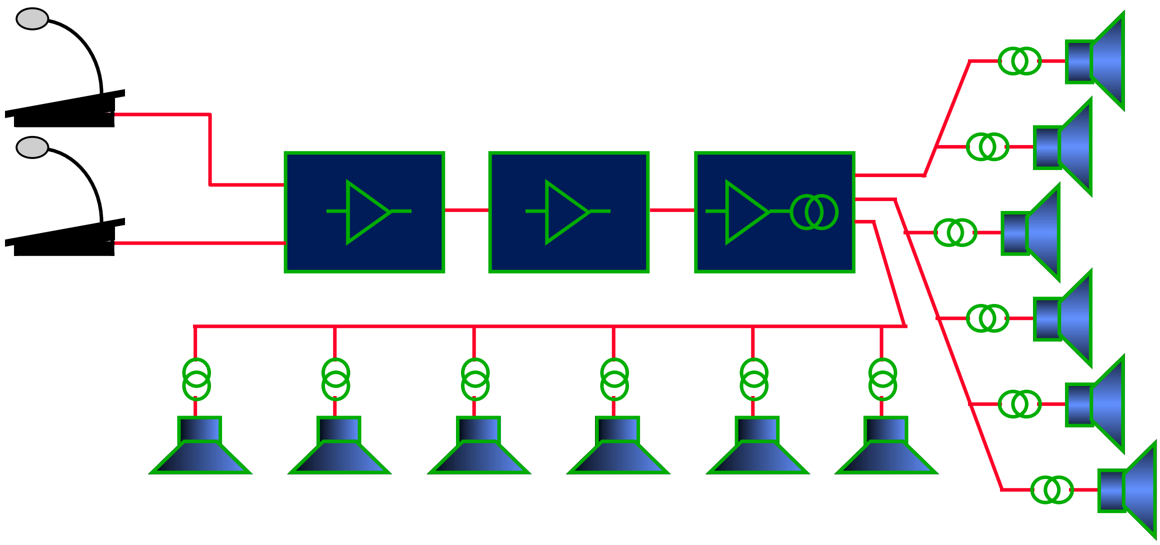
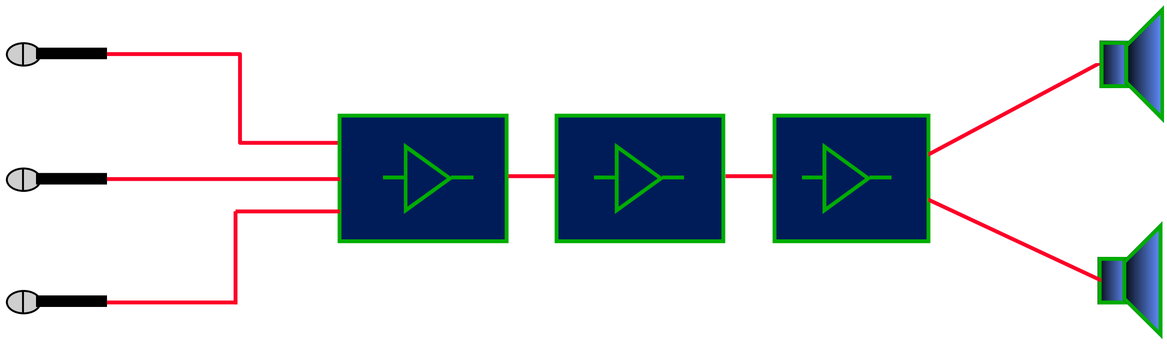
Sound System Reference Manual

Public Address



Sound Reinforcement



Philips Communication & Security Systems
Breda The Netherlands   Tel (+31) 76 5721 407

Frans van der Meulen

1958 Jubilee Edition 1998

Sound System Reference Manual

Introduction

Sound systems are used for amplification of speech or music to enhance intelligibility or loudness by electro-acoustic means in order to serve an audience with a higher degree of listening comfort.

A public address distribution system is designed primarily to carry live and recorded messages, signal tones and background music (if required), from several different sources, to a number of selectable remote areas. Common applications would be: hotels, restaurants, railway stations, airports, factories, oil platforms, office buildings, schools, shopping areas, ships, exhibition areas, etc.

A sound reinforcement system would normally be used to reproduce live voice (and often music) to a number of people who are generally located in the same room or area as the signal source. Typical applications are churches, lecture halls, political gatherings, conferences, etc.

Sound system design is a comprehensive subject combining a chain of devices: microphone, sound processing equipment, amplifier and loudspeaker, together with the acoustic environment, into a single system.

The **microphone** converts the acoustical vibrations, caused by an audio source, into an electrical signal. The **processing equipment** modifies the signal to compensate for deficiencies in the source or environment. The **amplifier** increases the level of the signal to one adequate for driving loudspeakers. The **loudspeakers** convert the electrical signal back into vibrations, which are greatly influenced by the **acoustic environment**, and in turn received by the ear of the **listener.**

This manual is intended to give readers with a technical background a reference to the various aspects of audio engineering and sound system design.

Frans van der Meulen - Public Address Consultant
Philips CSS - Kapittelweg 10 Tel (+31)76 5721 407
Breda - The Netherlands Fax (+31)76 5721 283
E-mail: Frans.van.der.Meulen@philips.com

TABLE OF CONTENTS

SOUND - THE THEORY

THE SOUND SYSTEM

MICROPHONES

AMPLIFICATION AND PROCESSING

HARDWARE INSTALLATION

LOUDSPEAKERS

THE ACOUSTIC ENVIRONMENT

A P P E N D I X

Definitions - Formulas - Symbols & Units - Materials

Applications - Simulations - Measuring instruments

Sound - The Theory

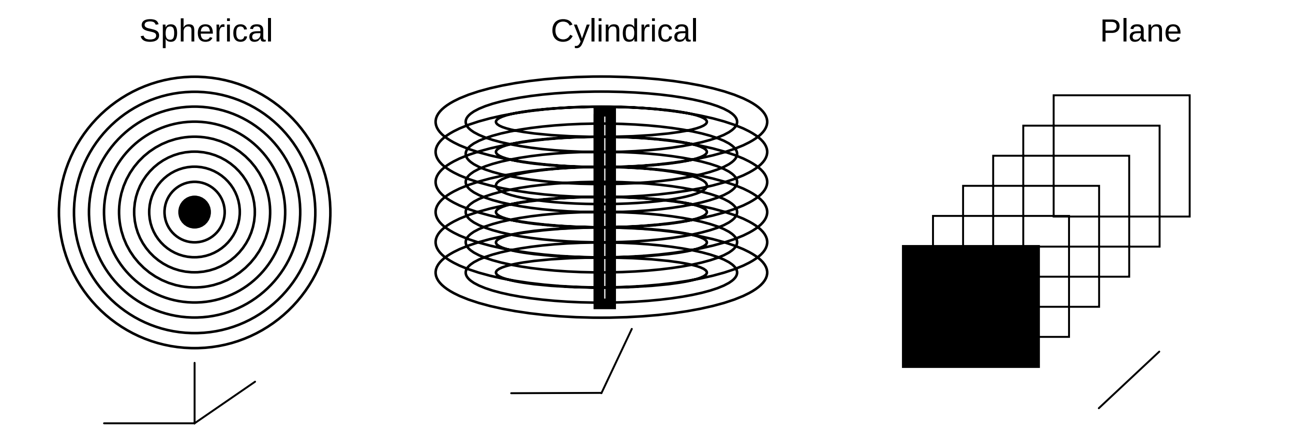
1.0 Basics

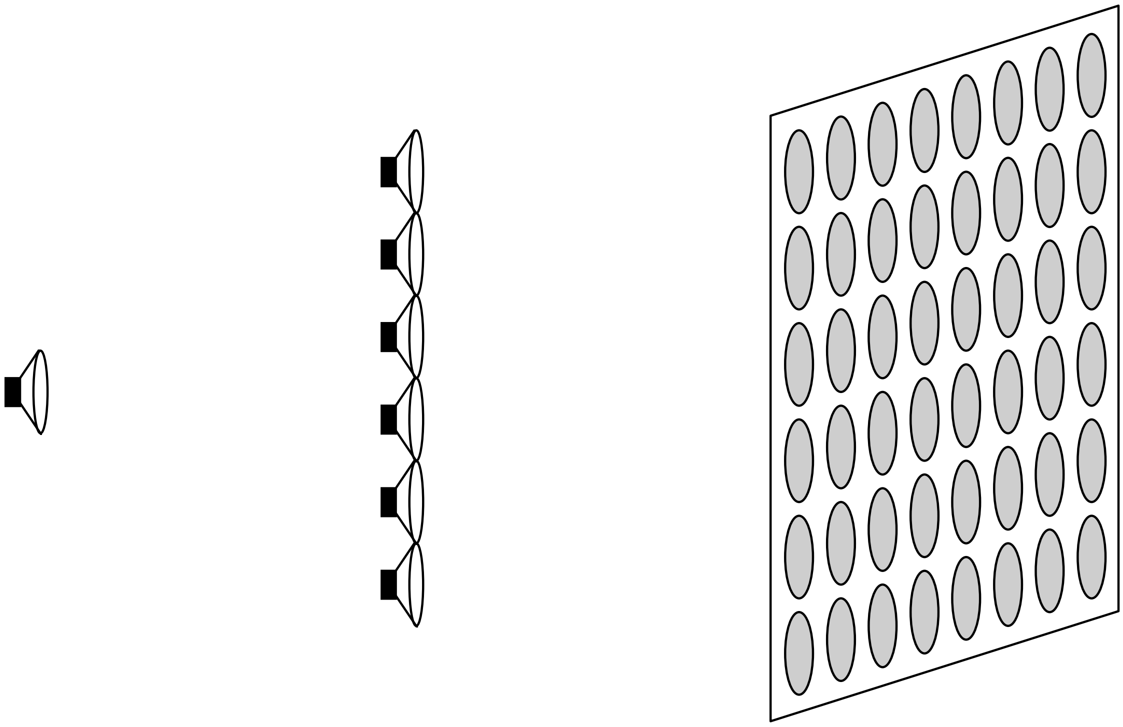
In order to design an efficient, effective and useful audio system, it is helpful to have a grasp of the way sound is received, processed, transmitted, and perceived by the listener. Sound waves are generated by air particles being set in motion by physical movement (such as the bow being drawn across a violin, the hammer hitting a piano wire or a vibrating cone of a loudspeaker, etc.). Once the particles start moving, they begin a chain reaction with other particles next to them. In this way, a movement of air is transmitted in all directions, by expanding and compressing air. So sound is energy that is transmitted by pressure waves in air.

We can make a distinction in the waveforms:

1. Spherical waves originated from an in all directions radiating point source.
2. Cylindrical waves originated from a line source.
3. Plane waves originated from a plane source.

Attenuation due to distance	
acc. square of distance	$= r^2$
acc. distance	$= r^1$
no attenuation	$= r^0$

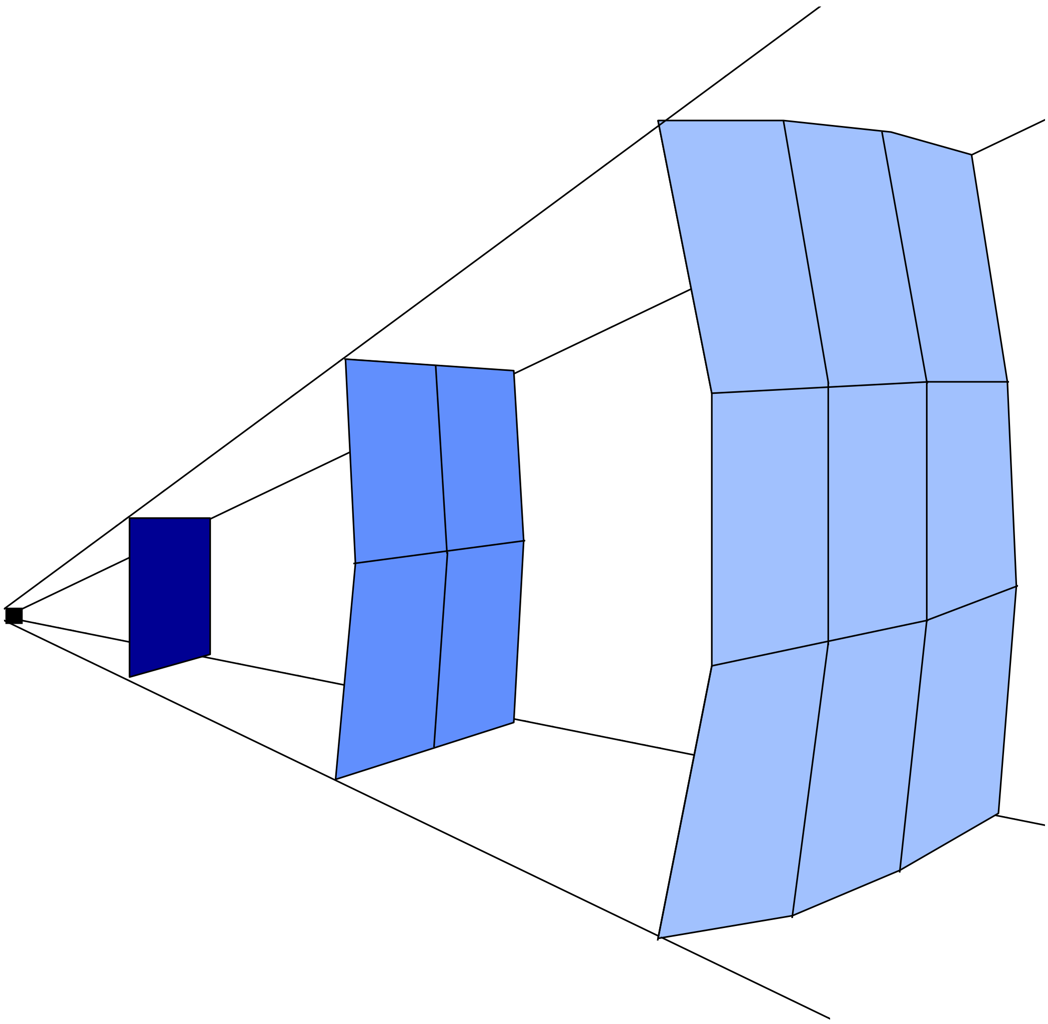




(Sphere)
Single Speaker

(Cylinder)
Array of speakers

(Plane)
Multiple speakers



On relative long distances from the sound source (r >> source dimensions >> wavelength), we apply generally the spherical wave attenuation r^2 but regard the waveform as a plane wave. The justification can be seen in above picture where the sound is moving in jumps of e.g. 1m. The total sound power (W) is in all cases the same but the sound pressure (N/m^2) or sound intensity (W/m^2) is decreasing with r^2.

Sound, as we refer to in this manual, generally consists of speech, music, alarm signals or attention tones.

Any kind of transmission or registration of sound, when converted into electrical signals, imposes a limitation on the dynamic range, frequency response, intelligibility and natural quality. The limitation of the dynamic range is the most prominent and the most important.

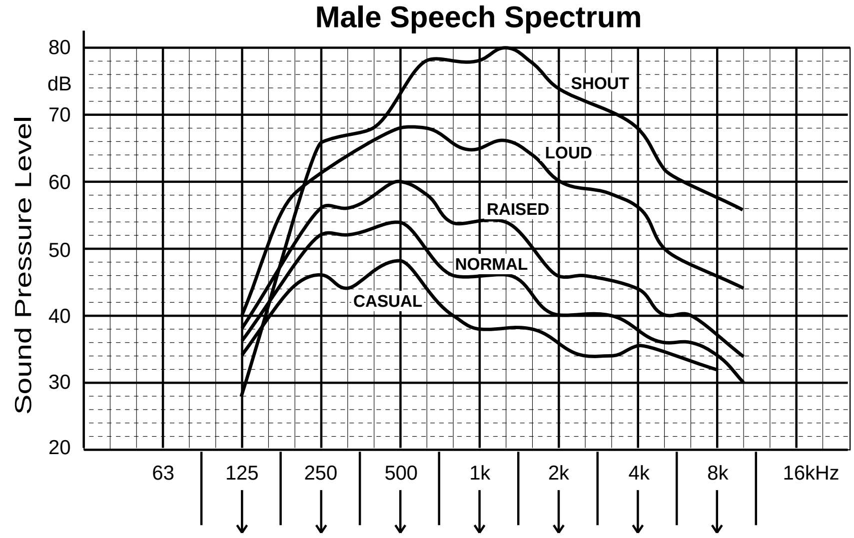
Dependent on our terms of reference dynamic range has different meanings:
1. In acoustics, the quietest sound level to the loudest one.
2. In music, the difference between pianissimo and fortissimo.
3. In sound engineering we prefer to express it as the difference between the maximum incidental peak value and the minimum value of the converted electrical signal.

With different measuring instruments (oscilloscope, level meter, VU-meter, etc.) we can analyse the signal, but dependent on the applied instrument's characteristics (integration time) different levels will be shown.

Sparks	=	= Integr. time 0 ms	= Oscilloscope with memory screen
Peak	=	= Integr. time < 5 ms	= Peak level meter (e.g. on mixing desks)
Fast	= Short Time Average	= Integr. time 125 ms	= Sound level meter (SLM)
VU	=	= Integr. time 270 ms	= VU-meter (e.g. on amplifiers)
Slow	=	= Integr. time 4 s	= Sound level meter (SLM)
LTA	= Long Time Average	= Integr. time 30 s	= Used for Amplifier cooling design

1.1 SPEECH

Speech consists of words and pauses. Words contain both vowels and consonants. Speech has loudness variation and frequency variation. Dependent on the voice strength the frequency spectrum (which is the lowest bass sound through to the highest treble one) is changing according the diagram. The lines in the graph represent the average level per 1/3 octave.



Contribution to Intelligibility						
	5%	13%	20%	31%	26%	5%
7%	22%	46%	20%	3%	2%	
Contribution to Speech Power						

The loudness of speech as it relates to frequency.

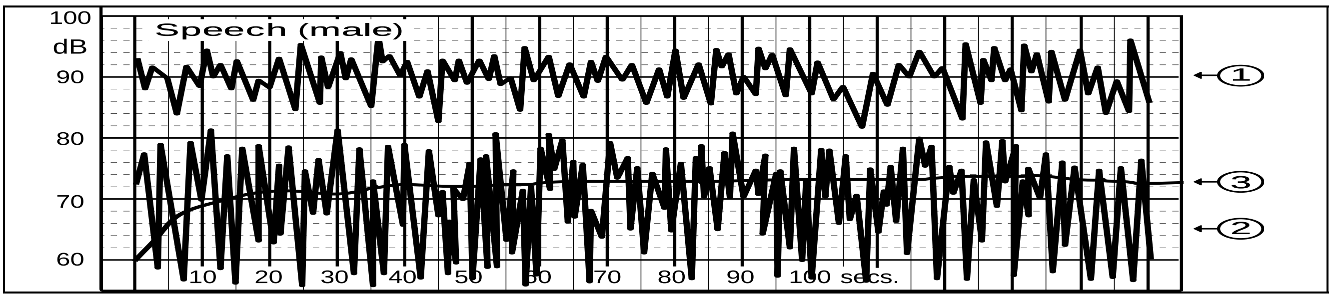
Let's look at loudness first. The vowels in a sentence have a frequency spectrum below 1000Hz, and they create the impression of loudness. The human mouth producing these sounds does so with a wide opening angle and in indoor situations can hit hard surfaces within range like walls and ceiling etc, so easily causing reverberation.

The consonants of the words in a sentence, having a frequency spectrum above 1000 Hz, provide the articulation. The human mouth produces these sounds with a narrow opening angle and, because of this, is rather directional.

Our principle aim is to deliver this complete speech spectrum to the listener's ears as unchanged as possible. Unfortunately various acoustical phenomena, which are discussed throughout this book, play their part in altering the speech spectrum, at times making it impossible for listeners to understand what is being said.
Because of this certain techniques are employed to compensate for these phenomena in order to make the speech intelligible.

1.1.1 Dynamic Range

The accompanying graphs show the speech pattern, versus time, of a trained announcer speaking at a fixed distance from the microphone, measured using different instruments.



Curve 1: peak value, rise time 1 msec. decay time 2.7 sec.
Curve 2: r.m.s. value, integration time 270 msec.
Curve 3: r.m.s. value, integration time 30 sec LTA.

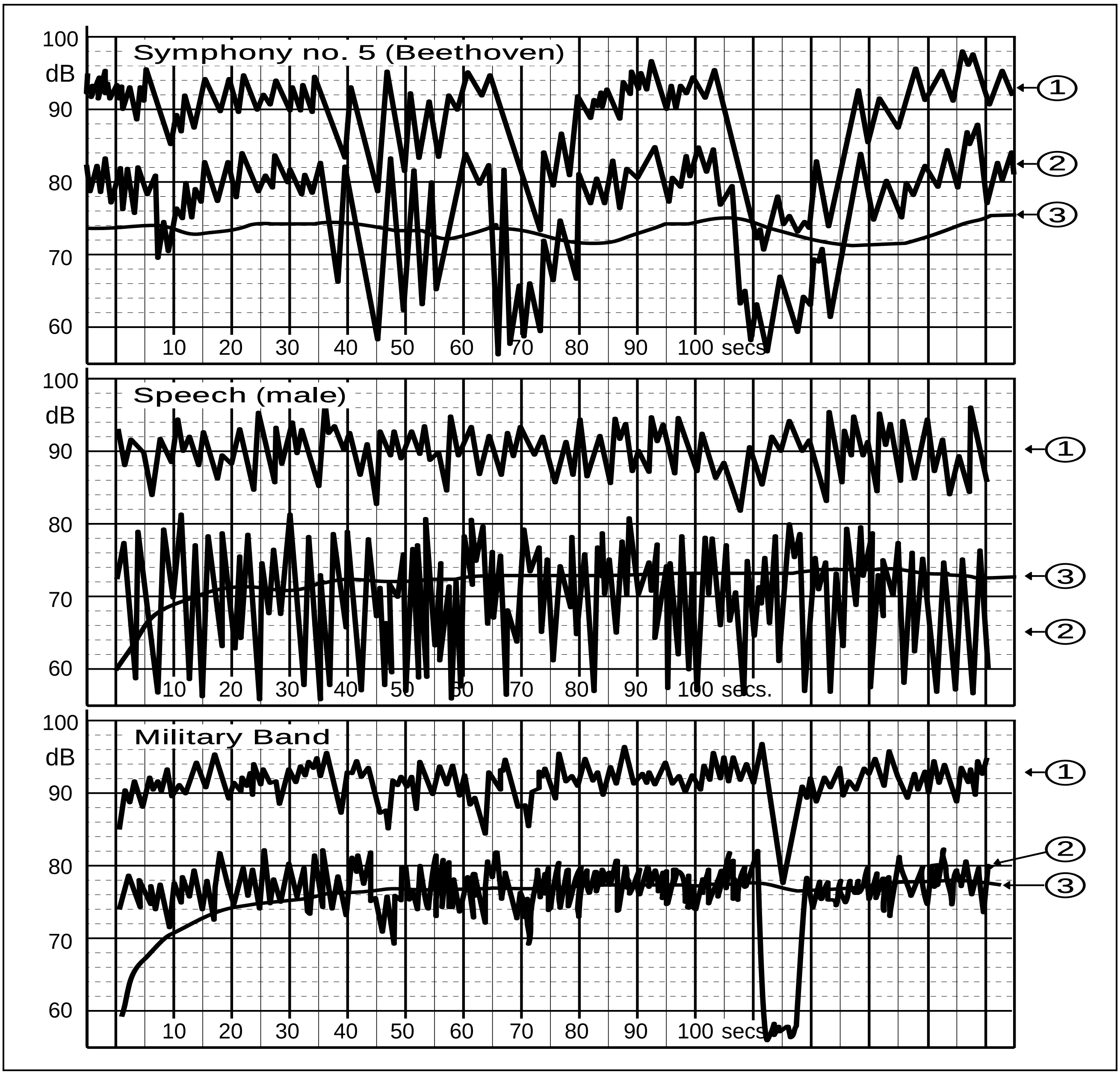
1.2 Music

As with any sound transmission, the two most prominent features of music reproduction are the dynamic range and the frequency response. If the dynamic range is limited, the music will appear emotionally flat, lacking both subtlety and excitement.

