Main Microphone Techniques For The 3/2-Stereo-Standard

Abstract

There is no doubt that 3/2-stereo main microphone arrays are a field with a great deal of uncertainty. In order to dispel doubts in this field, this paper presents five 3/2-stereo main microphone techniques and compares them by means of a listening test; three of them are known and two of them have been designed by the authors.

Introduction

Establishing a new sound-format usually involves a considerably higher amount of theoretical and technical work, and the ensuing higher production costs are not offset by higher sales figures because the new format must catch on approval before profits can be made. Why then should a new format be introduced? Is 2-channel-stereo not the nonpareil of sound transmission? Has it not met the expectations of consumers, professionals and hi-fi-enthusiasts for forty years? Why transmit sounds differently, not from two sources right and left as before - after all, we all have just two ears, a right one and a left one?

2-channel-stereo has never been free of shortcomings. From the time it was introduced a great number of critics have demanded an improvement of spatial impression either by a different distribution of the two channels (e.g. center + room) [1, 2, 3] or by more than two channels [4, 5, 6]. There are advantages, to be sure, but there are two weaknesses, above all, which have always called for improvements:

First of all there is the tiresome dependence of the listening location and secondly, the fact, that a recording room can only be reproduced within 60° between the loudspeakers. The remaining 300° are left to the reflections of the monitor room. 2-channal-stereo will never be able to convey the feeling of sitting in a concert hall..

It is at this point that the 3/2-stereo-standard comes into play: The additional center channel is intended to achieve a more stable localization with central sounds in connection with a widening of the listening angle. The surround-channels are meant to produce surrounding sound which conveys to the listener the impression of being in the center of the action, be it a concert hall or a movie scene.

So for transmission of music and ambience the chance presents itself to transmit musical instruments or other sounds which are to be positioned in the center via the center channel. Particularly the tone quality and localization stability of these center voices which, in case of 2-channel-stereo appear at the greatest possible distance from the loudspeakers, can be improved by means of the additional center channel. But the entire sound pattern too, which in case of 2-channel-stereo at the slightest change of the listening position skips into one of the two channels, becomes more stable [7].

It is here, of course, that the question must be asked what kinds of center and surround signals are required for significant improvements. Unskillfully placed center microphones or inadequate surround signals (e.g. too loud: a frequent mistake) can easily cause damage, especially with regard to localization.

We can split the question into two parts: The surround channels and the center channel.

The two surround channels are supposed to generate a surrounding sound field which gives the listener the impression of being present at the place of recording. The influence of room acoustics in the monitor room can have a disruptive effect, and this is why a monitor room which is as dry as possible is recommended for 3/2-stereo. As a matter of principle a distinction has to be made between music and ambience recording. As far as music is concerned the focus is on producing the best possible spatial impression. At the same time, consciously localization of the surround speakers is to be avoided (because in good concert halls we usually don't locate anything except the musicians and the coughing neighbor) and the listener must always, and in every listening position, have the impression that the musicians are sitting in front of him. For the reproduction of ambience however, localization from behind can be desirable sometimes. Even continuous localization of any direction might be desirable. But with only the two surround channels, especially at the sides, this can only be, for psycho-acoustical reasons, a rough approximation [8].

The center channel:

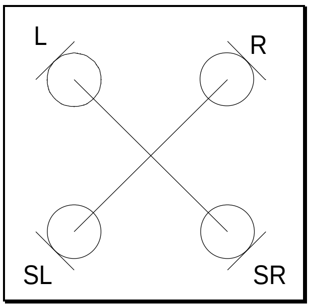
In film sound, as a rule, the dialogue is positioned in the center channel. But what about classical music? Can a soloist, as in films, be simply put in the center? Or, for that matter, a group of instruments? Or does a 3-channel main microphone arrangement exist? And if so, does the generation of phantom sound sources still function perfectly?

A first approach was simply to avoid the problem by not making use of the center channel. The point of view here was as follows: Three microphones could be seen as three stereo pairs Left-Center, Center-Right and Left-Right, and each of them would generate phantom images that usually don't match, which leads to a diminished localization quality. In fact this is true with certain main microphones arrangements, but we will see that we can steer clear of this problem by a skillful arrangement of the three microphones. Among these surround main microphone arrays without a center mike is the "Surround-Atmo-Mikrofon" (Surround-Ambience-Microphone) of the IRT [9] as well as Jerry Bruck's Surround Sphere [10]. They will be dealt with in greater detail later.

