Sound Recording

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Advanced Mixed Music Composition

Microphones

Types

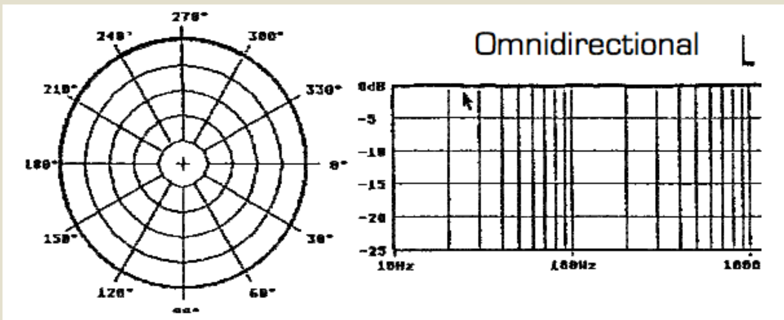
Dynamic: Operates via a diaphragm attached to a thin coil wrapped around a magnet. The magnetic flux of the coil moving in an electro-magnetic field creates a voltage.
• Pros: high moisture tolerance; rugged, simple construction; can handle very high sound pressure levels (good for drums, trumpets, etc)
• Cons: slower transient response, not very sensitive
• examples: Shure SM57 and SM58

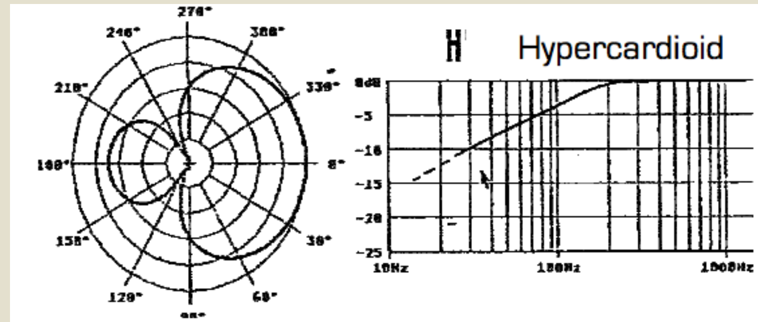
Condenser: Works on the principle of a capacitor: the diaphragm acts as one plate of a capacitor, and the vibrations produce changes in the distance between the plates. An electrical current (i.e. phantom power) is sent to the assembly, and as the diaphragm moves, the voltage of the current changes.
• Pros: quicker frequency response than dynamic microphones, especially to high frequencies (lighter diaphragm); very sensitive; fast transient response
• Cons: less rugged and more sensitive to temperature and humidity; operate using phantom power (or batteries)
• Examples: AKG C-414, Neumann 184, Shure SM-81

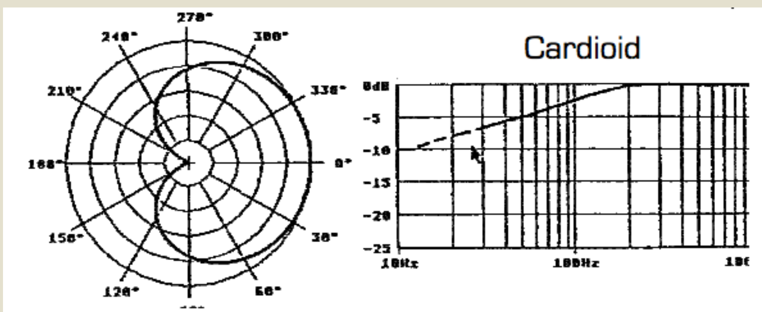
Ribbon: Works on same principle as dynamic microphone, but the diaphragm is a thin aluminum ribbon.
• Pros: although large, can pick up a lot of high frequency detail without sounding as harsh as condenser microphones; don't require phantom power.
• Cons: the most fragile of microphones
• Examples: RCA 44 and the AEA R44

