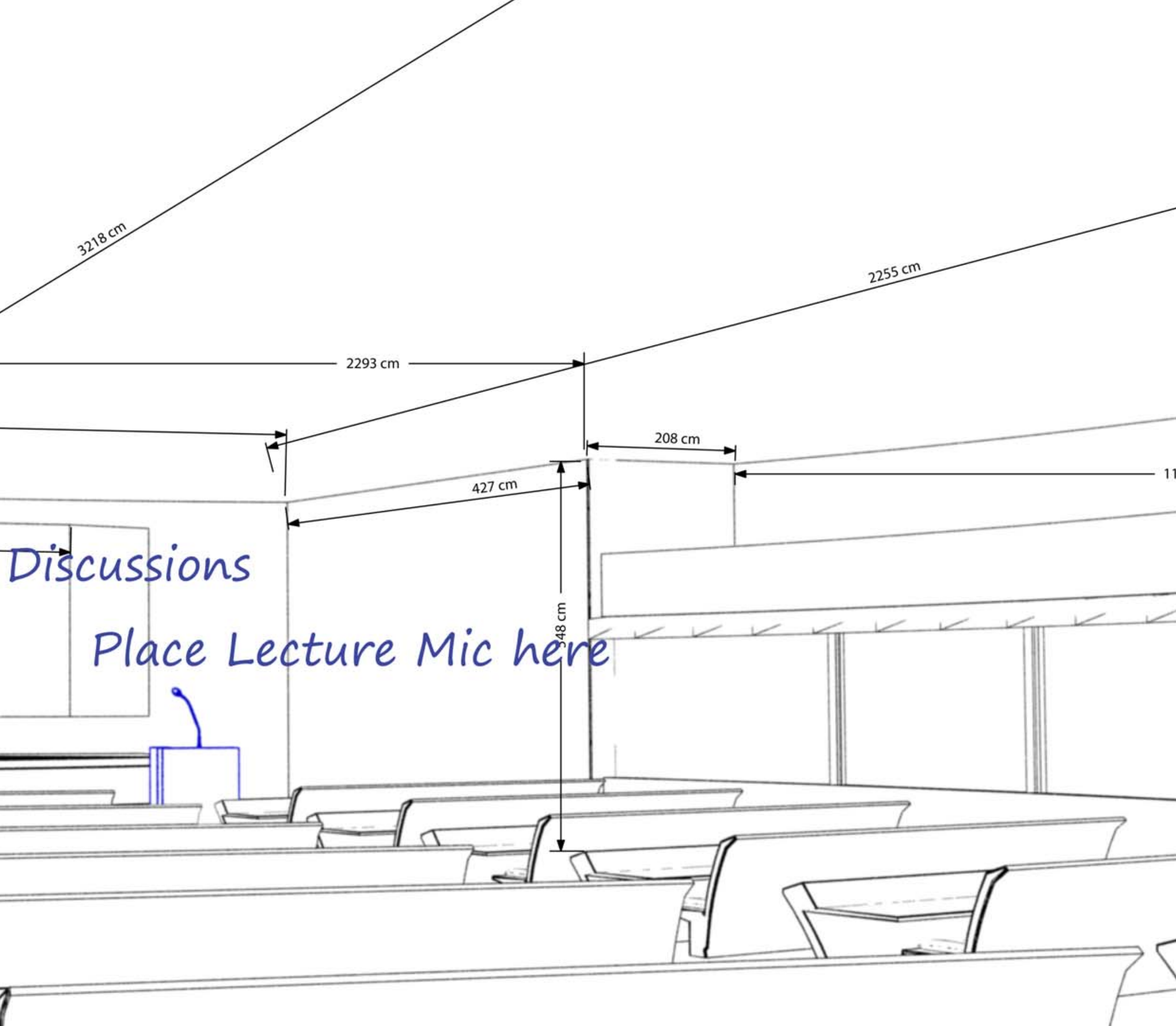

Microphone Technology for Conference Rooms

The Application Guide for Architects, Designers, Operators and Users

New products at the 2009 show

for sound and conference technology

DMS 700 – Digital Microphone System

Permanent installations: Outstanding features include optimum audio quality, an ultra-broadband UHF tuning range and encryption for sensitive audio transmissions.

AKG's revolutionary DMS 700 was developed to provide the required quantity of channels for any situation. Working with the new microphone system is made easy thanks to innovative and intuitive operation and unprecedented audio quality. It is the first professional digital wireless microphone system that complies with both US FCC and European ETSI regulations to ensure worldwide use.

The switching bandwidth of up to 155 MHz of transmitters and receivers gives the user the necessary flexibility, even in a crowded RF environment. The digital audio transmission eliminates distortions and significant noise levels that frequently occur in analog systems.

DHT 700 DPT 700 DSR 700

CS 5 Conference System

The new digital conference system CS 5 created by AKG is suited both for use as a straightforward discussion system and complex conference system. By focussing on few but versatile system components, the new AKG CS 5 conference system ensures maximum flexibility at a unique cost/performance ratio.

For reasons related to costs and design, user-defined expandability was right on top of the developers' priority list. Depending on your needs, you can expand the system by additional microphone stations while continuing to use currently operating components.

Optimized intelligibility, maximum comfort and low-fatigue work are ensured by AKG's Discreet Acoustics Modular Series. Just select the microphone mouthpiece from a choice of five different types to suit any desired use.

The optional simultaneous use of one microphone station by two persons helps reduce costs and save space on conference tables – a welcome effect.

The new AKG CS 5 meets all requirements for a perfect conference – optimized intelligibility, individual designing options for special demands, high level of reliability as well as optional conference line, recording and voting. After all, success is a matter of communication.

CS 5 BU (Base Unit)

CS 5 VU

Editorial

Success is a matter of communication

Even in our digitalized era, the spoken word is gaining importance: modern sound technology is at the heart of business meetings, conferences, conventions and public events, such as in court buildings or in Parliament.

Perfect intelligibility, individual design options for special requirements and a high level of operating reliability are of crucial importance.

Choosing the right microphone plays a central role. The microphone is the point where speech is converted into an electronic signal - the first link of the electronic transmission chain. Any loss of quality occurring at this stage is extremely difficult, in most cases impossible, to compensate for at a subsequent stage. This is why it is imperative to pay maximum attention to the selection of microphones, their positioning and the appropriate polar patterns. The right choices will improve voice quality and help reduce investment costs for the entire system.

Designing a conference system requires detailed discussions with all persons involved: architects and designers, acousticians, operators, users and operating staff. This approach is essential for learning about individual requirements, statutory and planning prerequisites as well as practical experience.

This Application Guide provides assistance for managing these tasks. It offers plenty of tips and tricks for practical implementation without requiring specific technical knowledge.

If you require more information, please send an email to hotline@akg.com or visit our website www.akg.com to browse for technical information that is being continuously updated and increased.

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Intelligibility Optimized

A conference room must be designed so as to provide optimum sound conditions. This can only be ensured by selecting the right microphones.

Intelligent Signal Processing

Conference applications need to process many different signals, such as: audio, wireless, infrared, simultaneous interpretation or external feeds. Accurate research is the prerequisite for optimum design.

Practical Applications

Regardless of the application, stable acoustic conditions are created by minimizing the distance between sound source and microphone.

Glossary

Helpful explanations of technical terms and checklists for planning.

Intelligibility Optimized

Once the system starts oscillating, it’s too late.

An ideal conference room should be **acoustically designed** to allow all persons present in the room to **hear all talkers perfectly.**

In most real conference rooms, however, **various types of sound sources compete with one another.** These include **voices; fans** in beamers, overhead projectors, computers, or air conditioning systems; **noises made by other delegates** setting down glasses, pushing back chairs, clearing their throats, or coughing; sound reflections from walls and ceiling; as well as ambient noise (footfall or traffic noise).

Human perception detects the periodic variations in air pressure we call “sound” through the ears. These stimuli then cause neuronal activity in brain cells. The resulting signals are selectively weighted, organized, and **passed on** if found to contain wanted information or immediately **rejected and not passed on** if found to be **unwanted information** (e.g., ambient noise).

This mechanism functions only as long as the **wanted sound** is appreciably louder than **unwanted noise. Below** a specific limit – a difference of less than **25 dB** between wanted sound and unwanted noise – the human brain finds it **difficult to distinguish between these two kinds of signals.** We do not hear clearly what is being said, we feel strained, and find it difficult to concentrate. In short, everybody hates to work in such a room. Appropriate equipment can help solve this problem: enter the sound system.

The **most important** job of a **conference or discussion system** is to ensure **perfect intelligibility.** It is a fact, however, that no audio system and no microphone can distinguish between **wanted** and **unwanted information** the way the human brain can. Every sound is picked up and amplified, no matter whether it is a **wanted signal** or **unwanted noise.**
Therefore, it is **important** to be aware of the **psychoacoustic aspects of wanted and unwanted sound** and to use this knowledge when **designing a conference system.**

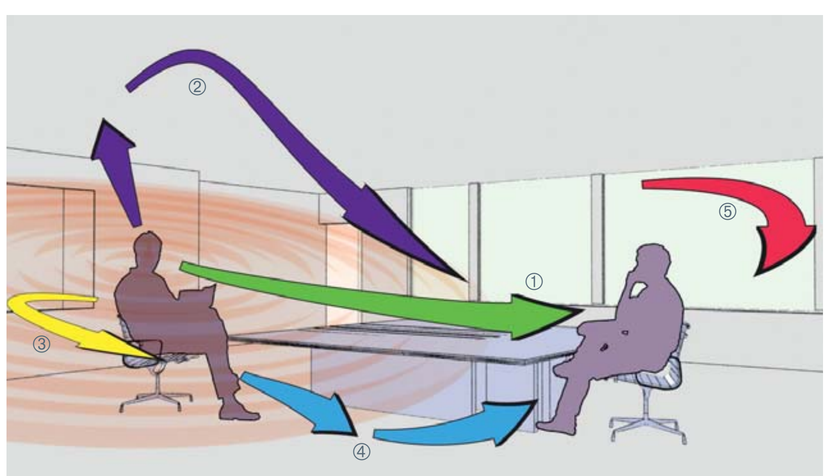
Intelligibility depends on the **loudness ratio between the wanted sound and unwanted noise.** Just **making the voice signal louder is not enough** to make it more intelligible. The point is that **the louder the wanted sound is compared to the unwanted noise, the more intelligible it will be.** In real life, all efforts must be made to achieve this objective. Therefore, it is the **highest priority** in system design.

Theoretically, it is **very easy** to reduce unwanted noise. Good acoustic treatment of the room and extremely quiet fans in projectors and computers automatically increase the perceived wanted sound level. Actual rooms, however, provide another, less obvious noise source: reverberation. **In fact, any room** generates a more or less significant amount of **reflections** depending on the absorptive quality of the room. These reflections form a **diffuse sound field** that, like noise, can **mask the wanted signal** and thus **reduce intelligibility.** The obvious idea of drowning out the reverberation by turning the gain of the wanted signal all the way up simply does not work.

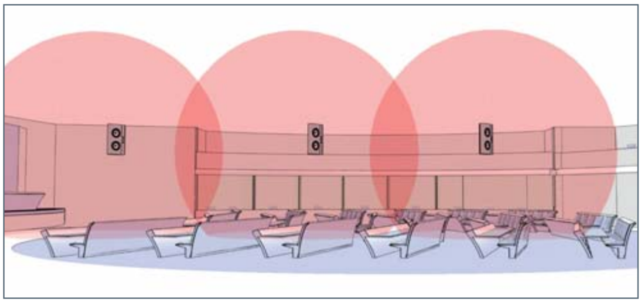
The reason is obvious, too: injecting **more sound energy** into a room also **increases the energy of the reflections,** i.e., the reverberant sound. In other words, **the higher the level of the wanted signal, the louder the unwanted noise.**

This explains why it is so important to think about microphone polar patterns and loudspeaker radiation patterns even at the design stage. At this point, we might take a look at the basic physics of all this.

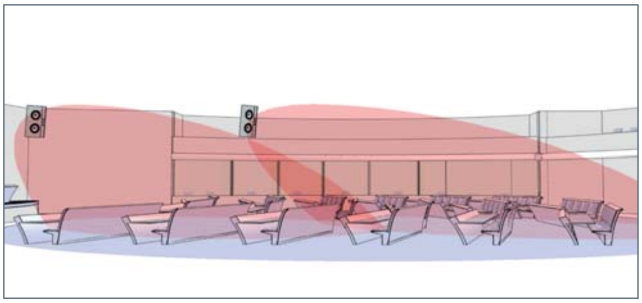
The **inverse-distance law** states that the sound pressure decreases by half (or 6 dB) as the distance is doubled. Conversely, this means that **the sound pressure in front of a microphone will double as the distance between the sound source and the microphone is halved.**



① Direct sound; reflections by: ② ceiling; ③ walls; ④ floors; ⑤ smooth surfaces (windows)



Non-directional loudspeakers: The sound disperses evenly through the room, producing strong reverberation effects.



Directional loudspeakers can be used for focused sound effects; less energy is left for reflections.

The **inverse-distance law** directly affects the **acoustics of any room** as it determines the **difference in level** between **the direct sound** and the **reverberation.** While **the direct sound level decreases by 50% every time the distance is doubled,** the reverberation level in the room remains more or less constant because the reverberation level is determined by the volume and absorption of the room. As a consequence, **the direct sound becomes less intelligible as you move away from the sound source** because the direct sound is masked by the reverberant sound field.

Due to the inverse-distance law, it makes no difference in terms of loudness whether you turn the amplifier gain up by 6 dB or reduce your distance from the loudspeaker by 50%. These two options do differ in another respect, though. Turning up the volume by 6 dB increases the sound energy in the room including the reverberation energy. **If you move closer to the loudspeaker, however,** the sound energy in the room remains constant and the **wanted signal to unwanted noise ratio improves,** resulting in better intelligibility.

