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NEW STUDIO CONDENSER MICROPHONES DESIGNED FOR THE DIGITAL AGE

Manfred Hibbing

Sennheiser electronic K.G., D-3002 Wedemark, West Germany

ABSTRACT

Digital recording may reveal inherent noise and distortions of condenser microphones previously concealed by the deficiencies of the analog recording technique. The causes for the imperfections of condenser microphones and measures to improve their characteristics will be discussed. The features of a new line of high-grade condenser microphones based on these improvements will be presented.

INTRODUCTION

In the past the quality of sound recordings was limited by the characteristics of the analog tape and record material, apart from losses induced by the copying and pressing procedures. Tape saturation, for instance, created additional harmonic and disharmonic distortion components, which affected the recording fidelity at high levels, whereas the linearity at low and medium levels was quite acceptable. However, the onset of these distortions was rather softly and extended to a wide level range which made it difficult to determine the threshold of audibility.

The distortion characteristics of the studio condenser microphones up to now were adequate to these properties of the analog recording equipment. Although exhibiting a high degree of technical sophistication these microphones show individual variations in the resolution of complex tonal structures, which only partly arise due to the specific frequency responses and directivity patterns. A remaining part is caused by non-linear effects inherent to the microphones.

These properties were mostly concealed by the distortions superimposed by the analog recording and playback processing. But the situation has changed essentially since the introduction of digital audio. The conversion of analog signals into digital information and vice versa is carried out very precisely, especially at high signal levels. Due to the linear quantization process the inherent distortions of digital recordings virtually decrease at increasing recording levels, which turns former distortion behavior upside down. This new reality, which is in total contrast to former experience with analog recording technique, contributes mostly to the fact that nowadays the specific distortion characteristics of the microphone may become obvious, whereas they have been masked previously by the more significant distortions of analog recording technique.

Another most significant feature of digital audio is the enlarged dynamic range, which has considerably reduced the noise floor of recent recordings. Unfortunately, due to this improvement the inherent noise of the microphones may become audible, as it is no longer covered up by the noise of the recording medium.

Sometimes it is argued that the ambient noise of the recording studios will be more significant than the inherent noise of the

microphones. Indeed, this is correct as far as low frequency disturbances are concerned which arise due to air conditioning and environmental vibrations. At higher frequencies, however, these interferences are effectively attenuated by the low pass characteristics of the air conditioning ducts and the inertia of the building mass, respectively. Thus only rumbling noise, if at all, but no hiss will be audible in a properly isolated studio. Consequently, if hiss becomes audible on digital recordings, it will be mainly generated by the microphones, provided the noise contribution of the microphone amplifiers is insignificant.
The preceding discussion suggests that the improvements of the digital recording era require also improved technical sophistication of studio condenser microphones. Minimal inherent noise combined with improved signal handling linearity are required besides the traditional design topics like balanced frequency responses and frequency independent directional characteristics.

IMPROVING THE LINEARITY OF CONDENSER MICROPHONES

PRACTICAL INVESTIGATIONS ON LINEARITY. Investigations on the linearity of condenser microphones customarily used in the recording studios were carried out by means of the difference frequency method using a twin tone signal (FIG. 1). This is a very reliable test method as the harmonic distortions of both loudspeakers which generate the test sounds seperately do not disturb the test result. Thus, difference frequency signals arising at the microphone output are doubtless generated by non-linearities of the microphone itself.
FIG. 2 shows the distortion characteristics of eight unidirectional studio condenser microphones which were stimulated by two sounds of 104 dB SPL (3 Pa). The frequency difference was fixed to

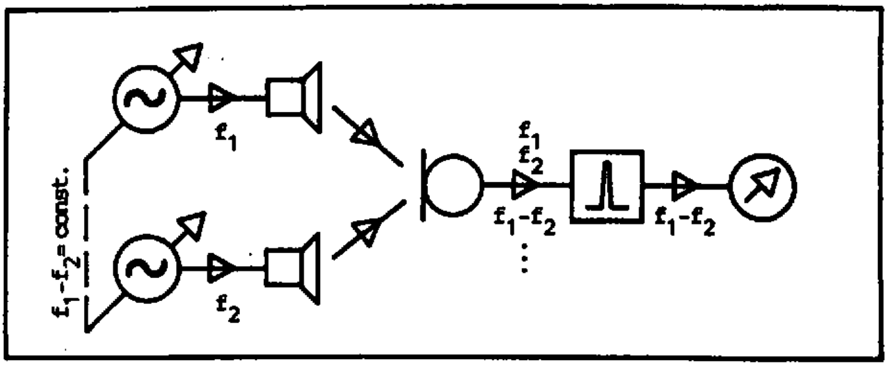

FIG. 1 - Difference Frequency Test Set-up
Two sounds of equal SPL are applied simultaneously and seperately to the microphone under test.
The main distortion component arising at the difference frequency is selected by a narrow band filter.

70 Hz while the twin tone signal was swept through the upper audio range. The curves show that unwanted difference frequency signals of considerable levels were generated by all examined microphones. Although the curves are shaped rather individually there is a general tendency to increasing distortion levels at high frequencies. Distortion figures up to 1% and more arise.