Frequently 3/2-stereo-recordings are also produced polymicrophonically. Instead of one main microphone separate microphones are set up for each single instrument or group of instruments, and these are then placed in panorama using the panpots of a mixer [11, 12]. This standard production procedure for popular music is often used for classical music productions as well for economical reasons or for the special direct sound. But still main microphone techniques are of much interest and there is no doubt that 3/2-stereo main microphone arrays are a field with a great deal of uncertainty. In order to dispel doubts in this field, five 3/2-stereo main microphone techniques will be presented and compared here by means of a listening test; three of them are known and two of them have been designed by the authors. An assortment of 88 participants, both laymen and experts were asked to judge the quality of the recording techniques. A statistical analysis of the answers enables significant statements about the quality and the characteristic properties of the recording methods.

Familiar Methods

The "Surround-Atmo-Mikrofon" (Surround-Ambience-Microphone) consists of a system of four cardioids which are arranged crosswised without a center microphone (see ill. 2). It is designed for ambience recording in a concert hall as well as in the field.

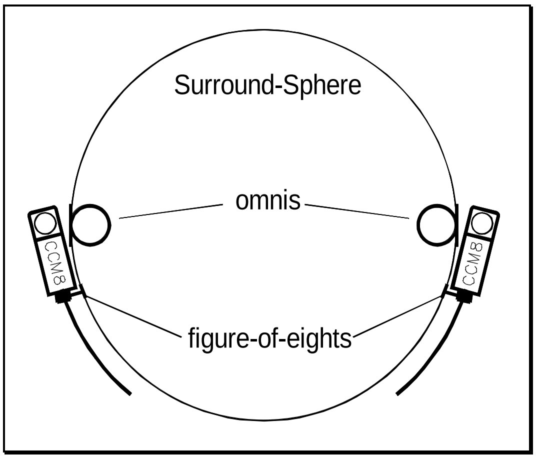


Ill.2: Surround-Atmo-Mikrofon (Surround-Ambience-Microphone) of the IRT

The underlying theory is explained by the developer G. Theile as follows:

"The symmetric design guarantees equivalent correlation for signal pairs Left - Right, Right - Rear Right, Rear Right - Rear Left, Rear Left- Left. The angle between the microphone axes is 90°. In order to obtain approximately the properties of an ORTF-microphone arrangement (the angle between microphones here is 110°) for the front and rear stereo sound stage, the distance between two capsules must be extended from 17 cm to 21 cm. [ ] If, however, one refers to the well-known Williams-curves and an recording angle of ± 45°, the resulting distance will be about 25 cm. In practice this difference is probably unimportant" [9].

The surround sphere presented by Jerry Bruck [10] is based on the Schoeps "Sphere" microphone KFM 6 [13]. Two figure-of-eights are added to the omnis built in the sphere, both in a forward direction. Thus they form on each side a MS-arrangement; the S-system should be properly called D, of course, because it provides for **d**epth, not **s**ide (ill. 3).



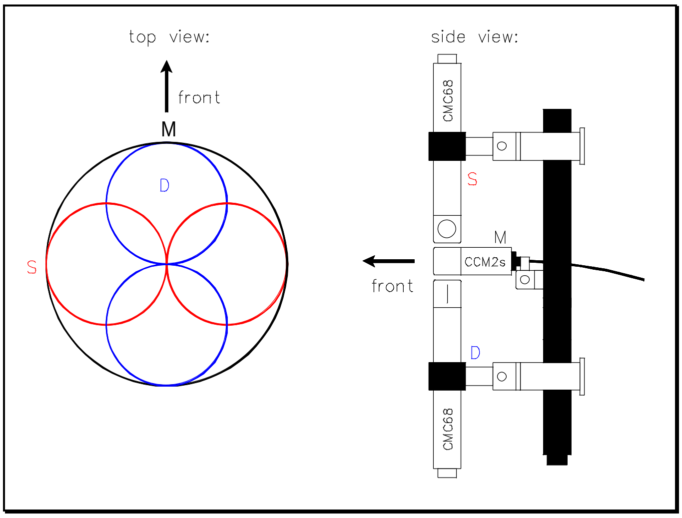
Ill. 3: Surround-Sphere by J. Bruck

A considerably older approach, M. Gerzon's Ambisonics systems was extended to include the 3/2-stereo-standard in 1992 [14]. Ambisonic is a psycho-acoustically optimized system, which does not transmit sounds by means of discrete channels, but, to put it simply, transmits the direction vectors of the sounds. This requires a coder at the sender and a decoder at the consumer, and provides for a relatively free loudspeaker arrangement. The standard arrangement (ill. 1) is only one of many possibilities which can be set at the user's decoder (a detailed explanation would certainly go beyond the scope of this article; it is given in [14] or in [15] in German).