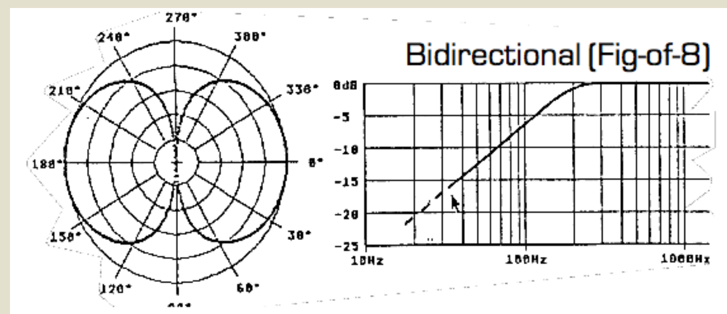
Microphones

Directional Patterns (a.k.a. polar pattern or directionality) and Frequency Responses









Note 1: Omnis generally have better low frequency response

Note 2: Polar patterns are frequency dependant

Note 3: The off-axis frequency response of directional mics varies significantly

Source: Williams, Michael. *Stereo Zoom*. 75th AES Convention in paris (1984), Preprint 2072

Microphones

Omnis vs. Directional

Polar pattern	Omnidirectional	Directional
Gain to feedback ratio	Lower	Higher
Feedback build-up	Slow	Fast
Off-axis colouration	Smooth and even	Typically less smooth
Proximity effect	No	Yes
Sensitivity to wind, handling & pop-noises	Low	Higher
Distortion	Low	Higher
Channel separation	Near field: Good Diffuse field: Less precise	Near field: Good Diffuse field: Good

Diaphragm Size

	Small Diaphragm	Large Diaphragm
Self Noise	Higher	Lower
Sensitivity	Low	High
SPL Handling capability	High	Lower
Frequency Range	Wide	Narrower
Influence on sound field	Small	Large
Dynamic Range	Higher	Lower

Also, small diaphragms generally have better fast transient response

Source: www.dpamicrophones.com/en/Mic-University/Technology-Guide/

Microphones

Choice of microphone should be done with regard to:

- Register and Nature of instrument
  - Should correspond to frequency characteristic of mic
  - See http://www.independentrecording.net/irn/resources/freqchart/main_display.htm
- Quality of recording room
- How much isolation do you need

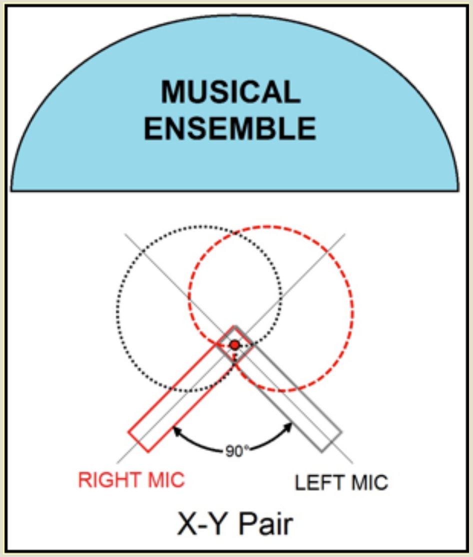
Stereo Micing Techniques

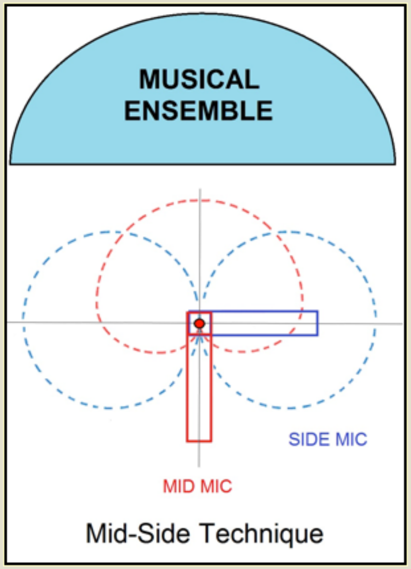
Coincident pairs

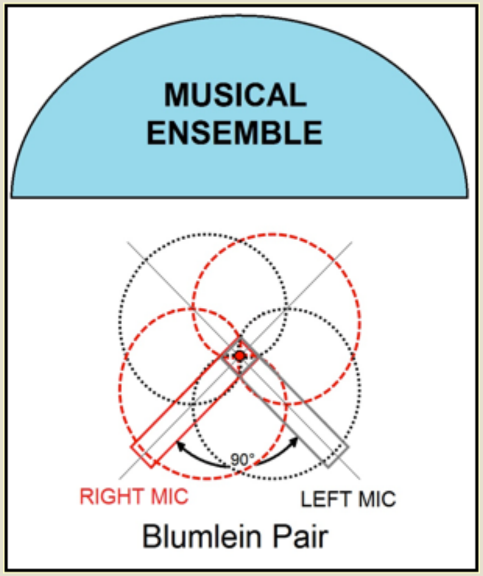
Two cardioid microphones (or fig 8 in the case of Blumlein pairs) are at the same place, typically pointing at an angle between 90° and 135° to each other. Captures IIDs only.

