FH MMA SALZBURG – AUDIO

MICROPHONES AND RECORDING TECHNIQUES

Figure 1: DNS 20-channel recording featuring: OSS main stereo pair (DPA 4006 with Jecklin Disk) and 18 support microphones (RODE NT 1000, Brauner Phantom, Audio Technica AT 3035)

1. TECHNICAL SPECIFICATIONS

1.1 CLASSIFICATION CRITERIA

A microphone, also called "mic" or "mike", is a transducer that converts sound waves into an electrical audio signal. Microphones can be classified per several criteria, such as:

- transducer type — condenser, dynamic, ribbon
- diaphragm size and address — small, large / front, side
- polar pattern — omni, cardioid, figure of eight, variable polar pattern, etc.
- technical specs — frequency and transient response, max SPL, sensitivity, S/N ratio, etc.

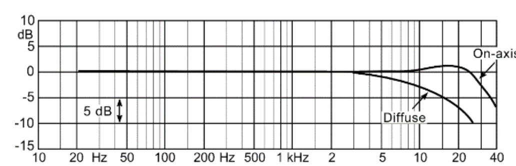
1.2 TECHNICAL SPECIFICATIONS

FREQUENCY RESPONSE

- The frequency response is usually measured "on axis", which means from a sound source right in front of the microphone; the "diffuse sound field" response might be quite different and affect how the microphone responds in a reverberant sound field.
- Depending on the purpose a microphone is built for, it might have a very flat frequency response, or have specific emphasis in certain ranges to fit specific purposes.
- Generally: omnidirectional microphones have flatter frequency response curves than cardioid types
- Condenser microphones in general have flatter frequency responses than dynamic microphones (this is due to the construction principle).
- Small diaphragm microphones usually have a wider frequency response than large diaphragm ones.

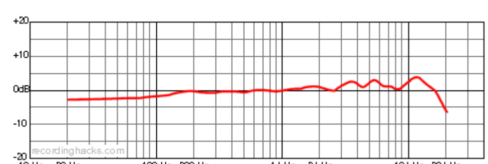
EXAMPLES:

- The **DPA 4006** (condenser, omni, small diaphragm) is a general-purpose instrumental/vocal microphone with an extremely flat frequency response from 20 to 20000 Hz.
  Its response is in fact so flat, that it can also be used as a measurement microphone.

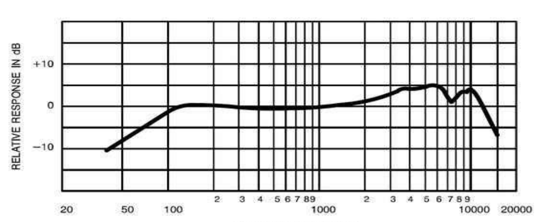


On-axis and diffuse-field frequency responses with pre-mounted Near-field Grid DD0251 fitted

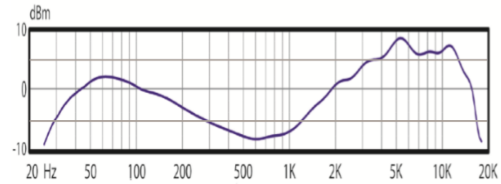
- The **RODE NT 1000** (condenser, cardioid, large diaphragm) is a general-purpose vocal and instrumental microphone, with a rather linear frequency response and very low self-noise.



- The **Shure SM 58** (dynamic, cardioid, small diaphragm) is a specialized vocal microphone; it features a "roll-off" in the bass range to compensate for the "proximity effect" (as it is usually used very close to the vocal source) and some emphasis in the 2-5 kHz range to increase clarity.



- The **Audix D-6** (dynamic, cardioid, large diaphragm) is a specialized bass-drum microphone and features a pronounced boost in the 50-100 Hz range (more bass drum “boom”) and another one in the 2-10 kHz range (more “click”).



TRANSIENT (OR IMPULSE) RESPONSE

- This quality defines how accurately the microphone can react to fast transients (= sudden changes in amplitude), as generated for example by drums, percussion or plucked instruments.
- Microphones with a lighter diaphragm (like condenser or ribbons types) react much better to transients than those with a heavier diaphragm construction (like dynamic microphones).

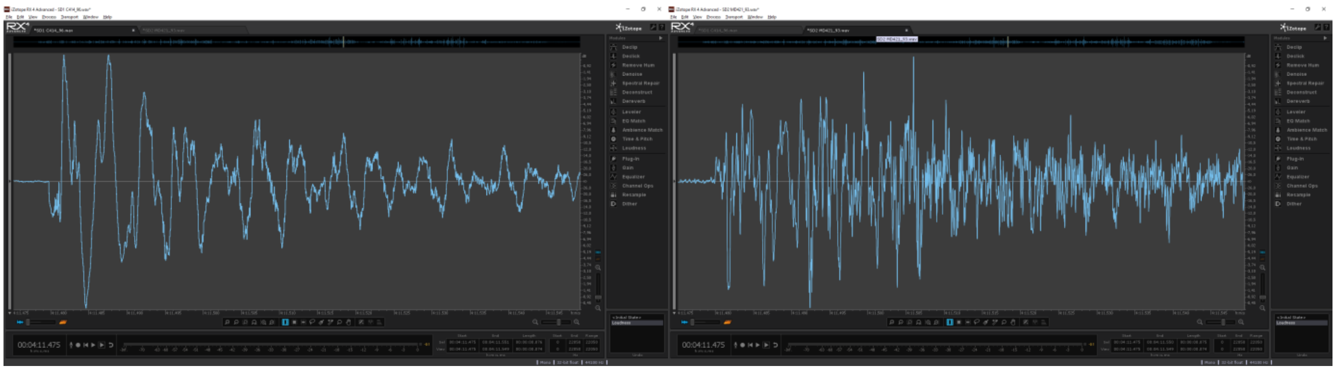


Figure 2: the same snare drum hit captured by a condenser mic (AKG C414, left) and a dynamic mic (Sennheiser MD 421, right)

- As you can see in the image above, that compares the transient response of a condenser mic (AKG C414) with that of a dynamic mic (Sennheiser MD 421), the initial part of the waveform looks very different (even if the mics were placed just a few cm apart); the condenser mic in this case captures the drum hit more accurately.
- This does not mean that the condenser microphone will be the best choice in every situation! Often the signal delivered by the dynamic microphone, while having a poorer transient and frequency response, might capture just what is musically relevant and fit better in the mix.

MAXIMUM SPL

- Measured in dB SPL, usually @ 1 kHz, 1% THD (Total Harmonic Distortion).
- This quality defines whether a microphone can also record very loud sound sources without distortion or even damage to the electronic components (for example, some microphones are designed to be placed in direct proximity of drums and percussion instruments).
- It is usually measured at 1 kHz with a tolerance of up to 1% THD (Total Harmonic Distortion); higher dB SPL values are preferable.
- Dynamic microphones have no electronics and their SPL limit is a physical limitation: the point where the diaphragm hits the magnet and can move no further.
- The Maximum SPL also dictates the higher limit of the microphone’s dynamic range.

EXAMPLES:

- the **RODE NT 1000** (condenser) has a maximum SPL of 140 dB, sufficient or record any orchestral instrument also from less than 1 distance; 140 dB SPL is the noise level of a Jet aircraft at about 50 m distance and is beyond the threshold of pain!

- the **Shure KSM 141** (condenser) is optimized for loud sound sources and has a max SPL of 164 dB; can be also placed in close proximity of a drum skin
- the **RODE NTR** (Ribbon) has a maximum SPL of 130 dB; it should not be placed too close to very loud sound sources
- the **Shure SM 57** (dynamic) has a maximum SPL of over 180 dB: that is like the noise level of the Space Shuttle launching, from about 10 m distance!!!

SENSITIVITY

- Measured in mV/Pa @1 kHz, this is the ratio between the input SPL and the level of the electrical output signal
- International standards have established 1 pascal (Pa) as 94 dB SPL: this reference point is now accepted for specifying the sensitivity of microphones.
- A higher sensitivity means less gain boost is required from the preamp to get from "microphone level" to "line level".
- Condenser microphones usually have a higher sensitivity than dynamic and ribbon ones.
- High sensitivity in combination with low Self-Noise levels are desirable when recording quiet sound sources, as they both contribute to a high S/N ratio.

