



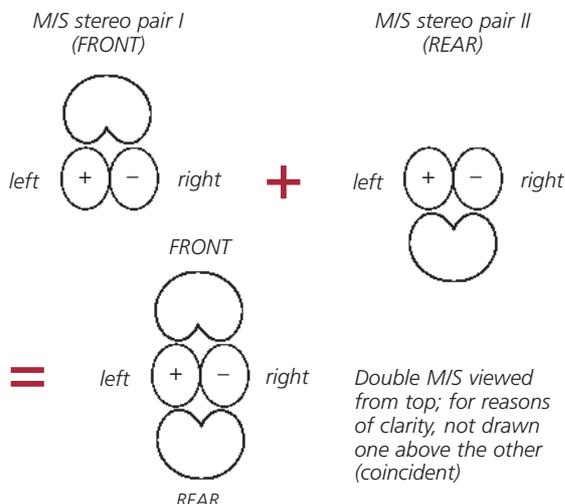
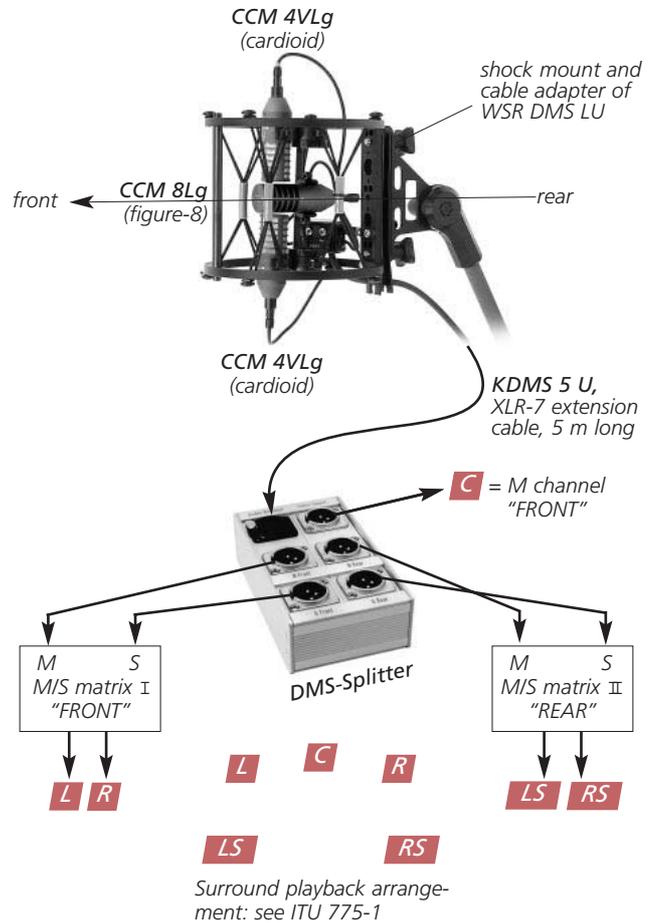
SURROUND RECORDING TECHNIQUES

"Double M/S" Surround with Double-MS Set

- requires only three microphones and channels for 5.0 surround
- very small, lightweight array
- can be well protected against wind (with "fur" Windjammer)
- allows post-production processing (if matrixing takes place after the recording)

"Double M/S" is an improved version of the well-known M/S stereo recording technique. In addition to a front-facing cardioid or supercardioid "mid" microphone and a figure-8 "side" microphone, a rear-facing directional microphone is set up. The front- and rear-facing microphones share the signals of the figure-8 microphone so as to form two complete, back-to-back M/S systems. One M/S system then provides the three front channels (the center channel signal being provided directly by the mid microphone of the front system), while the other system provides the two surround channels. A double M/S arrangement of this kind allows flexible processing of the stereo surround image width and post-production adjustment, just as with two-channel M/S recording.

