SCHOEPS
Mikrofone

DSP-4 KFM 360

Digital Microphone Processor for Surround System
with KFM 360 Sphere Microphone

Operating Instructions

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Schoeps GmbH
Spitalstraße 20
D-76227 Karlsruhe
Germany

Tel.: +49 (0)721 943 200
Fax: +49 (0)721 495 750
E-mail: mailbox@schoeps.de
Web site: www.schoeps.de

Overview

The DSP-4 KFM 360 is a 24-bit processor unit. Together with the sphere microphone KFM 360 and two bidirectional compact condenser microphones CCM 8L, it forms a compact 5.1 surround recording system with excellent spatial imaging capability.

The processor derives the six 5.1 surround channels from only four microphone signals (the two pressure transducers in the KFM 360 plus two figure-8s) and offers a wide range of selectable settings.

KFM 360 with two figure-8s (CCM 8L)

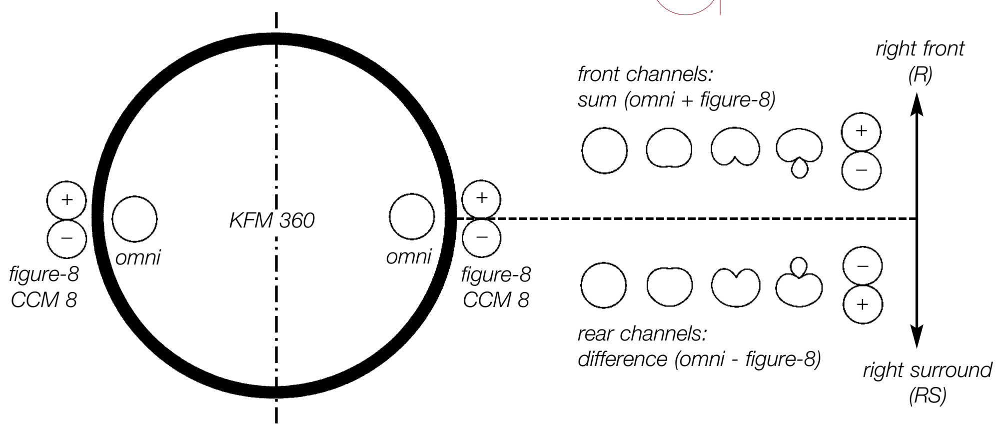
KFM 360:
– for 12 or 48 Volt phantom powering
– two pressure transducers built into the sphere
– diameter: 18 cm.
– pickup angle: ca. 120°

CCM 8L:
– for 12 or 48 Volt phantom powering
– to be aimed frontward, toward the sound source

The primary characteristics of the DSP-4 KFM 360 are:
– analog and digital inputs and outputs
– integrated 48 Volt phantom powering and microphone preamplification
– built-in equalization for the microphones
– digital processing can be performed during or after the actual recording
– analog control "feel"
– capability to store control "presets"
– directionality of the front and rear channel microphones is independently selectable
– the stereo basis width of the front channels is independently variable
– adjustable time delay and low-pass filtering for the rear channels
– sampling rate 44.1, 48 or 96 kHz, 24 bit

In addition to amplifying the microphone signals, in the "B: SURR" operating mode the DSP-4 KFM 360 provides compensation for the low-frequency response of the figure-8 microphones and appropriate diffuse-field equalization for the pressure transducers.

The front-channel outputs result from summing the outputs of the pressure transducers with the outputs of the figure-8s, while the rear-channel outputs result from subtracting these signals (see drawing on next page). The resulting "virtual microphones" face frontward and backward as do the actual figure-8s. The directional patterns of the front- and rear-facing "virtual microphones" may be set independently, from omnidirectional through cardioid all the way to figure-8 (though the extreme settings are of only limited usefulness). This permits flexible adaptation to the particular acoustic conditions of the recording venue, since the choice of directional pattern affects the amount of reverberant energy that will be



Derivation of right front (R) and right surround signals (RS) as proposed by Bruck (rough diagram)

picked up.

If the unprocessed microphone signals are recorded (four channels), the effective directional settings can be chosen retroactively during post-production playback.

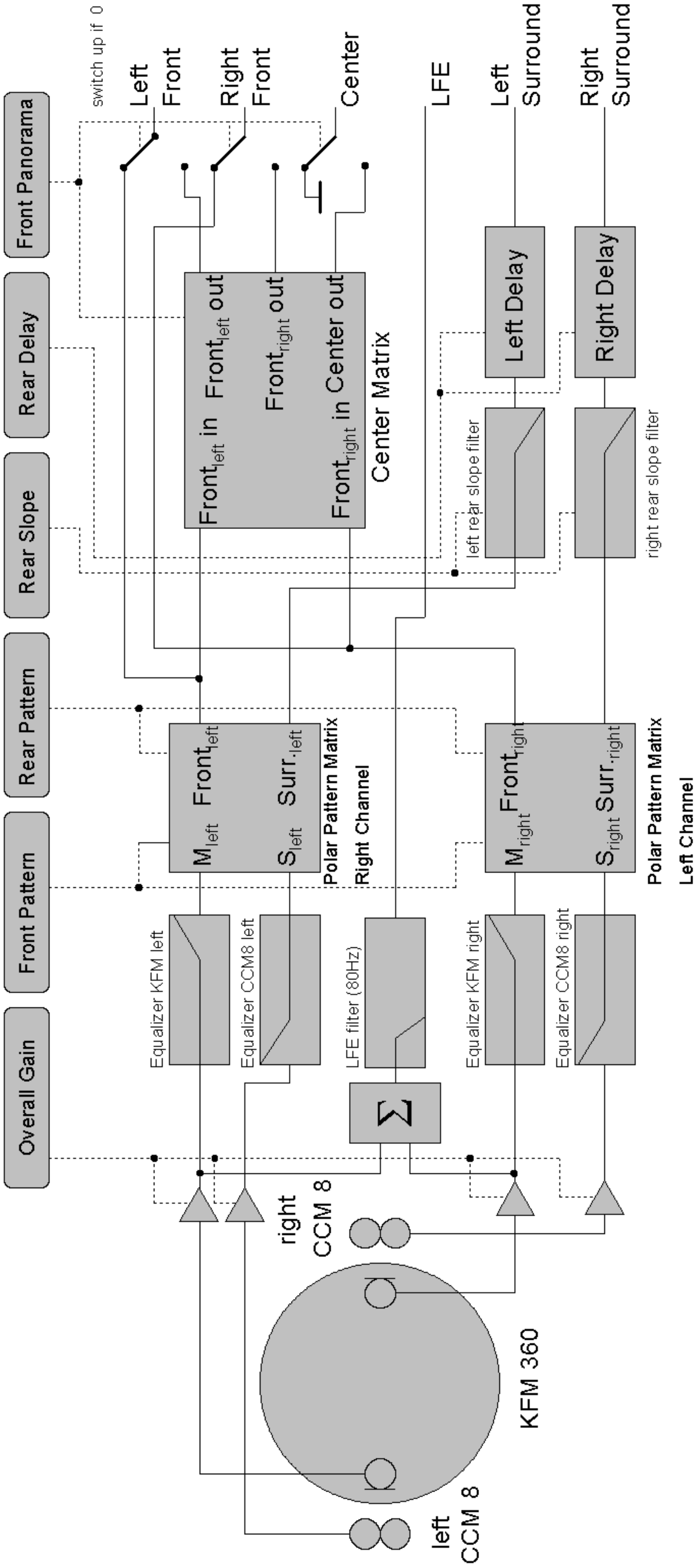
The center channel signal is derived from the front channels by a matrix. An additional channel contains only a low-frequency signal up to 80 Hz, which is obtained by summing the pressure transducer outputs and reducing the total level by 6 dB.