Because, in addition to the MPEG or AC-3 data reduction decoders, special Ambisonic-decoders are needed, this system is no more likely to gain widespread acceptance than earlier Ambisonic versions where in some cases all the information was packed into the two available stereo-channels.

For the comparison of main microphone techniques which we are presenting here, a MSD called recording technique based on Ambisonic theory was used which is, contrary to Ambisonic, decoded in the recording studio and not in the living rooms. Thus the transmitted signals are fully compatible with the 3/2-stereo-standard and the consumer does not need an Ambisonic decoder. On the other hand, the consumer now has no longer the choice between different speaker arrangements, he has to place the speakers according to the ITU-standard and if he doesn't (the normal case, have a look on your own 2-channel stereo system...) the result is not optimal. Because Ambisonic systems are always based on optimally adapted speaker feeds, we called the Ambisonic-based 3/2-stereo-recording-system "MSD" (Mid-Side-Depth).

MSD is recorded with an omni and two figure-of-eights according to ill. 4. It is then split in a high frequency and a low frequency band and matrixed according to the Ambisonic-coefficients [14] in the five 3/2-stereo channels. The matrixing can be visualized in simplified form as follows: The addition of two figure-of-eights arranged crosswise invariably produces a figure-of-eight the direction of which is determined by the ratio and the sign of the two figure-of-eights. If five figure-of-eights are produced in such a way in different directions, each of them can be added up with the Mid-Signal (an omni) to produce five cardioid-shaped directional patterns in five different directions, e.g. the five 3/2-stereo signals Left, Center, Right, Left Surround and Right Surround. Of course suitable mixing ratios (mathematically speaking: the coefficients) have to be chosen. If the psycho-acoustically optimized Ambisonic coefficients as calculated by Gerzon are chosen, the result is MSD/Ambisonics. However, things are not that simple: The point of Ambisonic is, of course, that matrixing is effected separately in two frequency ranges with varying coefficients in order to meet the different localization mechanisms of the ear below 700 Hz ("Makita localization") and above ("energy localization"). (for Ambisonic insiders: The signals M, S, D correspond to the signals W, X, Y, of the soundfield-microphone.)



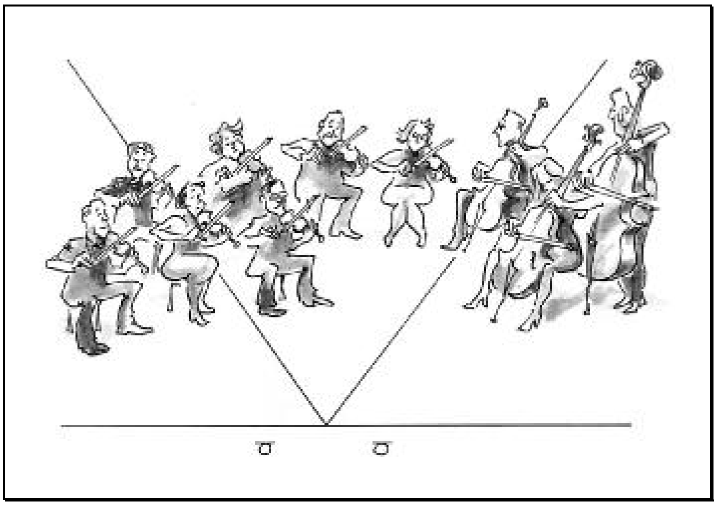
Ill. 4: MSD

The rating of this system in the listening test being poor, we will omit further details and turn to the next attempt to solve the "surround dilemma".

New methods

In many discussions about the subject of main microphone techniques for the 3/2-stereo-standard at last year's Annual Conference of German Sound Engineers and at various VDT-seminars there was a general agreement that a 3-channel main microphone for the 3 front channels of the 3/2-stereo-standard must be developed which could be complemented by ambience microphones or artificial reverberation for the surround channels. For ambience productions a five-channel microphone arrangement would be necessary.

A crucial quality of a 2-channel main microphone is its **recording angle**. The recording angle is specific for each main microphone and defines the field (symmetrical to the main axis) within which sounds are received in such a way that in the monitor room they will be localized between the two loudspeakers. An example: A sound source on the left edge of the recording angle is localized, when played back, at exactly the same place as another sound source, which is situated much further left (in the left loudspeaker). A further extension from the edge of the recording angle is not possible. Or in other words: Everything outside the recording angle is packed into one of the loudspeakers (ill. 5). The size of the recording angle is defined by the main microphones' diverse degrees of differences of intensity and phase delay.



Ill. 5 recording...