EXAMPLES:

- the **RODE NT 1000** (condenser) has a sensitivity of 16 mV /Pa (-36 dBV /Pa) @ 94 dB SPL
- the **Brauner Phantom** (condenser) has a sensitivity of 28 mV /Pa (-31 dBV /Pa) @ 94 dB SPL
- the **Shure SM 57** (dynamic) has a sensitivity of 1,85 mV /Pa (-54,5 dBV /Pa) @ 94 dB SPL
- the **Royer R-121** (ribbon) has a sensitivity of 3,16 mV /Pa (-50 dBV /Pa) @ 94 dB SPL
- the **RODE NTR** (ribbon) has an unusual high sensitivity of 29,85 mV /Pa (-31,5 dBV /Pa) @ 94 dB SPL

EQUIVALENT NOISE LEVEL / SELF-NOISE

- The Equivalent Noise Level is the sound pressure level required to create the same voltage that the Self-Noise from the microphone will produce.
- The microphone Self-Noise is mainly caused by the active electronic components.
- The Self-Noise is also higher in small diaphragm microphones, compared to large diaphragm ones.
- Dynamic microphones do not have the electrical components that condenser microphones have. Therefore, they essentially have a noise floor below 5 dB SPL. However, dynamic microphones also have a much lower sensitivity, therefore the S/N ratio might still be worse to that of a good condenser microphone.
- Low Self-Noise level in combination with high sensitivity are desirable when recording quiet sound sources, as they both contribute to a high S/N ratio.
- The Self-Noise also dictates the lower limit of the microphone's dynamic range.

EXAMPLES:

- The **Rode NT 1000** (condenser, large diaphragm) has a self-noise of just 6 dB-A (one of the lowest of any microphone)
- The **Brauner Phantom** (condenser, large diaphragm) has a self-noise of 11 dB-A
- The **DPA 4006** (condenser, small diaphragm) has a self-noise level of 15 dB-A
- The **Rode NT 5** (condenser, small diaphragm) has a self-noise of 16 dB-A
- The **Shure SM 58** (dynamic, small diaphragm) has a self-noise below 5 dB-A

1.3 DIAPHGRAGM SIZE AND FRONT/SIDE ADDRESS

SMALL OR LARGE DIAPHRAGM

Depending on the purpose, microphones can feature small or large diaphragm transducers.

SMALL DIAPHRAGM

- Small diaphragm microphones usually have:
  - higher self-noise
  - lower sensitivity
  - a wider frequency range
  - a higher maximum SPL level
- A smaller diaphragm does not mean lower quality, or worse reproduction of low frequencies: in fact, quite the contrary is true!
- Some of the microphones with the most linear frequency response are small diaphragm condenser with omni polar pattern (like the DPA 4006 or Schoeps CCM 2 capsule)

Figure 3: Telefunken ELA M-260, a small diaphragm front-address condenser microphone

LARGE DIAPHRAGM

- Large diaphragm microphones usually have:
  - lower self-noise
  - higher sensitivity
  - a narrower frequency range
  - a lower maximum SPL level
- A larger diaphragm is not better at reproducing low frequencies, but it has the advantage of being less sensitive to the “proximity effect”.
- It might also be less precise at reproducing high frequencies, which might give the erroneous impression of having “more low-end”.

Figure 4: Telefunken U47, a large diaphragm, side-address condenser microphone with switchable polar pattern – arguably one of the best microphones ever built

FRONT OR SIDE ADDRESS

Front or side address depends on how the diaphragm is mounted inside the microphone.

FRONT ADDRESS MICROPHONE

- It accepts sound into the “end” of the microphone.
- The microphone is simply “pointed” towards the sound source.

SIDE ADDRESS MICROPHONE

- Also called “side fire” microphone.
- It accepts sound from an angle perpendicular to the microphone.
- The forward side of the microphone is usually where the company logo is placed (like for Neumann, AKG, Telefunken) or a dot (RODE).

2. MICROPHONE TRANSDUCER TYPES

2.1 CONDENSER MICROPHONES

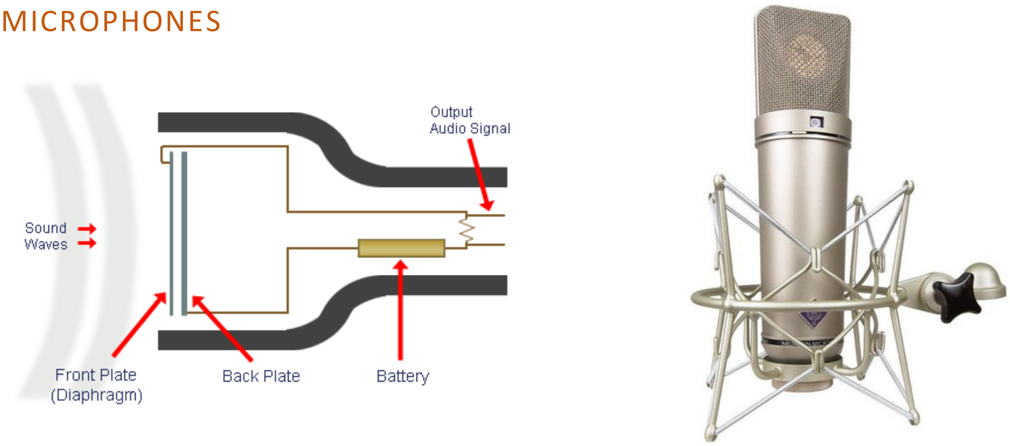


Figure 5: Cross section of a Condenser Microphone (left); Neumann U87 (right)

TRANSDUCER PRINCIPLE

In *Condenser Microphones* (also called *Capacitor Microphones*) the microphone diaphragm acts as one plate of a capacitor; the plates are biased with a fixed charge (typically 40-200 V) through a large resistance (>500 M Ohm). When the diaphragm vibrates in response to sound waves, it causes changes in the distance between the plates, which in turn affects the “capacitance” of the capacitor and produces accordingly variations in current at the resistance end. This current cannot be used directly as signal, due to the very high output resistance: for this purpose, a preamp amplifies the audio signal and transforms the impedance to about 200 Ohm.

Condenser microphones usually require 48 V phantom power for the capsule bias charge, as well as for the preamp. There are also models that require up to 140 V (for example the DPA 4004 high-SPL omnidirectional microphone requires 130 V).

Exception: in Electret Condenser Microphones (from *electro*static magne*t)* the plates are permanently charged (or “polarized”), therefore the phantom power is only required to operate the preamp.

CHARACTERISTICS

- Linear frequency response (especially the omni polar pattern), both in the very low and very high range.
- Very accurate impulse response (reacts well to fast transients, like in percussive sounds).
- Usually better S/N ratio than dynamic mics and low distortion.
- Overall: more transparent, detailed sound than dynamic mics, especially in the high range.
- High sensitivity, but lower headroom (maximum SPL level) than dynamic mics.
- Require phantom power, unless for models with an internal battery, or for “electret condenser” types.
- Double diaphragm condenser microphones are very flexible, as you can switch the polar characteristic between omni, cardioid and figure of eight; in some types, you can seamlessly blend through the different characteristics, with virtually infinite variations in polar pattern response.
- Not ideal for live usage: delicate, easier to get feedback because of wide frequency response, high sensitivity, etc.
- Nevertheless: should be used live for instruments with very strong energy in the high freq. range, like for example for cymbals ("drums overhead" configuration), tambourine, triangle, shaker, cabasa, etc.

EXAMPLES

- **large diaphragm condenser microphones**: Neumann U87, U89, M147, M149 TLM103; Audio Technica AT4050, AT3035; AKG C414; RODE NT 1000; Brauner VLM1, Brauner Phantom, Telefunken C-12, Telefunken CU-49, etc.
- **small diaphragm condenser microphones**: Neumann KM 183/184/185; AKG C480, C391B; RODE NT5, NT3; DPA 4006, etc.