To prevent the disturbing effect of rear-channel localization of sound sources, the rear channel signals can be reduced in level, delayed and/or processed by an adjustable low-pass filter.

In addition to these primary functions, each selector switch also has a secondary function that allows, for example, the relative front-to-back channel gain or the left/right balance to be adjusted. While recording there is also a monitoring function which permits twelve different signals to be heard through Output III, as determined by the setting of the REC MONITOR knob.

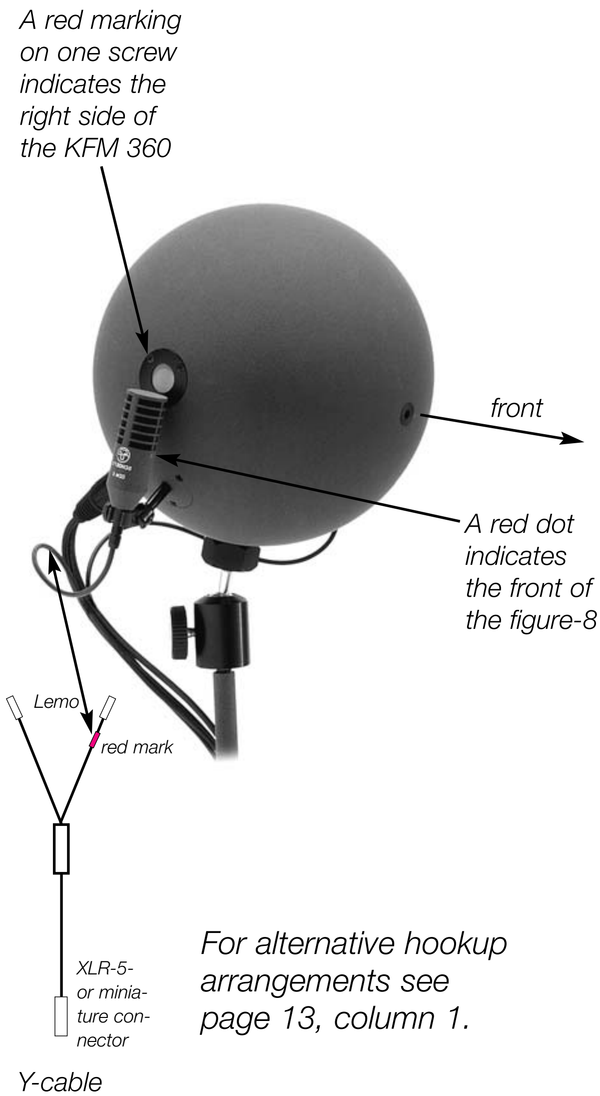
Suggestions concerning microphone placement

The KFM 360's microphones pick up both level and arrival-time differences between the left and right channels. However, the diameter of the KFM 360 has been made smaller than that of the sphere stereo microphone KFM 6 in order to increase the stereo pickup angle. This is because a surround microphone will often be placed closer to the sound source than a microphone might be for conventional stereo pickup. The rear-channel loudspeakers will reproduce components of the diffuse sound field, thus giving an impression of the hall's spatial character. This reduces the difficulty of finding a microphone position that will provide both a clear stereophonic image and also a good balance of direct and reverberant sound.



Signal processing flow in the DSP-4 KFM 360 (MODE B: SURR.)
(In the "REC" mode the equalization for omni and figure-8 is not active.)

Before powering on the unit for the first time, please make certain that the voltage selector is set correctly for your area. The voltage selector is on the rear panel beneath the on/off switch; see photos and instructions on page 12.



Setting up and connecting the KFM 360 and CCM 8Ls

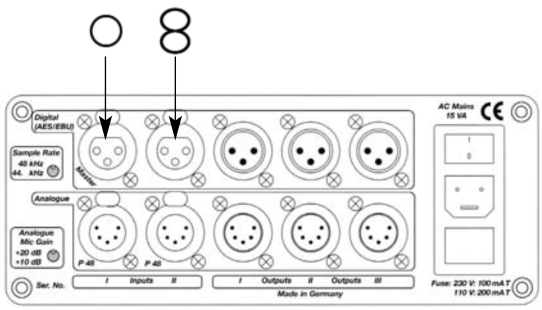
Note:
When the KFM 360 is suspended (e.g. from the ceiling), left and right are reversed.

Recording for Post-Production

Simply connect the microphones to a digital four-channel recording device that offers 48V phantom power and the needed gain.

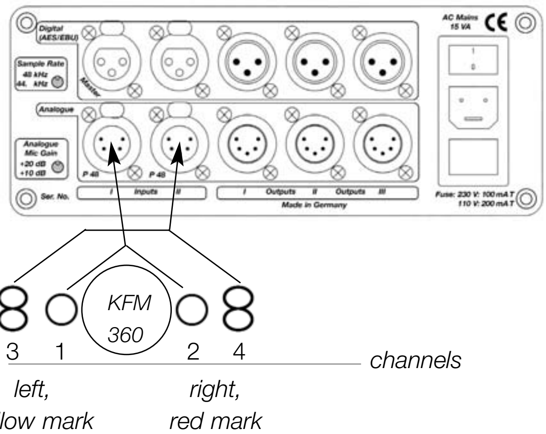
The signals of the KFM 360's omnis must be connected to input I and the figure-8s to input II (see bottom of this column; channel I = left side = yellow mark on the SCHOEPS Y-cable). This way each pair will correspond to one AES/EBU signal.