Pros:
• Mono compatible since it eliminates phase cancellation problems.
• Sound very focused in center (especially MS)

Cons:
• Stereo separation may be limited if the sound source is very wide.
• Relies heavily on off-axis response (colors the sound)







See Logic session “RecExamples” on how to decode MS technique

Stereo Micing Techniques

Near-Coincident pairs

Two directional microphones angled between 90° and 135° to each other. Captures IIDs and ITDS.

Pros:

• Very good separation

Cons:

• Less good mono-compatibility than coincident pairs

• Relies heavily on off-axis response (colors the sound)

Flavors:

ORTF (Office de Radiodiffusion-Television Francaise):
Cardioids, 17cm — 110°

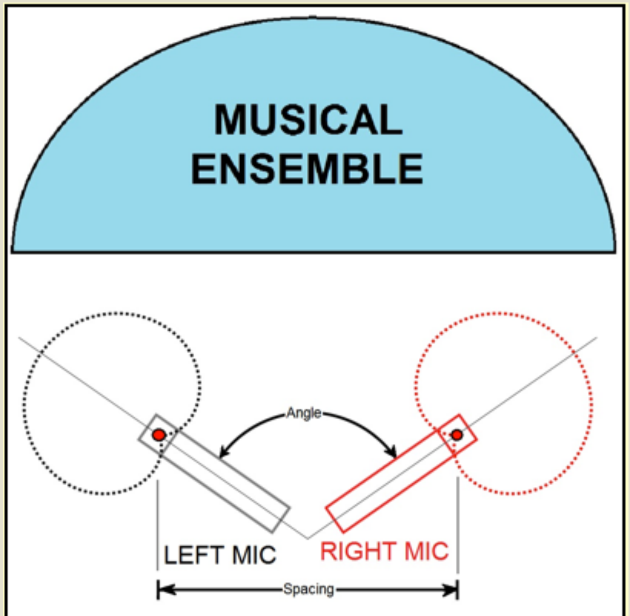
NOS (Nederlandse Omroep Stichting):
Cardioids, 30cm — 90°

RAI (Radio Audizioni Italiane):
Cardioids, 21cm — 100°

Faulkner array:
Fig 8, 20cm — 0° (very similar to spaced pair, could be used also with cardioids or hypercardioids)

Binaural:
Omnis inserted in ear canal of dummy head

Jecklin Disk:
Omnis spaced 16.5cm apart, with a 30cm sound absorbing disk in between



Near-Coincident Techniques

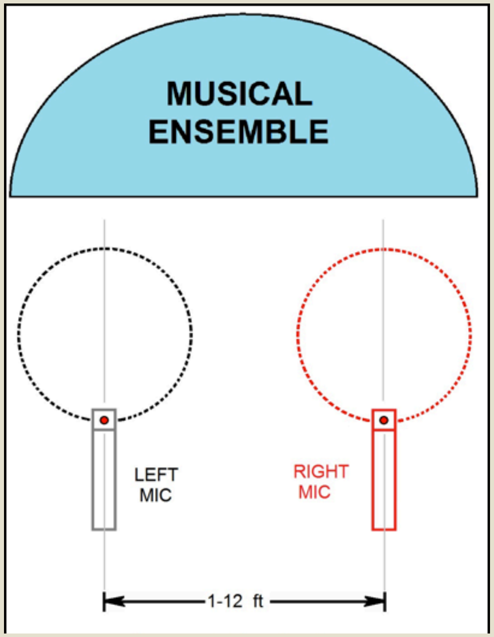
Stereo Micing Techniques

Spaced pairs

Two omni of cardioid microphones placed in parallel 1-12 feet apart. The distance depends on the size of the sound source. Captures both IIDs and ITDs.