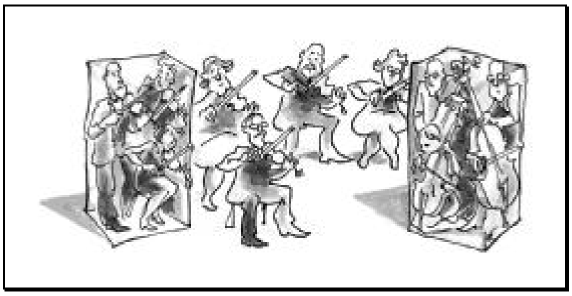
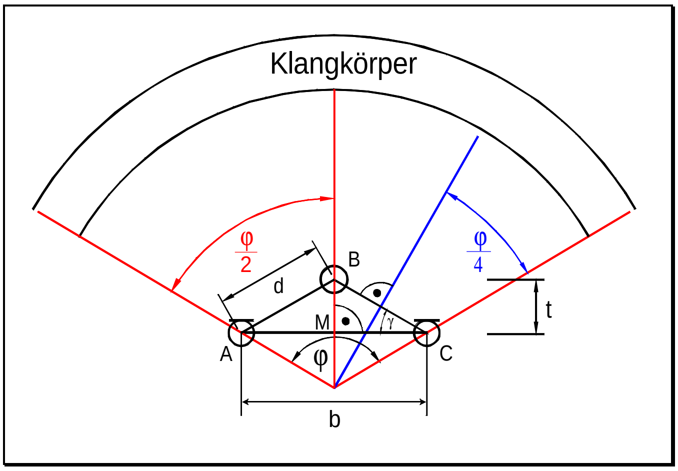


Illustration: Fabian Eckhardt

...and reproduction with recording angle chosen too small

A three-channel main microphone can be defined as a combination of two conventional 2-channel-stereo main microphones. Depending on the position of the sound source, two microphones (Left-Center or Center-Right) constitute a main microphone pair, whereas the phase delay respectively the intensity differences to the third microphone are so big, that the third signal is no longer used by the localization mechanisms of the brain. A sound source which is in the center of the recording angle of such a microphone arrangement must be localized on the extreme right of the stereo pair Left-Center and on the extreme left of the stereo pair Center-Right. The entire recording angle is thus divided into two parts of same size and each part is covered with one stereo-pair.

In order to make sure that there are no multiple localizations it is essential that there is always only one "active" microphone pair. The two recording angles must not overlap in the middle. An arrangement of three microphones in a line would not fulfill this requirement. The three microphones must be arranged in a triangle according to ill. 6, so that the mid-perpendicular of AB and BC are rotated by an angle of $\frac{\varphi}{4}$.



ill. 6: geometry of a 3-channal main microphone

This principle can be realized with omnis similar to AB-stereophony or with cardioids similar to ORTF-stereophony.

ABC

Among the merits of AB-stereophony, depth as well as the possibility to use omnis which ensure solely the best possible bass transmission. It is based on the assumption that in 2-channel stereo a phase delay of 36,7 µs between the channels causes the phantom sound source to move 1° off the main axis. Accordingly a sound is localized in a speaker if there is a phase delay of 1.1 ms. From this point the law of the first wave front applies which says that only the direction from which the first wave front arrives is relevant for the localization of a sound source [16]. As to 2-channel-stereophony the required distance d between two microphones for a recording angle of ω is defined as follows:

$$d = \frac{\Delta s}{\sin \frac{\varpi}{2}}$$

(Δs is the distance covered by sound in air in 1.1 ms: $\Delta s = c \cdot \Delta t = 340 \frac{m}{s} \cdot 1.1 ms \approx 37 cm$)

The distance d between A and B (and of course B and C) has to be adjusted so that its recording angle is $\frac{\varphi}{2}$:

$$d = \frac{37cm}{\sin\left(\frac{\varphi}{4}\right)}$$

To build up this microphone-array we need, in addition to the distance of every microphone pair, at least one of the distances b or t. Lets have a closer look at the triangle MCB:

First we realize $\gamma = \frac{\varphi}{4}$. Then it is quite easy to see:

$b = 2d \cdot \cos\left(\frac{\varphi}{4}\right)$ and

$$t = d \cdot \sin\left(\frac{\varphi}{4}\right) = 37cm$$

With these values a musical instrument situated in the center of the orchestra automatically appears on the right of the left microphone pair AB and on the left of the right microphone pair BC. So in the listening room it is localized in the center speaker.

