
STEREO MICROPHONE TECHNIQUES

By Bruce Bartlett

Stereo microphone techniques are used mainly to record classical-music ensembles and soloists on location. These methods capture a sonic event as a whole, typically using only two or three microphones. During playback of a stereo recording, images of the musical instruments are heard in various locations between the stereo speakers. These images are in the same places, left-to-right, that the instruments were at the recording session. In addition, true-stereo miking conveys:

*the depth or distance of each instrument

*the distance of the ensemble from the listener (the perspective)

*the spatial sense of the acoustic environment--the ambience or hall reverberation.

Why Record in Stereo?

When planning a recording session, you may ask yourself, "Should I record in stereo with just a few mics? Or should I use several microphones placed close to the instruments and mix them with a mixer?"

Stereo miking is preferred for classical music, such as a symphony performed in a concert hall or a string quartet piece played in a recital hall. For classical-music recording, stereo methods have several advantages over close-mic methods.

For example, I said that stereo miking preserves depth, perspective, and hall ambience -- all when you use several closeup mics, panned into position. But with a good stereo recording, you get a sense of an ensemble of musicians playing together in a shared space. Also, a pair of mics at a distance relays instrument timbres more accurately than closeup mics. Close-miked instruments in a classical setting sound too bright, edgy, or detailed compared to how they sound in the audience area.

Another advantage of stereo miking is that it tends to preserve the ensemble balance as intended by the composer. The composer has assigned dynamics (loudness notations) to the various instruments in order to produce a pleasing ensemble balance in the audience area. Thus, the correct balance or mix of the ensemble occurs at a distance, where all the instruments blend together acoustically. But this balance can be upset with multi-miking; you must rely on your own judgment (and the conductor's) regarding mixer settings to produce the composer's intended balance. Of course, even a stereo pair of mics can yield a faulty balance. But a stereo pair, being at a distance, is more likely to reproduce the balance as the audience hears it.

Other Applications for Stereo Miking

In contrast to a classical-music recording, a pop-music recording is made with multiple close mics because it sounds tighter and cleaner, which is the preferred style of production for pop music. Close miking also lets you experiment with multitrack mixes after the session. Still, stereo miking can be used in pop-music sessions for large sound sources within the ensemble, such as

*groups of singers

*piano

*drum-set cymbals overhead

*vibraphone, xylophone, and other percussion instruments

*string and horn sections

Other uses for stereo miking are

*samples

*sound effects

*background ambience and stereo dialog for film, video, and electronic news gathering

*audience reaction

*sports broadcasts

*radio group discussions

*radio plays

*drama

Goals of Stereo Miking

Let's focus now on stereo-miking a large musical ensemble and define what we want to achieve. One goal is accurate localization. That is, the reproduced instruments should appear in the same relative locations as they were in the live performance. When this is achieved, instruments in the center of the ensemble are reproduced midway between the two playback speakers. Instruments at the sides of the ensemble are reproduced from the left or right speaker. Instruments located half-way to one side are reproduced half-way to one side, and so on.

Figure 1 shows three stereo localization effects. In Figure 1(a), various instrument positions in an orchestra are shown: left, left-center, center, right-center, right. In Figure 1(b), the reproduced images of these instruments are accurately localized between the stereo pair of speakers. The stereo spread or stage width extends from speaker to speaker. If the microphones are placed improperly, the effect is either the narrow stage width shown in Figure1(c) or the exaggerated separation shown in Figure 1(d). (Note that a large ensemble should spread from speaker to speaker, while a quartet can have a narrower spread.)

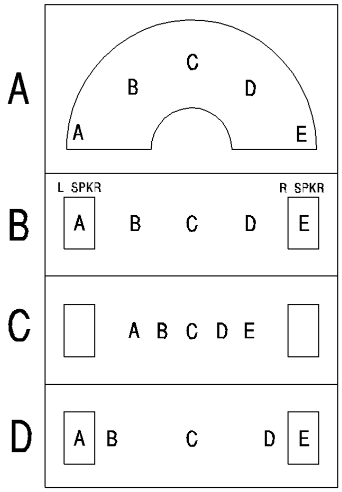

Figure 1. Stereo localization effects.

(a) Orchestra instrument locations (top view).

(b) Images accurately localized between speakers (the listener's perception).

(c) Narrow stage-width effect.

(d) Exaggerated separation effect.

To judge these stereo localization effects, it's important to position yourself properly with respect to the monitor speakers. Sit as far from the speakers as they are spaced apart. The speakers will appear to be 60 degrees apart, which is about the same angle an orchestra fills when viewed from the typical ideal seat in the audience (say, tenth row center). Sit exactly between the speakers (equidistant from them); otherwise, the images will shift toward the side on which you're sitting and will become less sharp. Also, pull out the speakers several feet from the walls: this delays and weakens early reflections which can degrade stereo imaging.

The reproduced size of an instrument or instrumental section should match its size in real life. A guitar should be a point source; a piano or string section should have some stereo spread. Each instrument's location should be as clearly defined as it was in the concert hall, as heard from the ideal seat. Some argue that the reproduced images should be sharper than in real life to supplant the missing visual cues. In other words, since you can't see the instruments during loudspeaker reproduction, extra-sharp images might enhance the realism.

The reproduced reverberation (concert-hall ambience) should either surround the listener, or at least it should spread evenly between the speakers (as shown in Figure 2). Typical stereo miking techniques reproduce the hall reverberation up front, in a line between the speakers, so you don't get a sense of being immersed in the hall ambience. To make the recorded reverberation surround the listener, you need extra speakers to the side or rear, an add-on reverberation simulator, or a head-related crosstalk canceller. However, spaced-mic recordings can artificially produce a sense of some reverb around you.

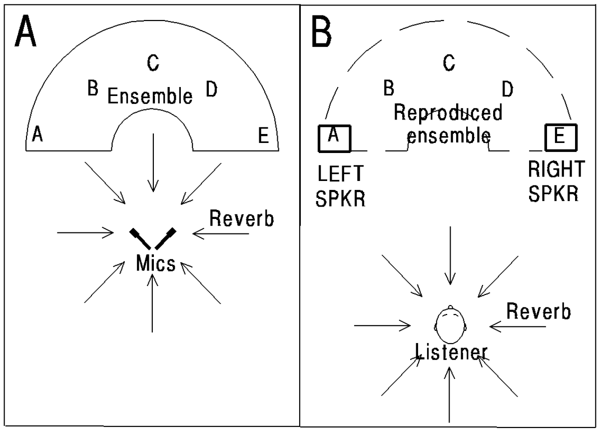

Figure 2. Accurate imaging: sound source location and size, and the reverberant field, are reproduced during playback. (a) Recording -- top view. (b) Playback -- top view.

There should also be a sense of stage depth. Front-row instruments should sound closer than back-row instruments.

Types of Stereo Microphone Techniques

There are four general mic techniques used for stereo recording:

*coincident pair

*spaced pair

*near-coincident pair

*baffled-omni pair or artificial head.

Let's look at each technique in detail.

Coincident Pair

With the coincident-pair method (XY or intensity stereo method), two directional mics are mounted with grilles nearly touching and diaphragms one above the other, angled apart to aim approximately toward the left and right sides of the ensemble (Figure 3). For example, two cardioid microphones can be mounted angled apart, their grilles one above the other. Other directional patterns can be used, too. The greater the angle between microphones, and the narrower the polar pattern, the wider the stereo spread.

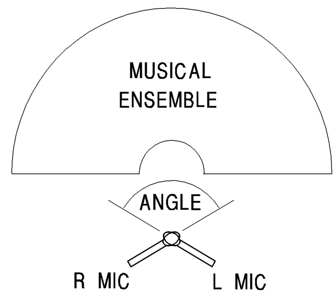

Figure 3. Coincident-pair technique (top view).

Let's explain how the coincident-pair technique produces localizable images. A directional mic is most sensitive to sounds in front of the microphone (on-axis) and progressively less sensitive to sounds arriving off-axis. That is, a directional mic produces a relatively high-level signal from the sound source it's aimed at and a relatively low-level signal for all other sound sources.

The coincident-pair method uses two directional mics symmetrically angled from the center line, as in Figure 3. Instruments in the center of the ensemble produce an identical signal from each microphone. During playback, an image of the center instruments is heard midway between the stereo pair of loudspeakers. That's because identical signals in each channel produce a centrally located image.

If an instrument is off-center to the right, it is more on-axis to the right-aiming mic than to the left-aiming mic, so the right mic will produce more signal than the left mic. During playback of this recording, the right speaker will play louder than the left speaker, reproducing the image off-center to the right -- where the instrument was during recording.

The coincident array codes instrument positions into level differences (intensity or amplitude differences) between channels. During playback, the brain decodes these level differences back into corresponding image locations. A pan pot in a mixing console works on the same principle.

If one channel is 15 to 20 dB louder than the other, the image shifts all the way to the louder speaker. So, if we want the right side of the orchestra to be reproduced at the right speaker, the right side of the orchestra must produce a signal level 20 dB higher from the right mic than from the left mic. This occurs when the mics are angled apart sufficiently. The correct angle depends on the polar pattern. Instruments part-way off-center produce interchannel level differences less than 20 dB, so they are reproduced part-way off-center.

Listening tests have shown that coincident cardioid microphones tend to reproduce the musical ensemble with a narrow stereo spread. That is, the reproduced ensemble does not spread all the way between speakers.

A coincident-pair method with excellent localization is the Blumlein array, which uses two bidirectional mics angled 90 degrees apart and facing the left and right sides of the ensemble.

A special form of the coincident-pair technique is the mid-side (MS) recording method shwon in Figure 4. A mic facing the middle of the orchestra is summed and differenced with a bidirectional mic aiming to the sides. This produces left- and right-channel signals. With this technique, the stereo spread can be remote-controlled by varying the ratio of the mid signal to the side signal. This remote control is useful at live concerts, where you can't physically adjust the microphones during the concert. MS localization accuracy is excellent.

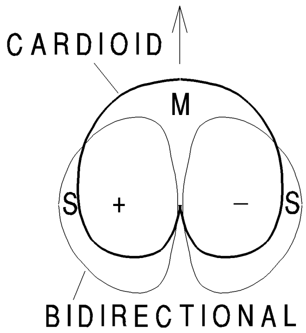

Figure 4 Mid-side technique. Left channel = mid + side. Right channel = mid - side. The polarity of the side mic lobes is indicated by + and -.

Mid-side recordings are sometimes said to lack spaciousness. But according to David Griesinger, this can be improved with spatial equalization, in which a circuit boosts the bass 4 dB (+2 dB at 600 Hz) in the L - R, or side, signal, with a corresponding cut in the L + R, or mid, signal. Another way to improve the spaciousness is to mix in a distant MS microphone -- one set about 30 ft from the main MS microphone.

A stereo mic uses two coincident mic capsules mounted in a single housing for convenience.

A recording made with coincident techniques is mono-compatible, that is, the frequency response is the same in mono or stereo. Because of the coincident placement, there is no time or phase difference between channels to degrade the frequency response if both channels are combined to mono. If you expect your recordings to be heard in mono (say, on radio or TV), you should consider coincident methods.

Spaced Pair

With the spaced-pair (or A-B) technique, two identical mics are placed several feet apart, aiming straight ahead toward the musical ensemble (Figure 5). The mics can have any polar pattern, but the omnidirectional pattern is the most popular for this method. The greater the spacing between mics, the greater the stereo spread.

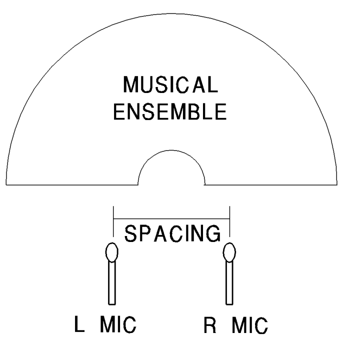

Figure 5. Spaced-pair technique.

Instruments in the center of the ensemble produce an identical signal from each microphone. During playback of this recording, an image of the center instruments is heard midway between the stereo pair of loudspeakers.

If an instrument is off-center, it is closer to one mic than the other, so its sound reaches the closer mic before it reaches the other one. So the microphones produce almost the same signal, except that one mic signal is delayed with respect to the other. If you send an identical signal to two stereo speakers with one channel delayed, the sound image shifts off center. With a spaced-pair recording, off-center instruments produce a delay in one mic channel, so they are reproduced off-center.

The spaced-pair array codes instrument positions into time differences between channels. During playback, the brain decodes these time differences back into corresponding image locations. It takes only about 1.5 milliseconds (msec) of delay to shift an image all the way to one speaker, so if we want the right side of the orchestra to be reproduced at the right speaker, its sound must arrive at the right mic about 1.5 msec before it reaches the left mic. In other words, the mics should be spaced about 2 ft apart, because this spacing produces the appropriate delay to place right-side instruments at the right speaker. Instruments part-way off-center produce interchannel delays less than 1.5 msec, so they are reproduced part-way off-center.

If the spacing between mics is, say, 12 ft, instruments slightly off-center produce interchannel delays greater than 1.5 msec, which places their images at the left or right speaker. This is called an "exaggerated separation" or "ping pong" effect.

On the other hand, if the mics are too close together, the delays produced will be inadequate to provide much stereo spread. In addition the mics will tend to favor the center of the ensemble because the mics are closest to the center instruments.

To record a good musical balance, we need to place the mics about 10 or 12 ft apart, but such a spacing results in exaggerated separation. One solution is to place a third microphone midway between the original pair and mix its output to both channels. That way, the ensemble is recorded with a good balance, and the stereo spread is not exaggerated.

The spaced-pair method tends to make off-center images relatively unfocused or hard to localize. Here’s why. Spaced-mic recordings have time differences between channels, and stereo images produced solely by time differences are relatively unfocused. Centered instruments are still heard clearly in the center, but off-center instruments are difficult to pinpoint between speakers. This method is useful if you prefer the sonic images to be diffuse or blended, rather than sharply focused.

There's another problem with spaced mics. The large time differences between channels correspond to gross phase differences between channels. Out-of-phase low-frequency signals can cause excessive vertical modulation of a record groove, making records difficult to cut unless the cutting level or low-frequency stereo separation is reduced. (This is not a problem with CDs or cassettes). In addition, combining both mics to mono sometimes causes phase cancellations of various frequencies, which may or may not be audible.

There is an advantage with spaced miking, however. Spaced mics are said to provide a warm sense of ambience, in which concert hall reverberation seems to surround the instruments and, sometimes, the listener. Here's why. The two channels of recorded reverberant sound are incoherent; that is, they have random phase relationships. Incoherent signals from stereo loudspeakers sound diffuse and spacious. Since reverberation is picked up and reproduced incoherently by spaced mics, it sounds diffuse and spacious. The simulated spaciousness caused by the phasiness is not necessarily realistic, but it is pleasant to many listeners.

Another advantage of the spaced-mic technique is the ability to use omnidirectional microphones. An omni condenser mic has more extended low-frequency response than a unidirectional condenser mic and tends to have a smoother response and less off-axis coloration.

Near-Coincident Pair

As shown in Figure 6, the near-coincident technique uses two directional microphones angled apart, with their grilles spaced horizontally a few inches apart. Even a few inches of spacing increases the stereo spread and adds a sense of depth and airiness to the recording. The greater the angle or spacing between mics, the greater the stereo spread.

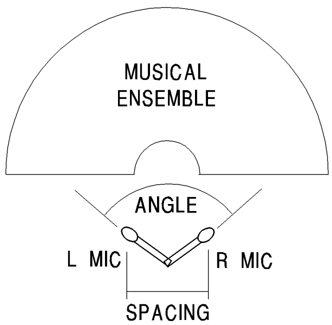

Figure 6. Near-coincident-pair technique.

Here's how this method works: angling directional mics produces level differences between channels; spacing mics produces time differences. The level differences and time differences combine to create the stereo effect. If the angling or spacing is too great, the result is exaggerated separation. If the angling or spacing is too small, the result is a narrow stereo spread.

The most common example of the near-coincident method is the ORTF system, which uses two cardioids angled 110 degrees apart and spaced seven inches (17 cm) apart horizontally. (ORTF stands for Office de Radiodif- fusion Television Française--French Broadcasting Organization.) This method tends to provide accurate localization; that is, instruments at the sides of the orchestra are reproduced at or very near the speakers, and instruments half-way to one side tend to be reproduced half-way to one side.

Baffled-Omni Pair

With this method, two omnidirectional mics are separated a few inches by a baffle between them. The baffle is a hard disk covered with absorbent foam (as in the Jecklin disk, Fig. 7). Or the baffle is a hard sphere with the mics flush-mounted on opposite sides (as in the Neumann and Schoeps spherical mics, Fig. 8.) Another format uses two Pressure Zone Microphones, ear-spaced on angled boundaries, with a foam baffle between the mics (as in the Crown SASS-P MKII – Fig. 9).

With the baffled-omni pair, the level, time, and spectral differences between channels create the stereo images. The omni condenser mics used in this method have excellent low-frequency response.

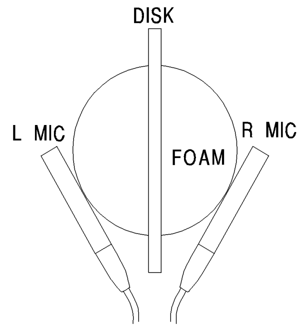

Figure 7. OSS system or Jecklin disk. Omnis spaced 16.5 cm (6.5 in.) and separated by a foam-covered disk of 28 cm (11 7/8 in.) diameter.

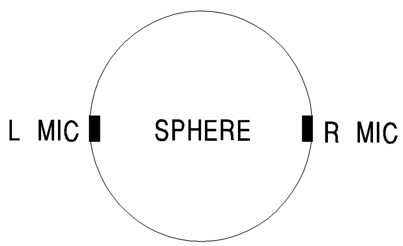

Figure 8. Schoeps spherical microphone.

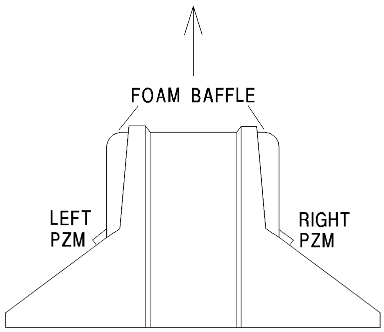

Figure 9. Crown SASS-P MKII stereo PZM microphone.

Comparing the Four Stereo Miking Techniques

The **coincident-pair technique** has the following features:

*It uses two directional mics angled apart with grilles nearly touching, one mic's diaphragm above the other.

*Level differences between channels produce the stereo effect.

*Images are sharp.

*Stereo spread ranges from narrow to accurate.

*Signals are mono-compatible.

The **spaced-pair technique** has these features:

*It uses two mics spaced several feet apart.

*Time differences between channels produce the stereo effect.

*Off-center images are diffuse.

*Stereo spread tends to be exaggerated unless a third center mic is used.

*It provides a warm sense of ambience.

*It may cause record-cutting problems.

The **near-coincident-pair technique** has these features:

*It uses two directional mics angled apart and spaced a few inches apart.

*Level and time differences between channels produce the stereo effect.

*Images are sharp.

*Stereo spread tends to be accurate.

*It provides a greater sense of "air" and depth than coincident methods.

The **baffled-omni-pair technique** has these features:

*It uses two omnidirectional mics a few inches apart, separated by a baffle.

*Level, time, and spectral differences between channels produce the stereo effect.

*Images are sharp.

*Stereo spread tends to be accurate.

*Low-frequency response is excellent.

Mounting Hardware

With coincident and near-coincident techniques, the microphones should be rigidly mounted with respect to one another, so that they can be moved as a unit without disturbing their arrangement. A device for this purpose is called a stereo mic adapter or stereo bar. It mounts two mics on a single stand with adjustable angling and spacing between mics..

Microphone Requirements

The sound source dictates the requirements of the recording microphones. Most acoustic instruments produce frequencies from about 40 Hz (string bass and bass drum) to about 20,000 Hz (cymbals, castanets, triangles). A microphone with uniform response between these frequency limits will do full justice to the music.

The highest octave from 10 kHz to 20 kHz adds transparency, air, and realism to the recording. You may need to filter out frequencies below 80 Hz to eliminate rumble from trucks and air conditioning, unless you want to record organ or bass-drum fundamentals.

Sound from an orchestra or band approaches each microphone from a broad range of angles. To reproduce all the instruments' timbres equally well, the microphone should have a broad, flat response at all angles of incidence within at least 90 degrees, that is, the polar pattern should be uniform with frequency. Mics with small-diameter diaphragms usually meet this requirement best. (Note that some mics have small diaphragms inside large housings.)

If you're forced to record at a great distance, a frequency response elevated up to 4 dB above 4 kHz might sound more natural than a flat response. Another benefit of a rising high end is that you can roll it off in post production, reducing analog tape hiss. Since classical music covers a wide dynamic range (up to 80 dB), the recording microphones should have very low noise and distortion. In distant-miking applications, the sensitivity should be high to override mixer noise.

For sharp imaging, the microphone pair should be well matched in frequency response, phase response, and polar pattern.

We've investigated several microphone arrangements for recording in stereo. Each has its advantages and disadvantages. Which method you choose depends on the sonic compromises you are willing to make.

For a practical application of these mic techniques, please see the article
Stereo Recording Procedures.

STEREO RECORDING PROCEDURES

By Bruce Bartlett

In the article *Stereo Microphone Technques*, I described stereo miking techniques and how they work. This article is more practical. It's divided into three parts:

1. On-location stereo recording of a classical-music ensemble.

2. The basics of stereo miking for popular music.

3. A troubleshooting guide to help you pinpoint and solve problems in stereo reproduction.

Let's start by going over the equipment and procedures for recording classical music.

Equipment

Before going on-location, you need to assemble a set of equipment such as this:

*microphones (low-noise condenser or ribbon type, omni or directional, free field or boundary, stereo or separate)

*MS matrix box (optional)

*recorder (open-reel, DAT, etc.)

*low-noise mic preamps (unless the mic preamp in your recorder is very good)

*phantom-power supply (unless your mic preamp or mixer has phantom built-in)

*mic stands and booms or fishing line

*stereo bar

*shock mount (optional)

*microphone extension cable

*Dolby noise reduction (optional)

*mixer (optional)

*headphones and/or speakers

*power amplifier for speakers (optional)

*blank tape

*stereo phase-monitor oscilloscope (optional)

*power strip, extension cords

*notebook and pen

*tool kit

First on the list are microphones. You'll need at least two or three of the same model number or one or two stereo microphones. Good mics are essential, because the mics--and their placement--determine the sound of your recording. You should expect to spend at least $250 per microphone for professional-quality sound.

For classical-music recording, the preferred microphones are condenser or ribbon types with a wide, flat frequency response and very low self-noise. A self-noise spec of less than 21 dB equivalent SPL, A-weighted, is recommended.

You'll need a power supply for condenser microphones: either an external phantom-power supply, a mixer or mic preamp with phantom power, or internal batteries.

If you want to do spaced-pair recording, you can use either omnidirectional or directional microphones. Omnis are preferred because they generally have a flatter low-frequency response. If you want to do coincident or near-coincident recording for sharper imaging, use directional microphones (cardioid, supercardioid, hypercardioid, or bidirectional). The baffled-pair technique uses omni mics.

You can mount the microphones on stands or hang them from the ceiling with nylon fishing line. Stands are much easier to set up, but are more visually distracting at live concerts. Stands are more suitable for recording rehearsals or sessions with no audience present.

The mic stands should have a tripod folding base and should extend at least 14 ft high. To extend the height of regular mic stands, you can either use baby booms or use telescoping photographic stands (available from camera stores). These are lightweight and compact.

A useful accessory is a stereo bar or stereo mic adapter. This device mounts two microphones on a single stand for stereo recording. Another needed accessory in most cases is a shock mount to prevent pickup of floor vibrations.

In difficult mounting situations, boundary microphones may come in handy. They can lie flat on the stage floor to pick up small ensembles or can be mounted on the ceiling or on the front edge of a balcony. They also can be attached to clear plexiglass panels that are hung or mounted on mic stands.

For monitoring in the same room as the musicians, you need some closed-cup, circumaural (around the ear) headphones to block out the sound of the musicians. You want to hear only what's being recorded. Of course, the headphones should be wide-range and smooth for accurate monitoring. A better monitoring arrangement might be to set up an amplifier and close-field loudspeakers in a separate room.

If you're in the same room as the musicians, you'll have to sit far from the musicians to clearly monitor what you're recording. To do that, you'll need a pair of 50-ft mic extension cables. Longer extensions will be needed if the mics are hung from the ceiling or if you're monitoring in a separate room.

If you use noise reduction, you'll also need a small stereo microphone mixer to boost the mics' signal level up to the line level required by the noise-reduction system. A mixer is also necessary when you want to record more than one source--for example, an orchestra and a choir, or a band and a soloist. You might put a pair of microphones on the orchestra and another pair on the choir. The mixer blends the signals of all four mics into a composite stereo signal. It also lets you control the balance (relative loudness) among microphones.

For monitoring a mid-side recording, bring an MS matrix box that converts the MS signals to L-R signals, which you monitor.

Note: be sure to test all your equipment for correct operation before going on the job.

Choosing the Recording Site

If possible, plan to record in a venue with good acoustics. There should be adequate reverberation time for the music being performed (at least 2 seconds for orchestral recording).This is very important, because it can make the difference between an amateur- sounding recording and a commercial-sounding one. Try to record in an auditorium, concert hall, or spacious church rather than in a band room or gymnasium. Avoid stage shells because they lack a sense of space.

You may be forced to record in a hall that is too dead: that is, the reverberation time is too short. In this case, you may want to add artificial reverberation from a digitial reverb unit or cover the seats with plywood sheets or 4-mil polyethylene plastic sheeting. Strong echoes can be controlled with carpets, RPG diffusors, or drapes. Dry climates tends to shorten the reverb time and dull the sound.

Session Setup

If the orchestral sound from the stage is bad, you might want to move the orchestra out onto the floor of the hall.

Take out your microphones and place them in the desired stereo miking arrangement. As an example, suppose you're recording an orchestra rehearsal with two crossed cardioids on a stereo bar (the near-coincident method). Screw the stereo bar onto a mic stand and mount two cardioid microphones on the stereo bar. For starters, angle them 110 degrees apart and space them 7 inches apart horizontally (the ORTF method). Aim them downward so that they'll point at the orchestra when raised. You may want to mount the microphones in shock mounts or put the stands on sponges to isolate the mics from floor vibration.

Basically, you place two or three mics several feet in front of the group, raised up high (as in Figure 1).

The microphone placement controls the *perspective* or sense of distance to the ensemble, the balance among instruments, and the stereo imaging.

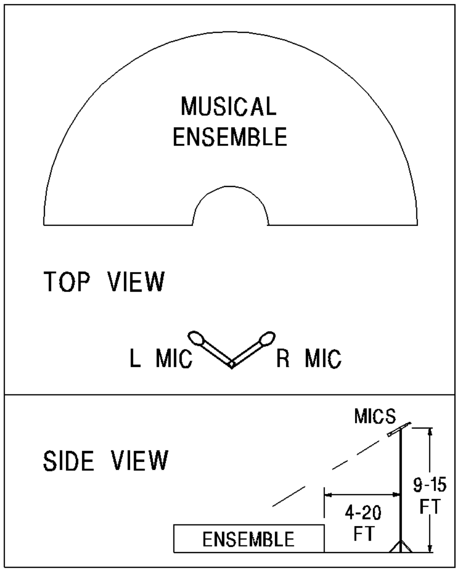

Figure 1. Typical microphone placement for on-location recording of a classical music ensemble.

As a starting position, place the mic stand behind the conductor's podium, about 12 ft in front of the front-row musicians. Connect mic cables and mic extension cords. Raise the microphones about 14 ft off the floor. This prevents overly loud pickup of the front row relative to the back row of the orchestra.

Leave some extra turns of mic cable at the base of each stand so you can reposition the stands. This slack also allows for people accidentally pulling on the cables. Try to route the mic cables where they won't be stepped on, or cover them with mats.

Live, broadcast, or filmed concerts require an inconspicuous mic placement, which may not be sonically ideal. In these cases, or for permanent installations, you'll probably want to hang the microphones from the ceiling rather than using stands. You can hang the mics by their cables or by nylon fishing line of sufficient tensile strength to support the weight of the microphones. Another inconspicuous placement is on mic-stand booms projecting forward of a balcony in front of the stage. For drama or musicals, directional boundary mics can be placed on the stage floor near the footlights.

Now you're ready to make connections. There are several different ways to do this:

*If you're using just two mics, you can plug them directly into a phantom supply (if necessary), and from there into your tape deck. You might prefer to use low-noise mic preamps, then connect cables from there into your recorder line inputs.

*If you're using two mics and a noise-reduction unit, plug the mics into a mixer or preamp to boost the mic signals up to line level. Then run that line-level signal into the noise-reduction unit connected to the recorder line inputs.

*If you're using multiple mics (either spot mics or two MS mics) and a mixer, plug the mics into a snake box. Plug the mic connectors at the other end of the snake into your mixer mic inputs. Finally, plug the mixer outputs into the recorder line inputs.

*If you're also using noise reduction, plug the mixer outputs into the inputs of the noise-reduction device and from there into the recorder.

*If you want to feed your mic signals to several mixers--for example, one for recording, one for broadcast, and one for sound reinforcement--plug your mic cables into a mic splitter or distribution amp. Connect the splitter outputs to the snakes for each mixer. Supply phantom from one mixer only, on the microphone side of the split. Each split will have a ground-lift switch on the splitter. Set it to *ground* for only one mixer (usually the recording mixer). Set it to *lift* or *float* for the other mixers. This prevents hum caused by ground loops between the different mixers.

*If you're using directional microphones and want to make their response flat at low frequencies, you can run them through a mixer with equalization for bass boost. Boost the extreme low frequencies until the bass sounds natural or until it matches the bass response of omni condenser mics. Connect the mixer output either into an optional noise-reduction unit or directly into your recorder. This equalization will be unnecessary if the microphones have been pre-equalized by the manufacturer for flat response at a distance.

Monitoring

Put on your headphones or listen over loudspeakers in a separate room. Sit equidistant from the speakers--as far from them as they are spaced apart. You'll probably need to use a close-field arrangement (speakers about 3 ft apart and 3 ft from you) to reduce coloration of the speakers' sound from the room acoustics.

Turn up the recording-level controls and monitor the signal. When the orchestra starts to play, set the recording levels to peak roughly around -10 VU so you have a clean signal to monitor. You'll set levels more carefully later on.

Microphone Placement

Nothing has more effect on the production style of a classical-music recording than microphone placement. Miking distance, polar patterns, angling, spacing, and spot miking all influence the recorded sound character. Let’s examine each aspect of mic placement.

Miking Distance

The microphones must be placed closer to the musicians than a good live listening position. If you place the mics out in the audience where the live sound is good, the recording will probably sound muddy and distant when played over speakers. That's because all the recorded reverberation is reproduced up-front -- on a line between the playback speakers -- mixed with the direct sound of the orchestra. Close miking (5 to 20 ft from the front row) compensates for this effect by increasing the ratio of direct sound to reverberant sound.

The closer the mics are to the orchestra, the closer it sounds in the recording. If the instruments sound too close, too edgy, too detailed, or if the recording lacks hall ambience, the mics are too close to the ensemble. Move the mic stand a foot or two farther from the orchestra and listen again.

If the orchestra sounds too distant, muddy, or reverberant, the mics are too far from the ensemble. Move the mic stand a little closer to the musicians and listen again.

Eventually you'll find a sweet spot where the direct sound of the orchestra is in a pleasing balance with the ambience of the concert hall. Then the reproduced orchestra will sound neither too close nor too far.

Here's why miking distance affects the perceived closeness (perspective) of the musical ensemble: the level of reverberation is fairly constant throughout a room, but the level of the direct sound from the ensemble increases as you get closer to it. Close miking picks up a high ratio of direct-to-reverberant sound; distant miking picks up a low ratio. The higher the direct-to-reverb ratio, the closer the sound source is perceived to be.

An alternative to finding the sweet spot is to place a stereo pair close to the ensemble (for clarity) and another stereo pair distant from the ensemble (for ambience). According to Delos Recording Director John Eargle, the distant pair should be no more than 30 ft from the main pair. If the distant pair is farther, its signal might simulate an echo. You mix the two pairs with a mixer. The advantages of this method are

*It avoids pickup of bad-sounding early reflections.

*It allows remote control (via mixer faders) of the direct/reverb ratio or the perceived distance to the ensemble.

*Comb filtering due to phase cancellations between the two pairs is not severe because the delay between them is great, and their levels and spectra are different.

Skip Pizzi recommends a "double MS" technique, which uses a close MS microphone mixed with a distant MS microphone (as shown in Figure 2). One MS microphone is close to the musical ensemble for clarity and sharp imaging, and the other is out in the hall for ambience and depth. The distant mic could be replaced by an XY pair for lower cost. Also, the distant mic could be recorded on separate tracks for use as surround channels.

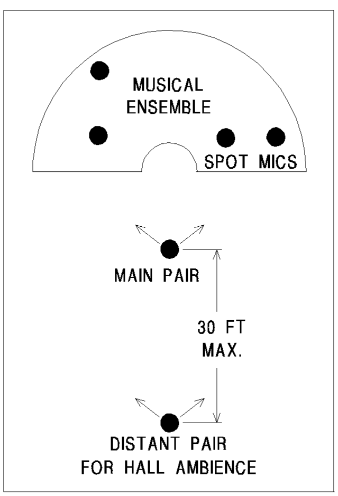

Figure 2. Double MS technique using a close main pair and a distant pair for ambience. Spot mics are also shown.

If the ensemble is being amplified through a sound-reinforcement system, you might be forced to mike very close to avoid picking up amplified sound and feedback from the reinforcement speakers.

For broadcast or communications, consider miking the conductor with a wireless lavalier mic.

Stereo-Spread Control

Now that you've settled on a miking distance, concentrate on the stereo spread. If the monitored spread is too narrow, it means that the mics are angled or spaced too close together. Increase the angle or spacing between mics until localization is accurate.

Note: increasing the angle between mics will make the instruments sound farther away; increasing the spacing will not.

If off-center instruments are heard far-left or far-right, that indicates your mics are angled or spaced too far apart. Move them closer together until localization is accurate.

If you record with a mid-side microphone, you can adjust the stereo spread by remote control at the matrix box with the stereo spread control (M/S ratio control). You can change the monitored stereo spread either during the recording or after:

*To change the spread during the recording, connect the stereo-mic output to the matrix box and connect the matrix-box output to the recorder. Use the stereo-spread control (M/S ratio) in the matrix box to adjust the stereo spread.

*To alter the spread after the recording, record the mid signal on one track and the side signal on another track. Monitor the output of the recorder with a matrix box. After the recording, run the mid and side tracks through the matrix box, adjust the stereo spread as desired, and record the result.

If you are set up before the musicians arrive, check the localization by recording yourself speaking from various positions on stage while announcing your position (e.g., "left side," "mid-left," "center"). Play back the recording to judge the localization accuracy provided by your chosen stereo array. Recording this localization test at the head of a tape is an excellent practice.

Monitoring Stereo Spread

Full stereo spread on speakers is a spread of images from the left speaker to the right speaker. Full stereo spread on headphones can be defined as stereo spread from ear to ear. The stereo spread heard on headphones may or may not match the stereo spread heard over speakers, depending on the microphone technique used.

Due to psychoacoustic phenomena, coincident-pair recordings have less stereo spread over headphones than over loudspeakers. Take this into account when monitoring with headphones or use only loudspeakers for monitoring.