If the frequency response is limited at the lower frequencies, the music will lack the depth to reproduce bass instruments fully. If it is limited at the higher frequencies, harmonics, which are vital for instrument recognition, will not be fully present, causing the music to sound dull.

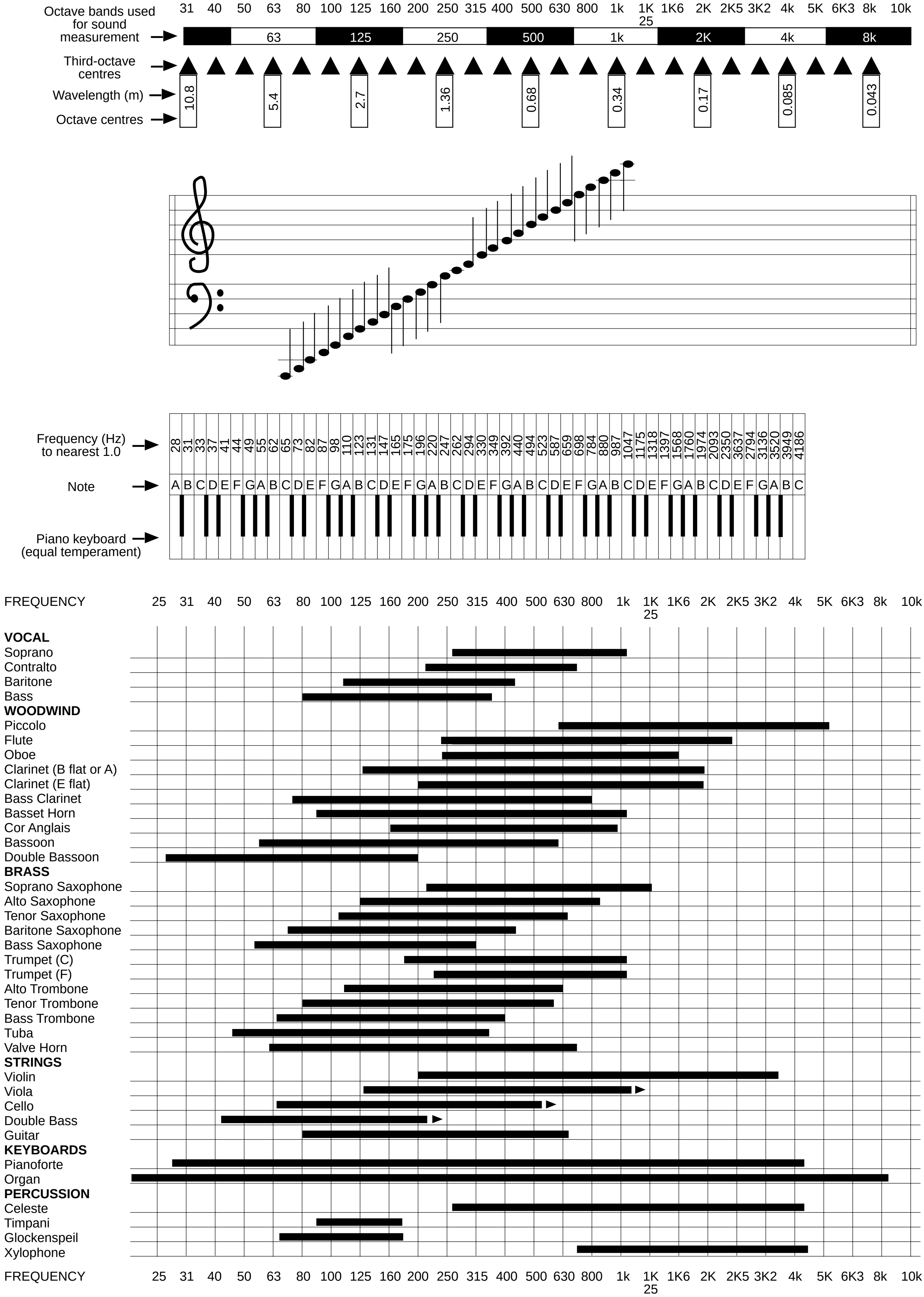
1.2.1 Dynamic Range

The accompanying graphs show comparative dynamic ranges of different styles of music and speech, measured simultaneously, showing: 1. Peak meter level. 2. VU meter level. 3. LTA level.
It can be seen that there is an average of 14 dB difference between the peak and VU level. Distortion during very short peaks is almost inaudible, so in practice 6 dB peak clipping is permissible. Therefore 0 VU or100% on a VU meter should correspond with a headroom of 8 dB under the distortion limit of the equipment.



Curve 1: peak value, rise time 1 ms, decay time 2.7 s.
Curve 2: r.m.s. value, integration time 270 ms.
Curve 3: r.m.s. value, integration time 30 s LTA.

1.2.2 Musical Range versus Frequency

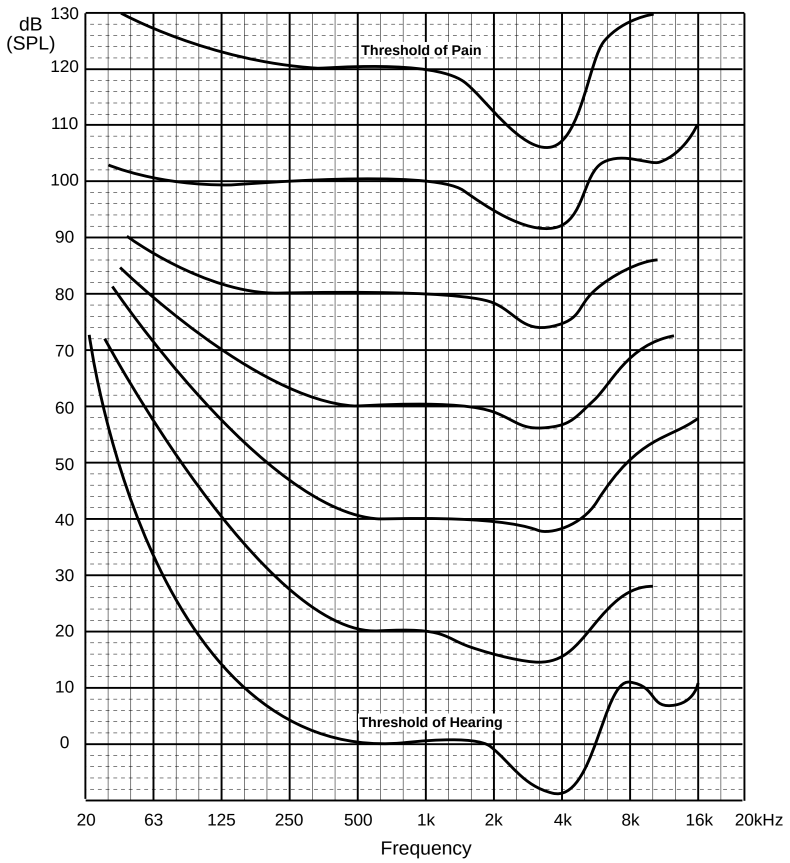


1.3 SOUND

Sound is a series of vibrations compressing and rarefying the air. Loudness is the subjective experience of sound level. Since, when we measure sound, we refer to changes in air pressure, a reference related to pressure must be used.
The reference used is the level of sound at 1 kHz, which is barely perceptible to people with normal hearing, being the quietest sound pressure that an average person can hear. This is called the 'threshold of hearing'. which at 1 kHz is: 20 μN/m^2 = 20 μPa = 2 x 10^{-5} Pa. (Pa = Pascal = N/m^2). Sound Pressure related to this reference level is expressed in dB (SPL).
As the sound pressure level is increased, a point is finally reached, just short of being painful to the ear, called the 'threshold of pain', which, using the 1 kHz reference frequency, corresponds to 20 Pa.
Since the 0 dB (SPL) absolute reference is 20 μPa;

$$20\ \text{Pa} \cong 20 \log \frac{20}{2 \times 10^{-5}} \cong 120\ \text{dB (SPL)}$$

1.3.1 Ear-characteristics



Equal loudness contours

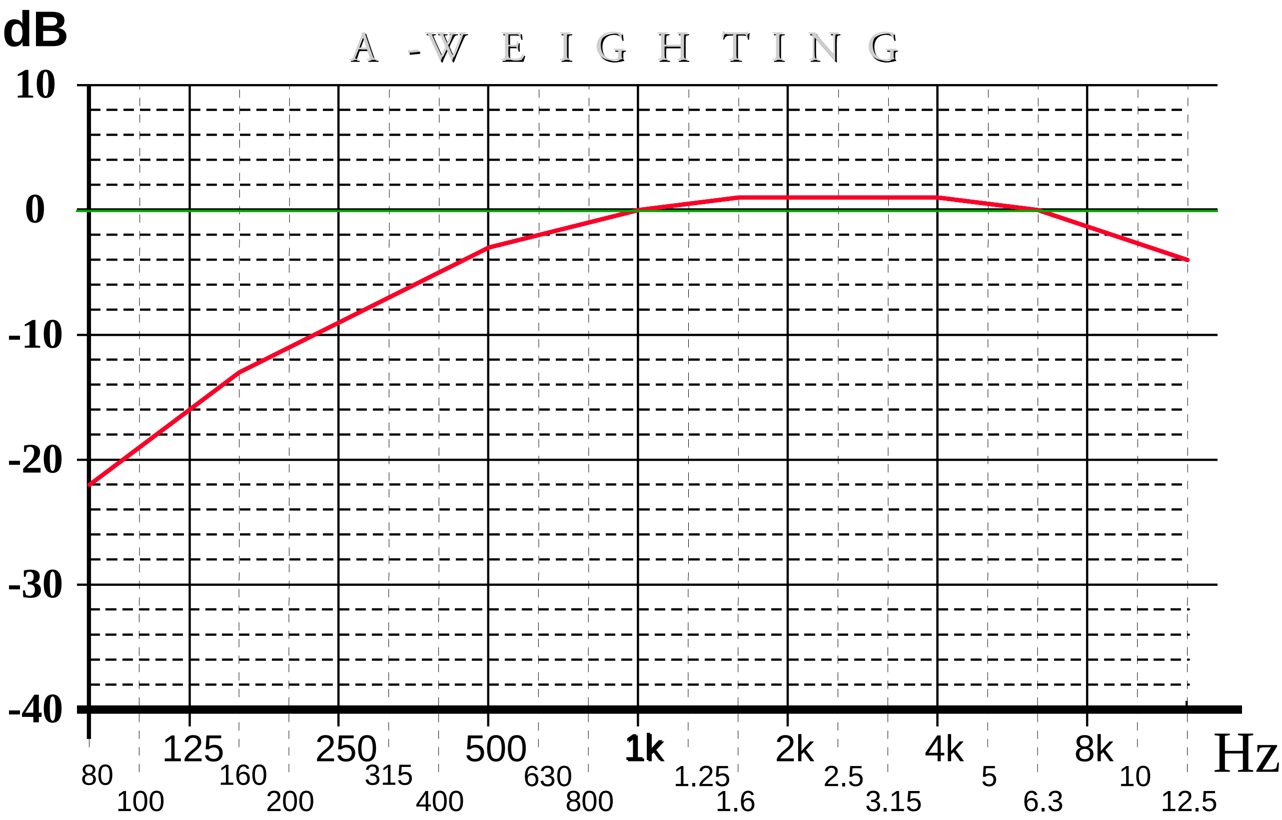
As is shown in the accompanying graph, the Sound Pressure Level at the threshold of hearing varies with frequency. Because of this it would require 60 dB (SPL) at 30 Hz to produce the same impression of loudness as 0 dB (SPL) at 1 kHz. The threshold of hearing represents the bottom limit of a series of 'equal loudness' contours, which are also shown. In studying the graph, we notice two important factors.

Firstly, very much more energy is needed to produce a bass signal of a given loudness, when compared with a 2 or 3 kHz signal - an important consideration in any system used to reproduce music.

The second point is that if noise, having a broad frequency spectrum, with a level of, say, 20 dB (SPL) is reproduced, the listener will have an impression which corresponds with the mirror image of the 20 dB equal loudness contour.
If the noise level is now raised a different impression of the same noise is received. This is due to the ear responding to the noise according to the changing curves of equal loudness. In other words, all frequencies are present in the signal but depending on the level, they will be heard in different relationships.

1.3.2 Weighting

In order to imitate this characteristic of the ear, a sound level meter often incorporates different filter curves which corresponds with this subjective hearing. There are three types of curves internationally standardised and they are called A-, B-, and C- weighting.

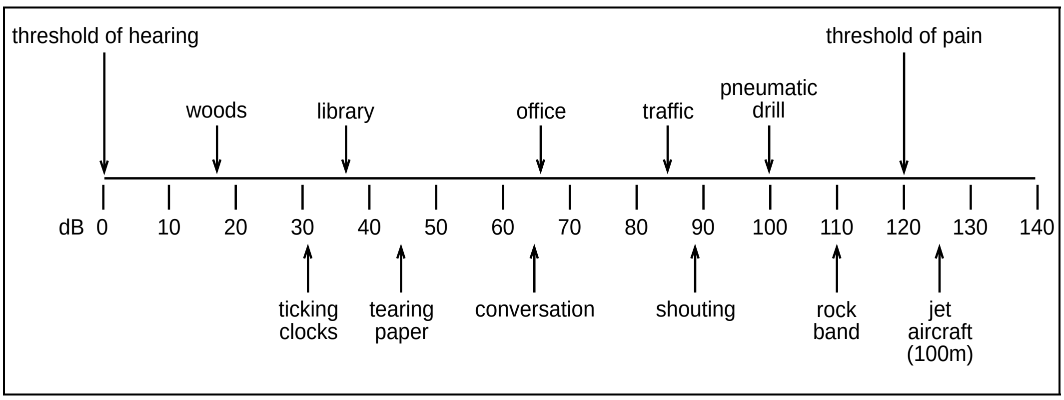


A-curve

This weighting should theoretically be used only for measurements below 40 dB (SPL).
Many simple sound level meters though are equipped with an A-curve filter only, and nowadays the majority of acoustic measurements are taken solely with A-weighting.
This is designated dBA (SPL).

1.3.3 Sound Pressure Level

The accompanying chart shows the sound pressures in dB(SPL), for several common sound sources.



1.4 SOUND PROPAGATION IN AIR

Sound could most simply be defined as a series of vibrations compressing and thinning the air. To be transmitted, sound relies on a vibrating object (vocal chords, loudspeaker, breaking window, etc.) which imparts its motion to surrounding molecules or particles.
Important physical parameters, which influence the propagation of sound in air are: f = frequency (Hz), v = velocity (m/s), λ = wavelength (m), p = pressure (Pa), T = temperature (K).

Velocity of sound

The velocity of sound is determined mainly by the temperature.
For normal conditions, in air, the velocity may be calculated by:

$$v = 20\sqrt{T}$$ where T is the temperature in Kelvin (0°C = 273 K)

This means that at 20°C

$$v = 20\sqrt{293} = 342{,}3 \cong 340 \text{ m/s}$$

The relationship between frequency, wavelength and velocity is given by: $\lambda = v/f$

Using these equations, it is seen that at 1 kHz at 20°C, the wavelength is

$$\lambda = \frac{340}{1000} = 0{,}340\text{m}$$

Air Absorption

Listening to sound on distance makes us aware of a frequency dependant attenuation due to air-absorption, the higher the frequency the more attenuation. This attenuation for a frequency of 500 Hz equals 0.3 dB per 100 metre, for 2000 Hz equals 1 dB per 100 metre and for 8000 Hz equals 7 dB per 100 metre (RH=70%).

Because the humidity effects the amount of water molecules in the air, also the attenuation of a sound signal is effected. This means that a relative humidity (RH) of 20% attenuates a 4 kHz signal by 0.09 dB/meter, whilst a relative humidity of 80% attenuates a 4 kHz signal by 0.02 dB/meter. The humidity level should certainly not be discounted since its effect can be quite dramatic.

Reverberation time

The effect of reverberation time (RT_{60}) in a room with volume (V) and surface (S):

$$\mathbf{RT_{60} = 0{,}161\ V/(\alpha S + 4mV)}$$

with: α = average absorption coefficient   m = attenuation constant (m^{-1})

Example: Room : 100x100x10 m $\alpha = 0{,}1$ V = 100.000 m^3 S = 24000 m^2
Relative Humidity = 60%   Temperature 20 ^{0}C

Freq. (Hz)	**m [1-2] (10^{-3} m^{-1})**	**4mV (m^2)**	**RT_{60} (s)**	**m [3] RH=60%**	**m[3] RH=20%**	**RT_{60} RH=20%**
125	0,12	48	6,62	0,07	0,10	6,60
250	0,28	112	6,41	0,15	0,23	6,46
500	0,51	204	6,18	0,37	0,56	6,14
1000	0,78	312	5,94	0,91	1,39	5,45
2000	1,49	596	5,37	2,25	4,28	3,92
4000	4,34	1736	3.89	5,6	14,5	1,96
8000	16	6400	1,83	16,2	47,1	0,76

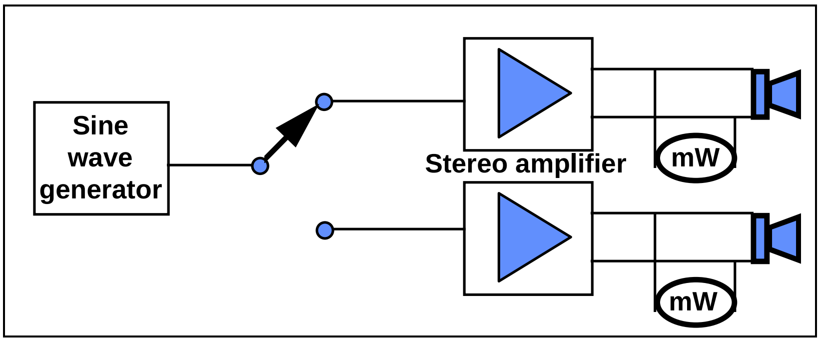
References: [1] Room Acoustics (1991) , Kuttruff. [2] Handbook of Chemistry and Physics (1973)
[3] Absorption of Sound in Air versus RH and T (1967), Cyril Harris.

2.0 Decibel Notation

2.1 DEFINITION

The use of the decibel (dB) notation system is common in sound and communications work. This system allows meaningful scale compression or expansion as required and greatly simplifies computations involving large quantities.
Our human senses - touch, sight, hearing, sense of weight, etc. - all function logarithmically. That is, in the presence of a stimulus the least perceptible change is proportional to the already existing stimulus (Weber-Fechner law).

2.1.1 Logarithmic characteristics of the ear



To evaluate the ear's behaviour in respect to sensitivity for level differences, we can experiment as follows:
The diagram shows two identical amplifiers and loudspeakers with a signal generator switched to one then to the other alternately. Initially the same power is supplied to each loudspeaker, e.g. 100 mW, and because of this both their signals are of equal loudness.

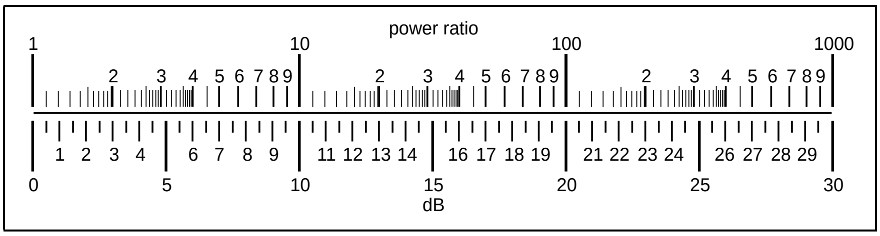
As the power to one of the loudspeakers is slightly increased no difference in loudness will be heard, whilst continuing to listen to them one at a time. Only when one loudspeaker receives 26% more power it will sound noticeable louder. At this point e.g. 126 mW is being fed to one loudspeaker and 100 mW to the other.

If the power of the other loudspeaker is also increased to 126 mW, the intensities will again be equal. If the power to the first loudspeaker is once again increased, no noticeable difference will be heard until it receives 26% more power (26% of 126 mW = 32 mW), which brings the higher loudspeaker output to 126 + 32 = 158 mW. In this way, the noticeable increase in loudness is obtained by raising the level in a given ratio, not by adding specific amounts of power. Increasing power in ten stages of 26% brings it to ten times its original level. This is a logarithm increase not a linear increase.

A power increase of a factor 10 is one Bel, with each power increase of 26% being one tenth of a Bel and called a decibel (dB). It must be appreciated that the dB is only a ratio, and that the ear hears the same difference between 1 W and 2 W as between 100 W and 200 W.

2.1.2 Power ratios

The Bel is defined as: Log P_1/P_2 , so the decibel (dB) is defined as: 10 Log P_1/P_2.



The amplification power ratio, expressed in dB, is given in the accompanying table. This shows, for instance, that 3 dB amplification doubles the power, and that a 100 times increase in power gives 20 dB amplification.

2.1.3 Voltage ratios

When 10V is connected to a 10 Ω resistor:

$I = U/R = 1A$

Power dissipated (P) = U x I = 10 W.

When the voltage is doubled and still connected to the 10 Ω resistor:

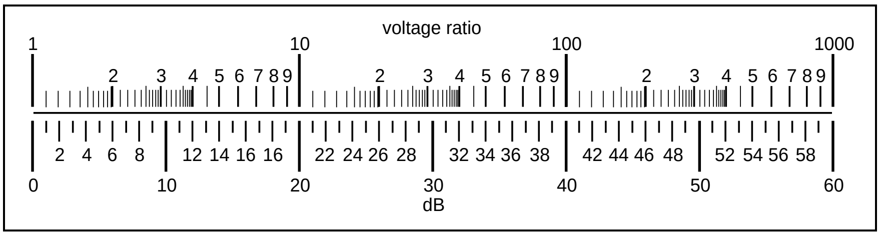
$I = U / R = 2A$

P = U x I = 40W, i.e. 4 times increase in power.

This shows that, in this case, doubling the voltage results in a quadrupled power; or to put it another way, a doubling of the power (3 dB increase) will *not* result in a doubling of voltage.
Since power is dissipated in the same resistor:

$$\text{Ratio (in dB)} = 10 \text{ Log } P_1/P_2 \quad = 10 \text{ Log } \frac{U_1^2/R}{U_2^2/R} \quad = 10 \text{ Log } \frac{U_1^2}{U_2^2} \quad = 20 \text{ Log } U_1/U_2$$

Because 10 Log power ratio = 20 Log voltage ratio, a gain of 3 dB gives a 2 x power gain, but only a 1.4 x voltage gain. In the same way, a 6 dB gain results in a 4 x power gain but only a 2 x voltage gain.



From this it can be seen that an amplifier (or attenuator) having a particular gain expressed in dB has a different multiplicative effect dependent on whether power gain or voltage gain is being considered.