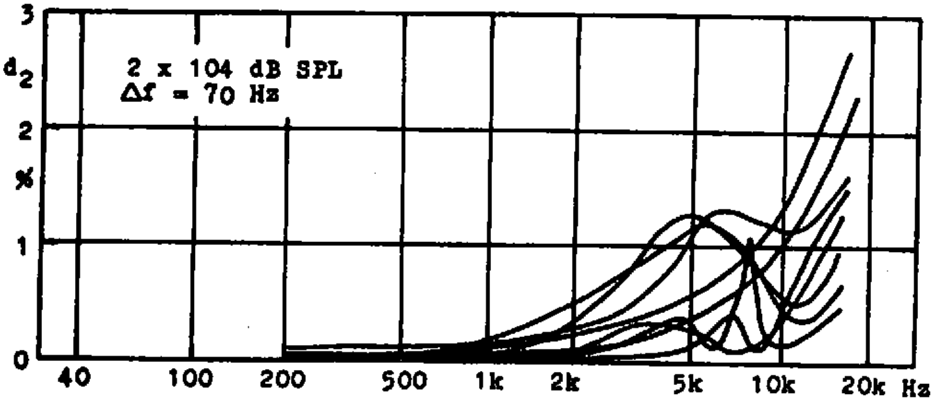

FIG. 2 - Difference Frequency Distortion vs. Frequency of Unidirectional Studio Condenser Microphones

The measurement results can be extended to higher signal levels simply by linear extrapolation. This means, for instance, that 10 times higher sound pressures will yield 10 times higher distortions, as long as clipping of the microphone circuit is prevented. Thus, two sounds of 124 dB SPL will cause more than 10 % distortions in the microphones. Sound pressure levels of this order are beyond the threshold of pain of human hearing but may arise at close-up miking. Despite of the fact that the audibility of distortions depends significantly on the tonal structure of the sound signals, distortion figures of this order will considerably affect the fidelity of the sound pick-up.
Of course, the sound pressure levels applied to the microphones have not been increased since the introduction of digital audio. Thus, improving the distortion characteristics of the microphones will less mean increasing the handling capability at extreme sond pressure levels, but rather improving the linearity in the total range of practical relevance.

CAUSE OF NONLINEARITY. FIG. 3 shows a simplified scetch of a capacitive transducer. Diaphragm and backplate form a capacitor, the capacity of which depends on the width of the air gap. From the acoustical point of view the air gap acts as a complex impedance. Unfortunately, this impedance is not constant but rather depends on the actual diaphragm excursion. Its value increases if the diaphragm is moved towards the backplate and decreases at the opposite movement. Thus the air gap impedance is varied by the motion of the diaphragm. This implies a parasitic rectifying effect superimposed to the flow of volume velocity through the transducer. The resulting non-linearity is mainly responsible for the distortion generated by condenser microphones.

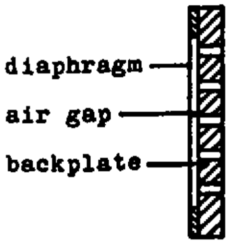

FIG. 3 - Scetch of a Conventional Capacitive Transducer

FIG. 4 - Scetch of a Symmetrical Push-pull Transducer

SOLVING THE LINEARITY PROBLEM. In order to improve the linearity of condenser microphones the most powerful principle is a push-pull design of the transducer as shown in FIG. 4. An additional plate equal to the backplate is positioned symmetrically in front of the diaphragm. Thus two air gaps are formed with equal acoustical impedances as long as the diaphragm is in its rest position. If the diaphragm is deflected by the sound signal, both air gap impedances are deviated opposite to each other. The impedance of one side increases while the impedance of the other side decrea-

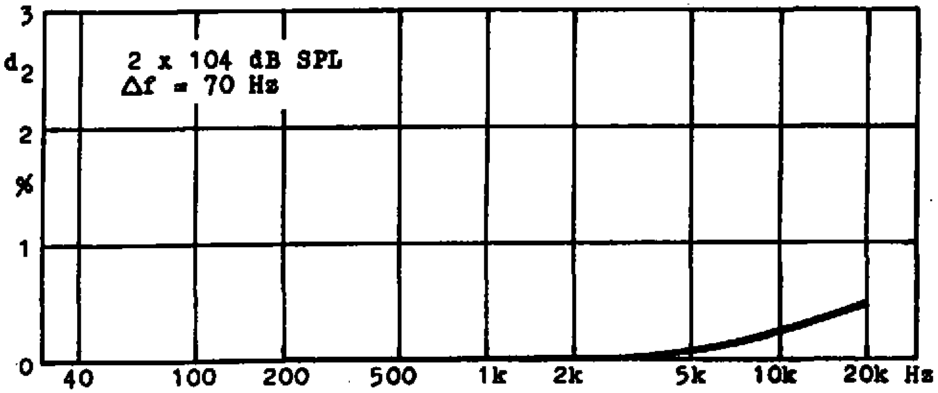

FIG. 5 - Maximum Difference Frequency Distortion of a New Condenser Microphone incorporating a SymmetricalPush-pull Transducer

see. The variation effects compensate each other regardless of the direction of the diaphragm motion, and the total air gap impedance remains constant. Thus the distortion of a capacitive transducer is fundamentally reduced by the push-pull design.
FIG. 5 shows the practical improvement resulting from this technique. The curve represents the maximum difference frequency distortion of a new unidirectional condenser microphone incorporating a symmetrical push-pull transducer. The measurements were carried out on a large number of samples at the same conditions as before. By comparison of FIG. 5 and FIG. 2 the improvement on linearity due to the push-pull design becomes obvious.

IMPROVING THE NOISE PERFORMANCE OF CONDENSER MICROPHONES

NOISE SOURCES. The inherent noise of condenser microphones is caused partly by the random incidence of the air particles at the diaphragm due to their thermal movement. The laws of statistics imply that sound pressure signals at the diaphragm can be evaluated by a precision which improves linearily with the diameter of the diaphragm. Thus, larger diaphragms yield better noise performance than smaller ones.
Another contribution of noise is caused by frictional effects in the resistive damping elements of the transducer. The noise generation of acoustical resistors is based on the same principles as the noise caused by electrical resistors. Thus high acoustical damping implies more noise than low damping.
Thirdly, noise is added by the electrical circuit of the microphone. This noise contribution depends on the sensitivity of the transducer. Obviously, high transducer sensitivity will reduce the influence of circuit noise. The inherent noise of the circuit itself depends on the operation principle and on the technical quality of the electrical devices.

