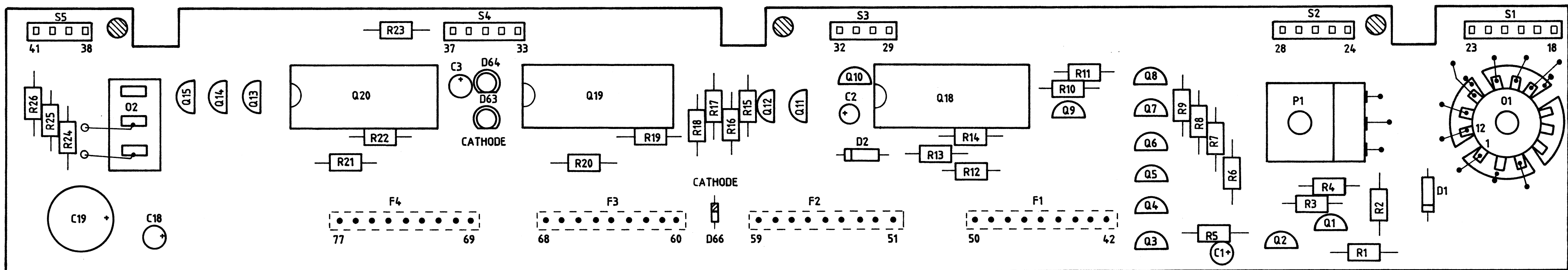
KONSTR.	
TEGN.	UDE apr.88
GODK.	
TEGN.NR.	88 03 25 61



PP	8804	DIV.		COMP. LAYOUT FOR MON 56	KONSTR	
PP	6610	R32, R42			TEGN	UDE
				np ELEKTROAKUSTIK AS	GODK	
					TEGN NR	
MÅLESTOK						86 11 09 62

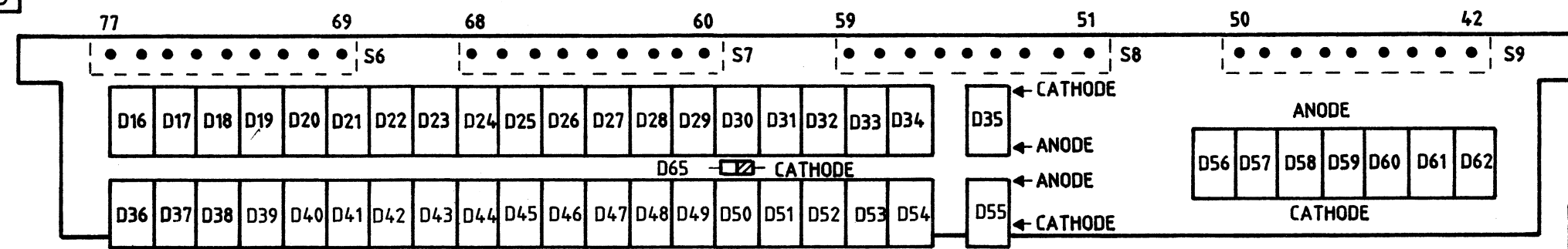


LV 28 PCB



LV 29 PCB

F1 - F12 : MOUNTED ON REAR SIDE



LED 18 PCB

MÅLESTOK	COMP. LAYOUT FOR LV 28, LV 29 & LED 18		KONSTR.	
			TEGN.	UDE apr.88
			GODK.	
			TEGN.NR.	
			ELEKTROAKUSTIK AS	
			86 11 08 62	









INTERFACE

OUTPUT R

OUTPUT L

TAPE INP R

TAPE INP L

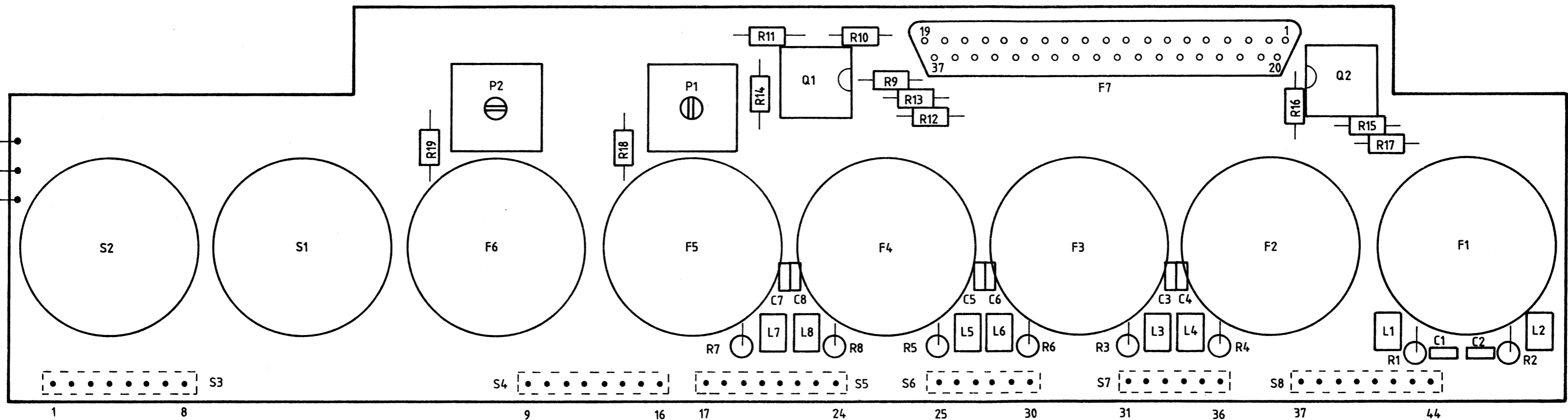
INPUT 4

INPUT 3

INPUT 2

INPUT 1

PHONES  
F8 & F9  
GND  
R(RING)  
L(TIP)



		COMP. LAYOUT FOR CP 06		KONSTR.	
				TEGN.	UDE maj.88
				GODK.	
				TEGN.NR.	
MÅLESTOK	2 : 1	ELEKTROAKUSTIK AS		83 10 05 13	