The Double-MS Set consists of a special shock mount with three CCM-L miniature microphones in a double M/S configuration, a windscreen and a Windjammer. (Instead of the CCM 4VL, a CCM 41VL supercardioid can be ordered.) A cable adapter from three Lemo sockets to an XLR-7 output connector is included as well as an XLR-7 extension cable (5 m long) and a DMS-Splitter. The latter is a passive device that simplifies connecting a double M/S array to phantom-powered preamplifiers with matrix circuitry (e.g. 2x Schoeps VMS 5U) or to the inputs of a mixer. It divides the signals from the three microphones to five outputs (the center channel plus the two M/S pairs, with two of the microphones serving dual functions), while preventing any overlap in the phantom powering of the microphones from the inputs to which it is connected.



WSR DMS LU
= shock mount with
windscreen (150 mm
dia.) and cable adapter



WSR DMS LU
with fur cover
(Windjammer)



1. OCT Front System

OCT=Optimized Cardioid Triangle by Dr. Theile

see AES 19th International Conference, pages 210 - 229
and also www.hauptmikrofon.de

OCT is a method for picking up the three front channels of a surround recording. It can be combined with any of several possible methods for obtaining the rear channels (and thus OCT surround), as will be shown on the following pages.

The preferred setup for OCT uses a forward-facing cardioid for the center channel. For the front L and R channels two supercardioid microphones are placed at opposite ends of an imaginary line running about 8 cm. behind the center microphone. These two microphones should be 40 – 90 cm apart, depending on the required recording angle, and must face squarely outward, away from center (see diagrams below and on the following pages).

Good separation between the center-to-left sector and the center-to-right sector is obtained with this method. For example, sound originating from half right is picked up only very weakly by the left microphone. Sound from the extreme right will be picked up directly on-axis by the right-facing supercardioid (0 dB) and by the forward-facing cardioid (attenuated by 6 dB due to its directional pattern). Finally it will be picked up, with a delay caused by the increased distance, on the rear lobe of the left-facing supercardioid. The polar pattern attenuation for this will

be 10 dB and the polarity will be inverted. These factors prevent the formation of annoying “phantom images” of sound sources in the wrong sector during playback.

A clean center channel is conveyed by this system, since front-incident sound is picked up mainly by the cardioid in the center. Because of their high directivity, the left and right microphones pick up front-incident sound only at a much lower level.

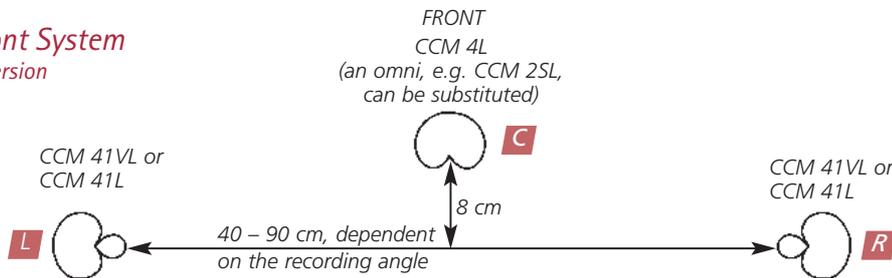
If the cardioid is placed 8 cm forward, the following recording angles will result, depending on the distance between the supercardioids:

40 cm: 160°	50 cm: 140°
60 cm: 120°	70 cm: 110°
80 cm: 100°	90 cm: 90°

It is better to err in the direction of greater spacing, so that one can be sure to avoid center-heavy images.

The fact that the supercardioids receive so much of their sound from off axis necessitates the use of small-diaphragm condenser microphones, as only this type of microphone has the requisite independence of frequency response from the angle of sound incidence. The CCM 41V and the MK 41V are particularly well suited for this application but the CCM 41 or MK 41 can also be used.