2.1.4 dB references

Though the decibel is only a ratio, it can be used to express absolute values if there is a given reference.
If, for example, a reference of 1 W is chosen, then 3 dB corresponds to 2 W, and 6 dB to 4 W and so on.

dBm - dBu

One of the common references used in the past, due to its frequent application in Telecommunications is 1 milliwatt (mW) across 600 Ohms, expressed as "dBm". (1 mW across 600 ohms = 775 mV.)
In practice however the resistance value is frequently ignored when dBm is quoted and the reference is 775 mV only, this makes this reference incorrect. In fact the dBu is referred to the 775 mV regardless of the impedance and is still commonly in use in studio engineering.

dBV

This is the favourite and common reference for electrical engineering. The reference is 1 Volt regardless of the impedance. Corresponding dB values are measured in dBV (e.g. 20 dBV = 10V).

dB(SPL)

There is another reference, which is used in the measurement of sound pressure levels. As we know sound is basically a series of vibrations compressing and rarefying the air. Since, when we measure sound, we refer to changes in air pressure, a reference related to pressure (the Sound Pressure Level) must be used.
The reference used is the level of sound , which is barely perceptible to people with normal hearing, being the quietest sound pressure that an average person can hear at 1 kHz. This is called the 'threshold of hearing'.
At this point the threshold of hearing is very low: $20\ \mu N/m^2 = 20\ \mu Pa = 2 \times 10^{-5}$ Pa. (Pa = Pascal = N/m^2) So Sound Pressure related to this reference level is expressed in dB (SPL).
As the sound pressure level is increased, a point is finally reached, just short of being painful to the ear, called the 'threshold of pain', which, using the 1 kHz reference frequency, corresponds to 20 Pa.
Since the 0 dB (SPL) absolute reference is 20 µPa;

$$20 \text{ Pa} \cong 20 \text{ Log } \frac{20}{2 \times 10^{-5}} \cong 120 \text{ dB (SPL)}$$

Other important levels are:

0,1 Pa ≅ 74 dB (SPL), and 1 Pa ≅ 94 dB (SPL)

2.2 CALCULATIONS

2.2.1 Addition and subtraction

When **adding** two unrelated sound sources, only their intensities (energy) should be added together.

$$L_S = 10 \text{ Log } [10^{L1/10} + 10^{L2/10}]$$

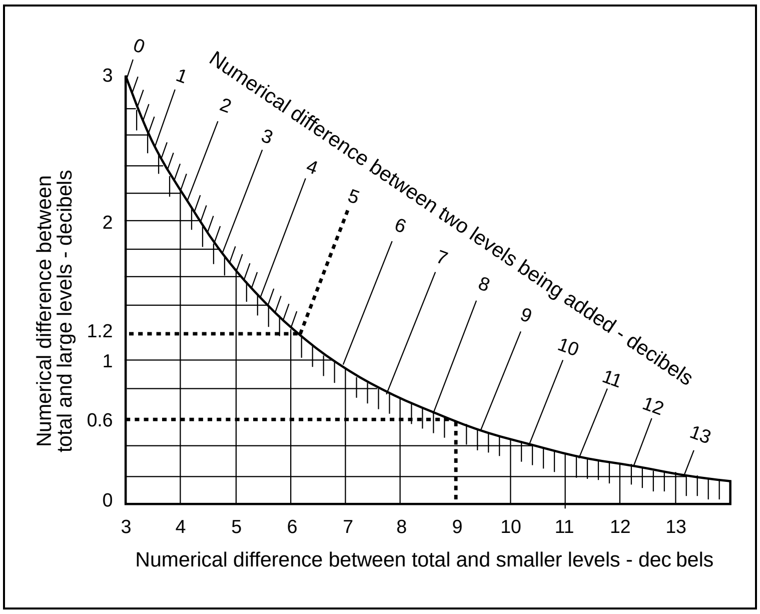
Two different noise sources both producing 90 dB (SPL) would be experienced as:

$$L_S = 10 \text{ Log } [10^9 + 10^9] = 93 \text{ dB (SPL)}$$

When **subtracting** two unrelated sound sources, only their intensities (energy) should be subtracted:

$$L_S = 10 \text{ Log } [10^{L1/10} - 10^{L2/10}]$$

The following graph shows how to add or subtract levels in dB's for **non-related** signals.



To add levels of non-related signals.
Enter the chart using the numerical difference between the two signal levels being added (top right of chart). Follow the line corresponding to this value until it meets the curved line, then move left. The figure shown on the vertical scale at the left of the chart is the numerical difference between the total and larger of the two signal levels. Add this value to the larger signal level to determine the total.
Example: Combine a 75 dB signal with one of 80 dB. The difference between these figures is 5 dB. The 5 dB line intersects the curved line at 1.2 dB on the vertical scale. This means that the total value is 80 + 1.2, or 81.2 dB.

To subtract levels of non-related signals.
If the numerical difference between the total and the smaller of the two levels is between 3 and 14 dB, enter the chart from the bottom. Using the numerical difference, follow the line corresponding to this value until it intersects the curved line, then follow the line to the left. The figure shown on the vertical scale at the left of the chart is the numerical difference between the total and the unknown (the larger) level. Subtract this value from the total to determine the unknown level.

Example: Subtract 81 dB from the 90 dB total. The difference is 9 dB. The 9 dB vertical line intersects the curved line at 0.6. Deducted from 90 dB total, this leaves 89.4 dB.
If the numerical difference between the total and the larger of the two signal levels is less than 3 dB, enter the chart from the left side. Then, at the intersection with the curved line, follow the line down to find the numerical difference between the total and the smaller level.

The Sound System

3.0 An Introduction

When assessing the requirements of any sound system it is important to have a firm grasp of what tasks the system will need to perform. Along with this, the acoustic environment will determine, to a great degree, what equipment should be specified. It is vital therefore to clearly understand the characteristics of the equipment available to meet these various needs.

This section contains a description of the basic components of the sound system, along with some technical specifications and, at times, advice on the techniques involved in installing and using the equipment.

In certain applications, for example a small church needing only speech amplification, we can reduce the equipment needed to a few microphones, one mixing amplifier and a few loudspeaker columns. The individual microphone volume levels would be controlled on the amplifier, which also allows tone-control of the loudspeakers. Once carefully set up, such a system should work without intervention, every time the amplifier is switched on. Other situations, for example an oil platform, require both sophisticated routing and switching systems, and a complete fail-safe redundancy backup system. Obviously, even though the sound quality should always be adequate, the complexity of calculating the type and quantity of equipment required depends upon the installation's requirements.

3.1 FUNCTIONAL REQUIREMENTS

Before starting to design a sound system it is vital to answer the following questions:

- Is the system required for speech alone, speech & music or music alone?
- Is the system required for announcements and/or for emergency purposes?
- How many calls must be made, at the same time, to different destinations?
- How many different music sources must be routed?
- What are the maximum and minimum ambient noise levels?
- What is the requirement in respect to loudness?
- What is the requirement in respect to speech intelligibility?
- What is the requirement in respect to annoyance due to excessive loudness?
- What is the requirement in respect to frequency response?
- What is the requirement in respect to sound orientation?

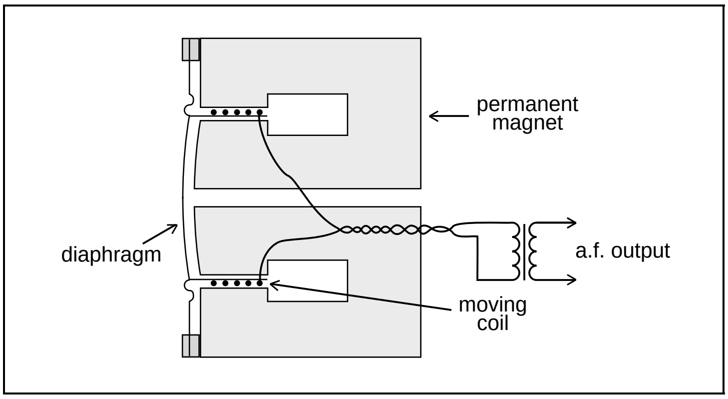
4.0 Microphones

4.1 Considerations when Selecting a Microphone

In any sound amplification chain, the first link is often the microphone, which converts acoustic vibrations into voltage variations. Three types of element are generally encountered in microphones used in a professional audio installation, Electrodynamic, Condenser, and Electret. The way an element is mounted in the microphone body determines the microphone's pick-up response pattern.

4.2 Microphone Types

4.2.1 Electrodynamic



The Dynamic microphone is based on the principle of a coil moving in a magnetic field.

Sound pressure causes the diaphragm to respond in rhythm with sound vibrations, so that the coil moves inside the air gap of a permanent magnetic field. This, in turn, induces a voltage in the coil. The pitch and intensity of the original vibrations determine the frequency and amplitude of this voltage. This means that the higher the frequency - the faster the coil moves, the louder the sound - the further the coil moves.

4.2.2 Condenser

The basic elements of the Condenser microphone are a thin metal flexible diaphragm, which forms one plate of a capacitor, whilst a solid metal plate forms the other.
The capacitance depends on the distance between the diaphragm and the plate. As the diaphragm moves, the distance between the diaphragm and the plate varies, which causes the capacitance to change accordingly. A steady D.C. polarising charge is maintained across the diaphragm and the plate. As the sound varies, this causes the capacitance to vary, which in turn causes the voltage to vary, causing the subsequent current flow to vary. A DC voltage, supplied by the mixing console or pre-amplifier unit, is carried on the microphone's standard two core screened signal cable, and is called Phantom Powering. This provides the polarising charge and also power for the microphone's FET amplifier.

4.2.3 Back Plate Electret

Though operating in a similar way to condenser microphones, the Philips Back Plate Electret (BPE) range of microphones feature a unique design. It is a combination of an uncharged, temperature independent, diaphragm and a **permanently** charged back plate electrode (which is achieved by sealing electret material onto a metal back plate).

4.2.4 Electret

Similar in operation to a condenser microphone, the diaphragm of the Electret microphone comprises a high polymer plastic film with a **permanent** electrostatic charge.

4.2.5 Choices

Because the microphone is such a fundamental part of the amplification chain, great care should be taken when making a choice. Normally a compromise must be made between reproduction quality and price, but it is wiser to economise on other equipment than on microphones.

Until recently **condenser microphones** have been used primarily in recording and broadcast studios, and rarely in public address systems. Having excellent reproductive qualities, condenser microphones tend to be comparatively expensive, in some cases fragile, and generally require a fairly powerful phantom power supply.

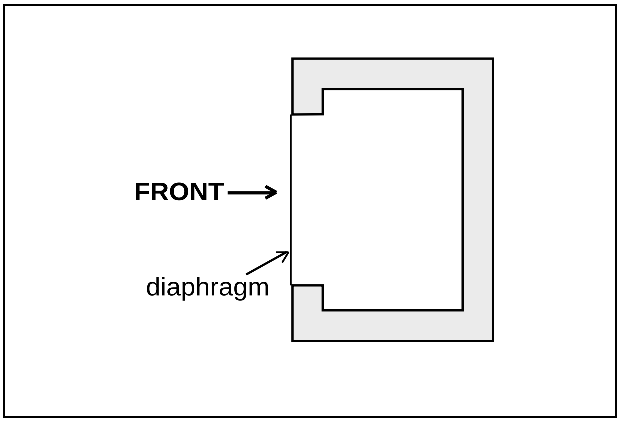
Like condenser microphones, **BPE microphones** require a supply voltage, but because they do not need a polarising charge, the current consumption is so low that up to four microphones can be powered by a single IEC268-15A (DIN4559-6) standard phantom powered input. BPE microphones have very good speech reproduction qualities, are rugged, and have low sensitivity to case noise, vibrations and hum fields.

The small FET amplifier contained within **Electret microphones** is often battery driven in consumer quality models, and phantom powered in professional models. The current drain is so small that battery life is usually several thousand hours. Though reproduction quality is lower than BPE microphones, the somewhat lower price makes them a viable alternative to dynamic microphones.

Until recently **Dynamic microphones** were the most popular for general use, requiring no phantom powering, being generally very rugged, and normally the least expensive. The lower sensitivity and, (in the case of less expensive models) low reproduction quality, mean that particular care should be taken when selecting dynamic microphones.

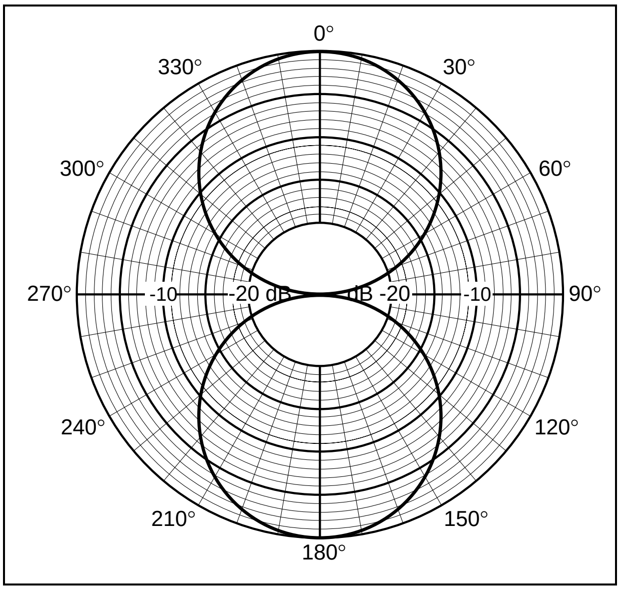
4.3 Pick-Up Response Patterns

The microphone shown in the accompanying illustration is sensitive to sound from any direction, responding to a voice from the front in just the same way as to the sound from the audience at the rear.



The force on the diaphragm is determined by the difference in pressure on its front and rear surfaces. Because the back of the element is totally sealed, the sound pressure variation leads directly to movement of the diaphragm, irrespective of which direction the microphone is facing.

Because it is responsive to sound from all directions it has what is called an "Omni-directional" response pattern.



Where the rear of the microphone is opened and the diaphragm is exposed to sound waves from the back as well as from the front, the polar plot is not omni-directional as before, but results in a **figure-of-eight** directional pattern.

Sound entering from the front will produce a frontal pressure, which is greater than, and out of phase with, the pressure due to sound entering the back. The difference will generate a maximum signal.

A sound source situated to the side however, puts the diaphragm under equal pressure from both sides and will tend to cancel itself out.

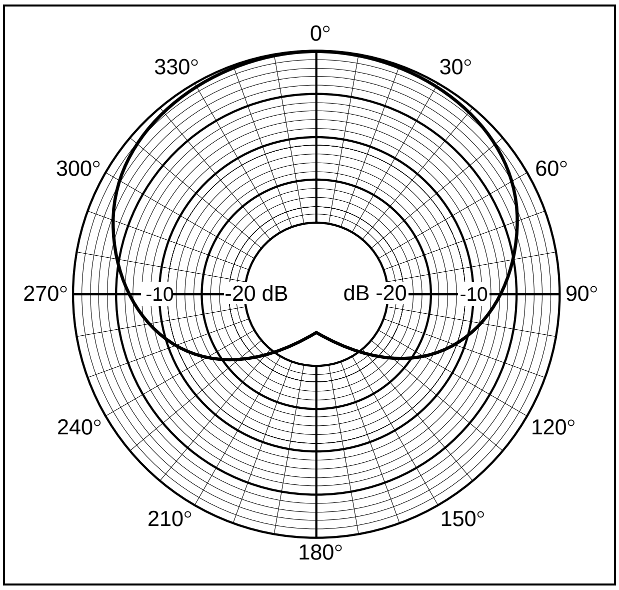
If the opening at the rear is adjusted in size and character by means of an acoustic filter, the polar response can be varied between the extremes of omni-directional and figure-of-eight. A response approximately halfway between these two is known as a Cardioid (heart shaped) response. The pattern known as a Hyper- Cardioid response is particularly sensitive to sounds which are generated at the front, and on axis with the microphone body. Other sounds, generated at the sides and back of the microphone are also picked up, but at a much reduced level.

4.3.1 Omnidirectional

Because of its construction, the Omni-directional microphone is sensitive to sound from any direction. It responds to a voice from the front in just the same way as to the sound from the audience at the rear. Because of their normally flat frequency response, irrespective of source distance, omni-directional microphones are often used for recording and measurement. They are used in situations where sound coming from several directions must be reproduced, and where either: a) the microphone is totally isolated from the loudspeakers, or b) the microphone is in close proximity to the sound source, so that the comparative level of any amplified signal it picks up is very small.

4.3.2 Cardioid

Unidirectional microphones with a **Cardioid** (heart shaped) directivity pattern are normally preferred in general public address distribution applications.

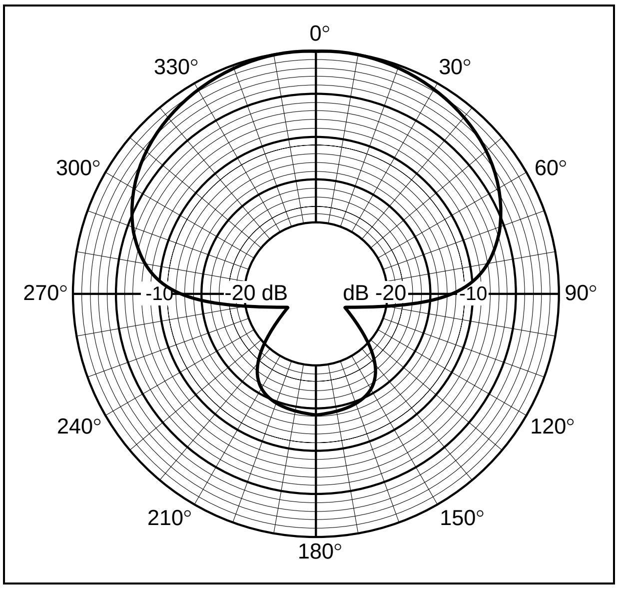


The **directivity factor** is the power ratio of the transformed frontal sound when compared to an omni-directional microphone with the same sensitivity for diffused sound. For cardioid microphones the directivity factor is max. 3 or the front to random sensitivity ratio 10 Log3 = 4.8 dB.

Careful tuning of the microphone ensures that whilst only a small amount of extraneous noise is picked up from the rear and sides of the microphone, the pick up pattern is wide enough to pick up sound from a fairly wide area at the front.

This allows a certain amount of freedom of movement for the speaker, without large drops in volume level.

4.3.3 Hyper-cardioid



The **hyper-cardioid** microphone operates in the same way as the cardioid microphone, but to a more extreme degree. For hyper-cardioid microphones the directivity factor is max. 4 or the front to random sensitivity ratio 10 Log4 = 6 dB.

Because of the high directivity of hyper-cardioid microphones, care should be taken in positioning to ensure that the operator is consistently speaking directly at front of the microphone.

Hyper-cardioid microphone characteristics present difficulties to the designers of Lavalier (Lapel) microphones, due to their sensitivity to local noise generated by contact with the user’s clothing.

Both hyper-cardioid and, to a lesser degree, cardioid microphones have a strongly increased sensitivity to low tones when the sound source is generated close to the microphone. This means that if an operator speaks very close to the microphone, their voice will become unnaturally bass in character, at times making the message unintelligible.

4.4 Special Microphones

A large number of special microphones are available, ranging from broadcast, through to the individual requirements of different musical instruments.
In the field of sound reinforcement and public address there are again several different types of microphone likely to be encountered for specialist applications.

4.4.1 The Lavalier and Lapel microphone

These microphones have been specially designed to reproduce speech, and are small, light, and designed to be worn (a) around the neck (Lavalier Microphone), or (b) clipped to a neck tie or jacket lapel (Lapel Microphone) without causing discomfort. With this in mind, they are particularly sensitive to high frequencies in order to compensate for the losses due to absorption by the user's clothing and made insensitive to the low toned noise caused when the microphone rubs against the clothing. The microphone capsules themselves are specially mounted in order to absorb shocks and therefore reduce noise being transmitted though the microphone due to movement on the speaker's clothes. Being omni-directional microphones, they are also suitable for use in such applications where a wide area needs to be monitored, such as in a conference recording system.

4.4.2 Noise cancelling microphone

This is essentially a hyper-cardioid microphone having an optimum speech characteristic, and is designed for extremely noisy environments such as touring buses, factories, and supermarket floors. This type of microphone must be held very close to the mouth, so filters have been built in to ensure that the frequency response is flat when the sound source is close to the microphone, and also that the bass content of the random noise is reduced.

4.4.3 Radio (Wireless) microphone

Great freedom of movement is provided for the microphone user by the use of a transmitter/receiver system. A FM signal provides a link between either a hand-held or lavalier/lapel microphone and a receiver connected to the sound system input. The hand held microphone has a built-in transmitter, while the lavalier model is connected to a small pocket transmitter, allowing full hands-free use.
When two or more radio(wireless) microphones are used in the same location, care should be taken to ensure that they each operate on a different transmission frequency, otherwise conflicts will occur.

5.0 Technical Principles

5.1 DIRECTIVITY

There is at times confusion between two terms of reference when microphones are being chosen for use in difficult acoustic environments where the risk of feedback must be reduced.

The response of a typical cardioid microphone at 500 Hz, as shown in 4.3.2, indicates that the response at the rear, on the 180° line, is some 23 dB less than that at the front. This is called the **front-to-rear** ratio. In 4.3.3 the response of a hyper-cardioid microphone is illustrated. Though the front-to-rear ratio is only 14 dB it is far more suitable for use in a very noisy environment. The reason is that the most ambient noise does not only come from the rear, but from the reverberant or diffuse field which is picked up at the sides of the microphone, and it is this field that the hyper-cardioid microphone, more than any other type, attenuates.

This is expressed in terms of what is called the **front-to-random** index where:

$$Fr = 20 \log S_f/S_d \text{ dB}$$

where S_f = free field sensitivity at 0° and S_d = average diffuse field sensitivity

The cardioid microphone typically has a front-to-random index of about 4,8 dB and the hypercardioid microphone has a front-to-random index of 5,8 dB.

5.2 SENSITIVITY

The sensitivity of a microphone is the output voltage for a given Sound Pressure Level at 1 kHz, in V/Pa.

Sensitivities vary considerably dependent on the type of design:

Studio Condenser	10 mV/Pa	(- 40 dB rel 1V/Pa)
BPE	3 mV /Pa	(- 50 dB rel 1V/Pa)
Electret	1,6 mV/Pa	(- 56 dB rel 1V/Pa)
Dynamic	1 to 2,5 mV/Pa	(- 60 dB to - 52 dB rel 1V/Pa)

5.3 INSTALLATION CONSIDERATIONS

5.3.1 Potential problems and causes

Problem	Cause
hum	Mains power cables
oscillation	100 V line output cables
crosstalk	other microphone cables

5.3.2 Solutions

The following steps help avoid these problems:

1. Use only two-core screened (shielded) cable for individual microphone signal cables and extensions.
2. Keep microphone cables away from mains power and loudspeaker cables. If it is necessary for the cables to cross, try to ensure that they cross at 90°, rather than running along side each other.
3. In installations with long microphone cables, use a cable transformer or line amplifier.

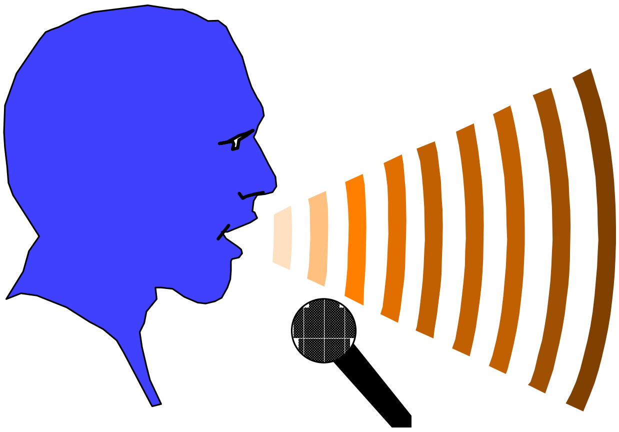
Also: Never position a microphone in the direct field of a loudspeaker, as this could cause acoustic feedback (howl around), described in chapter 9.2.