For post-processing, the two AES/EBU signals should be connected to the digital inputs I (omni) and II (figure-8):



You can use the DSP-4 KFM 360 as a microphone preamplifier and/or A/D converter

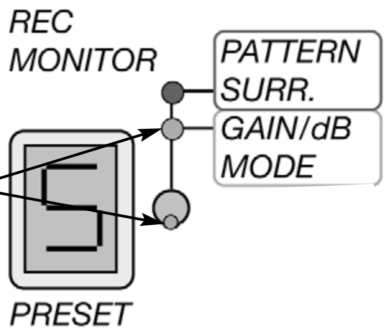
You can use the DSP processor as a microphone preamplifier together with its A/D converter.



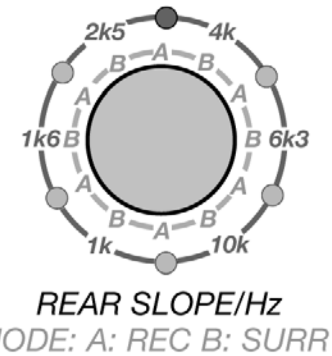
How to connect the KFM 360 and the CCM 8Ls to the processor

While F/REAR BALANCE and L/R BALANCE usually should be in the “0” position, the OVERALL GAIN knob controls the basic volume. It can be adusted when the fifth selector switch is turned so that only one LED in its surrounding circle lights up. The OVERALL GAIN serves to control mastering devices.

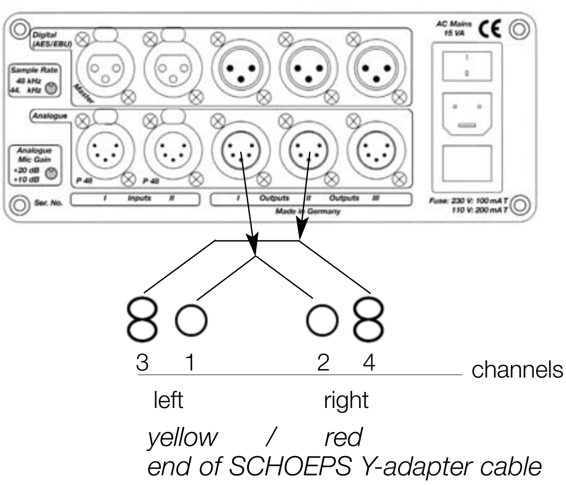
With the function control in the lower position (orange mode = “GAIN/dB MODE”)...



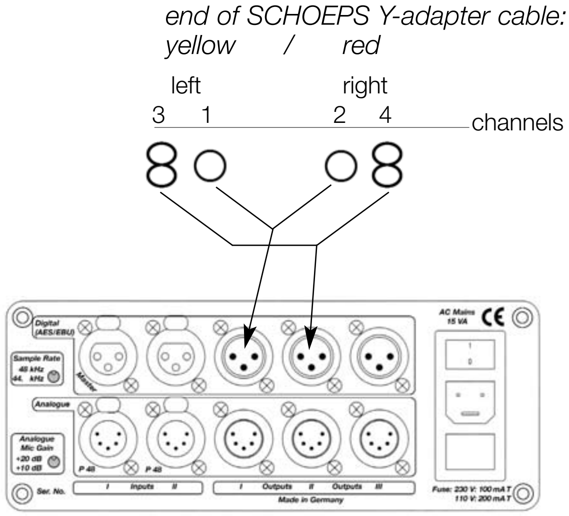
...this control must be set to “A” (only one LED will be lit).



Signals at the analog outputs (XLR-5):



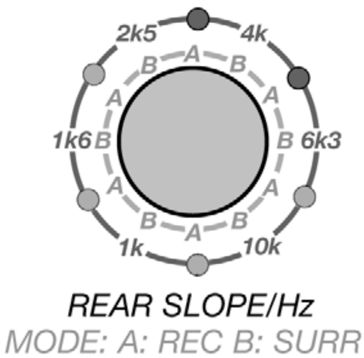
Signals at the digital outputs:



Processing and Recording simultaneously

If processing is to be done while recording, connect the microphones as shown on page 6. Then set the function control to the GAIN/dB mode (see illustration in column 1 on this page). Finally, turn the REAR SLOPE/Hz / MODE knob until a pair of LEDs in its surrounding circle lights up.

For processing the signals this control must be set to “B”, which is the case when any pair of LEDs lights up.



Processed analog outputs:

	channel	pins	+ phase at pin	color marking*
Output I:	left front	2, 3	2	yellow
	right front	4, 5	4	red
Output II:	center	2, 3	2	yellow
	LFE	4, 5	4	red
Output III:	left surround	2, 3	2	yellow
	right surround	4, 5	4	red

* with SCHOEPS Y-cable AKSU/2U (XLR-5 to 2x XLR-3)

Processed digital outputs:

	channel
Output I:	left front
	right front
Output II:	center
	LFE
Output III:	left surround
	right surround

Note:
The position of the function control determines whether you are using the function settings of the first (green) layer or the second (red) layer. Only the settings in the currently selected layer will be changed.

The center channel is not active when the “FRONT PANORAMA” is in the lowest position (function control in “PATTERN SURR.” position = green mode).

For lowest noise, the “Analogue Mic Gain” switch on the rear side should be in the upper position (+20 dB). The “+10 dB” position is reserved for exceptionally high levels.

The indicated polar patterns are nominal settings which will be precisely valid only if the gains of all the record/playback channels are carefully matched.