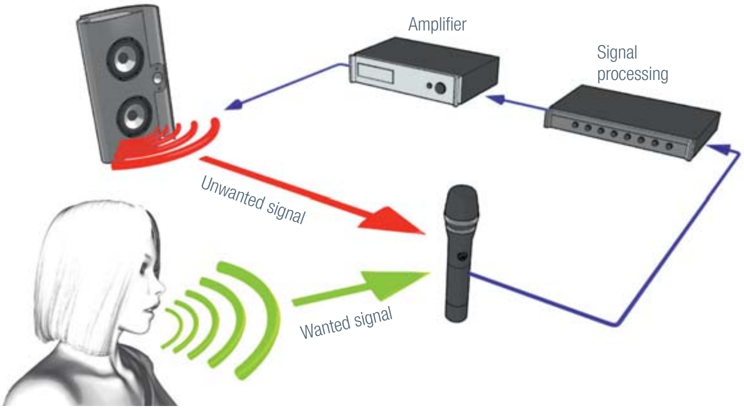
This is the idea behind **directional loudspeakers.** Instead of radiating the sound energy with equal intensity in all directions, they **focus the sound in one direction – exactly where it is wanted.** This technique provides **two benefits.** For one thing, persons sitting within the preferred radiation angle get the impression of being **closer to the loudspeaker.** For another, the sound energy is focused on the audience and there is **less energy available for reflections and reverberation.**

The input side (i.e., the microphone) of a sound system obeys the same physical laws.

Omnidirectional microphones capture sounds equally from all directions, including reflections and ambient noise. The noise is amplified by the same amount as the wanted sound and intelligibility is reduced dramatically. **As the reflections and the sound radiated by the loudspeakers are picked up by the microphone and amplified again, this signal regeneration may ultimately lead to an acoustic disaster called feedback.**

Feedback is a genuine vicious circle: Sound is picked up by the microphone, made louder by the amplifier, and radiated by the loudspeaker. The amplified sound is picked up by the microphone again and fed back into the amplification chain for another trip through the system.

The **first symptom of instability** in a sound system is a **ringing sensation or tone that gradually rises in level** until it culminates in an earsplitting, often painful, screeching noise. A **violent feedback** attack may even cause fits of tinnitus in listeners, affecting intelligibility on a “medical level” for quite a while.



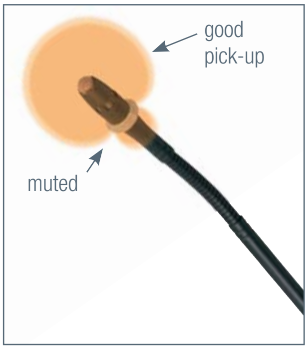
Feedback can be reduced/prevented by minimizing the distance between sound source and microphone while maximizing the distance between loudspeaker and microphone.

Unidirectional microphones can improve the situation** in more than one respect. They are **more sensitive to sound from one preferred direction** and thus **attenuate ambient noise coming from other directions.** This directivity also reduces the apparent distance from the sound source, thereby **increasing the wanted signal level.** As a beneficial result, the sound system volume can be set at a lower level so the microphone will pick up less sound from the loudspeaker and there will be less reflection energy in the room. All this adds up to **better intelligibility.** As an added bonus, there will be no risk of feedback.

Reducing the number of microphones that are open at the same time, say, from four to two (by 50%), has a similar effect. Halving the number of open microphones reduces the proportion of ambient and loudspeaker signals in the sum of all microphone signals in the mixing console or automatic mixer by 3 dB. Reducing the number of open microphones from two to one reduces the noise level by another 3 dB.

This means that **if four microphones are open,** feedback will start at a **gain setting 6 dB lower** than it would with a single open microphone. No matter how much power a sound system puts out, the usable gain before feedback (the sound pressure level that it can produce with no risk of feedback) with four open microphones is only 50% of what it would be with only one open microphone.

Omnidirectional response:
Sound is picked up equally from all directions.



Hypercardioid:
Boosts sound pick-up from one preferred direction while muting ambient noise.

The **automatic microphone mixer** presented on page 13 has been specifically designed for intelligent **management of open microphone channels.** It makes sure that only the number of microphones that are actually needed will be open simultaneously at any time. This also increases the usable gain before feedback dramatically in any room. It is good practice to keep this principle in mind when designing a conference system and to **keep the number of open microphones at a minimum at all times.**

SUMMARY: Why everything depends on selecting the right microphones:

> **The main purpose of a conference room** is to ensure **intelligibility of speech. The most important prerequisite** for that is **good acoustics.** In order for a sound system to provide the **desired results, reverberation** has to be kept to a **minimum.** This can be achieved by **using unidirectional microphones and loudspeakers** and **positioning them appropriately in the room.** Acoustic treatment of the walls and ceiling plays an important role, too.

> **The following points are of fundamental importance:**

- The **wanted signal level** must be **much higher** than the **unwanted signal level.**
- The **direct sound level** must be **much higher** than the **reverberation level.**

The **reverberation level** depends on the **level of the amplified wanted signal.**

> **Increasing the direct sound level** by adding more energy will always **increase the reverberation level, too** – which, after all, is the sum of all reflections of the direct signal.

> The microphone is the first and most important component of any electronic sound system. **Selecting the appropriate microphone** is therefore **one,** if not **the, most important decision** in designing a system. The **right types** *(see next section)* and **appropriate polar patterns of microphones and loudspeakers** increase **gain before feedback** and **minimize ambient noise.**

Choosing the right microphone

The optimum solution for every application

The performance of any sound system ultimately depends on what microphones are used. The **microphone** is the **point where the sound signal enters the electronic processing chain.** Any information lost during the conversion of the voice signal into an electrical signal is extremely difficult, in most cases impossible, to compensate for at a subsequent stage. A poorly miked up signal will not sound any better because it is amplified and reproduced by loudspeakers.

Some designers are said to select microphones for their esthetic appeal only or simply do not care. This kind of misjudgment may be expensive and can dramatically reduce the performance of any system. All other components such as mixers, amplifiers, and loudspeakers cannot do their job properly if the microphone is not at par. In the worst case, the wrong microphones may defeat the very purpose of a conference room.

Choosing the wrong microphones may **ruin intelligibility and thus overall system performance.** It therefore pays to learn about microphone specifications and what they mean in practice. After all, there are countless different types of microphones that look different and work differently. Knowing about these differences makes it easy to find an **esthetically pleasing, functionally suitable microphone for every kind of sound system.**

A microphone is an electro-acoustic transducer that converts the periodic variations in air pressure we call sound waves into an electrical signal. Over the years, two types of transducers have prevailed in the professional audio industry – **dynamic microphones** and **(electret) condenser microphones.**

When designing a sound system, it is important to know how these two types of microphones differ in day-to-day use. Most condenser microphones are much smaller and lighter than dynamic designs. Therefore, most miniature microphones, e.g., headset or lavalier models, use condenser transducers.

Microphones exposed to **high mechanical stress** and **extremely high sound pressure levels** are generally **dynamic types** because dynamic transducers are much more rugged than condenser capsules. Not surprisingly, most **handheld** microphones are dynamic designs.

Being very small, condenser transducers are used in more and more gooseneck microphones because esthetic reasons play an important role in this sector.

Note that condenser microphones need a supply voltage that is usually fed to the transducer by the mixer via the microphone cable. This is called **phantom powering.** Dynamic microphones, by contrast, need no supply voltage.

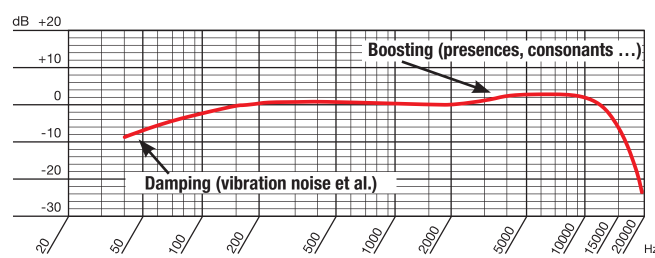
Condenser microphones are often used for **miking quiet sound sources.** Their **sensitivity is higher,** so they sound **louder** than dynamic microphones do at the same gain setting. (The "sensitivity" of a microphone is its output voltage at a given sound pressure level, usually stated in mV/Pa.) Conversely, a microphone with higher sensitivity needs a lower amount of gain on the mixer or amplifier to deliver the same loudness level. This is also the reason why highly sensitive microphones cause **lower noise.**

Most microphones are not equally sensitive to all frequencies. A microphone's "frequency response" shows how its sensitivity varies with the pitch (frequency) of the incoming sound. An ideal microphone would be equally sensitive throughout the entire audio frequency range. In real life, there are physical limits, though.

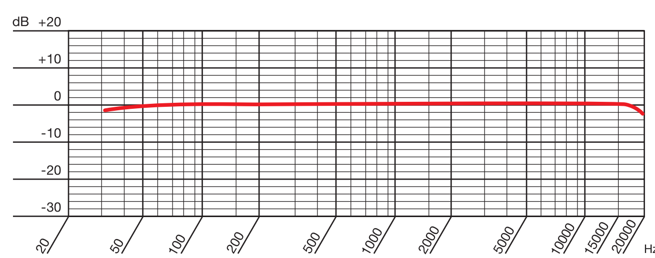
These limits (at the low and high ends of the frequency range) are, however, irrelevant when using today's high quality condenser or dynamic microphones for speech miking, for these limits are outside the frequency range of the human voice.

Paradoxically, a flat frequency response may not be the best solution for all applications. **A microphone's frequency response can be designed to attenuate certain parts of the frequency spectrum and boost others.** The essential information of speech is found within a range between 300 Hz and 3.2 kHz. The frequency response of most microphones is flat within this band. The consonants, which are essential for **intelligibility,** are sounds at frequencies between 2.5 and 15 kHz. Therefore, microphones optimized for speech pickup boost these frequencies.

Microphones are not equally sensitive to sounds from all directions *(see also the section on polar response below).* Talking into a microphone from the side will attenuate the high frequencies that are especially important for the intelligibility and presence of a voice. A high-frequency peak in the frequency



Speech-optimized frequency response



Extremely linear frequency response

response of microphones that are often talked into from the side by inexperienced users is therefore extremely helpful. Of course, microphones with a flat frequency response have their uses, too – they are excellent tools for experienced talkers.

The frequency response can also be designed to **roll off at low frequencies to minimize the pickup of mechanical and handling noise.** This also reduces **pop noise** (plosive sounds with a large proportion of frequencies below 150 Hz). Such a bass cut or rolloff is particularly beneficial in unidirectional microphones, to compensate for their **proximity effect.** Talking into such a microphone from very short distances will boost the low frequencies dramatically. Experienced talkers or vocalists use this effect to add more character to their voice. Unfortunately, the proximity effect also boosts pop noise. To avoid this effect, switch in the bass-cut filter (aka highpass filter) found on many condenser microphones as part of the preamplifier or on the mixer.

The overall performance of a sound system depends on the polar response of the microphone. The **polar diagram** shows **how the microphone responds to sounds coming from various directions.** Similarly to the effect of placing a hand behind your ear, some microphones prefer sounds from one (or two) direction(s) and more or less “ignore” sounds from other directions.

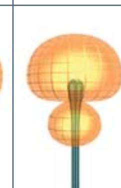
Unidirectional microphones therefore provide two major benefits. They **prefer the wanted signal** (the talker’s voice) and **attenuate unwanted noise** from around the talker, thereby **minimizing** the risk of **feedback** and **improving intelligibility.**

The polar diagram of a **cardioid microphone** illustrates in which direction its sensitivity is highest (0°) and in which direction it is more or less “deaf” (180°). Cardioid microphones are preferably used in situations with a **noise source or a loudspeaker exactly opposite the talker.** A hypercardioid microphone has its maximum rejection at 125° so it will efficiently reduce reflections from the floor or tabletop for instance.

The **hypercardioid** and shotgun are the polar patterns with the **highest front-to-back ratio** (directivity factor). Compared to an omnidirectional microphone, a hypercardioid picks up **four times more wanted sound** than ambient noise, so these microphones are the preferred choice for rooms with high noise levels. The diagram shows a comparison between various polar patterns.

Real-life **polar patterns are not the same for all frequencies.** The directivity is usually higher at high frequencies than it is at the low end. This has its implications for designing a system because most ambient noise is in the low-frequency band, so using a highpass filter definitely makes sense.

Microphones do not only differ in terms of electrical characteristics, they can also be classified by their type of body. No matter how good its condenser transducer may be, a handheld microphone is not the ideal choice for a nervous presenter using the microphone as a pointer.

CHARACTERISTIC	OMNI-DIRECTIONAL	CARDIOID	SUPER-CARDIOID	HYPER-CARDIOID
Polar response pattern				
Angle of max. rejection (null angle)	–	180°	120°	110°
Rear rejection (relative to front)	0	25 dB	12 dB	6 dB
Ambient sound sensitivity (relative to omni)	100%	33%	27%	25%
Distance factor (relative to omni)	1	1.7	1.9	2

This will at best cause wide variations in loudness. In the worst case, the presenter may even aim the microphone at the audience, driving the sound system into what everybody seems to fear most, the screeching oscillation caused by feedback. The best way to avoid this danger would be to use a **lavalier microphone** or a **head-worn microphone.** So, the easiest way to select the best microphone for a given application is to know exactly what the microphone will be used for and how.

For **talkers moving about in a room,** a wireless transmitter is usually the best option. This system replaces the microphone cable with a radio link. Handheld wireless microphones normally have a transmitter built into the microphone body. This is very convenient in situations where the microphone is to be passed on from person to person. Lavalier and head-worn microphones, however, are connected to a separate device called “bodypack transmitter”.