NOISE REDUCTION. The previous discussions suggest measures to improve the noise performance of condenser microphones. Firstly, the diaphragm diameter should not be too small. However, the larger the diameter the more the directivity at high frequencies will be affected. Thus a compromise has to be made. A transducer diameter of about 25 mm will reveal pleasing features in both respects.
A further measure is the reduction of the resistive damping of the transducer to a technically convenient minimum. In most directional condenser microphones a high amount of resistive damping is used to obtain a flat frequency response of the transducer itself and thus keeping the electrical circuit of the microphone rather simple. The drawbacks, however, are reduced sensitivity and increased noise.
A moderate resistive damping of the transducer, however, will be a more appropriate measure to improve the noise performance of condenser microphones. As this design leads to a frequency response of the transducer which is no longer flat, equalization has to be applied by electrical means in order to get a flat overall frequency response of the complete microphone. This design technique demands a more sophisticated elctrical circuit but serves with

outstanding noise performance.
A further measure to improve the noise characteristics of condenser microphones is the selection of a proper circuit design. It is well known that condenser microphones can be operated alternatively in the audio frequency (AF-) domain or in the high frequency (RF-) domain. It has turned out that the noise performance of the more sophisticated RF-design is superior to that of the AF-design. This is mainly a matter of the electrical impedance of the transducer, which is strongly different in both designs.
Electrically the transducer is acting as a pure capacitance. Its impedance decreases as the frequency increases. Thus the transducer impedance is low in a RF-circuit but high in an AF-circuit. Moreover, in a RF-circuit the electrical impedance of the transducer is constant due to the fixed frequency of the RF-oscillator, whereas in an AF-design it depends on the actual audio frequency, yielding very high values especially at low frequencies. Thus resistors of extremely high values are needed at the input of the microphone circuit to prevent electrical loading of the transducer. Unfortunately these resistors generate additional noise.
The RF-circuit, however, features a very low output impedance comparable to dynamic microphones. The output signal can thus be applied directly to normal bipolar transistors which enable excellent noise performance.

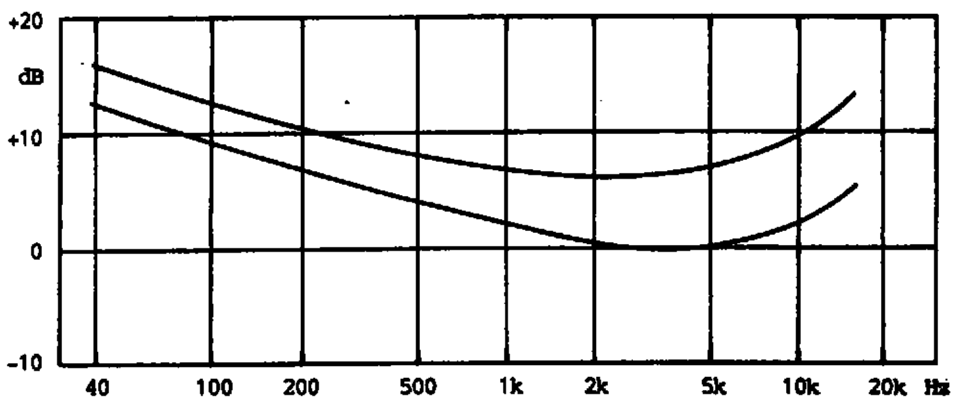

FIG. 6 - Third Octave Band Analysis of the Noise of Condenser Microphones refered to the Threshold of Hearing

FIG. 6 shows the practical improvements resulting from a low noise design based on the preceeding discussions. The upper curve shows the third octave band analysis of the noise created by a state of the art AF-condenser microphone. The vertical scale represents the equivalent SPL corresponding to the threshold of hearing. The lower curve shows the inherent noise of the new design which is reduced by about 6 dB at medium and high frequencies.

NEW CONDENSER MICROPHONES

A NEW LINE OF STUDIO CONDENSER MICROPHONES FEATURING IMPROVED NOISE AND DISTORTION CHARACTERISTICS

Based on these design details a new generation of RF-condenser microphones has been developed to meet the requirements of digital audio.

The microphone line comprises three types with different directional characteristics. The microphones are aligned to form a microphone family with identical technical data apart from differences dictated by the individual directional characteristics.

The MKH 20 is a pressure microphone with omnidirectional characteristics. The MKH 30 is a pure pressure-gradient microphone with a highly symmetrical bidirectional pattern which is enabled due to the unique symmetry of the push-pull transducer. The MKH 40 operates as a combined pressure and pressure-gradient microphone with an unidirectional cardioid pattern.

All microphones are phantom powered by 48 V and 2 mA. The outputs are transformerless floated.

The nominal sensitivity is 25 mV/Pa and can be attenuated to 8 mV/Pa (-10 dB) by an incorporated switch. The high sensitivity will significantly reduce line interference problems as well as influence of microphone amplifier noise. Due to the fact that the sensitivities of all microphone types are equal, combination or interchange of the different microphone types can be handled easily.

The maximum sound pressure level is 134 dB at nominal sensitivity and 142 dB at reduced sensitivity.

The equivalent SPL of the microphones is ranging from 10 to 12 dBA corresponding to CCIR-weighted figures of 20 to 22 dB. These figures represent improvements of about 6 dB compared to the previous state of the art.

The new microphones exhibit flat frequency responses up to 20 kHz. The pressure microphone cuts off below 20 Hz, whereas the directional types roll off below 40 Hz. The extended bass response of the directional microphones which is normally only feasible with larger diaphragm diameters is a benefit of the applied equalization technique. Additionally, this design avoids the poorer directivity performance associated with larger transducers at high frequencies.