6.0 Microphone Technique

Microphones in the Philips product range, are of advanced design, are very sensitive, and reproduce the human voice with great clarity. Many of these microphones have a hypercardioid response pattern, being particularly sensitive to sounds, which are generated at the front, and on axis with the microphone body. Other sounds, generated at the sides and back of the microphone are also picked up, but at a much reduced level. This characteristic gives them a high front to random response index. Due to the fact that they are so directional, hyper-cadioid microphones operate particularly well in difficult acoustic environments and in areas with high background noise.

In order to optimise these, or any microphone, it is important to be aware of certain operating techniques.

1. The microphone should be pointing directly at, but placed a little below, the speaker's mouth. This is to pick-up full spectrum sound including high frequencies and avoiding air blowing frontal on the microphones diaphragm and causing “plops”.
2. The best distance from which to speak into a microphone is approximately 15 to 40 centimetres. If that distance is reduced greatly, a phenomenon, especially common to (hyper)cardioid microphones, known as 'proximity effect' will occur. This is a very noticeable increase in the bass content of the signal, making the voice muffled, and at times unintelligible.



3. Speak at a consistent volume level.

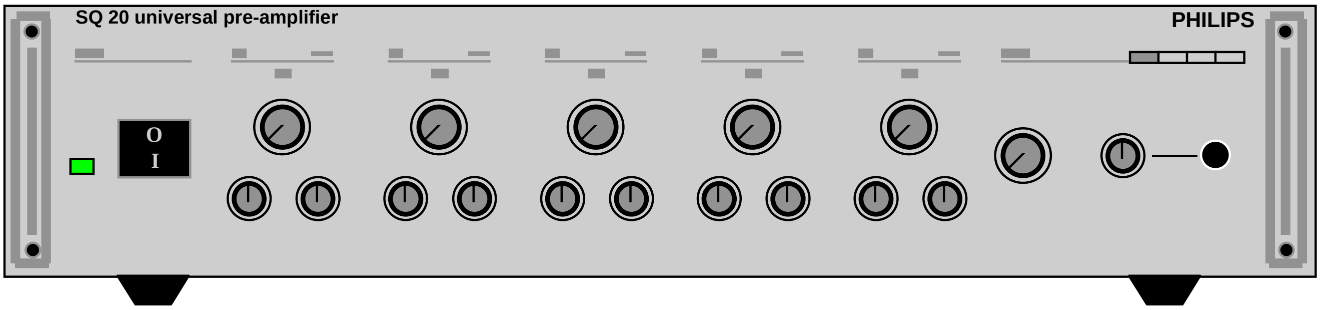
4. If the operator were to speak from a much greater distance than that recommended, the microphone would also pick up other sounds in the room, effecting the overall clarity. This is particularly unfortunate when the microphone is in the same room as the loudspeakers, due to the fact that the amplified signal could be picked up by the microphone and amplified again. If the amplification in this loop is allowed to continue, the disturbing phenomenon known as acoustic feedback, or 'howl around', will occur.

5. If feedback does occur, do not cover the microphone with your hand; this makes the situation worse. If you are very close to the microphone, moving backwards sometimes helps eliminate feedback. The operator should then reduce the amplifier volume slightly, or use a tone control or equaliser to attenuate the offending frequency somewhat.

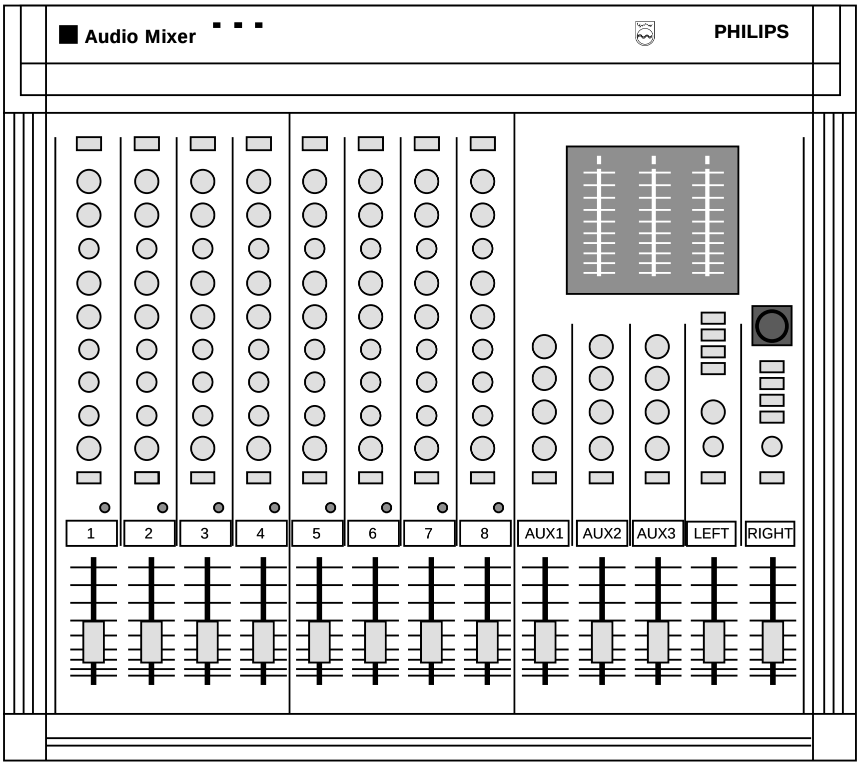
Amplification and Processing

7.0 Mixing Consoles

Certain installations involve a number of microphones, located in the same area, (for instance the stage or platform of an auditorium), which need to be amplified at the same time. For simple speech reinforcement systems a mixing pre-amplifier is fully adequate to fulfil the requirements.

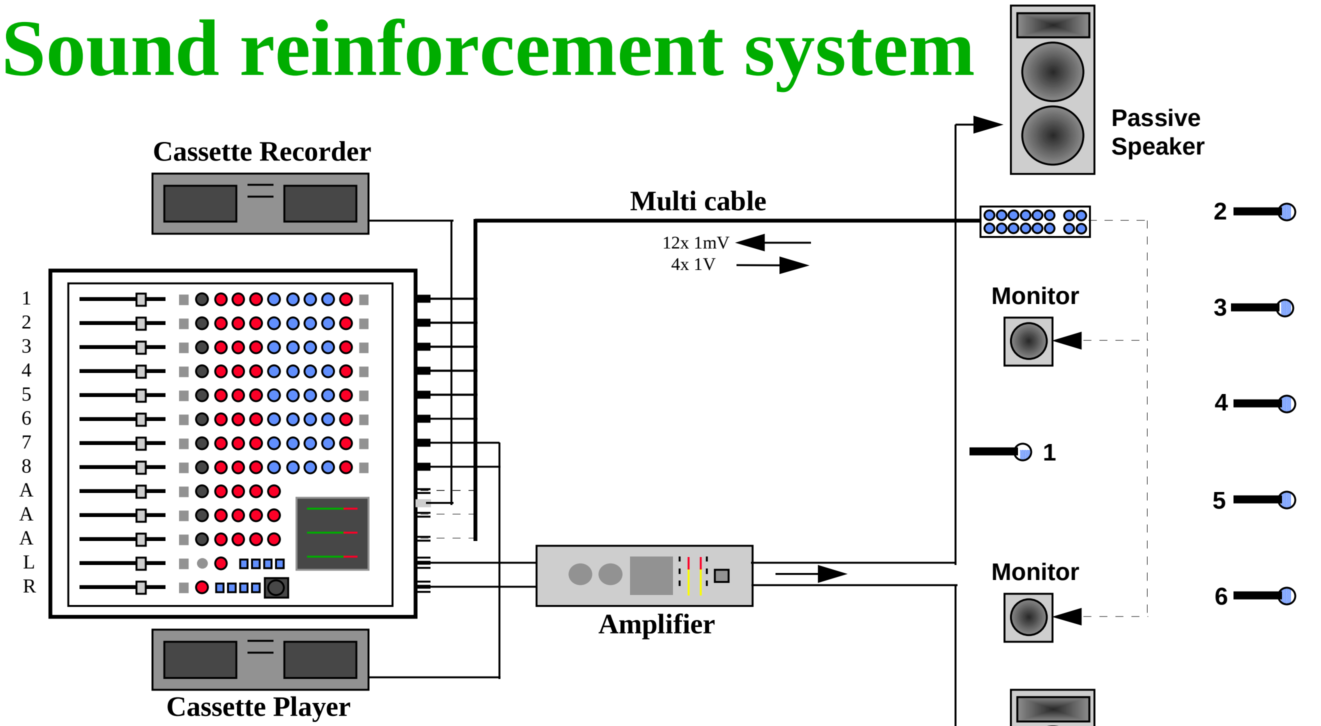


More elaborate installations involving a larger number of microphones a Mixing Console (or Mixing Desk) is the heart of this type of audio system, and is a device which takes the place of a simple pre-amplifier, being the control unit where all the microphones, cassette players, etc. come together. It accepts these various inputs and blends them together into one balanced whole. The final, mixed, sound is then sent to the input of power amplifiers, tape recorder and/or monitor loudspeaker(s).

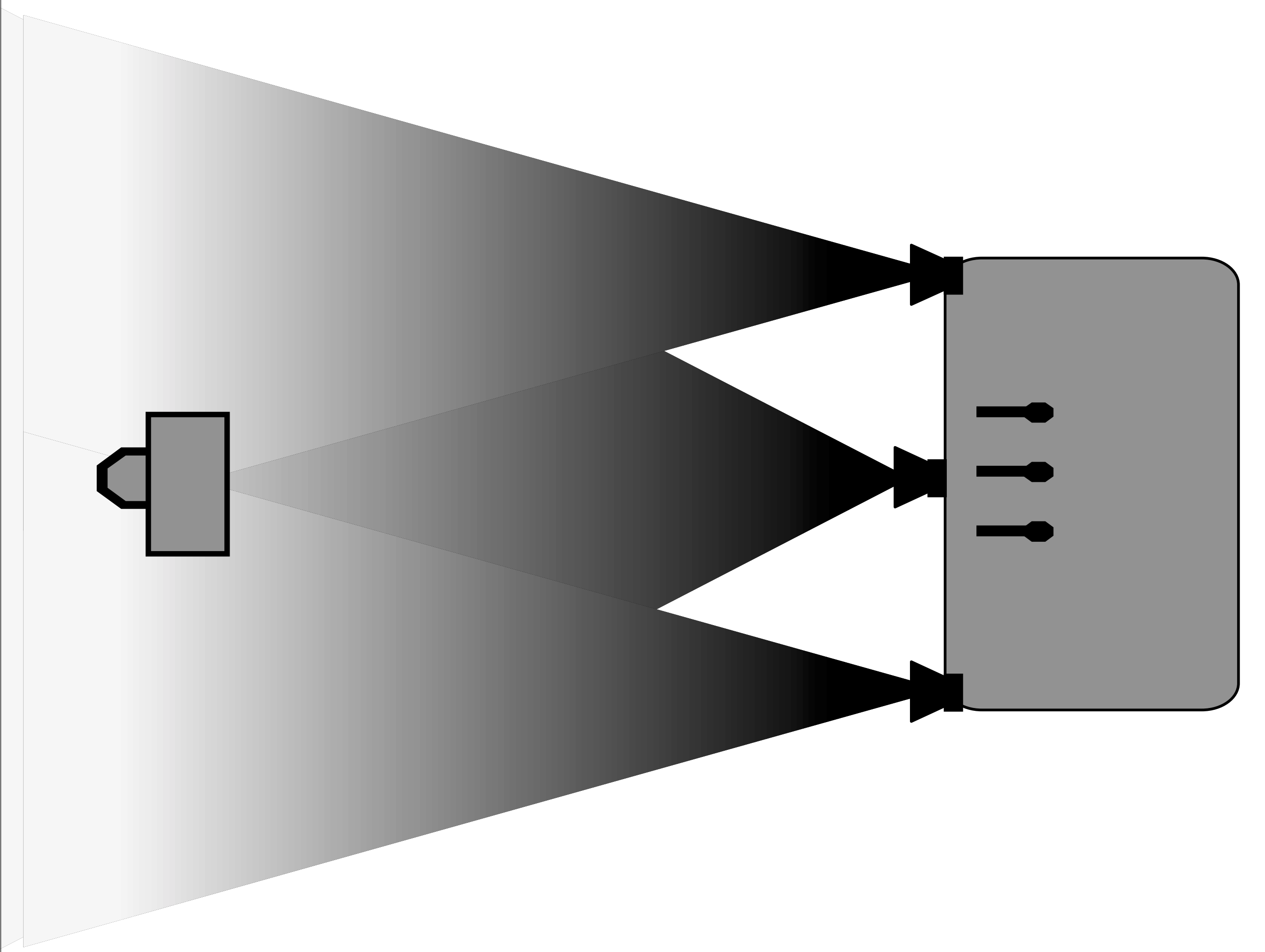


Mixing consoles range from simple units which accept 4 microphone inputs, have basic tone controls, and provide a mono output, to huge consoles having more than 60 input channels, each having very sophisticated equalisation, feeding a large number of sub groups, which in turn feed a selection of main outputs.
The latter type tends to be accompanied by several banks of audio processing equipment and is very much the domain of the professional mixing engineer.
In order to give the mixing engineer an undistorted judgement of the total sound, the favourite place for a mixing desk is in the middle of the auditorium.
On the next page a sound reinforcement system for an auditorium is shown.

Sound reinforcement system



Speech & music in small auditorium with recording & play back facilities



8.0 Amplifiers and Preamplifiers

Although quite often presented as a single unit, the public address amplifier must be considered as two separate sections: the pre-amplifier (voltage gain) and the output amplifier (power gain).
The **pre-amplifier** matches and amplifies the outputs of microphones, CD and cassette players, tuners, etc., to provide a voltage level suitable for driving the power amplifier. The pre-amplifier also normally incorporates the tone controls, input sensitivity adjustments, and master volume controls.
The **power amplifier**, often available as a separate unit, is used to amplify the output power of a pre-amplifier, distribution system, or mixing console to a level that will feed the loudspeakers properly. If necessary it is possible to link power amplifier inputs together so that a single input signal can feed a large number of amplifiers.

8.1 The Pre-amplifier

8.1.1 Inputs

The pre-amplifier is normally used for matching and amplifying small voltages, to provide a voltage level, usually 500 mV or 1 V, which is suitable for driving the power amplifier.
Typical inputs to the pre-amplifier may be:

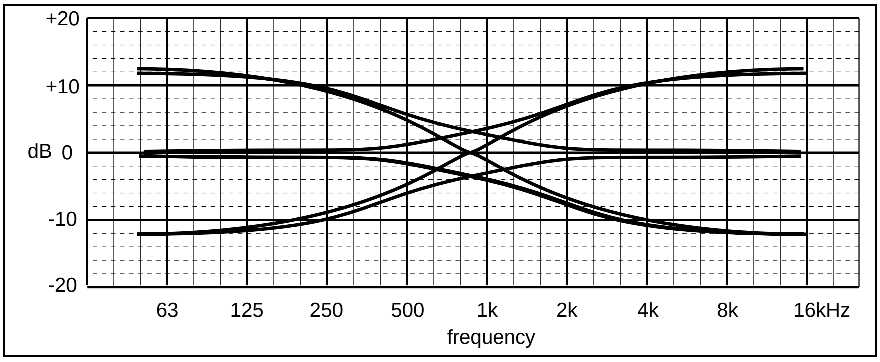
- moving coil (dynamic) microphone - 0,25 mV;
- electret or BPE microphone - 1 mV
- condenser microphone - 3 mV;
- dynamic pick-up - 5 mV;
- domestic source (tuner, cassette, CD, DCC etc) - 250 mV;
- professional tape recorder - 1,5V.

From this range of input requirements two inputs are often chosen: a microphone input with a sensitivity of 0,5 mV to 1.5 mV; and a music input of 100 mV to 1,5 V.
Tone controls, input sensitivity adjustments, and master volume controls are usually built into the pre-amplifier.

8.1.2 Tone controls

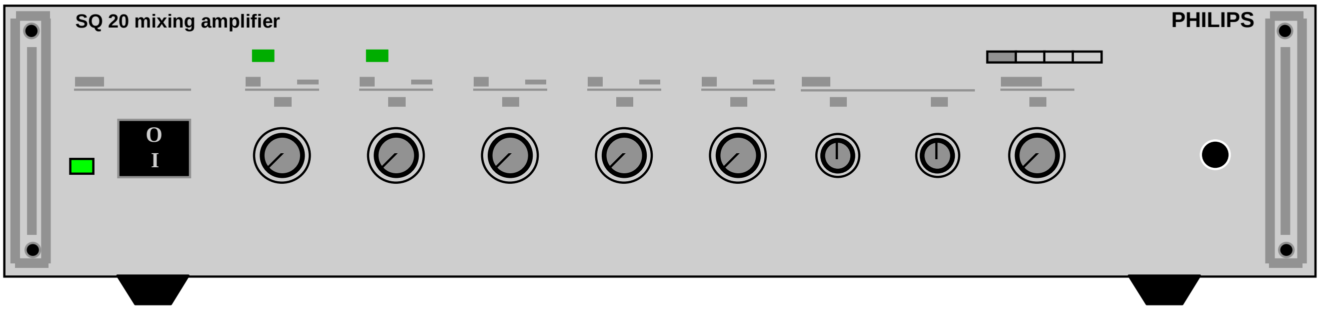
Tone control circuits vary the frequency characteristics of an amplifier.
The bass and treble tone control circuits, with which most people are familiar, are basically amplification and attenuation circuits, which operate over specific frequency bands. They operate as follows;

1. If the bass or treble potentiometer is turned to the right, from its 0 ('flat') position, the gain is increased, and the frequencies within its band of influence are amplified, giving an increase in volume of the respective bass or treble frequencies. The 'lifting' of the treble frequencies is particularly useful when it is desired to give speech greater clarity, helping it to 'cut through' noisy environments (see chapter 1.1 for information regarding the speech spectrum).
2. If the potentiometer is turned to the left; the respective bass or treble frequencies are attenuated. Bass attenuation is particularly useful in large rooms, where long reverberation times at low frequencies cause problems.
3. Bass lift and treble attenuation is rarely required. Bass lift could be used when amplifying music in a heavily damped room, where the bass frequencies would require reinforcement to give the music more depth. Care should be taken though not to overload the loudspeakers when amplifying the bass content of a signal.



Please note that some lower quality pre-amplifiers provide only attenuation, giving no amplification of either bass or treble frequencies.

In contrast to this, all Philips' professional preamplifiers provide both amplification and attenuation (see example next page).



8.2 The Power Amplifier

The power amplifier is used to amplify the output voltage of the pre-amplifier, distribution system, or mixing desk, to a level that will feed the loudspeakers properly. Depending on the design philosophy of the manufacturers, the input required to feed the amplifier at nominal full power can range from 100 mV to 10 V.

Many power amplifiers used in public address systems, and all amplifiers in the Philips product range use what is known as the 100 Volt line principle. This type of amplifier is favourable if long loudspeaker distances are involved. (This principle is discussed in the following section) Other power amplifiers, often used in sound reinforcement systems, provide a direct low impedance 2, 4 or 8 ohm output. If using the latter, make sure that the impedance of the loudspeakers matches that of the amplifier, and that the amplifier power is always lower than the loudspeaker power, so that the amplifier is not able to overload the loudspeakers.

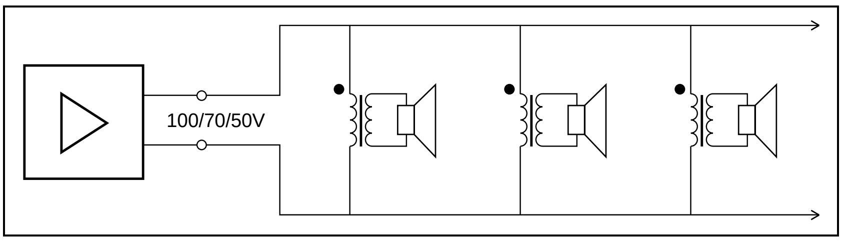
8.3 Amplifier/Loudspeaker Interface

As stated in 8.2, in order to interface loudspeakers with power amplifiers, all Philips amplifiers utilised what is known as the 100 Volt line matching principle, whilst certain amplifiers in the range also incorporate low impedance outputs. If the load is always constant, the loudspeakers can be connected in a series/parallel arrangement to exactly match the amplifier's low output impedance. However if the loudspeakers differ in power and impedance, or if the quantity of loudspeakers changes, it is very difficult indeed to match them to the power amplifier. In this type of situation, or in an application requiring long loudspeaker cable lengths, the 100 Volt line matching system is used.

In the 100 Volt line matching system, transformers, which are mounted in the power amplifiers, are tapped to step up the output voltage of the amplifiers from a low voltage to 100, 70 or 50 Volts. Transformers, mounted on the loudspeakers, then reduce this again to the original low voltage, acceptable to the loudspeakers.

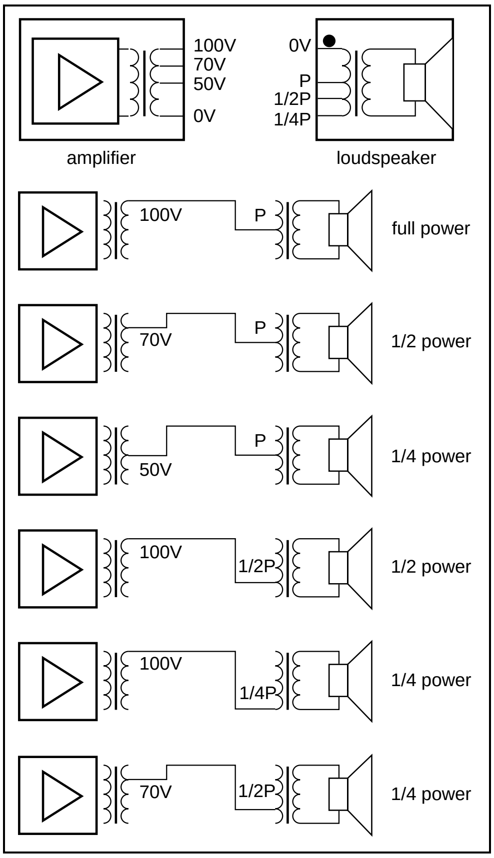
This system gives great flexibility in the design and use of public address systems for the following reasons:

1. By increasing the output power voltage of an amplifier, the amount of current (measured in amps) involved is reduced significantly. This means that even when high power amplifiers are used, line losses are kept low, and heavy duty cabling is not required.
2. Due to these low line losses, extremely long cable lengths are possible. This is a very important factor in a public address installation.
3. All loudspeakers may be simply connected in parallel.



So long as the total amount of watts drawn by the loudspeakers is not greater than the rated output power of the amplifier, it does not matter whether there is 1 loudspeaker or 150 loudspeakers connected to it at any time.

The 100V line principle can be compared to a normal domestic mains electricity power supply. In a mains supply, a constant supply voltage is present, and it is necessary only to plug an appliance into the mains socket for it to become operational. The amount of appliances plugged into a supply is irrelevant, so long as the total amount of power (wattage) drawn is not greater than that available.