Front panel

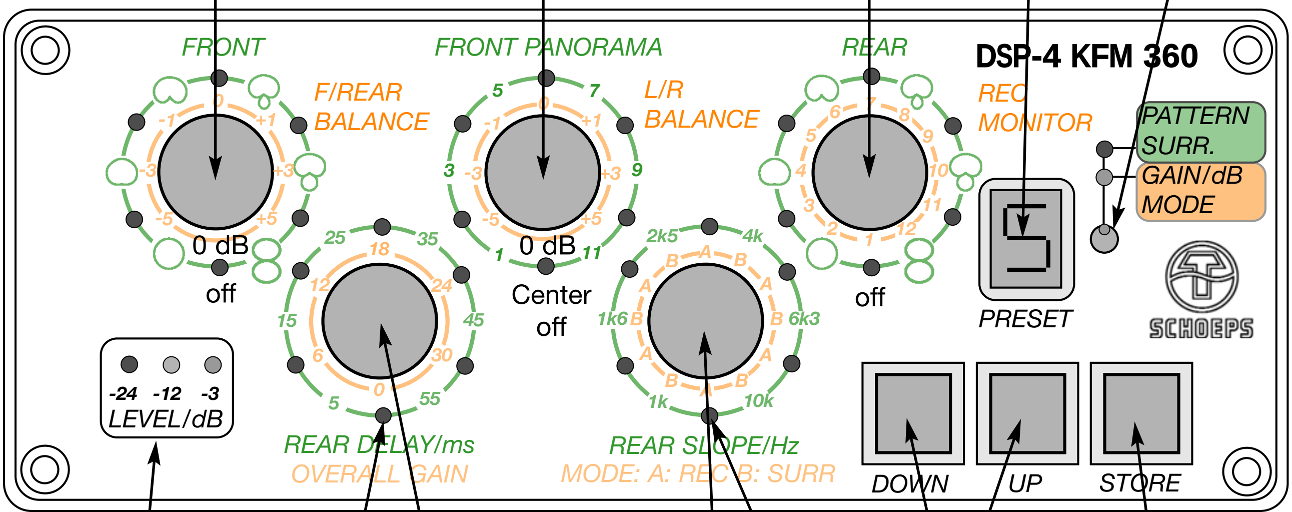
controls which are active in the surround mode (MODE B of selector switch 5)

controls which are active in the recording mode (MODE A of selector switch 5)

	Selector switch 1:	***Selector switch 2:***	***Selector switch 3:***
PATTERN SURR.	*polar pattern front microphones*	*Front stereo width pos. 12: Center off*	*polar pattern rear microphones*
GAIN/dB MODE	*Balance front/rear*	*Balance left/right*	*Monitor output selector*

*Number of the current **preset***

***Function control** (determines the set of functions of the five selector switches)*



0ms

∞ Hz (low-pass off)

***Indicators** for internal digital signal levels*

	Selector switch 4:	***Selector switch 5:***
PATTERN SURR.	*delay of the rear channels*	*Turnover frequency for rear-channel low-pass filter*
GAIN/dB MODE	*Overall gain control*	*A: Raw microphone signals (for post-production processing) B: Processing of microphone signals enabled*

*Pushbuttons for **selecting a preset***

*Pushbutton for **storing** the current **settings** as the current preset*

In the recording mode (MODE A of selector switch 5) this is the only control that is in effect.

Rear panel

Digital ***inputs***
(recorder playback)

Master* *(AES/EBU) for synchronizing the internal A/D converters*

Slave *(AES/EBU) switches over to the digital inputs*

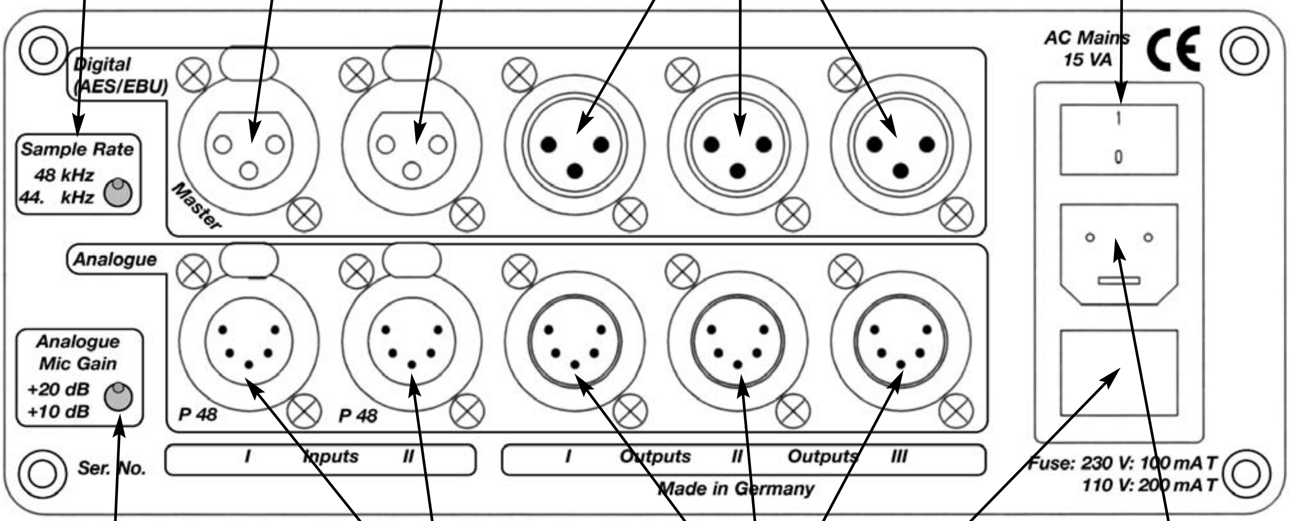
**The reason for having the master clock function and the slave input on two separate sockets is that this allows the converters to be slaved to an external source while the analog inputs are being used (see example on page 17).*

Sampling rate *selector (internal clock for A/D converters)*

Digital ***outputs***

Power on/off switch

Channel assignment just as with the analog in/outputs



Safety fuse / AC voltage selector

Power cord socket

Gain selector *for analog inputs*

Analog microphone ***inputs*** *with 48 V phantom powering (muted if digital master and slave inputs are connected)*

Input I:
channel 1: left omni
channel 2: right omni

Input II:
channel 1: left figure-8
channel 2: right figure-8

Analog ***outputs:***

	SURR mode (B):	REC mode (A):
Output I:		
channel 1:	*left front*	*left omni*
channel 2:	*right front*	*right omni*
Output II:		
channel 1:	*center*	*left figure-8*
channel 2:	*bass (LFE)*	*right figure-8*
Output III:		
channel 1:	*left surround*	*left monitor*
channel 2:	*right surround*	*right monitor*

AC power cord socket / Power on/off switch

AC power is connected to the back of the DSP-4P. A single module contains the on/off switch, the power cord socket and the main safety fuse holder. The processor can be powered by 230 VAC or 110 VAC. The normal factory setting of 230 VAC can be recognized from the fact that the "220 - 240 V" label is in the upright position, and that the arrowhead adjacent to it is the one pointing toward the small, white solid rectangle on the lower border below the fuse compartment. The voltage setting must be correct for the AC (mains) voltage in your region of operation.