The improved bass performance is most significant with respect to bidirectional microphones. The lack of bass response notoriously associated with these microphones may have contributed to the fact that they have not been used that much in the past.

Additional switching facilities are supplied to the new microphones to adjust them to varying recording situations. The directional microphones incorporate a switchable bass roll-off to compensate for the proximity effect at close-up miking.

A special feature of the omnidirectional microphone is a switchable diffuse field correction, which guarantees optimum results at both direct and diffuse sound field conditions. The normal switch position is recommended for a neutral pick-up at close-up miking. At larger recording distances where reverberations become significant the alternative switch position will be more suitable.

NEW CONDENSER MICROPHONES

The distinction between both recording situations arises due to the fact that omnidirectional microphones tend to attenuate off-axis sound signals at high frequencies. Thus diffuse sound signals with random incidence cause a lack of treble response which can be compensated by additional treble emphasis in the microphone circuit. Of course, frontally impinging sound is emphasized too by this measure, but this effect is negligible as long as the reverberant sound is dominant. Due to the switching facility the MKH 20 combines the characteristics of two types of microphones which otherwise are only seperately available, namely the direct field and the diffuse field pressure type.
Another useful feature is a supplementary ring which can be put upon the front end of the MKH 20. Due to the boundary effect the treble response is emphasized by 2 dB. Thus the treble response of the MKH 20 can be finely adjusted in 2 dB steps from 0 to +6 dB if both switch positions are combined with and without the ring.
The outstanding low noise performance of all microphone types enables a more detailed perception of low level components in acoustical signals. Thus a more refined replica of the filigree of sounds is rendered possible including a more detailed perception of subtle reverberations otherwise masked by the inherent microphone noise.
The improved analytic perceptibility of sound is also a consequence of the extraordinary low distortion characteristics of the new microphones, which prevent blending of sounds almost totally.

CONCLUSION

It has been demonstrated, that linearity and noise performance of studio condenser microphones can be further improved by means of symmetrically designed push-pull transducers combined with moderate acoustical damping. A line of three microphones with differing directional characteristics has been developed basing on these technical improvements. Standardized features and flat frequency responses ease combinations and interchange of the microphones. A world wide positive response has confirmed that the technical improvements are not only measurable but also clearly audible.

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Large Multi-Channel Wireless Microphone Systems:

Meeting the Need for 20 Channels in Theatre Applications

Hans Kuehn
Sennheiser electronic K.G., D-3002 Wedemark, West Germany

In theatres, film industry and TV-shows wireless microphones become more and more important. In the past, the use of 6 channels simultaneously seemed to be a limitation. Now, in major musical productions the number of channels has been increased to 30. What are the main problems with large multi-channel wireless systems and how could they be overcome?

1) The varying field strength is a wellknown problem. The field strength varies mainly by multi-path-propagation, by absorbtion and by shadowing. Considering only the multi-path-propagation effect, figure 1 shows a typical curve of the field strength at the receiving antenna. The variation of the field strength inside buildings with reflecting walls is 40 dB and more, but can be overcome by a good diversity system. Among the different diversity systems the so-called "true diversity" has proved to be the most efficient design. It is difficult to define an absolute value for the improvement caused by the diversity. Improvement can only be determined by statistical methods. Figure 2 shows the improvement by the true diversity system. By switching over in time, the drop outs of the actual field strength are less intense.
The improvement has a similar effect as an amplification of at least 25 dB of the wanted signal in case of emergency.
The diversity system is recommendable even if only one channel is in operation. Large multi-channel systems are only possible with diversity operation.

2) Non-linear effects in the RF-part of the receiver (or the antenna amplifier) cause problems caused by intermodulation. If in multi-channel operation several RF input signals exceed a certain level, the intermodulation products come up very quickly. We distinguish 3rd order, 5th order and higher order intermodulation products.

$$f_{im3} = 2\ f_1 - f_2$$

$$f_{im5} = 3\ f_1 - 2\ f_2$$

$$f_{im7} = 4\ f_1 - 3\ f_2$$

Figure 3 shows typical intermodulation products caused by two strong input signals. Looking at the intermodulation products, the frequency of a new channel should not be selected in a way that it is identical to the occuring intermodulation products of other signals. For a 6-channel system, for instance, 15 im3-products and 120 im5-products have to be taken into consideration. A computer programme will help to select suitable frequencies for multi-channel operation. This method, however, cannot be purely applied to large multi-channel systems. As shown in figure 4, the necessary RF-bandwidth would be unrealistically high with a 15 channel system, where only im3-products are being considered. Other additional design methods must be applied. If an RF bandwidth of 40 MHz is available for a 24 channel system, the bandwidth can be divided into 4 subgroups of 10 MHz, and

the 24 channels can also be divided into 4 subgroups of 6 channels. The subgroups can be separated from each other by highly selective RF-filters at the front end of the receivers (or the antenna amplifiers).
Figure 5 shows the filter curve of a three stage helical filter at the front end of a modern VHF highband receiver. Strong input signals 5 MHz either side of the receiving frequency are attenuated by at least 20 dB. In this way, they are normally reduced to an acceptable level. The subgroups then become independant from each other. Within the subgroups, however, the channel frequencies should be chosen with regard to im3-products and im5-products.
Which levels are critical considering im-products in the receivers? The manufacturer's technical data will give some hints. An intermodulation suppression of 60 dB means that in practice intermodulation products begin to come up at input levels of approximately 1 mV. Modern multi-channel receivers, as shown in figure 6, feature an intermodulation suppression of 80 dB. If the channel frequencies within one subgroup are selected with regard to im3, the critical input levels can be increased by appx. 10 dB. If, additionally, im5 is taken into consideration, another 5 dB are acceptable.
To prevent the receivers from getting unacceptably high input levels, the receiving antenna must be installed at a minimum distance to the transmitters.
Figure 7 shows what happens, if a 30 mW body-pack transmitter ir the 200 MHz range comes close to a half wave dipole. The receiving antenna should be positioned at a minimum distance of 6 m to the next transmitter. This condition is of high importance for good operation of large multi-channel systems. It goes without saying that the antenna amplifiers must also be of high quality, and should just compensate the cable losses and the distribution losses caused by the antenna splitters.