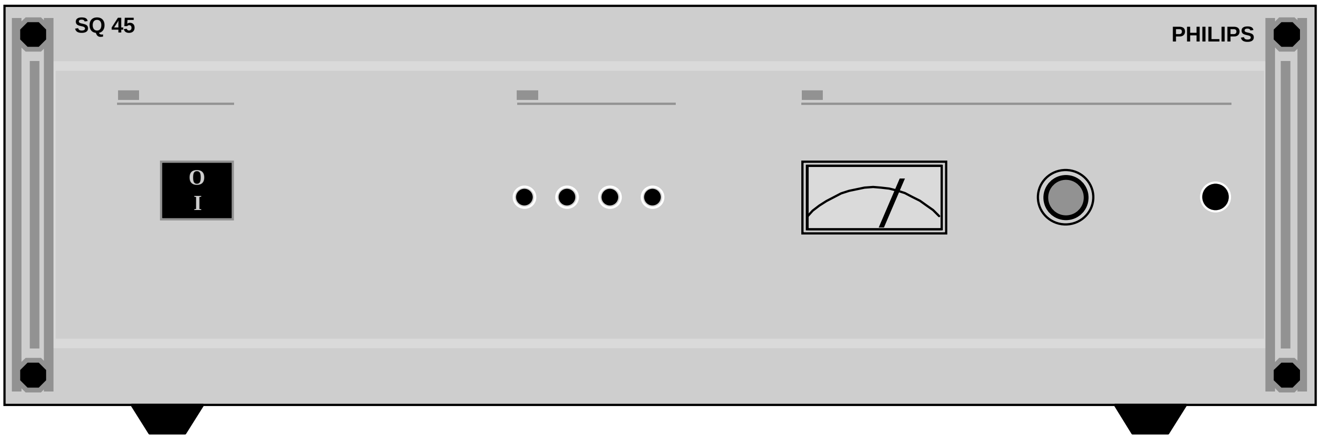


When loudspeakers are connected to the 100V amplifier tap, their full power is drawn, whereas if they are connected to the 70V tap, only 1/2 of their rated power is drawn. This means that the 70V tap enables the amplifier to power twice as many loudspeakers, with each loudspeaker producing 1/2 of its potential power.
Similarly, the 50V tap allows loudspeakers to draw 1/4 of their rated power, so that the amplifier is able to power 4 times more loudspeakers, with each producing 1/4 of its potential power.

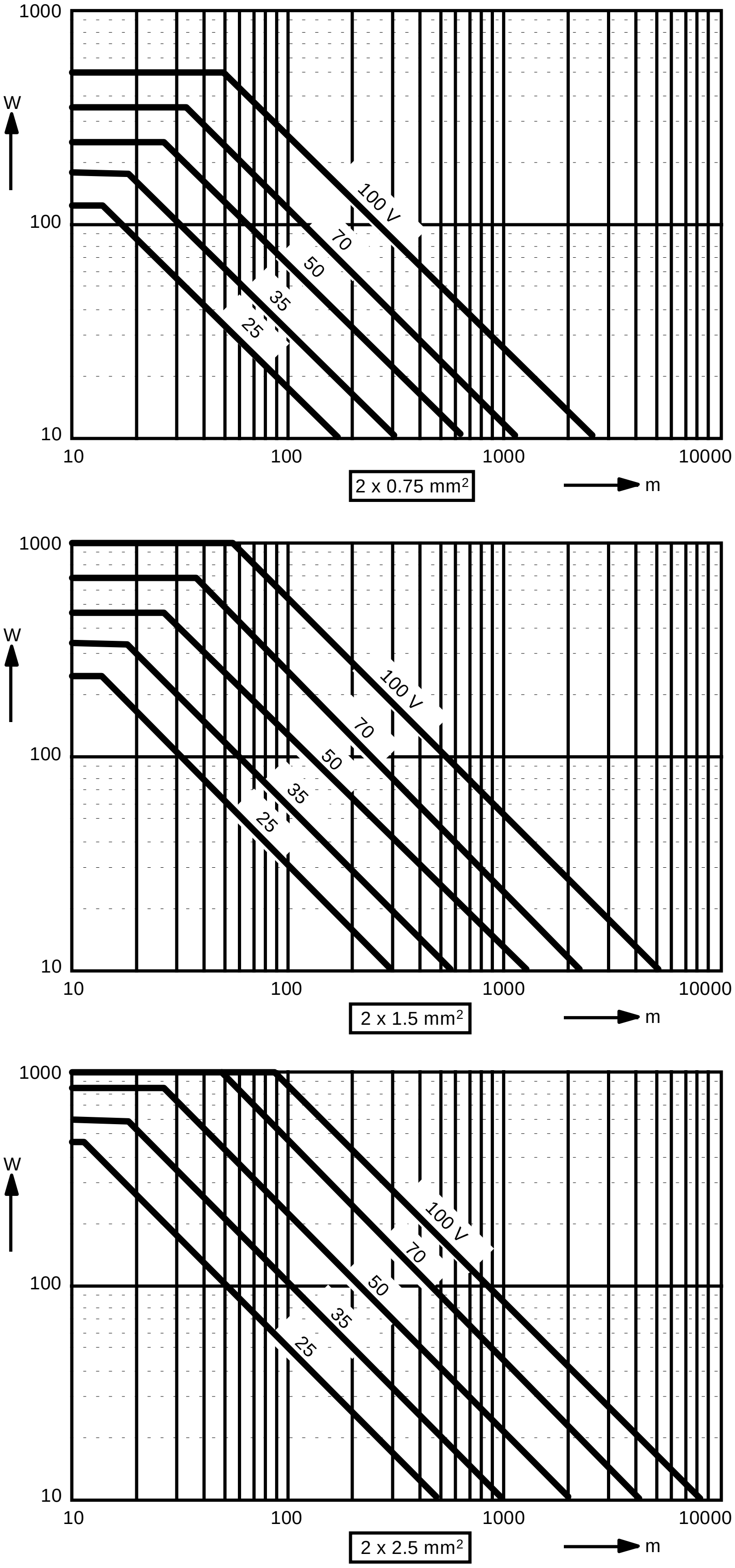
The transformers fitted to the loudspeakers have similar taps, but in this case the actual power which the loudspeaker will draw (e.g. P, P1/2, P1/4, or 6W, 3W, 1,5W), instead of the voltage, is printed beside the "power" (+) tap. These loudspeaker transformer taps are used in the same way as the amplifier transformer taps; matching the power drawn (in this case by each loudspeaker) to the amplifier power available.

When it is desired to reduce the power drawn by all of the loudspeakers, it is of course simpler and more efficient to utilise the amplifier transformer taps. It is possible though, by using the loudspeaker transformer taps, to reduce the power drawn by only a quantity of the loudspeakers, while the remainder draw full power.

Note:
When using the 100 Volt line matching system, the Rated Power of the amplifier corresponds to the Rated Load Impedance of the loudspeaker network. The total rated power required should be calculated, by simply adding the Rated Power of the connected loudspeakers together, taking into account the reduction in power drawn when using the loudspeaker power taps. **It is important that this total should not exceed the rated power of the amplifier.**



Maximum Permissible Cable Lengths



Cable lengths.

The maximum permissible cable lengths per size of cable are shown in the accompanying graph.

Example:

Assuming an amplifier of 100 W tapped at 100 V and using a cable of 2x0.75 mm^2 . The length of the cable should not exceed 250 m.

The values refer to a 10% voltage drop, with the entire load concentrated at one end of the cable. The lengths can be doubled when the load is distributed evenly along the cable.

Transformers.

Whilst considering the many advantages of the 100 V line matching system, it is important to realise that by inserting transformers into the signal chain, certain losses must occur.

Any transformer has an insertion loss. If for example, 10 W is required at the loudspeaker terminals, using a transformer with an insertion loss of 1.5 dB would require 14.13 W output from the amplifier.

The impedance of transformers also varies with frequency, which of course has an adverse effect on the overall system frequency response, and the demands placed upon the amplifiers, especially when reproducing bass frequencies.

9.0 Equalisers

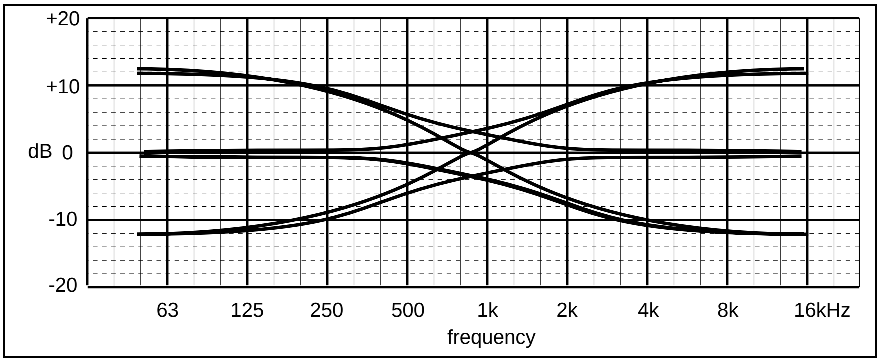
An Equaliser gives extensive control over the whole audio frequency spectrum by means of presence (gain) and absence (attenuation) filters and can be used for optimising the frequency response of the sound system.
It can even equalise the complete audio chain, from microphone to ear. Used with care, this would guarantee maximum amplification for the whole frequency spectrum, at the same time combating the problem of acoustic feedback by reducing the level of frequencies which cause it.

9.1 Equaliser Types

9.1.1 Basic tone controls

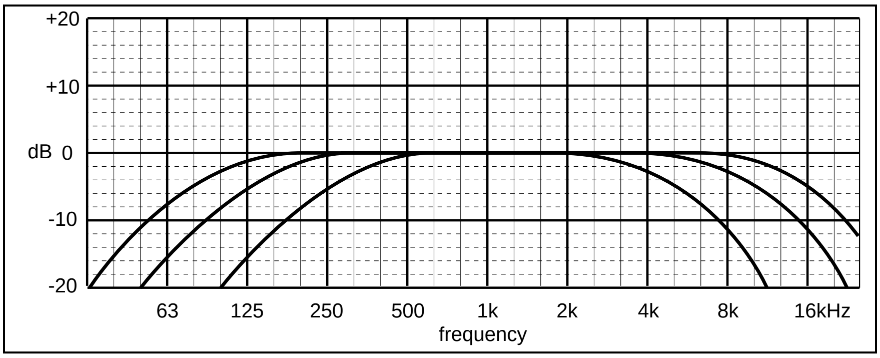
The bass and treble tone control circuits, with which most people are familiar, are basically amplification and attenuation circuits which operate over a specific (though fairly broad) frequency band. They operate as follows:

1. If the bass or treble potentiometer is turned to the right, from its 0 ('flat') position, the gain is increased, and the frequencies within its band of influence are amplified, giving an increase in the volume of the respective bass or treble frequencies. The 'lifting' of the treble frequencies is particularly useful when it is desired to give speech greater clarity, helping it to 'cut through' noisy environments
2. If the potentiometer is turned to the left; the respective bass or treble frequencies are attenuated. Bass attenuation is particularly useful in large rooms, where long reverberation times at low frequencies cause problems.
3. Bass lift and treble attenuation are rarely required. Bass lift could be used when amplifying music in a heavily damped room, where the bass frequencies would require reinforcement to give the music more depth. Care should be taken though not to overload the loudspeakers.



These treble and bass tone control circuits are very basic units, operating over wide frequency bands, raising or attenuating all of the bass or treble frequencies.

9.1.2 Band-pass filters

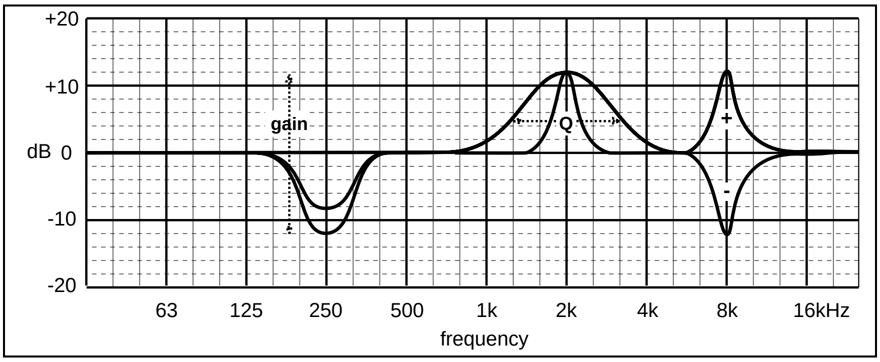


Bass and treble "Hi-Pass" and "Lo-Pass" (or "cut-off") filters are intended to restrict the frequency band. Their purpose is to severely attenuate all signals below or above a fixed (normally very low or very high) frequency.

In situations requiring control over specific frequency bands, a variety of equalisers are available see next page:

9.1.3 Parametric equaliser

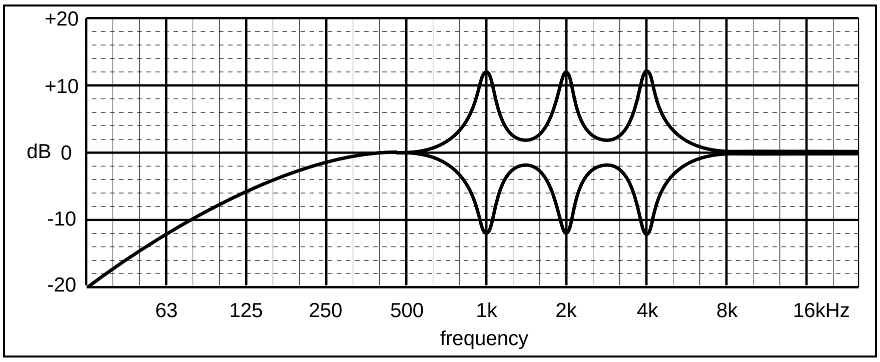
A parametric equaliser is a unit with 3 or 4 filters, and the possibility to adjust the frequency to be processed. The



processing consists of gain correction (+ , -),and a selection of the width (Q) of the frequency band. This makes it possible to alter, if necessary, a very small frequency band, without affecting the neighbouring frequencies. Because only a few filters are used, the overall response tends to be quite smooth. (See 9.2.4)

9.1.4 Parametric triple Q-filter

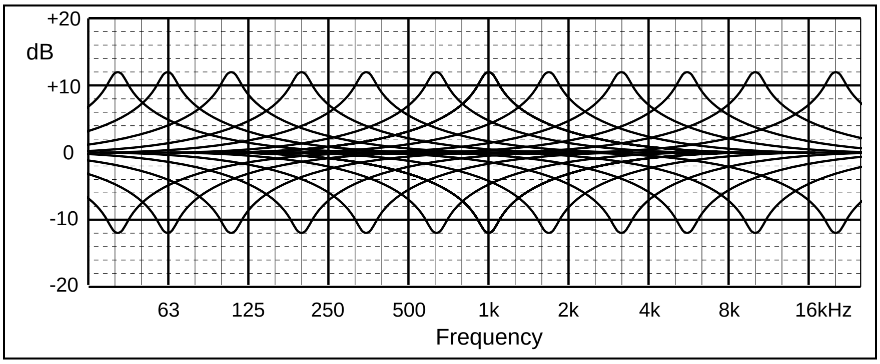
Basically a parametric equaliser but with pre-set fixed (speech) centre frequencies e.g. 1-2-4 kHz. This filter allows the operator to select the width & slope of the frequency band (Q) and presence or absence (Gain).



The unit is ideal for optimising the amplification of that part of the frequency-band that is responsible for speech intelligibility, it adds clarity and compensates for air absorption. An adjustable bass cut filter provides smooth roll-off of the bass content in the signal caused by e.g. speaking too close to a cardioid microphone.

9.1.5 Graphic equaliser

A Fixed Frequency or "Graphic" Equaliser often consists of 30 individual filter sections. Each control, which is



often a sliding potentiometer or "fader", effects a narrow frequency band (third octave). The "peak", or maximum effect is at the centre of each band, with the surrounding frequencies being effected to a proportionately lesser degree. The total frequency spectrum is covered, allowing the signal to be sculptured at several specific frequency bands as desired. To avoid excessive phase shifting, care should be taken to avoid extremes of variation between adjacent controls with, for example, one control at full attenuation and its neighbour at full amplification. The maximum level of both speech and music is in the 250-500 Hz frequency range. The level at these frequencies should be kept as near to 0 dB as possible to avoid distortion due to a general level increase.

9.2 EQUALISATION

9.2.1 Introduction

The increase in sound level which a sound amplification system can give to a performance in an auditorium is limited by acoustic feedback.

In many cases, e.g. when the performance requires a long microphone distance or with a somewhat noisy audience, the feedback limit prevents an adequate sound level being produced for comfortable listening. This situation can be annoying to both performers and audience because of an insufficient sound level or when the system amplification is increased, spurious ringing sounds.

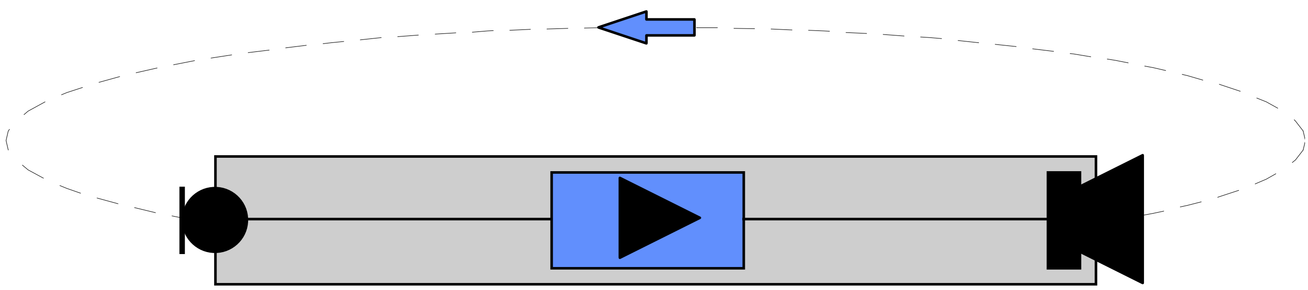
The use of sound equalisation to reduce acoustic feedback contributes toward the comfort of both performers and audience and will enhance the acoustic quality and increase the overall system gain.
Room conditions can also reduce intelligibility as they "colour" the sound by changing the frequency response. This effect can also be corrected by equalisation.

Though the selection of microphones, amplifiers, and loudspeaker types is vital when creating a system with smooth response, it is assumed that the sound system in question has already been optimised prior to conducting any equalisation measurements**.**

9.2.2 The acoustic feedback loop

The total system loop contains basically a sequence of the following elements:

- microphone
- amplifiers with volume control and possibly a tone control (e.g. mixer)
- loudspeakers
- acoustic transmission link between loudspeaker(s) and microphone



The acoustic transmission link consists of one (or two) direct path between the loudspeaker(s) and the microphone which is maintained by what is called the "direct sound field", and many other paths caused by reflections and multi-reflections which are maintained by what is called the "diffuse sound field".

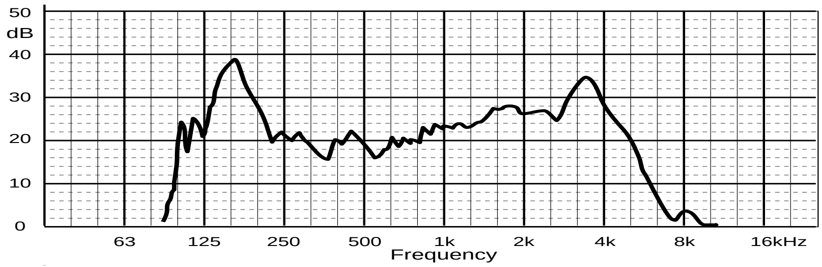
9.2.3 Resonant acoustic feedback

Acoustic feedback is spontaneous oscillation caused by the transmission of sound radiated by the loudspeaker(system output) back to the microphone (system
Spontaneous oscillations input).
When the gain of the sound system is gradually increased, a point will be reached where spontaneous oscillations (howling) start to occur.or resonant acoustic feedback can occur at any frequency for which:
a) the phase angle of the transmission through the acoustic feedback loop equals zero, and
b) the sound from the loudspeaker re-enters the microphone louder than the original sound (loop gain ≥ 1)
This cycle repeats itself, with increased amplification until the sound reaches the system's maximum loudness or until someone turns down the volume!

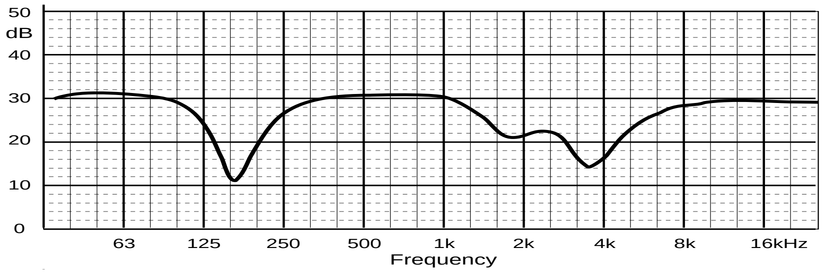
Even though a sound system is adjusted just below its critical gain, feedback will prolong the signal components at this critical frequency, producing ringing or howling sounds. To avoid ringing sounds during speech or musical performances, the gain has to be reduced to approximately 6 dB below the level at which spontaneous oscillation begins, this is called Feedback Stability Margin (FSM).

9.2.4 Principles of equalisation

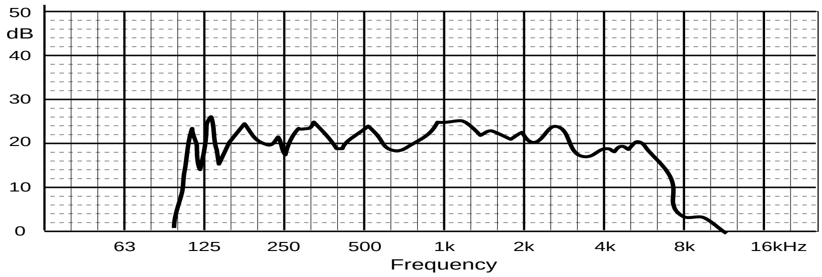
The ideal in any audio system is to obtain a flat frequency response over the complete audio frequency band. When considering the sound system equipment alone, a flat frequency response can be achieved within very fine limits, but when taking the sound system as a whole, with its associated acoustic link, changes are introduced to the feedback frequency response by the very nature of the auditorium. The cancellation of these changes in the frequency response, whether they be peaks or dips, is called "Equalisation".
Using a measuring set-up as explained on the next page (9.2.5) we can obtain e.g. the following loop response:



The dominant frequency where the acoustic feedback is likely to occur is 160 Hz and secondly 3.4 kHz.