Pros:
- Good frequency response (when used with omnis)
- Can cover wide source
- Captures much of the room sound (this is great if in a fantastic sounding room)

Cons:
- Poor mono-compatibility
- Captures much of the room sound (this is terrible if in a harsh sounding environment)
- Prone the "hole in the middle" effect (if mics are placed too far apart)



See Logic session "RecExamples" on how to decode MS technique

Stereo Micing Techniques

Stereo Recording Angle (SRA)

- Conceived by Michael Williams, it is defined as the "sector of the sound field in front of the microphone system which will produce a virtual sound image between the loudspeakers".
- Relies on Psychoacoustic data that describes combinations of IIDs and ITDs to "fool the brain" into perceive a sound source at a certain angle (based on **equilateral loudspeaker/listener setup**)

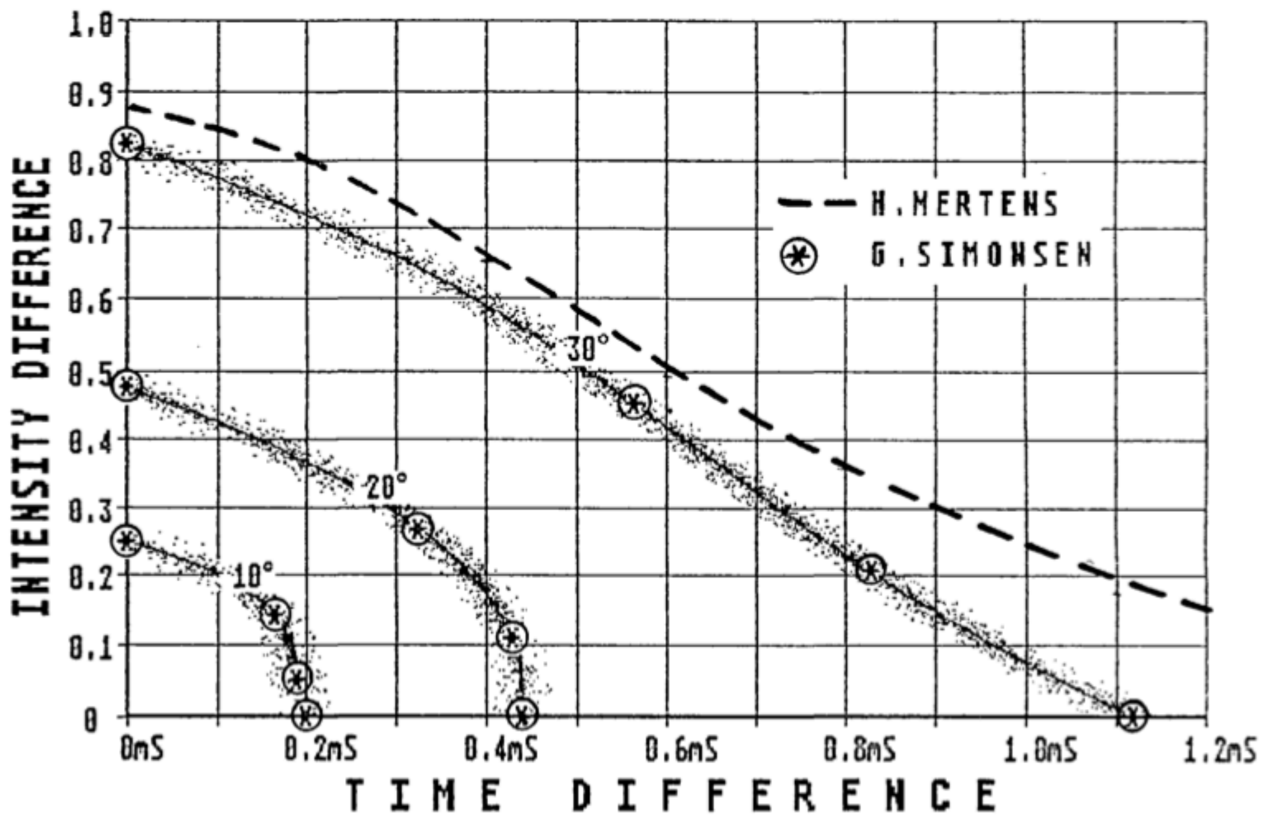


FIG.2

Source: Williams, Micheal. *United Theory of Microphone Sytems for Stereophonic Sound Recording*. 82nd AES Convention in paris (1987), Preprint 2466

Stereo Micing Techniques

Stereo Recording Angle (SRA)

• Williams extrapolated microphone angles and separation distances for given Stereo Recording Angles

• Every stereo micing techniques seen earlier is part of a larger continuum of possible microphone angle and distance combinations.