3) Receivers contain one or two local oscillators (single conversion or double conversion). A small part of the oscillator energy is radiated via the antenna or via the housing. Although this energy is extremely small (F.C.C. regulation), it is not considered negligible in designing large multi-channel systems. As the housings of all receivers within one system are connected to each other, this radiation can cause interferences, because this energy will find a bypass to the input filters. When calculating large multi-channel systems, the computer must also be fed with the spurious emission frequencies of the receivers. In order to get more safety, modern receivers have double screening: Inside the all-metal housing, hermetically sealed metal boxes contain the complete RF-circuitry (figure 8). By this method, the spurious emission is reduced by a further 20 dB below the F.C.C.-values.

4) Apart from the wanted radiation, transmitters also radiate some spurious emission. A VHF highband transmitter contains a 25 MHz oscillator. The 200 MHz carrier is generated by multiplying the 25 MHz frequency. During the multiplying process, not only the wanted 200 MHz carrier but also 200 MHZ ± 25 MHz is generated. For large multi-channel systems these spurious frequencies are not negligeable. To get more safety, the spurious emmission should be reduced by a further 30 dB below the value which is allowed by the F.C.C. Otherwise, the computer programme has to take into consideration also these spurious frequencies.

5) Intermodulation products are not only generated in receivers. Transmitters also have antennas which tend to pick up other signals. When these signals pass in a reverse way across the output filter of the transmitter, they are fed to a

non-linear component: the output stage transistor. In this way, transmitters can generate intermodulation products by themselves.
Figure 9 shows the intermodulation products of two hand-held transmitters (30 mW; 200 MHZ; 200.4 MHz) used at a distance of 1 m from each other. This shows that a good computer-aided frequency selection is of importance even if the receiver specifications meet the highest demands. A highly selective output stage is necessary to separate the subgoups from each other. With body-worn transmitters the problem becomes less critical. The human body absorbes appx. 10 dB of the transmitter energy. On the other hand, the transmitter working as a receiver is also handicapped by an antenna being close to the human body. Figure 10 is valid for body-worn transmitters and shows the transmitter intermodulation products of 2 transmitters (30 mW; 200 MHz; 200.4 MHz) when the distance is varied. This figure shows that actors with body-worn transmitters can come rather close to each other without significant problems of transmitter intermodulation products. The situation changes dramatically, if several transmitters in operation are put side by side on a desk. This should be avoided.

6) For large multi-channel systems the problem arises to actually have a complete overview about all transmitters and receivers. The most elegant way is to use a computer display (figure 11) monitoring all important functions of each channel.

7) External disturbing sources, like TV transmitters, taxi services, police services, digital equipment, etc., have of course also to be taken into consideration. Fortunately, the screening effect of buildings is rather high (30 - 40 dB for VHF highband carriers). For indoor application, this effect helps to keep rather strong outside signals at a low level inside. A significant problem sometimes occurs with some badly screened digital equipment working in the same room. These wide-band disturbing sources are able to interfere all channels. The only solution for this problem is replacement of the badly screened equipment by a better one.

8) To sum up, it must be said that large multi-channel systems demand excellent planning and, especially in the initial phase, good technical support. Observing all the above mentioned items, the perfect operation of a large multi-channel system can be garanteed, even under difficult conditions.

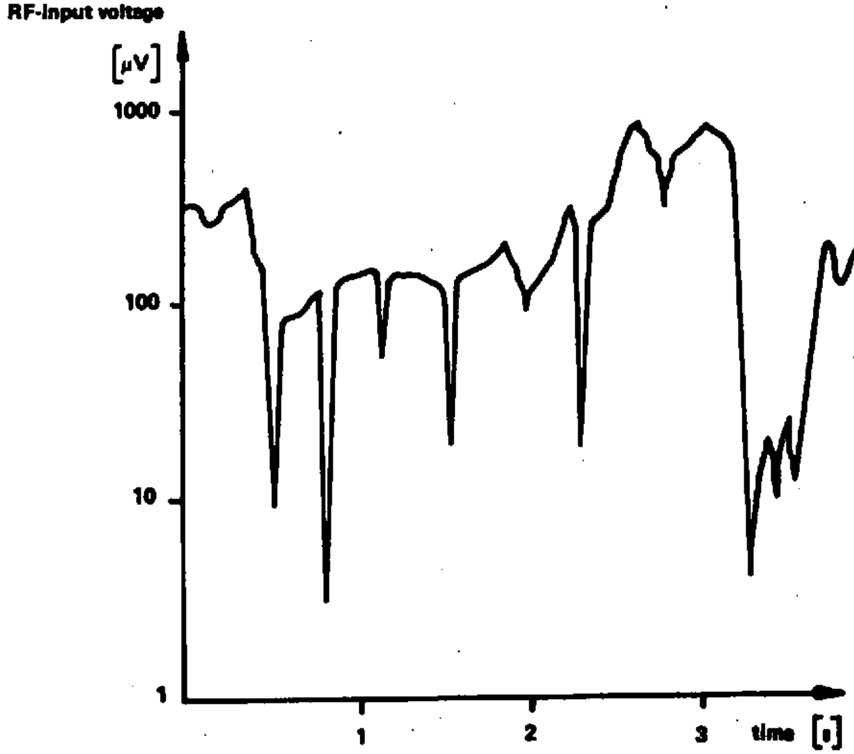

Fig. 1: Antenna input voltage of a receiver when moving the transmitter.