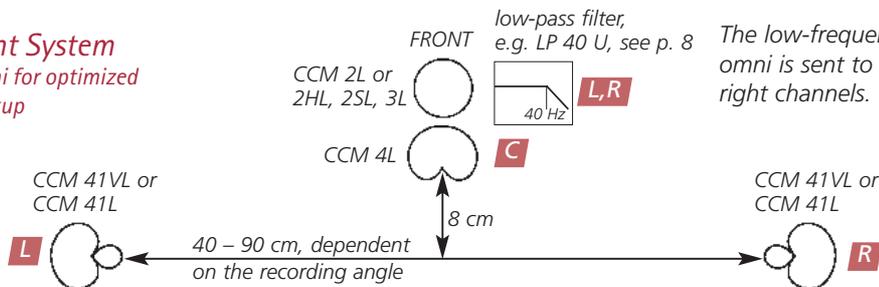
1.1 OCT Front System
basic version



It is possible to improve the extreme bass response of the supercardioids by adding signals from one (see 1.2) or two (see 1.3) omnis to L and R. Using an LP 40 low-pass

filter (cutoff frequency 40 Hz) and Schoeps omnis, the response curve below 100 Hz becomes substantially flat.

1.2 OCT Front System
plus omni for optimized bass pickup

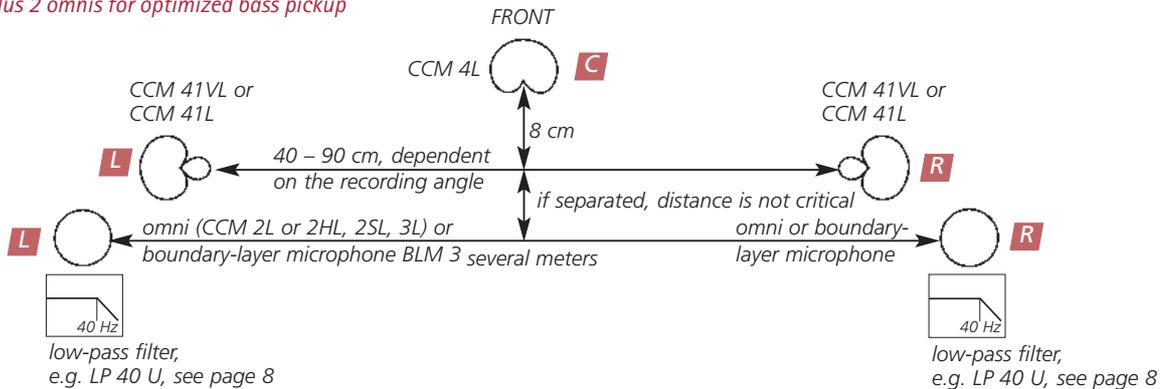


The low-frequency signal from the omni is sent to both the left and right channels.



1.3 OCT Front System

plus 2 omnis for optimized bass pickup



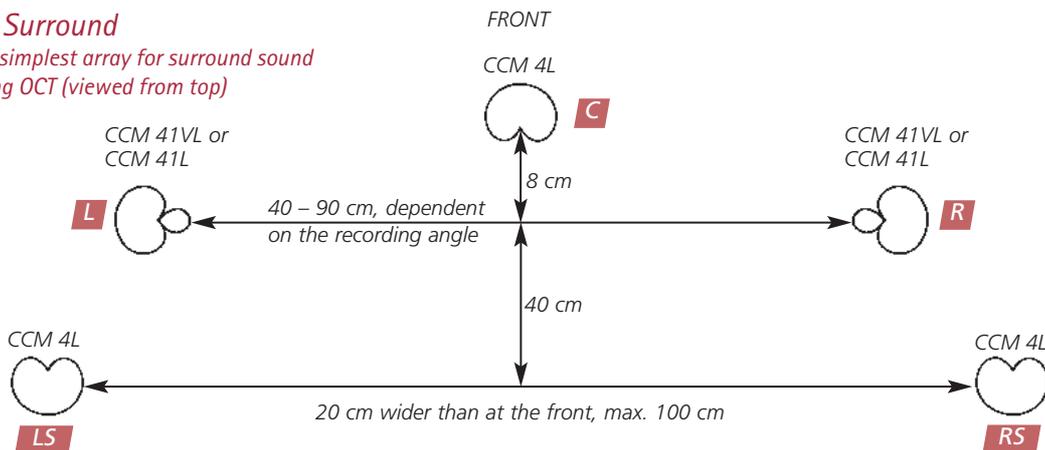
David Griesinger (Lexicon) has proposed that L and R bass signals be decorrelated by using additional widely-spaced pressure transducers. At low frequencies these not