If you are monitoring your recording over headphones or anticipate headphone listening to the playback, you may want to use near-coincident miking techniques, which have similar stereo spread on headphones and loudspeakers.

Ideally, monitor speakers should be set up in a close-field arrangement (say, 3 ft from you and 3 ft apart) to reduce the influence of room acoustics and to improve stereo imaging.

If you want to use large monitor speakers placed farther away, deaden the control-room acoustics with Sonex [tm] or thick fiberglass insulation (covered with muslin). Place the acoustic treatment on the walls behind and to the sides of the loudspeakers. This smooths the frequency response and sharpens stereo imaging.

You'll probably want to include a stereo/mono switch in your monitoring system, as well as an oscilloscope. The 'scope is used to check for excessive phase shift between channels, which can degrade mono frequency response or cause record-cutting problems. Connect the left-channel signal to the 'scope's vertical input; connect the right-channel signal to the horizontal input, and look for the lissajous patterns shown in Figure 3.

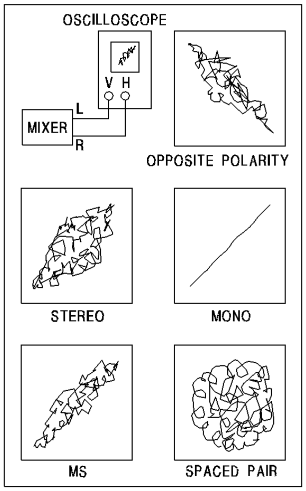

Figure 3. Oscilloscope lissajous patterns showing various phase relationships between channels of a stereo program.

Soloist Pickup and Spot Microphones

Sometimes a soloist plays in front of the orchestra. You'll have to capture a tasteful balance between the soloist and the ensemble. That is, the main stereo pair should be placed so that the relative loudness of the soloist and the accompaniment is musically appropriate. If the soloist is too loud relative to the orchestra (as heard on headphones or loudspeakers), raise the mics. If the soloist is too quiet, lower the mics. You may want to add a spot mic (accent mic) about 3 ft from the soloist and mix it with the other microphones. Take care that the soloist appears at the proper depth relative to the orchestra.

Many record companies prefer to use multiple mics and multitrack recording for classical music. This gives more control of balance and definition and is necessary in difficult situations. Often you must add spot or accent mics on some instruments or instrumental sections to improve the balance or enhance clarity (as shown in Figure 2). In fact, John Eargle contends that a single stereo pair of mics rarely works well.

A choir that sings with an orchestra can be placed behind the orchestra, miked with two to four cardioids. Or the choir can stand in the audience area facing the orchestra.

Pan each spot mic so that its image position coincides with that of the main microphone pair. Using the mute switches on your mixing console, alternately monitor the main pair and each spot to compare image positions.

You might want to use an MS microphone or stereo pair for each spot mic, and adjust the stereo spread of each local sound source to match that reproduced by the main pair. For example, suppose that a violin section appears 20 degrees wide as picked up by the main pair. Adjust the perceived stereo spread of the MS spot mic used on the violin section to 20 degrees, then pan the center of the section image to the same position that it appears with the main mic pair.

When you use spot mics, mix them at a low level relative to the main pair--just loud enough to add definition, but not loud enough to destroy depth. Operate the spot-mic faders subtly or leave them untouched. Otherwise the close-miked instruments may seem to jump forward when the fader is brought up, then fall back in when the fader is brought down. If you bring up a spot-mic fader for a solo, drop it only 6 dB when the solo is over -- not all the way off.

Often the timbre of the instrument(s) picked up by the spot mic is excessively bright. You can fix it with a high-frequency rolloff, perhaps by miking off-axis. Adding artificial reverb to the spot mic can help too.

To further integrate the sound of the spots with the main pair, you might want to delay each spot's signal to coincide with those of the main pair. That way, the main and spot signals are heard at the same time. For each spot mic, the formula for the required delay is

$T = D/C$

where

T = delay time in seconds

D = distance between each spot mic and the main pair in feet

C = speed of sound, 1130 ft per second.

For example, if a spot mic is 20 ft in front of the main pair, the required delay is 20/1130 or 17.7 msec. Some engineers add even more delay (10-15 msec) to the spot mics to make them less noticeable.

Setting Levels

Once the microphones are positioned properly, you're ready to set recording levels. Ask the orchestra to play the loudest part of the composition, and set the recording levels for the desired meter reading. A typical recording level is +3 VU maximum on a VU meter or -3dB maximum on a peak-reading meter for a digital recorder. The digital unit can go up to 0 dB maximum without distortion, but aiming for -3 dB allows for surprises.

When recording a live concert, you'll have to set the record-level knobs to a nearly correct position ahead of time. Do this during a pre-concert trial recording, or just go by experience: set the knobs where you did at previous sessions (assuming you're using the same mics at this session).

Multitrack Recording

British Decca has developed an effective recording method using an 8-track recorder:

*record the main pair on two tracks

*record the distant pair on two tracks

*record panned accent mics on two tracks

*mix down the three pairs of tracks to two stereo tracks

Stereo Miking for Pop Music

Most current pop music recordings are made using multiple close-up mics (one or more on each instrument). These multiple mono sources are panned into position and balanced with faders. Such an approach is convenient but often sounds artificial. The size of each instrument is reduced to a point, and each instrument might sound isolated in its own acoustic space.

To enhance the realism, mike parts of the ensemble in stereo. Overdub several of these stereo pickups. Such a technique can provide the feeling of a musical ensemble playing together in a common ambient space. Realism is improved for several reasons:

*The more-distant miking provides more natural reproduction of timbre.

*The size of each instrument is reproduced.

*Time cues for localization are included (with near-coincident and spaced techniques).

*The sound of natural room acoustics is included.

True-stereo recording works especially well for these sound sources:

*acoustic jazz combos and small folk groups (sometimes)

*soloist or singer/guitarist

*drum kit (overhead)

*piano (out front and in line with the lid, or over the strings)

*background vocals

*horn and string sections

*vibraphone and xylophone

*other percussion instruments and ensembles

If you record several performers with a stereo pair, this method has some disadvantages. You must adjust their balance by moving the performers toward and away from the mics during the session. This takes longer and costs more than moving faders of individual tracks after the session. In addition, the performances are not acoustically isolated. So if someone makes a mistake, you must re-record the whole ensemble rather than just the flawed performance.

The general procedures for true-stereo recordings are below:

1. Adjust the acoustics around the instruments. Add padding or reflective surfaces if necessary. You might prefer the sound obtained by putting the musicians near the center of a large, live room. This setup reduces early reflections but includes ambient reverberation.

2. Place the musicians around the stereo mic pair where you want them to appear in the final mix. For

example, you might overdub strings spread between center and far right and horns spread between center and far left. Try to keep the acoustic bass and lead instruments/singers in the center.

3. Experiment with different microphone heights (to vary the tonal balance) and miking distance (to vary the amount of ambience). Three to six feet distance is typical.

4. If some instruments or vocalists are too quiet, move them closer to the mics until the balance is good.

5. If an instrument lacks definition, consider giving it a spot mic. Mix it in at a low level.

Figure 4 shows a jazz group miked in stereo.

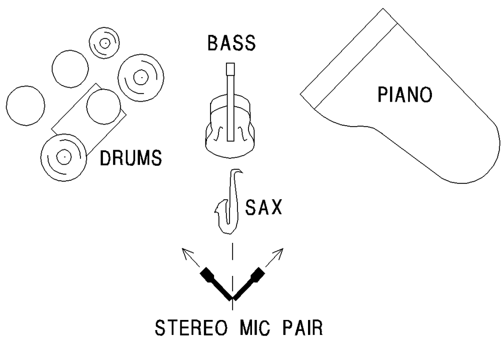

Figure 4. Stereo-miking a jazz group.

Troubleshooting Stereo Sound

Suppose that you're monitoring a recording. Something doesn't sound right. How can you pinpoint what's wrong and how can you fix it?

This section lists several procedures to solve audio-related problems. Read down the list of bad sound descriptions until you find one matching what you hear, then try the solutions until your problem disappears.

Before you start, check for faulty cables and connectors. Also check all control positions; rotate knobs and flip switches to clean the contacts.

Distortion in the Microphone Signal

*Use pads or input attenuators in your mixer.

*Switch in the pad in the condenser micr, if any.

*Use a mic with a higher "Maximum SPL" specification.

Too Dead (Insufficient Ambience, Hall Reverberation, or Room Acoustics)

*Place mics farther from performers.

*Use omnidirectional mics.

*Record in a concert hall with better acoustics (longer reverberation time).

*Add artificial reverberation.

*Add plywood or plastic sheeting over the audience seats.

Too Detailed, too Close, too Edgy

*Place mics farther from performers.

*Place mics lower or on the floor (as with a boundary microphone).

*Using an equalizer in your mixing console, roll off the high frequencies.

*Use duller-sounding mics.

*If using both a close-up pair and a distant ambience pair, turn up the ambience pair.

*If using spot mics, add artificial reverb or delay the signal to coincide with that of the main pair.

Too Distant (too much Reverberation)

*Place mics closer to the sound source.

*Use directional mics (such as cardioids).

*Record in a concert hall that is less "live" (reverberant).

*If using both a close-up pair and a distant ambience pair, turn down the ambience pair.

Narrow Stereo Spread (Fig. 5C)

*Angle or space the main mic pair farther apart.

*If doing mid-side stereo recording, turn up the side output of the stereo microphone.

*Place the main mic pair closer to the ensemble.

*If monitoring with headphones, narrow stereo spread is normal when you use coincident techniques. Try monitoring with loudspeakers, or use near-coincident or spaced techniques.

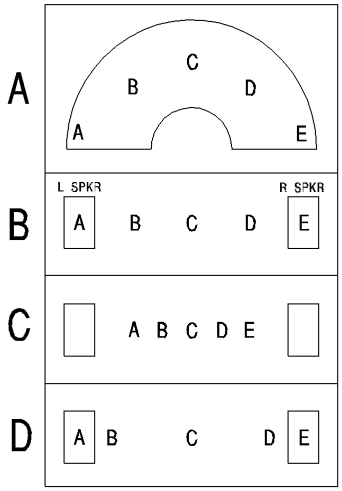

Figure 5. Stereo localization effects.

(a) Orchestra instrument locations (top view).

(b) Images accurately localized between speakers (the listener's perception).

(c) Narrow stage-width effect.

(d) Exaggerated separation effect.

Excessive Separation or Hole-in-the-Middle (Figure 5D)

*Angle or space the main microphone pair closer together.

*If doing mid-side stereo recording, turn down the side output of the stereo microphone or use a cardioid mid instead of an omni mid.

*In spaced-pair recording, add a microphone midway between the outer pair and pan its signal to the center.

*Place the mics farther from the performers.

*Place the loudspeaker pair closer together. Ideally, they should be as far apart as you are sitting from them, to form a listening angle of 60 degrees.

Poorly Focused Images

*Avoid spaced-mic techniques.

*Use a spatial equalizer.

*Use a microphone pair that is better-matched in frequency response and phase response.

*If the sound source is out of the in-phase region of microphone pickup, move the source or the microphone. For example, the in-phase region of a Blumlein pair of crossed figure eights is l45 degrees relative to center.

*Be sure that each spot mic is panned so that its image location coincides with that of the main pair.

*Use loudspeakers designed for sharp imaging. Usually these are signal-aligned, have vertically aligned drivers, have curved edges to reduce diffraction, and are sold in matched pairs.

*Place the loudspeakers several feet from the wall behind them and from side walls to delay and weaken the early reflections that can degrade stereo imaging.

Images Shifted to One Side (Left-Right Balance Is Faulty)

*Adjust the right-or-left recording level so that center images are centered.

*Use a mic pair that is better-matched in sensitivity.

*Aim the center of the mic array exactly at the center of the ensemble.

*Sit exactly between your stereo speakers, equidistant from them. Adjust the balance control or level controls on your monitor amplifier to center a mono signal.

Lacks Depth (Lacks a Sense of Nearness and Farness of Various Instruments)

*Use only a single pair of mics out front. Avoid multi-miking.

*If you must use spot mics, keep their level low in the mix, and delay their signals to coincide with those of the main pair.

Lacks Spaciousness

*Use a spatial equalizer.

*Space the microphones apart.

*Place the microphones farther from the ensemble.

Early Reflections too Loud

*Place mics closer to the ensemble and add a distant microphone for reverberation (or use artificial reverberation).

*Place the musical ensemble in an area with weaker early reflections.

*If the early reflections come from the side, try aiming bidirectionals at the ensemble. Their nulls will reduce pickup of side-wall reflections.

Bad Balance (Some Instruments too Loud or too Soft)

*Place the mics higher or farther from the performers.

*Move quiet instruments closer to the stereo mic pair, and vice versa.

*Ask the conductor or performers to change the instruments' written dynamics.

*Add spot mics close to instruments or sections needing reinforcement. Mix them in subtly with the main mics' signals.

*Increase the angle between mics to reduce the volume of center instruments, and vice versa.

*If the center images of a mid-side recording are weak, use a cardioid mid instead of an omni mid.

Muddy Bass

*Aim the bass-drum head at the microphones.

*Put the microphone stands and bass-drum stand on resilient isolation mounts, or place the mics in shock-mount stand adapters.

*Roll off the low frequencies or use a highpass filter set around 40 to 80 Hz.

*Record in a concert hall with less low-frequency reverberation.

Rumble from Air Conditioning, Trucks, and so on

*Temporarily shut off air conditioning. Record in a quieter location.

*Use a high-pass filter set around 40 to 80 Hz. Use microphones with limited low-frequency response.

Bad Tonal Balance (too Dull, too Bright, Colored)

*Change the microphones.

*If a mic must be placed near a hard reflective surface, use a boundary mic to prevent phase cancellations between direct and reflected sounds.

*Adjust equalization. Compared to omni condenser mics, directional mics usually have a rolled-off low-frequency response and may need some bass boost.

*If strings sound strident, move mics farther away or lower.

*If the tone quality is colored in mono monitoring, use coincident-pair techniques.

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SOUND ON SOUND
The World's Best Music Recording Magazine

Stereo Microphone Techniques Explained

Part 1

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Technique : Theory + Technical

PART 1: HUGH ROBJOHNS takes a historical look at stereo miking techniques and explains the whys and wherefores of the various methods available.

The first documented stereo microphone system was used (entirely by accident, in fact) at the great Electrical Exhibition in Paris in 1881. A French designer by the name of Clement Ader was demonstrating some improvements to an early telephone system, and stumbled across what we would now call the spaced-microphone stereo technique! Unfortunately, no one realised the significance of Ader's discovery and he went on to invent the inflatable bicycle tyre before playing with aeroplanes, calling his first plane 'A vion', which became the generic name for aeroplanes in the French language.

Most of the development of stereo recording as we know it today happened in the very early '30s, and almost simultaneously in America and the UK. In the USA, Bell Laboratories were working on systems using spaced microphones under the direction of Dr Harvey Fletcher. Meanwhile, in the UK, a very clever man called Alan Blumlein, working for EMI, was developing an alternative system which relied on coincident microphones.

Both methods were years ahead of their time and both had advantages and disadvantages. It was not until the invention of PVC in the '50s (which allowed micro-groove vinyl records to be produced) that either of these techniques were adopted commercially, but today both formats are alive and well, and are often used in concert with each other.

In this article, I'll be looking at what stereo microphone systems are trying to achieve, also taking a closer look at the coincident stereo ideas which have become the mainstay of many practical recording techniques. Next month, I'll talk about spaced microphone systems and combinatorial techniques.

WHAT IS STEREO?

The word 'stereophonic' is actually derived from Greek, and means 'solid sound', referring to the construction of believable, solid, stable sound images, regardless of how many loudspeakers are used. It can be applied to surround-sound systems as well as to simple two-channel techniques -- indeed, in the cinema, the original Dolby Surround system was called Dolby Stereo, even though it was a four-channel system! However, most people are conditioned to think of stereo as a two-channel system, and this is the definition I'll adopt in these articles.

There are basically three ways of creating stereo sound images over a pair of loudspeakers:

* The first is an entirely artificial technique based on Alan Blumlein's work, and uses pan pots to position the sound images from individual microphones by sending different proportions of each microphone to the two channels.

* The second technique (and one we will look at in more detail next month) is the use of two or more identical but spaced microphones. These microphones capture sounds at differing times because of their physical separation, and so record time-of-arrival information in the two channels.

* The third system is that of coincident microphones, and this has become the backbone of all radio, television, and a lot of commercial stereo recordings. This technique uses a pair of identical directional microphones , each feeding one channel. The microphones capture sound sources in differing levels between the two channels, much like the pan-pot system, but this time the signal amplitudes vary in direct relation to the physical angle between microphones and sound sources.

COINCIDENT MICROPHONES

Blumlein developed coincident techniques to overcome the inherent deficiencies (as he saw them) of the spaced microphone systems being developed in America. Since our hearing mechanism relies heavily on timing information (see 'The Human Hearing Process' box), Dr Harvey Fletcher thought it reasonable to use microphones to capture similar timing differences, and that is exactly what the spaced microphone system does.

However, when sound is replayed over loudspeakers, both ears hear both speakers, so we actually receive a very complex pattern of timing differences, involving the real timing differences from each speaker to both ears, plus the recorded timing differences from the microphones. This arrangement tends to produce rather vague positional information, and if the two channels are combined to produce a mono signal, comb-filtering effects can often be heard.

Blumlein demonstrated that by using only the amplitude differences between the two loudspeakers, it was possible to fool the human hearing system into translating these into perceived timing differences, and hence stable and accurate image positions. We all take this entirely for granted now, and are quite happy with the notion that moving a pan-pot or balance control to alter the relative amplitudes of a signal in the two channels will alter its position in the stereo image in an entirely predictable and repeatable way.

This process is used every day to create artificial stereo images from multi-miked recordings, but contrary to popular belief, the level difference between the two channels which is necessary to move a sound image all the way to one loudspeaker is not very much. Typically, a 12 to 16dB difference between channels is sufficient to produce a full left or right image, and about 6dB will produce a half-left or right image -- although the exact figures vary with individual listeners, the monitoring equipment and the listening environment.

To create stereo images directly from real life, Blumlein needed to develop a microphone technique which captured level differences between the two channels, but no timing differences. To avoid timing differences, the two microphones must be placed as close together as is physically possible -- hence the term 'Coincident Stereo'. The normal technique is to place the capsule of one microphone immediately above the other, so that they are coincident in the horizontal plane, which is the dimension from which we are trying to recreate image positions (despite hi-fi magazines' claims to the contrary, conventional stereo recording does not encode meaningful height information!). Amplitude differences between the two channels are created through the microphone's own polar patterns, making them more or less sensitive to sounds from various directions. The choice of polar pattern is the main tool we have for governing the nature of the recorded sound stage.

If you read books on stereo techniques, you'll find a variety of alternative terms used to describe the various methods in use. The kind of coincident stereo discussed here is also known as 'XY' recording (in America and parts of Europe), 'AB' recording (in the BBC and most other European broadcasters), 'crossed pairs', or just plain 'normal stereo'. The term 'AB stereo' takes on a different meaning in the USA, where it is often used to describe spaced microphone arrays -- beware of the potential for confusion!

PRACTICAL TECHNIQUES

In general, we aim to place sound sources around stereo microphones such that they occupy the complete stereo image. If you consider an orchestra, for example, it's usual to have the back row of the violins fully to the left, and the back row of the cellos or basses fully to the right.

To create this spread of sound using crossed cardioids to record the orchestra, it would be necessary to place them directly above the conductor in order to achieve the desired stereo image width. To take another example, crossed figure-of-eights would have to be positioned a long way down the hall to achieve the same stereo width (see Figure 1).

It should be obvious from these comments that in choosing the polar patterns for the microphones, you also determine the physical separation between sound sources and microphones for a given stereo width, and therefore the perspective of the recording. In the example above, the cardioids would give a very close-perspective sound, with little room acoustic and a distorted orchestral balance favouring the close string players over the musicians towards the rear and sides of the orchestra. In contrast, the figure-of-eights would give a much more natural and balanced perspective to the orchestra, but would also capture a great deal of the hall's acoustic, which might make the recording rather more distant than anticipated.

It's quite possible that neither of these basic techniques would produce an entirely satisfactory result, and a compromise might be to use crossed hypercardioid mics (with an acceptance angle of about 150 degrees). More likely, a combination of the two original techniques, plus a scattering of close 'spot' mics to reinforce the weaker sections of the orchestra (using pan-pots to match their stereo images to the main crossed pairs), would have to be used. The crucial point is that there is no absolutely correct technique, only an array of tools which you must choose and use to obtain the results you want.

COMBINING CROSSED PAIRS AND SPOT MICROPHONES

A very commonly-used technique is combining a crossed pair (to form the basis of a stereo image) with a number of close microphones (to give particular instruments more presence and definition in the mix). This applies equally whether we're talking about recording a philharmonic orchestra or a drum kit -- only the scale of the job changes; the techniques do not.

There are three things to consider with this combination technique: image position, perspective and timing.

The main stereo pair will establish image positions for each instrument and the close microphones must not contradict this, if we're to avoid confused and messy stereo images. The best technique I know for setting the panning for the close microphones is to concentrate on a particular instrument's image position in the main pair, then slowly fade up the corresponding spot mic and hear how the image changes. If it pulls to the right, fade the spot mic down, adjust the pan-pot to the left (or vice versa) and try again. With practice, you should be able to match image positions in three or four cycles, such that fading the spot mic up only changes the instrument's perspective, not its position.

Clearly, a microphone close to an instrument will have a completely different perspective to one further away, and this contrast is usually undesirable, as it draws undue attention to the instrument in question. The relative balance between the 'spot' mic and the main pair is critical, and it's surprising how little a contribution is required from the close mic in order to sharpen the instrument's definition, which is normally all you're trying to achieve. Remember, if you're aware of the close mic, it's too high in the mix.

"In general, we aim to place sound sources around stereo mics such as that they occupy the complete stereo image."

The last point is relative timing, but this is usually only a problem with large recording venues. Consider an orchestral recording again, where the main stereo pair of, say, hypercardioids, may be 50 or 60 feet away from the orchestra. As sound travels at about one foot every millisecond, the sound from the stereo pair will be about 60ms behind that from any close spot mics. The human hearing system is geared up to analyse the first arriving sounds, which means we naturally tend to be aware of sound from the spot mics before the main pair -- almost irrespective of how low they are in the mix. This is not the situation we want -- the spot mics are supposed to assist the main stereo pair, not the other way around!

The solution is to route all the spot mics to a stereo group (having balanced and panned them appropriately) and send the combined signal to a stereo delay line. Dial in a suitable delay (one millisecond per foot for the distance between the main pair and the most distant spot mic, and then add five to ten milliseconds for good measure). The output of the delay line is returned to the desk and mixed in with the main stereo pair to produce the final mix. By delaying the spot mics, you can cause their signals to be heard after the main stereo pair (by the five or ten milliseconds that were added), and they'll consequently be much harder to perceive as separate entities. In fact, delaying the close mics makes their level in the mix slightly less critical, as the hearing process takes less notice of them, although their panning is still crucial, of course.

This technique is extremely effective, but is rather time-consuming, and few people would bother with it if the main stereo pair was less than about 20 feet from any spot mic.

M&S COINCIDENT TECHNIQUES

There is an alternative coincident stereo technique, again developed originally by Alan Blumlein. This is the M&S, or Mid & Side, technique, mainly used by television sound recordists, but definitely worth knowing about, whatever you record.

M&S is a coincident technique in exactly the same way as the conventional systems already described. Instead of having directional microphones facing partially left and right, the M&S technique uses a pair of microphones, one of any polar pattern you like facing forwards and the other, a figure-of-eight, facing sideways. These two signals have to go through a conversion process before being auditioned on loudspeakers or headphones as normal left-right stereo.

The M&S system offers a number of practical advantages for television sound recordists (which are outside the scope of this article), but the single most useful aspect of the system for everyday recording tasks is that the perceived spread of sound sources across the stereo image can be controlled very easily from the desk.

The most common arrangement is to use a cardioid microphone facing forwards (the 'M' mic), together with a figure-of-eight microphone (the 'S' mic) facing sideways, and when these are converted into normal left-right stereo, they produce an identical acceptance angle to conventional crossed cardioids (see Figure 2). One important point to note: the polarity of the S lobe facing left should be the same as the polarity of the M mic. If this is not the case, the stereo image will be reversed.

As the balance between the M and S microphones is altered, so is the apparent distance between sound sources, as heard on the speakers (the effect is similar to adjusting the mutual angle between a conventional crossed pair of mics; see 'Terminology' box for more on this). This can be used to great effect, and it also allows the image width to be pushed outside the speakers by introducing an out-of-phase element to the signal, although this should be used with great care, as it will affect mono compatibility.

SOUNDFIELD MICROPHONE

This concept of M&S was extended in the design of the Soundfield microphone and its baby brother, the ST250. These microphones were originally developed for Ambisonic recording -- a technique which captures and reproduces true surround sound, with height information as well as 360-degree horizontal imaging (as opposed to the entirely artificial spatial positioning of the various cinema surround systems).

Unfortunately, Ambisonics has never really caught on and although a few companies are producing material suitably encoded (such as classical recordings from Nimbus), most people use the soundfield microphones as glorified, but stunningly accurate, stereo mics.

The soundfield microphones have an array of four cardioid capsules, arranged as the sides of a tetrahedron (two pyramids joined base-to-base), and these are combined electronically to produce four 'virtual microphones' called W, X, Y and Z. The first output (W) is designed to have an omnidirectional polar pattern, while the other three are figure-of-eights facing left-right, front-back and up-down. The way in which the W, X, Y and Z virtual microphones are created simulates extremely close spacing between capsules, so the stereo imaging is phenomenally accurate.

These four signals are combined together to produce a stereo output according to the settings on the control unit, in much the same way as the basic M&S arrangement described earlier. The omni (W) signal can be thought of as equating to the M microphone in a simple M&S pair, and the X, Y and Z signals equate to the S microphone, albeit with separate microphones for each direction (up/down, left/right and front/back).

The control unit allows the user to manipulate the Soundfield mic's characteristics to unprecedented degrees. The effective polar patterns of the simulated stereo pair can be selected, as can their mutual angle, and then this virtual stereo array can be pointed and tilted in any direction, simply by manipulating the way in which the four signals are combined. One of the most amazing aspects of the soundfield microphone is that by changing the balance between the W signal and all of the others, the mic can be made to appear to 'zoom in' to the sound source! It is even possible to record the four base signals individually (called the B-format) and then use the control unit to manipulate the microphone's characteristics on playback.

Next month, we'll look at spaced microphone arrays such as the Decca Tree and Binaural recording, as well as some of the more popular combinatorial techniques.

THE HUMAN HEARING PROCESS

The whole idea of stereo recording is to try to fool our auditory system into believing that a sound source occupies a specific position in space. So how does our hearing determine the positions of sounds around us in real life?

Without getting bogged down in the psychology and biology of the subject, we use three principal mechanisms to identify the positions of sounds around us. The first and probably most important one is that of differing arrival times of sounds at each ear, followed by level differences between the ears for high-frequency sounds, and finally, independent comb-filtering effects from the outer ear (the pinnae).

Since our ears are spaced apart on opposite sides of the head, any sound source off to one side will be heard by one ear fractionally before the other. Also, because there's a large solid object between the ears (the rest of the head), a 'sound shadow' will be created at high frequencies (above about 2kHz) for the distant ear.

Both these mechanisms highlight the possibility of confusion between the direction of a sound source at any given angle in front or behind the listener, since both the timing and level differences would produce the same results for both directions. To overcome this ambiguity, an automatic reflex action causes us to instinctively turn or tilt our heads slightly and the resulting changes in timing and level immediately resolve the confusion.

The third mechanism was discovered relatively recently, and is the reason for the bizarre shape of the pinnae. (I always knew they had to be there for something other than supporting glasses and earrings!) As sounds arrive at the outer ear, some of the sound enters the ear canal directly, while some is reflected off the curved surfaces of the outer ear and into the ear canal. Since the reflected sound has to travel fractionally further, it is delayed, and in combining with the original sound, produces a comb-filter effect, resulting in characteristic peaks and notches in the frequency response. These frequency-response anomalies depend on the particular direction of sound arrival, and it is thought that we build a 'library' memory of the comb-filter characteristics which can be used to help provide crude directional cues.

This whole concept of directional perception is the foundation of the sophisticated signal processing used in systems like QSound and RSS, which try to create surround sound information from a conventional two-channel stereo system. Modifying the frequency response of recognisable sounds to simulate the effects of the pinnae can trick us into perceiving sounds from locations outside the normal stereo spread between the loudspeakers.

TERMINOLOGY, RIGGING AND CALIBRATION

Blumlein performed all his experiments using microphones with figure-of-eight polar patterns (only these and omnidirectional mics were available at the time). Most of the time, the figure-of-eight microphones were arranged at 90 degrees to each other, such that one faced 45 degrees left, and the other 45 degrees right. The angle between microphones is called the 'Mutual Angle', and 90 degrees is the most commonly used. It is possible to change the mutual angle over a small range, to adjust the precise relationship between the physical sound source positions in front of the microphones and their perceived positions in the stereo image, although the effect is often very subtle and few people find it necessary to make such adjustments.

The usable working area in front of the microphone is defined by the polar patterns of the microphones, and is called the 'Acceptance Angle'. The diagrams below show the typical acceptance angles for figure-of-eights and cardioids crossed at 90 degrees. Note that because the figure-of-eights are bi-directional, with opposite polarity lobes, they have two acceptance areas and two out-of-phase areas at the sides.

It is essential to calibrate the microphones and their channels at the desk before attempting to record anything in stereo. Even nominally identical microphones will have slightly differing sensitivities, and the input channels in the desk could be set up completely differently -- so it is important to run through a line-up procedure (which is far quicker to do than to read -- honest!)

What we need to achieve is identical signal levels in the left and right desk channels for a given sound pressure level in front of the microphones. The easiest and most accurate technique starts with setting the microphones' polar patterns to the desired response (if using switchable mics) and connecting them to two desk channels (or a stereo channel, if available). Turn the pan pots on paired mono channels fully left and right and use a fader clip (or some other means, such as a large bulldog clip) to mechanically fix the two faders together so they track accurately. Rig the microphones one above the other with their capsules as close together as possible, and turn them to face in the same direction while someone speaks in front of them (about two feet away and at their mid-height, if possible, to ensure minimal level differences).

In the control room, switch the loudspeaker monitoring to mono (do not use the channel pan pots, because their centre positions may not be accurate), and adjust one mic channel for the typical operating gain you expect to need, with the fader in its normal operating position. Check that there is no EQ in circuit in either channel and switch a phase reverse into the second channel. Adjust the second channel's gain until the combined output from the microphones is as quiet as possible -- there should be a very obvious null point (it will never completely cancel, because of inaccuracies in the microphones and desk channels, but it should get extremely quiet).

Next, remove the phase reversal and loudspeaker mono-ing, and with the two mics still facing forward, have your talking assistant wander in a complete circle all the way around the microphone array. If the stereo image moves away from the centre, the mics have incompatible polar patterns and will not produce accurate stereo images. Select another pair of microphones and start over.

Finally, rotate the microphones to face 45 degrees left and right (make sure the microphone connected to the panned-left channel is turned to face the left of the sound stage) and have your assistant confirm the image boundaries and left-right orientation. Having completed the line-up, do not re-plug the microphones, or adjust the channel gains, as the calibration will be destroyed and you'll have to go through the entire process all over again! In practice, this whole procedure should take about a minute and should become routine.

A lot of engineers use a 'stereo bar' as a more convenient way to mount a pair of mics from a single mic-stand. Although this technique introduces small timing differences into the recording, it is a perfectly acceptable technique, provided the microphones face outwards rather than inwards after the line-up process. The reason for this is that each microphone casts a sound shadow at high frequencies across the other, and if they face inwards this is likely to degrade the stereo image (particularly if the mics in question are physically large, such as C414s, or U87s). If the mics face outwards, the sound shadow will fall on the rear of each microphone, where it is relatively insensitive anyway (assuming cardioid or hypercardioid patterns) and will not cause imaging problems.

DECODING M&S PAIRS

To decode the M&S signals to normal left and right, pan the M microphone to the centre and split the S microphone to feed a pair of adjacent channels (or a single stereo channel). Gang the two S channel faders together, pan them hard left and right, and switch in the phase reverse on the right channel.

Listening with the monitoring switched to mono, balance the gains of the two S channels for minimal output (make sure there is no EQ switched into either channel). Once the two S channels have been aligned, revert to stereo monitoring, fade up the M channel and adjust the balance between the M and S signals for the desired image spread.

Putting a phase reverse in the M channel will swap the stereo image over -- left going to the right and vice versa -- and the image width can be varied from mono, through normal stereo, up to extra wide, simply by moving the S fader up and down.

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SOUND ON SOUND
The World's Best Music Recording Magazine

Stereo Microphone Techniques Explained

Part 2

Published in SOS March 1997

PART 2: HUGH ROBJOHNS continues his history of stereo recording techniques with a look at the development of spaced microphone arrays.

Last month we investigated the various coincident techniques for stereo recording developed by Alan Blumlein in the early 1930s; this time we'll be covering some alternative techniques using spaced microphone arrays.

WALL OF SOUND

Some of the earliest stereophonic experiments were made in America under the direction of Dr Harvey Fletcher at Bell Laboratories, as mentioned in the first part of this feature. One of the techniques investigated was the 'Wall of Sound', which used an enormous array of microphones hung in a line across the front of an orchestra. Up to 80 microphones were used, and each fed a corresponding loudspeaker, placed in an identical position, in a separate listening room.

The operating principle was that the array of microphones 'sampled' the wave-fronts of sound emanating from the orchestra, and these exact wave-fronts were recreated by the loudspeakers in the listening room. The results were extremely good, with remarkably precise imaging and very realistic perspectives. However, the technology of the '30s was such that recording or transmitting 80 discrete signals was simply not practical.

Consequently, the initial microphone array was systematically simplified to find the minimum number of microphones that produced acceptable results. The general consensus was that three microphones and three loudspeakers represented the best compromise between high-quality imaging and practicality.

Today, the three-spaced-microphone technique is still in widespread use (one form being the Decca Tree) and the three-loudspeaker arrangement is the standard method of frontal sound reproduction in every cinema!

MONO COMPATIBILITY

So, what is the disadvantage of spaced microphone techniques compared with the coincident systems? Well, the main problem has to be mono compatibility. Any array that has multiple microphones spaced apart from each other will capture the sound from a given source at different times. If the outputs from all of the microphones are mixed together (to produce a single mono signal), the sound will become coloured because of a process known as 'comb filtering' -- the beginnings of phasing or flanging. In a severe case, the comb filtering may alter the sound to such an extent that an orchestra will sound as if it's at the other end of a long cardboard tube. The greater the number of combined microphones, the worse the effect is likely to be.

However, if you can guarantee that recordings produced with a spaced microphone array will not be combined to make a mono signal, the comb-filtering problem becomes totally irrelevant -- the argument adopted by many of the organisations that record classical music.

Since virtually all of the classical music catalogue is on CD or cassette these days, mono replay is no longer an important consideration -- when was the last time you saw a mono CD or cassette player? Whether you're listening on a serious hi-fi, in the car, on a mini-system, or through a 'Brixton briefcase', it will almost always be in stereo. Even broadcast music is in stereo on Classic FM or Radio 3 for the vast majority of listeners.