Due to its spherical polar pattern, the **CK 77 WR** is very small and especially well protected against moisture (perspiration) and vibration noise.

With its cardioid polar pattern, the **CK 55** is distinctly larger than the CK 77 WR, since directional microphones always require a somewhat larger design.

The **HC 577** headset provides maximum transmission quality due to the especially short distance between microphone and sound source.

The shorter the distance between the microphone and the talker's mouth, the lower the risk of feedback. **Head-worn microphones** also minimize handling noise and **keep the distance between the sound source and transducer constant,** ensuring a consistent signal level. Therefore, head-worn microphones appear more and more frequently in TV talk shows. Where head-worn microphones are unacceptable for visual reasons, a unidirectional (cardioid) lavalier model is a good alternative.

Gooseneck microphones are particularly suited for applications where many different talkers will speak at the same spot (e.g., a lectern), for it is obvious into which end of the microphone the user has to talk, and the microphone is usually close enough to the speaker's mouth to ensure good intelligibility. Selecting a **microphone with a polar pattern optimally suited** for the application will further improve intelligibility and gain before feedback.

Thanks to a choice of different lengths and a flexible joint, the gooseneck can be optimally adjusted to any position.

Boundary microphones are suited for discreet mobile use as well as for permanent installation.

Gooseneck microphones are available in various lengths and can therefore be bent so that the talker will stand or sit in front of the microphone in a comfortable position and at the ideal distance. They are available in various colors to blend in with the existing interior decoration. Various types of installation accessories can be used to minimize vibration noise (e.g., if a microphone is mounted on a piece of furniture). Also on the market are plug-in type gooseneck microphones for quick setup and takedown.

At some venues, no microphones must be seen. In these cases, **boundary microphones** can be an alternative to gooseneck types as they can be **mounted almost invisibly,** for instance, on a lectern, or in a table. Boundary microphones pick up **all signals arriving from above the boundary** (in this example, the tabletop). In other words, they have a **pickup angle of 180 degrees.** When installing a boundary microphone, it is important to make sure that the surface in which it is installed is as large as the wavelength of the lower frequency limit.

In real life, boundary microphones differ from gooseneck mics in that they are more sensitive to unwanted noise from the tabletop (users knocking on it, shuffling papers, etc.) and place the transducer further away from the talker. Also, many users drop papers on top of a boundary microphone so it cannot function properly anymore. In areas near loudspeakers, boundary microphones may not be the best solution, because they increase the risk of feedback. Sound systems with boundary microphones need to be designed extra carefully, and it may be a good idea to use special designs with a limited acceptance angle.

In some rooms, even nearly invisible boundary microphones may be perceived as too obtrusive or it may be impossible for architectural reasons to run cables to the delegates' desks. In such cases, system designers may specify special **overhead microphones** suspended from the ceiling or boundary microphones flush-mounted in the ceiling.

This solution has its acoustical drawbacks, however. Loudspeakers are often installed in the walls or ceiling and many of them may be closer to the microphone than the talker is. In addition, all other (unwanted) sound sources are at least as close to the microphone as the wanted sound source. Such a system would perform poorly in terms of gain before feedback and intelligibility.

Overhead microphones HM 1000 (omnidirectional) und CHM 21 (cardioid)

SUMMARY: The right microphone for every situation

> Microphones come in **many different types,** sizes, and shapes.

> They are **electro-acoustic transducers** that convert sound waves into electrical signals.

> There are two basic varieties: **dynamic microphones** and (electret) **condenser microphones.**

> Dynamic microphones are more rugged than condenser transducers, while the latter are **smaller** and are therefore used in head-worn, lavalier, and slim-line gooseneck microphones.

> Condenser microphones need a **supply voltage;** dynamic microphones need no powering.

> **Unidirectional microphones** can be used to pick up sound from a **preferred direction** and **reject** sound from other directions:

- Use a **cardioid** if there are any noise sources or loudspeakers opposite the talker.
- Use a **hypercardioid** if there is any noise coming from below (the floor, lectern, tabletop).
- Use a **supercardioid** in rooms with high noise levels from all directions.

> For **talkers moving about,** it is best to use a **wireless microphone.**

> **Head-worn microphones** provide a consistent signal level, and **lavalier microphones** minimize handling noise.

> For **changing talkers** speaking from the same microphone position, **gooseneck microphones** are a good choice. They come in different lengths.

> Where **microphones must not be visible, boundary microphones** can be installed. Like hanging microphones, they too are **more likely to pick up noise.**

Intelligent signal processing

Automatic equipment is today’s standard

The preceding chapters explained in detail why it is so important to **capture every sound with the greatest possible accuracy.** The next step is **signal processing. Intelligent signal processing equipment** is widely used today. Depending on the type of application, microphone signals may be fed to the mixer via cables or wireless transmission so they can be processed as required.

Wireless microphone systems:

Wireless microphone systems are needed where hardwire microphones cannot be used for esthetic or practical reasons. In most cases, **combinations of** a **lavalier or head-worn microphone** and **wireless transmitter** are used. Wireless microphones are also recommended for situations where a microphone needs to be passed on from one talker or member of the audience to the next.

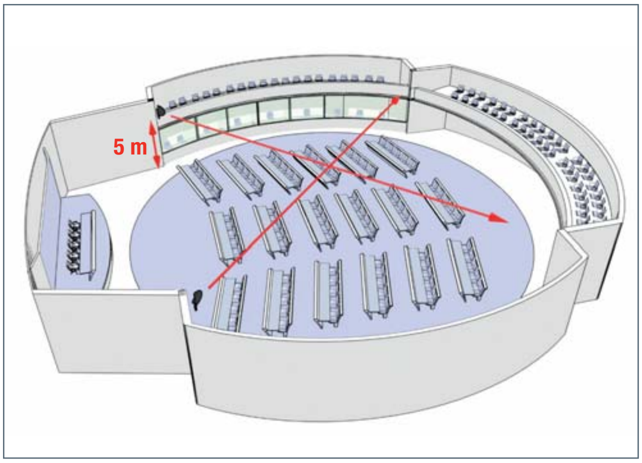
Any wireless microphone system comprises at least one transmitter and one receiver. If several wireless microphones are to be used simultaneously, note that **each transmitter** must be assigned to a **separate receiver tuned to the same frequency** as the transmitter. The transmitter may be integrated in a handheld microphone or a separate bodypack transmitter. A bodypack transmitter allows you to connect a variety of different microphone types. Many handheld transmitters are available with a choice of dynamic and/or condenser microphone elements and polar patterns.

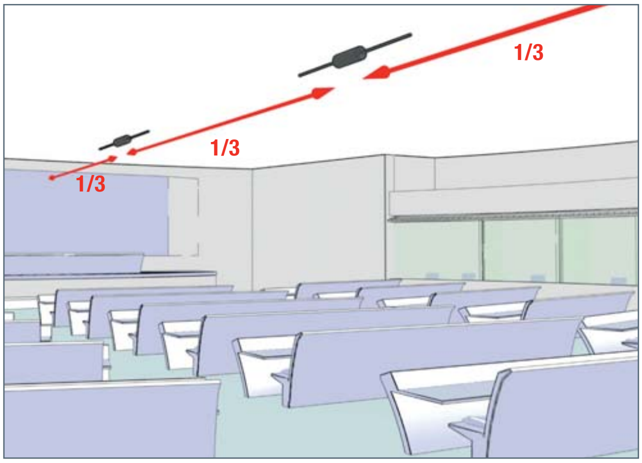
The criteria for selecting a type of microphone and polar pattern for a wireless system are the same as those for its hardwire counterpart. The difference, however, is the **mobility of the user.** Since talkers are free to move around in the room (and beyond), they may pass into the **coverage angle of a loudspeaker** or other source of unwanted noise. A microphone with the appropriate **polar pattern** can help reduce the risk of feedback.

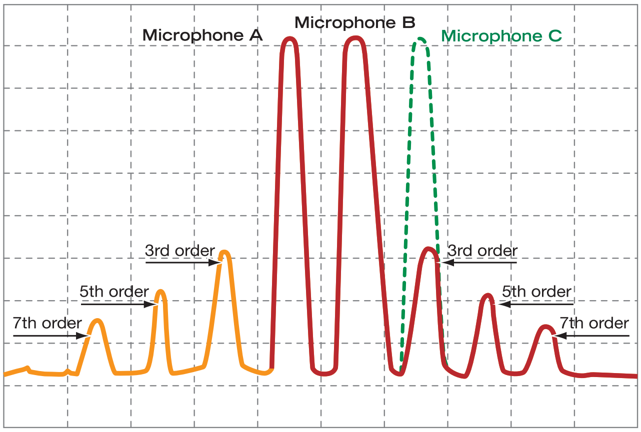
Wireless microphones use analog or digital transmission technology. Advanced digital systems use **digital encryption to prevent interception.** (Radio waves can penetrate walls so an unencrypted signal may be listened to outside the room by anyone with a suitable receiver.) Particularly in conferencing, encryption is a very important feature. In addition, digital systems provide higher audio quality and reception reliability than analog systems do.

The antenna is the “ear” of any wireless system. Accordingly, it is important not only to **position the antenna correctly,** but also to select the antenna that is **best suited to a given application.** The transmitter signal does not always arrive at the receiver in good condition. **Reflections** and **shadow effects** (due, e.g., to large metal grid structures) can **weaken the radio signal** or even **cancel it out completely** (the places where this happens are called “dead spots”). Any persons present in the room may also attenuate the radio signal, because radio waves behave in similar ways as light waves, differing only in wavelength. Therefore, it is good practice always to **make sure that the talker can see the receiver from wherever they use the microphone.**

Other kinds of disturbances are **interference** and **intermodulation.** These may arise when a receiver picks up **several RF signals at different frequencies** at the same time. “Interference” is a type of disturbance caused by external sources such as TV transmitters, cell phones, or WLAN networks. Using **several wireless microphones** simultaneously may also cause system-internal disturbances called **“intermodulation”.**







In large systems with many wireless microphones these disturbances may have serious consequences because intermodulation problems grow proportionately with the number of wireless microphones used at the same time. Additions, multiplications, or subtractions of the frequencies radiated by the transmitters produce new frequencies that may **cause additional disturbances.**

Flawless operation of a multichannel wireless system **depends on efficient frequency management software** of the type integrated in all AKG wireless microphone systems. The software algorithm selects the RF frequencies to be used simultaneously in such a way as to prevent intermodulation.

(For detailed information on frequency management and AKG wireless microphone systems, visit www.akg.com or consult the free AKG Setup Guides for wireless microphones.)

Intermodulation

Using several RF sources at the same time may cause intermodulation in the frequency spectrum. If a third RF source (microphone) operates at the same frequency as third-order products of microphones A and B, disturbances induced by intermodulation are very likely to occur. You will therefore need to check all required radio frequencies prior to the event.

This is easily accomplished with the aid of the fully automatic Auto Setup Mode.

DMS 700 – Digital Microphone System

SUMMARY: Perfect wireless transmission

> **Wireless microphone systems** comprise at least one **transmitter** and one **receiver**.
Take care to ensure **direct intervisibility** between transmitter and receiver.

> The transmitter may be integrated in a **handheld microphone** or a separate **bodypack transmitter.**
A bodypack transmitter allows you to plug in a variety of **different microphone types.**

> **Handheld transmitters** are available with **dynamic and/or condenser microphone elements** and **various polar patterns** so as to prevent feedback.

> Wireless microphones use **analog or digital transmission technology.**
Digital encryption ensures **safety from interception.**

> **Flawless operation** of a multichannel wireless system depends on **efficient frequency management.**

Automatic microphone mixers:

In a **“moderated” conference,** the chairperson, an engineer at the mixing desk, or the intelligent electronics of the conference system **limits the number of microphones that are open at the same time.** Many small systems do not provide this control function. Besides the acoustical problems caused by too many open microphones, other unpleasant effects may plague meetings with no form of moderation.

NOM limitation (limitation of the number of open microphones) is necessary because each additional open microphone destabilizes the sound system, reducing **gain before feedback** and increasing the unwanted noise level by 3 dB. If all microphones were open and all talkers were to speak into their microphones at the same time, all statements would finally be completely unintelligible.

An **automatic microphone mixer** can **prevent a number of problems.** For instance, many talkers forget to switch off their microphone after finishing their statement (not all microphones have an on/off switch, either). Some talkers may even forget to switch their microphone on before talking.

Automatic mixers operate as **“electronic switches”.** The microphone signal at the input will not be fed to the amplifier unless the signal meets certain input parameters. Otherwise, the channel remains silent. The automatic mixer thus **distinguishes between active and inactive microphones,** muting (or attenuating) the inactive microphones to **optimize the overall system level.**

Simple automatic mixers use a **noise gate** in each channel that feeds the microphone signal to the output stage as soon as the input level exceeds a defined threshold. This is a very simple and efficient technique. On the down side, however, these mixers would open all microphone channels whose input levels exceed the turn-on threshold. In the worst case, even a loud fan in a small room could open all microphone channels. The fan noise would be amplified and the risk of feedback would rise sharply within seconds.

To prevent this situation, **good quality automatic mixers** feature a **NOM limitation algorithm.** This function **balances the levels of all open microphone channels to maintain constant overall system gain** and prevent feedback.

Intelligent mixers also provide an algorithm that not only evaluates the level but also the spectrum of a microphone signal. This **“noise detection” algorithm** can detect speech activity in a channel and **open only this channel** while those channels where only background noise of the same level is detected remain muted. A similar algorithm is very useful for panel discussions as it will open only the one microphone where the loudest speech signal is detected. This function, known as **“best mic on”,** also **prevents unwanted comb filtering effects.**

The **“last mic on” function,** by contrast, keeps the last active microphone open in order to **feed some ambience to the loudspeakers.** This is generally perceived as a **pleasant “acoustic backdrop”** – as opposed to the breathing sound associated with the opening and complete silencing of microphones.