only increase the low-frequency pickup but also accentuate the difference between R and L. This results in an increased sense of spaciousness.

Just as there are many possible arrangements for front channels in surround recording, so there are several ways to record the rear channels.

2. OCT Surround

the simplest array for surround sound using OCT (viewed from top)



The surround cardioids face rearward to avoid picking up direct sound. Time-of-arrival and level differences between each side's cardioid and hypercardioid pair produce a stereophonic representation of lateral sounds to

match the forward image. Imaging will therefore remain correct for listeners who turn towards the L/LS or R/RS sectors. This produces a convincing spatial perspective.



3. OCT Front System

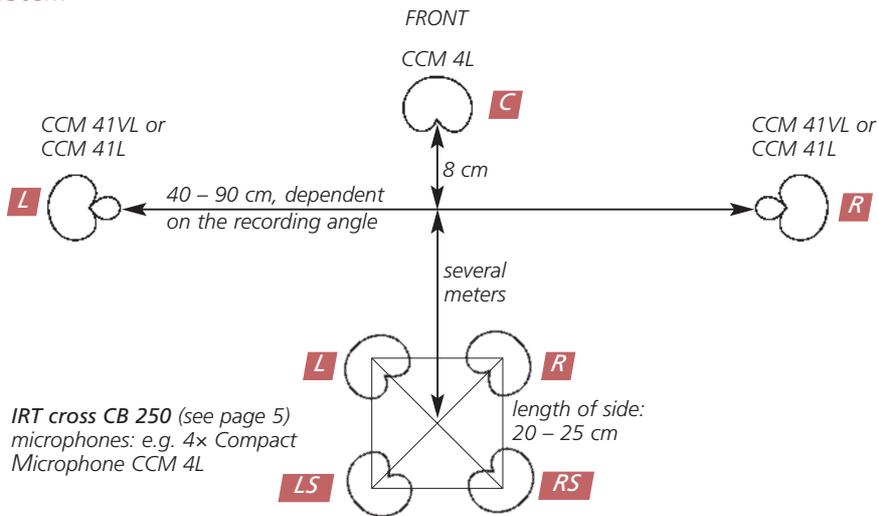
plus 4-channel ambience array

In the following two microphone arrangements, a group of four additional directional microphones is placed several meters behind the front OCT system. In each case, signals from the front two microphones of the additional group are blended into the main L and R front signals without

further processing. This helps to prevent dissociation between the front and rear images, while the separation between the arrays allows optimal placement of each for direct and for ambient pickup respectively.