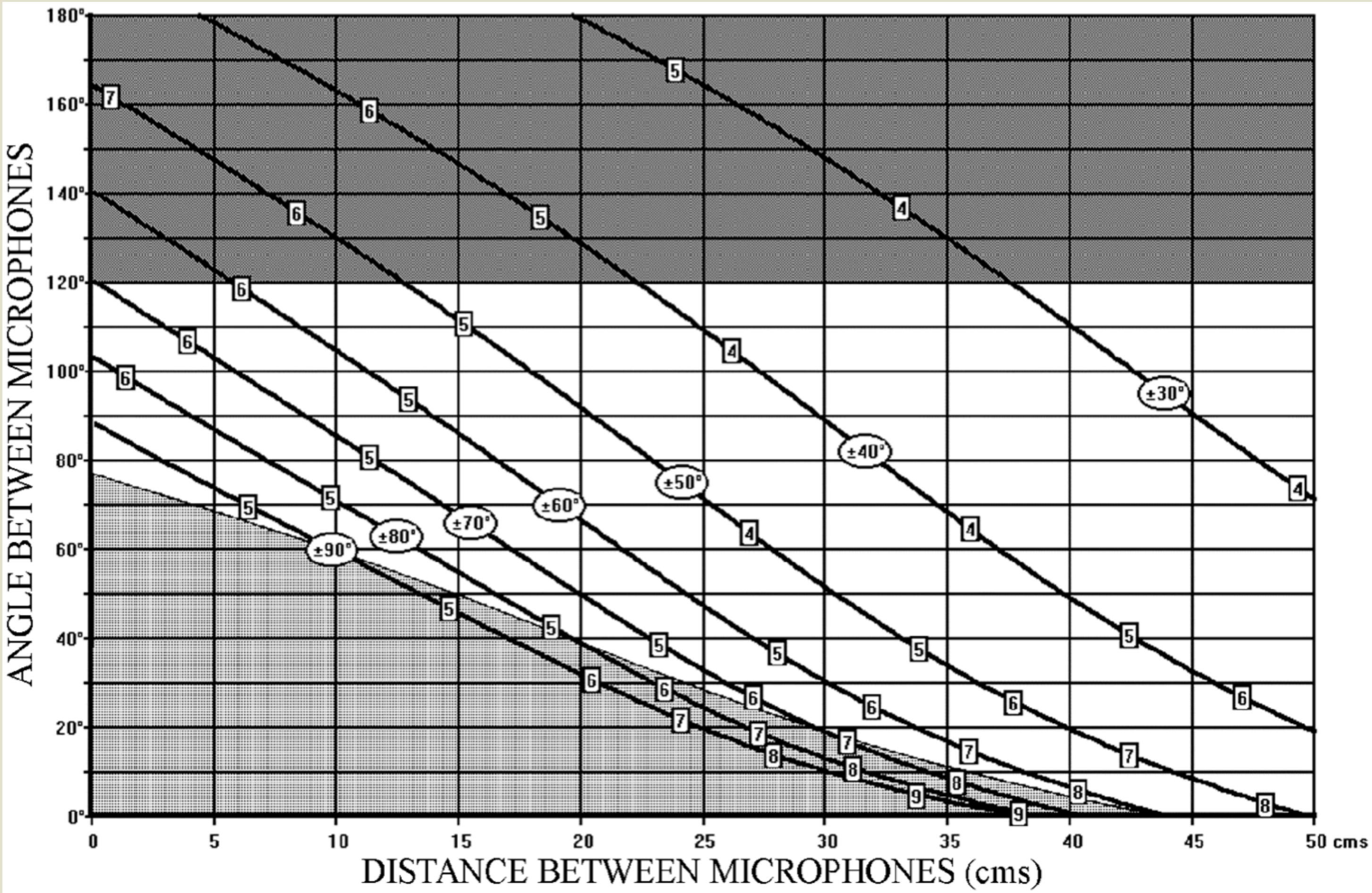


Figure 9 – SRA diagram for cardioid microphones
showing angular distortion and reverberation limits