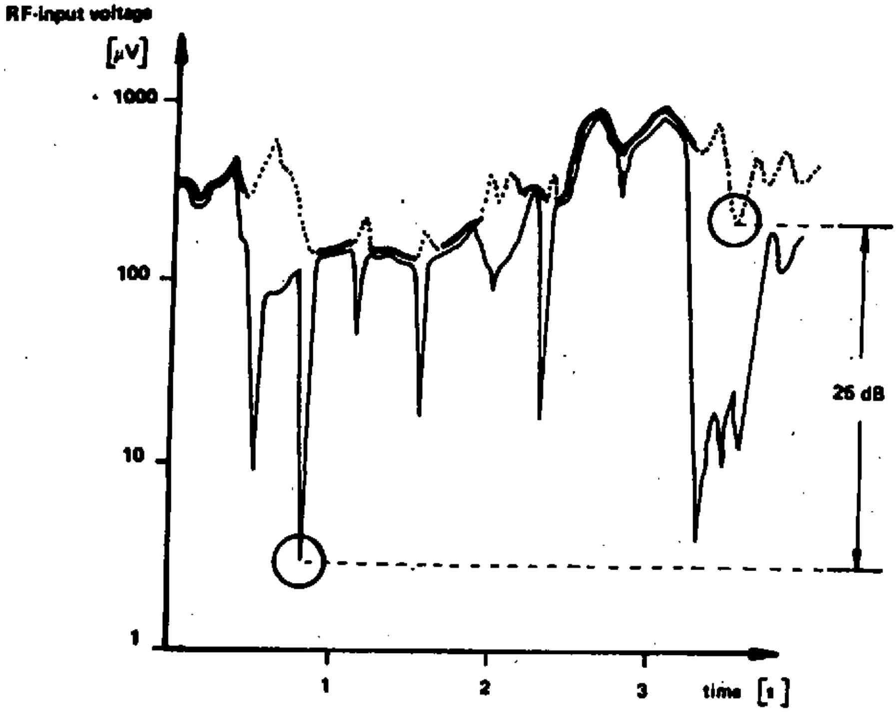

Fig. 2: Effect of switch-over diversity operation
RF-level at antenna 1 ————
RF-level at antenna 2 ....

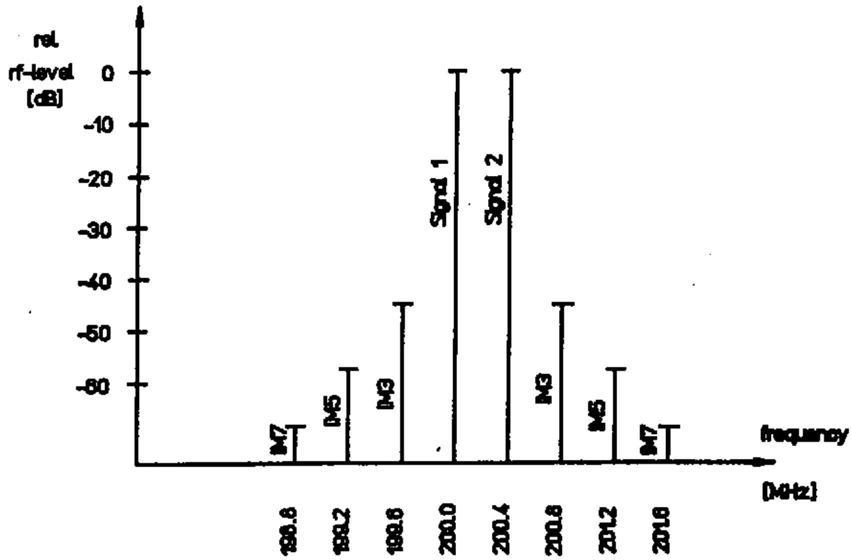

Fig.3 Intermodulation products in a receiver caused by two strong input signals.

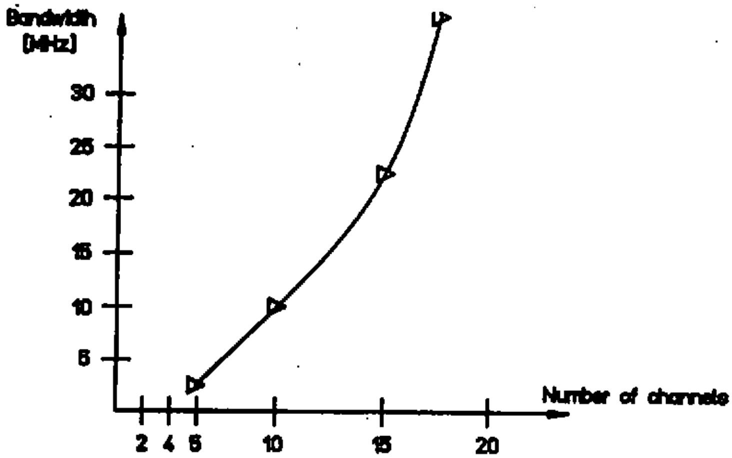

Fig.4 Necessary bandwidth of a multichannel system if only IM3 products are taken into consideration.