2.2 DYNAMIC MICROPHONES

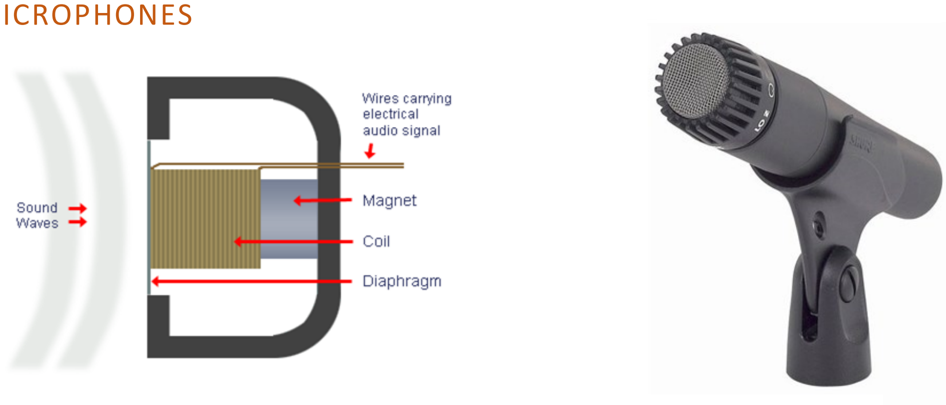


Figure 6: Cross section of a Dynamic Microphone (left); Shure SM-57 (right)

TRANSDUCER PRINCIPLE

In *Dynamic Microphones* (also called *Moving-Coil Microphones*) the microphone diaphragm is connected to a ring-shaped induction coil, which is wrapped around a permanent magnet. When the diaphragm vibrates in response to sound waves, the coil moves in the magnetic field, generating per induction a varying electrical current in the coil; this current has already a resistance of about 200 Ohm, so it can be used directly as audio signal.

The principle is similar to that of a loudspeaker, only reversed: in a loudspeaker, applying current to the coil moves the speaker membrane; in a dynamic microphone the movement of the diaphragm/coil generates current per induction.

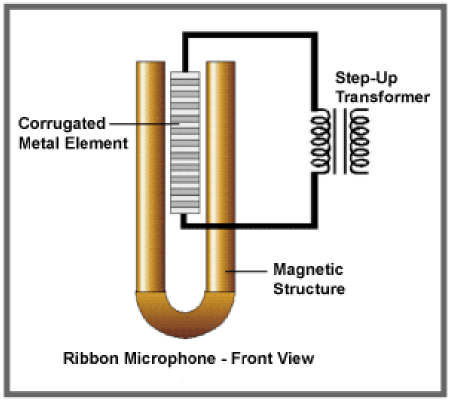
CHARACTERISTICS

- Poor response in the high frequency range ("roll off" might start around 15 kHz or earlier).
- The impulse response is not very accurate (dynamic mics do not react well to fast transients, due to the high mass of the diaphragm + coil).
- Usually worse S/N ratio than condenser mics.
- Overall: less transparent and detailed sound than condenser microphones.
- On the other hand: rounder, softer sound than condenser, therefore good for harsh signals (for example, a distorted e-guitar cabinet).
- Lower sensitivity than condenser mics, but much higher headroom (can be used in close proximity of very loud instruments, like drums or percussion, without clipping or being damaged).
- Do not require phantom power, or batteries.
- Very robust, easy to use in live P.A. situations
- Hard to get a feedback: as it is used very close to the sound source and requires less preamp gain boost.

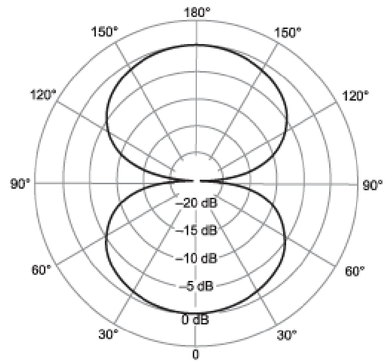
EXAMPLES

- **small diaphragm dynamic microphones**: Shure SM57, SM58 (vocals, snare drum); Sennheiser MD421 (snare drum, toms, Latin percussion, guitar amp), E865; Heil PR30 (snare drum, toms, guitar amp, vocals), etc.
- **large diaphragm dynamic microphones**: D112 (bass drum), Audix D-6 (bass drum), Telefunken M82 (bass drum)

2.3 RIBBON MICROPHONES



How a ribbon microphone works



Bi-directional polar or pickup pattern

Figure 7: Ribbon Microphone transducer principle (left) and polar diagram (center); Royer R-121 (right)

TRANSDUCER PRINCIPLE

Ribbon Microphones use a thin electrically conductive “ribbon” (a corrugated metal foil usually made of aluminum, duralu-minum or nanofilm) placed between the poles of a magnet; they produce a variable voltage by means of electromagnetic induction, therefore the principle has similarities to that of moving-coil dynamic microphones.

The main difference is that the ribbon is very light (unlike the heavy diaphragm + coil in dynamic microphones), therefore the sound character of ribbon microphones is quite similar to that of condenser microphones (detailed, good reproduction of higher frequencies), but with a softer top-end.

Most basic ribbon microphones detect sound in a bi-directional (figure of eight) pattern, because the ribbon is open on both sides (front and rear). Some models provide a unidirectional polar pattern, by adding an acoustic trap or baffle on one side of the ribbon.

Most ribbon microphones do NOT require 48 V phantom power (in fact, phantom power should not be turned on even accidentally, as it could damage older ribbon microphones). There are however exceptions: active ribbon microphones (such as the Royer Labs SF-2 and SF-24) include a preamplifier that boosts the signal to condenser microphone levels and require 48 V phantom power.

CHARACTERISTICS

- Detailed sound character, good response in the high frequency range.
- Softer/rounder top-end compared to condenser microphones.
- Good transient response.
- Relatively high self-noise and low sensitivity (except active ribbon types).
- Figure of eight polar pattern is common.

EXAMPLES

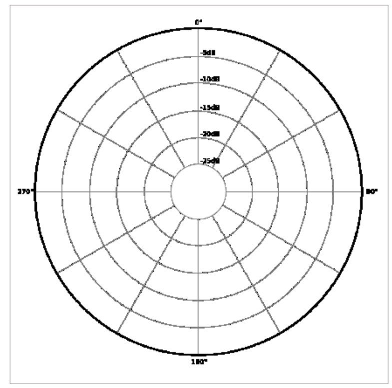
- **high end / reference class:** Royer Labs R-121, R-122, SF 12 (stereo), SF 24 (stereo)
- **high class / modern:** RODE NTR
- **mid class:** Cascade Fat Head, Fat Head II, X-15 (stereo, similar to SF 12)
- **budget / vintage:** No Hype Audio LRM-1, LRM-2

3. POLAR PATTERNS

3.1 POLAR PATTERN TYPES

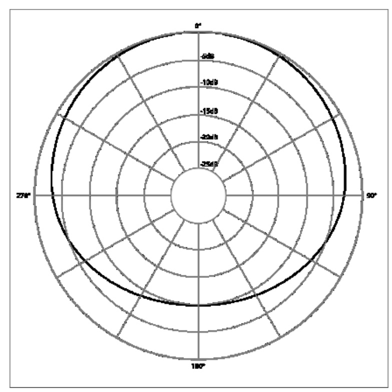
A microphone's *directionality* or *polar pattern* (German: *Richtcharakteristik*) indicates how sensitive it is to sound waves arriving at different angles about its central axis. In the following diagrams, the central axis (0°) is always shown pointing upwards, and the rear (180°) downwards.

OMNIDIRECTIONAL



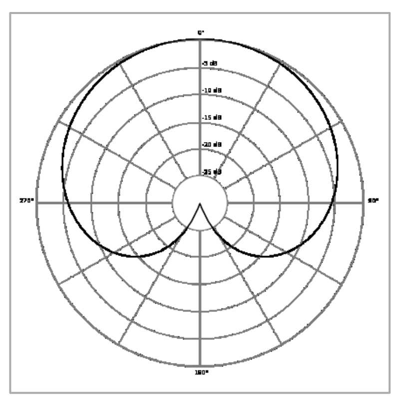
- German: *Kugel*
- Omnidirectional – pure pressure transducer.
- Theoretically, omni pattern microphones have the same sensitivity from sound coming from all directions, over the complete freq. range; this is not quite accurate: most omnidirectional capsules (for example, the DPA 4006) are more sensitive to high freq. for sound coming on-axis (0°), and could therefore be defined as being "mildly directional" in the high range.
- Omnidirectional microphones usually have a very flat response across the complete frequency spectrum.
- Example of stereo setup usage: A-B, OSS.