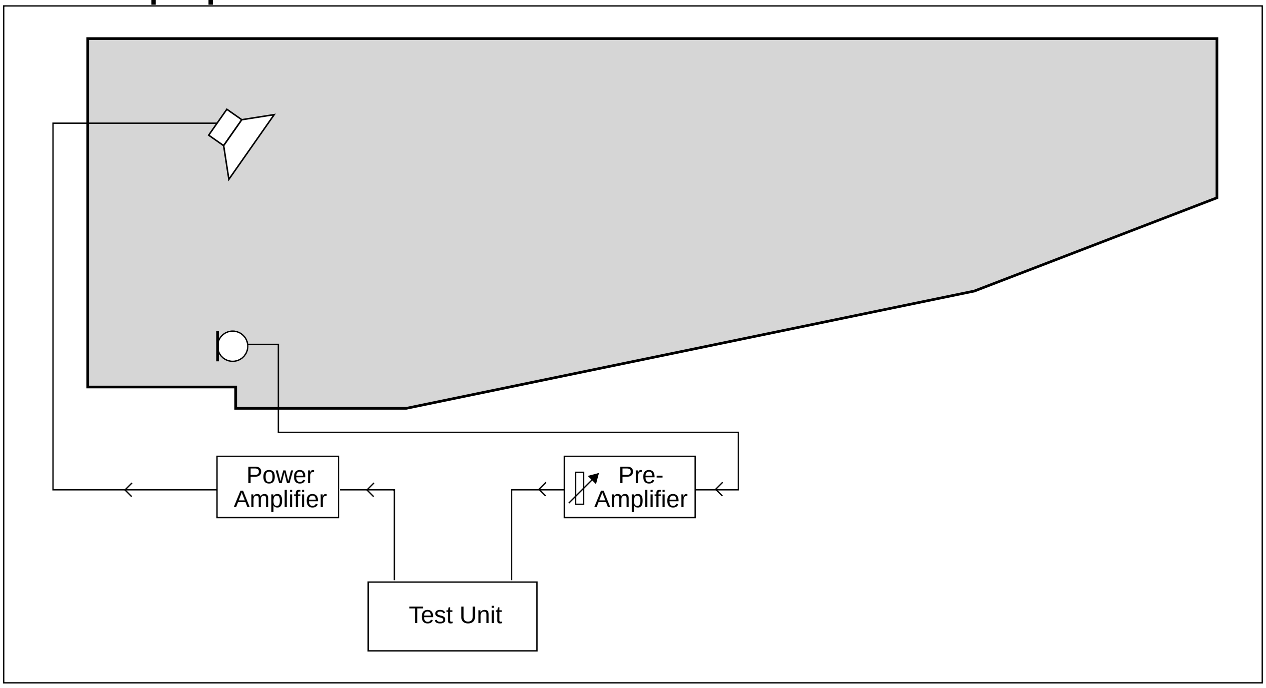


By equalising the loop response now with an "mirror imaged" filter response, the overall gain can be increased. In order to maintain the optimum signal to noise ratio, increasing the gain at dips in the frequency response, as well as reducing the resonant peaks, should be considered. This can be done manually by means of a graphic or parametric equaliser or automatically by a so called intelligent feedback exterminator which work with a number of narrow band filters adjusted dynamically at the critical frequencies and maintaining a FSM of 6dB.



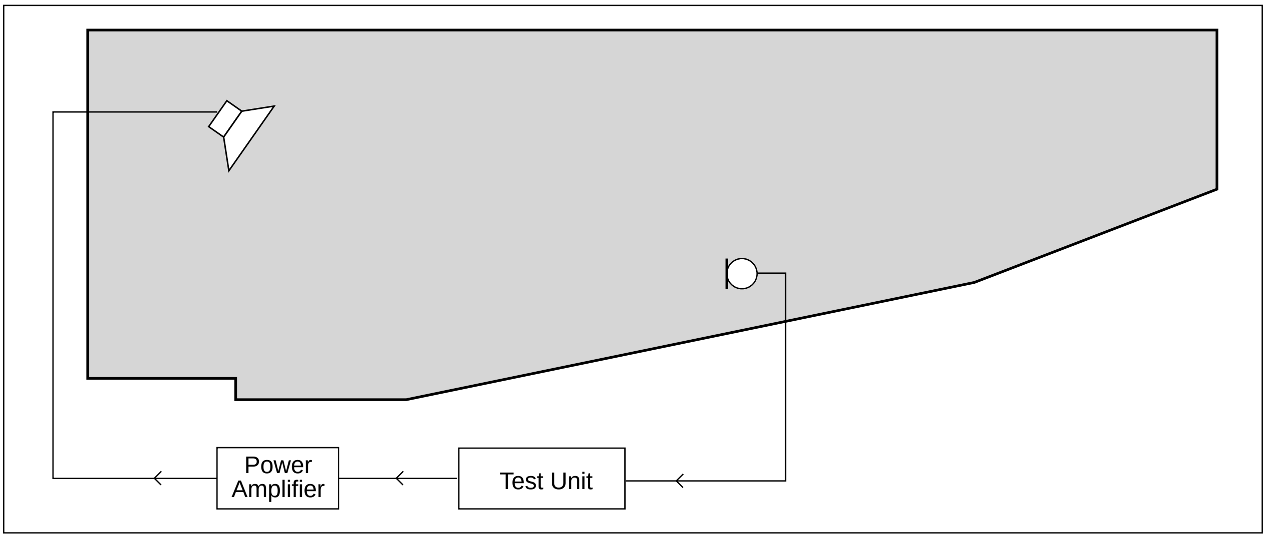
It is impossible to lay down hard and fast rules as to which equalisation method should be used, as the requirements will be vary from one auditorium to another. The prime objective is to obtain a flat frequency response of the loop to obtain max. possible gain for all frequencies and preserve the signal to noise ratio.
A listening test after equalisation is important because a flat loop response is not always a flat listening result, a high frequency roll off is sometimes required (3 dB/octave > 1 kHz).

9.2.5 Loop equalisation



In a speech reinforcement system facing the problem of acoustic feedback, we equalise the whole loop, which consists of the system microphone(s) - amplification - loudspeaker(s) and room.
The test unit produces a 1/3 octave “warbled” tone, which glides from 20 Hz to 20 kHz, and is fed into the power amplifier. The corresponding output, reflected by the room surfaces, is received by the system microphone and plotted on the test unit recorder. Another method is to inject pink noise in the sound system and measure with a 1/3 octave Real Time Analyzer. This is a good method for adjusting a 1/3 octave graphic equalizer in the system.

9.2.6 Loudspeaker equalisation



In a system used for playing pre-recorded music, we concentrate our measurements on the loudspeaker reproduction. In this case we use a calibrated measuring microphone at the audience position (averaged), and equalise only the power amplifier, loudspeaker(s), and room. The most convenient method is to inject pink noise in the sound system’s line input and measure with a 1/3 octave Real Time Analyzer on the audience position. A 1/3 octave graphic equalizer is the easiest to adjust but difficult to hide for unauthorised tempering.

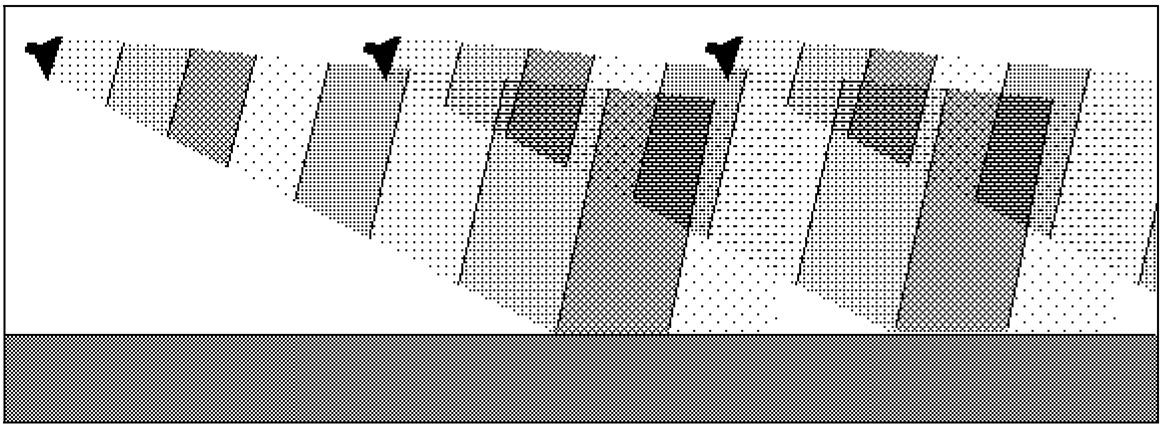
9.2.7 Loudspeaker equalisation & Loop equalisation

For sound systems used for music reproduction and sound reinforcement, where there is a need to optimise both, the loudspeaker equalisation should be carried out first, and secondly an additional equaliser should be used in the system microphone channel.

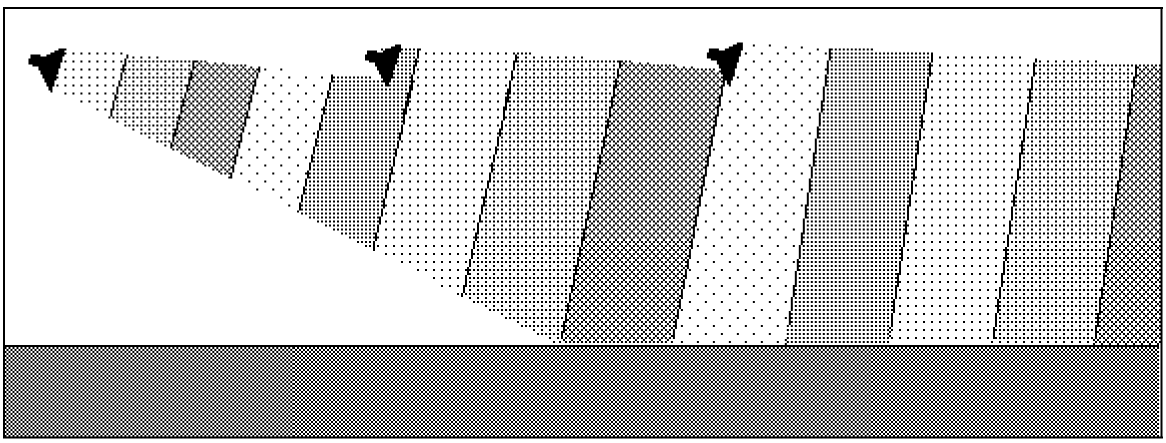
10.0 Time Delay

When a sound reinforcement system in an large auditorium, with loudspeakers located at the left and right hand side of the stage and dispersed at intervals along the length of the auditorium, a problem of timing becomes apparent. When all loudspeakers produce their sound at the same time, the listener hears the speaker's voice coming from the direction of the closest loudspeaker, instead of from the stage.
This conflict between the visual and audible experience is rather uncomfortable. To overcome this disturbing effect, the sound from each (group of) loudspeaker(s) must be delayed using time delay equipment. If the timing is set properly, (based on the speed of sound travelling at 5 meters per 15 milliseconds), and the sound of the loudspeakers arrives later (5-15 ms) and not more than 10 dB louder than the original speakers voice, the sound will appear to originate from the front of the auditorium or area, where the speaker is located.

Another problem occurs at railway stations, where the aural announcement origin is located at the closest loudspeaker, but is then followed by arrival of sound from the other loudspeakers, causing echoes and reverberation. To overcome this disturbing effect, the sound from each (group of) loudspeaker(s) must be delayed using time delay equipment. If the timing is set properly, the sound will be synchronised with the furthest loudspeakers and will benefit the intelligibility considerably. The most effective way of doing this is to use the loudspeakers located in the middle of the platform as the starting point. The other loudspeakers which should be pointing away from this centre position, should be delayed proportionally so that the sound appears to come from this centre position. The loudspeakers should be selected carefully and angled for a minimum of backward radiation.



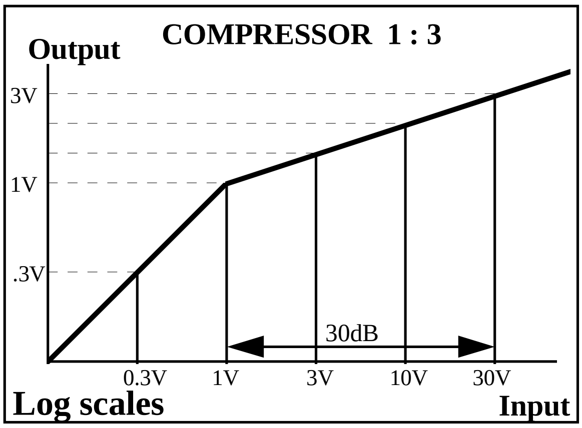
Railway platform without delayed loudspeaker signals

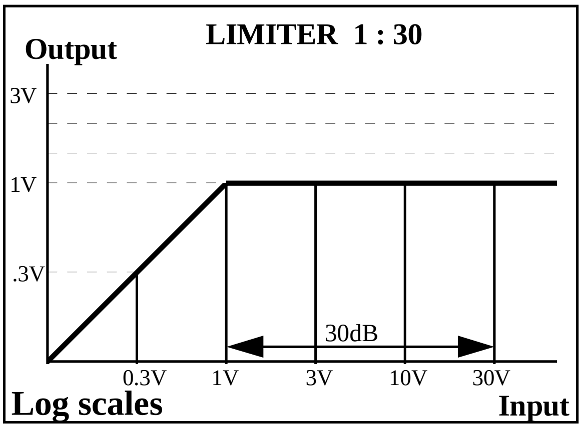


Railway platform with correctly delayed loudspeaker signals

11.0 Compressor/Limiter

A compressor and a limiter are input signal dependent attenuators. The dynamics of input levels below the threshold are not affected, but the dynamics of levels above are reduced. The attack time is 1 ms, while the adjustable release time is dictated by the application, short for speech (100 ms), long for music (>1s).





A **compressor** reduces input signal variations above the threshold level to about one third (in dB's) without introducing distortion.
(30 dB input variation gives only 10 dB output variation).
A compressor is ideal for background music applications to reduce the (often unwanted) large dynamic range of recordings or broadcastings.
The release time should then be set on >1s to avoid music sounding unnatural (pumping).

A **limiter** effectively restricts the output level to e.g. 1V for all input levels above the threshold level without introducing distortion. A limiter is ideal for mounting in call stations to guarantee a fixed maximum output level, independent of the person speaking (male/female/distance/loudness). To utilise this maximum peak level with the full capability of the sound system, it is necessary to align the rest of the chain in such a way that also the maximum undistorted output level of the amplifier is reached.

12.0 Automatic Volume Control

Automatic Volume Control (AVC) regulates the loudness of a P.A. announcement relative to the ambient noise level. This guarantees maximum intelligibility and minimum annoyance.

The ambient noise level is continuously measured by a microphone connected to the sensor input of the AVC unit, which uses this measurement to set the attenuation of the signal path. During periods of low ambient noise, the PA system gain is reduced by the AVC-unit, and during periods of high ambient noise, the PA-system gain is restored to its nominal maximum. A blocking circuit 'freezes' the input sensor while an announcement is being made, ensuring that the announcement itself is not measured by the unit as ambient noise.

The control range of the AVC, with attenuation values from 6 to 21 dB, depends on the maximum loudness of the sound system. 80 dB(SPL) is regarded as a comfortable maximum listening level. If the loudspeaker system is set up so that a maximum of 89 dB SPL can be achieved, then a control range of 9 dB would be the right choice. An AVC unit with 21 dB control range would only be used in PA systems which can produce a maximum level of 101 dB(SPL), being 21 dB above the comfortable listening level of 80 dB(SPL).

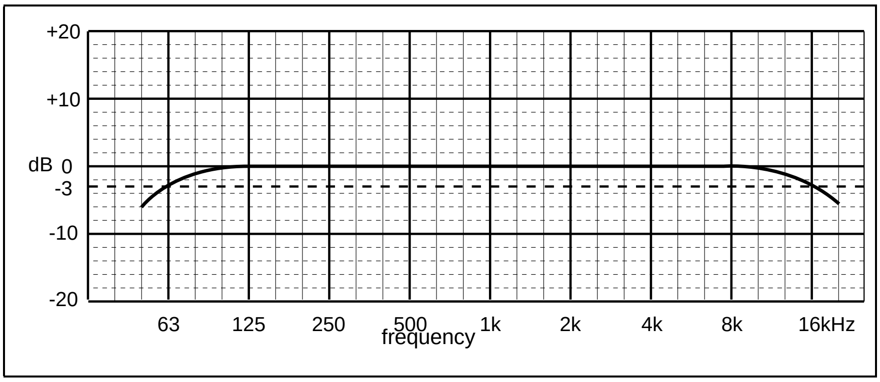
The AVC unit is factory pre-set, therefore only the sensing input microphone gain in the corresponding loudspeaker-zone and the reset time (blocking) needs to be adjusted.

If a microphone is located inside the loudspeaker zone to which it is addressed, the gain should be carefully set to avoid acoustic feedback. The system should be checked during periods of high ambient noise and low level talking into the microphone in order to ensure that no acoustic feedback, automatic attenuation(AVC), or limiting (Callstation) occurs.

13.0 Technical Considerations

13.1 SPECIFICATIONS

13.1.1 Frequency Response



This graph illustrates the typical flat response of an amplifier suitable for music reproduction.
The written specification of this type of frequency response should state the frequencies at the points where the curves have dropped by 3 dB.
In our example, the frequency response is from 63 Hz to 16 kHz.

When this specification relates to power amplifiers the level at which it is measured should be 10 dB below the rated output power.

Specifications should be read carefully. If a manufacturer chooses -6 dB points as reference, he is able to quote a frequency response range which extends much wider than more ethical competitors.

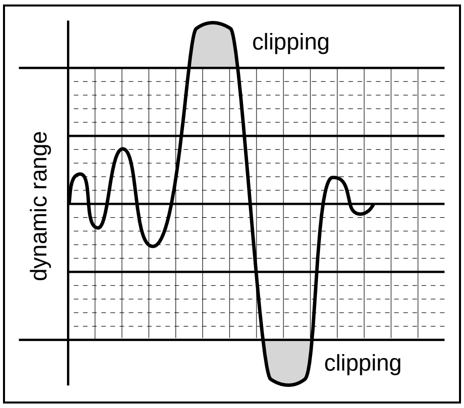
13.1.2 Power bandwidth

The Power bandwidth is the frequency range in which the amplifier can deliver its rated power (-3dB) with a maximum distortion level (THD) as stated by the manufacturer (0.5% for PA amplifiers).

13.1.3 Linear distortion

If an amplifier is not capable of amplifying the full frequency spectrum equally, the amplified waveform will be altered in a similar way as when tone controls are used. This unwanted modification of the signal is called **linear distortion**, which in its extreme could give rise to a guitar input producing a 'piano' sound output.

13.1.4 Non linear distortion or clipping (THD)



This graph shows an amplifier with too much input signal.

If the amplifier is overdriven, a clipping of the output voltage is likely to occur. This effect, called **non-linear distortion,** happens when the input signal exceeds the dynamic range of the amplifier. When the voltage is clipped, the normal curve of the signal wave is squared off, producing extra harmonics of the fundamental. This is commonly referred to as Total Harmonic Distortion (THD).

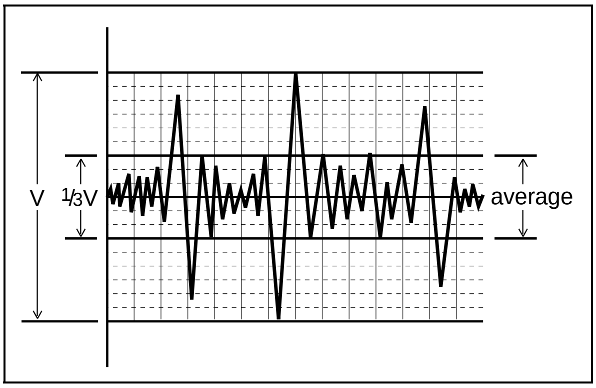
The result is an audible change, making the sound uncomfortably raw.

Another problem occurs when the current continues to rise, causing too much energy to be fed into the loudspeakers (beyond their Power Handling Capacity (PHC) limits), which could cause them to be damaged.

13.1.5 Rated Output Power

Rated Distortion Limited Output Power is the power which the amplifier is capable of dissipating in the rated load impedance, at a given frequency or frequency band (1 kHz), without exceeding the rated Total Harmonic Distortion (THD). This is defined in publication IEC 268-3.

Emotional speech, or certain passages of music, can cause pronounced audio signal peaks. Such instantaneous features of speech and music have to be reproduced without distortion. Generally, allowance must be made for speech attaining voltage peak values of approximately three times its average.
This may be expressed as 20 Log 3 = 10 dB. 10 dB, as a power ratio, means that the peak power is roughly 10 times that of the average power. This is called the 'rated power' of an amplifier. A 100 W amplifier, for instance, having a input sensitivity of 100 mV, will produce 100 W output when the input voltage reaches 100 mV.



This 100 W is the maximum output power which the amplifier can produce whilst still keeping distortion below its specified limit.

Under normal conditions, however, the average input voltage will only be 33 mV (allowing up to 100 mV for peaks) and the average output power will only be 10 W (allowing up to 100 W for peaks).

This means that, on average, an amplifier normally operates at only one tenth of its rated (or peak) value. In our example this will be 10 W.

13.1.6 Temperature Limited Output Power (TLOP)

The IEC 65 standard states that an amplifier, running under worse case conditions, should at least be able to run continuously at 12½% of its Rated Output Power without any components overheating.
This means that 100 W amplifiers, located in ambient temperature of 45° C, with + 10% mains over-voltage, stacked on top of each other, in a 19 inch rack frame, should be able to run continuously for 24 hours per day at 12.5 W average power without overheating.

13.2 ADJUSTING SIGNAL LEVELS IN A SYSTEM CHAIN.

1 **Microphone** in a Callstation is often combined with a pre-amplifier and limiter to optimise the signal in the transport cable. The max. output signal is due to the limiter restricted to 0 dBV (=1V).
The potmeter affecting the gain before the limiter should be adjusted to the announcer and/or acoustic feedback. The limiter is activated by the peaks in the signal therefore the average level of speech will be around -8 dBV but the peak level is close to 0 dBV.

2 **Routing controller** (SM30 or SM40) has 0 dBV input sensitivity for 0 dBV output. The input adjusters should always be in maximum position and only be changed in the seldom situation that you do not want the full power out of the system for this corresponding microphone input.
The attention and alarm signals are separately adjustable to an average level of -8 dBV (can be checked as 0 VU on the amplifier). If signal processing is applied (tone controlling, equalising, time delay etc.) take care of their gain settings to avoid unwanted gain or attenuation for speech/music (can be checked with pink noise).

3 **Amplifier** needs 0 dBV at the input in order to deliver 100 Volt to the rated load impedance. For this rated outputlevel we specify THD - Power bandwidth - S/N ratio etc, acc. IEC 268-3 DIN45500 FTC etc.
The Temperature Limited Output Power acc.IEC 65 is specified as 9 dB below the rated output power under extreme working conditions and is a measure for the cooling capacity of the amplifier power stages (heat-sinks and/or ventilators). The VU-meter has an integration time of 240 ms and adjusted so that 40 Volt (sinewave rms) reads 0 VU (=8 dB below 100 Volt); therefore in practice, speech and/or music should give not more than 0 to +3 VU readings as maximum in order to guarantee that short peaks in the signal (exceeding 100V) do not cause unacceptable audible distortion.

4 **Loudspeakers** are (for reasons of electrical power transport and installer requirements) generally connected via a 100 Volt line system. Matching of the loudspeaker requirements to the available amplifier power is done via the tapping-down possibilities (1/2P-1/4P)(70-50V).
Generally the Power Handling Capacity acc. IEC268-5 will be greater than the Rated Power of the loudspeaker in order to avoid damaging of the loudspeaker during excessive signal overload (acoustic feedback!).
Rated Power of the amplifier corresponds via 100 Volt to the Rated Load Impedance of the loudspeaker network. Therefore the total rated power of the connected loudspeakers (taking the powertapping into account) should not exceed the rated power of the amplifier.