LOVELY OMNIS

So classical music, in particular, is often recorded with spaced microphone arrays because mono compatibility is not an issue -- but why would anyone want to use spaced arrays? What is wrong with coincident systems which offer mono compatibility as standard?

As we saw last month, all coincident systems have to use directional microphones in order to create the necessary level differences between the two channels of the stereo system. Directional microphones rely on the pressure-gradient principle, which has an inherent problem with low frequencies. The mechanical design of the microphone diaphragm assembly has to compensate for the inadequacies of the pressure gradient by making the diaphragm resonate at very low frequencies. Although this can achieve an acceptable frequency response, it generally compromises the sound quality, restricting the smoothness and extension of the very lowest frequencies.

"The principle of binaural recording is to replicate the way our ears capture sounds, and replay those sounds directly into the corresponding ears."

On the other hand, omnidirectional microphones do not suffer from any of these compromises. They have very smooth and extended low-frequency regions, with very even off-axis responses, both of which are very desirable characteristics. The only problem, of course, is that omnidirectional microphones do not work terribly well as coincident pairs because they do not produce level differences proportional to the angle of incident sound. Omnidirectional microphones can only be used to record in stereo if you space them apart and deliberately record timing differences.

SPACED STEREO

I mentioned last month that reproducing the kind of timing differences captured by a spaced microphone array over a pair of loudspeakers would confuse our hearing system and therefore not produce good stereo images. This was only a half-truth, I'm afraid! It is true that replaying a stereo recording with timing differences between the two channels leads to a confusing set of time-of-arrival differences for our ears, but the sound is normally still perceived as having width and a certain amount of imaging information, and it usually sounds a lot more spacious than a coincident recording.

The problem (as far as Alan Blumlein was concerned, anyway) is that, apart from the mono compatibility issues, the imaging is not very precise and often seems to huddle around the loudspeakers rather than spreading uniformly between them. In really bad cases, the recording may even appear to have a hole in the middle!

If I were to compare the two main types of stereo recording as if they were paintings (a ludicrous thing to do, but I'm going to anyway!), then good coincident recordings are like etchings or line drawings -- very precise imaging, lots of detail, leaving nothing to the imagination. On the other hand, spaced microphone recordings are more like water colours -- the detail is blurred, and the essence is more about impression than reality.

Many people specifically prefer the stereo presentation of spaced-pair recordings, finding them easier to listen to than coincident recordings. There's nothing wrong in that -- as far as the recording engineer is concerned, this is just another technique with a collection of advantages and disadvantages over the alternative formats. It's up to you which technique you use, and as long as you are aware of the characteristics of each system, you are in a position to choose wisely and should be able to achieve the sound quality you seek very quickly.

PRACTICAL SPACED TECHNIQUES

The simplest spaced-microphone technique is to place an identical pair of omnidirectional mics a distance apart in front of the sound source; most engineers would generally choose a spacing of between a half and a third of the width of the actual sound stage. For example, if you set out to record an orchestra, typical positions might be a quarter of the way left and right, either side of the centre line. The distance between orchestra and microphones will depend on the acoustics of the environment and the kind of perspective you want to achieve. For the recording, each microphone feeds its corresponding track on the stereo machine.

The potential problem with this arrangement is a hole in the middle of the stereo representation. The simplest way to avoid this disastrous situation is to bring the mics closer together, but this will affect the spaciousness of the recording, the whole thing tending to become rather narrow and lifeless. The optimal position is often a little harder to find than might be imagined at first.

Although most people use the spaced technique purely so that they can take advantage of the qualities inherent in omnidirectional microphones, there is no reason why you should not use directional microphones in a spaced array -- a very well-known classical music recording engineer, Tony Faulkner, often uses figure-of-eight microphones, for example. The advantage of using directional mics is that it is possible to reject some unwanted signals (typically reverberation) while retaining most of the other advantages of spaced-mic recordings.

"Binaural recording effectively transports our ears directly to the recording venue."

Other spaced techniques that use directional microphones include the ORTF format and the NOS system (both named after the European broadcasting companies that developed them). These are often called 'near coincident' techniques because they combine the level difference recording characteristics of directional coincident microphones, with spaced arrays.

In the case of the ORTF technique, the basic configuration uses a pair of cardioid microphones with a mutual angle of 110°, spaced about 17cm apart. The NOS variant has a 90° mutual angle and a spacing of about 30cm, and -- just for the record -- the Faulkner array uses a pair of figure-of-eights, both facing directly forward, but spaced by about 20cm.

I often use the ORTF or NOS techniques, generally with good results, and I recommend that you experiment around these basic arrangements to find out what works best for you.

BINAURAL RECORDING

Binaural recording is one of those techniques that seem to have a cyclical life. It becomes very popular for a while, then seems to disappear without trace, only to be re-invented a few years later...

Binaural recording is a basic two-microphone spaced-pair technique, but it is rather specialised in that it only works effectively when listened to through headphones. The principle is to replicate the way our ears capture sounds, and replay those sounds directly into the corresponding ears.

Our ears have a hemispherical polar pattern, largely dictated by the lump of meat cunningly positioned between them. As we saw last month, the head creates sound shadows and timing differences for the two ears, so a binaural recording format has to replicate those actions.

The easiest technique is simply to clip a couple of small omnidirectional microphones (tie-clip mics are perfect) to the ears of a willing victim (the arms of a pair of glasses would be less painful!). If recording an orchestra or band, the human mic stand would have to be persuaded not to move their head, but stunning results can be obtained if the microphones are recorded on a cassette or DAT Walkman as you go about your daily chores. Crossing a busy road can cause very entertaining reactions in the listener -- and see what happens if the recording includes your morning ablutions: listening to the binaural sound of someone brushing their teeth is an experience in itself!

A rather more practical method is to use a Jecklin Disc (also known as a 'Henry' in some circles), which mimics the fundamental acoustic aspects of the average head. The disc can be made from perspex or plywood, typically about 25-30cm in diameter, with a mounting point for the microphone stand on one edge, and fixings for a pair of microphones arranged through its centre.

The surface of the disc should be covered in some kind of absorbent padding to avoid reflections from the disc surface back into the microphones, and the mic capsules should be mounted about 15-18cm apart on opposite sides of the disc.

The operating concept is that the microphone-spacing matches that of our ears, and the disc provides the sound-shadowing effects of the head; thus the whole technique should be able to capture signals in the microphones which will closely match those of our own ears. When replayed over headphones, the signals from the disc mics are fed directly into our ear canals, bypassing the effects of our own head- and ear-spacing -- effectively transporting our ears directly to the recording venue.

"Placing spaced microphones is really a black

art."

In practice, the results vary from being incredibly three-dimensional and realistic, to forming a stable and solid image behind, but not in front of, the listener's head. The differences are probably due to the difficulty of accurately matching the dimensions of the recording system to the listener's own physique. If you really want to go overboard, the state-of-the-art binaural technique uses a fully bio-accurate dummy head (a common alternative name for the binaural system is 'dummy-head recording'). Several manufacturers produce anatomically accurate heads, often with the complete torso. Even greater accuracy can be achieved by using carefully shaped pinnae around the microphone capsules and even replicating the various mouth, nose and sinus cavities within the human head!

Interestingly, binaural recordings replayed over loudspeakers manage to convey a sense of stereo width and movement without having any accurate imaging qualities. This facet of the technique is often used to advantage in the production of sound effects for radio and television. In general, sound effects -- especially atmospheric effects -- should convey the environment, but must not distract from the foreground dialogue or action.

Imagine a scene from a radio drama where two actors appear to be conversing on a busy town street. If coincidentally recorded effects were used, the sounds of footsteps and buses travelling across the sound stage could be intensely distracting. However, a binaural recording of the same atmosphere, while being very realistic over headphones, is far less distracting over loudspeakers. Scale, width, perspectives and movement are all conveyed to the listener, but in a laid-back manner that is often far more effective.

SUMMING UP

Spaced techniques allow the engineer to take advantage of the inherent quality of omnidirectional microphones in stereo recordings, particularly their extended and even low-frequency response, and their smooth off-axis pick up. The only problem to be aware of is the potential for comb-filtering effects when spaced microphones are combined, and the more the mics, the worse the effect is likely to be, although it is very hard to predict the audible results.

There are no rules about spacing microphones -- it really is a case of trying an arrangement and listening carefully to the results, then moving things about until you find what you are after. Thinking about the physics of the whole thing helps but, in practice, placing spaced microphones is really a black art, and the best results are almost always obtained by trial and error.

Binaural techniques are a lot of fun, and can be stunningly realistic, but most people prefer to listen over loudspeakers rather than headphones, so the technique is of limited practicality.

Most of my own best recordings have used spaced systems to overcome the weaknesses of coincident techniques (generally, excessive precision and a lack of spaciousness), and my personal favourite arrangement is the near-coincident ORTF setup. This rarely fails to provide the kind of sound I like, and is usually a good starting point from which to build the final sound.

I have used variations on this theme to form the basis of recordings of everything from solo acoustic guitars, complete drum kits, Leslie speakers on Hammonds, and live jazz bands in clubs, up to full concert orchestras.

However, your likes and dislikes, in terms of stereo sound stages and imaging details, are bound to be different from mine, so don't take my word for it, go out and experiment -- it really is the only way!

DECCA TREES

The most commonly used spaced-pair technique is probably the Decca Tree. This was developed many years ago (some time in the early '50s, in fact) to allow the use of omnidirectional microphones to record in stereo.

The basic arrangement is to mount three microphones in a triangular pattern, the central microphone being forward of the others. Dimensions are not particularly critical but, typically, the two rear microphones are about 140cm apart, with the central microphone about 75cm forward of them. The exact dimensions may be varied to suit the size of the sound stage being recorded, and, depending on the polar accuracy of the omnidirectional microphones used, it may help to angle the outer microphones towards the edges of the sound stage so that the microphones' best high-frequency response favours these parts.

The left microphone is recorded on the left channel of the stereo recording, and the right mic on the right channel, as you would expect; the central microphone is distributed equally between the two channels. Although combining the central microphone with the two edges is potentially risky in terms of comb-filtering effects, the hazards are far outweighed by the advantage of a very stable central portion of the sound stage, avoiding any possibility of a 'hole in the middle'.

This extra stability is not only due to the mere presence of an additional microphone covering the centre of the sound stage, but also because the central microphone is forward of the others. The slightly closer proximity to the sound stage means that it will capture sounds before they arrive at either of the other microphones. On replay, this will cause the sound stage to build from the centre, expanding to the edges as the out-rigger microphones capture the sound fractionally later. It is a very subtle effect -- one that works at a subliminal level -- but is crucial to the effectiveness of the Decca Tree format.

COMBINATION TECHNIQUES

In most cases, one primary technique rarely produces the results we want in our recordings. Last month we saw how combining a coincident pair with spot microphones was an effective technique. A similar combination of spaced mics and spot mics also works well, but many engineers prefer to add a spaced array to the full coincident/spot combination. This adds more spaciousness and ambience to the recording and is simply achieved by placing a pair of omnidirectional out-riggers towards the left and right edges of the sound stage. The effect of the omnis is to provide a much richer and more substantial sound, while the coincident pair provides most of the imaging accuracy and the spot mics highlight the inner detail and lift the weaker instruments.

The idea can be used on a drum kit, where spot mics are placed close to each drum head to capture the slap and attack of the sticks on the skins, and a pair of omnidirectional microphones is placed some distance away to give a broad and spacious stereo image. If the microphones are placed low down, towards the floor, they will tend to favour the drums rather than the cymbals, and if they are higher up the reverse is true.

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STEREO RECORDING TECHNIQUES

To enable the recording and/or transmission of sound events, the acoustical signals must be transduced into electrical signals, recorded (transmitted) in this form and finally reconverted into acoustical signals.

There are different terms according to the number of electrical transmission channels used: **monophony** in the case of one channel; **stereophony** in the case of two channels; **quadrophony** in the case of four channels. The number of transmission channels corresponds to the number of possible mutually independent acoustical signals in recording and replay.

Stereophonic recording (transmission) can be done either in
"room-related stereophony" or "head-related stereophony" (dummy-head stereophony).

1 Head-related stereophony (dummy - head stereophony)

Head-related stereophony is a simple and logical technique: Technically speaking the sound field which exist at the location of the two ears of a listener in the recording room is recorded by an artificial head, a so-called dummy head, and is reproduced directly at the ears of a listener in the replay room with the help of earphones.

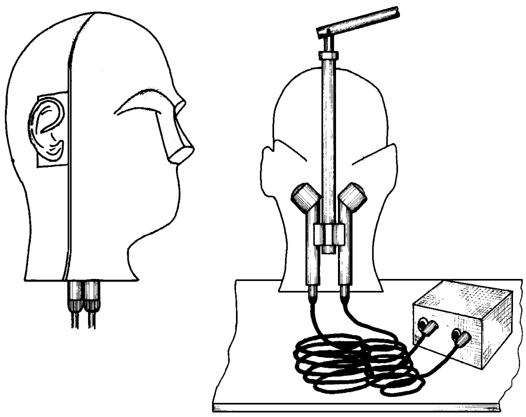


Fig. 1
Dummy head

The dummy head simulates the human head in acoustic sense as far as possible. In place of ear drums the dummy head has suitable microphones whose directional characteristics and other physical properties correspond to the properties of human ears (Fig. 1). Characteristic to and advantageous only for this technique is the fact, that all sound incidence directions, including above and behind, as well as distances, can be exactly reproduced.

The precise natural representation of the room sound (initial reflections and reverberation) is possible only with this technique. The transmission of all sound source directions with only two channels is possible, since the dummy head linearly distorts the signals dependent on the direction of sound incidence, corresponding to the linear distortion of the ear-signals for various directions of sound incidence.

On the other hand we still have the disadvantage that the sound picture, played back by earphones, is always related to the listener’s head and not to the listening room, so that a sound source always follows every movement of the listener's head.
A further disadvantage is that optimal reproduction is only possible with head phones (Fig. 2); replay arrangements with two or four loudspeakers have the substantial disadvantage that the listening area is extremely small.

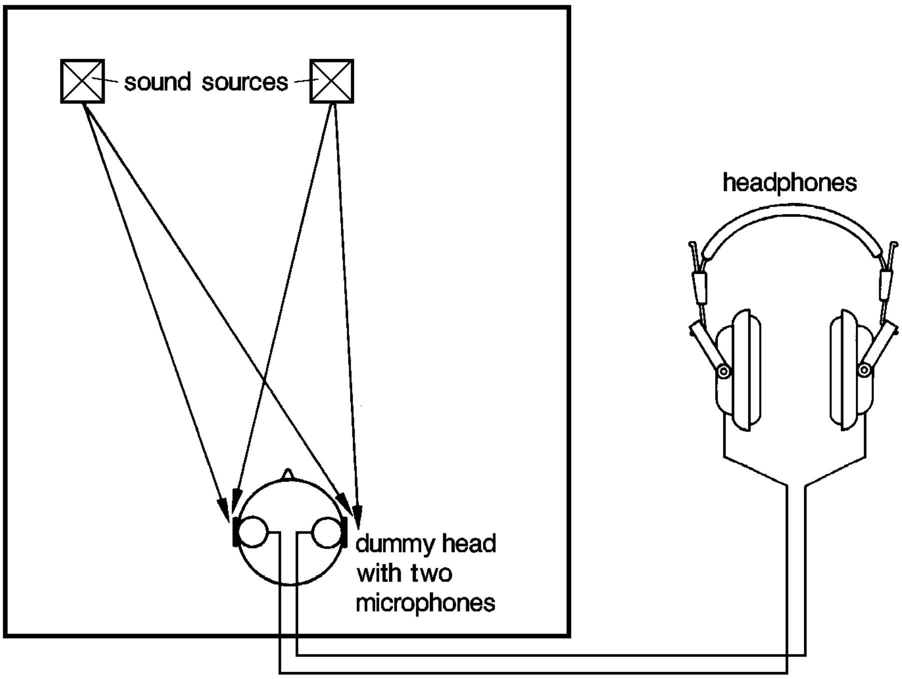


Fig. 2

Sennheiser has constructed another type of dummy head, using two miniature electret microphones at the ends of a support (in the form of a U) so that they can be hung into the external ears of a human head or of a dummy head, just in front of the eardrums. Thus the recorded sound is the one entering the auditory canal (Fig. 3).

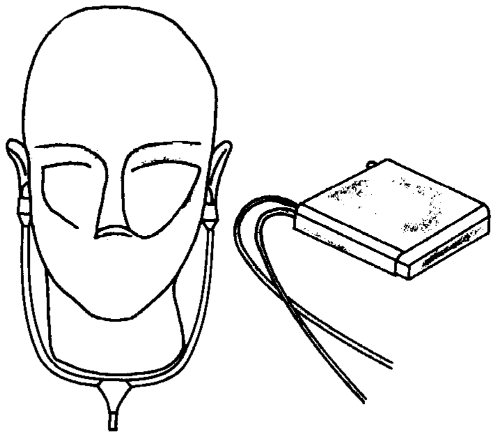


Fig. 3

Sennheiser thinks that by placing the microphones at the level of the eardrum, the sound information has to pass the auditory canal once during recording and a second time during listening via headphones. That is one time to many; so the sound has to be picked up at the entry of the auditory canal.

The microphones with their special support are thus separated from the head. One can wear them also oneself during the recording, what by no means is very practical.

The microphones used are omnidirectional with a hump of 3 dB at about 10.000 Hz to compensate the high frequency attenuation of the head.

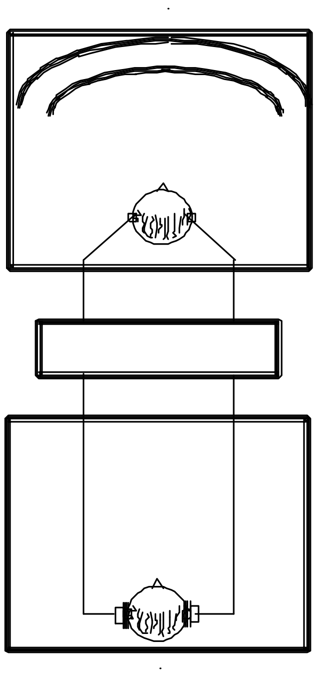


Fig. 4

Dummy head Recording

(transmission)

2 Room-related stereophony

Room-related stereophony tries to reproduce the sound field of the recording room with loudspeakers in the replay room.

In principle this works better the more transmission or recording channels are used and the less influence the replay room has on the sound picture. Initially the limitation of the transmission to only two channels has been made for economic reasons, since additional expense does not bring adequate gain in the reproduction of the sound picture.

As in the case of natural listening, sound localization is realized on the basis of intensity and/or delay-time differences between both channels (see Sound perception 3.1 and 3.2).

Whether intensity or delay-time stereophony or a combination of both is used will be determined solely by the recording technique, not by the listening arrangement. With a suitable loudspeaker arrangement, room-related stereophony produces a sound pattern stationary in the room. The sound picture will be optimally reproduced only inside the stereo listening area (see Sound perception 4).

There are different recording techniques for the room-related stereophony:

Intensity stereophony:	single microphone technique
	XY-coincident microphone technique
	MS-coincident microphone technique
Delay time stereophony:	AB – technique

Due to the arrangement and the directional characteristics of the microphones used for intensity stereophony, the stereophonic sound impression is mainly determined by intensity differences or level differences between the stereo signals L (left) and R (right).

Three microphone techniques can be used for intensity stereophony:

1. several mono microphones, whose signals at the replay side will appear along the "stereo basis"
2. one or several coincident-microphones *) switched on XY - technique
3. one or several coincident-microphones *) switched on MS – technique

Coincident microphones unite two independent microphone systems, one on top of the other.

With AB - stereophony the stereophonic sound impression is determined due to delay time differences between the signals of two mono microphones, set up at a certain distance from each other.

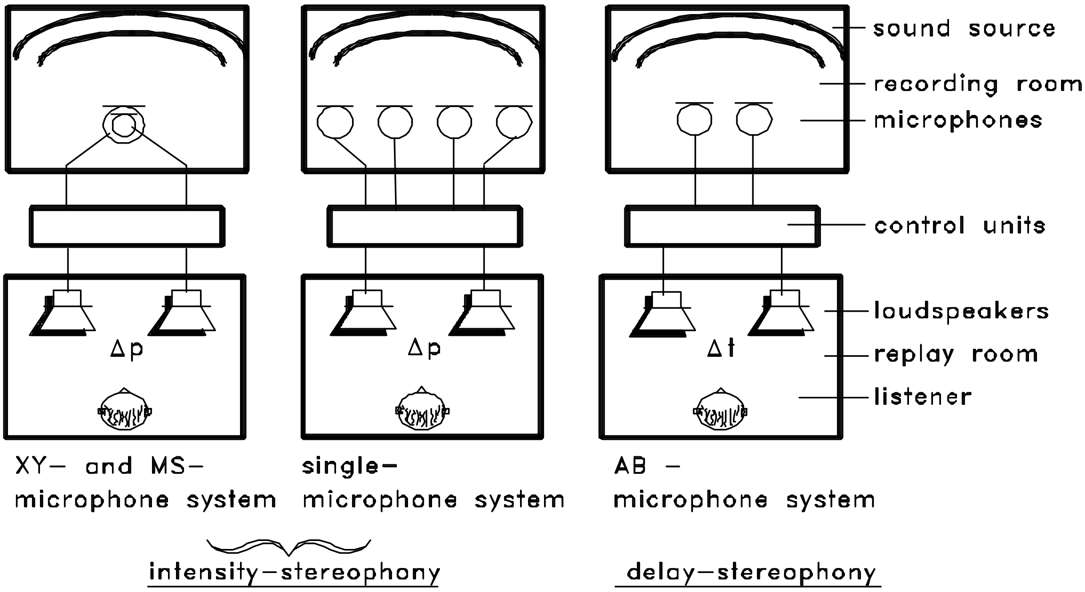


Fig. 5 *Stereo-Recording Systems*

3 Level suppression due to delay differences

If two microphones have different distances from a sound source, one will find a level increase or decrease if both microphone signals are summed up for certain distance differences and frequencies:

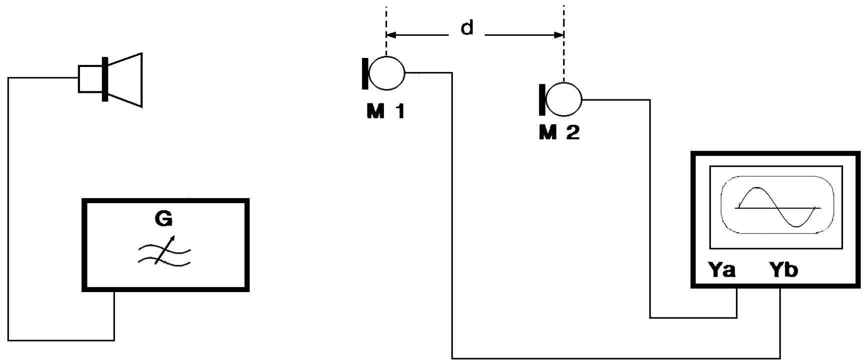


Fig. 6

Summation of two microphone signals for different distance-differences and constant frequencies:

d	0	8.5	17	25,5	34	42,5	51	cm
Σu	200	40	160	80	130	90	108	%

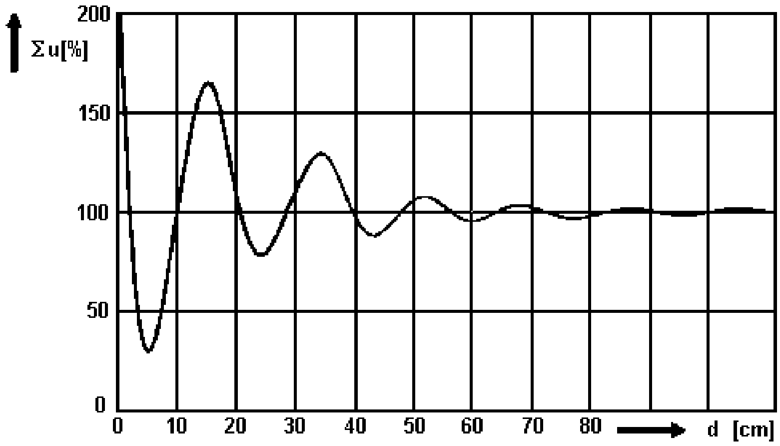


Fig. 7

Level suppression: At distance differences λ/2 and the odd multiples
Level increase: At distance differences 0, λ and the multiples of λ

4 Stereo Signals

Stereophony intends to give the listener a life-like impression of a sound event. This requires the transmission of at least two information signals.

With commonly used intensity stereophony replay happens with the help of two separate loudspeakers. The "L-signal" is led to the left loudspeaker and the "R-signal". to the right one. The directional impression depends on the intensity relation between the two signals. In principle tere are the following voltage relations in both channels during replay of stereophonic impressions:

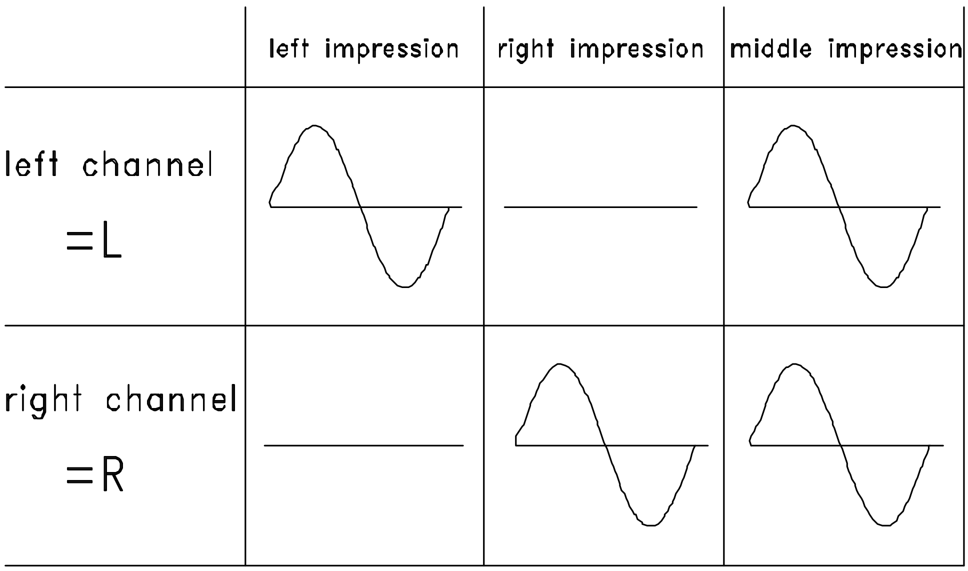


Fig. 8

To get the "middle" impression, the signals L and R have to be identical. If the voltage in the L-channel decreases andincreases in the R-channel, the sound source seems to move more and more to the right side.

If stereophonic productions are broadcast by a transmitter, it must be guarantied that both the stereo-listener and the mono-listener receive a sound impression artistically seen satisfactory. (compatible system).

For example, if the mono-listener receives only the left channel, he lacks in the information from the right channel and vice versa. Consequently, the mono-listener needs both information, L+R. Such a signal is called mono or M signal.

$$\mathbf{M = L + R}$$

In Fig. 9 we see the M signal in its correct and wrong polarity:

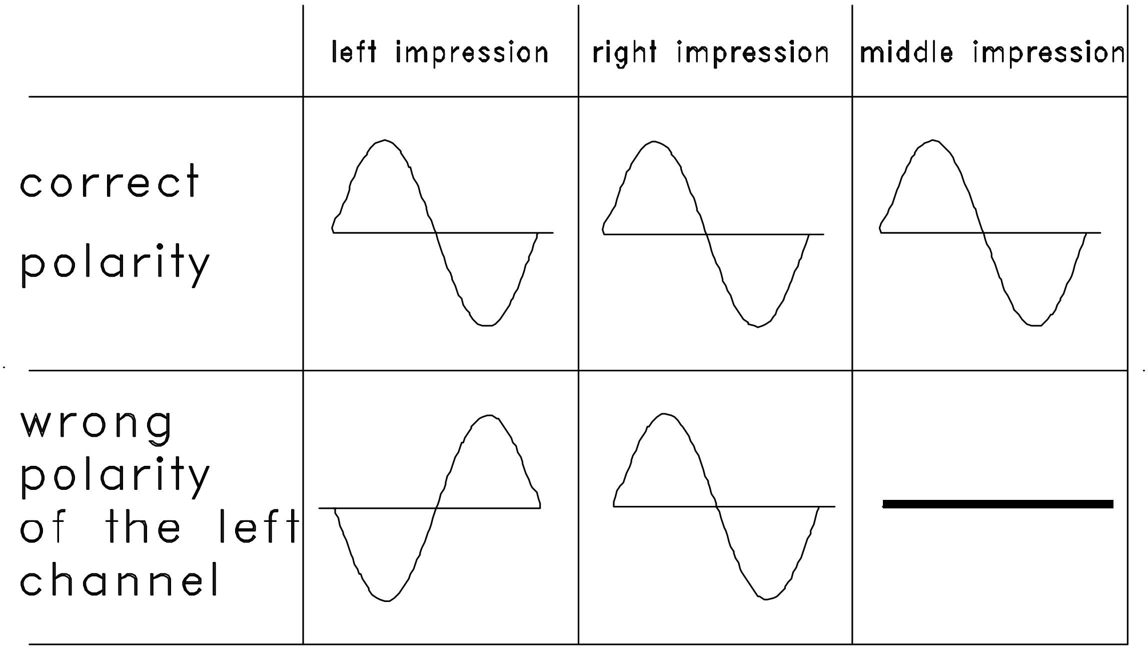


Fig. 9

l) The wrong polarity of the M signal during a transmission of the left impression will not be noticed by the listener.
Also a change of the wires – a - and - b - in the mono-technique cannot be noticed.

2) During the transmission of a middle impression, the M signal has the desired amplitude if the polarity is correct.

If there is a wrong polarity in one channel the M-signal becomes 0.
This means complete suppression of information for the mono-listener.

A wrong polarity of the R channel produces a negative R signal.
Consequently:

$$\boxed{M = L + (-R) = L - R}$$

This means that the amplitude of the M signal is too small, if not 0. The M signal is no longer the sum, but the difference of both. But the difference is smaller then the sum (if R is different from 0).

For stereophonic reproduction the M signal alone is insufficient because the two loudspeakers need the signals L and R. But M is the sum of both.

Besides the M-signal a broadcasting transmitter needs a further information, the so-called S signal (side signal).
From both signals M and S, the signals L and R can be reconverted:

$$\boxed{S = L - R}$$

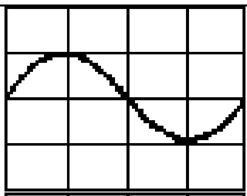
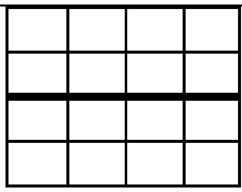
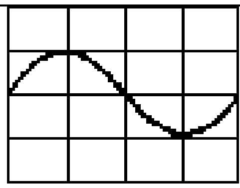
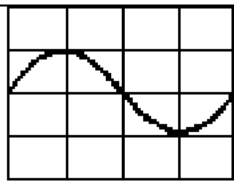
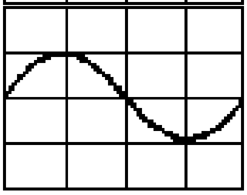
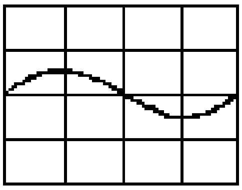
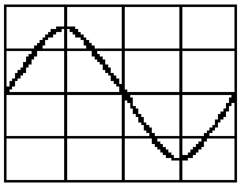
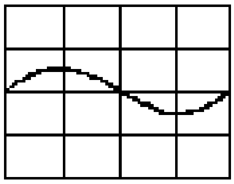
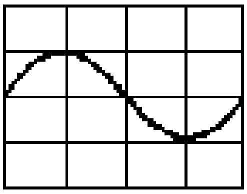
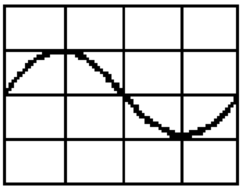
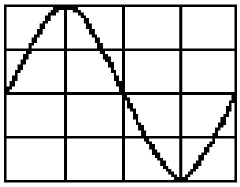
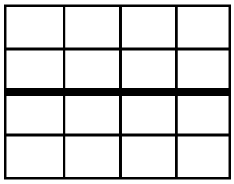
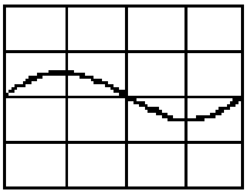
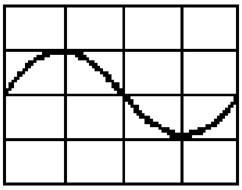
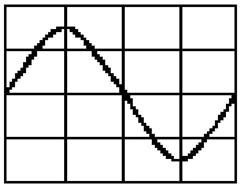
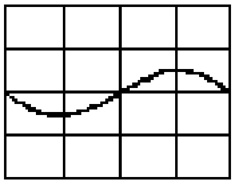
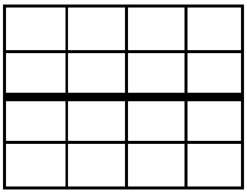
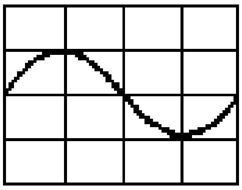
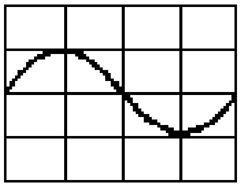
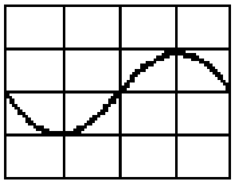
$$M + S = (L + R) + (L - R) = 2L$$
$$M – S = (L + R) - (L - R) = 2R$$

A broadcasting transmitter transmits M and S.
Stereophonic replay occurs at L and R.

During a monophonic replay a wrong polarity in the L or R channel causes a bigger or smaller decrease of volume. It can lead to a complete suppression of a sound source.

With stereophonic replay a wrong polarity (for example in the right channel) leads to a signal -R instead of +R in the loudspeaker. This will not lead to volume (level) differences, but a localization of different sound sources is no longer possible; the result is a sound impression which is completely different compared to the original sound event.

4.1 Stereo signals summary

Direction	**L**	**R**	**M**	**S**
left				
half-left				
middle				
half-right				
right				

S-Signal:

Amplitude	S ="0"	Localisation in centre spot
Amplitude	S = M	Localisation at a side
Polarity	+	Localisation left
Polarity	-	Localisation right

M-Signal:	Left	100%
(from sound source	Right	100%
with full level)	Middle	200% (not compatible)

5 Intensity stereophony

In case of intensity stereophony the stereo signals L and R have different levels; delay and phase differences are unwanted recording or transmission faults, especially in respect to the demands of compatibility.

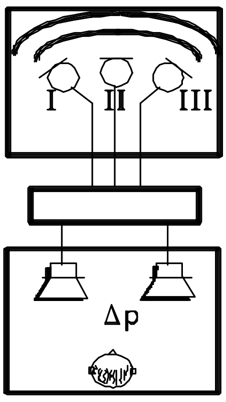
5.1 Single microphone technique

The *single microphone technique* uses several, mainly directive, mono microphones. In the control room (with the help of basis and direction control) the respective signals are composed to the final stereophonic sound impression. Microphone I for example is exclusively assigned to the left channel, microphone III to the right channel, whereas the signal of microphone II as Middle signal is equally distributed to both channels. A basic condition for the *single microphone technique* to function perfectly is a high degree of acoustical separation of the different microphones. Two single microphones can be considered to be sufficiently separated if their cross talk attenuation is bigger than 15 dB. This separation can be improved with the help of sound absorbing barriers or bunks, or by skillfully placed sound sources.