WIDE CARDIOID OR SUB-CARDIOID



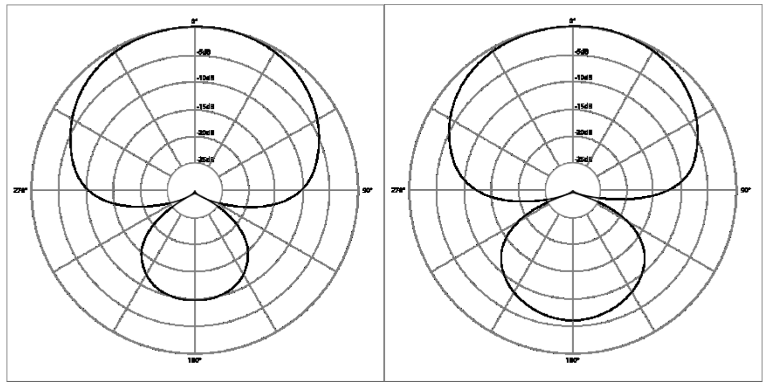
- German: *Breite Niere*
- A cross between Omnidirectional (Omni) and Unidirectional (Cardioid) patterns, with mild directivity.
- Very useful when you want to record instrumental groups in an orchestra and you wish to get enough separation, but without focusing too much on a single instrument in front of the microphone (like with a regular cardioid).

CARDIOID



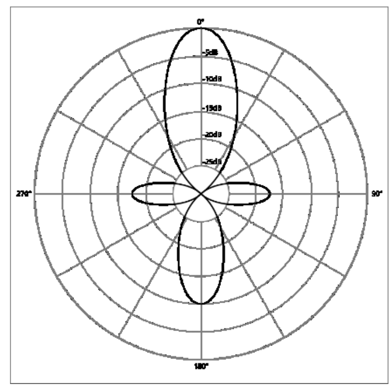
- German: *Niere*
- Unidirectional, with pronounced directivity.
- This polar pattern has maximum sensitivity on-axis (0°), -6 dB from the sides (90° and 270°) and minimum sensitivity from the rear (180°).
- This is one of the most common capsule types and can be used in countless situations. Standard close-up support microphone.
- Example of stereo setup usage: X-Y, ORTF, NOS.

SUPER-CARDIOID & HYPER-CARDIOID



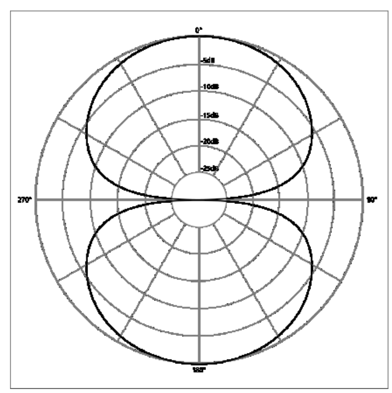
- German: *Superniere* & *Hyperniere*
- A cross between unidirectional (cardioid) and bidirectional (figure of eight), with strong directivity.
- These polar patterns have a stronger directional sensitivity than the cardioid: there is more attenuation from the sides (-10 to -15 dB), but they also react to signals from the rear (180°), similarly to the figure of eight.
- Note: the rear-lobe response (lower part of the diagram) is out-of-phase.

SHOTGUN



- A cross between unidirectional (cardioid) and bidirectional (figure of eight), with extreme directivity.
- Example of usage: mounted on video cameras, for location recording (example: to interview somebody in the middle of a noisy crowd).

FIGURE OF EIGHT



- German: *Achter Charakteristik*
- Bidirectional pattern – pure pressure gradient transducer.
- These microphones types have two symmetrical sensitivity lobes, with maximum sensitivity on-axis (0°) and from the rear (180°); minimum sensitivity from the sides (90° and 270°).
- Note: the rear-lobe response (lower part of the diagram) is out-of-phase.
- Example of stereo setup usage:
  - Blumlein (2 bidirectional microphones),
  - M/S (a bidirectional microphone is used for the S+/S- signals, in combination with a directional microphone for the M signal,

3.2 CONSTRUCTION PRINCIPLES

OMNIDIRECTIONAL

- **Pressure Transducers** (German: *Druckempfänger*): the output voltage is proportional to the variations in sound pressure. An omnidirectional capsule enclosure is like a closed can, with the diaphragm sealing the front end.
- The air pressure inside the sealed microphone capsule is always constant, therefore the diaphragm only reacts to outside variations in sound pressure.
- The diaphragm always responds in-phase to sound pressure variations, regardless whether sound waves come from the *front*, *side* or *rear*.

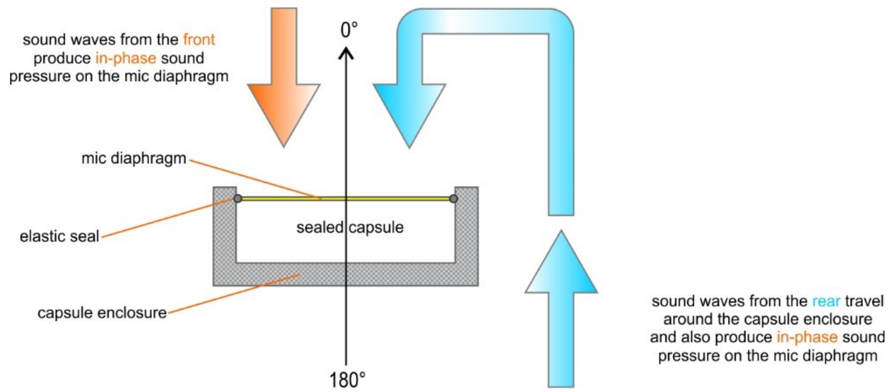


Figure 8: Cross-section of a Pressure Transducer (Omnidirectional) microphone capsule

BIDIRECTIONAL (FIGURE OF EIGHT)

- **Pressure Gradient Transducers** (German: *Druckgradientenempfänger*): the output voltage is proportional to the *pressure differential* (gradient) between the two sides of the mic diaphragm.
- A bidirectional capsule is open to both ends: when incident sound waves arrive on-axis (0°) or completely off-axis (180°), the pressure differential on the mic diaphragm is at its maximum; the microphone reacts with the same sensitivity to sound waves coming from the *front* (max in-phase response) or from the *rear* (max out-of-phase response).
- When incident sound waves are coming from the *side*, the pressure differential is null, as the sound waves produce equal amounts of sound pressure on *both* sides of the diaphragm (in-phase as well as out-of-phase); this causes phase cancellation and results in minimal sensitivity from the side.

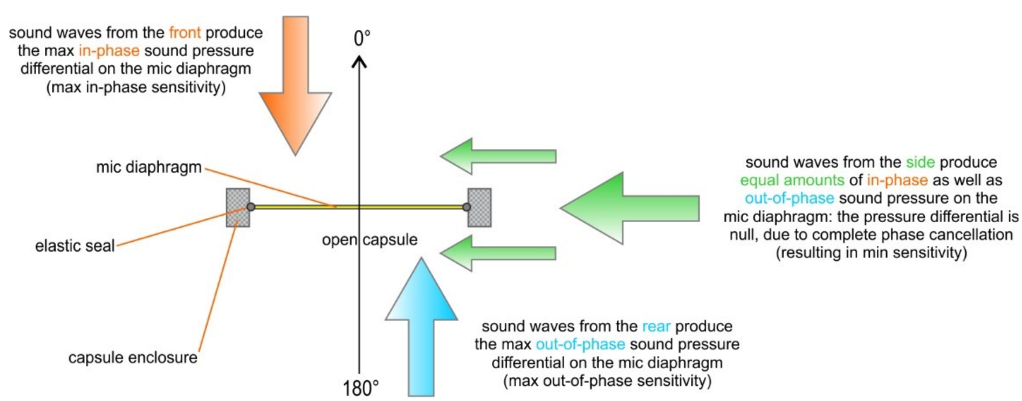


Figure 9: Cross-section of a Pressure Gradient Transducer (Bidirectional / Figure of Eight) microphone capsule

UNIDIRECTIONAL (CARDIOID)

- **Unidirectional Microphones** are conceptually a **superposition of a pressure transducer** (*omni*) **and a pressure gradient transducer** (*figure of eight*).
- From a construction point of view, unidirectional microphones use a partially sealed capsule enclosure with specially designed side/rear vents, which allow part of the sound waves coming from the side or the rear of the microphone to reach the rear of the diaphragm (with a short time-delay due to the longer signal path).
- Sound waves coming from the *front* produce the highest in-phase pressure differential on the diaphragm; this results in maximum sensitivity from the front.
- Sound waves coming from the *side* mostly reach directly the front of the mic diaphragm (in-phase), but travel also through the side/rear vents and reach the rear of the diaphragm (out of phase); this causes a reduced pressure differential due to partial phase cancellation and results in reduced sensitivity from the side (damping: -6 dB for cardioid; -10 dB for super-cardioid; -15 dB for hyper-cardioid).
- Cardioid: Sound waves coming from the *rear* produce equal amounts of in-phase as well as out-of-phase sound pressure on the diaphragm, causing complete phase cancellation: the pressure differential is null and results in minimal sensitivity from the rear.
- Super-Cardioid and Hyper-Cardioid: Sound waves coming from the *rear* produce more out-of-phase than in-phase sound pressure on the diaphragm, causing an out-of-phase sensitivity from the rear (but less pronounced than from the front).