5 **Automatic Volume Control** (AVC) regulates the loudness of the P.A. announcement in relation to the ambient noise. This guarantees maximum intelligibility and minimum annoyance. During low ambient noise level the PA-system gain is gradually reduced with 9 dB by the AVC-unit.
The 9 dB controlrange is chosen to assure a good performance for ambient noise levels upto 75 dB(SPL). The loudspeakersystem should be set-up such that a calculated SPLtotal of 89 dB can be achieved, being 9 dB above comfortable listening level of 80 dB(SPL). The AVC-unit is factory pre-set, therefore only the gain of the sensing input should be adjusted to the applied microphone(s) in the corresponding loud-speakerzone and the resettime (blocking).
The AVC-unit should be by-passed for announcement microphones placed in the addressed loudspeakerzone for acoustic feedback stability. If not, the system gain will be reduced by the control range of the AVC during low ambient noise. Acoustic feedback stability should then be checked during high ambient noise levels and low level talking in the microphone in order to avoid any automatic attenuation or limiting.

The AVC-unit with 21 dB control range is only for those PA-systems which can produce a maximum level of 101 dB(SPL) being 21 dB above comfortable listening level of 80 dB(SPL).

Hardware Installation

14.0 Grounding and Screening

14.1 EARTHING (GROUNDING)

14.1.1 Safety and system earth's

In order for a sound system to operate satisfactorily and safely, care must be taken to ensure that it is adequately earthed (grounded).

The earth path provided by the mains cable is the 'protective', or 'safety' earth, which takes any potentially hazardous positive voltage down to ground if an electrical fault occurs.

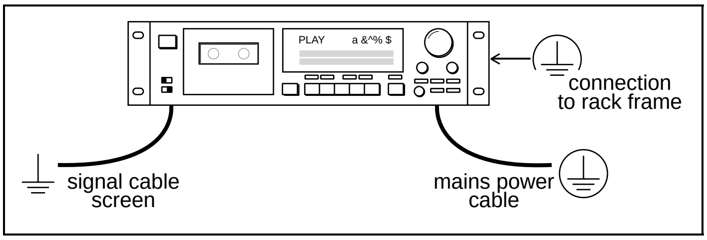
With professional audio equipment this must also act as a 'system earth', being connected to the system's screening (shielding) network, and taking all of the interference, 'collected' by the screening, down to ground. It is therefore vital that this is a 'clean' or 'noiseless' earth.

Unfortunately the mains earth is often contaminated with interference, caused by other types of equipment which use this common earth. If possible, an alternative clean earth should be established. The best way to ensure this is to make a separate path to earth by driving a long copper pole into the ground, and connecting this to the amplifier or system rack(s) with an adequate earth wire.

14.1.2 Earth (ground) loops

Incorrect earth wiring in a public address distribution system can cause malfunction of the equipment by introducing hum, distortion, or instability. It can even result in an overload condition, which may cause complete electrical breakdown of components. The main reason for the occurrence of these problems is the inadvertent introduction of earth (ground) loops in the earth wiring. Earth loops exist where multiple connections to earth are made from any one part of the system. Once the system is installed and these problems become apparent, it is often very difficult to trace the source of the problem. Because of this, it is vitally important to design the system so that an earth loop is not built in.

In a system, where several units are powered directly from the mains supply, earth loops can be caused by the mains wiring. A typical example of this would be a tape recorder or professional DCC or CD player, mounted in the 19 inch rack frame.



In this case three different paths to earth will have been established:

- via the mechanical connection of the chassis to the rack unit.
- via the mains earth wire.
- via the signal cable screen.

An earth loop would be the result.

Special measures must be taken to remove the earth loop, but at the same time, for safety reasons, the earth connection to the units must be maintained.

To ensure that each unit within the system has only one path to earth:

1. Even though many domestic, and some professional, music sources and auxiliary equipment have no electrical connection to the mains earth, there is often the possibility of a mechanical connection when, for instance, the chassis of the unit makes metal to metal contact with the rack frame.

   To avoid intermittent contact, and guarantee maximum security, each amplifier and music source chassis should be securely earthed (if necessary with short lengths of wire) to the rack.

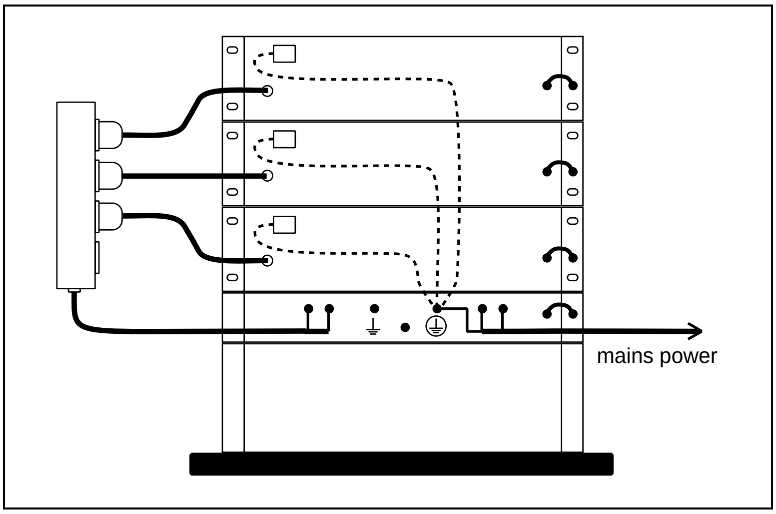
2. The earth of the power cable carrying the electricity supply from the mains should be securely connected to the rack, and all the mechanical earth's connected at that point.

3. In any amplification equipment, two earth's are present. One is the 'electrical' earth, connected to the 0V side of the circuit. The other is the 'mechanical earth', connected to the unit's chassis.

4. On amplifiers which are stacked, or mounted in a 19 inch rack, all electrical earth's should be connected to the mechanical earth at the same point. These earth's should not be wired together, but an individual wire should be run from each amplifier to the earth connection point.

4a. Where a separate clean earth is available, these electrical earth's should be wired to this ground connection point.

4b. In instances where the only available earth is the mains safety earth, all the electrical earth's should be connected at the point where the power cable carrying the electricity supply from the mains is connected to the rack.



5. When mains powered domestic music sources (CD players, etc.) are used in a system, a 1:1 isolating (or "galvanic separation") transformer should be fitted in the signal cable.

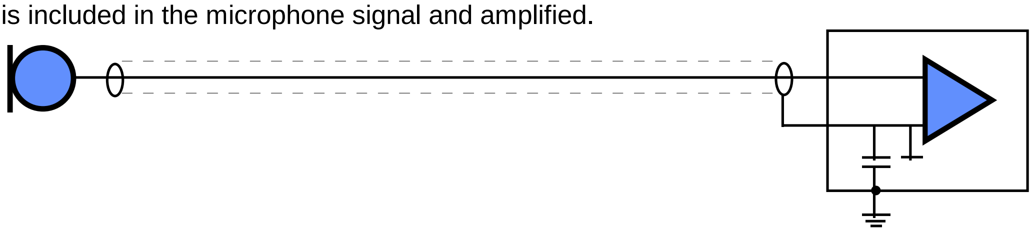
6. An alternative to this is to connect the signal screen to earth at one end of the cable only. This should be connected at the amplifier or preamplifier input, but not at the output of the signal source (CD player, cassette machine etc.).

14.1.3 Microphone Earth Loops

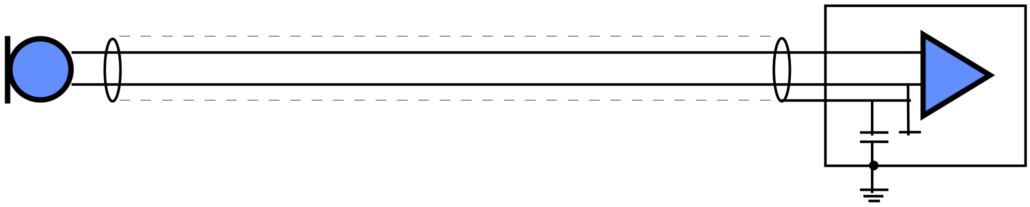
Earth loops are a common source of hum in installations with long microphone and connection cables.

To prevent earth loops it is wise to follow the next considerations:

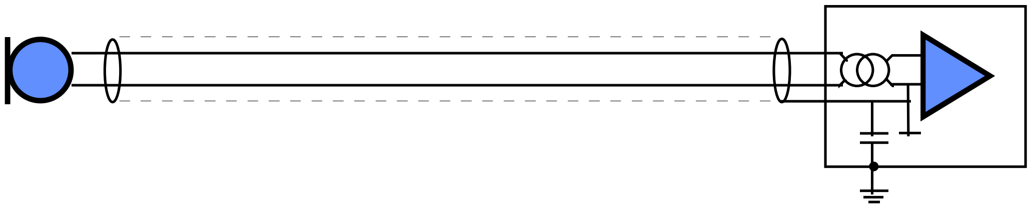
1. If it is necessary to use **single screened** cables, these should only be used for very short microphone cables in noise-free environments. With such cables, the screen is used as the return connection for the microphone, and because the screen is then earthed at the pre-amplifier input, any noise or hum induced in the cable is included in the microphone signal and amplified.



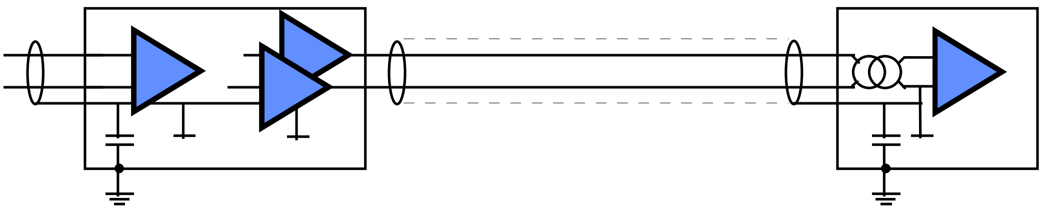
2. Normally **twin core screened** cable should be used. The screen, which is connected to earth at the pre-amplifier input, is not used as the return for the microphone. In most cases, no problems with hum pick-up will occur if the microphone is wired in this way. In severe environments, with intense magnetic fields, it is possible that noise or hum will be induced, not only in the cable screening, but also in the cable cores.



3. For 100% security, a twin core screened cable, with its screen connected to the pre-amplifier's earth, should be connected to a **balanced pre-amplifier** input. This is a pre-amplifier fitted with a separating transformer at its input. Such a transformer gives a common mode rejection greater than 30 dB to the microphone signal lines and therefore cancels out any hum present on those lines.
Do not connect the cable screening wire to the metal body of the microphone connecting plug.



4. Example how to connect electrical equipment in a chain, the screening is only on the receiving side of the equipment connected to electrical earth, this is to avoid ground loops (hum).



14.2 Radio and Mains Born Interference

Philips' amplifiers and distribution systems contain extensive protection against external interference sources and, in normal circumstances, will not be effected by them. However, extraordinary radio-born and mains electricity supply conditions may cause problems which have to be solved individually.

Problems may be expected when:

- An electrical field strength exceeds 1 V/m. This would be the case when the system is installed:
  a. Within a 20 km radius of a 1 MW medium wave radio transmitter.
  b. Within a 5 km radius of a 100 kW FM or television transmitter.
  c. Within a 100m radius of a 0.5 W citizens band transmitter (e.g. 27 MHz), depending on the directivity of its antenna.
  d. Near medical equipment. For instance; within 100 m of a 27 MHz, 1 kW radio-therapy unit.

Frequencies above 200 MHz (like radar or relay connections >1 GHz) seldom cause problems.
A factor which normally decreases the interference influence, is the screening property of the building, especially when metal construction materials or reinforced concrete are used.

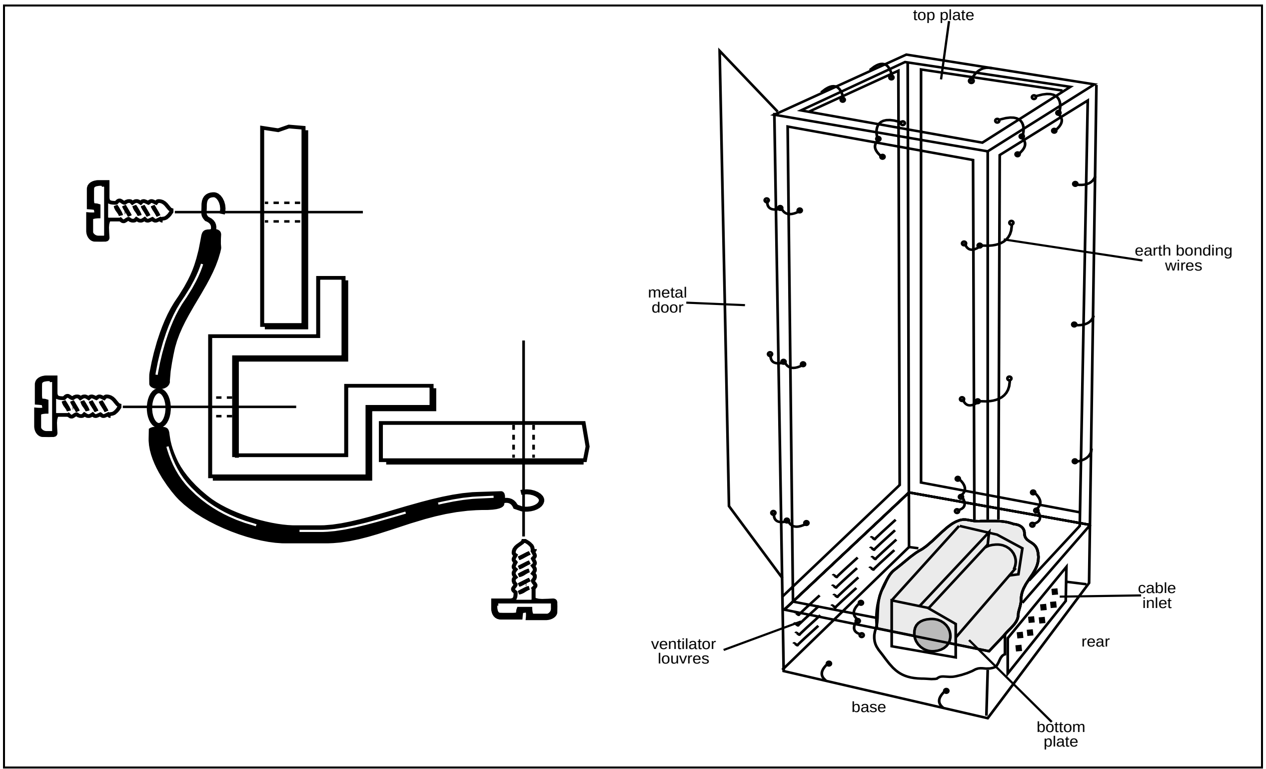
- When voltage spikes on the mains electricity supply exceed 800 V.
  This can occur when highly inductive or capacitive loads are switched on and off on the mains network.

Problems of this kind can normally be solved by installing a good mains filter. A variety of special application versions exist for this kind of situation.

14.2.1 Prevention of Interference

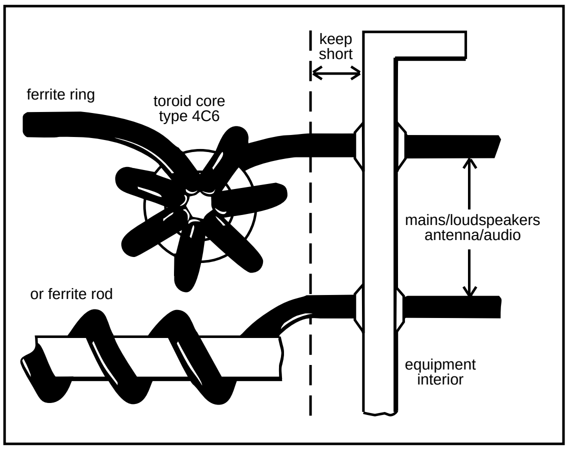
There are two basic methods of preventing radio born interference:

1. Screening the source of radiation:
   Generally the most effective method, but only feasible when the offending cause is located 'in house', as would be the case with medical equipment etc. The equipment and the patient would have to be located inside a Faraday cage, within which the 'radiation area' would be confined.
2. Screening the effected equipment:
   The 19 inch rack unit, within which the distribution system, and/or amplifiers of a sound system would be mounted, makes an ideal screen for the electronic circuitry. The rack used must have a top and bottom plate, and an all metal door. The only holes in the outside surfaces should be for ventilation, and these should be in the form of louvers or small holes, rather than one large opening. The rack can form a more efficient screen when all of the component parts (covers, construction bars, etc.) are electrically connected.



This is done by using short lengths of wire to join each part to its neighbour. On large surfaces, such as cover panels, these connections should be made in several locations (e.g. 6 wires on side/rear covers). In extreme cases it may be necessary to remove paint, and use self-tapping screws every 5-10 cm to make the cabinet 100% RF immune. See accompanying illustrations for examples.

14.2.2 Interference introduced via cables



Any cable, whether signal, loudspeaker, or mains, is a potential antenna for radio born interference.

Where feasible, disconnect and reconnect each cable in turn, until the offending cable(s) is (are) found.

The simple modification illustrated can effectively cancel this problem. Experiment with the amount of windings to find the optimum RF damping.

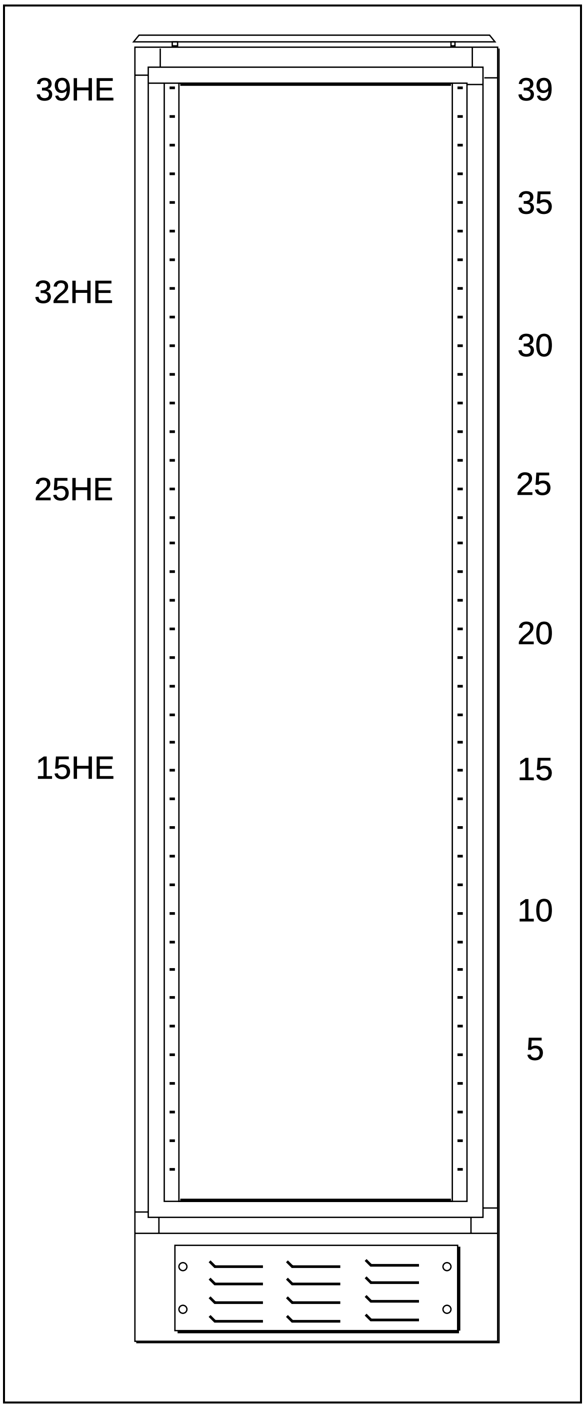
14.2.3 Interference introduced inside rack unit

In some cases, hum can be induced into a signal line from the radiation effects of mains electricity voltage cables and transformers. Care should be taken when planning the internal wiring of the rack unit, to keep input wiring, where possible, away from mains wiring and transformers.

14.2.4 Interference induced from 100 V loudspeaker wiring

Signal wiring, both inside the rack, and in external cable ducts, should be kept separate from 100 V loudspeaker wiring. If this is not done, inductive & capacitive coupling might occur, causing the system to oscillate.

14.3 NINETEEN INCH RACK UNITS



Philips public address equipment, like distribution controllers amplifiers, monitor panels and many auxiliary equipment, are designed to be mounted in a cabinet with a standard front panel width, called a "Nineteen Inch Rack". Some music source equipment (background music players, CD players, radio tuners, etc.) can be modified, using bolt on accessories, to fit also into these 19 inch racks.

To simplify calculation of 19 inch rack space required, a standard height 'HE' equal to 44.55 mm (1.75 inches) has been chosen. Most power amplifiers for instance, are 3 HE in height, requiring 133.65 mm of rack space. The use of this height standard eases the problem of calculating the number of amplifiers, modular distribution units, or panels that will fit into a given rack.

Certain rules should be observed when planning the equipment layout of the rack.

1. Cassette front loaders, tuner scales, and other frequently used equipment should be mounted at a height which makes the front panel clearly visible to the operator.

2. If power amplifiers are mounted beneath rack frames containing microprocessor controlled distribution units, a heat shield should be installed above them. This is necessary to deflect hot air currents, which could otherwise cause instability in the microprocessor units.

3. When several high power amplifiers are used, a fan unit should be mounted in the bottom of the rack to ensure adequate ventilation.

Loudspeakers

The loudspeakers used in the audio reproduction chain are a vital factor in determining the overall quality and success of a sound system. Because if this, it is vital to understand the different types of loudspeakers available, and their particular strengths and weaknesses.
Philips offer a wide range of loudspeakers in their product range, but all are designed and rigorously tested to reproduce speech clearly, and to provide a very high level of reliability.

15.0 Loudspeakers

15.1 Loudspeaker Types

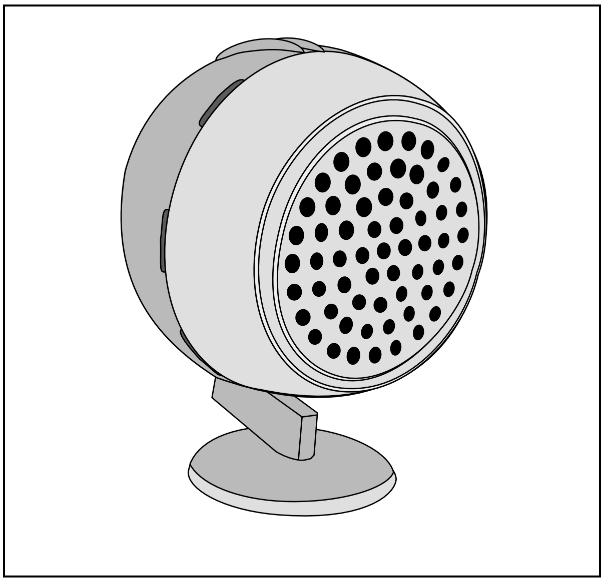
Cone loudspeakers are the most commonly used units, which in order to function properly must be mounted in correctly designed enclosures (cabinets or boxes). Dependent on the enclosures in which they are mounted, the characteristic of handling a wide frequency range makes them particularly suitable for the reproduction of music and speech. Loudspeakers with larger cone diameters generally give better low frequency reproduction. The fact that they are less efficient, and do not produce a high SPL, compared with diaphragm (horn) type loudspeakers, limits their use in areas of high ambient noise, or where the loudspeaker must be mounted a great distance from the listeners. The following units are based around cone loudspeakers:

15.1.1 Standard loudspeaker cabinets

Standard (infinite baffle) loudspeaker cabinets, are in principle a sealed box containing 1 cone loudspeaker, and have a typically wide dispersion pattern. Their shape makes them convenient for mounting on walls or pillars, or suspending vertically from the ceiling, to give a wide beam of sound.