The *single microphone technique* can also be used together with other microphone techniques. It is especially favorable

a) if the desired arrangement of the sound sources in the stereophonic impression does not correspond to the actual arrangement in the recording room.
b) if the different sound sources differ widely in their natural loudness.
c) if the signals of various sound sources have to be differently manipulated by the recording engineer (use of different filters, equalizers, delays, reverberation).

The use of many microphones is also called polymicrophony.



SINGLE-MICROPHONE TECHNIQUE

Fig. 10

5.2 XY-microphone technique

The XY-microphone technique uses a *stereo coincident microphone*, whose two microphone systems are set to the same directional characteristic, either cardioid or bi-directional. The main sensitivity axes of the two systems are symmetrical to the fictitious central axis, which is directed to the center of the sound source (orchestra). The respective main sensitivity axes form together with the central axis the angle of displacement [Φ]. The angle of displacement depends on the necessary recording or acceptance angle [2Φ]. The recording or acceptance angle is the angle within which effective, definite stereophonic resolution is possible. It thus includes the recording range, which during playback is represented between the two loudspeakers.
The angle of displacement influences the recording angle (recording range):

The recording range increases with decreasing angle of displacement.
The sum M of the stereo signals L (left channel) and R (right channel) will be a perfect mono signal.

5.3 MS-microphone technique

Like the XY-microphone technique, the MS-microphone technique uses a coincident microphone. In this connection the one system (S system) has to be set to the bi-directional characteristic and to an displacement angle of 90°: the other system (M system) can be set to the omnidirectional, bi-directional or cardioid characteristic, but the angle of displacement has to be 0°.
The two systems do not immediately deliver the signals L (left channel) and R (right channel), but the signals M and S (Middle component and Side component signal, or sum and difference signal), which after reconversion deliver the L and R signals.

$$L = M + S$$
$$R = M - S$$

The M signal is the immediate mono signal which, in contrast to the XY-technique, is delivered from a single microphone.

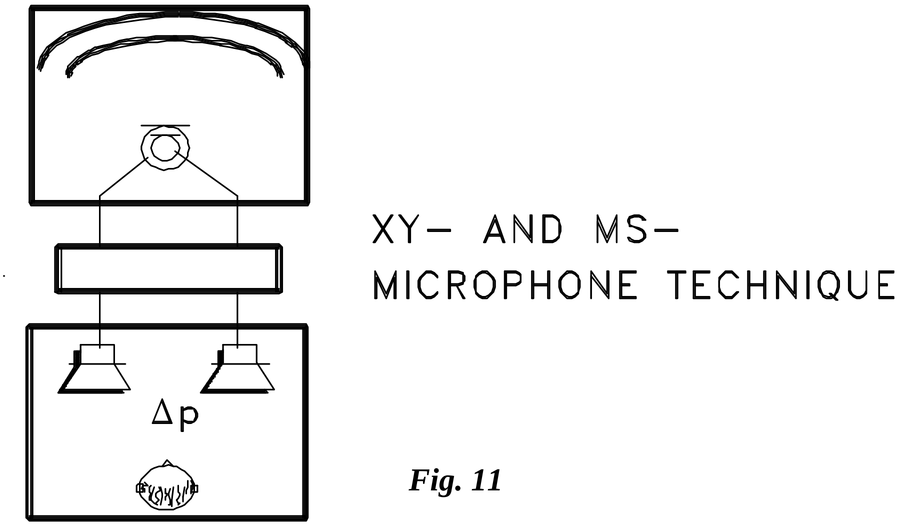


Fig. 11

6 Equivalent directional characteristics combinations

Physically seen, the XY and MS recording techniques are equivalent under certain circumstances. As we have already seen, there is a simple relation between the signals L, R, M and S; they can be converted into each other. this conversion is also called stereo conversion.

$M = L + R \quad L = M + S$

$S = L - R \quad R = M - S$

Since M and S can be converted into L and R in the same way as L and R into M and S, both conversions can be done with the same device (sum-difference transformer or differential transformer).

Due to this equivalence, certain XY-combinations correspond theoretically to certain MS-combinations. But in practice one will find some differences between equivalent XY-MS combinations. Since the microphone voltages are frequency dependant, all MS-combinations reproduce the higher frequencies better in the range of the symmetry axis for example, the XY-combinations better in the lateral ranges.

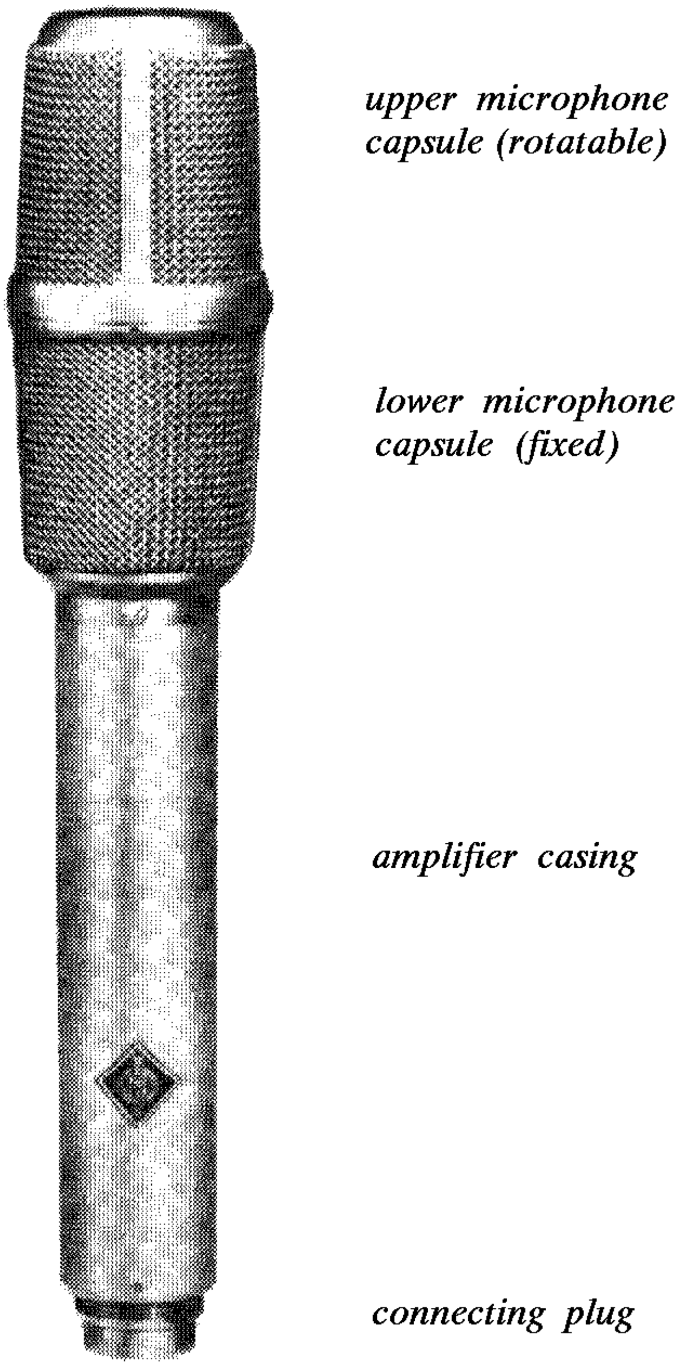


Fig. 12

Coincident microphone

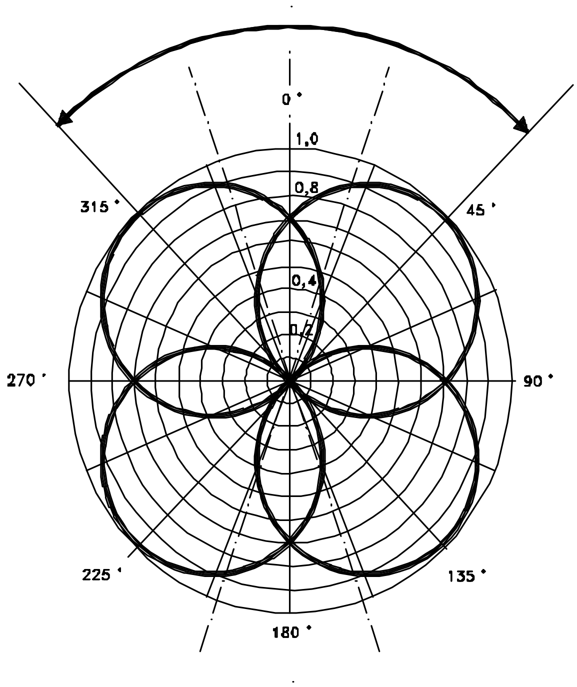


Fig. 13
Coincident microphone

<u>Definition of terms for upright standing coincident microphones</u>

1. fixed capsule System I (below)	channel I	delivers in XY-Technique: L signal delivers in MS-Technique: M signal
2. revolving capsule System II (above)	channel II	delivers in XY-Technique: R signal delivers in MS-Technique: S signal
3. angle of displacement:		angle at which the main sensitivity directions of the microphone capsules are turned away from the symmetry axix (mechanically) <u>Adjustment</u>: XY-combination: freely selectable MS-combination: M = 0° ; S = 90°
4. recording range:		range that at replay will be represented inside the “Stereo Basis”
5. recording angle:		angle that includes the recording range

7 Use of coincident microphones in practice

7.1 Representation of sound sources on the stereo basis

The directions of sound incidence and distances of sound sources are picked up by the coincident microphone in a polar plane, which means that only sound sources located on a circle around the microphone will be represented during playback along a straight line (basis line) between the two loudspeakers, whereas sound sources that are arranged along a straight line in front of the coincident microphone will emerge from the basis line:

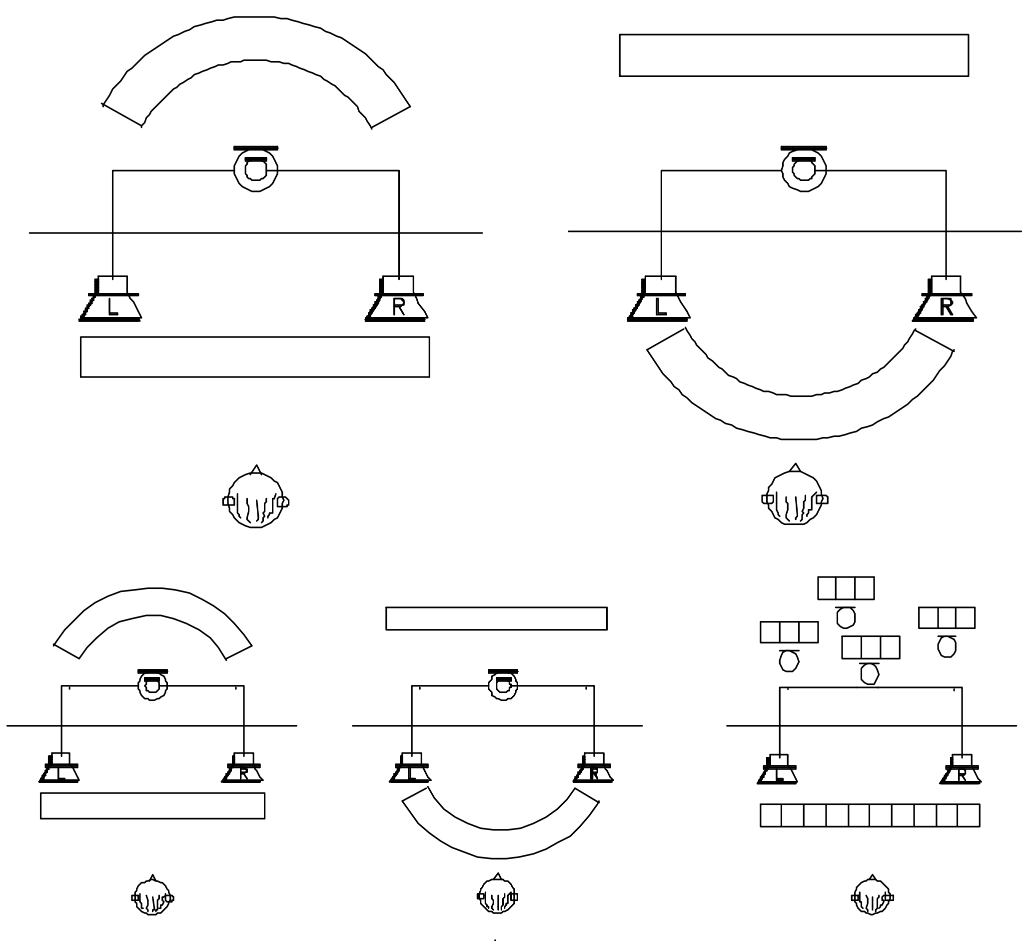


Fig. 14

7.2 Orientation of the coincident microphone

As we have already seen, coincident microphones consist of two single microphones, (mounted on a common axis), that can be turned individully. Their directional characteristic is variable. The fixed (lower) single microphone or capsule delivers the X- or M-signal; for a recording in the XY-system it is turned to left, in the MS-system to the center (or most important part) of the sound source. The revolving (upper) capsule delivers the Y- or S-signal; for a recording in the XY-system it is turned right, in the MS-system to the left at an angle of 90° to the axis of the M-signal.

If the coincident microphone is suspended upside down, the M-signal remains unchanged, whereas after sum-difference conversion the L and R signals would be laterally inverted if the S-capsule is not turned by 180°. For an upside down XY-arrangement, the fixed (now upper) capsule has to be turned to the left and the revolving (now lower) capsule has to be turned to the right.

In the XY-system, the angle of displacement [δ] has to be selected according to the desired recording range (Chapter 6). The capsules are symmetrical to the axis that faces to the center of the sound source.

In the MS-system, the M-capsule faces towards the center of the sound source and the S-capsule (*always set to bi-directional*) at a right angle (90°) to the M-capsule so that its positive side (in phase with M) is facing left.

Delay time differences between the two systems of a coincident microphone and the therefore resulting phase differences between the stereo signals falsify the sound picture: with the MS-recording system the stereophonic sound picture and at the XY-recording system the stereophonic **and** monophonic ones. This can only be avoided if sound hits the common axis of rotation of the two microphones exactly vertically:

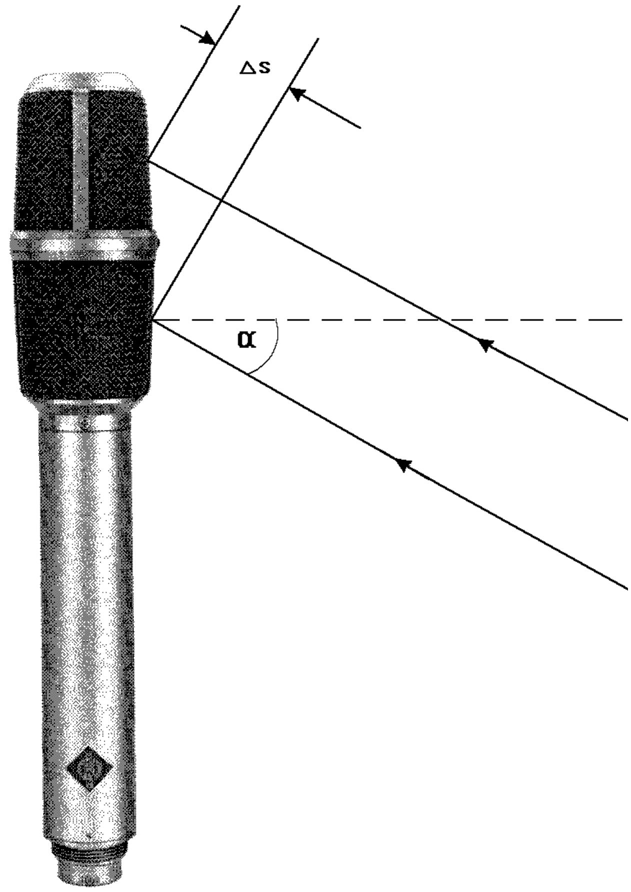


Fig 15

At a slanting sound incidence, there is a distance difference Δs, or a delay time and phase difference, which causes a frequency dependent levelreduction of the mono signal. If the deviation is about 15° (angle α in Fig. 15), and the distance between the middle points of the capsules is about 40 mm, frequencies around 15.000 Hz are cancelled.

With stereophonic replay, delay time differences between the two XY-capsules will have a negative effect only in extreme situations; they will reduce the accuracy of directional perception. A phase difference between M and S of less than 90° will cause, after conversion into L and R, a reduction of cross talk attenuation, which means a displacement of sound sources to the middle. At 90° phase shift a total cross talk arises, which means reproduction occurs only in the middle (mono); at 180°, L and R are reversed. This can lead to the effect that lateral sound sources that are arranged vertically will be represented horizontally during playback: e.g.the sound of footsteps beneath a person's voice.

7.3 Practical limitations of XY/MS equivalencies

In theory, the MS and XY microphone systems are equivalent, and their signals can be converted into each other by the sum-difference-transformation, but practical experience has shown some restrictions. The main reason for these restrictions is the deviation of the real polar diagrams from the mathematically defined ideal forms.

Fig. 16 shows the frequency dependency of the directional patterns of a coincident capacitor microphone with variable directional patterns:

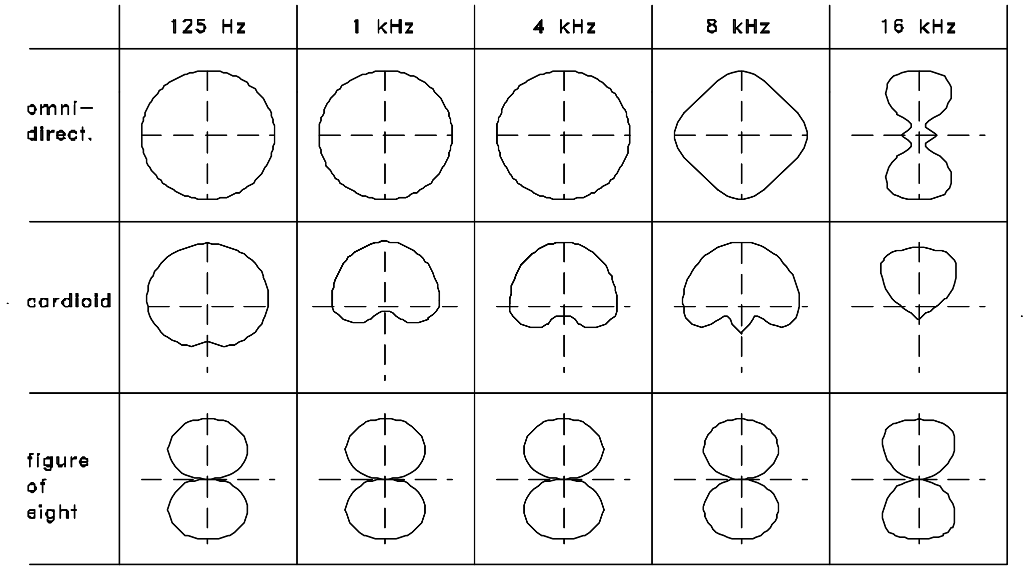


Fig. 16

Between 1 and 4 kHz, the real directional patterns correspond very well to the ideal form. At lower frequencies, the cardioid mike approaches the omnidirectional pattern; at higher frequencies the omnidirectional mike becomes less sensitive at its sides and thus approaches the hyper cardioid characteristic; the figure-of-8 characteristic is the least frequency dependent one.

In principle, the deviations of the real patterns from the ideal ones in both the MS and XY systems will cause a frequency dependent displacement of sound sources during replay. With the help of Fig. 13 these influences can easily be explained. For instance: at lower frequencies the MS combination according to Fig. 13, 1c changes into the combination of 1b, which means an increase of the recording range; at higher frequencies the MS combination according to Fig. 13, 1b changes into the combination of 1d, which means a decrease in recording range.

Due to the frequency dependency of the microphone signals, all MS combinations reproduce the higher frequencies better in the range of symmetry axis, all XY combinations better from the sides.

The equivalent mono-signal (Fig. 13), also delivers information about how diffuse sound (reverberation, noise of the audience) will be recorded during a stereophonic recording.

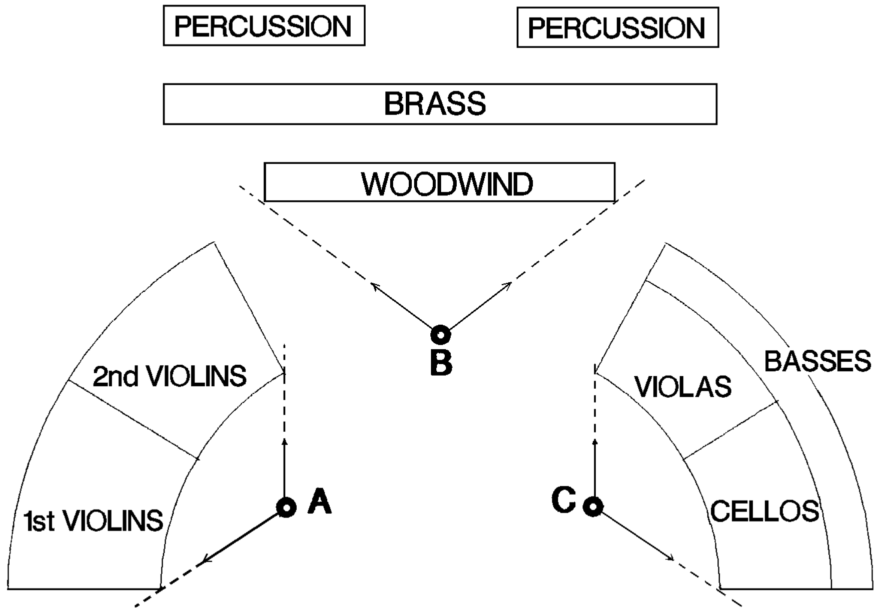
Despite the theoretical equivalence of XY and MS systems, in practice the MS system offers some advantages:

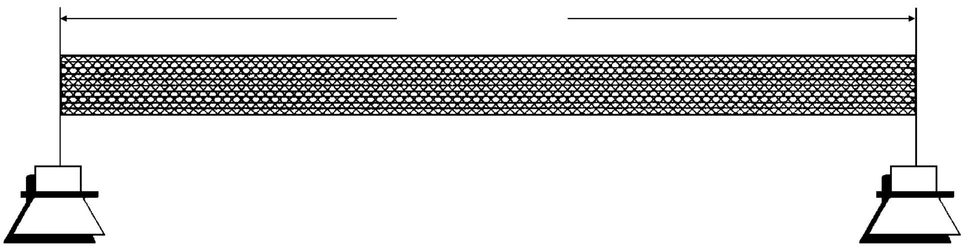
- with remote-controllable settings of the directional patterns, all necessary settings can be remote-controlled in the MS system, with the XY system the angle of displacement has to be set at the microphone itself

- in the MS system, the M mike is always directed towards the center of the sound event, at the XY system the two mikes are directed towards the rims of the sound event. Thus the brilliance of a recording is reproduced better with the MS system.

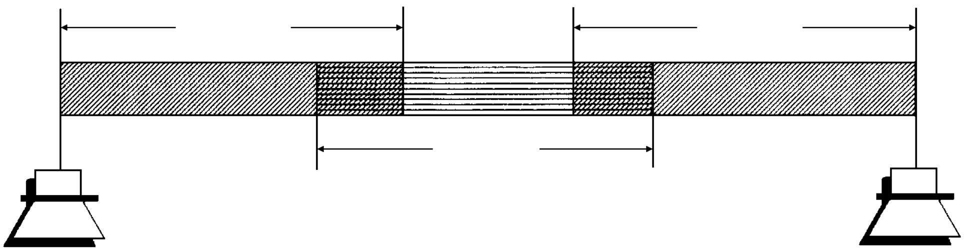
- a practice-oriented microphone arrangement with a not too big recording range and a sufficient suppression of diffuse sound (Fig. 13, 1c and 2c) will not be satisfactory in the XY system since the necessary hyper cardioid characteristic is normally very frequency dependent or is not available.

8 Mixing of several stereo plus support microphones





REPRODUCTION WITHOUT DIRECTION- AND BASIS CONTROL



REPRODUCTION WITH DIRECTION- AND BASIS CONTROL

Fig. 17

8.1 Use of support microphones

Recordings of larger sound sources, like for orchestras, choirs, etc., involve high demands on the transparency and brilliance of sound, on an even utilization of the whole stereo basis, on a good and high directional resolution and on the reproduced room illusion.
To realize the desired sound impression for the listeners it is very often necessary, to set up several main stereo microphones, or tu use additional support mikes, especially in rooms with unsatisfactory acoustics. Mono or stereo microphones can be used as support microphones.
It is advisable to limit the number of support microphones to as few as necessary.

Mono support microphones

Mono support microphones are suitable for sound sources without audible spatial extent, like single instruments or soloists. The mono signal has to be placed in the desired direction with the help of a panorama-potentiometer (pan-pot) or direction-mixer.
The microphone must be acoustically separated from other te other microphones, to avoid a double reproduction, falsification of directional information or ambiguity. If sufficient separation cannot be realized, then, with the help of a pan-pot or direction mixer, the sound source has to be placed into the direction that corresponds to its real position in the recording room.

Stereo support microphones

If the support microphone has to include a sound source of a certain spatial extent (group of instruments or choir, a piano, organ or harpsichord), an additional stereo microphone will be suitable. Yet its directional allocation has to coincide with that of the main microphone. Correspondingly its basis-width and reproduction direction have to be properly set with the help of the direction mixer.
A basic problem of a stereo support microphone is its poor suppression of sound sources in its back and the diffuse reproduction of sound sources that are outside its recording range. Therefore, in most cases, the use of several mono support microphones would be advisable.

8.2 Pan-pot and direction mixer (equivalent settings)

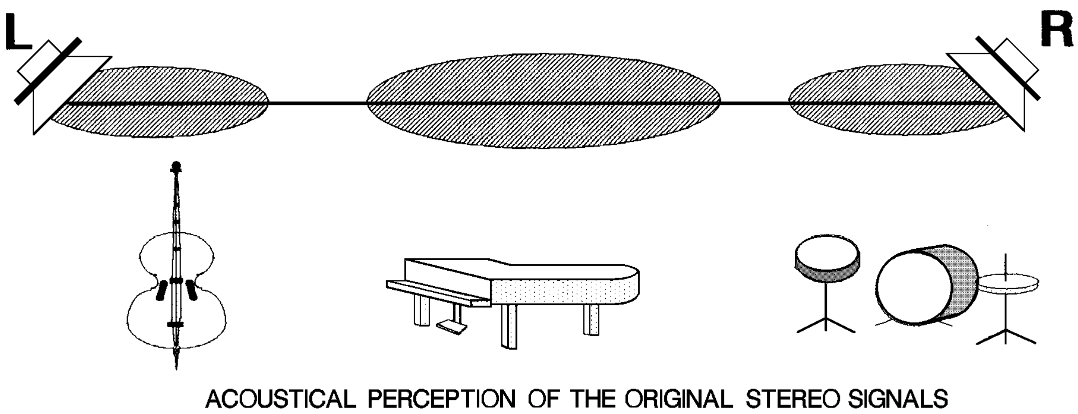


Fig. 18

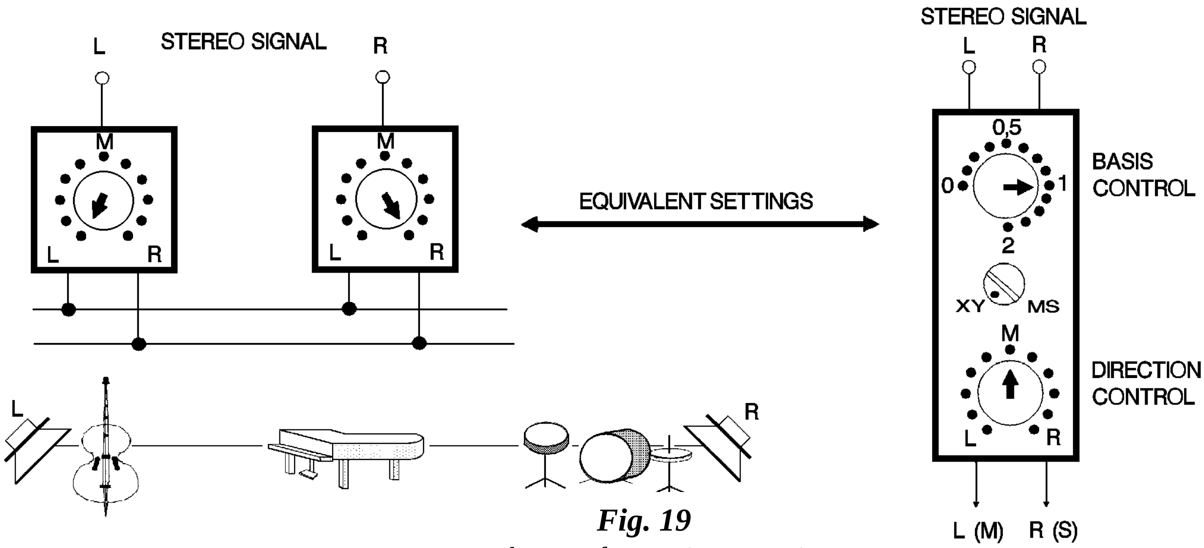


Fig. 19

No change of acoustic perception

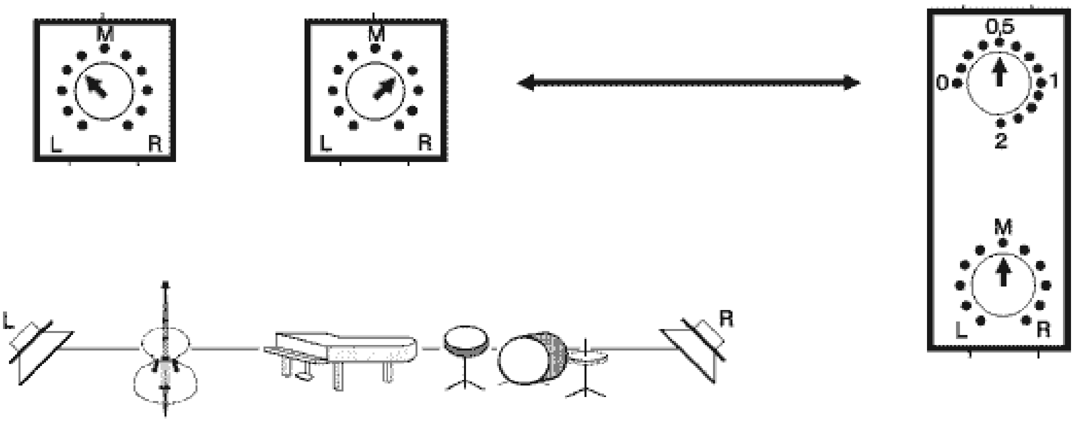


Fig. 20

Reduced Basis width

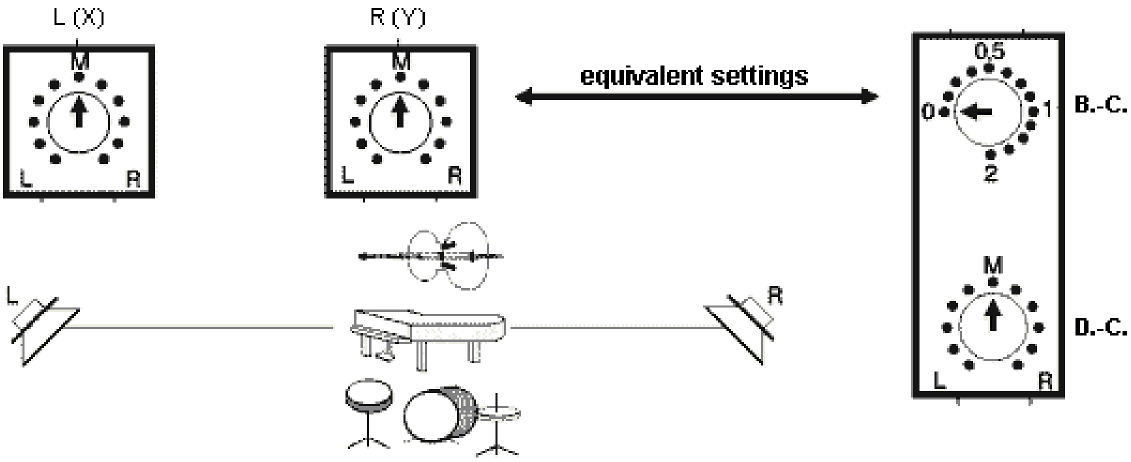


Fig. 21

Basis width=0 (Punctual Perception from the middle)

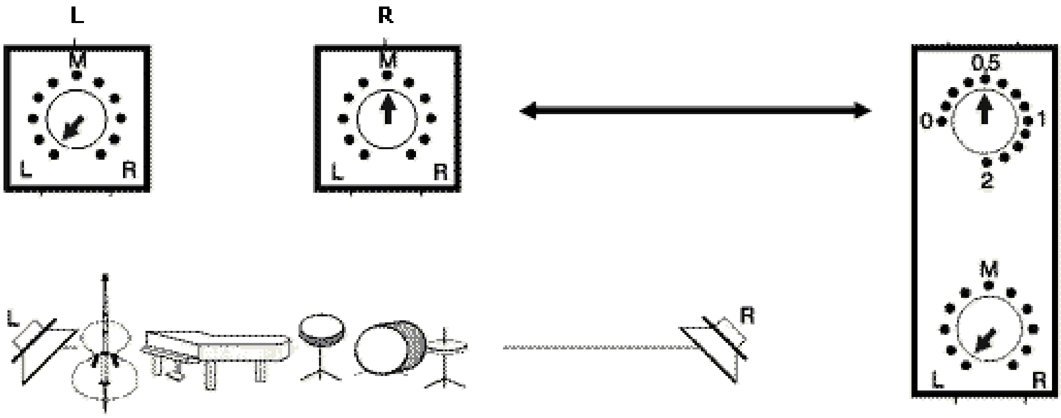


Fig. 22

Full Basis but reversed

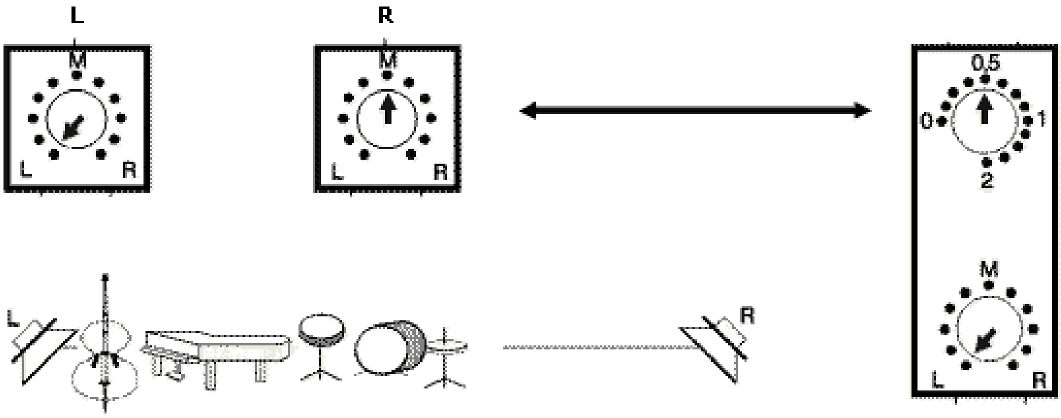


Fig. 23

Shifting to left-half left

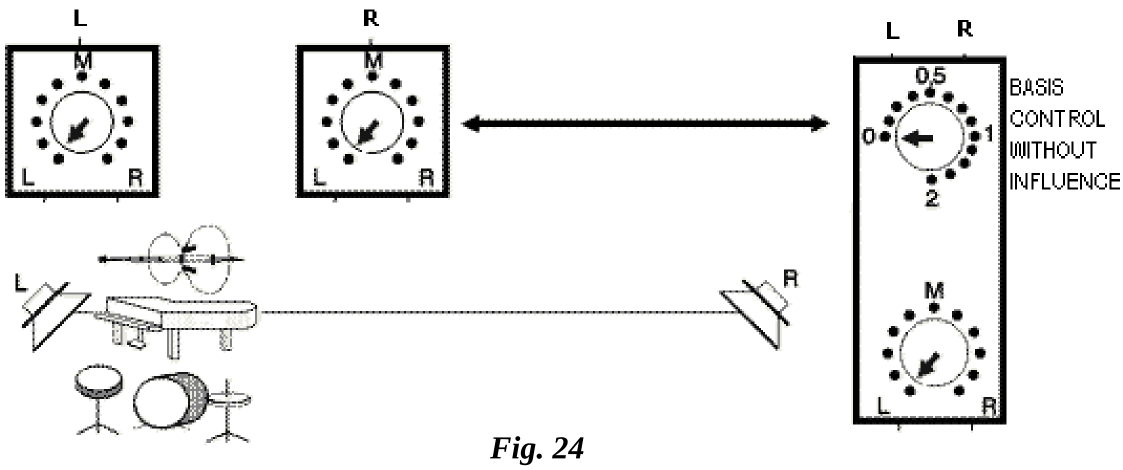


Fig. 24
Shifting to completely left

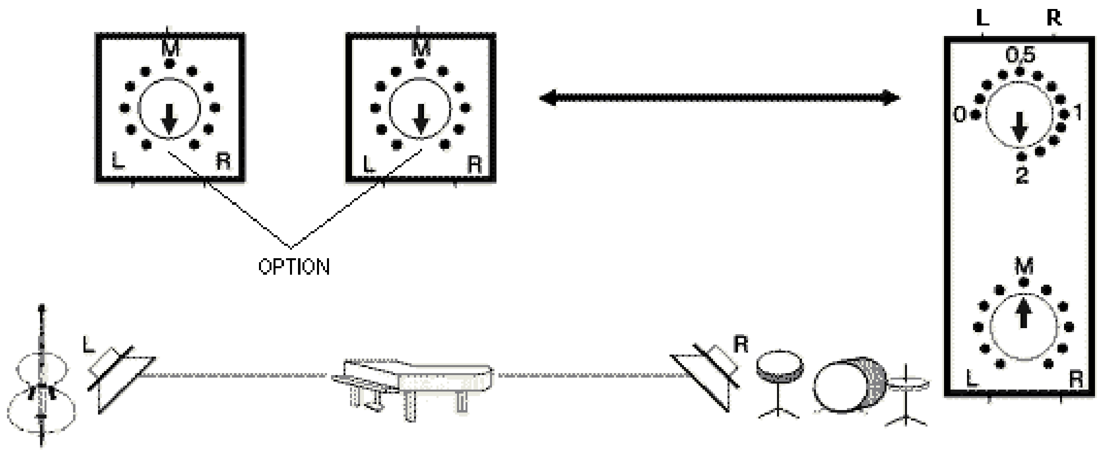


Fig. 25
Too big basis

8.3 Principle of a panorama potentiometer

The use of support microphones

As already mentioned above, mono support microphones are recommendable for the recording of sound sources without audible spatial extent, for example a single instrument or a soloist. The mono signal has to be integrated into the sound picture with the help of a panorama potentiometer (PAN-POT) or direction mixer.