Highly refined automatic mixers do not even switch the microphones on and off completely. Instead, they attenuate the signals of inactive microphones such that the operation of the mixer remains completely inaudible. Many automatic mixers provide one or more priority levels to which each microphone can be assigned, allowing, for instance, the president’s microphone to override all other microphones. Advanced auto mixers even feature equalizers and other audio processors.

SUMMARY:

Intelligent electronics for optimized communication

> Limitation of the number of open microphones is necessary to avoid feedback. **Automatic microphone mixers** operate as **regulators** for microphone management, optimizing the overall system level.

> Automatic mixers use **noise gates** to activate microphones as soon as a specific sound level has been reached. The **NOM limitation algorithm** prevents activation through background noise and maintains **constant overall gain.**

> The **“noise detection” algorithm** activates only the required microphone, thus **preventing** unwanted **comb filtering effects.**

> The **“last mic on” function** keeps one microphone open in order to allow some ambient sound. Advanced automatic mixers **attenuate** microphone signals and feature equalizers for **sound regulation.**

High-performance conferencing

Automatic mixing, conference systems, infrared language distribution systems

Automatic mixers can do a lot more than just turn microphones on and off. Besides the usual discussion applications, they are also used in broadcast studios and multimedia suites. While most auto mixers are used as standalone devices, they can also be daisy-chained to provide more microphone inputs.

Discussion systems are easier to use and offer **sophisticated features.** These highly specialized systems can control anything from just a few up to hundreds of microphones. The **microphones** usually come **installed in dedicated microphone stations** that may also feature **loudspeakers** and **voting buttons.** These systems are controlled by highly specialized circuitry optimized for discussion applications, using, like auto mixers, noise gates and NOM limitation.

Professional discussion systems are even more powerful. **Microphone stations** are **connected** by special cables to a **closed-circuit bus or several single-ended lines.** This reduces wiring cost, **improves system reliability,** and minimizes setup and takedown times for mobile systems.

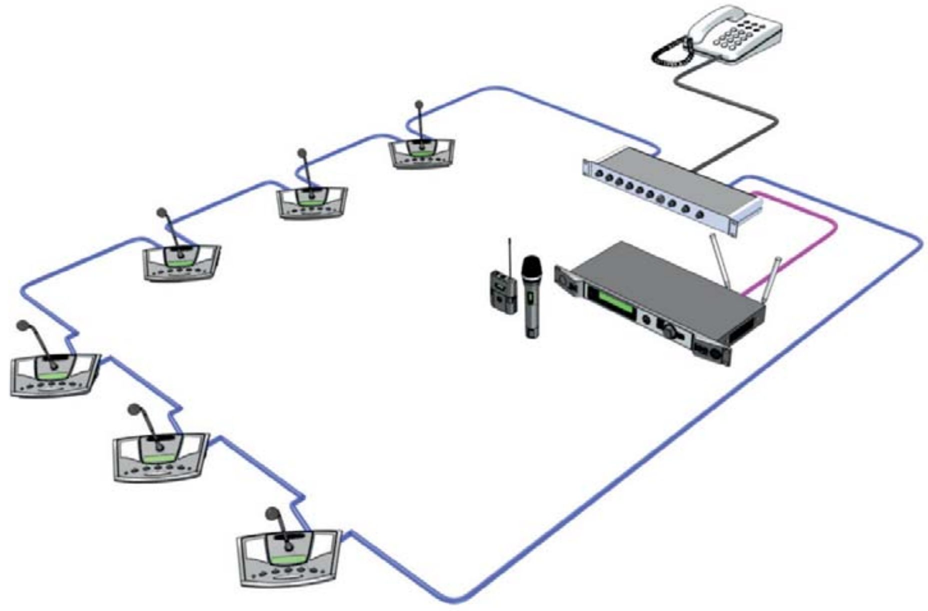
Discussion-system microphone stations are usually available with a **gooseneck** or **boundary microphone.** Which type and polar pattern to choose depends on the kind of system envisaged. When designing a **mobile system,** it is important to find out as much as possible about the room beforehand. All the same, many decisions can only be made at the venue itself, especially where the space available for setting up the sound system components is limited. It is always a good idea to bring some **extra equipment** in case some unexpected needs arise at the last minute.

Discussion systems offering a choice of **gooseneck lengths** and **microphone polar patterns** provide considerable **advantages.** If any acoustic problems arise, microphones can still be replaced at the last minute. Loudspeakers built into the microphone stations are both economically and acoustically beneficial. They save the need for external loudspeakers and system designers do not need to worry about loudspeaker positioning and the related feedback problems. Mobile systems with integrated loudspeakers can be set up extremely quickly.

Another important issue in the operation of discussion systems is the question of **how disciplined delegates are going to be in using the microphones.** The more people attend a discussion with no moderator, the more likely the meeting may get out of hand. To anticipate situations where several delegates speak at the same time, trying to drown out the others, it is **absolutely necessary to use a mixer with NOM limitation.** Unlike automatic mixing, the NOM limitation function does not reduce gain but will not open any more microphones as soon as the selected number of microphones are open.

Small discussion systems are normally run in automatic mode today. The control unit allows the NOM limit to be set, and the delegates can open their own microphones by pressing a button on the microphone station – as long as the NOM limit has not been reached. Once the limit is reached, all other requests for the floor are put on a waiting list and the appropriate microphones opened as soon as another delegate turns their microphone off.

Well-appointed microphone stations provide a **priority function.** The microphones on these stations can always be activated irrespective of NOM limitation. This type of microphone station is normally reserved for the chairperson of a meeting and allows them to mute all other microphones.



These microphone stations also allow the president or a technician to give, or refuse, the floor to speakers. Delegates can ask for the floor by pressing a button on their microphone station. The chairperson can then give the floor to them or mute their microphone to ensure that no more than the desired number of talkers will speak at a time.

Conference systems:

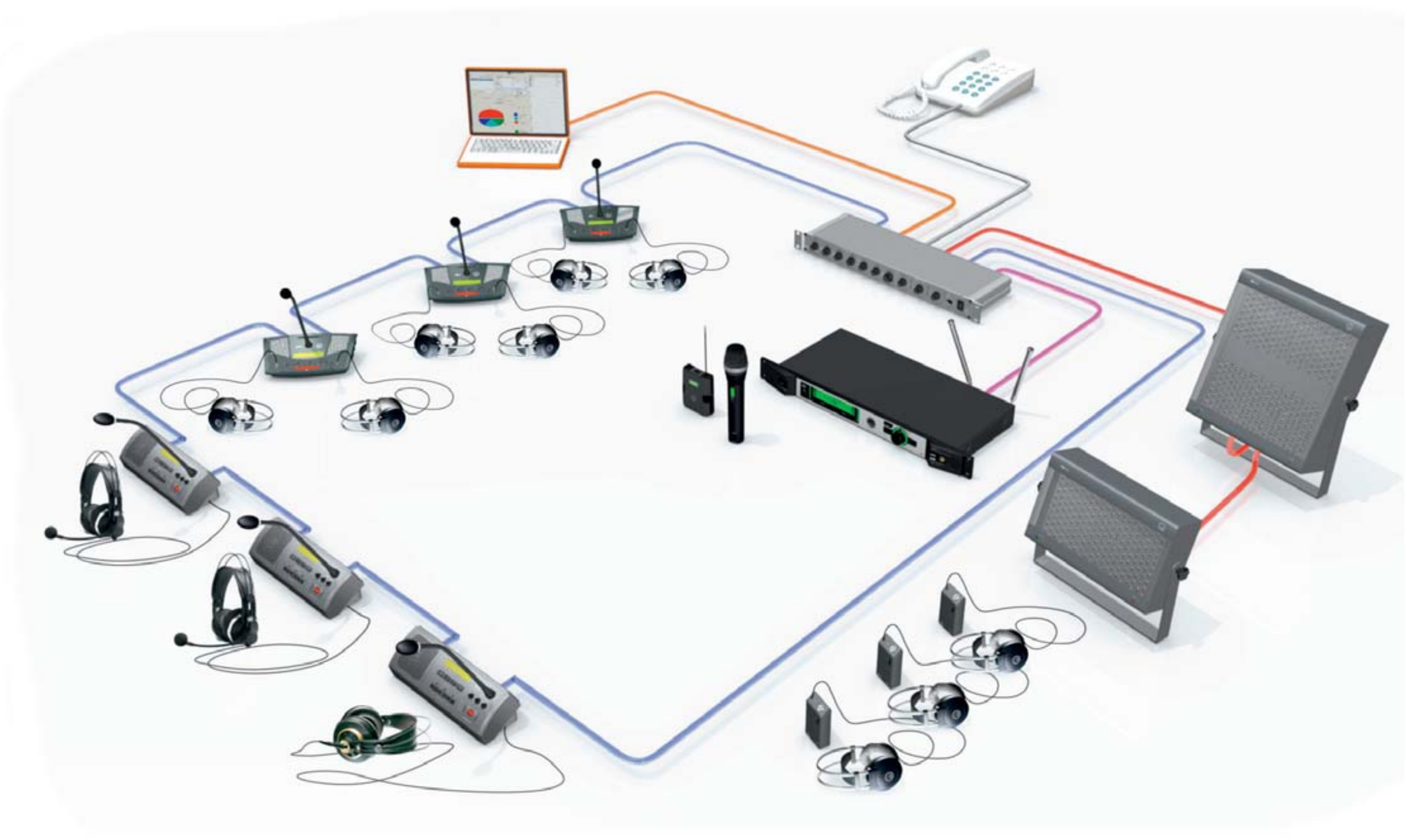
Designed for use at large international conferences, **conference systems** are **discussion systems with such added functions as recording and voting.** They can be **controlled by a computer** that may also provide a database of all delegates. This makes it easy to make lists of speakers, carry out votes, and record the entire conference.

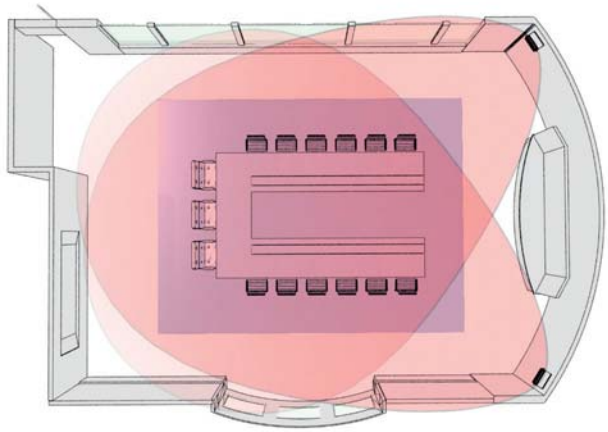
International conferences with hundreds of delegates speaking many different languages need interpreters. **Simultaneous interpretation** depends on appropriate equipment, and good conference systems include **interpretation booths tailored to the needs of the interpreters.**

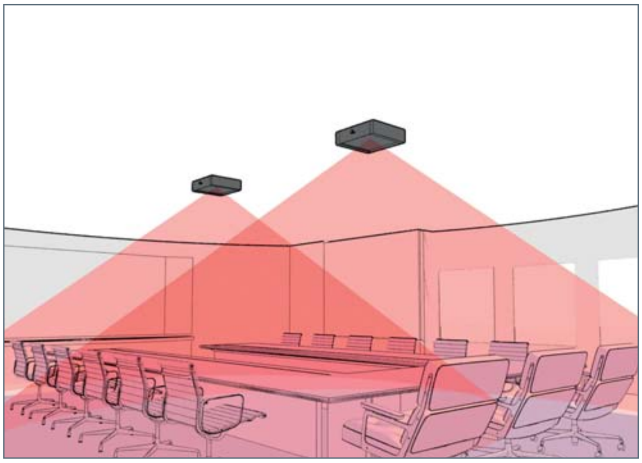
The control desks in each booth must allow each interpreter to **listen to the person having the floor, or to another interpreter** if necessary. If a speaker uses a language that only one booth is capable of interpreting, all other booths will have to use this booth's interpretation as their source ("relay") language. A **list of language pairs and relay languages** helps the system designer to include **all the interpretation equipment required** to **ensure a smoothly functioning conference system.** The job of a conference interpreter is extremely strenuous and responsible. Therefore, the necessary equipment needs to assist them in doing their job without distracting them.

The interpreted languages are transmitted to the delegates by a **language distribution system.** These systems can also be used to help persons with limited hearing. The **signals are distributed via cables, induction loops, infrared, or radio.** Permanently wired systems and systems using induction loops are more expensive and limit the mobility of delegates. Therefore, conference system designers today prefer infrared or radio systems.

A highly convenient solution is to incorporate a language selector in the microphone station, allowing the delegate to choose the interpretation they wish to listen to on their headphones.







Infrared language distribution systems:

Infrared language distribution systems **allow delegates to move around freely in the conference room.** Also, these systems can be **set up and taken down very quickly,** which pays off quickly, too.

Unlike radio signals, **infrared signals cannot escape from an enclosed conference room.** Unless somebody opens a window or door, **nobody outside the room can listen in on the signal.**

Infrared (IR) systems need no communication authority approvals, and there can be no interference from other radio equipment or cell phones. However, intense sunlight, dimmed fluorescent tubes, energy saving lamps, or photoflashes can affect the transmission quality.