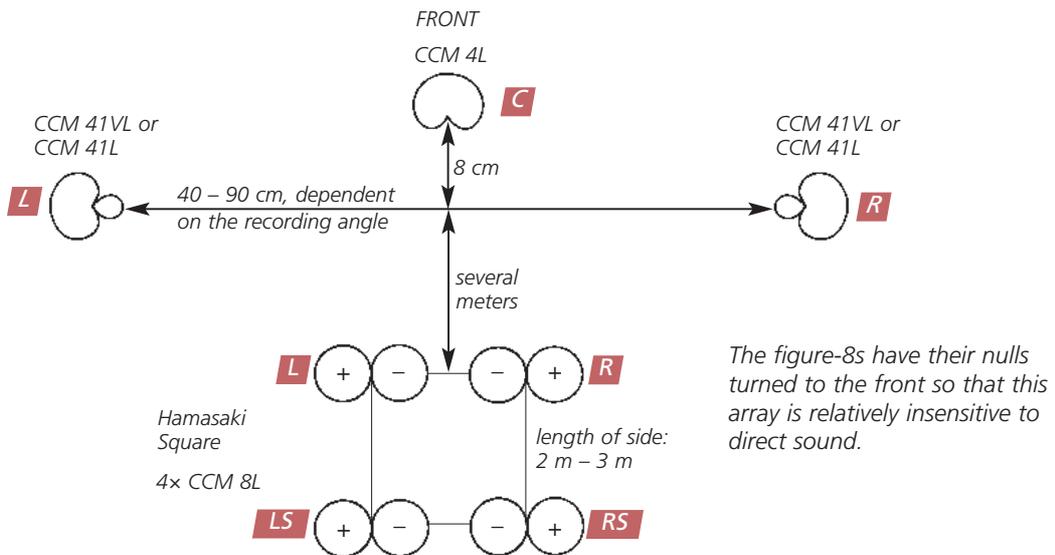
3.1 OCT Front System

+ IRT cross



3.2 OCT Front System

+ Hamasaki Square





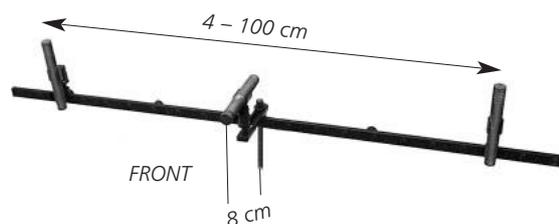
Mounting Bar for OCT (by Dr. Theile, IRT, Germany) or A/B MAB 1000

- for recording the three front channels for surround
- can also be used for A/B stereo
- recording angle: 90 - 160°
- distance between microphones: 4 – 100 cm
- can be expanded for five-channel OCT surround

This 1-meter-long mounting bar has 3/8" threaded attachments for three stand adapters or shock mounts (not included). Each side of the bar is engraved with markings every 2.5 cm so that distances of 5 cm, 10 cm, 15 cm, etc. can easily be set between the two outer microphones.

Recommended stand adapters or shock mounts: SG 20 or A 20 for CMC-series microphones; SGC or AC for CCM-series microphones.

Thread: 3/8"
Length: 1000 mm
Weight: 500 g



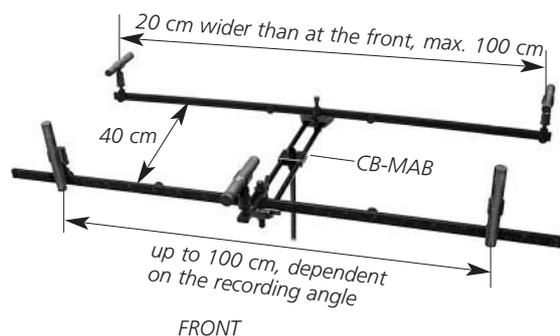
OCT bar
Center: cardioid (MK 4) aimed forward
2x supercardioid (MK 41V or CCM 41, aimed to the left and right)

OCT Surround Arrangement with 2 x MAB 1000 + CB-MAB

- for recording five-channel surround
- the simplest and most compact arrangement for surround recording with OCT

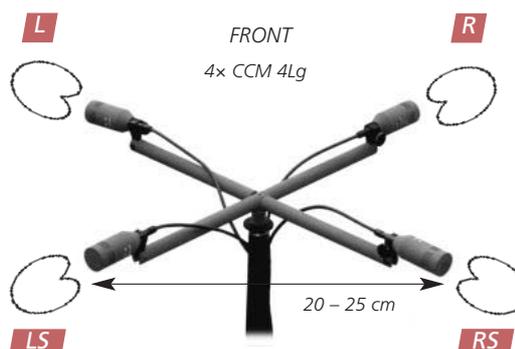
This OCT surround mounting bar set consists of two MAB 1000 mounting bars (one for the front-channel microphones and one for the rear-channel microphones) held 40 cm apart by a supporting beam with 3/8" threaded attachment. It is recommended that the rear-channel microphones be placed 20 cm farther apart than the front-channel microphones.