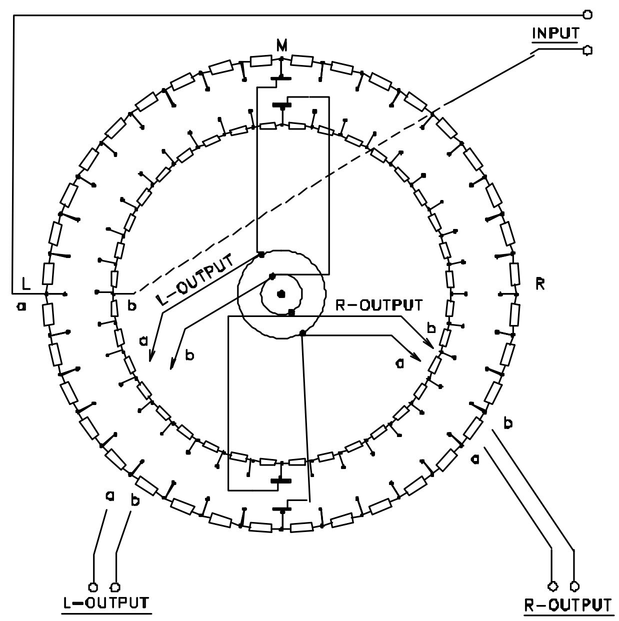
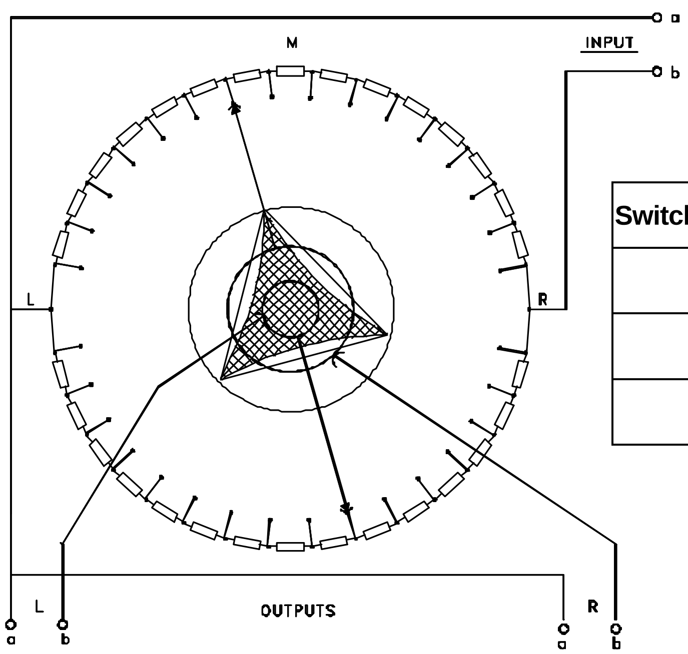


Fig. 26



Switch position	Output left	Output right
L	-10dBu	-∞dBu
M	15dBu	-15dBu
R	-∞dBu	-10dBu

Fig. 27

8.4 Practical use of stereo support microphones

There are four basic arrangements for the use of several stereo support microphones (coincident microphones):

1. In the so-called "staggered arrangement", additional coincident microphones are set up in the symmetry plane of the main microphone (Fig. 28). This requires an adjustment of the represented basis width of the support microphones to the representation of the main microphone in order to avoid a repeated representation of a sound source on the stereo basis. In this case, the basis width of the support microphone has to be set smaller than that of the main microphone due to it's shorter distance to the supported sound source.

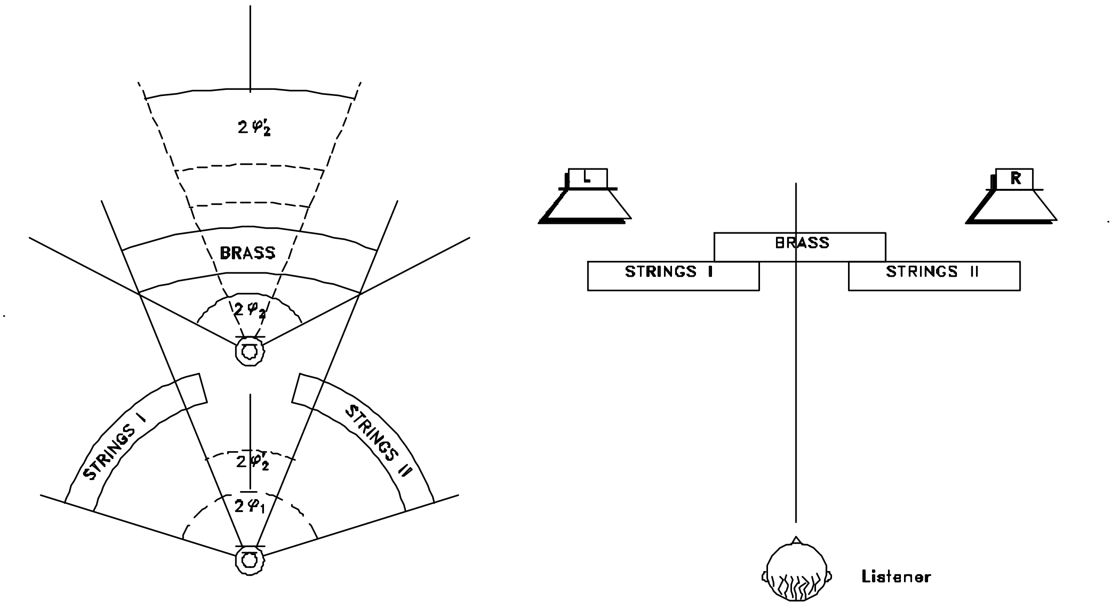


Fig. 28:

Coincident support microphone on the symmetry axis of the main microphone. 2 F_1 and 2 F_2 = opening angle of the microphones; 2 F'_2 = reproduction angle of support microphone, to be set with direction control.

2. One ore several coincident support microphones are set up outside the symmetry plane of the main microphone (Fig. 24). Main and support microphones have to be brought into correspondence as far as basis width and direction is concerned. As in the preceding example, one has to ensure a sufficient back attenuation of the support microphones, otherwise there will be repeated representations of certain sound sources.

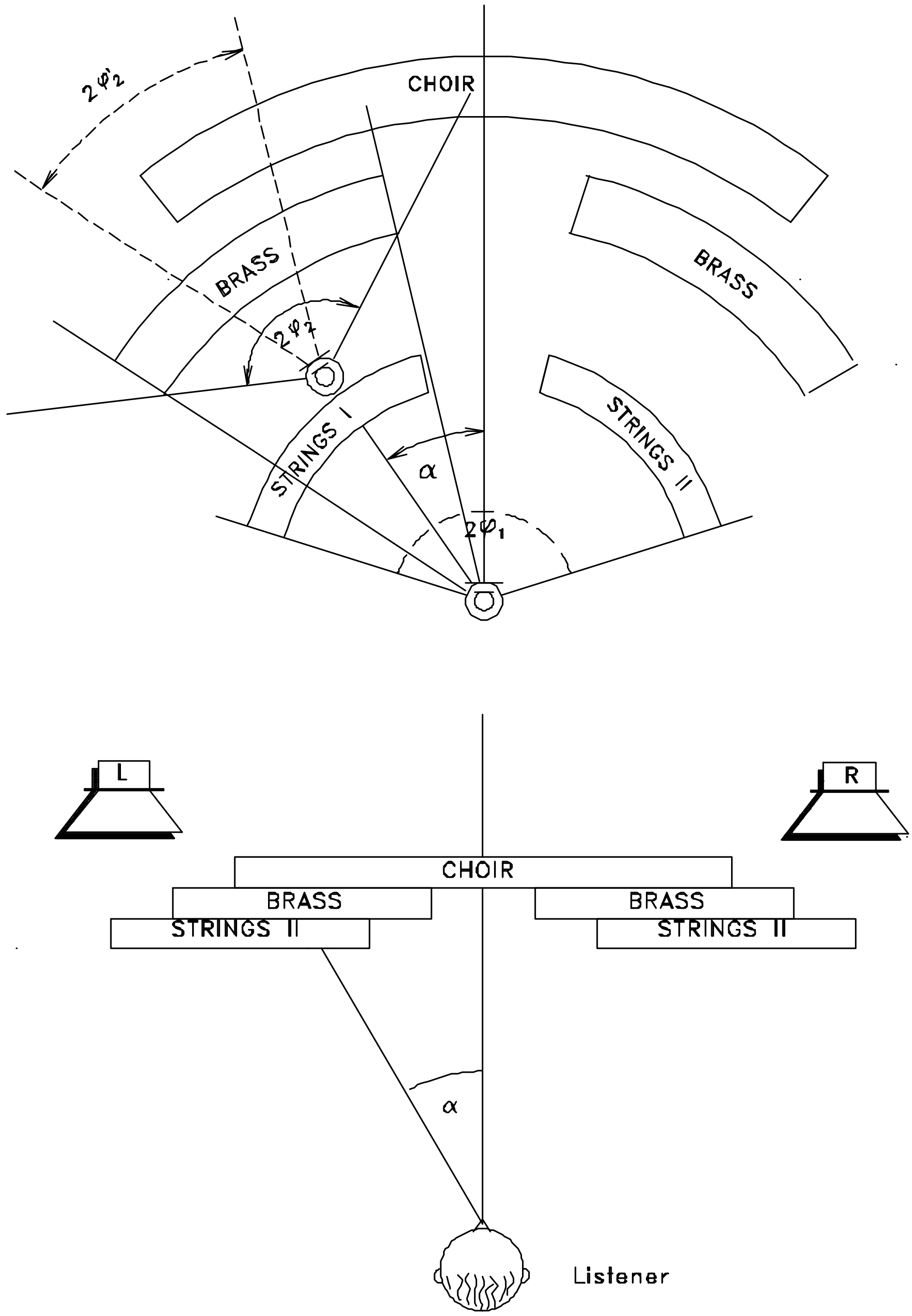


Fig. 29

Coincident support microphone outside the symmetry plane of main microphone. 2 F_1 and 2 F_2 = opening angle of the microphones, 2 F'_2 = reproduction angle (basis width) to be set with direction control, á = direction of support microphone to be set with direction control.

3. Sectorial use of several equal coincident microphones (Fig. 30). The orchestra is devised in sectors; each sector is recorded with it's own microphone; the microphone signals are mixed together respective of the corresponding basis and direction control.

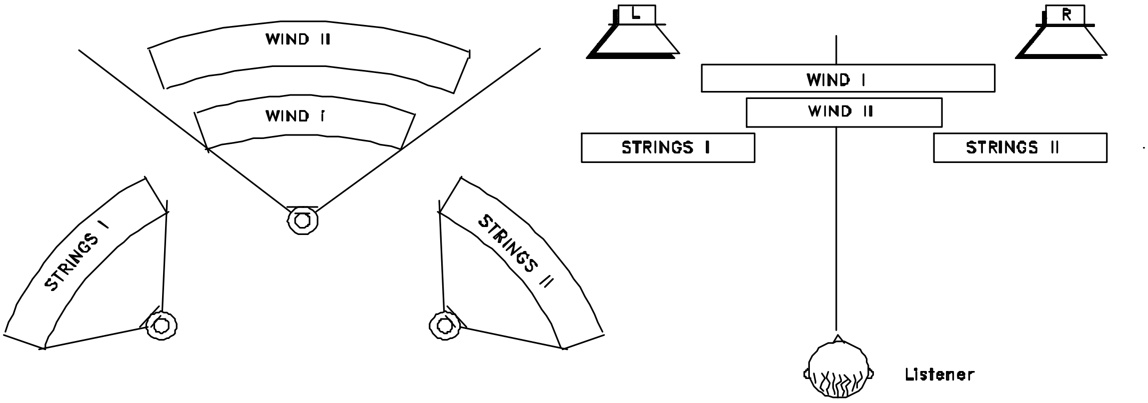


Fig. 30

Set-up of several equal coincident microphones in sectors.

4. Integration of separated sound sources into the total sound picture (Fig. 31): On the one hand there is often not enough room to arrange all partial sources in the recording range of the main microphone. On the other hand it is useful to arrange all sources around the conductor. The sound signal of a separated source is picked up with its own coincident microphone.

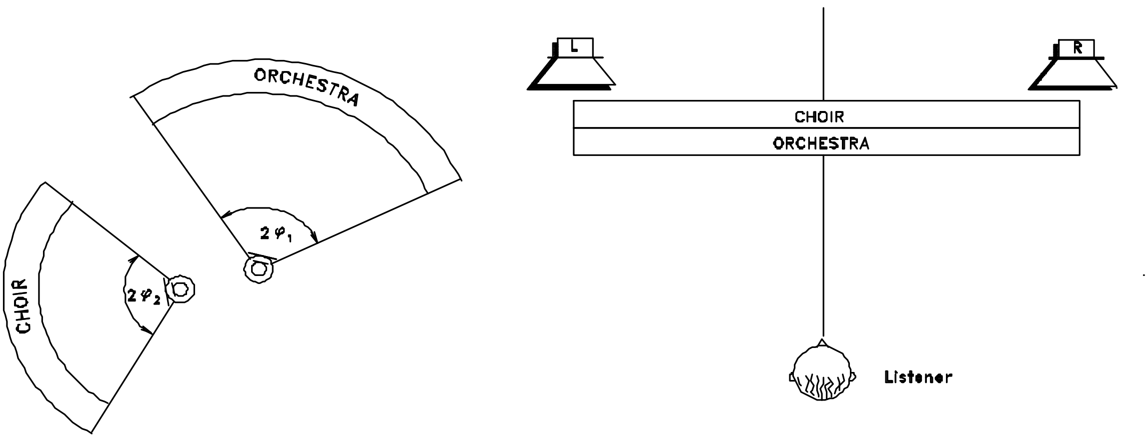


Fig. 31

Integration of separated sources, 2 F_1 and 2 F_2=opening angles of coincident microphones.

Common to all these applications is the fact that the better the microphones are separated acoustically, the better the basis and direction control can be handled. As far as the coincident microphone is concerned there is practically no effective possibility of suppressing sound arriving from the back in stereo recordings. Due to the better back attenuation of directive AB-support microphones, such arrangements have proved quite satisfactory.
XY combinations are preferable if large sound sources have to be recorded at a short distance (e.g. wood wind players of an orchestra), whereas MS combinations are better suited for soloist and smaller instrumental groups.

An example of a possible arrangement with several support microphones for a symphony orchestra is shown in Fig. 32. A stereo coincident microphone in XY-technique is used as the main microphone. Directive mono microphones (to be set with pan pots) are used in the other instrumental groups since they allow better acoustical separation.

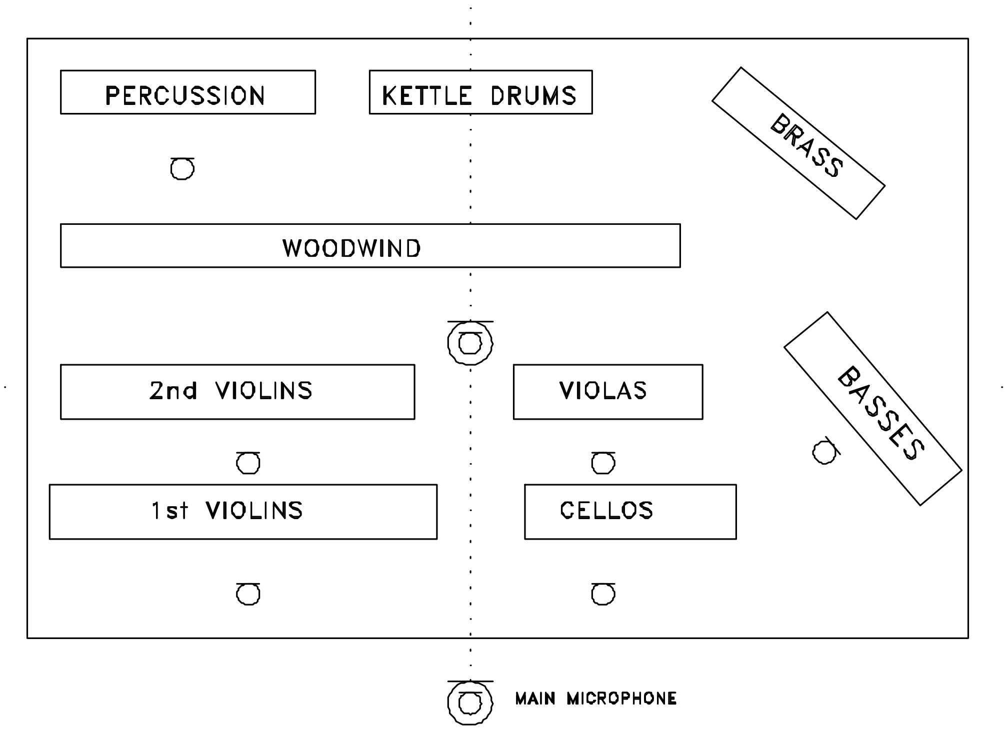


Fig. 32

Example of microphone arrangement with several support microphones for a symphony orchestra.

Orchestras with sound sources of very different loudness, like rock pop and jazz bands, require far more support microphones than classical symphony orchestras or chamber music groups. During recordings with support microphones for all partial sound sources of the orchestra, the main microphone often serves no purpose and can be dropped. These support microphones are direction and basis-controlled stereo microphones and direction-controlled mono microphones; they are set up in the near soundfield of the instruments and have to record the direct sound of the corresponding instruments, but they have also to be insensitive to the sound of other instruments and to indirect (diffuse) sound.

NOTE: In practice there are many more factors that influence the microphone set-up, (Acoustics of the recording room, arrangement of seats, problems with field of view, loudness balance, musical and artistic requirements, etc.), so that the above mentioned rules have to be adjusted for each individual case.

The main goal of a stereophonic recording is not to fulfil basic laws of recording techniques, but to realize the desired sound picture.

9. Delay time stereophony (A-B stereophony)

The recording system of delay time stereophony is called an AB-system. Two microphones are set up side by side (microphone basis) at a certain distance from the sound source. (Fig. 33)

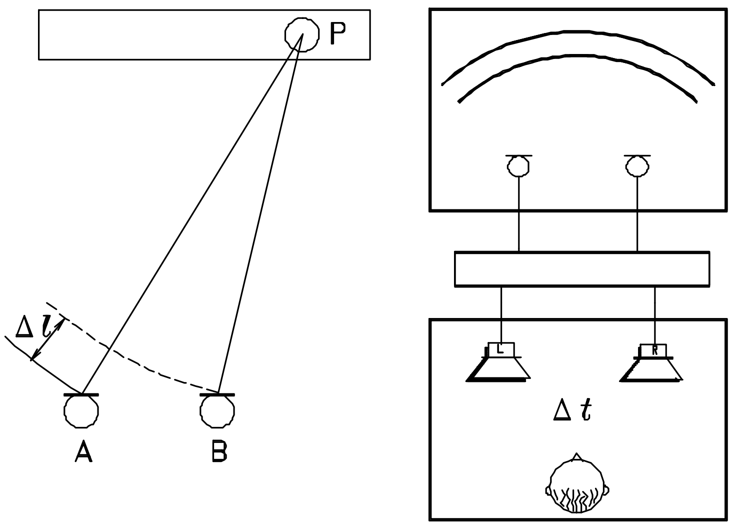


Fig. 33

Due to the distance difference Δl, resulting between a point P of the sound source and the two microphones A and B, delay time and phase differences occur between the microphone signals, or. between the loudspeaker signals at replay, due to which the stereophonic effect becomes possible. Delay time differences are always combined with more or less strong intensity differences. A and B are mono microphones with unidirectional bidirectional or omnidirectional characteristics; both microphones must have the same characteristic. They immediately reproduce the signals of the left and right channels, L and R. The superimposing of L and R to a mono signal can lead to sound distorting obliteration, which means that there is no fundamental guarantee of compatibility in the AB-system. Further disadvantages are poor directional resolution, the displacement of remote lateral sound sources toward the middle and the risk of a so-called "hole" in the middle. On the other hand it’s easier to realize big reproduction width and an impressive room illusion of the recording.
Due to compatibility problems, delay time stereophony is rarely used. So we will skip the practical use of Delay time stereophony.

10 Problems of level control - forming of the M-signal

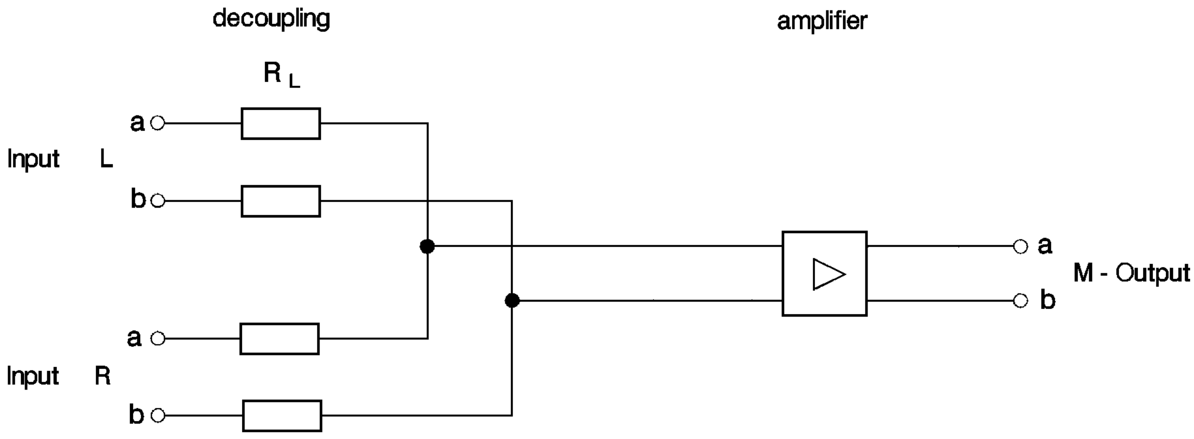


Fig. 34

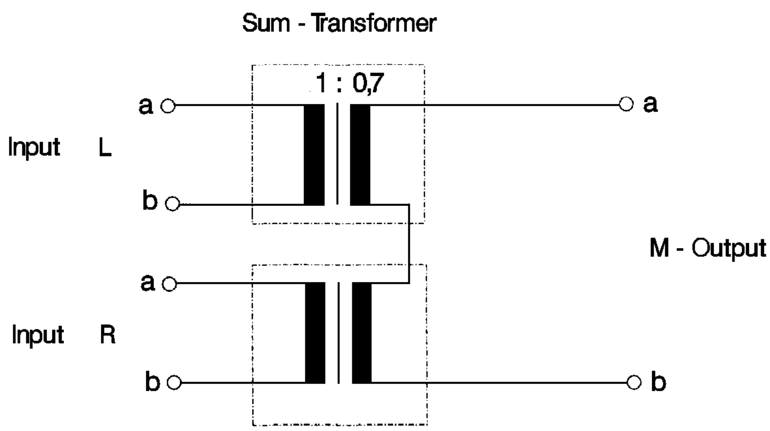


Fig. 35

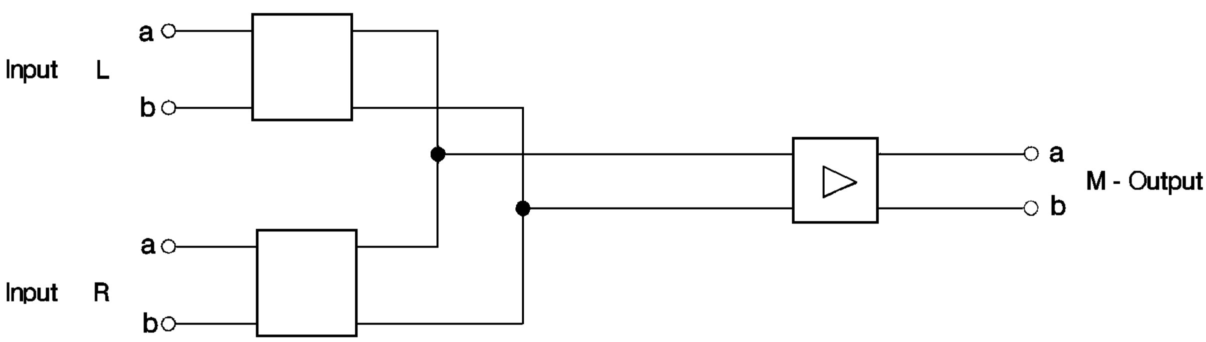


Fig. 36

11 Correlation

11.1 Correlation degree

In stereophony, correlation means the interrelationship between the signals of the left and right channels. The size of this relation is the correlation degree.

If we compare two sine oscillations of the same frequency and phase relation with each other, we see that both have the same tendency. (Fig.37):

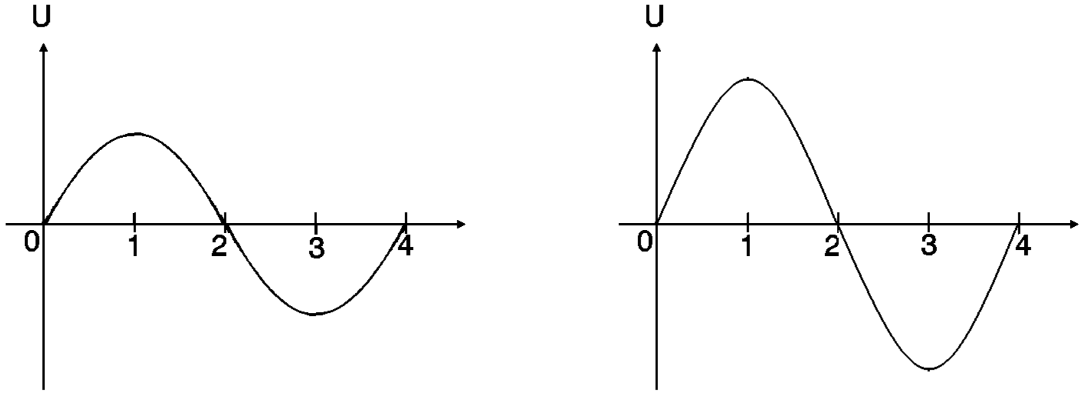


Fig. 37

During the period t = 0....1 both oscillations rise from 0 to their maximal positive value; during t = 1....2 both fall from their maximal positive value to 0; during t = 2....3 both fall from 0 to their maximal negative value, etc..

Mathematically seen, both oscillations in Fig. 37 show the same positive behavior:

if one rises, the other rises

if one falls, the other falls.

In our example, this relation lasts for as long as we examine it. In mathematical terms this means the value 1. If the same tendency exists only during a part of the examination time, the value will also be only a part of 1.

Follows: The two oscillations in Fig. 37 have the correlation degree +1.

If two oscillations are identical in frequency and phase, they have the correlation degree +1.

The two oscillations in Fig. 38 have the same frequency, but a phase difference of 180°. These two signals are also correlated, but they show an opposite, mathematically negative, behavior.

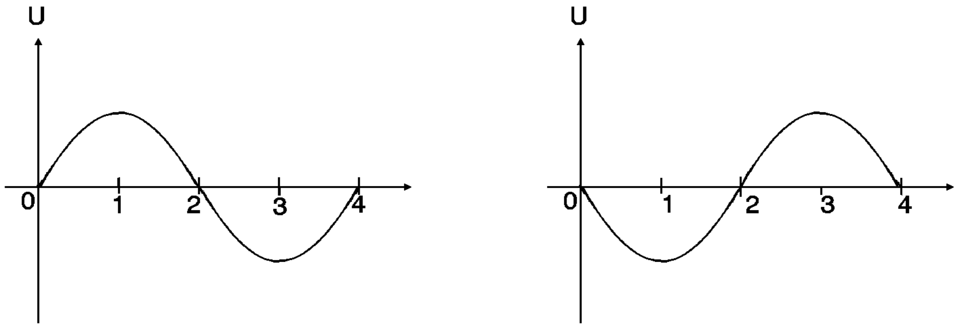


Fig. 38

During t = 0...1 the left oscillation rises to the positive peak value and the right one falls to the negative peak value; during t = 1...2 the left one falls to 0 and the right one rises from the negative peak value to 0, etc.. Since this opposite behavior continues permanently, the two oscillations have the correlation degree -1.

If two oscillations are identical in frequency but of opposite phase (180°), they have the correlation degree -1.

Fig. 39 shows two signals with different frequencies:

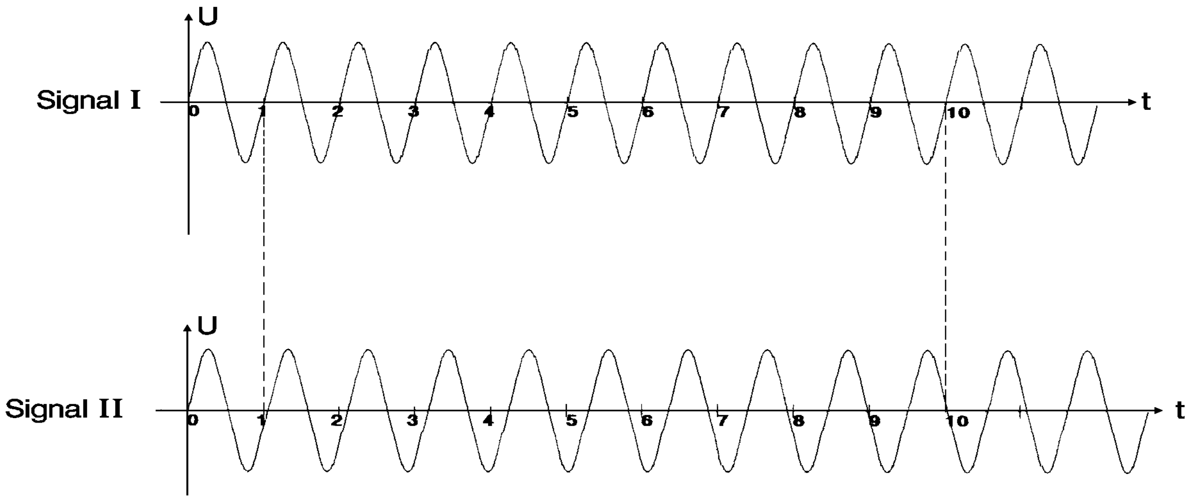


Fig. 39

During t = 0....1, which means during the first period of signal I, signal II has the same tendency, except the small part at the end of it's period. Consequently, the correlation degree for this short time is almost +1.

By period 10 both oscillations have diverged so much that the phase difference already reaches 180°, meaning that the following period (11) of signal I has the opposite tendency as signal II; thus the correlation degree for this moment is about -1.

Since this phase shift happens continuously, the correlation degree also changes continuously from +1 to -1.

If we determine the mean value of the correlation degree over a longer time, the result will be a value of 0.