Therefore, special attention should be paid to these **extraneous light sources (stray light).** IR systems comprise several components. A central unit modulates the output signal of each interpretation booth onto the IR channel assigned to the corresponding language. Coaxial cables feed the resulting signal to IR radiators that use infrared diodes to broadcast the signal to the room. Portable, battery operated receivers allow the delegates and audience to select the desired language and listen to it on headphones. In **designing an IR system,** the **main objective is to cover the entire conference room adequately with IR light.** To achieve this object, it is necessary to know the coverage angle and reach of each IR radiator. Based on the radiators' specifications, the number of IR radiators needed can be shown in a diagram.

SUMMARY: Sophisticated technology boosts communication

> Professional **discussion systems are highly specialized** and **easy to operate.** They can control up to hundreds of microphones. Microphone stations are connected by special cables to a closed-circuit bus or several single-ended lines in order to increase **system reliability.**

> The **microphone stations** are available with **gooseneck or boundary microphones,** with a free choice of polar patterns. **Integrated loudspeakers** are cost-efficient and less prone to feedback.

> In the **automatic mode,** the number of open microphones (NOM) at any one time is set at the control unit. The **priority function** allows certain microphones to be activated at all times while muting all others.

> **Conference systems** are discussion systems with such added functions as **recording and voting.** They are controlled by a computer that also provides a **database** of all delegates.

> Conference systems can also support **simultaneous interpretation.** Interpreters can choose between **different language channels.** The interpreted languages are transmitted to the delegates by a **language distribution system.**

> Signals are distributed via **cables, induction loops,** infrared or radio. Conference system designers today prefer **infrared or radio systems.**

> **Infrared language distribution systems** need no communication authority approvals, are less prone to interferences, safe from interception, and they allow delegates to move around freely. Coaxial cables feed their signals to the transmitters that use infrared diodes to broadcast the signal.

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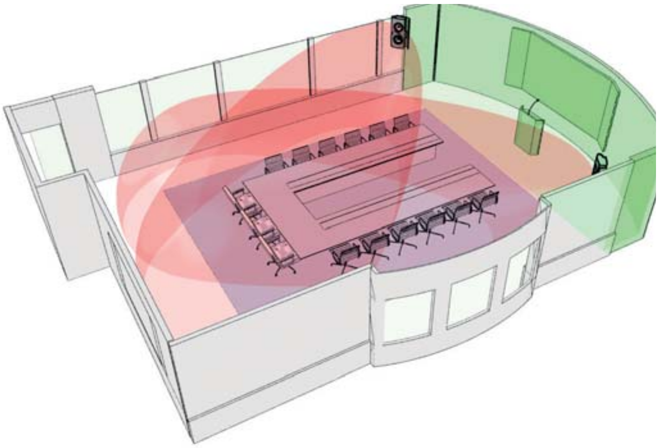
Applications:

Lectern

Sound systems with lecterns are **simple to design and use.** The positions of lectern and loudspeakers are known beforehand and will normally not change during the event (unlike a talker with a wireless microphone). This minimizes the risk of sudden feedback problems.



For **perfect intelligibility,** focus on the following points: **loudspeakers** must **never** radiate directly **into the microphone.** To **avoid reflections** that might trigger a feedback loop, the lectern should not be placed in front of strongly reflecting surfaces (windows, concrete walls). A **padded wall background** is a much better solution. The microphones are placed on the lectern or may even be permanently mounted. In both cases, good **vibration noise insulation** is essential; failing this, any sound caused by touching the lectern (especially knocking sounds) will be picked up by the microphone and transmitted via the system.

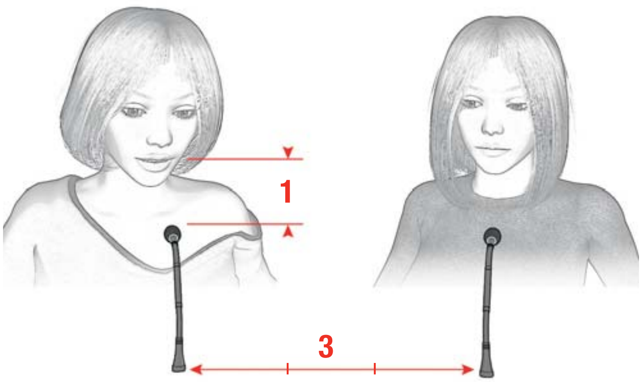
For permanent installation, there is a wide variety of options. The **correct rubber mix of the absorbing element** is essential: insufficient internal attenuation of desktop-mounted or built-in devices will fail to **disrupt noise transmission.** Apart from vibration sound, it is important to avoid direct transmission of background noise such as papers being shuffled. This is ensured through the focused use of directional microphones.

Lecterns are used with two types of microphones: **boundary microphones** and **gooseneck microphones.** The former can be mounted **almost invisibly** so as not to interfere with the overall design. However, basing the design on visual considerations will cause acoustic problems: since the boundary is the tabletop itself, background noise such as the shuffling of papers will be transmitted very loudly. Besides, the talker's mouth is rather too far from the microphone, while papers are very close to or – in the worst case – even on top of the microphone.

Gooseneck microphones provide **considerable advantages** for the use **on lecterns:** the microphone is **close to the mouth,** and even inexperienced users will see it at first glance. This is a very valuable feature, since most talkers are not really used to talking into microphones and frequently do not know which direction to choose. Moreover, the adjustable gooseneck allows the microphone to be **positioned more accurately;** microphones with a **strong directional effect further increase** the **proportion of wanted sound.**

Some talkers tend to move their heads from side to side. This is frequently compensated through **close spacing of two microphones.** However, this is **not a good idea** because it leads to a number of problems: the microphones are roughly at the same distance from the talker as from each other. Since the talker is practically never in the exact middle between the two microphones, the sound will hit the microphones with a slight delay but at nearly the same level. This will cause partial **cancellation of the audio signal** in the mixing desk – the so-called **comb filtering effect.** This effect is particularly strong in the 1-2 kHz range (path difference 8 -15 cm), causing hollow sound and poor intelligibility while also increasing the feedback tendency of the system.

There are two options for avoiding such interferences: to increase the pickup range (because of head movements), we need to apply the **3:1 rule.** This means that two adjoining microphones must be placed at least three times as far from each other as the sound source from either of the two microphones. This rule also applies to **discussion systems** whose microphone stations are **spaced very closely together.**



If, however, two microphones are used in order to ensure **system reliability** (redundancy), the **microphone elements** should be spaced as closely as possible – **preferably one above the other.** Thus the unwanted comb filtering effects are reduced to a high-frequency field where they are not perceived anymore. In any case, an **automatic mixer should be used** for two microphones; yet **a single directional gooseneck microphone** is the better solution.

Miking the audience

Optimum protection against vibration noise

Unlike conventional rubber mixtures (Fig. 1) whose vibration damping effect is very slow, the special AKG rubber mixture (see right) absorbs vibrations almost immediately (Fig. 2).

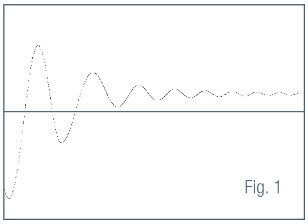


Fig. 1

Fig. 2

SUMMARY:

A time-tested classic: The lectern

> Sound systems with lecterns **minimize feedback** and are **easy to design.**

> Loudspeakers must **not radiate directly into the microphone.** The lectern should **not be placed in front of strongly reflecting surfaces.**

> Microphones need high-quality **insulation against vibration noise.**

> The use of **directional microphones** helps avoid the **transmission of background noise.**

> Lecterns use **boundary** and **gooseneck microphones:**

- **Boundary microphones** allow nearly invisible installation but are prone to **acoustic problems.**
- **Gooseneck microphones** can be placed close to the mouth, providing **great acoustic benefits.**

> When **two microphones** are used to compensate for **head movements,** follow the **3:1 rule.**

> When **two microphones** are used, space them **as closely as possible** to **ensure failure safety** and use an **automatic mixer.**

However, a single strong **directional gooseneck microphone** provides a better solution.

For miking **contributions from the audience** there are two options:

From an **acoustic** point of view, it is **best to place one or several microphone stands in the room,** choosing their positions so as to **increase feedback protection.** This will help avoid handling noise or other unwanted sounds.

Passing a wireless mike around is more comfortable for participants, enabling them to talk directly from their seats. The microphone **signal** should be **checked** by a **sound engineer** or **automatic mixer** in order to mask unwanted handling noise at least in part.

As long as the use of **wireless mikes** is **restricted to the stage,** this will cause **few problems,** since it is easy to accomplish **full antenna coverage.** This limited area can be easily kept free of direct disturbing sources.

Using them in the **audience area** of a large conference room requires a **more sophisticated antenna network** that needs to be taken into account at the designing stage. Because of the good absorption of radio waves by the human body, it is imperative to ensure proper antenna positioning **(line of direction equals line of sight!)** to **prevent dropouts and shadow effects.**

Wireless microphones react **sensitively** to interferences of other radio waves; such disturbing sources are found abundantly in the audience: computers, cell phones, wireless mice, bluetooth signals or W-LAN networks. These bothersome high-frequency transmitters are closer to the microphones (receivers) than the antennae (transmitters) themselves, thus causing especially strong disturbances. Such **interferences** can only be **prevented** through **professional frequency management,** i.e. by avoiding these frequency bands altogether!

If none of these can be used, **suspended or directional microphones from a longer distance** may be used as a **last resort.** However, experience has shown that this will cause **substantial problems** and requires controlled activation (automatic mixer, technician).

SUMMARY:

Voices from the audience – ideal gain

> **Best option:** placing one or several microphone stands in the room.

> **More comfortable option: passing** a **wireless mike around** whose signals should definitely be checked by a **sound engineer** or an **automatic mixer.**

> **Proper antenna positioning** (line of direction equals line of sight!) **prevents dropouts and shadow effects.**

Applications:

Moving around

Fig. 1

Fig. 2

The challenge of miking lecturers **moving freely about a room** is much larger than capturing the speech of a user at a lectern. Three ominous standard situations keep cropping up to the despair of technicians.

The **speaker's position vis-à-vis the loudspeaker is not fixed:** S/he suddenly appears in front of the box, causing loud **feedback screeching.**

The **microphone's position vis-à-vis the sound source** (mouth) **is not fixed:** the lecturer walks about with a handheld mike in front of a projection screen. How many seconds will it be before s/he uses the **microphone as a pointer** for the first time, thus **disrupting sound transmission?**

The **microphone's position in the speaker's hand** is **not fixed,** either; s/he **puts it aside** to distribute information, causing two loud noises – when the mike hits the tabletop; and when it falls to the floor.

Generally, **hand-held microphones** are **a real headache** at **conferences:** they are best used only for **passing around the audience** or as stand-in for a gooseneck microphone (on a table stand or floor stand).

In all **other cases,** it is better to use **headsets** (Fig. 1) or **clip-on microphones** (Fig.2) with an attached radio transmitter. The speaker has his/her **hands free,** and the **distance from the microphone** (as well as the sound level) remains **constant.**

Directional microphones work even better: they **attenuate ambient noise.** This ensures **maximum feedback protection** with **optimum intelligibility.**

The only downside is their visual appearance. Although these microphones are available in skin-colored design, many speakers do not want to appear with an unfamiliar object in front of their face.

Clip-on microphones (Fig.2) offer an alternative, but their drawbacks are obvious: the distance to the microphone increases, the wanted signal softens, while the unwanted signal gets louder. The system must be turned up, which increases the feedback tendency. Besides, microphones attached to the clothes transmit friction sounds on the microphone element or cable (as known from radio or TV).

Fig. 4

Fig. 3

Speakers who move about are best miked with the aid of bodypack transmitters (Fig. 3) combined with headsets or lavalier microphones, or with a handheld wireless microphone (Fig. 4).

SUMMARY:

Mobile miking – a challenge

> The **best option** for applications with moving speakers is the **headset,** best of all with a directional microphone.

> **Second best** is the **clip-on microphone,** preferably with a directional polar diagram.

Designing
a discussion or conference system

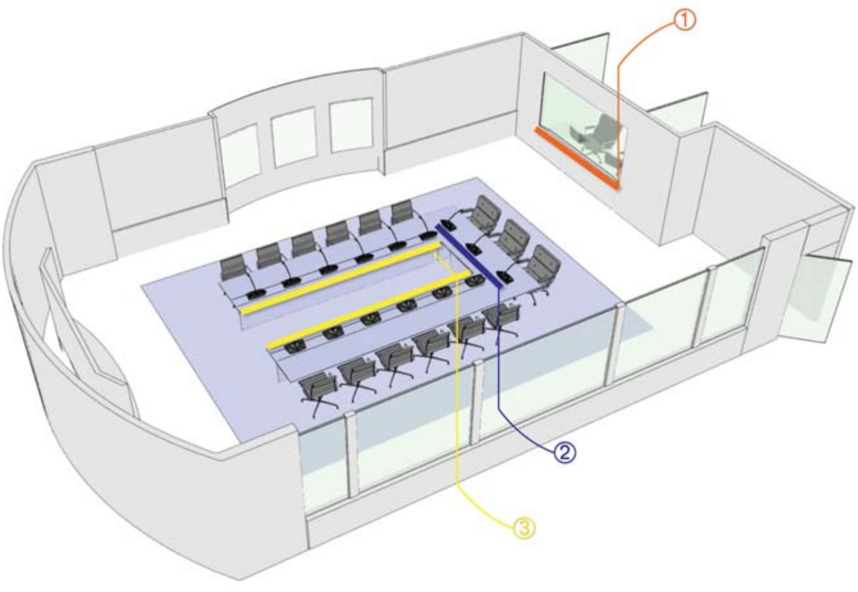
When **designing discussion or conference rooms,** you need to include **various sets of issues, relating** them to each other in a **sensible manner.** First of all, identify **architectural** conditions and requirements. Do you want microphones, loudspeakers and transmitters to be visible or do you prefer concealed installation? This question affects the selection of design types and polar patterns.