Source: Williams, Micheal. *United Theory of Microphone Sytems for Stereophonic Sound Recording*. 82nd AES Convention in paris (1987), Preprint 2466

Stereo Micing Techniques

Stereo Recording Angle (SRA)

• Stereo micing will have compression and expension effects on the perceived stereo image vis-à-vis the actual size of the sound source

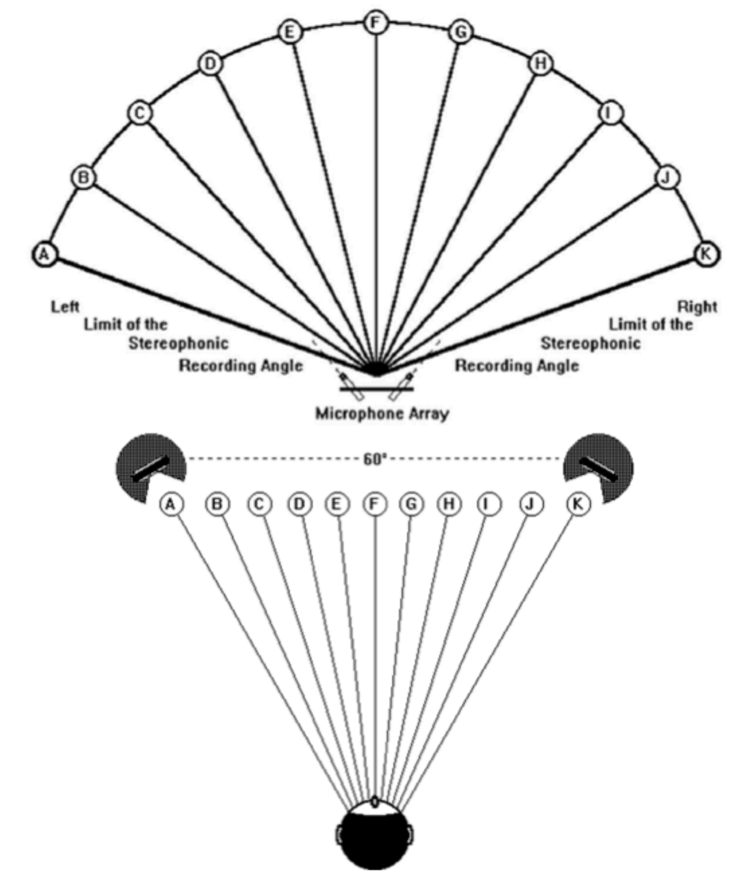


Figure 11 – Angular compression in reproduction (but without angular distortion)

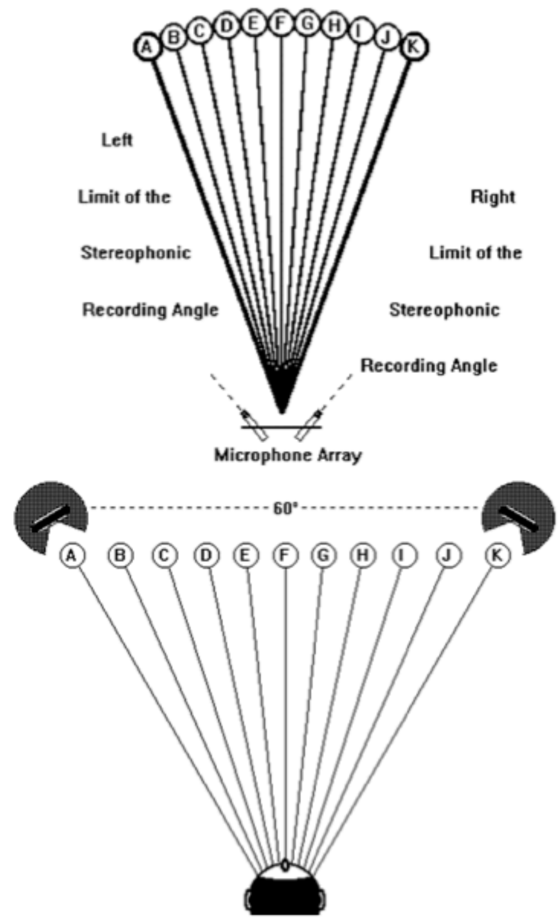


Figure 12 – Angular expansion in reproduction (but without angular distortion)