230V setting = factory setting

110V setting

The AC voltage setting / Changing the fuse

The fuse compartment on the back of the DSP-4P provides the means for setting the power supply voltage:

To change the setting, unplug the AC power cord from the socket and insert the tip of a flat-blade screwdriver into the slot at the lower edge of the power cord socket (see photo above on the right side). Carefully pull the fuse compartment out. To facilitate changing the fuse, unfold the small tabs that are on either side of the fuse holder:

When the proper value fuse has been inserted, turn the fuse holder so that the intended voltage label is right-side-up, and carefully slide the fuse compartment back into the DSP-4P.

Please note:
Both a 100 mA and a 200 mA fuse are supplied, each in the corresponding position of the compartment.
The fuse rating for 230-Volt operation must be 100 mA, and for 110-Volt operation, 200 mA (slow blow in both cases); fuse size is 5 x 20 mm or 6 x 30 mm. It is important for the fuse to be a 230-Volt type even if the actual AC voltage to be used is much lower. Also, when changing the unit from 110- to 230-Volt operation it is important to install the lower-rated (100 mA) fuse. A 200 mA fuse at 230 Volts would not provide adequate protection; operating the unit in this manner is potentially unsafe and will void the unit's warranty.

Under no circumstances should any other type of value of fuse be used other than the types and values given above.
If fuses burn out continually, send the unit to the factory, dealer or representative for repair.

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Connecting the microphones

Analog inputs

Input I: XLR-5; the KFM 360 is connected here.

Input II: XLR-5; The two figure-8s are connected here. The following combinations of connecting cables can be used:

– 2 adapter cables of type K 5 LU (as included with the CCM 8 Lg) plus one Y-adapter AK 2U/SU (2x XLR-3 to XLR-5)
– 1 Y-adapter KLY 250/0 SU, which goes from two Lemo connectors on the microphone ends of the cable (250 mm long each) and terminates in an XLR-5M cable 10 cm long; XLR-5 cables can be used as extensions.
– 1 Y-adapter KLY 250/5SU (as above, but with a 5-meter XLR-5M output cable)
– 1 Y-adapter KLY 250/0 I (as KLY 250/0 SU, but with a miniature output connector instead of an XLR-5M). Adapter cable KS 5 IU is also available, which goes from this type of miniature output connector to an XLR-5M.

Both ends of the right-channel cable (II) are marked in red on all Y-adapters.

Note: The analogue inputs are exclusively intended for connecting condenser microphones. Hence the 48 V phantom powering is permanently enabled.
If, however, a device is connected here without any decoupling, this can lead to a serious damage of its outputs.

Digital inputs:

These inputs are designed for connecting a four-channel recorder with AES/EBU outputs. The raw (unequalized) microphone signals can thus be processed after the recording. Each input should receive the signals from one pair of the microphones. Channel assignments match those of the analog inputs: Input pair I = KFM 360; input pair II = the two figure-8s.

A four-channel recorder will use both the master and the slave input sockets. The processor will thus receive word clock from the external recorder, and the fact that a connection has been made to the slave input will switch the unit from its analog to its digital signal inputs for audio signals.

If a two-channel digital recorder is connected to the master input, an additional connection to the slave input will be needed in order to cause the processor to switch over to its digital audio inputs. This is important when deriving a center-channel signal from a two-channel KFM 360 recording or when the processor is being used as a two-channel D/A converter.

Setting the analog signal levels

The "Analogue Mic Gain" switch on the left side of the rear panel sets the input gain for all four analog inputs. "+20 dB" is the normal setting for 10 mV/Pa condenser microphones. Only for sound pressure levels above 120 dB (with microphones of approximately that sensitivity), or when a recorder's line-level analog outputs are connected for post-processing, should the "+10 dB" gain setting be used.

Setting the level of input signals prior to processing

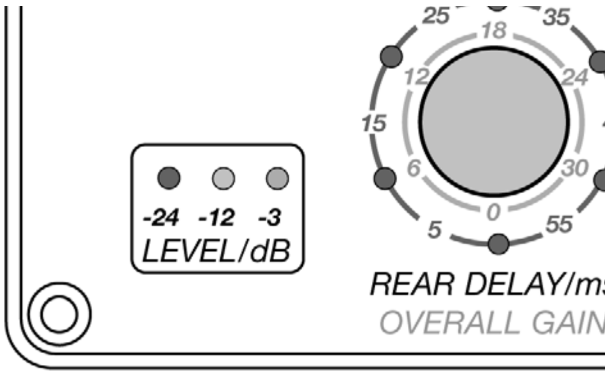
The level of input signals-including digital signals-can be adjusted in 3 dB steps prior to any other processing within the DSP-4 KFM 360. Up to 33 dB of gain may be applied at this stage. The gain is set by using the “REAR DELAY/ms”, “OVERALL GAIN” control in its secondary function (set by the function control switch which has got to be in the “GAIN/dB MODE” position).

Note: The higher the gain, the smaller is the dynamic range (headroom).

Level indicators

The "Level" LEDs indicate the digital signal levels within the DSP-4P. At any given moment they display the highest level occurring anywhere within the signal-processing system.

The -3 dB LED should light only during the highest peak inputs, i.e. only very rarely. Otherwise, in an extreme case digital clipping could occur, which might become audible.



Connecting the Outputs

MODE: A: REC

The signals at outputs I and II are unprocessed, i.e. without frequency response compensation. This operating mode leaves output III free for use as a monitor output. In mode "B: SURR" output III will carry the rear channel signals, so it may be necessary to reroute these signals to the front or main loudspeakers. See following page for further details concerning the monitoring function.