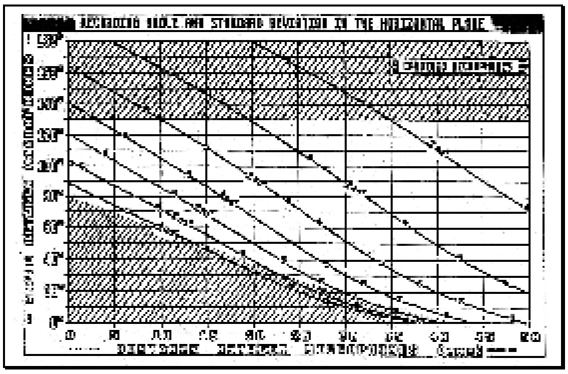
Here are some calculated values for given recording angles:

entire recording angle in °	***microphone distance d in cm***	***microphone distance b in cm***
100	87,5	158,5
120	74	128
140	64,5	105,5
160	57,5	88

table 1: some calculated values for ABC recordings

Mixed stereophony

A main microphone which produces differences of intensity *and* phase delay is makes use of the distance and the angle between cardioid microphones. The relationship between theses two parameters and the recording angle results from the Williams-curves (ill. 7) for cardioid microphones. Each curve represents a recording angle that can be achieved by different combinations of the angle between microphones (y-axis) and microphone distance (x-axis). Because of the symmetry Williams defines a recording angle of e.g. 160° as ± 80°.



Ill. 7: William's curves [17], [18]

Ideal Cardioid Arrangement for music recording "ICA-3"

The structure o the microphone arrangement shown above (ill. 6) can also be calculated for three cardioids. Here, in addition to the distances, the direction of the microphones is of great importance, too. (Of course, with omnis the direction is also of some importance, because for high frequencies omnis are not omni-directional at all. Every sound-engineer knows that it doesn't make sense to set omnis pointing towards the ceiling, just because it's more convenient. But for theoretical considerations, omnis are regarded to be omni-directional.)

Since the central microphone points to the center of the orchestra and thus points to the right edge of the recording area Left-Center and towards the left edge of Center-Right, the two outer microphones must point to the edges of the recording area, too. So in contrast to 2-channel-stereo, in 3/2-stereo the angle between microphones of the stereo pairs Left-Center and Center-Right is fixed. It's always identical with half of the entire recording angle. With the angle between microphones given, the distance d of the stereo-pairs can be easily obtained from the Williams curves or calculated by the formula:

$$d = \frac{-7.6075\left(1-\cos\left(\frac{\varphi}{2}\right)\right)^3 + 16.2605\left(1-\cos\left(\frac{\varphi}{2}\right)\right)^2 + 29.342\cos\left(\frac{\varphi}{2}\right) + 8.738}{\sin\left(\frac{\varphi}{4}\right)} \quad [17]$$

In this formula φ is the entire recording angle. An example: If the microphone-array should be situated right in front of the orchestra, φ is 180°. When looking up the required microphone distance in the curves by Michael Williams, you have to be careful: The recording angle for each of the two stereo-pairs in that case is 90°. So the appropriate curve in William's diagram is the ±45° one.

The distance b of the outer microphones and the height of the triangle t can be obtained from the formulae given above as with ABC.

One has to bear in mind that, in contrast to ABC, with ICA-3 the distance t is not constant, it depends on the value of a.

entire recording angle j in °	***microphone distance d in cm***	***microphone distance b in cm***	***height of the triangle t in cm***
100	69	126	29
120	53	92	27
140	41	68	24
160	32	49	21

table 2: some calculated values for ICA-3 recordings

The Surround Channels

In order to generate surround signals suitable for music productions it would seem best to use omni-directional microphones. The distance between these microphones should be rather big, so that two almost uncorrelated spatial signals are produced. The distance between the surround mikes and the main mike determines the relationship of diffuse and direct sound in the surround channels, and it should be rather big in order to avoid a localization of the surround speakers; here again one should bear in mind that the delay caused by the big distance might be perceived as an echo. This can be easily counteracted, however, by means of a digital delay of the front signals, so that the time difference between the direct and surround signals lies between 10 and 40 ms [19]. As a general rule: The bigger the audience the recording is intended for, the bigger should the time difference between front and surround signal be, for the reason that even the listener sitting next to a surround speaker must be reached first by the front signals.

In our test recordings, however, we used omnis as surround mikes in the described way only for ABC. ICA 3 was complemented by the "Surround Atmo Mikrofon" used as a ambience mike (with the same delay-adjustment as in ABC). The surround sphere and MSD produce the surround signals by way of matrixing of the front signals.