Example for five possible measured values:

$$\frac{+1 + 0{,}5 + 0 + (-0{,}5) + (-1)}{5} = \frac{1{,}5 - 1{,}5}{5} = 0$$

For almost all, in sound recording techniques interesting signals we can state that:

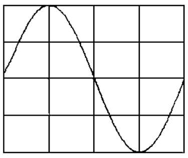
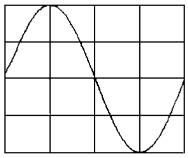
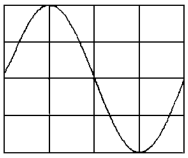
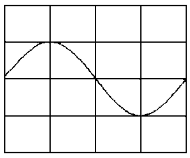
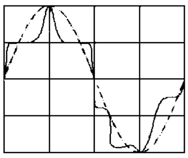
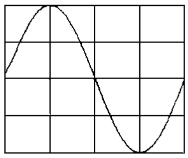
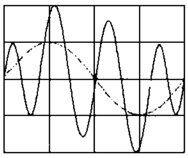
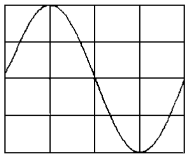
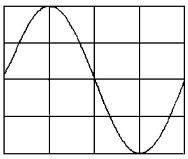
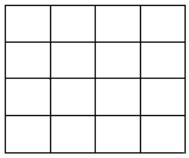
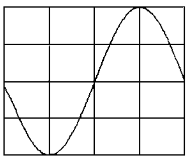
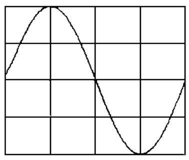
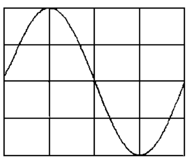
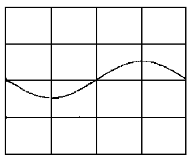
Two signals with different frequencies have a mean correlation degree of 0.

Let's transfer the knowledge of correlation degree to stereophony:

Relation between left and right channels		**corr. degree**
different frequencies		**0**
identical frequencies	**in phase**	**+1**
	in antiphase	**-1**

According to this, a correlation degree meter has to indicate 0 for different frequencies in the left and right channels. For identical frequencies with the same phase in both channels it has to indicate +1; and for identical frequencies with opposite phases it has to indicate -1. The indication has to be independent of the corresponding levels; it should not vary over the whole dynamic range.

Correlation degree of stereo signals

EXAMPLE	L	R	r
flute			
flute			
flute piccolo			
flute piccolo			
flute			
flute			
flute			

11.2 Correlation degree meter

In the preceding chapter we pointed out the demands on a correlation degree meter. The basic function of a correlation degree meter, that is to measure the phase difference between two signals, is fulfilled by a synchronous rectifier. But there are additional functional groups in the complete correlation degree meter.

So a precise reading is only possible if sufficiently big and constant signals are available at the inputs of the synchronous rectifier. That's why the signals of both stereo channels have to first be amplified and then rectified.

The amplified and rectified L-signal is used as an indicator signal, whereas the amplified and rectified R-signal is used as a switch signal.

The synchronous rectifier delivers a pulsating dc-voltage that is first integrated and then lead to the indicating instrument.

Correlation degree meter connected to MS-signals:

What happens if a correlation degree meter is mistakenly connected to the MS-channels?

1) Sound source from the right: R = +1 and L = 0 ---> M = L + R = +1
S = L - R = -1

Antiphase signals have a correlation degree of –1

2) Sound source from middle: R = +0,5 and L = +0,5 ---> M = L + R = +1
S = L - R = 0

If one of both signals is zero, the correlation degree is also zero.

3) Sound source from the left: R = 0 and L = +1 ---> M = L + R = +1
S = L - R = +1

In-phase signals have the correlation degree +1

This shows that the correlation degree can be measured only between the L and R channels.

11.3 Stereoscope

The stereoscope has become more common for display and control of stereo signals in recent years. It displays phase, direction and intensity of stereo signals on an oscilloscope tube. The basic circuit diagram is shown in Fig. 40.

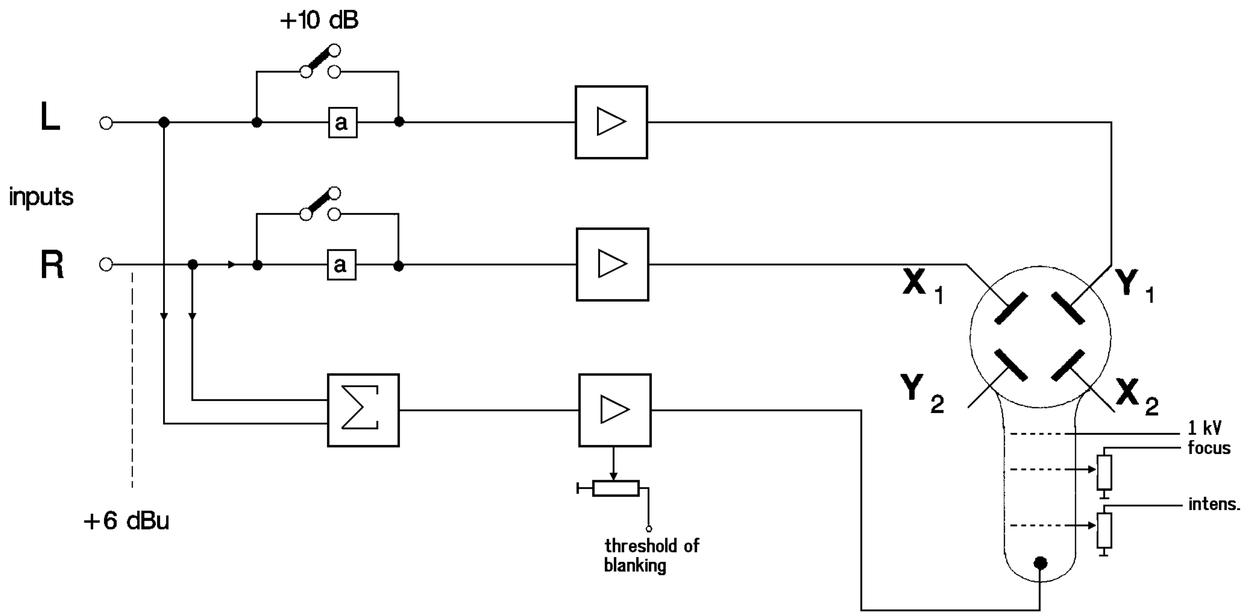


Fig. 40

In this example the input of the stereoscope is connected to an LR-signal of +6 dBm. For a higher accuracy of reading at smaller input levels, or for the head alignment of tape recorders with the help of an alignment tape, the input sensitivity can be increased by 10 dB. If the input level falls short of a level of -30 dB, the electron beam of the tube is blanked to avoid a burning-in of an ion spot on the screen during modulation pauses. The blanking threshold is variable. Also, focus and intensity of the beam can be adjusted externally.
Compared to a normal oscilloscope, the tube of the stereoscope is turned by 45°.

Fig. 41

Stereoscope: Typical Displays

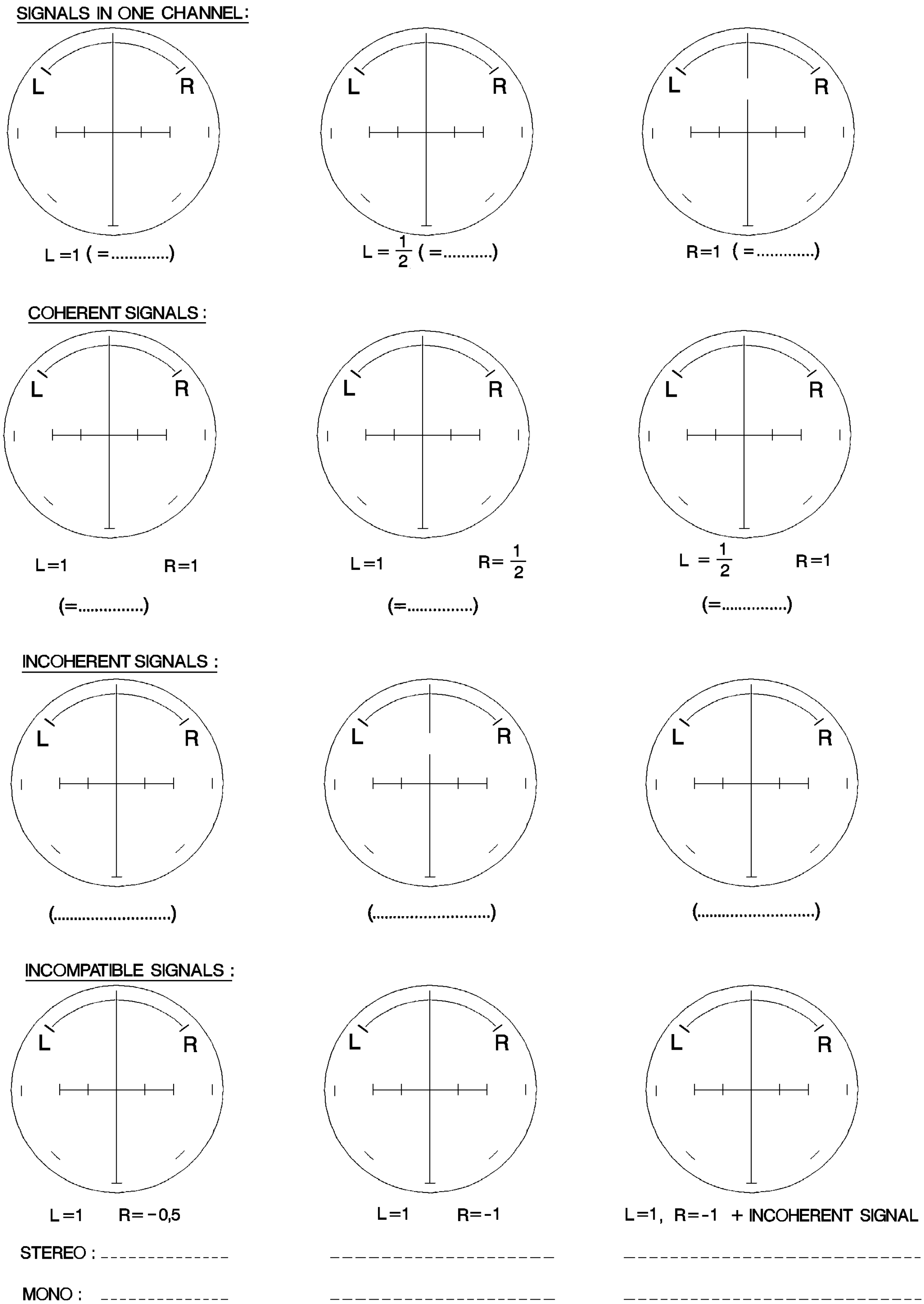
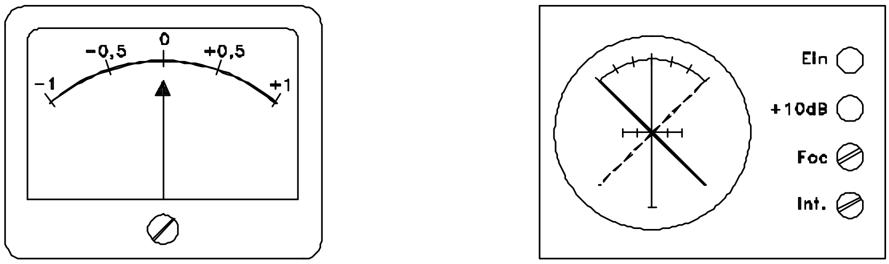


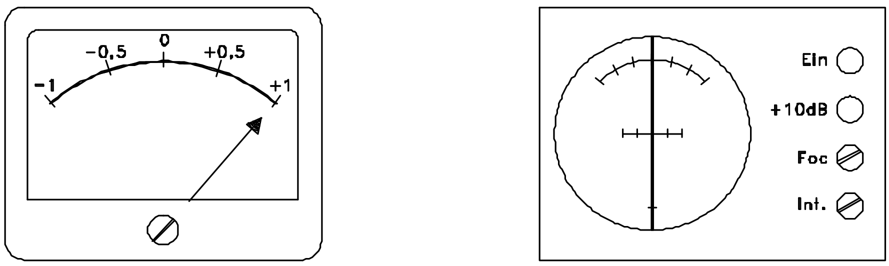
Fig. 42

Comparison

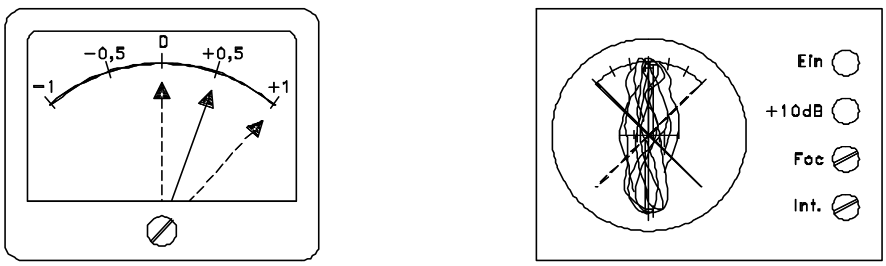
correlation degree meter **stereoscope**



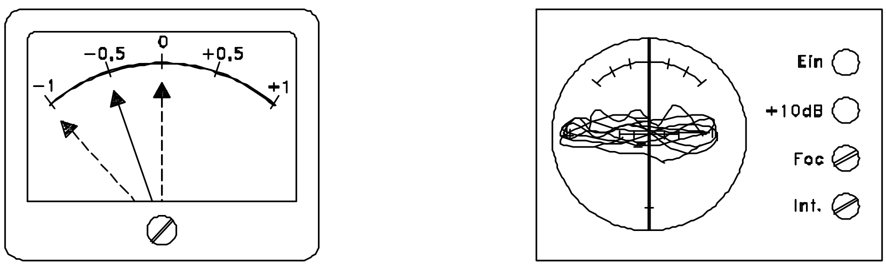
Signal in one channel only – left or right



Identical signals in both channels



Correct stereo signal (compatible)



Phase shifted parts in the stereo signal (not compatible)

Fig. 43

Typical Displays of Stereo Signals on the Stereoscope

1. Different displays with a 1 kHz test signal:

M-Signal (mono)

only left channel modulated

only right channel modulated

M-Signal, one channel antiphase

2. Full stereo signal with big basis width.

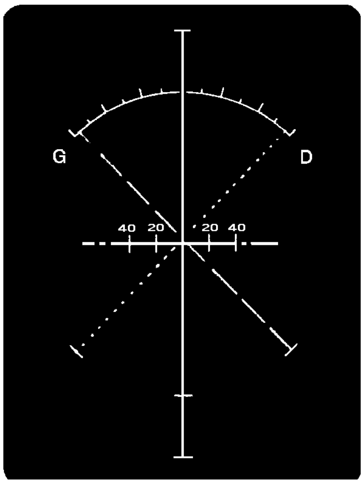


Fig. 43.1

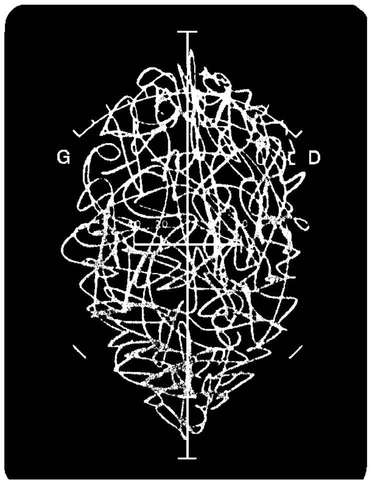


Fig. 43.2

3. Stereo signal with soloist group half-left. The small soloist signal shows a recording with mono mike.

4. Stereo signal with stereo soloist signal of wrong polarity (antiphase). Soloist signal is cancelled at mono replay.

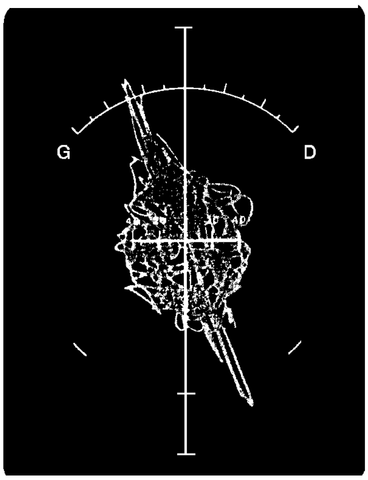


Fig. 43.3

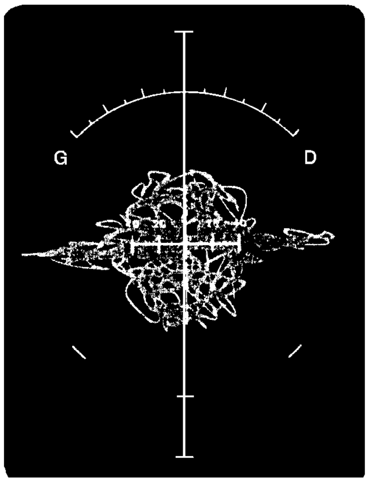


Fig. 43.4

5. Pseudo-stereo signal. (Mono signal is converted into pseudo-stereo signal with help of different filters).

6. Limiters in the stereo- or soloist channel do not have the same working characteristics. Left channel is limited earlier than right channel.

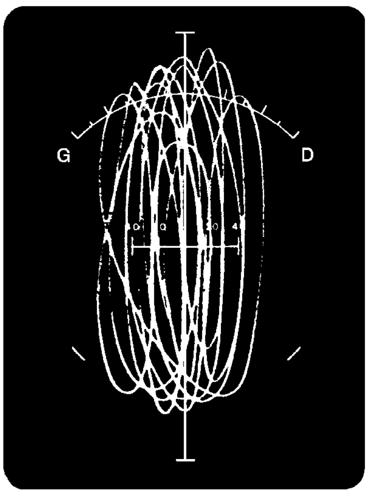


Fig. 43.5

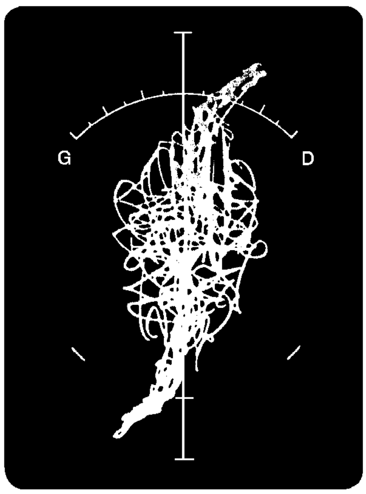


Fig. 43.6

CROWN BOUNDARY MICROPHONE APPLICATION GUIDE

PZM, PCC, SASS AND BOUNDARIES

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INTRODUCTION

A boundary microphone is a miniature microphone designed to be used on a surface such as a piano lid, wall, stage floor, table, or panel. Mounting a miniature mic on a surface gives several benefits:

- A clearer, more natural sound quality
- Extra sensitivity and lower noise
- Consistent tone quality anywhere around the microphone
- Natural-sounding pickup of room reverberation

Crown boundary microphones include the PZM, PCC, MB, and SASS series microphones. This guide explains how they work and how to use them. For information on the CM, GLM, and LM models, please see the *Crown Microphone Application Guide.*

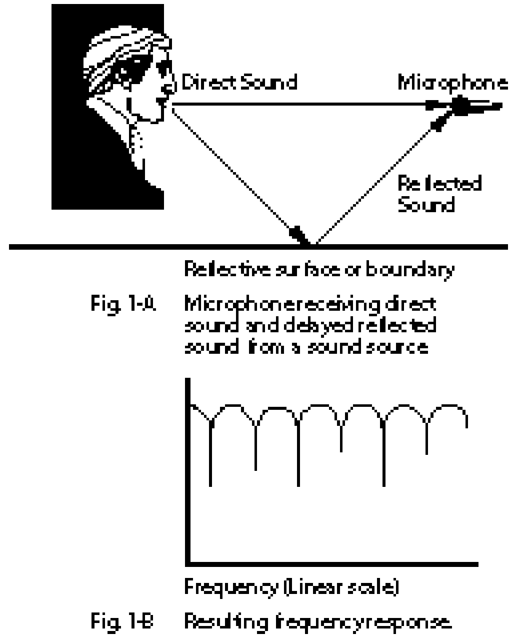


Fig. 1

BACKGROUND

In many recording and reinforcement applications, the sound engineer is forced to place microphones near hard reflective surfaces. Some situations where this might occur are recording an instrument surrounded by reflective baffles, reinforcing drama or opera with the microphones near the stage floor, or recording a piano with the microphone close to the open lid.

When a microphone is placed near a reflective surface, sound travels to the microphone via two paths: (1) directly from the sound source to the microphone, and (2) reflected off the surface (as in Fig. 1-A). Note that the reflected sound travels a longer path than the direct sound, so the reflected sound is delayed relative to the direct sound. The direct and delayed sounds combine at the microphone diaphragm.

All frequencies in the reflected sound are delayed by the same time. Having the same time delay for all frequencies creates different phase delays for each frequency, because different frequencies have different wavelengths. For example, a time delay of 1 millisecond causes a 360-degree phrase shift for a 1000-Hz wave, but only a 180-degree phase shift for a 500-Hz wave. Fig. 2 illustrates this point.

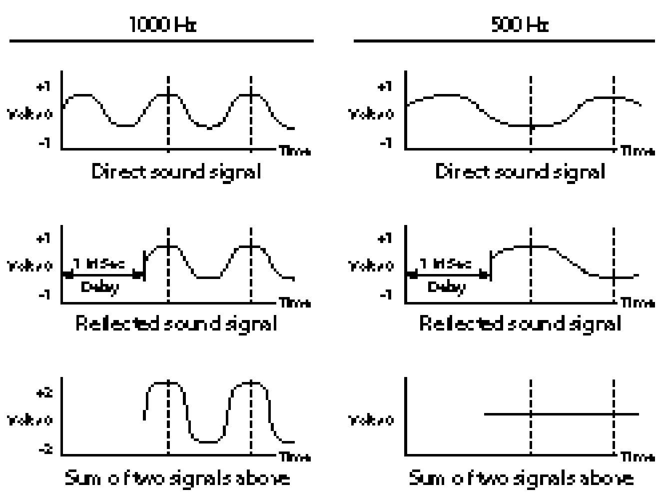


Fig. 2 - Example of wave addition and cancellation at two different frequencies.

At frequencies where the direct and delayed sounds are in-phase (coherent), the signals add together, doubling the pressure and boosting the amplitude 6 dB. At frequencies where the direct and delayed signals are out-of-phase, the signals cancel each other, creating a dip or notch in the response. There results a series of peaks and dips in the net frequency response called a comb-filter effect ,so named because the response looks like the teeth of a comb. (Fig. 1-B).

This bumpy frequency response colors the tonal reproductions, giving an unnatural sound. To solve this problem, we need to shorten the delay of the reflected sound so that it arrives at the microphone at the same time the direct sound does.

If the microphone is placed on the reflective surface (as in Fig. 3), the direct and reflected sound paths become nearly equal. There is still a short delay in the reflected sound because the center of the microphone diaphragm (where the two sound paths combine) is slightly above the surface. Consequently, the high frequencies may be cancelled, giving a dull sound quality.

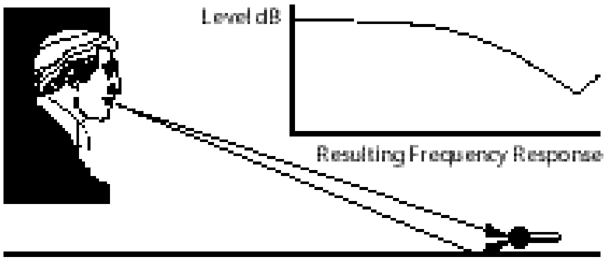


Fig. 3 - Conventional microphone on floor receiving direct sound and slightly delayed reflected sound.

HOW THE BOUNDARY MIC WORKS

By orienting the diaphragm parallel with the boundary (as in Fig. 4), the diaphragm can be placed as close to the boundary as desired. Then the direct and reflected waves arrive simultaneously at the microphone sound entry (the slit between the microphone diaphragm and the boundary). Any phase cancellations are moved outside the audible band, resulting in a smooth frequency response.

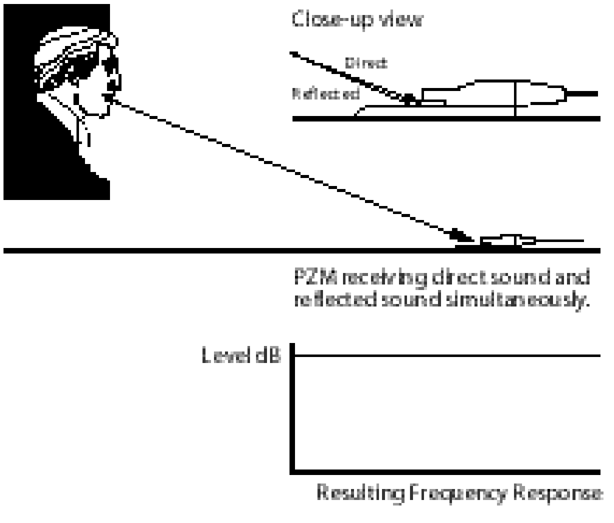


Fig. 4

The technique of mounting a microphone in this manner is called the Pressure Recording Process™ (invented by Ed Long and Ron Wickersham). They developed the first microphone to use this process. The first manufactured microphone using the principle was the Pressure Zone Microphone® (developed by Ken Wahrenbrock). PZM®s are now manufactured by Crown International, the first company licensed to the build microphones using the Pressure Recording Process.

The Pressure Zone is the region next to the boundary where the direct and reflected waves are in-phase (or nearly so). There may be a slight phase shift between the direct and reflected waves, as long as the frequency response is not severely degraded.

The Pressure Zone can be defined another way: The Pressure Zone is the distance from the boundary that the microphone diaphragm must be placed to achieve the desired high-frequency response. The closer the diaphragm is placed to the boundary (up to a point), the more extended is the high-frequency response. Let's show some examples.

For a frequency response down a maximum of 6 dB at 20 kHz, the mic-to-boundary spacing should be .11." Or you could say the Pressure Zone is .11" thick. This spacing corresponds to ⅙ wavelength at 20 kHz.

For a response down 3 dB maximum at 20 kHz, the spacing should be .085" (⅛ wavelength at 20 kHz).

For a response down 1 dB maximum at 20 kHz, the spacing should be .052" (1⁄13 wavelength at 20 kHz).

Note that the thickness of the Pressure Zone is an arbitrary number depending on frequency. For example, the direct and reflected waves of a 100-Hz tone are effectively in-phase within a Pressure Zone 10" thick.

The Crown PZM microphone-to-boundary spacing is only .020", which relates to 1 dB down at 52 kHz.

Pressure doubling

As stated earlier, comb-filtering is eliminated when the direct and reflected waves add together in-phase. There is another benefit: the sound pressure doubles, giving a 6 dB increase in acoustic level at the microphone. Thus the effective microphone sensitivity increases 6 dB, and the signal-to-noise ratio also increases 6 dB.

Consistent tonal reproduction independent of source height

The microphone placements shown in Figs. 1 and 3 cause another problem in addition to rough response. As the sound source moves up or down relative to the surface, the reflected path length changes, which varies the comb-filter notch frequencies. Consequently, the effective frequency response changes as the source moves.

But with the PZM, the reflected path length stays equal to the direct path length, regardless of the sound-source position. There is no change in tone quality as the source moves.

Lack of off-axis coloration

Yet another problem occurs with conventional microphones: off-axis coloration. While a microphone may have a flat response to sounds arriving from straight ahead (on-axis), it often has a rolled-off or colored response to sounds arriving from other directions (off-axis).

That fault is mainly due to the size of the microphone and its forward orientation. When sound strikes the microphone diaphragm on-axis, a pressure boost occurs at frequencies where the wavelength is comparable to the microphone diameter (usually above about 10 kHz). This phenomenon is called diffraction. Sounds approaching the microphone from the sides or rear, however, do not experience a pressure boost at high frequencies. Consequently, the high-frequency response is greater on-axis than off-axis. The frequency response varies with the position of the sound source.

Since the PZM capsule is very small, and because all sound enters the capsule through a tiny, radially symmetric slit, the response stays constant regardless of the angle at which sound approaches the microphone. The effective frequency response is the same for sounds from the front as it is for sounds from other directions. In other words, there is little or no off-axis coloration with the PZM. The reproduced tone quality doesn't change when the sound source moves.

As further benefit, the PZM has an identical frequency response for random-incidence sound as it has for direct sound. Direct sound is sound traveling directly from the source to the microphone; random incidence sound is sound arriving from all directions randomly. An example of random-incidence sound is ambience or reverberation – sounds reflected off the walls, ceiling, and floor of the recording environment.

With most conventional microphones, the response to reverberant, random-incidence sound is rolled off in the high frequencies compared to the response to direct sound. The direct sound may be reproduced accurately, but the reproduced reverberation may sound duller than in real life.

This fact leads to some problems in recording classical music with the microphones placed far enough away to pick up concert-hall ambience. The farther from the sound source the microphone is placed, the more reverberant is the sound pickup, and so the duller the sound is. The effective microphone frequency response may become duller (weaker in the high frequencies) as the microphone is placed farther from the sound source.

This doesn't occur with the PZM when it's used on the floor. The effective response stays the same regardless of the mic-to-source distance. The response to ambient sound (reverberation) is just as accurate as the response to the direct sound from the source. As a result, the total reproduction is brighter and clearer.

Reach

"Reach" is the ability to pick up quiet distant sounds clearly. "Clearly" means with a high signal-to-noise ratio, a wide smooth frequency response, and a high ratio of direct sound to reverberant sound.

As described earlier; the PZM has several performance attributes that contribute to excellent reach. The signal-to-noise ratio is high because the signal sensitivity is boosted 6 dB by the on-surface mounting. The frequency response is wide and smooth because comb filtering is eliminated, and because reverberant sound is picked up with a full high-frequency response. The direct-to-reverberant sound ratio is high because the direct sound is boosted 6 dB near the surface, while the reverberant sound, being incoherent, is boosted only about 3 dB.

If the PZM element is mounted in a corner, the direct sound is boosted 18 dB, while reverberant sound is boosted only 9 dB. This gives the PZM a 9 dB advantage over a conventional omnidirectional microphone in the ratio of direct-to-reverberant sound. In other words, distant sources sound closer and clearer with the PZM than they do with a conventional omnidirectional microphone.

Low vibration sensitivity

The low mass and high damping of the PZM diaphragm make it relatively insensitive to mechanical vibrations such as table and floor thumps and clothing noise. The only pickup of theses sounds is acoustic pickup through the air, not mechanical pickup through the microphone housing.

Small size

In addition to the acoustic benefits of the PZM, there are psychological benefits related to its low-profile design. Its inconspicuous appearance reduces "mic fright." Since the PZM does not point at the performers, they may feel more relaxed in not having to aim their instruments at the microphone.

PZMs can be hidden in theatre sets. In TV-studio applications, the PZM practically disappears on-camera. PZMs reduce clutter on the conference tables and lecterns, giving the feeling that no microphones are in use.

THE PCC

The Phase Coherent Cardioid (PCC) is a surface-mounted supercardioid microphone which provides the same benefits previously mentioned for the PZM. Unlike the PZM, however, the PCC uses a subminiature supercardioid mic capsule. Its directional polar pattern improves gain-before-feedback, reduces unwanted room noise and acoustics, and rejects sound from the rear.

Fig. 5 shows the difference in construction and polar patterns of the PZM and PCC.

In the Crown PCC, the microphone diaphragm is small enough so that any phase cancellations are above the audible range (Fig. 6). This results in a wide, smooth frequency response free of phase interference. In contrast, the mic capsules in conventional microphones are relatively large. As a result, reflections are delayed enough to cancel high frequencies, resulting in dull sound (Fig. 3).

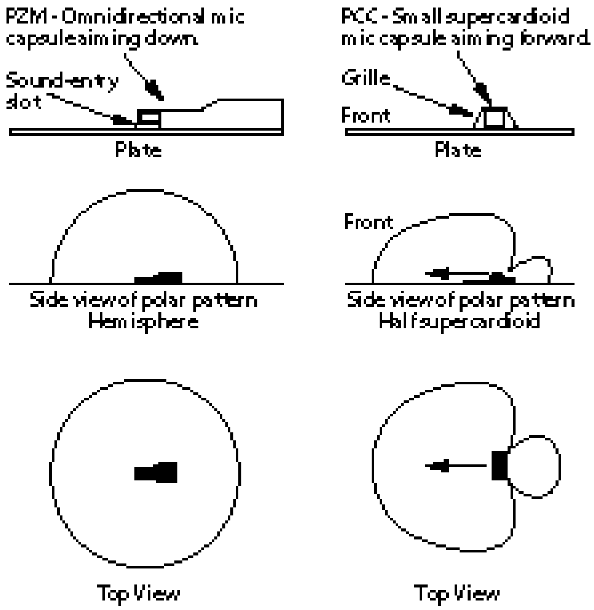


Fig. 5

Technically, the PCC is not a Pressure Zone Microphone. The diaphragm of a PZM is parallel to the boundary; the diaphragm of the PCC is perpendicular to the boundary. Unlike a PZM, the PCC "aims" along the plane on which it is mounted. In other words, the main pickup axis is parallel with the plane.

Because of its supercardioid polar pattern, the PCC has nearly a 6 dB higher direct-to-reverberation ratio than the PZM; consequently, distant sources sound closer and clearer with the PCC than with the PZM.

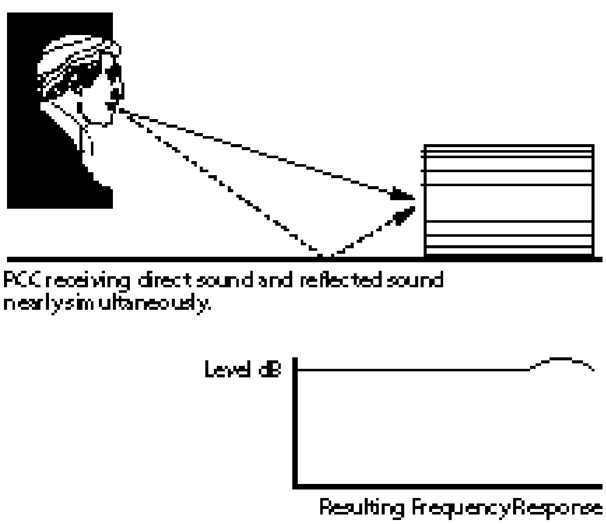


Fig. 6

BOUNDARY MICROPHONE TECHNIQUES FOR RECORDING

Before placing microphones, work on the "live" sound of the instrument or ensemble to be recorded. Do what you can to improve its sound in the studio.

To determine a good starting microphone position, close one ear with your finger; listen to the instrument with the other, and move around until you find a spot that sounds good. Put the PZM there, or put it on the floor or table at that distance.

Moving the microphone around the instrument will vary the tone quality, because an instrument radiates a different spectrum in every direction. Place the PZM to get the desired tone quality, then use equalization only if necessary. Note that the response of the PZM does not change with the angle of incoming sound, but the spectrum of the instrument does change depending on how it is aimed at the PZM.

To reduce pickup of acoustics, leakage from other instruments and background noise, move the PZM closer to the sound source. Mike only as close as necessary, since too-close placement may result in an unnatural tonal balance. Move the PZM farther from the source to add ambience or "artistic leakage" to the recording.

To further reduce pickup of unwanted sounds, mount the PZM on a large baffle or acoustic panel placed between the PZM and the offending noise source.

Use the smallest number of microphones that provide the desired sound; use as few as necessary. Sometimes this can be done by covering several sound sources with a single microphone. The wide polar pattern of the PZM allows you to pick up several instruments or vocalists on a single microphone with equal fidelity.

Follow the 3:1 rule: When multiple microphones are mixed to the same channel, the distance between the microphones should be at least three times the mic-to-source distance. This procedure reduces phase interference between the microphones. For example, if two microphones are each placed 2 feet from the instruments they cover, the mics should be at least 6 feet apart.

PZMs used in multiple-microphone applications may pick up a lot of leakage ("off-mic" sounds from other instruments). This leakage usually sounds good, however, owing to the PZM's uncolored off-axis response. Artistic usage of leakage can add pleasing "liveliness" to the recording.