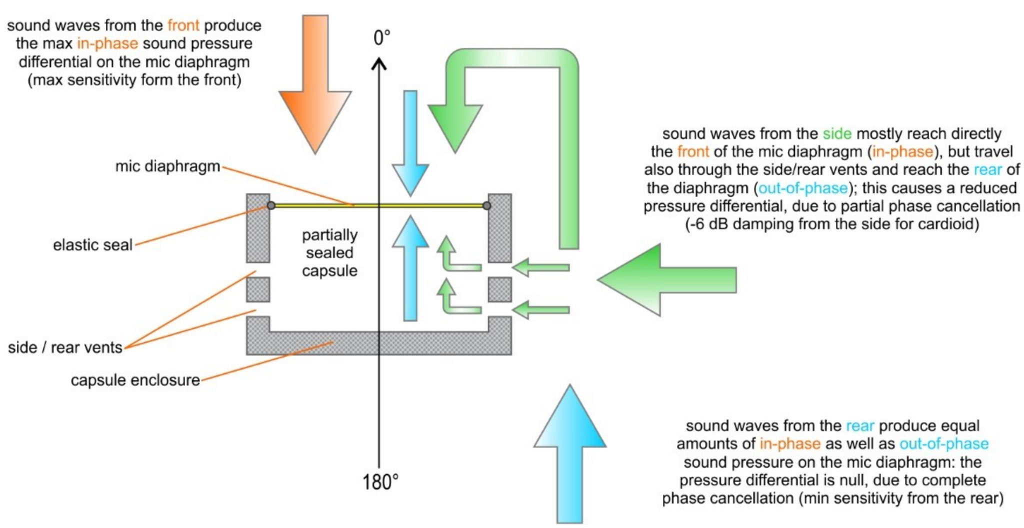


Figure 10: Cross-section of a Pressure Gradient Transducer (Unidirectional / Cardioid) microphone capsule

Figure 11: Telefunken TK61 omnidirectional capsule (left) and
TK60 cardioid capsule (right), showing the characteristic side/rear vents (slits)

4. STEREO RECORDING TECHNIQUES

4.1 OVERVIEW OF STEREOPHONIC RECORDING PRINCIPLES

INTERAURAL TIME DIFFERENCE (ITD)

- Example: **A-B**
- The principle is based on differences in *time delay* between the L and R channel, that are caused by the spacing between the 2 microphone capsules (incident sound waves at different angles will reach the two microphone capsules with varying time delays).
- German: *Laufzeitstereophonie*.

INTERAURAL AMPLITUDE DIFFERENCE (IAD)

- Examples: **X-Y**, **M/S, Blumlein**
- The principle is based on differences in *amplitude* (peak, level) between the L and R channel, that are caused by the varying sensitivity of the cardioid polar pattern at different incident angles.
- Also called **Interaural Level Difference (ILD)** stereophony.
- German: *Pegeldifferenzstereophonie*.
- The German term *Intensitätsstereophonie*, often found in German publications, is inaccurate and should not be used: the IAD principle is based on differences in *amplitude* (peak, level) between channels, and not on differences in *intensity* (= sound power per unit of area).

COMBINATION OF ITD AND IAD

- Examples: **ORTF**, **NOS**, **OSS**
- Our hearing system works in a similar way to the OSS (Optimal Stereo Sound) setup, using omnidirectional transducers (our ears) and a combination of ITD and IAD to determine the direction of a sound source (see under for details).
- In addition, our hearing also process differences in sound color / filtering caused by the outer ear (*pinna*, or *auricle*), which lets us gather additional information about the sound source, such as its 3D spatial position (height and distance information).
- The max ITD between our ears (that are on average 17 cm apart) is about 0,5 ms; therefore, both ORTF and OSS use exactly this distance between the microphone capsules.

4.2 INTERAURAL TIME DIFFERENCE (ITD)

A-B

SETUP

- 2 *omnidirectional* microphones
- Distance between capsules: 40 to 80 cm (as main microphones); up to 2-3 m (for capturing ambience sound)
- Angle between capsules: usually 0° (parallel)

SOUND CHARACTERSTICS

- Wide stereo image, but poor localization.
- The L-R signals are not correlated (different phase, as the mic capsules are not coincident) and not mono-compatible.
- If an A-B recording must be converted into mono, the best solution is to use just one of the two channels.
- The impression of having a “hole in the middle” of the stereo image might occur when the microphones are very far apart (2-3 m), but this is not an issue when recording ambience / room sound with an A-B setup.
- A-B sounds “roomier” than X-Y or M/S: due to the omnidirectional pattern of the microphones, more room ambience information is being captured.

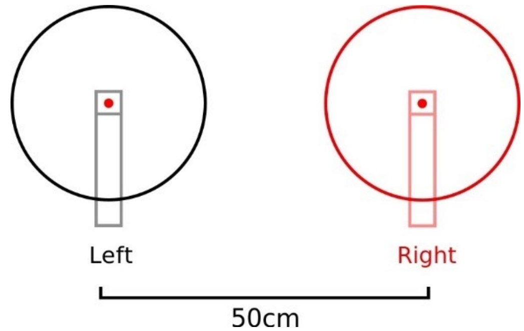


Figure 12: A-B stereo setup with DPA 4006

4.3 INTERAURAL AMPLITUDE DIFFERENCE (IAD)

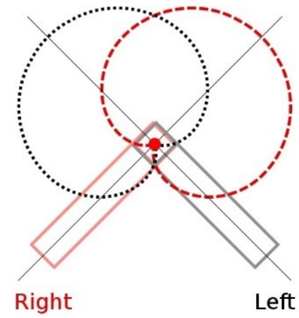
X-Y

SETUP

- 2 *cardioid* microphones
- Distance between capsules: 0 cm (coincident)
- Angle between capsules: 60° to 120° (typical 90°)

SOUND CHARACTERISTICS

- Relatively narrow stereo image, but excellent localization.
- The L-R signals are perfectly correlated (same phase, as the mic capsules are coincident), therefore excellent mono-compatibility.
- X-Y sounds “drier” than A-B or OSS: due to the cardioid pattern of the microphones, less room ambience information is being captured.



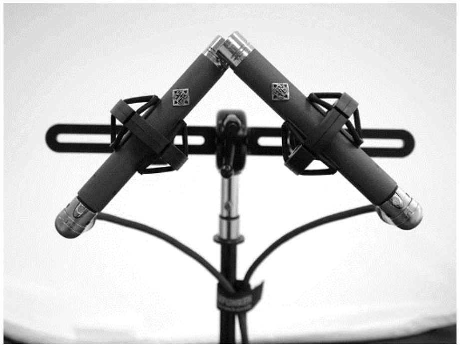
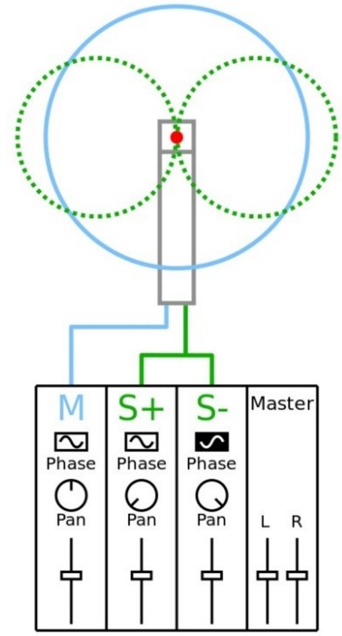


Figure 13: X-Y stereo setup with Telefunken ELA M-260

M/S (*Mid/Side*)

SETUP

- A *cardioid* microphone for the **Mid** signal and
  a *figure of eight* microphone for the **Side** signal
- The diaphragms are angled 90° from each other



DECODING AND BALANCE CONTROL

- To decode a M/S group you use three mixer channels:
  - one for the Mid signal, panned center
  - one for the S+ (in-phase) signal, panned full Left
  - one for the S- (out-of-phase) signal, panned full Right
- S- (out-of-phase) is the same as the S+ (in-phase) signal, but reversed in phase; phase polarity switches might not be available on low-budget mixers.
- Adjusting the balance between the Mid and Side signals, it is possible to seamlessly blend between mono and full stereo signal (there is more control than with a X-Y setup).