Accessory (included): KMAB 1000, robust carrying case for the two MAB 1000 mounting bars and the CB-MAB supporting beam



Mounting Bar for "IRT Cross" CB 250

The IRT microphone cross is an arrangement for ambient recording. Its primary characteristic is a transparent and spacious rendering of the acoustic environment. It is useful as a four-channel arrangement for room tone and can also be used for full surround (e.g. 360° cinema sound recording or for other playback arrangements in which the speakers are placed in the four corners of a square).

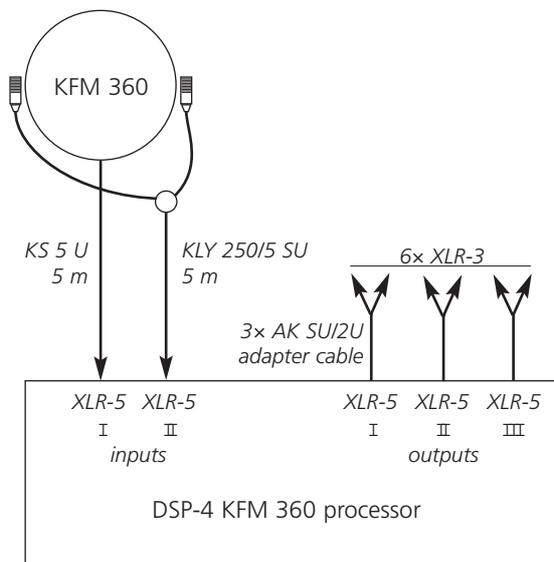




KFM 360
plus 2 CCM 8Lg Compact Microphones (figure-8)



DSP-4 KFM 360
processor unit with built-in A/D and
D/A converters



Surround Microphone System by Bruck with KFM 360 and DSP-4 KFM 360

- compact, highly effective arrangement
- extensive capability for post-processing in digital domain (many important parameters can be adjusted after the recording has been made)
- digital and analog inputs and outputs
- analog-style controls
- user preference settings can be stored and recalled

The complete surround microphone system consists of:

- KFM 360 sphere microphone with SGC-KFM suspension for two figure-8s
 - two figure-8 microphones (CCM 8L) and
 - the DSP-4 KFM 360 processor.
- These components are also available separately.

The central unit in this system is the KFM 360 sphere microphone. It uses two pressure transducers and can also be used for stereophonic recording. Its recording angle is ca. 120°, thus permitting closer miking than the pure stereo microphone KFM 6 (90°). The necessary high-frequency compensation for the pressure transducers is built into the processor unit.

Surround capability is achieved through the use of two figure-8 microphones, which can be attached next to the pressure transducers by an adjustable, detachable clamp system with bayonet-style connectors (SGC-KFM). The main axes of these two microphones should be aimed precisely forward.

The DSP-4 KFM 360 processor derives the four “corner” channels (L, R, LS, RS) from the microphone signals. A center channel signal is obtained from the two front signals. An additional output channel is provided which carries only the frequencies below 70 Hz. To avoid perceiving the presence of the rear loudspeakers one can lower the level of their signals, delay them and/or set an upper limit on their frequency response.

The front stereo image width is adjustable, and the directional patterns of the front-facing and rear-facing pairs of “virtual microphones” (see “Operating principle” below) can be chosen independently of one another.

The processor unit offers both analog and digital inputs for the microphone signals. In addition to providing gain, it offers high-frequency compensation for the pressure transducers as well as low-frequency compensation for the figure-8s.