For the next step, you need to define the desired **functions of the system:** will microphones be activated automatically or manually? What about quantity and mode of operation (mobile or stationary?) Will voting take place? Are the loudspeakers to be placed in the room, or are they part of the microphone station? Which are the external sources for feeds? Will you need channels for simultaneous interpretation?

Depending on the **available budget,** you can now define whether each conference delegate needs to have his/her own microphone, and how many microphone channels will operate simultaneously. Having specified their quantity, you can now choose the type and scope of performance of the control unit (auto mixer, discussion or conference system). This is also the time to plan the amount of additionally required channels, such as music feed, computers, DVD, telephone or video conference systems.

Obtaining this information requires **in-depth discussions with all persons involved: architects and designers** know about statics, construction plans, color selection and design. **Acousticians** measure the reverberation period and the expected level of background noise in the room. They are familiar with peculiar acoustic phenomena such as flutter echoes and can specify sound-absorbing surfaces. **Operators and users** should be contacted in order to identify the type and organisation of the meetings, desired interaction facilities, as well as the number and qualification of the delegates. The **operating staff** is an important source of information for typical procedures and their structures; they will also be familiar with the causes of previous problems.

In certain cases, **mobile discussion units** are a compact and low-cost alternative. All components of a mobile unit are perfectly matched to ensure fast assembly of the unit. A simple mobile unit can be reached with just a **few boundary microphones.** These are arranged on the conference table at regular intervals and monitored by an auto mixer.

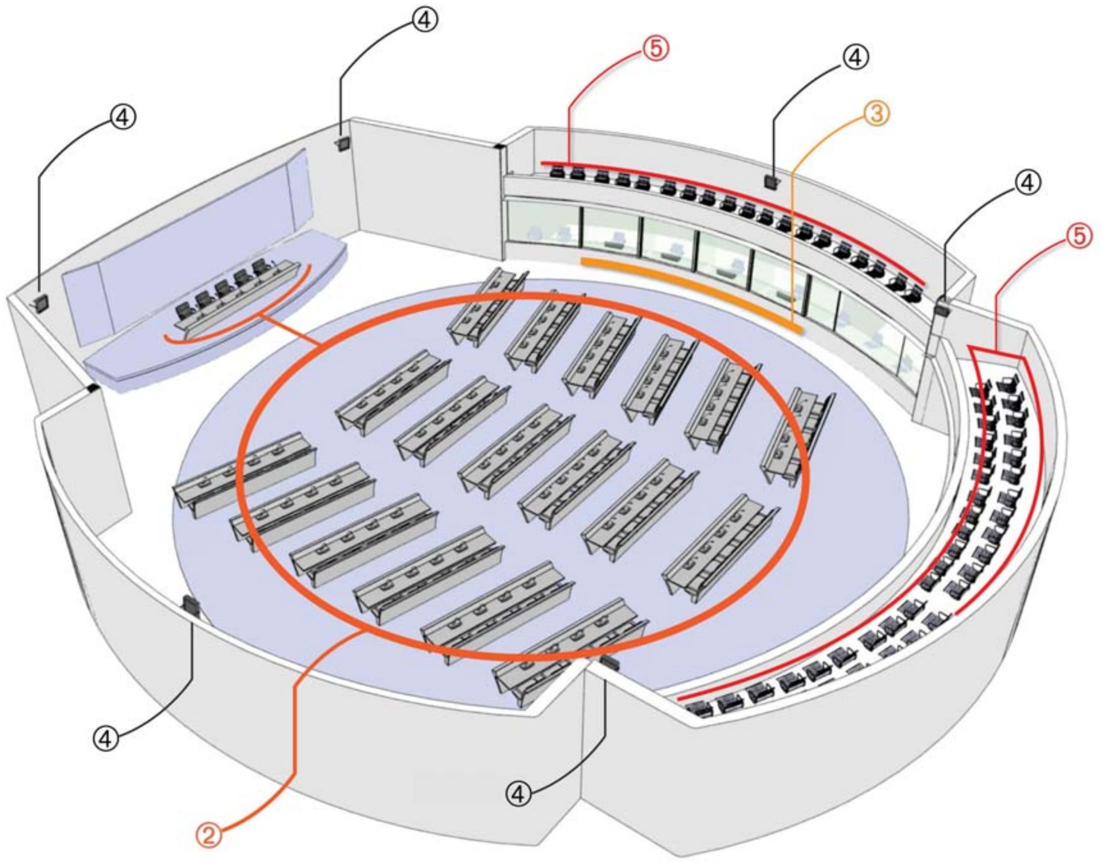


Discussion system –
Just a few components will do:

1 Basic control unit
2 Moderator’s microphone station
3 Delegates’ microphone station

With just a few components, a discussion system offers lots of operating comfort and sophisticated technology. And what’s best: If you want to expand the system subsequently, this can be done without difficulty – with a professional discussion system combined with the existing components!

The basic control unit manages the entire system. You can define and freely configure the number of persons able to speak at a time (NOM). From his/her microphone station, the moderator controls the discussion with the call-to-order button, arranges votes or sets priorities for the sequence of speakers. Delegates participate in discussions and votes from their microphone stations.



Conference system – A professional conference system can be expanded almost ad lib.:

1 Basic control unit
2 Chairperson‘s microphone station
3 Interpreter‘s microphone station
4 Infrared transmitter
5 Infrared receiver

The expandable conference system is up to any challenge and provides all technological options imaginable.
There is a technical equipment room with basic control unit, event technology equipment for video feeds, illumination and telephone conference circuits. Apart from the standard functions of a discussion system, a conference system allows many more people to attend as listeners without speaking themselves. IR devices provide wireless transmission of the interpreted signal to IR receivers that let you choose among several languages.

For acoustic reasons (distance from the microphone, ambient noise) **gooseneck microphone stations** ensure even **better sound results.** It will be best to plan for each delegate to have his/her own microphone. Under certain circumstances, **loudspeakers integrated in the microphone stations** will suffice as a sound system. In such cases it is possible to do without further components such as mixing desk, amplifier or external loudspeakers.

For conference rooms with permanently installed seating and tables, it will be best to provide for the **permanent installation of microphones or microphone stations.** The **wiring** is **concealed from view,** thus ensuring improved protection against cable breaks or inadvertent disconnection of microphones by delegates.

Vibration noise transmission between the microphones/microphone stations and the tabletop must be **interrupted.** For acoustic planning, array the microphones/ microphone stations so as to avoid interferences (feedback). Unless done beforehand, this problem is frequently difficult to remedy and will involve high expenses. In such cases, the best solution is using discussion stations with integrated loudspeakers.

International conferences come with much more sophisticated requirements with regard to planning and technology. The highly complex organisational and structural procedures must be taken into consideration by means of appropriate **design specifications.** The use of **interpreters** will require proper constructional, acoustic and technological resources.

Depending on the number of languages used, you must ensure the availability of a **sufficient amount of interpreting channels and booths.** Each language requires its own room. For the **language distribution system,** a **channel scheme** will be set up in order to define the language assignment to individual channels.

When **planning** a conference **in detail,** it is essential to identify a **source language,** because interpreters cannot be expected to be proficient in all languages used on the floor. One interpreter is chosen as reference interpreter whose interpretation serves as the basis for all other interpreters (relay interpreting).

Professional conference systems have a **conference management programme** that is used for pre-planning the conference and establishing a database of the delegates. Individual delegates will be assigned speaking authorization or priority level, and their voting competence will be defined. This software is also used for recording the conference procedure (speakers' lists, voting results …).

Large-scale conferences will also require **external feeds from various sources** as well as the integration of various pulse generators (camera control, AMX/Crestron linkup, sound web integration …). These functions are ensured through integration in a larger media control system.

SUMMARY: **Proper planning is the foundation of every system**

> It is essential to begin with **in-depth discussions with all persons involved** (concerning **architecture, acoustics, functionality** and **budget),** including:

- **Type and organisation** of the meetings
- Desired **interaction facilities**
- Delegates' **number and qualification**
- **Typical event procedures** and their structuring characteristics

> **Mobile discussion units** may be a **compact and low-cost alternative.**

> Interpreters need a sufficient quantity of **interpreting channels and booths:** one room per language. A **channel scheme** is set up for the **language distribution system.**

> Professional conference systems have a **conference management programme.**

> Large-scale conferences require external **feeds from various sources,** which is why they are integrated in a larger **media control system.**

Glossary

Explaining technical terms – from A to Z

Antenna Cable
Cable specifically designed for RF signals. Used for connecting a remote antenna to a receiver. Antenna cables are typically coaxial and symmetrical. Signal attenuation depends on the frequency band of the signal as well as the length and quality of the cable and is quoted for a 100-m run of cable.

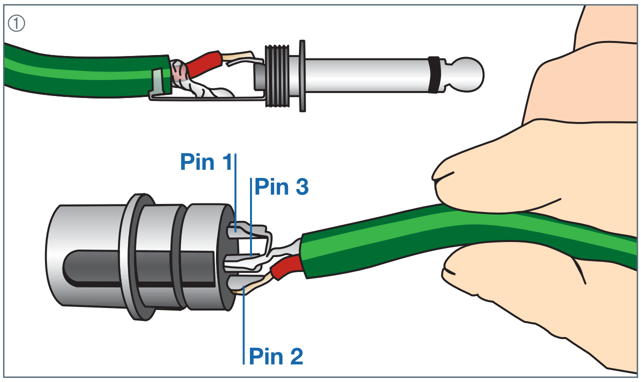
Antenna Splitter
Electronic network specifically designed for RF signals. Distributes an antenna output signal to several receivers. Powered antenna splitters use an amplifier to compensate for cable attenuation while passive antenna splitters have no amplifier.

Balanced/Unbalanced Connections
Microphones can be connected to an amplifier with either balanced or unbalanced cables. In a balanced cable, the signal is carried by the two inner conductors and the shield is not part of the signal path. Even with long cable runs, any external interference signal (such as power line hum) would be induced equally in both conductors and thus be canceled. Unbalanced cables use only one center conductor as the “hot” wire, the shield being the ground (“cold”} lead. While this arrangement works well with cables up to 10 meters in length low-frequency, long-wave hum interference may be picked up by longer cables which act as a long-wave antenna.

BNC
Connector specifically designed for RF lines.

Booster
Amplifier for RF signals. Boosters are connected between a transmitter output and the antenna in order to increase radiated power (custom product).

Condenser Microphone
The transducer element consists of a vibrating diaphragm (metalized foil) only about a ten thousandth of an inch thick and a fixed metal electrode (back plate). The two electrodes make up a capacitor (condenser) charged by an externally applied DC voltage 1“ polarizing voltage or carrying its own permanent charge. The sound waves driving the diaphragm will vary the capacitance of the capacitor and consequently the microphone output voltage will vary in step with the sound waves.



Condenser microphones, also called “capacitor microphones”, need an impedance converter (preamplifier) to match the very-high-impedance condenser transducer to low-Z inputs. Condenser microphones usually have a flat frequency response, high sensitivity, and good transient response. They require a power supply. All AKG condenser microphones are designated by the letter(s) “C” or “CK” in front of the model number.

Connecting AKG Microphones
All handheld microphones are low-impedance 1,200 to 620 incorporating a balanced output on a 3-pin male XLR connector. Conforming to IEC 268-12, pin 1 is ground, pin 2 high, and pin 3 low. The output is compatible with all mixers, tape recorders, etc.

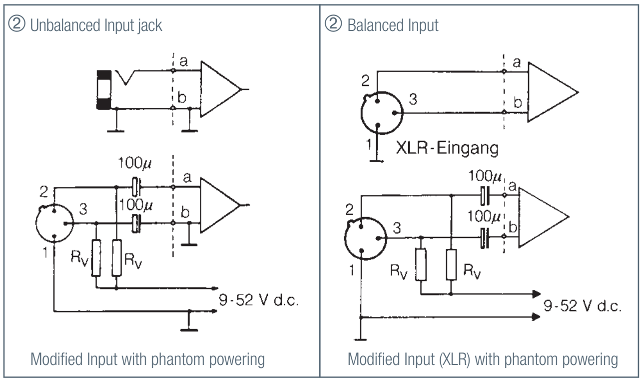
To connect an AKG microphone to an input jack, wire the microphone cable as follows: connect the sleeve of the jack plug (ground) to the cable shield and the shield to pins 1 and 3 on the XLR connector. The center (“hot”) wire connects pin 2 to the jack plug tip (see diagram 1).

If your installation uses pin 3 as “high” or “hot”, bridge pins 1 and 2 for unbalanced connections and make sure to follow the same convention for all cables in order to avoid phase reversal problems.

Very old sound systems sometimes have high-impedance microphone inputs. Should the signal of a low-impedance microphone be too weak, insert a 1:10 step-up transformer at the amplifier input.
Long cable runs used with high-impedance equipment cause high-frequency loss. The same applies if you connect a microphone to a high-impedance guitar amplifier input.

Connecting Condenser Microphones
Condenser microphones – except for the battery powered C 1000 S – require an operating voltage that needs to be fed through the microphone cable (phantom powering). This can be done in several ways:

1. From a mixer with built-in phantom power (9 to 52 V).



Modified Input with phantom powering

Modified Input (XLR) with phantom powering

2. By modifying the mixer or tape recorder to provide phantom power: find a regulated DC voltage between 9 and 52 V in the power supply. All modern AKG condenser microphones accept any voltage within this range. Wire the input(s) as shown. Current consumption of the phantom circuit is negligible (about 1 mA per mic). Replace the input jacks with XLR sockets if possible. While stereo jacks will work as well, there may be a risk of mistaking them for send/returns or the like.