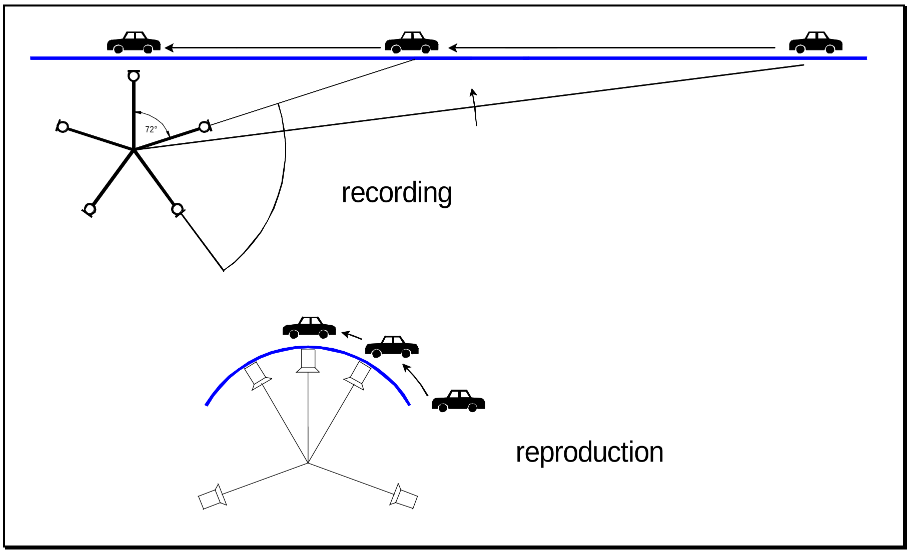
The risk of the badly irritating rear localisation can be further reduced by filtering the surround signals: There are areas on frequency axis, that are emphasised by the transfer function of the external ear for sounds arriving from behind [16]. This filtering is our only way to determine front and rear directions when we do not rotate the head. So de-emphasizing these areas is an easy way to reduce the risk of irritating rear localisation. This was realised for the methods ABC, ICA and Surround-Sphere with -9dB at 1,2 kHz and 12 kHz (The Dolby-Surround decoder tries something similar with a high-cut at 7 kHz in the surround channel, but in a psycho-acoustically sub-optimal way).

Ideal Cardioid Arrangement for ambience recording "ICA 5"

With ambience productions matters differ in some respects. With a five-channel ambience production it is not an orchestra of limited dimensions that is to be recorded. Each and every sound round the mike belongs to the ambience. This is why 360° must be presumed to be the standard recording angle. How can this angle be distributed effectively among the five stereo bases in the 3/2-stereo-standard?

The obvious solution (and the one used with the "Surround-Atmo-Mikrofon" of the IRT) would seem to divide the whole recording angle into five equal parts, resulting, with ICA-5, a recording angle of each basis of 360° / 5 = 72°.

In practice, however, this distribution has its drawbacks. Let's assume a car is moving across the setting from right to left in a straight line. One would the expect the noise caused by the car to appear in the right speaker, move via the center speaker to the left speaker and disappear there. This, however, is not the case with such an microphone arrangement. Acoustically, the car seems to move round a bend rather than in a straight line (ill. 8).



Ill.8: inadequate image distortion

If the distance from the mike set-up is very big, the car is - acoustically - in the stereo basis between the right front mike and the right rear one. So it is represented somewhere right of the listener. When it approaches it moves gradually forward until it appears in the right loudspeaker. After passing in front the same procedure is repeated on the left in inverse order. This is why a sound moving in a straight line describes a curve in reproduction.

An even worse distortion arises if the entire recording angle is distributed in such a way that the individual recording angles correspond to the loudspeaker arrangement of the reproduction standard.

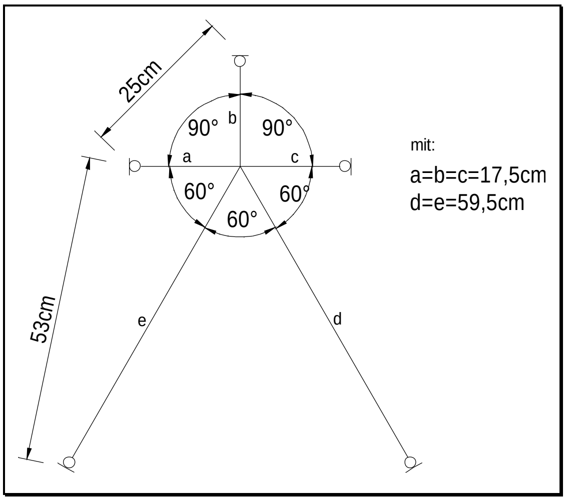
The following distribution has a more natural effect:

The front stereo bases left-center and center-right are each recorded at a recording angle of 90°. Because of that the whole front stage which is likely to be seen in the image is represented between the right and left speakers. The remaining 180° behind the mike arrangement are distributed at 60° each among the stereo bases left - rear left, rear left - rear right, rear right - right.