As with M/S recording, matrixing can be performed during post-production. All matrixing in the DSP-4 is performed in the digital domain.

Operating principle:

The front and rear channels result from the sum (front) and difference (rear) of the omnidirectional and figure-8 microphones on each side respectively (see illustration on page 7). The four resulting “virtual microphones” which this process creates will seem to be aimed forward and

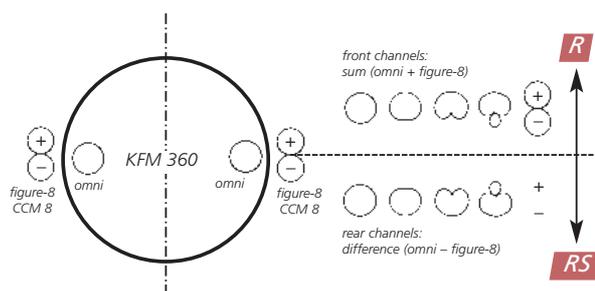


backward, as the figure-8s are. At higher frequencies, however, they will seem to be aimed further outward. Their directional pattern can be varied anywhere from omnidirectional to cardioid to figure-8; the pattern of the two rear-facing virtual microphones can be different from that of the two forward-facing ones. Altering the directional patterns alters the sound as well, in ways that are not possible with ordinary equalizers. This permits a flexible means of adapting to a recording situation – to the acoustic conditions in the recording space – and this can even be done during post-production if the unprocessed microphone signals are recorded.

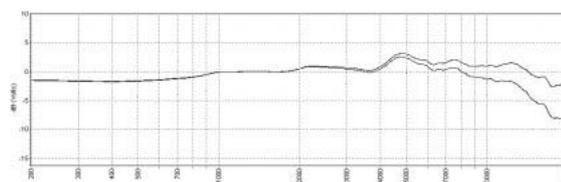
The signals from the four "virtual microphones" comprise a type of surround reproduction which is valid in its own terms but lacks the center channel and low-frequency channel of the standard 5.1 approach, so these additional facilities are provided by the digital signal processor.

The complete KFM surround set consists of:

- KFM 360 sphere microphone
- 1× KG ball-and-socket joint for mounting on a stand
- 2× CCM 8 L Compact Microphone (as a matched pair); included: K 5 LU adapter cable (Lemo / XLR-3), SGC miniature stand clamp with swivel and wooden box
- 2× SGC-KFM mounting clamp for CCM 8
- DSP-4 KFM processor with mains cable and wooden box
- 3× AK SU/2U adapter cable (XLR-5F to 2× XLR-3M) for connecting the processor's analog outputs
- 1× KS 5U stereo cable for connecting the KFM 360 (5 m long; advantage: only one cable is necessary for two channels)
- 1× KLY 250/5 SU, Y-cable, 5 m long, for connecting the two CCM 8 to the processor



Operating principle:
derivation of right R and right surround RS (rough diagram)



Frequency response curves showing the effect of the processor's built-in corrective equalization (before / after)

When a KFM 360 is ordered separately, it comes with 1× KG, 2× SGC-KFM, 1× AK SU/2U (XLR-5F to 2× XLR-3M)

Available separately:

- for the KS 5 U and KLY 250/5 SU cables: extension cables (e.g. KS 10 U, 10 m; KS 20 U, 20 m)
- For the KFM 360: KKFM wooden box and HKFM suspension device

Technical specifications:

KFM 360

- for 12 V / 8 mA, 48 V / 4 mA phantom powering
- 2 pressure transducers, built into the sphere
- diameter: 18 cm
- recording angle: ca. 120°
- weight: 800 g

CCM 8L

figure-8, for technical specifications: see catalog

Processor:

Analog inputs: 2× 2, XLR-5F, balanced, transformerless, 48 V phantom powering, 20 kOhm:

Analog Mic Gain: 10 dB +20 dB
Maximum input level: -4 dBu -14 dBu

Maximum SPL

KFM 360: 130 dB-SPL 120 dB-SPL

(The digital output level can be adjusted.)