SOUND CHARACTERSTICS

- Relatively narrow stereo image, but excellent localization.
- Perfect mono compatibility, therefore M/S is an excellent choice for radio and TV productions.
- The S+ and S- signals are opposite in phase and erase each other when the final L-R stereo (decoded) channels are mixed together, leaving out only the Mid signal. In other words:

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

Figure 14: Blumlein stereo setup with Cascade Fat Head II

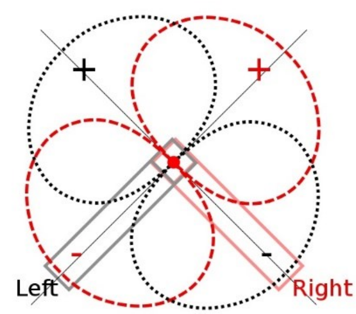
BLUMLEIN

SETUP

- 2 *bidirectional* microphones
- Distance between capsules: 0 cm (coincident)
- Angle between capsules: 90°

SOUND CHARACTERISTICS

- Realistic stereo image, with excellent front localization
- The L-R signals are perfectly correlated (same phase, as the mic capsules are coincident), therefore excellent mono-compatibility.
- More ambience information than X-Y, due to the rear (reversed phase) sensitivity
- Unfortunately, any instrument placed in the rear appears on the wrong side of the stereo field, and phase reversed.

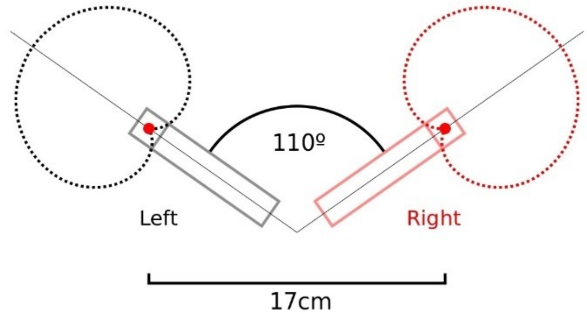


4.4 COMBINATION OF ITD AND IAD

ORTF (*Office de Radiodiffusion-Télévision Française*)

SETUP

- 2 *cardioid* microphones
- Distance between capsules: 17 cm
- Angle between capsules: 110°
- You can experiment varying the distance (15-30 cm) and the angle (60-120°) between the mic capsules
- Remember: to maintain a similar recording stereo base, distance and angle between capsules should be adjusted (inversely proportional)
  - smaller distance -> wider angle
  - greater distance -> narrower angle



SOUND CHARACTERSTICS

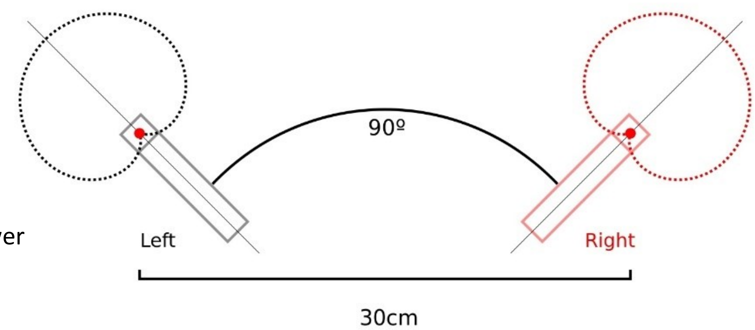
- Balanced stereo image, with good localization.
- The L-R signals are still quite correlated, due to small ITDs between the mics, therefore still acceptable mono compatibility.
- ORTF sounds “drier” than A-B or OSS: due to the cardioid pattern of the microphones, less room ambience information is being captured.

Figure 15: Superlux S-502 ORTF microphone

NOS (*Nederlandse Omroep Stichting* – English: Dutch Broadcast Foundation)

SETUP

- 2 *cardioid* microphones
- Distance between capsules: 30 cm
- Angle between capsules: 90°
- Similar principle to ORTF, but the capsules are more far apart, while the angle is narrower



SOUND CHARACTERISTICS (COMPARED TO ORTF)

- Due to the wider spacing between the mic capsules, NOS sounds wider than ORTF, but localization is not as accurate.
- Less correlation between the L-R signals, therefore less mono-compatible.
- Due to the smaller angle between the capsules, the microphones are pointed more towards the recording source, which can result in a small improvement in sound color accuracy.

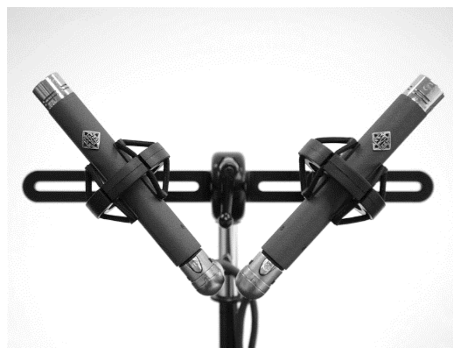
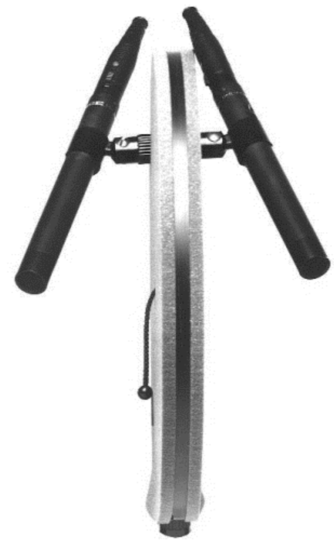


Figure 16: NOS stereo setup with Telefunken ELA M-260

OSS (*Optimal Stereo Signal*)

SETUP



- 2 *omnidirectional* microphones
- Distance between capsules: about 17-20 cm
- Angle between capsules: 30-45°
- The "Jecklin Disk" (about 30-35 cm in diameter) is placed between the microphones to dampen the mid/high frequencies of side sound waves and create the required interaural amplitude differences.

SOUND CHARACTERISTICS

- Balanced stereo image with very natural localization.
- OSS captures nice, deep and spatial sound, due to the omni polar pattern of the microphone used.
- The L-R signals are still quite correlated, due to small ITDs between the mics, therefore still acceptable mono compatibility; with greater distance between the capsules, stronger coloration (comb filtering) might occur.
- Although the principle might appear to be similar, OSS should not be confused with *binaural recordings* done with a "dummy-head", which are only compatible for reproduction over headphones!

Figure 17: OSS setup with Jecklin Disk

5. PRACTICAL APPLICATIONS AND RECORDING SETUPS

5.1 POTENTIAL ISSUES THAT CAN AFFECT THE RECORDING QUALITY

PROXIMITY EFFECT

- The “Proximity Effect” is an undesired boost of the lower frequencies that occurs when a microphone with directional polar pattern is placed very close to the sound source (quite noticeable under 30 cm distance).
- The frequency response of the “Gradient Component” of a directional microphone would normally increase 6 dB per octave; to compensate for this, the diaphragm is damped -6 dB per octave.
- As long as the sound source is far from the microphone (Far-Field Response), the Gradient Component is considerably larger than the Inverse Square Component (the sound pressure level of this component drops rapidly, -6 dB for each doubling of the distance) and the Overall Frequency Response is linear (with damped diaphragm).
- The Inverse Square Component is however already linear in response; due to the diaphragm equalization (-6 dB damping per octave), it appears to have much more energy in the low frequency range.

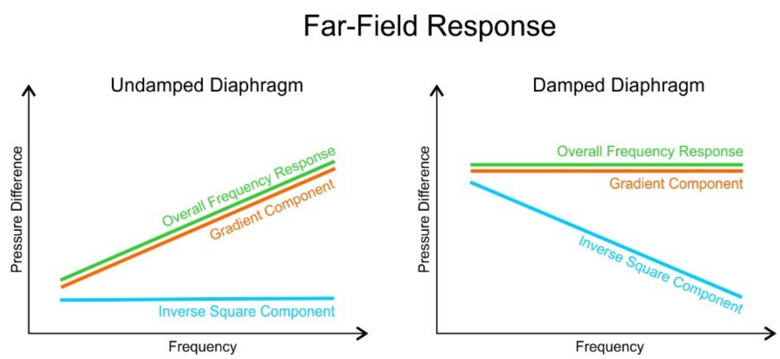


Figure 18: Far-Field Response of a Pressure Gradient Microphone, showing the frequency response of the Inverse Square Component and the Gradient Component, with Undamped and Damped Diaphragm. The Overall Frequency Response only depends on the Gradient Component.