MODE: B: SURR

Processed signals are present at the outputs as follows:

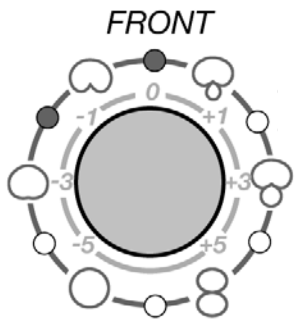
Output I:	channel I:	L (front left)
	channel II:	R (front right)
Output II:	channel I:	Center
	channel II:	Bass (LFE)
Output III:	channel I:	LS (rear left)
	channel II:	RS (rear right)

Settings:

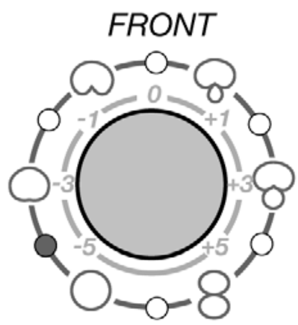
Setting the directional pattern: (FRONT, REAR)

To make this setting, the function switch must be in the upward position ("Pattern, SURR.").

The directional patterns of the front-channel microphones can be set with rotary selector switch 1 ("FRONT") in the upper row, and the patterns of the rear-channel microphones can be set with rotary selector switch 3 ("REAR"). The selected directional pattern is shown by LEDs that surround each rotary switch. For example:



selector switch 1 in “cardioid” position



selector switch 1 in position between omni and wide cardioid

Note: If the knob for the front channels is turned until the LED directly underneath it lights up, these channels will be turned off.

If, however, the corresponding knob for the rear channels is put into this position, their characteristic will be identical with those of the front channels.

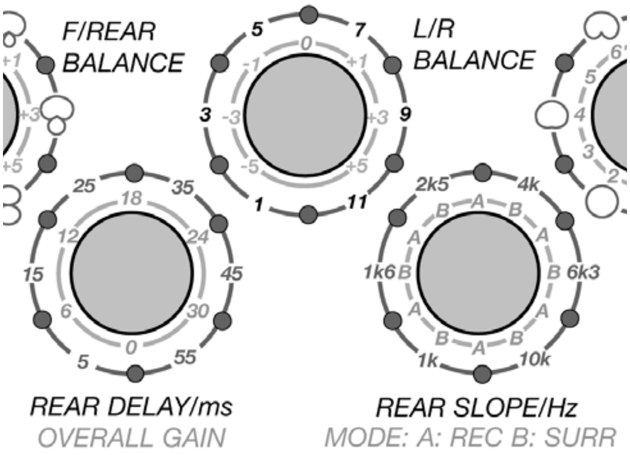
The Center Channel

The center channel is formed by matrixing the front-channel signals. It can be turned off by setting the "FRONT PANORAMA" selector switch to its lowest position. However, it will then no longer be possible to adjust the stereo soundstage width.

Additional processing for the front channels:

The center channel and the "FRONT PANORAMA" setting

The center channel is formed by matrixing the front-channel signals and thus contains their correlated signal components (those common to the two channels). The use of this channel makes it necessary to decorrelate the front left and right loudspeaker signals in order to keep the stereo image from becoming weighted too heavily toward the center. The degree of decorrelation is controlled by the "FRONT PANORAMA" knob. If this knob is set to the bottom position, the center channel signal is not produced, the matrix is turned off and the decorrelation circuitry will have no further effect on the L and R signals.



The "L/R BALANCE" control has the same function as in a conventional stereo system, except that its bottom-center setting has the same effect as the top-center setting: the balance of the two channels is not altered.

Additional processing for the rear channels:
F/REAR BALANCE, REAR DELAY/ms
REAR SLOPE/Hz

To avoid hearing the rear loudspeakers--or more precisely, to avoid localizing sound sources in them in a distracting manner--there are three independent settings which can be used in any desired combination:

1. Lowering the rear channel level relative to the front channels (F/REAR BALANCE);
2. Delaying the rear channels relative to the front channels by up to 55 ms (REAR DELAY/ms)--in the bottom-center setting, the delay is 0.
3. Rolling off the high-frequency response of the rear channels (REAR SLOPE/Hz) at a 12 dB/octave slope--in the bottom center setting, no high-frequency rolloff is applied

Please note: Although adjustable delay is a feature normally included only in more ambitious digital mixing desks, the DSP-4 KFM 360 processor offers such a function. However, the amount of the delay should not be changed while the unit is recording in Surround mode, because such a change may produce audible side effects.

Additional Low-Frequency Channel

The LFE ("low frequency enhancement") channel contains the blended, low-pass-filtered information from the two pressure transducers of the KFM 360 ((L + R)/2). It does not correspond in function to the LFE channel of a cinema surround sound system [i.e. special effects such as earthquakes].

The -3 dB point of the low-pass filter is 80 Hz. It is critically damped with a fourth-order rolloff (24 dB/octave).

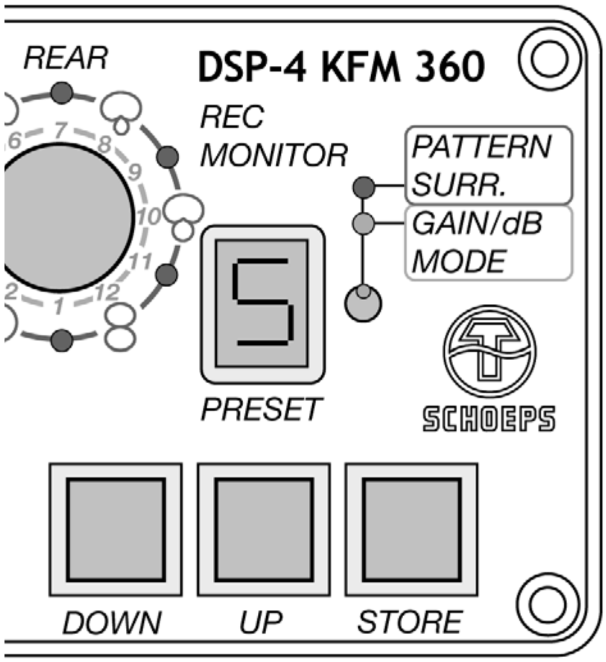
Monitor function REC MONITOR

In recording mode "MODE: A: REC" the input signals are passed through and are presented unchanged (apart from the possible A/D and D/A conversions) at outputs I and II. Output III in this mode is used for monitoring. Selector switch 3 allows the following options to be chosen:

1. KFM 360 with frequency response correction applied
2. Figure 8s with frequency response correction applied
3. L and R (FRONT PANORAMA not used; no center channel signal available)
4. L and R (FRONT PANORAMA active)
5. Center channel
6. LS and RS without low-pass filter
7. LS and RS with low-pass filter

The next two monitor settings allow the effect of the delay to be monitored:

8. LS channels with low-pass filter: left channel with delay / right channel without delay
9. SR channels with low-pass filter: left channel with delay / right channel without delay
10. KFM 360 with no frequency response correction applied
11. Figure 8s with no frequency response correction applied
12. Additional Low-Frequency Channel (LFE)



Buttons and display for preset functions

Storing and Recalling Presets

Certain groups of settings will probably reveal themselves as your favorite ways of using the DSP-4P. To set these up rapidly, the unit is capable of memorizing ten groups of settings and of calling them up instantaneously as "presets" 0 through 9.