Analog outputs: 3× 2, XLR-5M, balanced, transformerless, 100 Ohms, max. 6 dBV (2VRMS):

REC mode: outputs I, II: unprocessed input signals, output III: Monitor;

SURR. mode: output I: L, R; output II: C, LFE; output III: SL, SR

Maximum analog output level: +6 dBu

Digital inputs/outputs: channel assignment just as with the analog outputs; data format: 24 bit AES/EBU

Digital inputs: 2× 2, XLR-3F

Digital outputs: 3× 2, XLR-3M synchronization/sample rate:

- with internal synchronization: 44.1/48 kHz
- with external synchronization (through master inputs): 25 – 50 kHz

Elevation of the digital level: max 33 dB

Dynamic range:

- A/D converter: 98dB (peak)/CCIR
110dB (RMS)/unweighted
- D/A converter: 100dB (peak)/CCIR
113dB (RMS)/unweighted

Characteristic impedance of the cable connected to the digital outputs: 110 Ohms

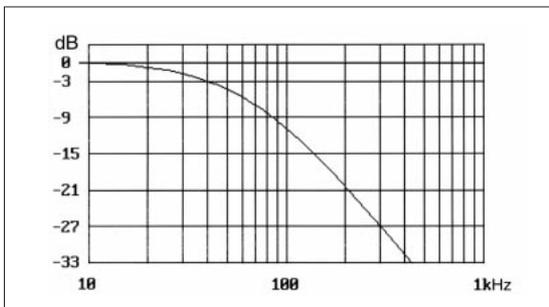
Mains voltage: switchable 110 – 120 V / 220 – 240 V

Power consumption: 15 VA

Fuse: 230 V: 100 mA slow blow; 110 V: 200 mA slow blow

Dimensions (W×H×D in cm): 22 × 9 × 24.6
about 8.7" × 3.5" × 9.7"

Weight: 2.7 kg (about 6 lbs)



Active Low-Pass Filter LP 40

- for filling in the bass region of the frequency response of directional microphones, especially when used with the OCT System (see pages 2 – 4)
- for phantom-powered microphones

All SCHOEPS omnidirectional microphones operate as pressure transducers. No directional microphone can equal their low-frequency response, which is essentially perfect down to as low as 16 Hz. But when using directional microphones, one can extend low-frequency response by adding in signals from omnidirectional microphones that have been low-pass filtered with a device such as the LP 40 (see illustration at left).

The LP 40 filter is designed to extend the frequency response of the SCHOEPS supercardioid with the lowest possible ripple. This makes it ideal for use, among other applications, with the OCT front surround system. SCHOEPS omnidirectional microphones with the LP 40 filter can also be used to extend the response of other pressure-gradient transducers such as cardioids.

Technical Specifications:

Filter: in-line low-pass, critical, 2nd order (12 dB/octave)

Cut-off frequency (-3 dB): 40 Hz

Output impedance: 40 Ohms @ 1 kHz with SCHOEPS

CCM Compact Microphones and CMC Standard

Colette microphone amplifiers

Gain: 0 dB

Maximum output voltage (depends on current drawn by microphone): ca. 1 V for 4 mA

Connectors: XLR-3

Dimensions: length: 94 mm, diameter: 20 mm

Surface finish: Nickel



ORTF



X/Y

Miniature Positioning Device for Stereo Recording M 100 C

Allows use of the following stereo recording methods:

- **ORTF** with two axially addressed capsules or microphones: MK 4 capsules + KC Active Cables or CCM 4 compact microphones
- **X/Y** as above or with radially addressed capsules or microphones such as the MK 4V or CCM 4V U/L

The device can be tilted forward or backward as needed.
Threads: 5/8"-27 NS with adapter for 3/8" and 1/2"