The car mentioned in the example above appears, if recorded by such a microphone arrangement, only in the frontal stage.

Basically the same results can be obtained with omnis, too, but it requires very big distances between the microphones, and this will prove unworkable in most recording situations (about 2 m diameter). Therefore only the arrangement with five cardioids was realized as shown in ill. 9.

The distance of the individual stereo bases can be obtained from the Williams-curves [17].



Ill. 9: ICA-5

Testing Procedure

Judgments of individuals will always be subjective. In order to obtain more objective results, an experiment with 88 participants was carried out. Statistics then permit to extract a more objective result. The test participants were asked to answer five questions about the tonal quality of the recording techniques on the basis of five examples. Each technique was compared with all the others in sets of two examples at a time. A relative, symmetrical, five-grade ordinal assessment scale with the grades "much better", "better" and "same" was used. From the total of the answers the median, which determines the winner of the comparisons on pairs can be ascertained, as well as the quartile deviation which is a measure of dispersion. In addition the validity and reliability of the test were analyzed separately with regard to each question. The integration of all the pair-comparisons makes it possible to establish a ranking list for each quality mark of the recording techniques.

First of all the individual questions and the ranking of the techniques:

ICA = Ideal Cardioid Arrangement; ABC = AB-stereophony for 3/2-stereo; BSS = Bruck's Surround Sphere; SAM = "Surround-Atmo-Mikrofon" (surround ambience microphone); MSD = Mid-Side-Depth

How well is the spatial impression of the concert hall conveyed?

ABC > ICA ³ BSS ³ MSD

How can you localize the individual instruments?

ICA > ABC > BSS > MSD

How do you rate the sound as a whole?

ICA < ABC > BSS >> MSD

(The examples used in the test consisted of extracts from "Also sprach Zarathustra" by R. Strauss, performed by the "Duesseldorfer Symphoniker" at the Tonhalle at Duesseldorf.)

Which example gives you a more precise impression of the driving car? (One drives right to left, another drives from front right to rear right)

ICA > SAM > BSS >> MSD

Which example makes you feel more like being right in the middle of the station? (ambience from Duesseldorf main station during peak hours)

ICA 8 SAM 8 BS > MSD

The interpretation of this ranking and the analysis of the extra comments from the participants permitted answering of the following questions:

What is the advantage of the center channel with regard to localization in the frontal stage area?

During the test, the participants were seated in four different positions (ill. 10), and these were analyzed separately.

The center channel improves localization on the outside sets in particular. Participants generally confirmed that techniques with the center channel ICA and ABC are superior than the four-channel techniques "Surround-Atmo-Mikrofon" and Surround-Sphere. MSD, in spite of it's center channel, was worst and this proves that the center channel has advantages only with suitable recording techniques. If methods ICA and ABC are used, the area in which a satisfactory localization is possible is enlarged. There was no decrease in quality of localization due to the additional center channel. ICA is also indisputably better than the four-channel techniques in the ideal listening position, whereas ABC is placed on the same level as the surround sphere.

Is localization outside the frontal stage area possible?

It was particularly fascinating for the participants to follow a car driving past them from front to rear. Altogether, however, the localization of sounds from the sides were felt to be worse than of sounds in the frontal stage area. For the recording of sounds which are to be represented outside the frontal stage L-C-R, ICA and the "Surround-Atmo-Mikrofon" can be recommended.

How is a spatial impression transmitted?

The transmission of the spatial impression of a concert hall with the 3/2-stereo.standard is much better than with 2-channel-stereo. As with 2-channel-stereo, omnis produce the best spatial impression in 3/2-stereo, too: Asking for spatial impression, ABC was rated significantly better than all the other techniques. The "Surround-Atmo-Mikrofon" used as an ambience microphone for orchestra recordings didn't prove to be of particular advantage. The reduction of undesired rear localization by filtering the surround channels was deemed to be helpful.

How do the tested recording techniques differ in tonal quality?

As far as the overall impression is concerned the participants definitely preferred ICA and ABC, followed by the surround sphere. In contrast to techniques that use ambience microphones placed far away from the orchestra a certain amount of rear localization cannot be completely avoided with the surround sphere, due to the high amount of direct sound in the surround channels, and this has a marked negative effect on the overall impression.

MSD performed worst. The sound was described as "centered" and "too narrow".

There is no clear preference for either ICA or ABC. ABC has the advantage of better spatial impression as well as the better performance in the bass reproduction of omnis. ICA has the advantage of a more sophisticated localization. Moreover, ICA is considered to be "softer". The choice between the two techniques in a special recording situation is similar to the choice between AB and ORTF in 2-channel-stereo.

How much influence has the position of the listener on the result?