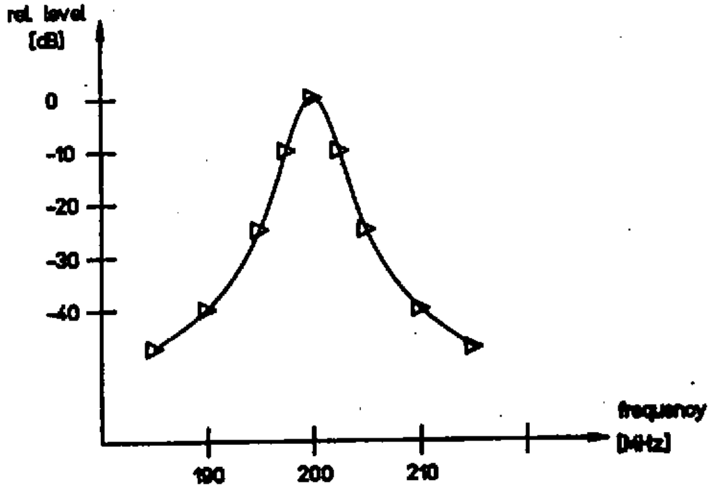

Fig. 5 Selection curve of a 3-stage helical filter at the front end of a receiver.

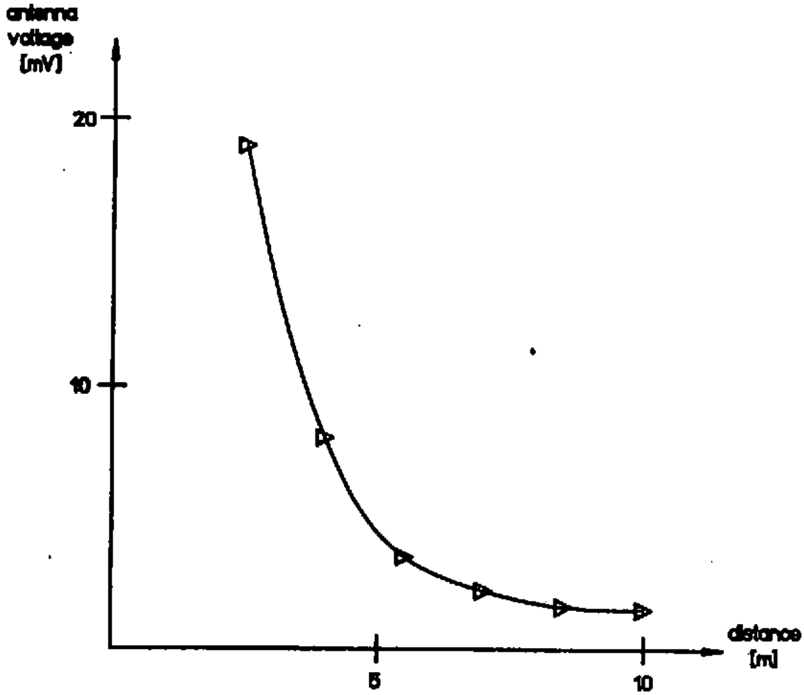

Fig.7 Antenna voltage of a $\lambda/2$-dipole if a body worn transmitter comes close to it (30mW, 200MHz).

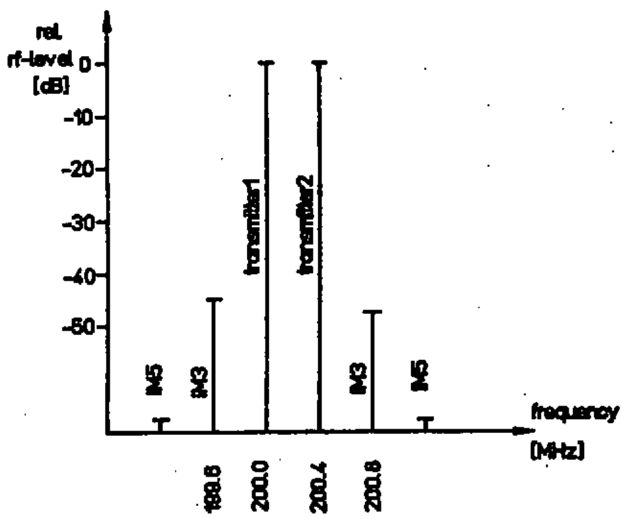

Fig.9 Intermodulation products generated by 2 transmitters (handheld, 30mW, 200MHz, 200.4MHz, distance 1m).

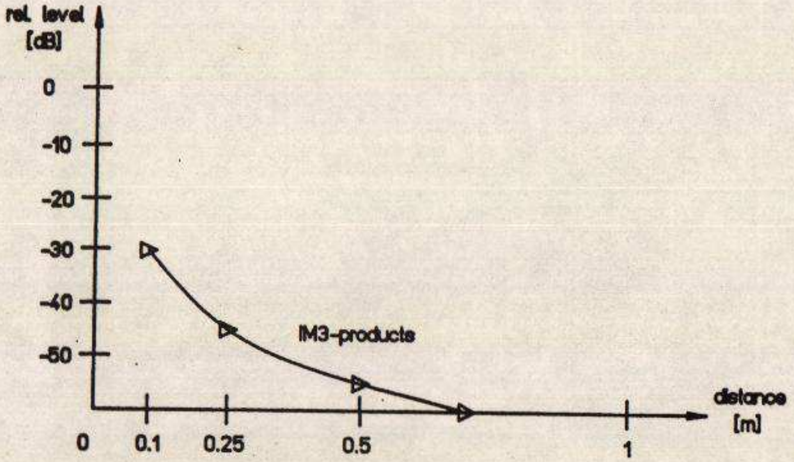

Fig.10 IM-3 products generated by two body-worn transmitters when approaching to eachother(30mW, 200MHz, 200.4MHz).