Source: Williams, Micheal. *United Theory of Microphone Sytems for Stereophonic Sound Recording*. 82nd AES Convention in paris (1987), Preprint 2466

Close Micing Issues

Proximity Effect

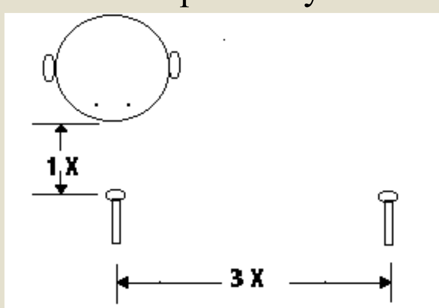
• An increase in bass response whenever a directional mic is brought within 1’ of the sound source. This bass boost (which is often most noticeable on vocals) proportionately increases as the distance decreases. If you want to compensate for this effect, use the low-frequency roll-off filter switch (if available).

Phasing

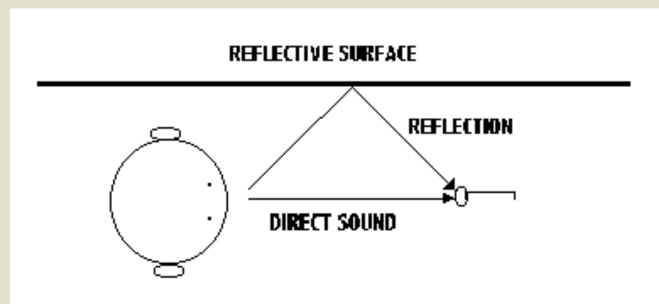
• Results from comb filtering that occurs when a signal is summed with a delayed version of itself.
• Situations where phasing can occur:

1-While recording

a) One source captured by two mics



b) Reflections from surround surfaces



• Can be mitigated, sometimes, by delaying one of the signals (only when the summing occurs post-recording)
• Use 3-1 rule as a guideline for avoiding phasing problems

*** Note that phasing can also occur while mixing:

a) Layering different sounds in the sequencer may produce comb filtering
b) Panning stereo recordings (especially non mono-compatible spaced pairs)

General Sound Recording Tips

- Get good levels, -6dB in 24bit is a good range to aim for
- Don't over compress when mixing
- Use a reference CD to build up your “studio ear” and compare what you’re doing
- Usually don't boost, cut
- EQ out portions of the sound that are not in the instrument's range
- You can clip at the track level, but you can never clip at the output (put a limiter at the output)
- Tame high peaks at the track level so that you don't have to limit too much on the output
- If you find yourself wanting to boost sounds past clipping, then it's not a level problem, but a spectral problem
- Aim for a well balanced sound, even low-mid-highs (watch out for the 2-5k range, where we are most sensitive; too much in that region can lead to listener fatigue)
- Don’t forget!!! use a reference CD

Impulse Responses

Definition

• An impulse response (IR) describes the reaction of the system as a function of time.
• In the audio world, IR most commonly refer to the reflective characteristics of a room, or the filtering properties of a particular system (can be a room or a device).
• IR are used in conjunction with convolution software.

How do we record them?

• Three typical methods:

1) Record and Impulse sound (balloon pop, starter pistol, wood stick slap, etc.) in the system
2) Record a sine sweep
3) MLS (Maximum Length Sequence) and IRS (Inverse Repeated Sequence) which use pseudo-random white noise (these are more complicated and will not be discussed further here).

Why would we want to record them?

• To create a "concert version" of your piece, but in the studio
• To create a virtual concert hall, to give an idea of how the piece will translate from the studio to the hall
• To create novel effects (instrumental resonances, object resonances, processing device emulation, etc.)

Impulse Responses

Technique 1: Impulse sound

Pros: Easy, quick, not much gear needed
Cons: Room must be very quiet, hard to meter because of fast transient, hard to repeat the exact same impulse sound

Technique 2: Sine Sweep

Pros: More forgiving to ambient noise (why?), easier to repeat, generally gives more accurate results
Cons: more involved gear wise, longer to record