The advantage of the center channel is more obvious in the side positions than in the center positions. In the rear position there is a stronger tendency towards rear localization. The widening of the listening area horizontally must be bought at the expense of a shortening in depth. The shortening of the listening area, however, is diminished if the surround signals are aftertreated in a suitable manner. The listening position does not affect the general sound quality of the recording techniques. There was the same favorite techniques on all listening positions with ever pair of comparisons.

Conclusion

For the best possible 3/2-stereo-recording ICA-5 can be recommended. The IRT's "Surround-Atmo-Mikrofon" is recommended only in situations in which only four transmission channels are available and no dominant single sound source within the ambience is to be localized. The attempt to imitate Ambisonics with traditional studio equipment and in this way to get by with 3 channels only was not successful. The tested distribution of the entire recording angle with ICA-5 for ambience recordings (360° $\Rightarrow$ 180° of the 3 front channels and 180° on the rest) has proved to be practicable and advantageous. Experimenting with different kinds of distributions might be helpful if special ambiances are to be recorded.

For orchestra recordings ABC and ICA-3, complemented by two ambience microphones for the surround channel, can be recommended. Which technique in particular is to be chosen depends on whether priority is given to precise localization or to the spatial impression coupled with voluminous bass performance. Artificial reverberation was not tested because the quality of the equipment which is used is of crucial importance. There is no doubt that in many concert halls experiments with artificial reverberation are essential. In principle the difference between ABC and ICA-3 is equivalent to the difference between AB and ORTF.

[1] H. Buttenberg: „Wege zum echten Klangerlebnis“, Radio und Fernsehen, Berlin, 7 (1958), 9, S. 289-292

[2] V. Aschoff: „Probleme der elektroakustischen Einkanalübertragung“, Bericht der Arbeitsgemeinschaft für Forschung des Landes Nordrhein-Westfalen, Ber. 39, 1954

[3] H. Lauridsen und F. Schlegel: „Stereophonie und richtungsdiffuse Klangwiedergabe“, Gravesaner Blätter, 1956, Nr. V, August, S.28-50

[4] R. Vermeulen: „Stereo Reverberation“, Philips technical Review, 17.März 1956

[5] D. Kleis: „Experimente zur Verbesserung der Raumwirkung von Schall“, Presented at the 3. Tonmeister-tagung 1954, Detmold

[6] A. Jonquct und J.P. Pignon: „Holophony and Tetraedrophony – some new views about 3-D-Stereophony“, Presented at the 56th AES Convention, Paris 1977

[7] A. Silzle und G. Theile: „HDTV-Mehrkanalton: Untersuchung zur Abbildungsqualität beim Einsatz zusätzlicher Mittenlautsprecher“, Presented at the 16th Tonmeistertagung 1990

[8] Ziegelmeyer u.a.: „Darstellung seitlicher Schallquellen bei Anwendung des 3/2-Stereo Formates“, Presented at the 19th Tonmeistertagung 1996

[9] G. Theile: „Möglichkeiten und Grenzen der 3/2-Stereo Aufnahme“, 19th Tonmeistertagung, Karlsruhe 1996

[10] J. Bruck: „Solving the Surround Dilemma“, 19th Tonmeistertagung, Karlsruhe 1996

[11] M. Haßler: "Musikaufnahme und -wiedergabe mir dem 5-Kanal-HDTV Audiosystem 3/2", Diplomarbeit FH Düsseldorf 1993

[12] S. Schneider: "Practical Experiences with Multichannel Postproduction", Presented at the 100th AES Convention 1996, May 11-14, Copenhagen

[13] G. Theile: „Das Kugelflächenmikrofon“, Bericht zur 14. Tonmeistertagung 1986, S. 277-293, Bildungswerk des Verbands Deutscher Tonmeister

[14] M. A. Gerzon und G. J. Barton: „Ambisonic Decoders HDTV“, AES Preprint 3345, 92nd Audio Engineering for Society Convention, Wien (März 1992)

[15] U. Herrmann und V. Henkels: "Vergleich fünf verschiedener Surround-Hauptmikrofonverfahren", Diplomarbeit an der FH Düsseldorf, März 1997

[16] J. Blauert: „Räumliches Hören“, Hirzel Verlag 1974

[17] M. Williams: „Unified theory of microphone systems for stereophonic sound recording“, AES Preprint 2466 (H-6), 1987

[18] J. Wuttke: Betrachtung bekannter Grundlagen und Theorien stereofoner Aufnahmetechnik, Presented at the 15th Tonmeistertagung, 1988

[19] Gerhard Steinke: "Quo vadis, HDTV Sound", Presented at the 17th Tonmeistertagung, 1992