Please note: The groups of stored settings include the position of the five main knobs in their primary functions, all of which are controlled in the “green mode”: FRONT pattern, FRONT PANORAMA, REAR pattern, REAR DELAY and REAR SLOPE/Hz. The settings of the secondary functions are not stored.

Storing presets:

The numeric display will always show the number of the most recently recalled or stored preset. The current settings will be stored as that number of preset whenever the "Store" button is pressed. If a group of settings has been associated with a given preset number, a small red point will light up within the display alongside the numeral.

If you intend to store a group of settings as a certain preset number, press the “store” button and keep it pressed while using the up and down buttons to find the desired number. If then you release the "store" button, the settings will be stored as the indicated preset. To store a group of settings as the preset number which is already being displayed, merely press and release the “store” button.

Recalling presets

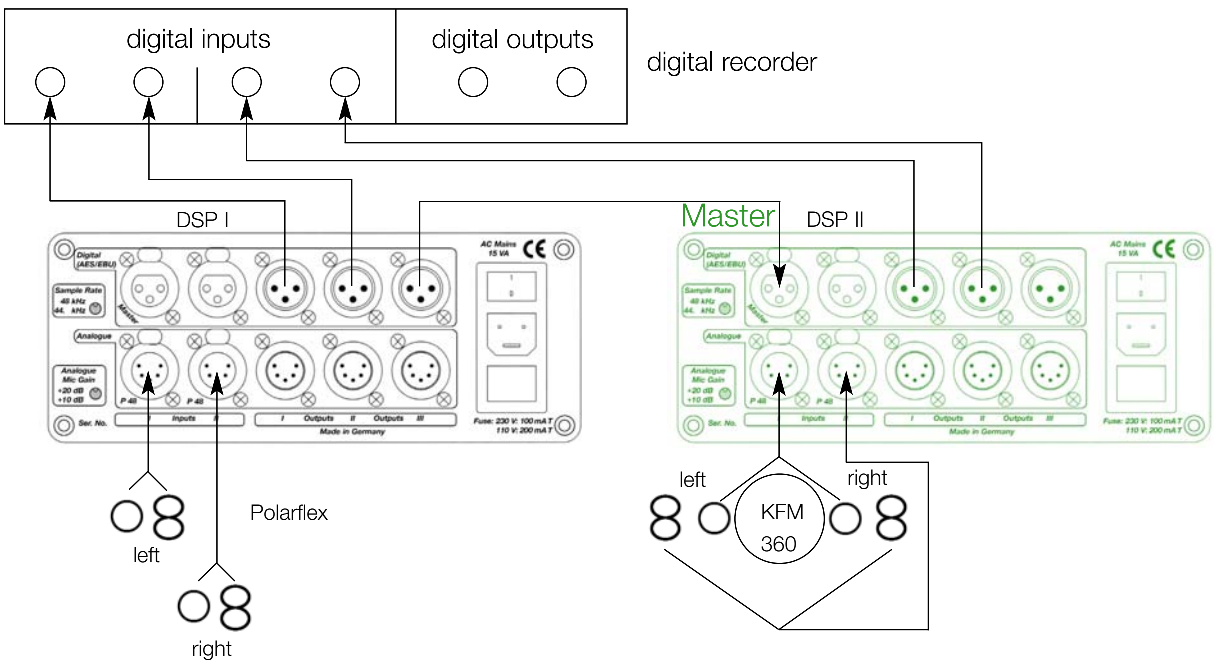
Stored presets are recalled simply by pressing the "Up" and "Down" buttons until the desired preset number appears. The current knob settings are replaced immediately by the stored settings of the indicated preset number.

Please: If settings are called up and then changed by turning one or more of the knobs, these changes will be lost if the user presses the "Up" or "Down" button without first storing the new settings as a preset.

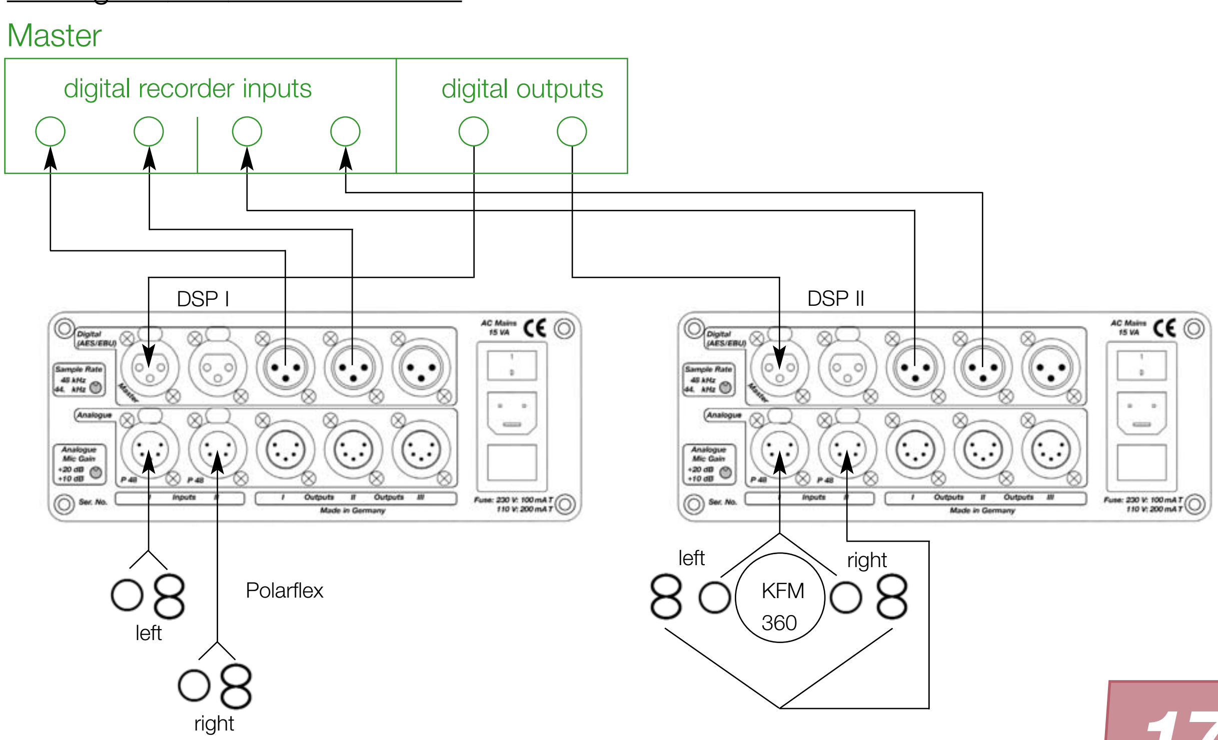
How to synchronize two DSP-4 KFM 360 or a DSP-4 KFM 360 + DSP-4P PolarFlex processor with one recorder