When a PZM is mounted on a wall, the wall becomes a part of the microphone. When a PZM is mounted in a corner, then all three walls become part of the microphone. The Pressure Zones of all the walls combine to reinforce the sound pickup. Use this fact to your advantage by mounting the PZM capsule at the junction of multiple boundaries whenever possible.

The following are some guidelines for PZM placement in various recording applications. Many were provided by users. Although these techniques have worked well in many situations, they are just suggestions.

Acoustic guitar, mandolin, dobro, banjo:

- On panel in front, about 2 feet away, guitar height.
- On panel in front and overhead to avoid interference with audience viewing.
- On floor (for soloist).

String section:

- On panel above and in front of the entire section.
- On panel midway between two instruments, about 6 feet high.

Fiddle or Violin:

- On panel in front or overhead.
- On music stand.

Cello or acoustic bass:

- On panel on floor, in front, tilted toward performer.
- On panel in front and above.
- On floor (for soloist).

String quartet:

- Spaced pair on floor about 3 to 6 feet apart.
- Spaced pair on panels in front and above, spaced 3 to 6 feet apart.

Harp:

- On panel about 2½ feet away, aiming toward treble part of sound board.

Sax, flute, clarinet:

- On panel in front and slightly above.
- On music stand.

Horns, trumpet, cornet, trombone:

- On wall, on hard-surface gobo, or on control-room window. Performers play to the wall or gobo a few feet away. Since their sound bounces off the wall back to them, they can hear each other well enough to produce a natural acoustic balance.
- On panel in front of and between every two players, 1 to 2 feet away.
- On music stand.
- Tuba – on panel overhead.

Grand Piano:

- Tape a PZM to the underside of the lid in the middle (Fig. 7). Put the lid on the long stick for best sound quality. To reduce leakage and feedback, put the lid on the short stick or close the lid and cover the piano with a heavy blanket.
- For stereo, use two PZMs taped under the lid – one over the treble strings near the hammers, one over the bass strings well away from the hammers. Microphone placement close to the hammers emphasizes attack;

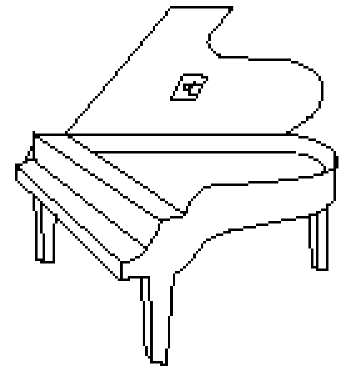


Fig. 7

placement far from the hammers yields more tone.
- To pick up the piano and room ambience with a single microphone, place a PZM on a panel about 6 to 8 feet from the piano, 4 feet high. Put the lid on the long stick, and face the panel at the piano with the panel perpendicular to the lid.
- To add ambience to a close-miked piano, mix in a PZM or two placed on a wall far from the piano.
- For singers who accompany themselves on a piano, mount two PZMs on opposite sides of a panel. Place the panel about where the music rack would be. For stereo, use a longer panel with two microphones on each side of the panel.

Amplifier/speaker for electric guitar, piano, bass:

- On panel in front of amp.
- On floor a few feet in front of amp. For an interesting coloration, add a panel a few feet behind the microphone.
- Inside the cabinet.

Leslie organ cabinet:

- Two PZMs on either side of the rotating horn, inside the top of the cabinet. Place another PZM inside the bottom cabinet.

Drum set:

- On panel or hard gobo, 1 to 2 feet in front of set, just above the level of the tom-toms. Use two microphones 3 feet apart for stereo. The drummer can balance the sound of the kit as he or she plays. Also hang a small-plate PZM vertically in the kick drum facing the beater head, with a pillow or blanket pressing against the beater head. The high sound pressure level will not cause distortion in the PZM's signal.
- Try two PZMs overhead, each mounted on a 1-foot square panel, angled to form a "V," with the point of the "V" aiming down. Omit the panel for cymbal miking.
- Two PZMs on a hard floor, about 2 feet to the left and right of the drummer.
- Tape a PZM to a gauze pad and tape the pad to the kick drum beater head near the edge. This mic will also pick up the snare drum.
- See percussion below.

Percussion:

• Use a PZM strapped to the chest of the player. The microphone is carried by the percussionist as he or she moves from instrument to instrument.

Xylophone, marimba, vibraphone:

• Use two PZMs above instrument, over bass and treble sides, with or without panels.

Lead Vocal:

• In the studio, mount a PZM on a wall, control-room window or panel a foot in front of the performer. The panel can be used in place of a music stand to hold the lyric sheet. Use the supplied foam windscreen to prevent "popping" sounds from the letter "P."

• To reduce leakage in the vocal mic, (1) overdub the vocal, (2) use gobos, or (3) use a well-damped isolation booth with one hard wall to mount the PZM on. Note: the PZM does not have proximity effect (up-close bass boost). Use console EQ to add extra warmth if necessary.

Background harmony vocals:

• On a wall or panel.

• Use one or two on both sides of a gobo, with singers surrounding the gobo.

Combos, small groups:

For small musical groups with a good natural acoustic blend, such as bluegrass, old-time, ethnic groups, blues groups, or barbershop quartets.

• On floor–two for stereo about 3 to 5 feet apart (Fig. 8).

• On panels in front, or on panels on the floor, angled toward performers.

Drama, theatre, opera:

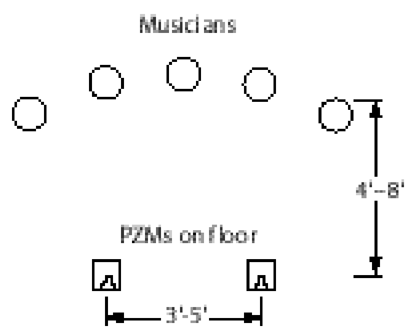


Fig. 8

• Try one to three PCCs across the front edge of the stage, about 1 foot from the edge of the stage (Fig. 9). One or two PCCs are usually sufficient for small stages, and they clearly pick up stage action for dressing room cues. Place two PCCs about 20 feet apart; place three PCCs about 15 feet apart.

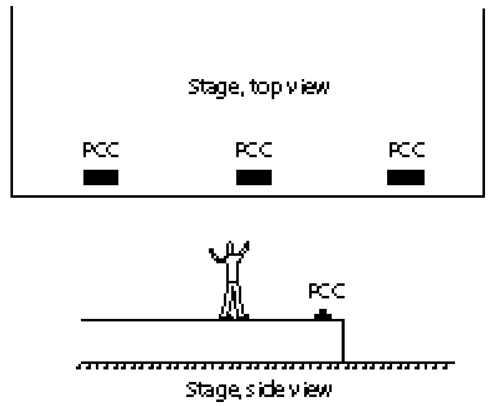


Fig. 9

For maximum clarity and maximum gain before feedback, turn up only the microphone nearest the person talking.

Show the performers and the custodian where the PCCs are located so they aren't kicked or mopped.

To reduce pickup of the pit orchestra, put a 2' x 2' piece of 4" thick Sonex foam about 1" behind each PCC.

The excellent "reach" of the PCC provides clear pickup of rear-stage action in most cases. But if you need extra reinforcement, place PZMs on the rear wall, on panels overhead, on a table under a table cloth, behind posts, under eaves, or on scenery.

Orchestra Pit:

•Tape two PZMs to the wall on either side of the conductor's podium, about 20 feet apart, facing each section of the orchestra.

• Use a separate PZM on a panel for each section of the orchestra.

Orchestra, marching band, jazz ensemble, pipe organ:

These large sound sources typically are recorded at a distance, using two microphones for stereo pickup.

• Mount a PZM 6 inches from the edge of a 2-foot square panel. Mount another PZM similarly on another panel. Tape together the panel edges nearest the microphones, forming a "V." Aim the point of the "V" at the center of the sound source. Angle the panels about 70 degrees apart (as in Fig. 10). This assembly is called a PZM wedge.

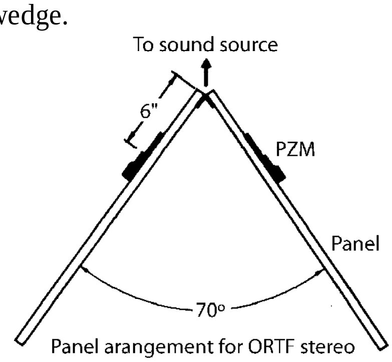


Fig. 10

The wedge can be suspended, or can be placed on edge on the floor, with the PZMs at the junction of the floor and the vertical panels (as in Fig. 11).

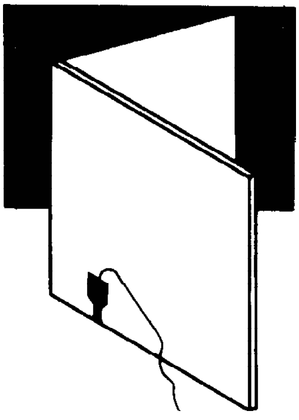


Fig. 11 - Near-coincident-stereo arrangement on floor

- Place two panel-mounted PZMs about 3 to 12 feet apart, 14 feet high, and 5 to 20 feet from the first row of players. Place the microphones farther apart to widen the stereo spread, closer together to narrow the spread.
- For noncritical documentary recordings, PZMs can be taped to the proscenium arch, the backstage wall, or the floor in front of the ensemble.

Choirs:

- Try two PZMs on panels, 5 feet above and 3 to 15 feet in front of the choir. Coverage is wide and the response is uncolored off-axis.
- For small choirs singing in an open area, place PZMs or PCCs on the floor in front of the group.
- For choirs seated on one side of a church chancel or small chapel facing the other side of the chancel or chapel, mount a PZM on the wall opposite the choir.

Ambience:

- One or two PZMs on the walls give an uncolored sound.
- One or two PZMs on the walls of an echo chamber provide ambient richness and naturalness.

Audience:

- On panels suspended over left and right sides of audience.
- Two PZMs on the front face of the stage about 4 feet apart.

Altars:

- Place a PZM or PCC on the altar table (perhaps under the table cloth).

Recording a conference:

Note: These suggestions are for recording, not for teleconferencing sound reinforcement. For teleconferencing sound reinforcement applications, see the *Crown Microphone Application Guide for Teleconferencing and Distance Learning.*

For maximum clarity, hold the conference in an acoustically "dead" room with carpeting, drapes, and acoustic-tile ceiling.

- Lay a single PZM in the middle of the table (Fig. 12).

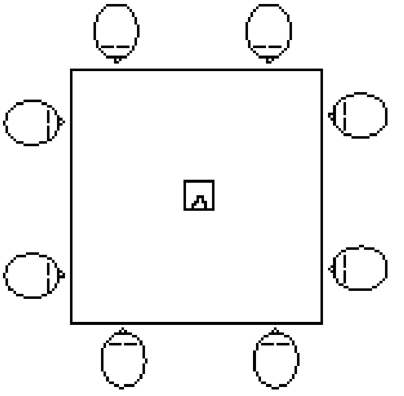


Fig. 12

- On a long table, use one PZM in the middle of every 4 to 6 people. No person should be more than 3 feet from the nearest microphone.
- For permanent installations, use the PZM-20R, a recessed microphone with all electronics and cabling under the table. Before installing it, first check that the pickup will be adequate by testing a regular PZM lying on the table.
- Try a PCC-170, a PCC-130 or a Mini Boundary mic at arm's length for every one or two people.
- For more clarity, feed the PCCs into an automatic mixer.
- If table placement is undesirable, try mounting a PZM on the ceiling.
- Remove the plate from a PZM-6D. Install the capsule/holder in an upper corner of the room as in Fig. 13. This arrangement increases microphone output by 12 dB and gives surprisingly clear reproduction. Large rooms may require such a pickup in all four corners.

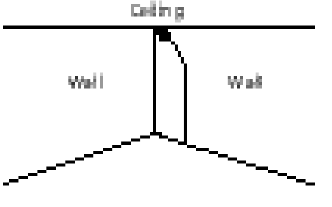


Fig. 13

Security/Surveillance:

- Try the ceiling or corner placements mentioned above.
- Use a PZM-10, PZM-10LL, PZM-11, or PZM-11LL. The PZM-10 flush-mounts in a ceiling or wall. The PZM-11 mounts in an electrical outlet box.

Lectern:

- Place a PCC on the lectern shelf top, outside of any cavities (Fig. 14). If the lectern has a raised edge, place the PCC at least twice as far from the edge as the edge is high. Set the BASS TILT switch to FLAT or BOOST, according to your preference.

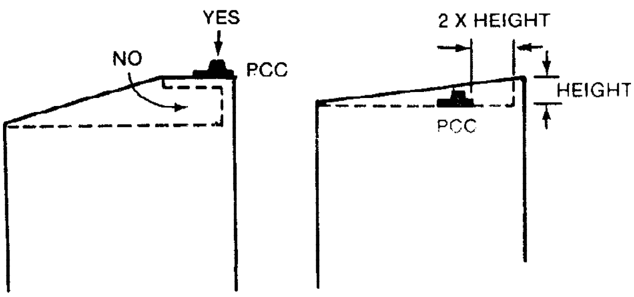


Fig. 14

- For lecterns with raised edges, you can modify your PZM as follows: remove the capsule holder by removing the two screws on the underside of the plate. Save the screws and plate for possible reassembly. Mount the capsule holder in the corner of the recess, with the holder pointing into the corner (as in Fig. 16). This configuration makes the pickup more directional but allows less talker wandering.

Fig. 15 shows how the frequency response of the PCC varies depending on where it is placed.

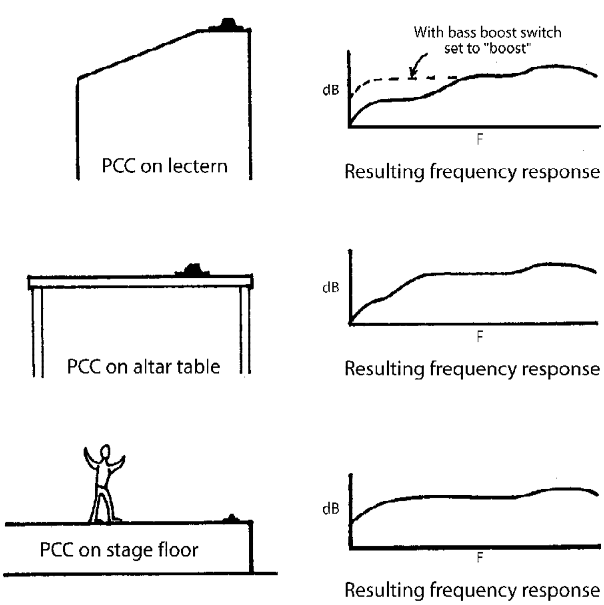


Fig. 15

Note: If you need to return the PZM for service, reattach the capsule/holder to the plate. The plate contains the identification number necessary for warranty service.

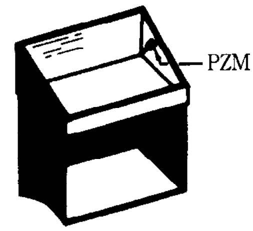


Fig. 16
Corner placement for lectern

Courtrooms:

- A PCC on the bench or witness stand can be permanently mounted and permits freedom of movement without lost speech. It provides excellent clarity and intelligibility. It also is far less intimidating to the witness than traditional microphones.

Sport events:

Basketball –

- On the basketball backboard under the hoop to pick up the sound of the ball hitting the backboard.
- On the floor just outside the boundary at center court to pick up foot and ball noises and audience reaction.
- On a 2' x 2' panel suspended over center court, using two PZMs on either side for stereo pickup.

Football –

- A PZM pyramid aimed at the field clearly picks up the quarterback calling the plays.

Boxing –

- Mount a PZM on a corner post or panel overhead.

Bowling –

- Place a PZM on the back wall of the alley, high enough to avoid being hit, to pick up the pin action.

Golf –

- Try a PZM on the ground near the tee. Insulate the mic from the ground to avoid ground loops.

Hockey –

- Tape a PCC-160 to a post, aiming down, to pick up action near the mic.

Indoor sports –

- Sports such as weight-lifting or fencing can be picked up with a PZM on the floor.

BOUNDARY MICROPHONE TECHNIQUES FOR SOUND REINFORCEMENT (P.A.)

Conventional Crown microphones (such as the LM, CM, and GLM series) work better than PZMs and PCCs for sound reinforcement of musical instruments and vocals. Please see the *Crown Microphone Application Guide* for suggestions on using conventional Crown microphones.

PZMs and PCCs can be used for sound reinforcement in many applications. These are described below.

Altar table:

• Place a PCC on the altar table as in Fig. 17. The PCC is available in black or white.

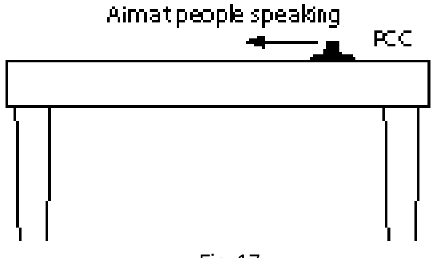


Fig. 17

Conferences, teleconferences, group discussions, interviews:

See the *Crown Microphone Application Guide for Teleconferencing and Distance Learning.*

Drama, theatre, musicals, opera

• Try one to three PCC-160s across the front edge of the stage, about 1 foot from the edge of the stage (Fig. 18). One or two PCCs are usually sufficient for small stages, and they clearly pick up stage action for dressing-room cues. Place two PCCs about 20 feet apart; place three PCCs about 15 feet apart.

For maximum clarity and maximum gain before feedback, turn up only the microphone nearest the person talking.

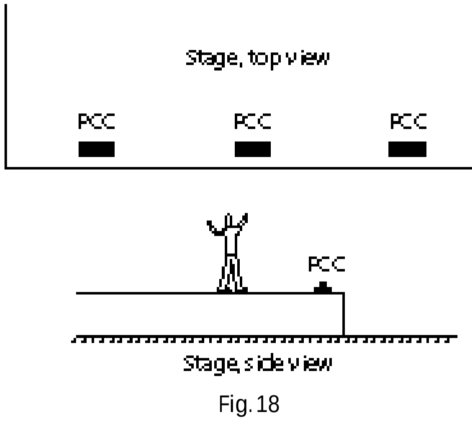


Fig. 18

Show the performers and the custodian where the PCCs are located so the PCCs are not kicked or mopped.

To reduce pickup of the pit orchestra, put a 2' x 2' piece of 4" thick Sonex foam about 1" behind each PCC.

The excellent reach of the PCC provides clear pickup of rear-stage action in most cases. But if you need extra reinforcement, place PZMs on the rear wall, on panels overhead, on a table under a tablecloth, behind posts, under eaves or scenery.

Courtroom proceedings:

• Place a PCC-170 or PCC-130 on the witness stand and judge's bench.

Horns

• Tape a PZM on the music stand above the sheet music.

Grand piano:

• Tape one or two PZMs or PCCs to the underside of the lid, about 8" horizontally from the hammers (see Fig. 19). To reduce feedback, close the lid.

Upright piano:

• Tape two PZMs 3' apart on the wall, 3' up. Place the piano frame 1" from the wall so that the PZMs pick up the soundboard.

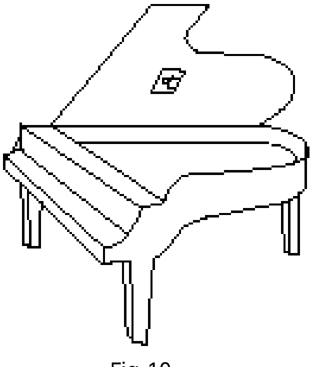


Fig. 19

Drum set or percussion:

• Tape a PZM to the drummer's chest. Hang a PZM vertically in the kick drum with the microphone side of the plate aiming toward the beater.

PZM BOUNDARIES

You can greatly broaden your range of PZM® applications by mounting the PZMs on one or more *boundaries*. A boundary is a stiff, nonabsorbent surface such as a floor, table, or plexiglass panel. PZM boundaries are usually constructed of clear acrylic plastic (plexiglass) to make them less them less conspicuous, but any stiff, sound-reflective material can be used.

By adding boundaries to a PZM, you can tailor the microphone's frequency response and directional pattern. Such flexibility makes the PZM one of the world's most versatile microphones.

This section explains the theory, benefits and drawbacks of single and multiple boundaries. Also covered are construction methods for several types of PZM boundary assemblies.

Credit is due Ken Wahrenbrock for his pioneering work in multiple boundary experiments, and for many of the boundary array suggestions in this section.

A PZM is designed to be mounted very near a boundary to prevent acoustic phase cancellations. The boundaries mentioned in this guide will degrade the frequency response and polar patterns of conventional microphones. Only PZMs can be used effectively in multiple boundaries.

The size, shape and number of boundaries all have profound effects on the performance of a PZM mounted on those boundaries. Let's discuss these effects in detail.

Sensitivity Effects

Imagine a PZM mic capsule in open space, away from any boundaries. This microphone has a certain sensitivity in this condition (Fig. 20).

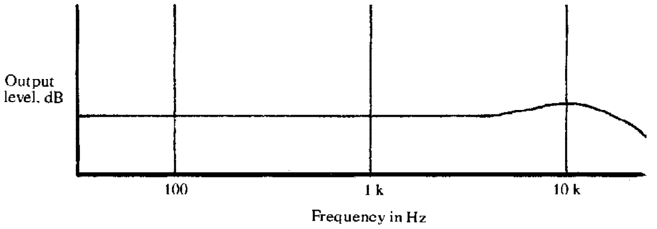


Fig. 20

Now suppose the PZM capsule is placed very near (within .020" of) a single large boundary, such as a wall. Incoming sound reflects off the wall. The reflected sound wave adds to the incoming sound wave in the "pressure zone" next to the boundary. This coherent addition of sound waves doubles the sound pressure at the microphone, effectively increasing the microphone sensitivity 6 dB.

In short, adding one boundary increases sensitivity 6 dB. *This is free gain.*

Now suppose the PZM capsule is placed at the junction of *two* boundaries at right angles to each other, such as the floor and a wall. The wall increases sensitivity 6 dB, and the floor increases sensitivity another 6 dB. Thus, adding two boundaries at right angles increases sensitivity 12 dB.

Now let's place the PZM element at the junction of *three* boundaries at right angles, such as in the corner of the floor and two walls. Microphone sensitivity will be 18 dB higher than what it was in open space. This is increased gain with no increase in noise!

Note that the *acoustic* sensitivity of the microphone rises as boundaries are added, but the *electronic* noise of the microphone stays constant. Thus, the effective signal-to-noise ratio of the microphone improves 6 dB every time a boundary is added at right angles to previous boundaries.

If a PZM is in the corner of three boundaries that are NOT at right angles to each other, the sensitivity increases *less than* 6 dB per boundary. For example, a PZM-2.5 boundary is built with two panels at 135 degrees. This panel assembly is at right angles to a base plate. The net gain in sensitivity from these three boundaries is approximately 16 dB rather than 18 dB.

Direct-to-Reverb Ratio Effects

We mentioned that sensitivity increases 6 dB per boundary added. That phenomenon applies to the *direct* sound reaching the microphone. *Reverberant* or *random-incidence* sound increases only 3 dB per boundary added. Consequently, the direct-to-reverb ratio increases 3 dB (6-3dB) whenever a boundary is added at right angles to previous boundaries.

A high direct-to-reverb ratio sounds close and clear; a low direct-to-reverb ratio sounds distant or muddy. Adding boundaries increases the direct-to-reverb ratio, so the subjective effect is to make the sound source audibly closer or clearer. That is, "reach" is enhanced by adding boundaries.

Frequency-Response Effects

The size of the boundary on which the PZM is mounted affects the PZM's low-frequency response. The bigger the boundary, the better the bass. Specifically, the response begins to shelve down 6 dB at the transition frequency F_T, where

$F_T = 750/D$

D is the boundary dimension in feet. The response is down 6 dB at the frequency F_{-6} where

$F_{-6} = 188/D$

For example, if the boundary is 2 feet square,

$F_T = 750/D = 750/2 = 375$ Hz.

$F_{-6} = 188/D = 188/2 = 94$ Hz.

That is, the microphone starts to shelve down at 376 Hz and is down 6 dB at and below 94 Hz. (See Fig. 21).

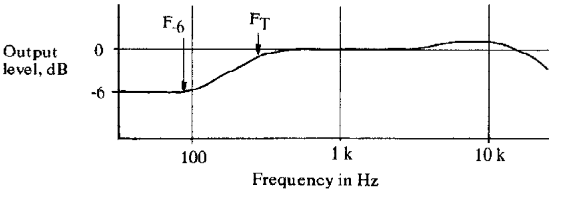


Fig. 21

Below 94 Hz, the response is a constant 6 dB below the upper-mid frequency level. Note that there is a response *shelf*, not a *rolloff*.

If a PZM is mounted on a 4' square boundary,

$F_T = 750/4 = 178$ Hz

$F_{-6} = 188/4 = 47$ Hz.

This result has been loosely called the "4' – 40 Hz" rule. Fig. 22 shows the PZM response on various sizes of boundaries.

What if the PZM is on a rectangular boundary? Let's call the long side "Dmax" and the short side "Dmin." The response is down 3 dB at 188/Dmax, and is down another 3 dB at 188/Dmin.

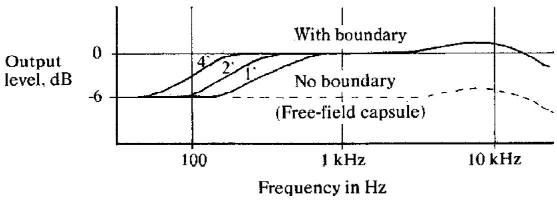


Fig. 22

As Fig. 23 shows, the low-frequency shelf varies with the angle of the sound source around the boundary. At 90 degrees incidence (sound wave motion parallel to the boundary), there is no low-frequency shelf.

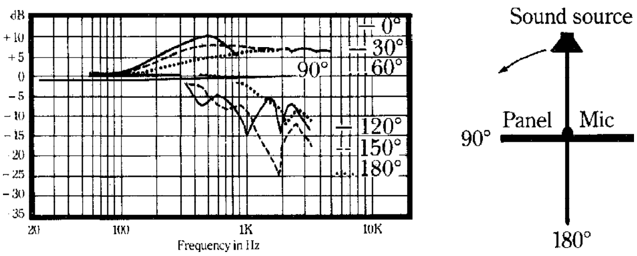


Fig. 23

The depth of the shelf also varies with the distance of the sound source to the panel. The shelf starts to disappear when the source is closer than a panel dimension away. If the source is very close to the PZM mounted on a panel, there is no low-frequency shelf; the frequency response if flat.

If the PZM is at the junction of two or more boundaries at right angles to each other, the response shelves down 6 dB *per boundary* at the frequency mentioned above. For example, a two-boundary unit made of 2-foot square panels shelves down 12 dB at and below 94 Hz.

There are other frequency-response effects in addition to the low-frequency shelf. For sound sources perpendicular to the boundary, the response rises about 10 dB above the shelf at the frequency where the wavelength equals the boundary dimension (see Fig. 23).

For a square panel, F_{peak} = .88C/D, where C = the speed of sound (1130 feet per second) and D = the boundary dimension in feet. For a circular panel, F_{peak} = C/D. As example, a 2' square panel has a 10 dB rise above the shelf at .88C/D = 88 x 1130/2 = 497 Hz.

Note that this response peak is only for the direct sound of an on-axis source. If the sound field at the panel is partly reverberant, or if the sound waves strike the panel at an angle, the effect is much less. The peak is also reduced if the mic capsule is placed off-center on the boundary.

Fig. 23 shows the frequency response of a PZM mounted on a 2' square panel, at various angles of sound incidence. Note several phenomena shown in the figure:

- The low-frequency shelf (most visible at 30 and 60 degrees).
- The lack of low-frequency shelving at 90 degrees (grazing incidence).
- The 10 dB rise in response at 497 Hz.
- Less interference at increasing angles up to 90°.
- Greater rear rejection of high frequencies than low frequencies.

What are the acoustic causes of these frequency-response effects?

When sound waves strike a boundary, pressure doubling occurs at the boundary surface, but does not occur outside the boundary. Thus there is a pressure difference at the edge of the boundary. This pressure difference creates sound waves.

These sound waves generated at the edge of the boundary travel to the microphone in the center of the boundary. At low frequencies, these edge waves are opposite in polarity to the incoming sound waves.

Consequently, the edge waves cancel the pressure doubling effect.

Thus, at low frequencies, pressure doubling does not occur; but at mid-to-high frequencies, pressure doubling does occur. The net effect is a mid-to-high frequency boost, or – looked at another way – a low-frequency loss or shelf.

Incoming waves having wavelengths about six times the boundary dimensions are cancelled by edge effects; waves of wavelength much smaller than the boundary dimension are not cancelled by edge effects.

Waves having wavelengths on the order of the boundary dimensions are subject to varying interference vs. frequency; i.e., peaks and dips in the frequency response.

At the frequency where the wavelength equals the boundary dimension, the edge wave is in phase with the incoming wave. Consequently, there is a response rise (about 10 dB above the low-frequency shelf) at that frequency. Above that frequency, there is a series of peaks and dips that decrease in amplitude with frequency.

The edge-wave interference decreases if the incoming sound waves approach the boundary at an angle.

Interference also is reduced by placing the mic capsule off-center. This randomizes the distances from the edges to the mic capsule, resulting in a smoother response.

Directional Effects

The polar pattern of a PZM on a large surface is hemispherical. The microphone picks up equally well in any direction above the surface plane, at all frequencies.

By adding boundaries adjacent to this PZM, you can shape its directional pickup pattern. Boundaries make the PZM reject sounds coming from behind the boundaries. In addition, making the PZM directional increases its gain-before-feedback in live reinforcement applications. Directional PZMs also pick up a higher ratio of direct sound to reverberant sound, so the resulting audio sounds "closer" and "clearer."

In general, sound pickup is fairly constant for sound sources at any angle in front of the boundaries, and drops off rapidly when the source moves behind the boundaries.

For sounds approaching the rear of the panel, low frequencies are rejected least and high frequencies are rejected most.

A small boundary makes the PZM directional only at high frequencies. Low frequencies diffract or bend around a small boundary as if it isn't there. The bigger you can make the boundary assembly, the more directional the microphone will be across the audible band.

The bigger the boundary, the lower the frequency at which the PZM becomes directional. A PZM on a square panel is omnidirectional at very low frequencies, and starts to become directional above the frequency F, where F = 188/D and D is the boundary dimension in feet. Sound familiar? That's the same equation used to predict the 6-dB-down point in the frequency response.

Boundaries create different polar patterns at different frequencies. For example, a 2' square panel is omnidirectional at and below 94 Hz. At mid-frequencies, the polar pattern becomes supercardioid. At high frequencies, the polar pattern approaches a hemisphere (as in Fig. 24). Two boundaries are more directional than one, and three are more directional than two.

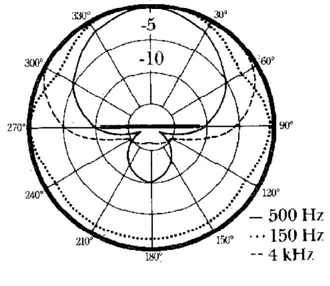


Fig. 24 – Polar response of 2–foot square boundary

With multiple boundaries, the shape of the pickup pattern approximates the shape of the boundary assembly. For example, a V-shaped boundary produces a polar pattern with a lobe whose sides are defined by the sides of the "V." Note, however, that the polar pattern varies with frequency.

This "V"-shaped boundary works like a horn loudspeaker in reverse. Speaker horn theory applies to microphone horns. For instance, if you want a constant directivity boundary horn, the horn must flare out exponentially like a well-designed loudspeaker horn.

Disadvantages of Boundaries

Boundaries must be large to be effective. Their size and weight makes them cumbersome to mount or hang. Large boundaries are also visually conspicuous, but this problem is reduced by using clear plastic.

Many users claim that the sound quality and flexibility if multiple-boundary PZMs outweigh the disadvantages. For those users who need a directional PZM but prefer not to use boundaries, Crown makes the PCC®-160 and PCC-170, which are supercardioid surface-mounted microphones. They use a directional mic capsule, rather than boundaries, to make the microphone directional. The PCC-130 is cardioid.

Summary

- Microphone sensitivity increases 6 dB for every boundary added at right angles to previous boundaries (less than 6 dB if not at right angles).
- For a flat panel, the frequency response begins to shelve down at the transition frequency F_T = 720/D, where D = boundary dimension in feet. The response shelves down 6 dB at and below the frequency F_{-6} = 188/D. This shelf disappears if the sound source is at the side of the panel, or if the source is very close to the microphone (less than a panel dimension away).
- For a square panel, the frequency response rises about 10 dB above the low-frequency shelf (when the source is perpendicular to the boundary) at the frequency F = .88C/D, where C = the speed of sound (1130 fps) and D = the boundary dimension in feet.
- The PZM/panel assembly is omnidirectional at and below the frequency F = 188/D, where D = the boundary dimension in feet. The panel becomes increasingly directional as frequency increases.
- Use the biggest boundaries that are not visually conspicuous. Big boundaries provide flatter response, better bass, and more directionality than small boundaries.
- The flattest response for a single panel occurs for angles of incidence between 30° and 90° to the axis perpendicular to the boundary.
- For the flattest response, place the PZM 1⁄3 of the way off-center (say, 4" off-center for a 2' panel). For flattest response on multiple boundaries, place the tip of the PZM cantilever touching against the boundaries (leaving the usual gap under the mic capsule).
- To increase directionality and reach, increase boundary size or add more boundaries.

Construction Tips

You can obtain clear acrylic plastic (plexiglass) from a hardware store, plastic supplier or a fabrication company. Plastic ¼" thick is recommended for good sound rejection. Many vendors can heat and shape the plastic according to your specifications. They use their own adhesives which are usually proprietary.

Cyanoacrylate adhesive (“Super Glue”) or RTV (“Sealastic”) have worked well in some instances. Or you can join several pieces of plastic with metal brackets, bolts and nuts.

If you intend to hang or “fly” the boundary assembly, drill holes in the plastic for tying nylon line. To prevent cracks in the plastic, use ceramic drill bits or start with small drill-bit sizes and work up. You may want to paint the boundary edges flat black to make them less visible.

When making a multi-boundary assembly, be sure to mount the PZM mic capsule as close as possible to the junction of the boundaries. Let the tip of the cantilever touch the plastic, but leave the usual gap under the mic capsule.

NOTE: Some older PZMs include a small block of foam under the mic capsule for acoustical adjustment. If your PZM has this foam block, trap it under the mic capsule before screwing the PZM cantilever to the boundary.

The PZM model used for multi-boundary assemblies is the PZM-6D. When drilling the screw holes for the cantilever, make them 5/32" diameter, .563" center-to-center, and countersunk .250" x 90°.

2-Foot-Square Flat Panel

This boundary (Fig. 25) is most often used for directional pickup of solo instruments, choirs, orchestras, and bands. Two PZMs back-to-back on a panel form a “bipolar” PZM for coincident stereo. Place the assembly about 14 feet above the stage floor.

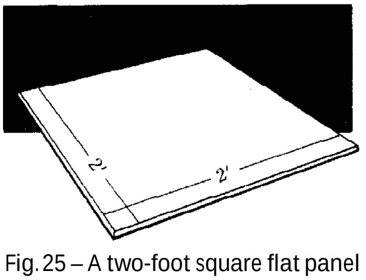


Fig. 25 – A two-foot square flat panel

For near-coincident stereo miking, place two panels with edges touching to form a “V” (Fig. 10). Aim the point of the “V” at the sound source. Mount a PZM about 4" off-center on each panel, toward the point of the “V” for better stereo imaging. This assembly provides a higher direct-to-reverb imaging. This assembly provides a higher direct-to-reverb ratio (a closer perspective) than the bipolar PZM mentioned above. It also rejects sounds approaching the rear of the panels.