- As the sound source moves closer to microphone, the Inverse Square Component becomes larger, eventually overtaking the Gradient Component. Once this happens, the Inverse Square Component contributes to the Overall Frequency Response causing a boost of the lower frequencies.

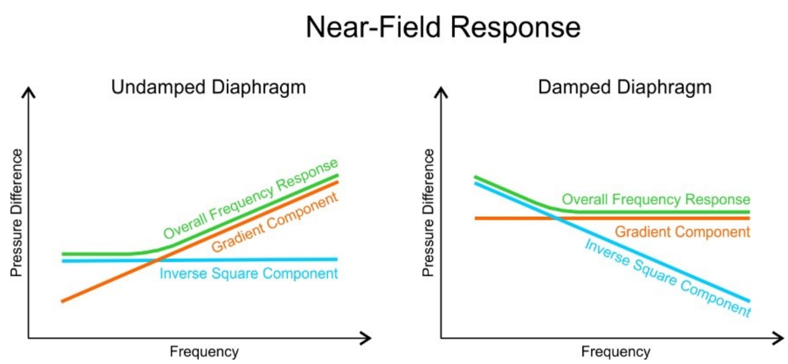


Figure 19: Near-Field Response of a Pressure Gradient Microphone, showing the frequency response of the Inverse Square Component and the Gradient Component, with Undamped and Damped Diaphragm. The Inverse Square Component is larger than the Gradient Component and contributes to the Overall Frequency Response causing a boost of the lower frequencies.

- Large diaphragm microphones are not as sensitive to the proximity effect: therefore, they are ideal for close vocal or instrumental recording.
- True Omnidirectional (single diaphragm) microphones are not affected by the proximity effect.
- Some directional microphones (like the Shure SM-57 and SM-58) are designed to be used extremely close to the sound source (voice, drums, percussion, guitar amps, etc.) and feature an optimized frequency response with a "roll-off" in the low range to compensate for the proximity effect, as well as a boost in the mid-high range to add clarity. Other models (like the Neumann U87, or the AKG C414) feature a switchable low-cut filter.

CLIPPING AND DISTORTION

- Consider the SPL range of the sound source (quiet, loud) to be recorded and choose your microphone carefully!
- Remember: condenser and ribbon microphones are usually topping around 135-140 dB maximum SPL, while most dynamic ones can easily handle 160-170 dB SPL.
- When recording drums very close to the drum skin, peaks beyond 140 dB SPL can occur.
- When recording guns and rifles with live ammunition (example: sound design for a videogame or a movie), peaks beyond 140 dB SPL at 1m distance can occur.
- Closer to the muzzle, SPL peaks up to 160-180 dB SPL can occur. As can be expected, such peaks can not only destroy most microphones, but also cause irreparable damage to the hearing. You should in any case wear hearing protection when working with high SPL sound sources!
- Even if the microphone is not damaged, when the SPL exceeds the maximum that can be handled by the electronic or mechanic components, clipping and distortion will occur.
- In the past, this meant the recording was compromised beyond repair. Nowadays it is possible to partly reconstruct the peaks of a clipped recording using special Sound Restoration Software (such as iZotope Studio RX)

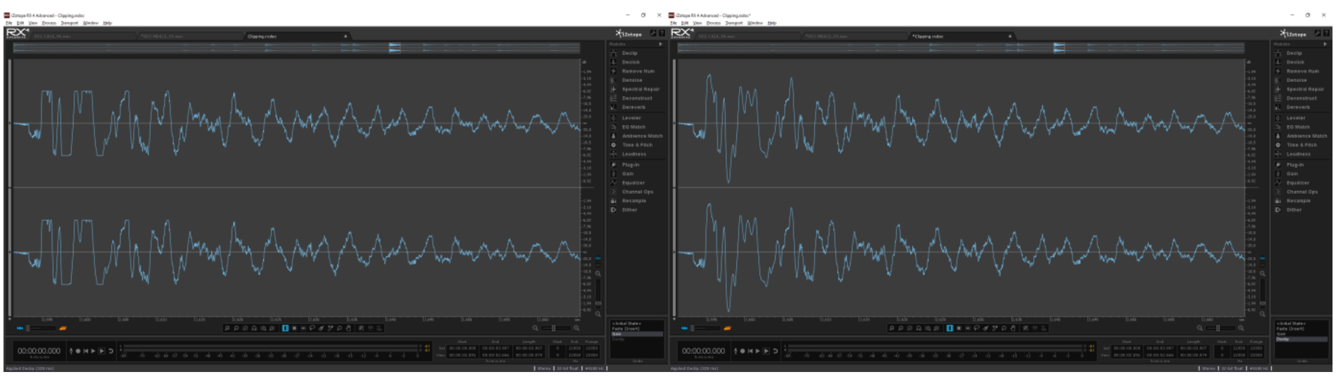


Figure 20: A clipped snare drum hit, before and after being processed with the "DeClip" function in Studio RX

COMB FILTERING

- Microphones should never be placed near large and hard reflecting surfaces (walls, floor, ceiling): the reflected sound wave will reach the microphone with just a small time-delay compared to the sound wave coming directly from the sound source.
- The interference from the direct and delayed sound wave cause a type of phase interference called "comb filtering" (from the shape of the resulting frequency response); this sounds like a very unnatural, slightly metallic type of coloration / distortion.
- Comb filtering can also occur when mixing together signals from multiple microphones, placed at different distance from the sound source.
- Comb filtering, once it occurs, cannot be removed from the recorded signal using EQs or any other form of post-processing: it is very important to be aware of this, and take the necessary measures *before* the signal is recorded.
- It can be empirically demonstrated that comb filtering will not be noticeable if the delayed sound wave is at least 18 dB quieter than the original one.

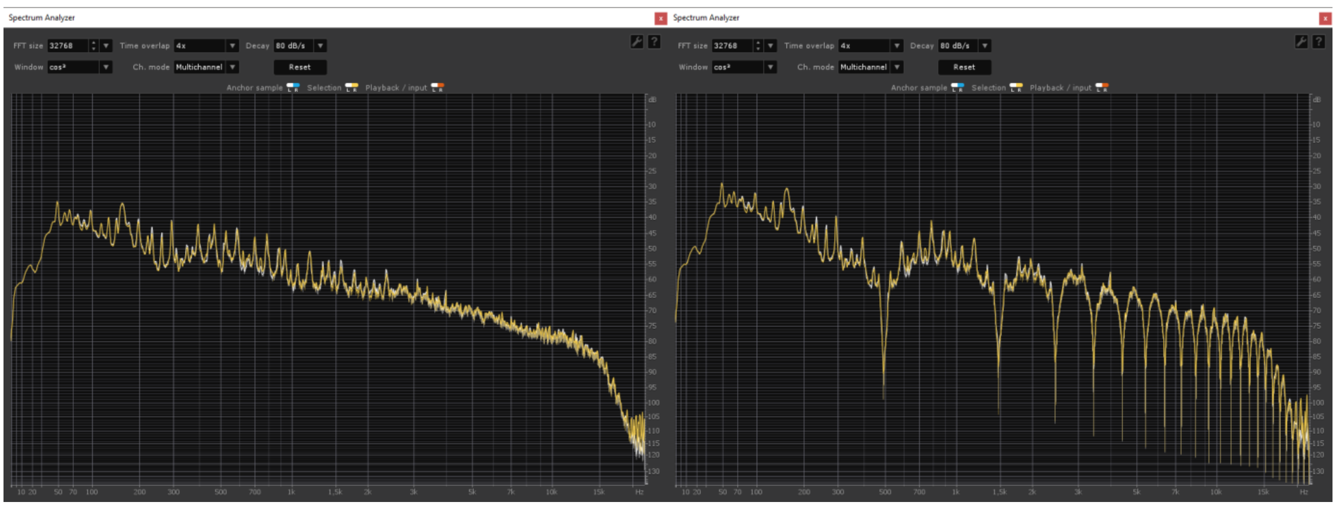


Figure 21: Left, the normal spectrum of a sound source; right the resulting "comb filtering" after adding the same signal, delayed just 1 ms (that corresponds to an additional sound path of 34 cm)

- Exception: **Boundary Layer Microphones** (Grenzflächemikrophone), or **PZM** (Pressure Zone Microphone) are designed to work best when placed directly on surfaces; they use the boost in level in proximity of a surface to optimize sensitivity, and because they just cannot get the reflected wave from the surface (as they are placed on the surface itself), they sound very natural and uncolored (no comb filter effect).