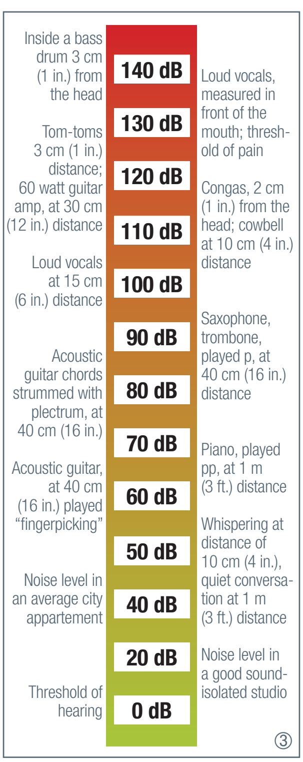
Use the following standard resistances (IEC 26815) for Rv:

Voltage	Resistance	
12 V (±2 V)	680	+10%
24 V (±4 V)	1.2 k	±10%
48 V (±4 V)	6.8 k	±10%

Make sure to use resistor pairs whose combined actual value is within 0.4 % of the specified value!

3. By inserting N 62 E or N 66 E AC power supplies between the mixer and microphones.

4. By using the B 18 battery power supply which is ideal for outdoor recording.



Crosstalk
The undesired coupling of signals from one channel to another channel.

dB SPL
Decibel Sound Pressure Level. A measure of the sound level referenced to 20 µPa (the sound pressure corresponding to the threshold of human hearing). A 6-dB increase in SPL would sound about twice as loud.

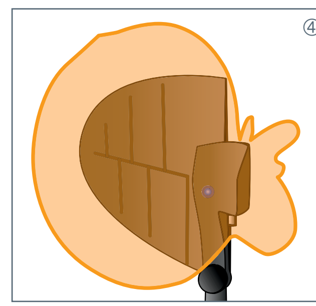
Deep Fade
Massive decline of received signal strength due to cancellation of the carrier in multipath transmission situations.

Directivity Factor
The directivity of a microphone can be expressed in terms of the amount of sound energy it absorbs out of a diffuse sound field. The directivity factor indicates how much less sound energy is absorbed by a directional microphone than an omnidirectional microphone.

Distortion
Dynamic microphones virtually never distort the signal. To be precise, their distortions at very high sound pressure levels (<130 dB) cannot be measured because loudspeakers are incapable of reproducing such levels distortion free. For this reason, we state no maximum SPL for dynamic microphones.

However condenser microphones with their built-in preamplifier may overload at high sound levels. When close miking (from a few inches) loud instruments such as drums or trumpets the microphone sensitivity should be reduced. With the C 535, simply use the preattenuation switch.

Directional Antenna
Antenna whose sensitivity is highest within a limited angle in front of the antenna. Directional antennas are used mainly where standard receiving antennas cannot be mounted within the range of the transmitters so the transmitter signals must be picked up from greater distances (e.g., in open-air arenas).



Diversity
Reception technique that ensures clear reception even in difficult environments. Diversity receivers use several antennas for the same carrier frequency and some models use several receiving sections, too.

Downtime
Period of time during which a system is inoperative.

Dropout
Momentary loss of signal due to squelch operation or interference.

Dynamic Microphone
A coil attached to a diaphragm is driven by the sound waves and vibrates between the poles of a magnet. This movement induces in the coil a voltage which corresponds to the sound pressure.

Dynamic microphones handle high sound levels without overloading and are very rugged. They require no operating voltage. Dynamic microphones from AKG are designated by the letter “N” in front of the model number. Also known as “moving coil microphone”.

Glossary

Explaining technical terms – from A to Z

Electret Condenser Microphone
Condenser microphone that needs no polarization voltage. Instead, a special metalized plastic "electret foil", in which a permanent electrical charge has been stored by application of heat and a high polarizing voltage, is used either for the diaphragm or the fixed electrode. The latter type is called "back plate electret microphone".

Electromagnetic Wave Spectrum
Range of frequencies of electromagnetic radiation.

Environment
Dynamic microphones will generally stand up to extreme environmental conditions such as temperatures from -25 °C to +70 °C and high humidity.

Condenser microphones, however, are susceptible to humidity and condensation. When an object is damp and colder than its environment, condensation water will form on its surface. Drops of condensation water inside the transducer or high-impedance preamplifier will cause crackling noises.

Storing condenser microphones:

1. Store the microphone in a dry and warm place. It should never be colder than its environment. If it has been transported in a cold car or van, allow it to warm up before use.

2. The supplied silica gel absorbs humidity. It will maintain this property as long as you keep it in the sealed package and may be regenerated in the oven if necessary.

3. Be sure to protect condenser microphones from rain when using them outdoors.

Equivalent Noise Level
Since condenser microphones incorporate a preamplifier, they introduce a low amount of self-noise which appears at the microphone output as an unwanted signal voltage. This noise voltage is measured using standard weighting filters and the result stated as the equivalent noise level in dB. An equivalent noise level of 20 dB, for instance, means that the self-noise of the microphone is as loud as a sound at 20 dB SPL (see dB SPL).

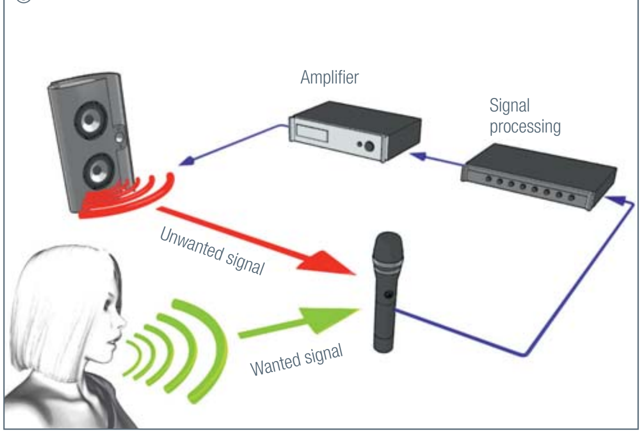
Noise level in quiet recording studio:
A low equivalent noise level means that the microphone's self-noise is low. The self-noise voltage is weighted either conforming to IEC 268-1 and DIN 45 405 using the filter according to CCIR 468-3 with the "quasi-peak" value being quoted, or in accordance with IEC 651 or DIN 45 412 using the A-weighting curve with the rms value being quoted. Studio engineers seem to prefer the CCIR weighting while A-weighting is still accepted as well.

ERP
Equivalent Radiated Power,
a measure of a transmitter's RF output.

Far-Near Difference
The difference between the shortest and the longest distance between stage and antenna.

Feedback
When a microphone picks up amplified sound from a loudspeaker this signal will be reamplified, picked up again, etc., until the commonly known shrill howling (sometimes a lower midrange rumbling) sets in. In small rooms, feedback is usually caused by reflections. In this case, acoustic treatment of the walls should help. On stages with correctly set up FOH speakers it is the monitor speakers that may cause feedback. A very good hypercardioid microphone (e.g. a D 3900) may sometimes provide a few extra dB's of gain-before-feedback. Place the monitors slightly off-axis (135-) where the microphone is least sensitive.

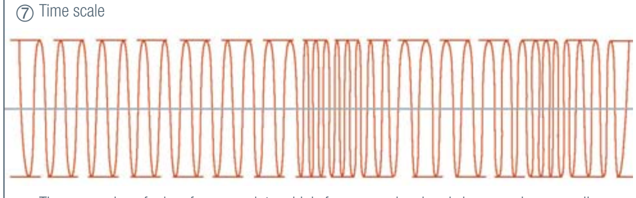
⑥



Frequency Management
Organization of frequency resources.

Frequency Modulation
A technology that alters (modulates) carrier frequencies to transmit information.

⑦ Time scale



The conversion of a low-frequency into a high-frequency signal and vice versa is accomplished via frequency modulation, i.e., by applying a high-frequency sinus carrier wave (HF) to the low-frequency audio signal (LF).

Frequency Range
The frequency range of a microphone is usually stated as the upper and lower frequency limits within which the microphone delivers a useful output signal.

Frequency Response
Microphones are not equally sensitive to all notes. The frequency response indicates the relationship between sensitivity and pitch. The 0-dB reference being the output voltage at 1 kHz, the frequency response is measured at constant sound pressure level, from about 20 Hz (lowest note) to 20 kHz (above the upper limit of human hearing).

Hum Sensitivity
Magnetic fields from amplifiers, long power cables, and lighting systems in particular may induce hum in microphones.

A microphone's hum sensitivity gives an indication of how susceptible it is to this kind of interference. Values are 3 µV/5 µT for dynamic microphones with hum suppression coil, 30 µV/5 µT for dynamics with no suppression coil (D 90, D 95, D 190), and up to 10 µV/5 µT for condenser microphones.

In practice, though, it is the microphone cables, most of all unbalanced ones, and mixer inputs, that are most likely to pick up hum.

Impedance
Frequency dependent AC resistance of a microphone. Always quoted at 1 kHz the actual impedance at other frequencies may differ slightly from this reference value. Also known as "source impedance".

Intercept Point
The Intercept Point (IP) provides a measure for an amplifier's resistance to intermodulation distortion. IP 3, for example, is the reciprocal value of the third-order coefficient of an amplifier's nonlinear transmission polynomial.

Interference
Disturbance in transmission caused by extraneous signals.

Intermodulation
A nonlinear (multiplicative) combination of signals with different carrier frequencies that will produce completely new frequencies, called intermodulation products.

Limiter
Electronic circuit that prevents subsequent circuits being overloaded by excessive signal levels that would also cause distortion.

Line Microphone
The directivity factor of conventional unidirectional microphones is limited by the laws of physics. This can be overcome by installing a slotted tube in front of the diaphragm ("interference tube"). Off-axis sounds are canceled through interference, which results in an ultradirectional polar pattern.

Matching
Microphones should operate in an open circuit. This is the case if the input impedance of the preamplifier or mixer is at least 2 to 5 times as high as the microphone's rated impedance. The appropriate value is quoted in the specifications of each microphone as "recommended load impedance".

Maximum SPL
The highest sound pressure level (loudness) a microphone can handle without introducing more than a specified amount of "Total Harmonic Distortion" (1 %), in other words, without distorting the signal. Usually measured at 1 kHz, except for the C 460 B ULS Series where it is quoted from 30 Hz to 20 kHz.

Mechanical Noise
See "Vibrational Noise".

Memory Effect
The loss of capacity which occurs in nickel-cadmium storage batteries if they are not completely discharged prior to recharging.

Modulation/Demodulation
A sine-wave carrier starting at a time of minus infinity and ending at a time of plus infinity contains no information. However, any change in amplitude or frequency at any time (e.g., a pulse-like change) adds information to the carrier.

This process is called "modulation". The process by which a receiver detects and extracts this information from the carrier is called "demodulation".

Multichannel System
A wireless microphone system that allows several radio microphones to be operated simultaneously in the same room.

Noise Burst
Brief disruption of the desired signal by noise from a transient interference source (e.g., ignition spark).

Noise Skirt
An ideal carrier spectrum would be a line.

As the carrier is modulated, the noise inherent in the switching signals makes the transients look ragged. This raggedness ultimately frequency-modulates the carrier with noise. Once that happens, the carrier spectrum is no longer a line but a noise spectrum that tapers off to either side of the wanted frequency, which is why this part of the spectrum is called a "noise skirt".

NOM
(Number of Open Mics Limit)
Good auto mixers feature a special NOM limitation algorithm (Number of Open Mics Limit). The NOM limitation automatically adjusts the level of the microphones open at the same time, thus ensuring constant overall system gain without feedback.

Glossary

Explaining technical terms – from A to Z

Phantom Power
to IEC 2681 5/DIN 45596
Condenser microphones require an operating voltage. It can be fed to the microphone either by a-b powering or phantom powering. In a-b powering, the operating voltage is fed to the balanced audio wires without using the shield. a-b powering is incompatible with dynamic microphones since the operating voltage would flow through the moving coil and destroy it.

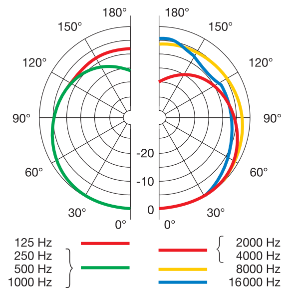
In phantom powering, the negative terminal is connected to the cable shield and the positive terminal is split via decoupling resistors to the balanced audio wires. Since both audio wires carry the same potential, no current will flow through the coil of a dynamic microphone so there is no risk of destroying it even if the phantom power is accidentally left on.

When adding phantom power to a single ended (grounded) input or an input with no front-end transformer, either capacitors or an optional transformer need to be wired into the audio lines as shown below, to prevent leakage currents from entering the input stage.

Polarity
If you use more than one microphone for a recording, they should be of the same polarity. This means that if the diaphragms move in the same direction, the output voltages of all microphones should have the same polarity. If they don't there will be signal cancellation effects causing sound coloration – particularly in the bass range – as soon as you mix the microphone output signals together.

Polar Pattern
The "polar pattern" of a microphone indicates its sensitivity to sounds arriving from different directions. Omnidirectional microphones "hear" equally well in all directions while all others prefer sound from one (unidirectional) or two (bidirectional) directions. The polar diagram shows the three-dimensional "hearing performance" of a microphone as a single curve. It is sufficient to plot only one half of the curve (0° through 180°) since the other half (180° through 360°) is symmetrical. In this way, the directivity can be shown for several different frequencies (broken, dotted, solid lines).

⑧



At 150° off-axis, the sensitivity is 17 dB down (referenced to 0°) at 125 Hz (solid line), and 10 dB down at 8 kHz (dashdotted line, righthand half). 150° means 150° left, right, up and down (see diagrams on the next page).