If it should be necessary to synchronize two DSP-4 processors, connect the units as shown here:

DSP no. I is the master of DSP II and the digital recorder:



The digital recorder is the master of DSPs I and II:



Modes of operation

Since the DSP-4 KFM 360 offers both analog and digital inputs and outputs, several operating modes are possible:

Input	Output
Analog signals from microphones Digital signals from a recorder	Passed-through input signal (analog and digital) Processed signal (analog and digital)
Internal clock: 44.1, 48 or 96 kHz **External clock:** via MASTER AES/EBU input (25 - 100 kHz)	
Digital input signals (e.g. from a recording device)	Passed-through input signal (analog and digital) Processed signal (analog and digital)
The DSP-4KFM 360 will switch over to its digital inputs if a signal is presented at the "Slave" AES/EBU input socket. **The unit connected to the MASTER AES/EBU socket will determine the clock frequency (25 - 100 kHz).**	

Specifications:

Analog inputs (2 pairs, with 48 Volt phantom powering)

"Analogue Mic Gain":	**"+10 dB"**	**"+20 dB"**
Gain:	+10 dB	+20 dB
Maximum input level:	-5 dBU	-15 dBU
Maximum sound pressure level with CCM 2 S (omnidirectional) :	130 dB SPL	120 dB SPL

Analog outputs (3 pairs, balanced):
2 pairs of pass-through outputs, one pair of processed signal outputs
Maximum analog output level: +4.5 dBU
Digital inputs (2 pairs):
Data format: AES/EBU
Digital outputs (3 pairs):
2 pairs of pass-through outputs, one pair of processed signal outputs
Data format: AES/EBU
Synchronization/clock frequency:
with internal synchronization: 44,1/48/96 kHz
with external synchronization (via master inputs): 25 - 100 kHz
Maximum digital gain: 33 dB
Dynamic range:
A/D converter: 98 dB (peak)/CCIR
110 dB (RMS)/A-weighted
(108 dB (RMS)/unweighted)
D/A converter: 100 dB (peak)/CCIR
113 dB (RMS)/A-weighted
(110 dB (RMS)/unweighted)
Characteristic impedance of the cables which are connected to the digital outputs: 110 Ohm
AC mains voltage: switchable 110-120 V / 220-240 V
Power consumption: 15 VA
Fuse: 230 V: 100 mAT; 110 V: 200 mAT
Dimensions (W×H×D): 220 × 90 × 246 mm
(8,66 × 3,54 × 9,69 ")
Weight: 2.7 kg (6 lbs)

Important

The DSP-4P is designed for indoor use only. To prevent damage to the unit and shock hazard to the user, do not allow liquids to enter the housing.

Do not operate the unit in strong, direct sunlight because of the risk of overheating. For the same reason the openings for cooling must not be covered.

Proper use: We hereby state that the DSP-4P is for use only in studios, domestic installations, concert halls, churches etc.; it is not designed for use in vehicles of any kind, especially in airplanes, airborne equipment or public transport, nor in hospitals or in any other setting in which the existing equipment may be sensitive to electromagnetic radiation.

Cables: For the digital connections use only shielded cable with 90% or better shield coverage in order to prevent undue radiation of radio-frequency signal energy. All SCHOEPS cables meet this requirement.

If any such cables should exceed 20 meters in length, they must have a characteristic impedance of 110 Ohms and must also be terminated with 110 Ohms according to the standard AES 3-1992. This is automatically the case with the DSP-4 KFM 360.

Declaration of conformity

For the DSP-4P Schalltechnik Dr.-Ing. SCHOEPS GmbH herewith declare that it complies with the directive 89/336/EWG on EMC (electromagnetic compatibility) of the EC council.

This product is not subject to further directives.

For the judgement of this product in respect of EMC the following standards are applicable:

EN 55103–1, EN 55103–2

The manufacturer accepts full responsibility for this declaration.

Warranty

We guarantee our products for a period of twenty-four months, except for cables, batteries and cells (including rechargeable batteries and cells) and any other products of other manufacturers for which SCHOEPS is only the reseller; for these products the period of guarantee is six months. The guarantee period begins on the date of purchase. Please provide your bill of sale in all cases as proof of guarantee; without it, repairs will be undertaken only at the owner's expense.

We reserve the right to satisfy all warranty requirements regarding defects of workmanship or materials by means of repair or partial or complete replacement of the unit, at our sole discretion.

Excluded from this guarantee are defects due to misuse (e.g. incorrect operation; mechanical damage), abuse or “acts of God.” This guarantee is nullified in the event of tampering by unauthorized persons or agencies.

To secure your rights under this guarantee, send the unit with all included accessories and proof of purchase, at your expense, either to SCHOEPS (if you are a customer in Germany), or to our representative (if you are a customer outside of Germany).

In exceptional cases you can, by prior arrangement with SCHOEPS, send a unit directly to us from a foreign country. But since any return shipment to a foreign customer must be prepaid, this would be slower especially when the conditions for service under guarantee are not met; all payment must then be carried out before the repaired item could be returned to the customer.

This guarantee does not affect any contractual agreements which may exist between the buyer and seller of the equipment.

This guarantee is world-wide.

SCHOEPS
Mikrofone

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