Figure 22: Beta 91 by Shure, a PZM microphone specially designed to record the bass drum

USING MULTIPLE SUPPORT MICROPHONES

- How to avoid comb filtering when using multiple support microphones: use the so called **4x Rule**.
- The combination of 4x the distance (a drop of -12 dB), plus setting the microphones at such an angle that allow to take advantage of the side attenuation in a cardioid polar pattern (-6 dB), gives -18 dB difference for the recording level of an instrument (in Figure 11, the Cello) in its own support microphone, and the "leaking" level in the support microphone of the nearby instrument (in Figure 11, the Viola).
- This way, the same signal will be captured by the support microphone of the nearby instrument with a -18 dB drop in level (plus additional attenuation due to different pan settings) and there will be no noticeable comb filtering.

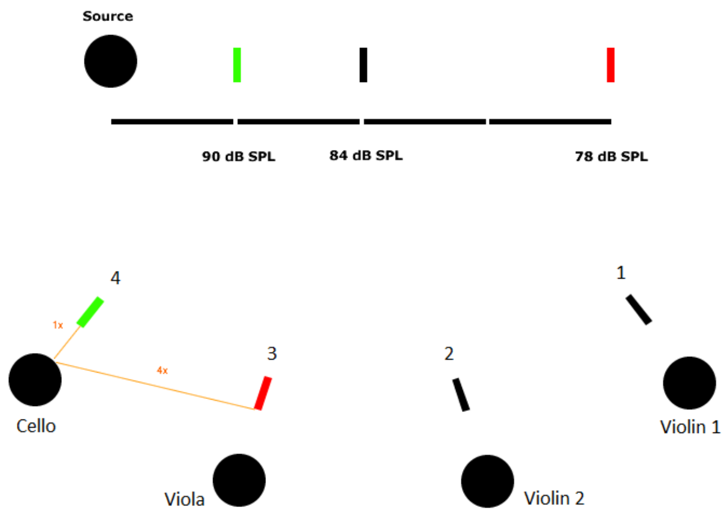


Figure 23: the "4x Rule" to avoid comb filtering when recording with multiple support microphones.

5.2 SPEECH RECORDING TIPS

To record speech professionally, please follow these simple guidelines:

- Use a large diaphragm cardioid condenser microphone, such as NT 1000, AKG C414, Neumann U87, AT 4050, etc. or a specific vocal/speech dynamic microphone, such as Shure SM-58.
- Use professional digital recording equipment that can provide stable phantom power.
- Record at 24-bit quantizing precision, 44.1 (for CD) or 48 kHz (for video / DVD) sampling frequency.
- Set the recording level manually, do not use automatic input level regulation!
- Leave enough headroom to avoid "clipping" (set the level so that the average is about -12 dBFS and the peaks about -6 dBFS; the level should never reach 0 dBFS).
- Place the microphone about 10-30 cm from the speaker, depending on microphone type and directional pattern
- Do not place the microphone directly in front of the mouth to avoid wind and pop noises (exception: SM-58 dynamic microphone, and other mics designed for extreme close setup).
- Use a pop filter (for example from K&M).
- Avoid reflecting surfaces between microphone and speaker; for example, if you must record on a table, place some absorbing material (like a thick blanket, or acoustic foam) on it to avoid reflections and comb-filtering.

6. RECORDING SETUPS

6.1 JAZZ ENSEMBLE (SOLO VOCALS, VIOLIN, ACOUSTIC BASS, DRUM KIT, VIBRAPHONE)

DRUM KIT

Figure 24: Jazz Drum Kit, recorded with Shoeps CMC 541 overheads (for cymbals and toms) in NOS configuration (the RODE NT 1000 are for Vibraphone)

Figure 25: Drum Kit: detail of the Schoeps CMC 541 overheads in NOS configuration

Figure 26: Snare drum, recorded with AKG C414 and Sennheiser MD421; bass drum, recorded with Neumann Fet47 and Sony C37A

ACOUSTIC BASS

Figure 27: Acoustic bass, recorded with Brauer Phantom and internal DPA pick-up

VIBRAPHONE AND VOCALS

Figure 28: Vibraphone, recorded with 2 RODE NT 1000 (sort of A-B stereo); Solo Vocals, recorded with Brauner Phantom

6.2 ELECTRIC GUITAR

MINI-AMP

Figure 29: DNS recording: electric guitar Vox mini-amp, recorded with Cascade FatHead II (ribbon) and Heil PR-30B (dynamic) microphones

LARGE 4-SPEAKERS AMP

Figure 30: Electric guitar amp (with 4 speakers), recorded with Royer R-121 (ribbon) and Shure SM-57 (dynamic) microphones, both placed very close to the amp speaker cone. Source: http://www.royerlabs.com/rectips_electricguitar.html

Figure 31: the same combination of amp and microphones, this time with the Shure SM-57 placed up close, and the Royer R-121 at about 50 cm distance, capturing a less direct, more “airy” sound. Source: http://www.royerlabs.com/rectips_electricguitar.html

6.3 CLASSICAL TRIO

PIANO AND TWO VIOLINS (OVERVIEW)

Figure 32: DNS recording, featuring a OSS main stereo pair (DPA 4006), grand piano with NOS (2x RODE NT 1000) and two violins (2x Brauner Phantom)

GRAND PIANO (DETAIL)

Figure 33: Steinway D grand piano, detail of the NOS stereo pair (RODE NT 1000)

6.4 MIXED INSTRUMENTAL/VOCAL ENSEMBLE

GRAND PIANO, GUITARS, WOODWINDS, TRUMPET, PERCUSSION, CHILDREN CHOIR, SOLI

Figure 34: DNS recording setup for a mixed ensemble conducted by Albert Anglberger in the Große Universitätsaula (Salzburg), featuring OSS main stereo pair (DPA 4006), NOS for the grand piano, A-B-C-D group (NT 1000) for the children choir (on the right), 2x AT 3035 for the percussion, 4x NT 1000 for the woodwinds and the trumpet, AKG C414 for the acoustic guitar, Sennheiser MD 421 for the electric guitar amplifier, 2x Brauner Phantom for solo vocals

6.5 SOLO VOCALS

Figure 35: DNS recording @ die:mischbatterie for Placido Domingo, featuring OSS main stereo pair and vintage Neumann U-47 valve condenser microphone

Figure 36: detail of the Neumann U47 and K&M pop filter

6.6 CLASSIC ORCHESTRA

STRINGS, WOODWINDS, BRASS, TIMPANI

Figure 37: DNS recording setup for the Mozarteumorchester in the Odeion Saal (Salzburg), featuring OSS main stereo pair (DPA 4006 with Jecklin Disk) and 12 support microphones (RODE NT 1000, Brauner Phantom and Audio Technica AT 3035)

Figure 38: another view of the same orchestra recording setup

6.7 WIND SYMPHONY ORCHESTRA

WOODWINDS, BRASS, PERCUSSION, SOLO CONTRABASS

Figure 39: DNS orchestra recording setup for the Bläserphilharmonie Salzburg in the Mozarteum Grosser Saal (Salzburg), featuring OSS main stereo pair (DPA 4006) and 17 support microphones (Neumann TLM 103, AKG C414, RODE NT 1000, Brauner Phantom, Audio Technica AT 3035)

Figure 40: another view of the same orchestra recording setup

RECOMMENDED LITERATURE

- EDERHOF, Andreas: *das Mikorfonbuch* – GC Carstensen 2004 (ISBN 3-910098-28-2)
- HENLE, Hubert: *das Tonstudio Handbuch* – GC Carstensen 2001 (ISBN 3-910098-19-3)
- HÖMBERG, Martin: *Recording Basics* – PPV Medien 2002 (ISBN 3-932275-21-7)

WEBSITE

- http://www.digitalnaturalsound.com/fh-multimediaart/audio.html
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