The frequency response of a flat panel is the smoothest of all the boundary assemblies in this booklet. For a 2-foot square panel, there is a 10-dB rise above the low-frequency shelf at 497 Hz for direct sound at normal incidence (Fig. 23). F_{-6}=94 Hz.

The polar pattern is omnidirectional at low frequencies, supercardioid at mid frequencies, and hemispherical at high frequencies (see Fig. 26).

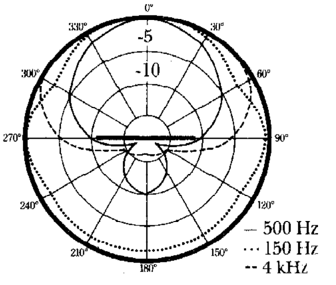


Fig. 26 – Polar response of a 2 foot-square boundary.

Random Energy Efficiency = –3 dB at high frequencies. The assembly has 3 dB less reverb pickup than an omnidirectional microphone in open space at the same distance.

Distance factor =1.41. That is, the microphone/panel can be placed 1.41. times as far from the source as an omnidirectional microphone for the same direct-to-reverb ratio.

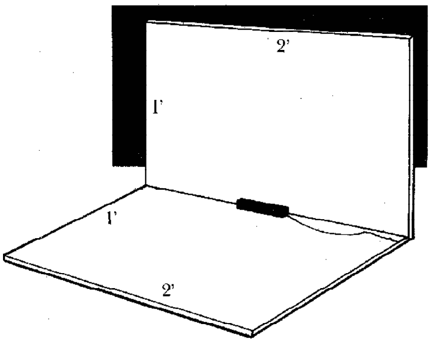


Fig. 27 – PZM-2

PZM-2

This model uses two panels at right angles to each other. One of the panels is placed on a large flat surface such as a table or floor.

One configuration uses a 1'x2' vertical panel. When this vertical panel is placed on a horizontal surface, the vertical panel is “reflected” in the horizontal surface. The panel and its reflection appear to be a 2'x2' panel with a 94-Hz shelving frequency.

Random Energy Efficiency =–6 dB. The assembly has 6 dB less reverb pickup than an omnidirectional mic in open space as the same distance.

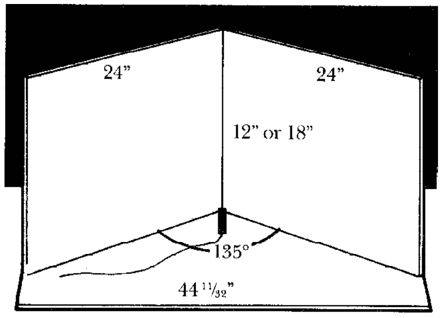


Fig. 30 – PZM 2.5

PZM-2.5

This model provides about 10 dB of forward gain at mid frequencies compared to a PZM on the floor. The assembly is placed on a large horizontal surface such as a stage floor.

An 18"tall unit works well for speech pickup of drama, musicals, and opera; cello, string bass, and kick drum.

F_{-6}=160 Hz for 12"tall model. Polar pattern (12" model): See Figs. 28 and 29.

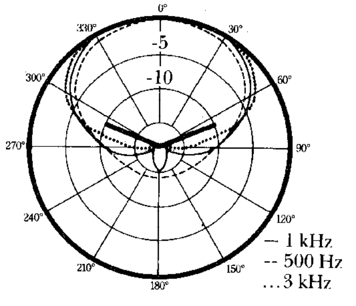


Fig. 28 – 12" tall PZM-2.5: horizontal-plane polar response.

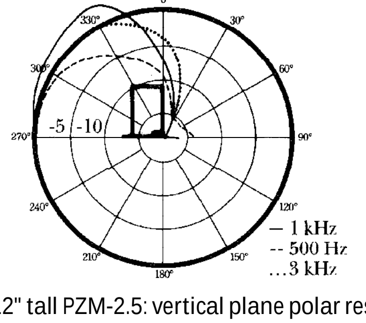


Fig. 29 – 12" tall PZM-2.5: vertical plane polar response.

PZM-3

This model has a tighter polar pattern than the PZM-2.5, so it can be used to isolate soloists. Again, the assembly is placed on a large horizontal surface such as a stage floor.

F_{-12} (two 1' square panels on floor) =94 Hz.

Random Energy Efficiency =–9 dB. The assembly has 9 dB less reverb pickup than an omnidirectional microphone in open space at the same distance.

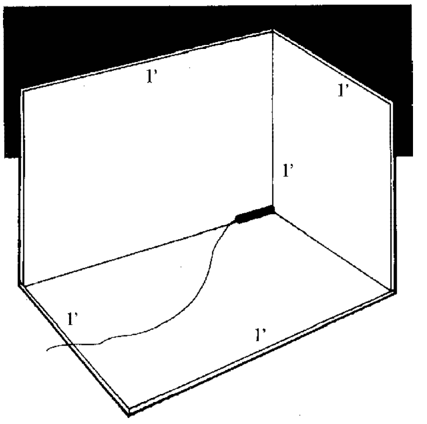


Fig. 31 – PZM 3

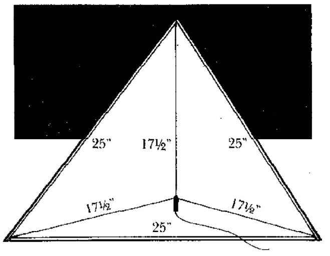


Fig. 32 – PZM Pyramid

PZM Pyramid

This model can be made of three or four sides. It emphasizes mid frequencies and is recommended only for speech. Its highly directional pattern makes it useful for long-distance pickup of quarterback calls. Pyramids also have been hung over stages for pickup of rear-stage dialog.

Since a plexiglass pyramid can be quite heavy, you may want to make it out of sheet metal.

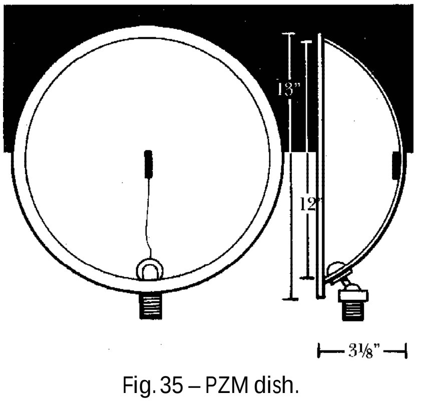


Fig. 35 – PZM dish.

PZM Dish

The PZM Dish has an uneven response on-axis, but is useful for its excellent directionality at mid-to-high frequencies. Dishes have been used over orchestral sections for isolation, and for long-reach speech applications.

The dish is not a parabolic microphone. The PZM is placed on the dish, rather than at the focus of a parabolic surface. The dish obtains its directionality from diffraction (blocking sound waves from certain directions), while a parabolic microphone obtains its directionality by focusing sound energy from a particular direction on the mic capsule.

F_{-12}=250 Hz.

Frequency response: See Fig. 33.

Polar pattern: See Fig. 34.

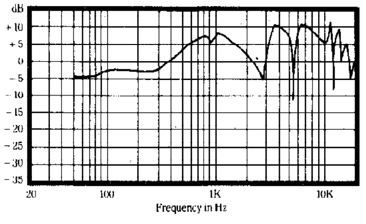


Fig. 33 – PZM dish frequency response.

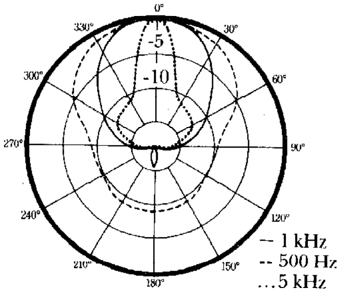


Fig. 34

PZM Cone

This model is highly directional and emphasizes mid-frequencies. It has been used as a "follow" mic for a roving TV camera, and provides a close-up audio perspective.

Random Energy Efficiency (for a cone with a 90-degree included angle) =–8.3 dB. The cone rejects reverb by 8.3 dB compared to an omnidirectional microphone in open space at the same distance.

Distance factor: 2.6. That is, the cone can be placed 2.6 times as far from the source as an omni mic for the same direct/reverb ratio.

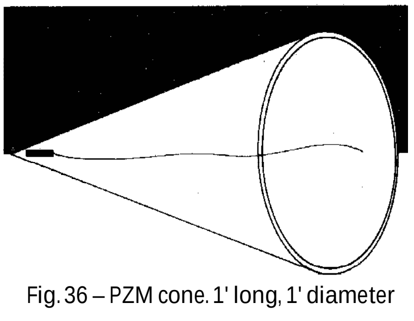


Fig. 36 – PZM cone. 1' long, 1' diameter

1560, 2260, 4060, 7260

These models have the same basic shape – two panels angled 60 degrees apart—but have different sizes. In general, the bigger the panels, the better the low-end response and the lower in frequency the directivity extends.

The 1560 is typically used on lecterns. Its response and polar patterns are shown in Figures 37, 38 and 39.

The 7260 has been used for stereo pickup of xylophones or brass sections. It is assembled in two halves for easier transport.

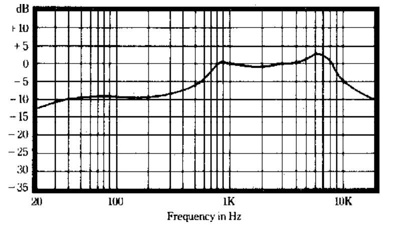


Fig 37 – Frequency response of PZM-6D on 1560 boundary.

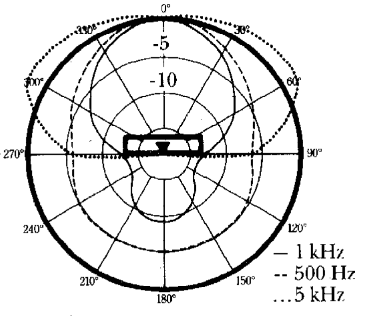


Fig 38 – 1560: horizontal-plane polar response.

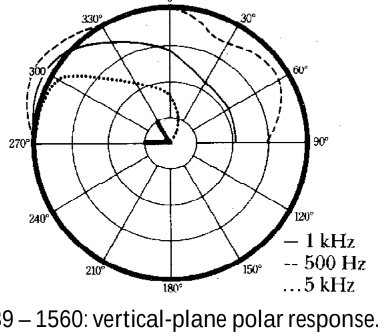


Fig. 39 – 1560: vertical-plane polar response.

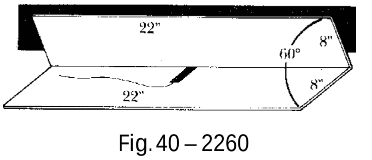


Fig. 40 – 2260

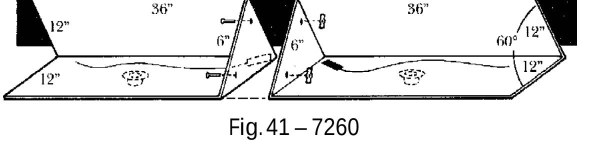


Fig. 41 – 7260

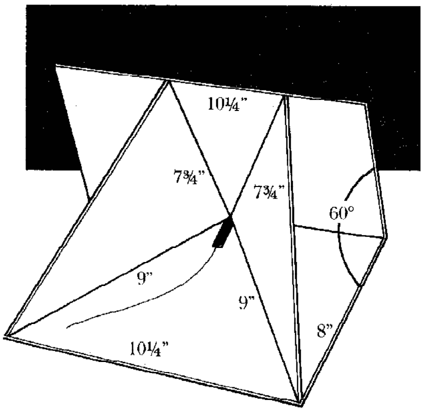


Fig. 42 – 1560 with side boundaries.

1560 with Side Boundaries

This is a basic 1560 modified with two side boundaries at 45 degrees on each side (Fig. 42). The side boundaries provide additional discrimination of loudspeakers to either side of the lectern.

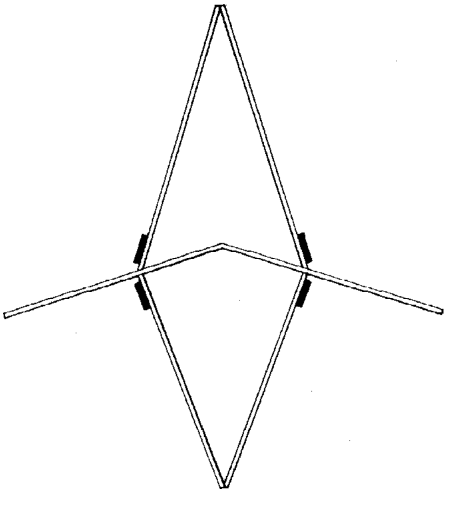


Fig. 43

L^2 Array

This multipurpose array (Fig. 43) was designed by recording engineer Mike Lamm. Mike has used this array extensively for overall stereo or surround pickup of large musical ensembles.

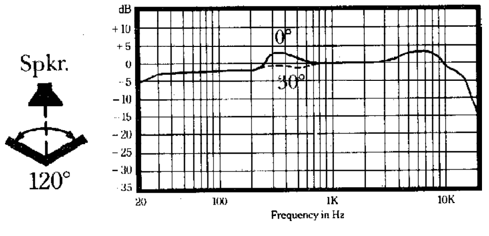


Fig. 44 – Frequency response of L^2 array (with PZM-6D capsule). 120° between boundaries.

The hinged, sliding panels can be adjusted to obtain almost any stereo pickup pattern. A complete description of the L^2 Array is in AES preprint 2025 (C-9), "The Use of Boundary Layer Effect Microphones in Traditional Stereo Miking Techniques," presented at the 75th Convention of the Audio Engineering Society, October 1983. The frequency response is shown in Figures 45 and 46.

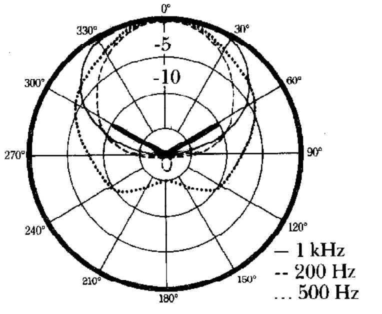


Fig. 45 – Polar response of L^2 array, 120° between boundaries.

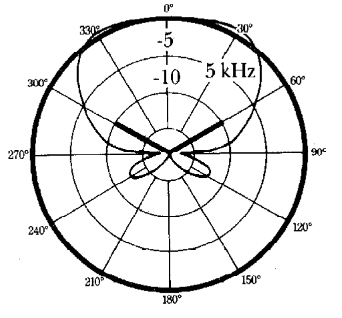


Fig. 46 – Polar response of L^2 array, 120° between boundaries.

L^2 Floor Array

Here's another stereo PZM array (Fig. 47) designed by recording engineers Mike Lamm and John Lehmann. It simulates the O.R.T.F. stereo mic technique. According to one user, "You can take this array, set it down, and just roll. You get a very close approximation of the real event."

Suspending the inverted array results in less bass and more highs, while placing it on the floor reverses the balance.

When this array is used on a stage floor, the construction shown in Fig. 48 is useful. It has decreased side pickup and increased pattern overlap. The axes of the left and right polar patterns may be at any desired angle, just so the 120° boundary angle and 6.7-inch capsule spacing are maintained.

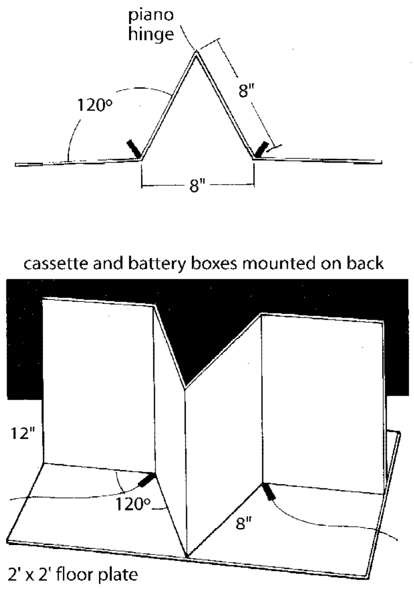


Fig. 47 – L² floor array – designed by Mike Lamm and John Lehmann can be set on the floor, set on a C stand or hung inverted from the ceiing.

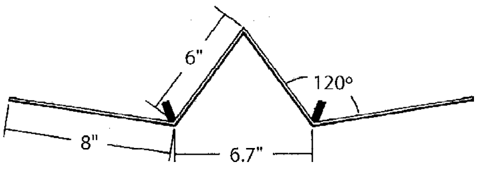


Fig. 48 – Another version of the L² floor array.

PZM Wedge or Axe

This stereo PZM array has been used extensively by recording engineers Mike Lamm and John Lehmann. It simulates the O.R.T.F. stereo microphone technique. Stereo imaging is precise and coverage is even.

Place the mic capsules 2½" below the center of the panels to smooth the frequency response. To compensate for the bass shelving of the panels, boost the bass +6 dB at and below 141 Hz.

A panel containing a ⅝"-27 Atlas flange can be fastened to the bottom of the array for stand mounting.

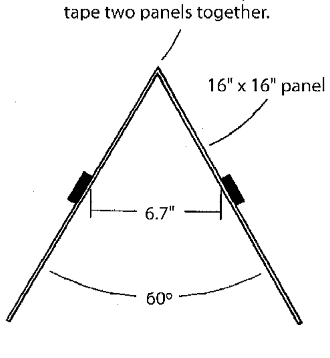


Fig. 49 – PZM wedge or axe (top view).

Aim the point of the wedge at the center of the sound source, and raise the array about 14 feet off the floor.

The wider the angle between boundaries, the more frontal is the directivity. Increasing the angle makes the sound source appear to be closer to the listener. The array is mono compatible to a great extent.

A wedge used by Gary Pillon of General Television Network, Detroit, uses 2'-square panels. The mics are 4" off-center toward the point of the "V". To compensate for bass shelving of these panels, boost the bass +3 dB at 94 Hz.

Pillon Stereo PZM Array

This stereo PZM array was devised by Gary Pillon, a sound mixer at General Television Network of Detroit, Michigan. A documentary recording he made with this array won an Emmy. The assembly can be stand-mounted from the backside or handheld, if necessary.

The stereo image, which is partly a result of the 8" capsule spacing, is designed to be like that produced by a binaural recording, but with more realistic playback over loudspeakers. Ideally, this device would mount on a Steadicam platform and give an excellent match between audio and video perspectives.

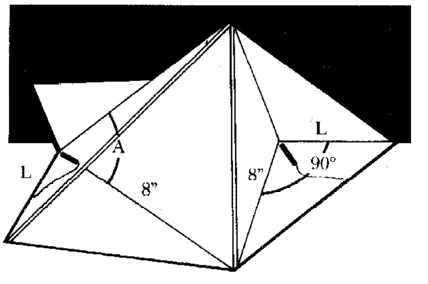


Fig. 50 – Pillon stereo PZM
L = 8" and A = 90° for speech use
L = 12" and A = 120° for music use

THE SASS® PZM STEREO MICROPHONE

As explained earlier, one way to record in stereo with PZMs is to mount two PZMs on a wedge: two 2'-square panels, angled apart to form a "V". This arrangement can be cumbersome. But recent research has led to a unique application using PZMs on a smaller head-size boundary: The Crown SASS microphone.

The Crown SASS® or Stereo Ambient Sampling System is a new kind of stereo microphone. It does an excellent job recording sounds in stereo, such as:

- orchestras, choirs, symphonic bands, pipe organs
- news events
- sports ambience and crowd reaction
- background sounds for films
- stereo samples for keyboards
- stereo sound effects

Before explaining how the SASS makes such good stereo recordings, let's describe how stereo itself works.

How Stereo Works

Normally you listen to stereo over two speakers, one placed in front of you to the left, and one to the right.

When you listen to a stereo recording of an orchestra, you can hear strings on the left, basses on the right, and woodwinds in the middle. That is, you hear an image of each instrument in certain locations between speakers (Fig. 51).

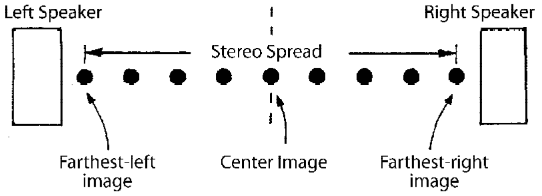


Fig. 51 – Stereo images.

If you send the same audio signal to the two speakers, you hear an image in the middle between the two speakers.

How do recording engineers make the images appear left or right?

One way is to make the signal louder in one channel than the other. For example, if you feed the same signal to both channels, but turn up the volume of the right channel, the image shifts to the right speaker.

Another way is to delay the signal in one channel. If you feed the same signal to both channels, but delay the left channel one millisecond, the image shifts to the right speaker.

So, various image locations can be created by recording loudness differences and/or time differences between channels. We want a sound source on the right to make a louder signal in the right channel than the left. Or we want a sound source on the right to make a signal sooner in the right channel than the left.

This is done with stereo microphone techniques. There are three basic stereo techniques; coincident pair, spaced pair, and near-coincident pair.

With coincident-pair miking, a pair of directional microphones is placed with grilles touching, one mic above the other, and angled apart (Fig. 52). A sound source toward the right will produce a stronger signal from the mic aiming toward it than from the mic aiming away from it. Thus, the right channel will be louder and you'll hear the image to the right.

With spaced-pair miking, a pair of microphones is placed several feet apart, aiming straight ahead (Fig 53). Sounds from a source toward the right will reach the right mic sooner than the left mic, simply because the right mic is closer to the sound source. Thus, the left channel will be delayed and you'll hear the image to the right.

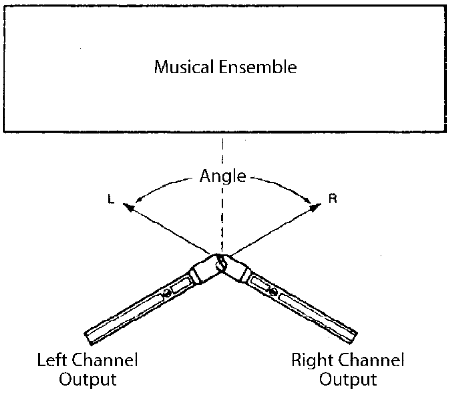


Fig. 52 – Coincident-pair stero miking.

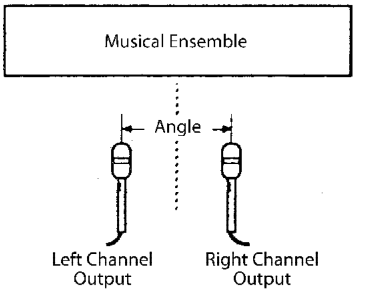


Fig. 53 – Spaced-pair stereo miking.

With near-coincident miking, a pair of directional microphones is angled apart and spaced apart a few inches horizontally (Fig. 54). A sound source on the right will be louder in the right channel AND delayed in the left channel. These two effects add together, so you'll hear the image to the right.

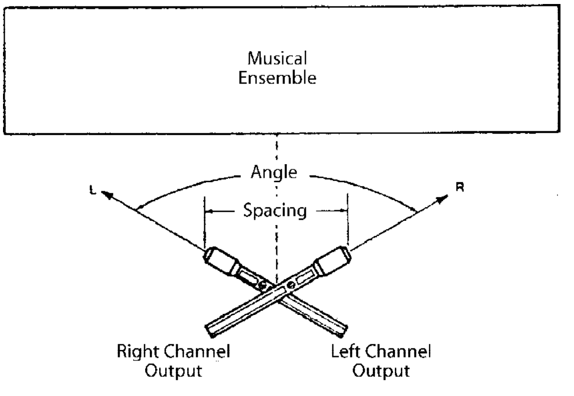


FIg. 54 – Near-coincident stereo miking.

How the SASS Works

Back to the SASS. It uses two small microphones spaced a few inches apart. Each microphone is on a surface that blocks sound from the rear, and these surfaces are angled apart (Fig. 55). In other words, the surfaces make the microphones directional. So the SASS is like a near-coincident pair, in which two directional mics are angled apart and spaced horizontally a few inches.

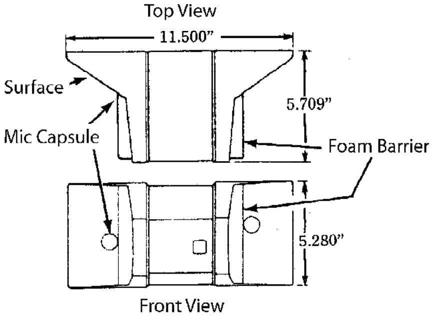


Fig. 55 – Parts of SASS

The surfaces make the microphones directional only at mid-to-high frequencies. At low frequencies, the microphones pick up all around them—they are omnidirectional.

The SASS produces stereo in different ways at different frequencies. At low frequencies, the SASS acts like a spaced pair, producing time differences between channels to make a stereo effect. At high frequencies, the SASS acts like coincident pair, producing mostly loudness differences between channels to make a stereo effect. At mid-frequencies, the SASS acts like a near-coincident pair, using both loudness and time differences to make stereo.

This is the same way the human hearing system works. Our ears are omnidirectional at low frequencies, directional at high frequencies (because the head blocks sounds), and are spaced apart a few inches.

Since the SASS hears sounds the same way our ears do, it produces very natural stereo with easy-to-localize images. It also gives a pleasing sense of spaciousness, a sense of the environment in which the sound was recorded. Both these attributes can be heard over loudspeakers or over headphones.

The coincident-pair method gives a narrow stereo spread over headphones. The spaced-pair method makes images that are poorly focused or hard-to-localize when heard with speakers (Fig. 56). The SASS has neither of these problems. It gives accurate, wide stereo over headphones, and makes images that are sharp and correctly placed when heard with speakers or headphones.

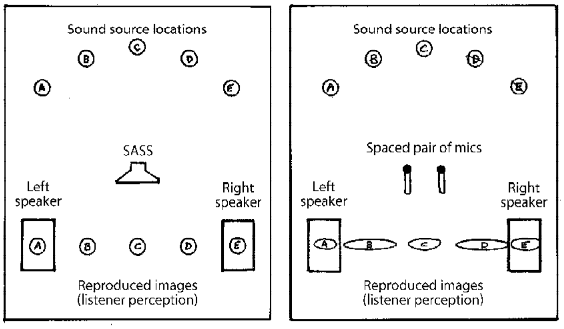


Fig. 56

The SASS Is Mono Compatible

It's important for stereo recordings to be mono compatible. That is, the tone quality must be the same whether the program is heard in mono or stereo.

With spaced-pair or near-coincident recordings, the microphones are spaced apart. Sound arrives at the two microphones at different times. Thus, the left and right signals are in phase at some frequencies, and out-of-phase at other frequencies. If the two channels are combined for mono listening, the out-of-phase frequencies cancel out. This makes dips in the frequency response (Fig. 57). The non-flat response gives a filtered, colored tone quality to whatever is recorded.

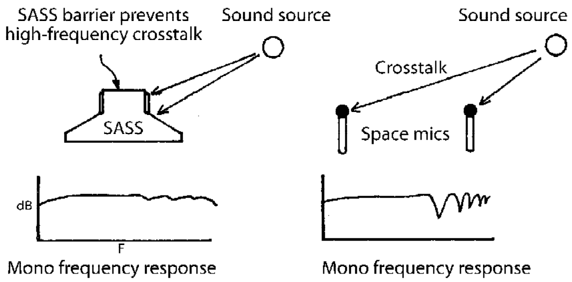


Fig. 57

Recordings made with the SASS do not have this problem. That's because it's made with a special block of dense foam between the mic capsules. This foam barrier absorbs sound. It prevents sound from the right side from reaching the left microphone, and vice versa. Thus, the signal is much louder in one channel than the other.

For a phase cancellation to be complete when two channels are combined to mono, the levels in both channels must be about the same. But the levels in both channels are different in the SASS (due to the foam barrier between capsules), so phase cancellation in mono is relatively slight (Fig. 57). Thus the tone quality stays the same in stereo or mono with the SASS.

Better Bass Response

All directional microphones have reduced output in the deep bass. Thus, stereo methods that use directional mics have weak deep bass. The SASS does not have this problem because it is not directional at low frequencies. It has excellent bass response extending to 20 Hz.

Simple to Use

Mid-side stereo microphones require a matrix box between the microphone and recorder. This box converts the mid and side signals from the microphone to left and right signals for stereo recording. The SASS already has left and right outputs, so it needs no in-line matrix box.

It's easy to tell where to aim the SASS by looking at it. In contrast, some stereo microphones are difficult to aim properly.

Excellent Performance

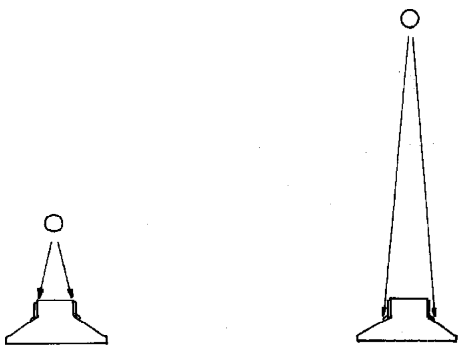
The SASS has very wide-range, smooth frequency response (20 Hz–18 kHz), and very low pickup of mechanical vibrations and wind noise.

Summary

The SASS is a stereo microphone using PZM® technology. The unit provides excellent stereo imaging, has a natural tonal balance, is mono-compatible, is easy to use, and costs less than the competition. It comes with a carrying case and a full line of accessories.

HOW TO USE THE SASS

Do not place the SASS closer than 3 feet from the sound source, or the center image will be weak or muffled (Fig. 58).



Center sound source too close to SASS: mic capsules can't "hear" it, because its sound is blocked. Result: weak center image.

Center sound source far from SASS: both mic capsules hear it. Results: strong center image.

Fig. 58

In live-to-2-track recording, we recommend that the final placement decisions be made while monitoring on loudspeakers for more-accurate imaging.

If the correctly monitored stereo spread is excessive (because of close mic placement), run the SASS signals through a stereo mixer with pan pots, and pan the two channels toward center until the stereo spread is correct. This can be done during recording or post-production.

Large musical ensembles (orchestra, band, choir, pipe organ)

Place the SASS 4 to 15 feet from the front row of musicians. Angle it down so that it will be aimed at the performers when raised, and raise it about 15 feet high on a microphone stand (Fig. 59). Closer placement to the performers will sound more edgy, detailed, and dry: farther placement will sound more distant, blended and reverberant. Try to find a spot where you hear a pleasing balance between the direct sound from the ensemble and the hall ambience.

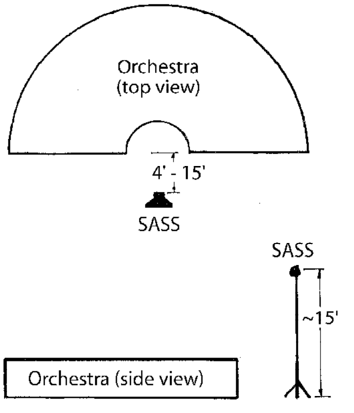


Fig. 59

Because the SASS is quite sensitive to the sides as well as the front, closer placement will not be as dry as with directional microphones. Hence, the SASS can be placed into an ensemble farther than is ordinarily possible, providing greater detail and spread, if that is desired, without feeling forced or unnatural. The center of the sound image and the hall reverberation are still retained.

If you are recording a choir that is behind an orchestra, experiment with the stand height to find the best balance between the two sources. The strings project upward while the choir projects forward, so you might find a better balance at, say, 9 feet high rather than 15 feet.

Small musical ensemble or soloist (quartet, small combo, background harmony vocals, solo piano, harp, or guitar)

Place the SASS 3 to 8 feet away at ear height. Move closer for less reverb and noise, farther for more hall acoustics. For a grand piano, place the SASS in-line with the lid. Placement near the hammers sounds more trebly; placement near the tail sounds more bassy.

Drum set

See Fig. 60. Place the SASS above the level of the snare drum, below the cymbals, aiming at the snare drum about 3 feet away, midway between the mounted tom and floor tom. You may need to boost a few dB around 10 kHz - 15 kHz. Add another microphone of your choice in the kick drum. The SASS also works well as an overhead mic.

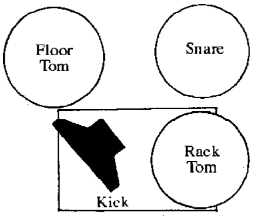


Fig. 60

Moving vehicles (car hood, handlebars)

We recommend that you devise a shock-mount system to be used under the microphone. Also be sure to put on the windscreen, and enable the low-cut switch (or use low-cut filters in your mixer).

Electronic News Gathering (E.N.G.)

You can often record an announcer and ambience with the SASS alone, without an extra handheld or lavalier mic on the announcer. If the ambient noise level is too high, use a mixer to blend a close-up microphone (panned to center) with the SASS.

The SASS will give a slight but not noticeable boost to the appropriate side if the talent moves away from frame center. If the SASS is camera mounted, use the windscreen to subdue wind noise caused by camera movement.

Because of its light weight, the SASS can be mounted on a fishpole, floor stand, boom stand, or tripod, in addition to the handgrip.

Samples and sound effects

If you sample a sound source in a particular off-center position, the SASS will accurately reproduce the image location. Recorded ambience will sharpen the image, but is not necessary.

If you record ambience along with the sample, the ambience will be reproduced whenever the sample is played. So you may want to make several samples of one source at different distances to include the range of added reverberance or ambience.

Suppose you're sampling in stereo and picking up ambience. If the sample is pitch-shifted, the direction of the images and perceived size of the room will be affected by most pitch-changing algorithms. You can minimize these effects by sampling at intervals of one-third octave or less.

When looping, try to control the room ambience so it is consistent before and after the sample (unless reverberant decay is desired as part of the sample).

When recording a moving sample or effect, experiment with the distance between the microphone and the closest pass of the sound source. The closer the SASS is to the path of the subject, the more rapidly the image will pass the center point (almost hopping from one channel to the other). To achieve a smooth side-to-side movement, you may need to increase the distance.

CHOOSING THE RIGHT CROWN BOUNDARY MICROPHONE

Application	Suggested Model
Recording	PZM-30D Rugged, use with detachable cable. PZM-6D Inconspicuous, permanently attached cable.
Multiple Boundary Recording/P.A.	PZM-6D Unscrew cantilever from plate.
Stereo Recording	SASS-P MK II Stereo PZM microphone.
Stage Floor	PCC-160
Conference Table Models MB-1, MB-2, and MB-4E require an MB-100 or MB-200 interface. The MB-200 interface allows remote sensing of switch closure, so it can be used with a video switcher.	PCC-170 Supercardioid. PCC-170SW Supercardioid with push on/off membrane switch. PCC-130 Small cardioid. PCC-130SW Small cardioid with push on/off membrane switch. Mini Boundary Microphones : Very small mics with a half-supercardioid pattern. Five models are available: MB-1 : Plugs into a brass cup in the table. MB-2 : Plugs into a jack in the table. MB-3 : Tubular; mounts in ceiling, wall or table. MB-4 : For temporary use. Has a thin cable with XLR connector. MB-4E : Lowest cost. Cable fits through small hole in table.
Piano Sound Reinforcement	PZM-30D, PZM-6D or PCC-160 on underside of lid.
Ambience	PZM-30D or PZM-6D on walls.
Security/Surveillance	PZM-11, PZM-11LL, PZM-10, PZM-10LL