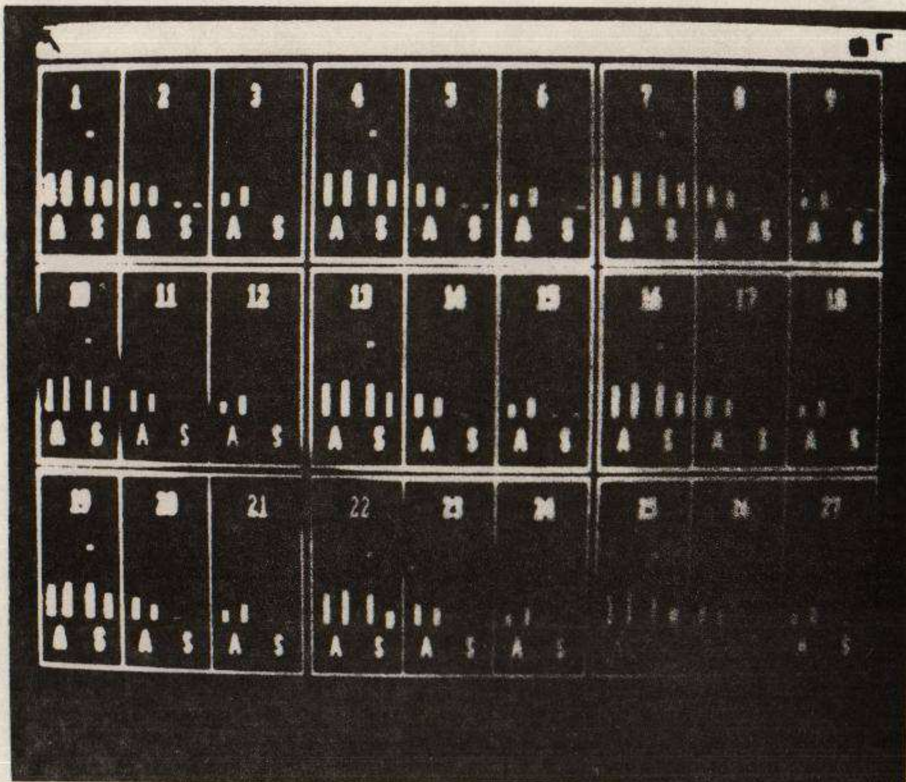

Fig. 11 Displaying all relevant functions of a 27 channel diversity system on a computer monitor

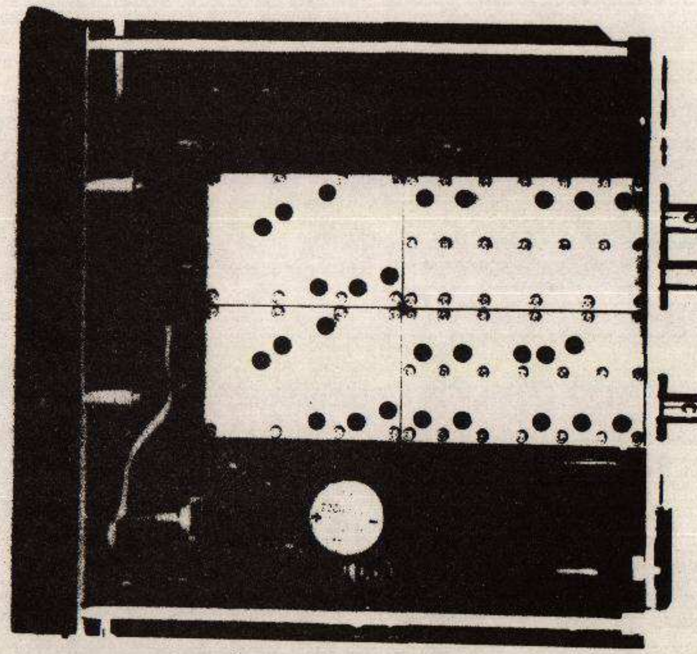

Fig. 8 Diversity receiver with double screening. The metal housing is removed. The metal boxes containing the complete rf-circuitry can be seen.

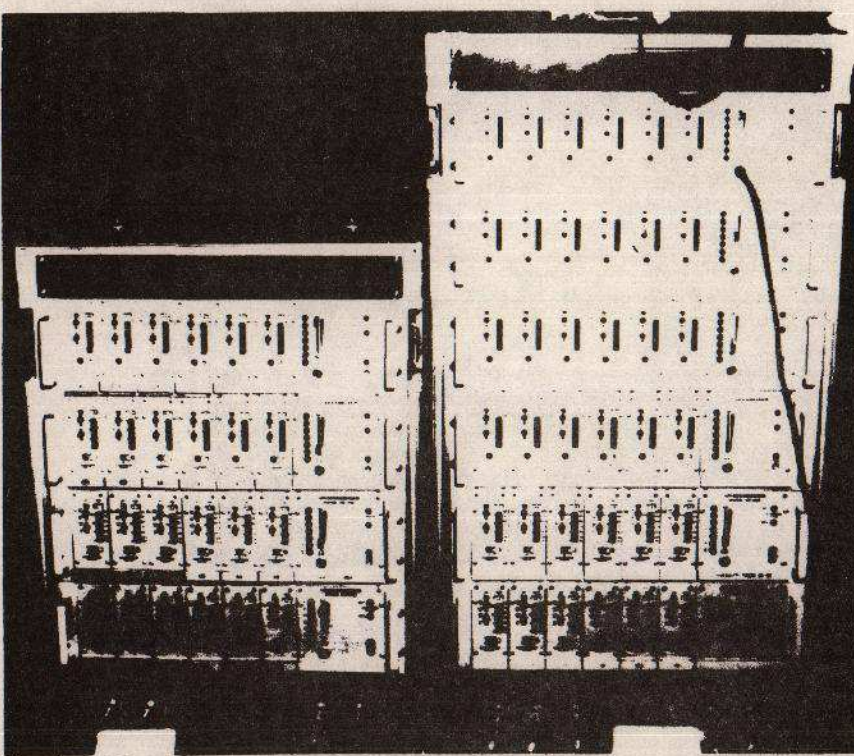

Fig. 6 30 channel diversity system