Pop Noise
In order to avoid those unpopular pop noises on stage, remember the following:

- Talk across the microphone head.
- Interestingly, pop noises are worst about 2 in. from the mic. So move either closer or further away.
- Perhaps use an extra foam windscreen.

Pressure Gradient Microphone
If both the front and rear of a diaphragm are exposed to a sound field, then the force that vibrates the diaphragm results from the difference between the sound pressures in front and to the rear of the diaphragm (called the pressure gradient).

The magnitude of the driving force depends on the distance between the front and rear sound entries, the frequency, and the angle of incidence and is therefore a directional variable which can be utilized to design directional microphones. Cardioid, figure eight, or hypercardioid polar patterns can be achieved by incorporating appropriate sound paths.

Pressure Microphone
If only one side (front) of a microphone diaphragm is exposed to a sound field and the other (rear) side sealed off by a soundproof case, the diaphragm will be vibrated by changes in sound pressure only. Sound pressure being a non-directional (scalar) variable, the microphone is equally sensitive in all directions. The resulting polar pattern is called omnidirectional 1.

Proximity Effect
In unidirectional microphones, as the working distance decreases, the output voltage rises more markedly at the low frequencies than throughout the rest of the frequency range. This is due to the fact that the diaphragm is vibrated by the pressure gradient between its front and rear surfaces and the pressure gradient is related to the curvature of the wave fronts.

This effect, known as "proximity effect", begins to become audible at a few hundred Hz and at extremely close working distances, the output level may be up to 15 dB higher at 50 Hz than at 1 kHz. This corresponds to about 6 times the normal output voltage.

Reflection
When a signal wave hits an obstacle, it will be reflected, i.e., bounce off the obstacle's surface at an angle equal to the angle of incidence.

Remote Antenna
Antenna that is connected by a special antenna cable to the antenna input socket on a receiver rather than directly to the antenna input socket.

Room Radius
In a room within which a sound is generated, e.g. by a loudspeaker, every point is characterized by its own unique ratio of direct sound and sound reflected from the walls.

The distance from the sound source at which the direct and reflected sound energies are equal is called the “room radius”. Outside the room radius the overall sound pressure level is constant throughout the room in the form of a “diffuse sound field”.

Transient
Temporary change in voltage or current occurring as a voltage or current source is switched on or off, e.g., a transistor controlled by a pulse signal.

Sensitivity
A microphone’s output voltage at any given sound pressure level. A more sensitive microphone will sound louder at the same gain setting (the feedback risk being proportionately higher). High sensitivity (condenser microphones) is needed to drive the mixer adequately when far miking quiet sound sources.

Sensitivity is commonly given in mV/Pa or dBV (referenced to 1 V/Pa) and measured at 1 kHz.

Here are some examples:

D 58	0.7 mV/Pa (-63 dBV)
D 190	1.6 mV/Pa (-56 dBV)
C 1000 S	6.0 mV/Pa (-44 dBV)
C 535	7.0 mV/Pa (-43 dBV)
C 451 EB comb	9.5 mV/Pa (-40.5 dBV)
C 460 Bcomb ULS/61	10.0 mV/Pa (-40 dBV)
C 562 BL	20.0 mV/Pa (-34 dBV)

Shadow loss
Signal loss which occurs in wireless transmission if an obstacle blocks the line-of-sight transmission path between transmitter and receiver.

Signal Loss
Signal loss in a cable may be due to ohmic resistance, dielectric leakage or radiation loss.

Signal-to-noise (S/N) Ratio
The S/N ratio is the difference between the reference sound pressure level of 94 dB (1 Pa sound pressure) and the equivalent noise level. Contrary to the equivalent noise level, a lower S/N ratio means higher noise and therefore a narrower dynamic range.

Squelch
Electronic circuit that switches the receiver off when the received signal is too weak so the associated extraneous noise and the self-noise resulting from the receiver being switched off will be inaudible. The squelch threshold is usually user adjustable within a preset range.

Tone coded squelch, tone code squelch, tone squelch
These terms denote a circuit that will open the audio path only when it detects a system-specific tone within the demodulated signal. This tone is higher than 20 kHz, the upper end of the range of human hearing, and is added to the audio signal by the transmitter.

Total Harmonic Distortion (T.H.D.)
A measure of the non-linear distortion of a signal (e.g. a sine wave) that occurs when a microphone or input is overloaded producing harmonics (overtones) at multiples of the fundamental frequency.

Transient Response
The ability of a microphone to follow sudden sound events immediately. Transient response depends on diaphragm mass, transducer damping factor, etc.

UHF
Ultra High Frequency

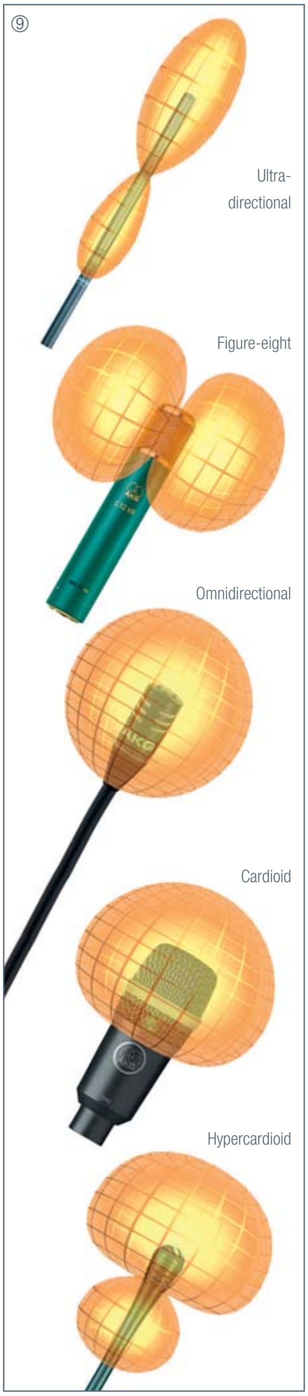
VHF
Very High Frequency

Vibrational Noise
In addition to air-borne sound, microphones also pick up mechanical noise such as impact, footfall, handling, or cable noise. Such unwanted noise can be reduced by special design features (transducer shock mount, compensation systems, bass cut)

Vocal Microphone
A microphone specifically designed for vocal use on stage. It incorporates a pop screen, a transducer shock mount to reduce handling and impact noise, and is particularly rugged so it will survive the occasional drop from the stand.

Many vocal microphones have an upper midrange (3 to 8 kHz) peak to make the voice cut through. In the studio, vocals are ideally recorded from 30 cm (1 ft.) or even farther, usually with condenser microphones.

Wavelength
The distance between two consecutive peaks (or troughs) of a sine wave.



Checklists

Questions on designing a meeting system

We have compiled a number of questions that may be helpful for designing a sound system.
They should serve as points to remember in combination with the content of this Guide.

1.) Questions referring to the event:

> How many persons will typically attend an event?
> Do you want each participant to have his/her own microphone?
> How many microphones will you need in all?
> How many microphones should be for stationary/mobile use?
> How many audio sources should be fed in? (language, live music, feeds, telephone conference, video conference …)
> Will the seating be the same for all events?
> Will there be several concurrent events in adjacent rooms?
> How many languages will be spoken?
> Will you need simultaneous interpretation; if so, for how many languages?
> Will a sound engineer be present at the event?
> Will there be a chairperson (conference manager)?
> Do you want to carry out votes?
> Can you describe a typical event procedure?

2.) Questions referring to the room and its acoustics:

> Which are active/passive room areas?
> Do you have an architect's accurate drawings of the room? (hardcopy, electronic)
> Do you know of any structural limitations, e.g., with regard to the installation of loudspeakers or infrared transmitters?
> Are there any requirements/limitations concerning the design and/or color selection of the components?
> Where will the loudspeakers be installed?
> What is their directional characteristic?
> Where should/may further loudspeakers be installed?
> Where will you need microphones?
> Do you have (a) permanently installed conference table(s), or can you design the room flexibly, for different applications?
> Do you have permanently assigned active and passive areas, or is the room used differently for different events?
> What has been done with regard to sound insulation?
> What are the sources of unwanted noise? (projectors, computers, air conditioning …)
> What is the expected level of interference due to the noise sources?
> Are there any acoustically critical areas? (reflections caused by hard walls, windows, smooth surfaces …)
> How long is the reverberation period?
> How is the reverberation period affected by the participants' presence?
> Are there any disturbing reflections in the room (flutter echo …)?
> What is the necessary acoustical gain (NAG)?

3.) Questions referring to microphone selection:

> Are microphones, loudspeakers and transmitters allowed to be visible, or will concealed installation be preferred?
> What microphone designs and polar patterns can be used?
> What is the preferred kind of installation (permanent, mobile)?
> Will there be a lectern?
> Will you need a wireless microphone system? (speakers moving about, or audience microphone)
> Will it be possible to use headsets or lavalier microphones?
> What mechanical features will be required?
> What visual features will be required?
> Which are the room areas where microphones will be used?
> Should every participant have his/her own microphone?
> Should the microphones feature controls? (on/off switch, voting button …)

4.) Questions referring to operation:

> Will the events be serviced by a technician or an automatic mixer?
> Which internal communication facilities will be required?
> How are the visual and acoustic conditions in the areas of technical service, control desk and interpreters' booths?
> Have any typical problems occurred so far, and which?

Room for notes:

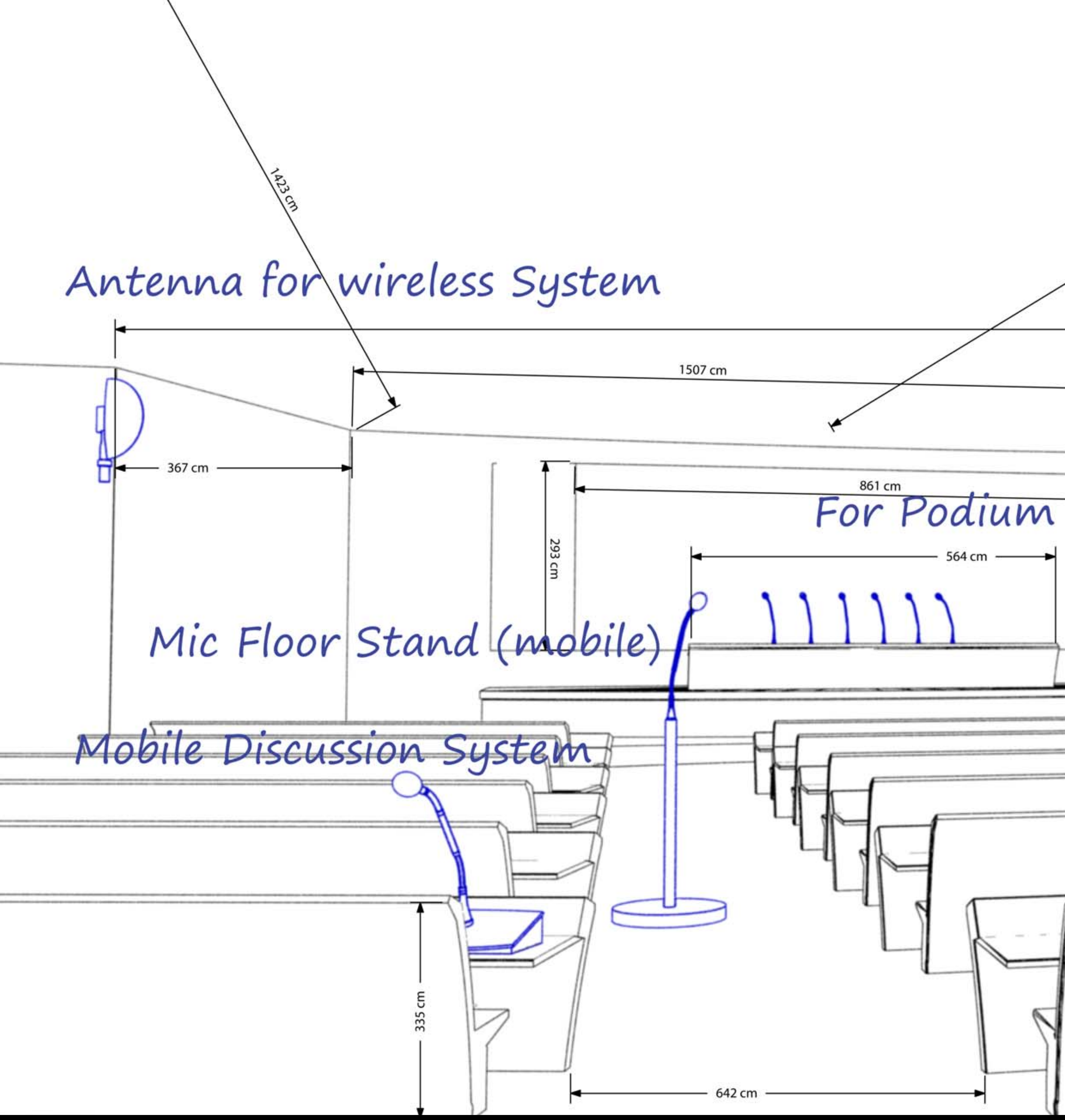
5.) Questions referring to the type of microphone system:

5.a) Questions referring to the wiring:
> Are microphone connections with phantom powering available?

5.b) Wireless microphones:
> Will you need audio signal encryption?
> How many wireless microphones will be required at the same time?
> Which wireless transmitters will be required (handheld devices, bodypack devices)?
> What microphones will be connected to the wireless system? (headset, lavalier, instrument miking …)
> Where can you install the antennae?
> Where is the control center with the radio receiver?

6.) Questions referring to type and function of the system control unit:

> What kind of control unit will be required? (automatic mixer, discussion or conference system)
> What functions should be available?
> How many microphone channels will be operated at the same time?
> How many interpreting channels will be required?



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