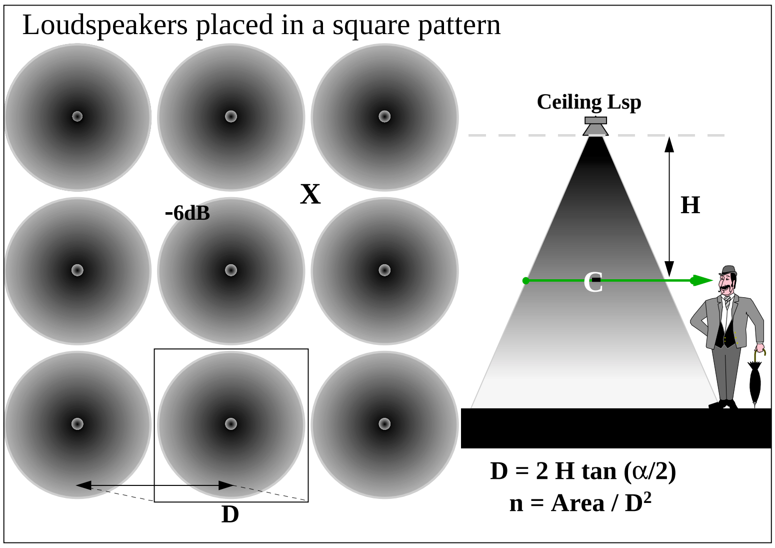
The bass response of the sealed enclosure is very much dependent on its inside volume. Normally, a large sealed enclosure will provide better bass response than a small one. In high quality reinforcement and Hi-Fi installations, enclosures are "tuned" to the resonant frequency of the (bass) loudspeaker, often by building in a bass opening or elongated port having very critical dimensions.

One versatile, and popular version, is the Philips **LBC 3003**, a cylindrical ABS plastic enclosure which ideal for mounting indoor, as well as in outdoor environments (splash-waterproof version). Where higher SPL and more directional control are required, these units may be mounted together in a column configuration.



The Philips **Cardioid Sound Projector LBC3002** is a single cone loudspeaker, mounted in a spherical ABS enclosure, designed specifically for the reproduction of speech. Whereas normal cone loudspeakers produce a very wide beam of sound at low frequencies and a narrower beam at high frequencies, due to its unique inbuilt acoustic filtering slots, the Cardioid Sound Projector produces a well defined beam of sound over all frequencies.

15.1.2 Ceiling loudspeakers



A ceiling loudspeaker is a cone loudspeaker, mounted on a front panel, which may be recessed into a ceiling or hollow wall. They can be spaced at regular intervals to give a fairly even coverage of sound.

A common used calculation-method leads to the mutual distance between the speakers:
$D = 2\,H \tan(\alpha/2)$
(H = Ceiling height to Ear height
and α = opening angle at 4 kHz)

And the total number of the speakers:
$n = Area / D^2$

The accompanying tabel shows the level variations which can be expected for different opening angles.

Opening Angle at 4 kHz = 60 ⁰ Level variation = 4,5 dB

Ceiling Height	in m	3	3,5	4	4,5	5	5,5	6
Mutual Distance D	in m	1,7	2,3	2,9	3,5	4	4,6	5,2
Covered Area	in m^2	3	5,3	8,3	12	16	21	27

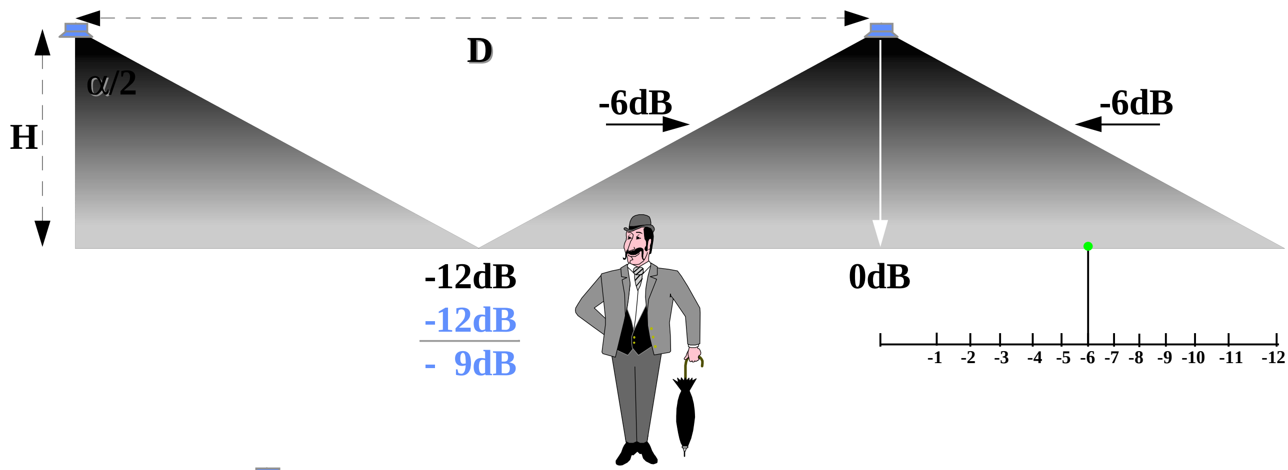
Opening Angle at 4 kHz = 90 ⁰ Level variation = 5 dB

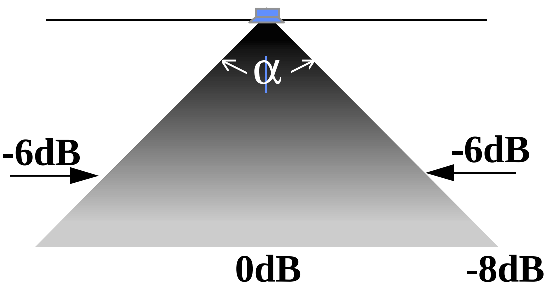
Ceiling Height	in m	3	3,5	4	4,5	5	5,5	6
Mutual Distance D	in m	3	4	5	6	7	8	9
Covered Area	in m^2	9	16	25	36	49	64	81

Opening Angle at 4 kHz = 120 ⁰ Level variation = 7 dB

Ceiling Height	in m	3	3,5	4	4,5	5	5,5	6
Mutual Distance D	in m	5,5	7	9	10,5	12	14	16
Covered Area	in m^2	30	49	81	110	144	196	256

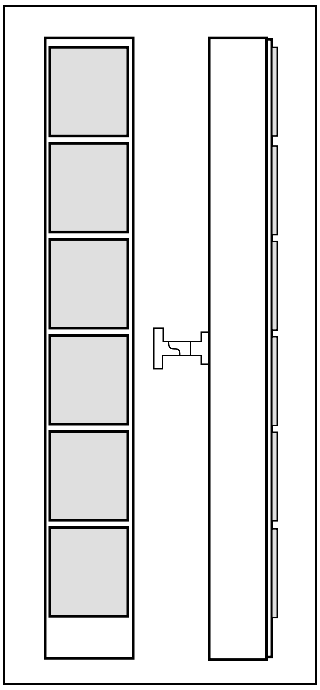
Note:
a: Caution should be taken when mounting these units in particularly high ceilings (> 5 meters) and in noisy environments. The level of sound reaching the listeners may be unacceptably low, due to the distance involved, and the limited maximum SPL available from the units.
b: It is difficult to obtain good results from a ceiling loudspeaker system in rooms with a reverberation time of more than 2 seconds (see chapter 18 for indoor acoustics).





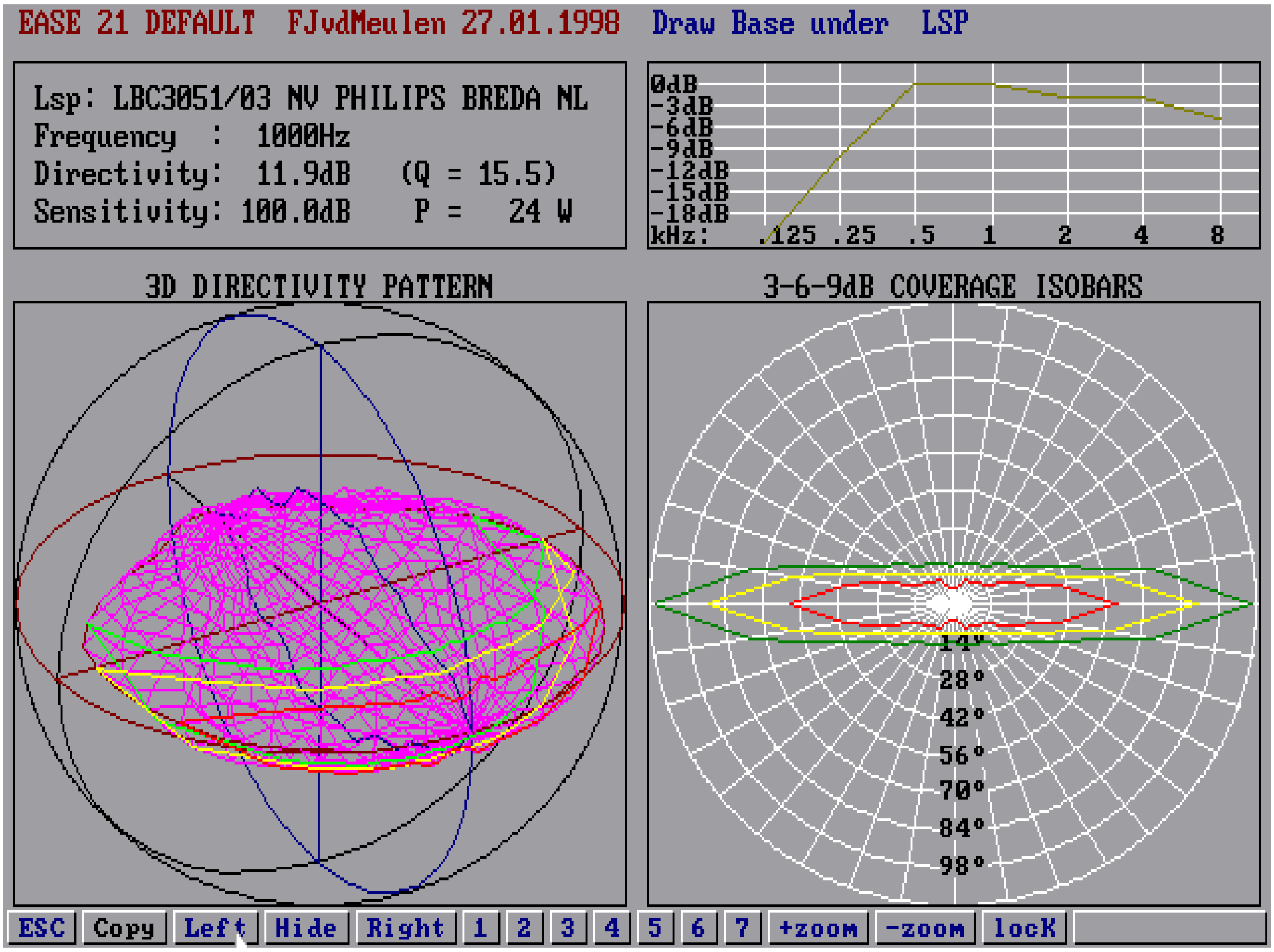
The level variation for wide opening angles however is due to the extra distance attenuation more than for small opening angles.
A special ceiling speaker program CSP calculates the actual levels under and between the speakers, and can be set for 6-5-4-3-2-1 dB variation. For every Philips ceiling speaker type the required number of speakers and mutual distance is calculated.

15.1.3 Sound columns



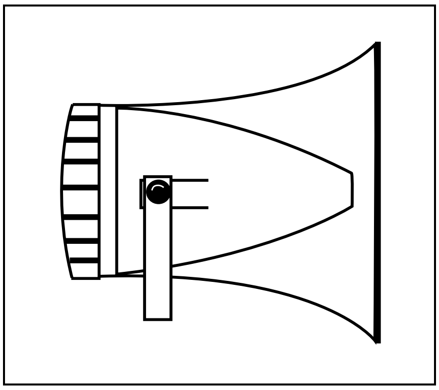
Sound columns are a group of (usually 4 to 10) loudspeakers mounted close together in a vertical array. Due to an interesting acoustical phenomenon, though the beam of sound emitted horizontally is approximately the same as a normal cone loudspeaker, the beam of sound emitted vertically is narrow (10-15^0) and therefore very directional, especially at higher frequencies. Column loudspeakers are particularly useful in situations where a great degree of control is required over the vertical spread of sound, and no spill of sound is acoustically allowed. A typical instance would be in reverberant environments (e.g. churches) where it is desirable to beam the sound down onto the listeners, without it reflecting off hard walls and ceilings.

Unfortunately the bass frequencies are less directional than the higher frequencies, and spread much wider than the useful loudspeaker opening angle. In reverberant environments this wide spread of low frequencies can excite a reverberant field, causing great problems with intelligibility. In situations where the microphone is in the same room as the loudspeakers, this can also cause acoustic feedback. This can be overcome by the use of equalisation (described in chapter 10), reducing the volume of the bass frequencies in the signal. Though this is acceptable for speech purposes, it would have an adverse affect on the quality of music reproduction, so care should be taken not to completely eliminate the bass content of the signal if music is to be amplified.



A more suitable alternative would be the use of Philips **Cardioid Loudspeaker Columns (e.g. LBC3051)**. Using the same principle as the Philips Cardioid Sound Projector, the beam of sound at low frequencies is tightly controlled, making it very similar to that of higher frequencies. This makes the unit ideal for environments with difficult acoustics, and gives the designer a greater degree of predictability when calculating intelligibility.

15.1.4 Horn loudspeakers

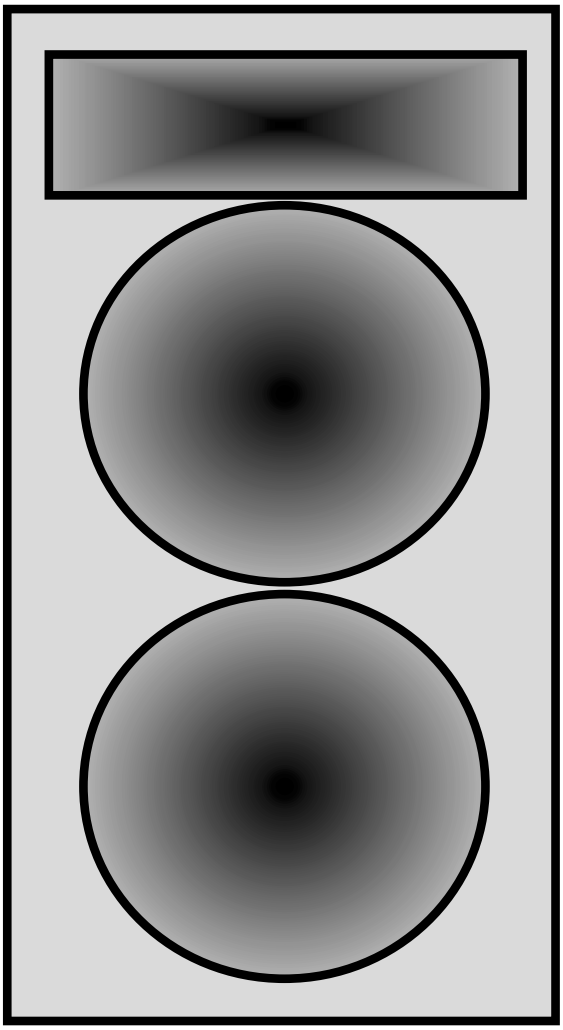


Horn (or 'diaphragm') loudspeakers, are different to cone loudspeakers in that the sound produced is generated by a small, thin metal diaphragm, and amplified by the shape and size of a folded horn. They produce a very powerful, concentrated, beam of sound enabling them to reach listeners at a great distance.

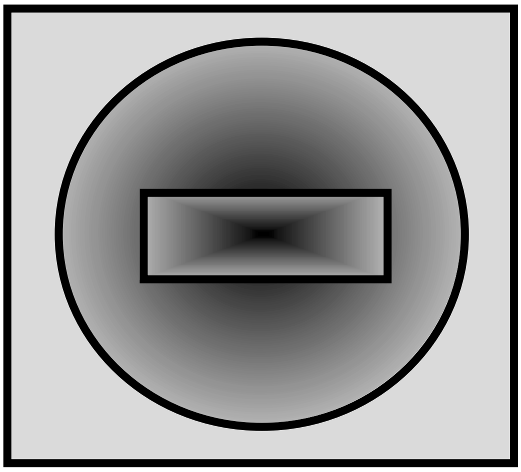
Because the diaphragms are normally mounted in moulded plastic or metal folded horn enclosures, they can be easily rendered weatherproof, which allows them to be used outdoors and in dusty and humid environments. They may be mounted on masts or higher buildings and/or arrayed in a column to produce a directional vertical beam.

The diaphragm loudspeakers used in public address installations have the limitation of having a fairly restricted frequency range, giving a diminished output at low frequencies due to the diameter of the horn and at high frequencies due to folding of the horn. This makes them generally unsuitable for satisfactory music reproduction, but can to some degree be compensated for by combining them with cone loudspeakers.

15.1.5 Full range high power loudspeakers

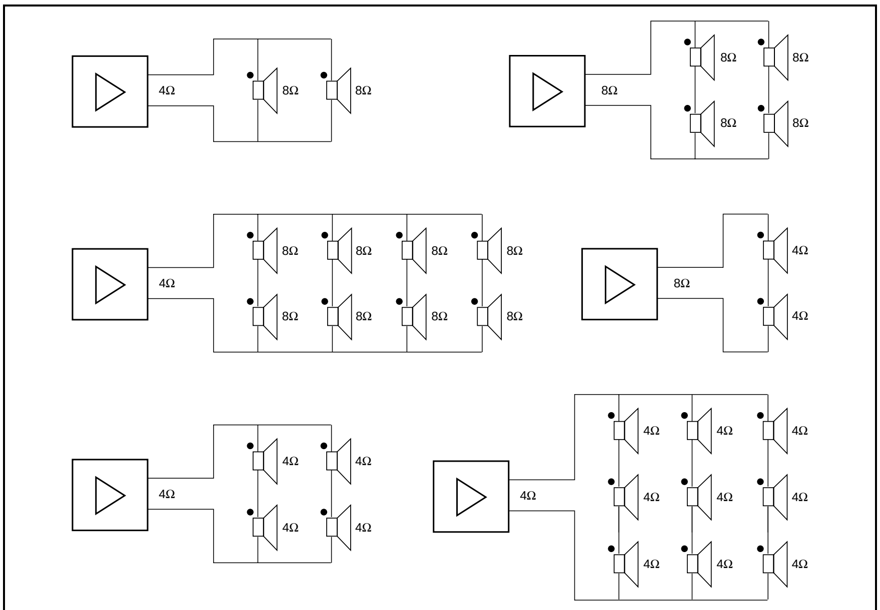


Loudspeakers with diaphragms mounted directly onto the mouth of an exponential horn are often used as the treble component of a "full range" multiple loudspeaker enclosure. The audio signal is fed through a suitable crossover filter which eliminates the bass content, which could damage the diaphragm. These enclosures, often grouped together in a cluster, are used in installations to produce full range high power sound.



Combining the horn in the centre of the woofer loudspeaker has the advantage of a compact stackable or arrayable unit.

15.2 Matching Loudspeakers to Amplifiers



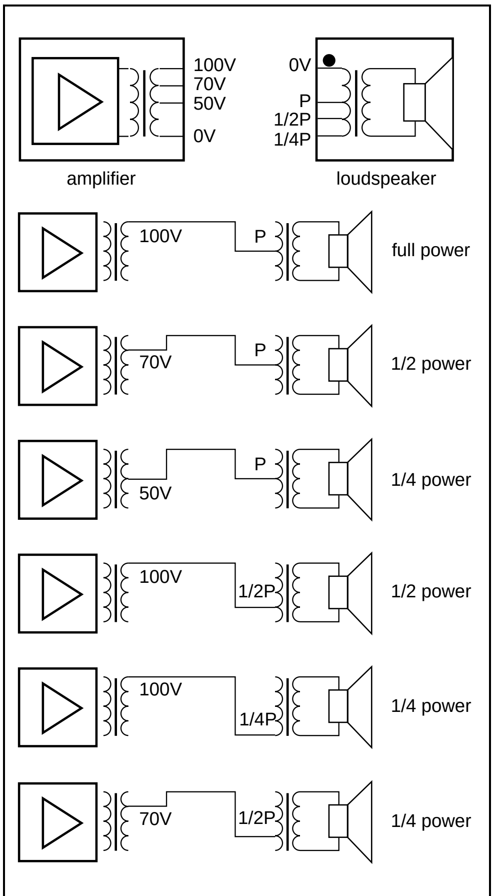
matching loudspeakers to amplifier low impedance output

Two systems are available for connecting loudspeakers to amplifiers:
-The direct low impedance system
and
-The 100 Volt Line Matching System (which is normally used in public address emergency & announcement systems).

The loudspeakers could be connected in a series/parallel arrangement, as illustrated, to exactly match the amplifier's low output impedance.

This is only a feasable solution if the power leads to the loudspeakers are reasonable short, otherwise line losses are considerably.

If the loudspeakers differ in power and impedance, it is very difficult indeed to match them to the power amplifier. In this type of situation, or in an application requiring long loudspeaker cable lengths (e.g. public address systems), the 100 Volt line system should be used.



When loudspeakers are connected to the 100V tap on the amplifier's line matching transformer, their full power is used, whereas if they are connected to the 70V tap, only 1/2 of their rated power is used. This means that the 70V tap enables the amplifier to power twice as many loudspeakers, with each loudspeaker producing 1/2 of its potential power. Similarly, the 50V tap allows loudspeakers to use 1/4 of their rated power, so that the amplifier is able to power 4 times more loudspeakers, with each producing 1/4 of its potential power.

The transformers fitted to loudspeakers have similar taps, but in this case the actual power which the loudspeaker will draw (e.g. P, P1/2, P1/4, or 6W, 3W, 1,5W), instead of the voltage, is printed beside each power (+) tap. A reduced loudspeaker volume can be set by using these taps. For instance if the same type of loudspeakers are powered from a common amplifier, and it is desired to have one of them producing less volume than the others, then it is a simple matter of connecting the signal to either the 1/2 or the 1/4 power (+) tap. This would reduce the output of the loudspeaker by 3dB or 6dB respectively.

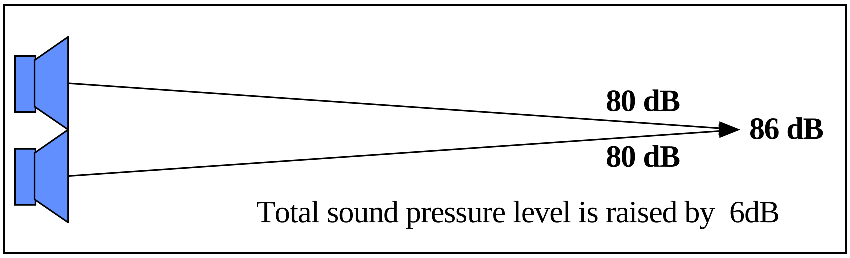
Note:
When using the 100 Volt line matching system, the Rated Power of the amplifier corresponds to the Rated Load Impedance of the loudspeaker network. The total rated power required should be calculated, by simply adding the Rated Power of the connected loudspeakers together, taking into account the difference in power drawn when using the loudspeaker power taps. **It is important that this total should not exceed the rated power of the amplifier.**

Loudspeakers in the Philips product range are manufactured with a Power Handling Capacity (PHC) according to the IEC268-5 standard. These loudspeakers are actually capable of withstanding power input greater than the PHC, which enables them to avoid damage during times of excessive signal overload (acoustic feedback!)

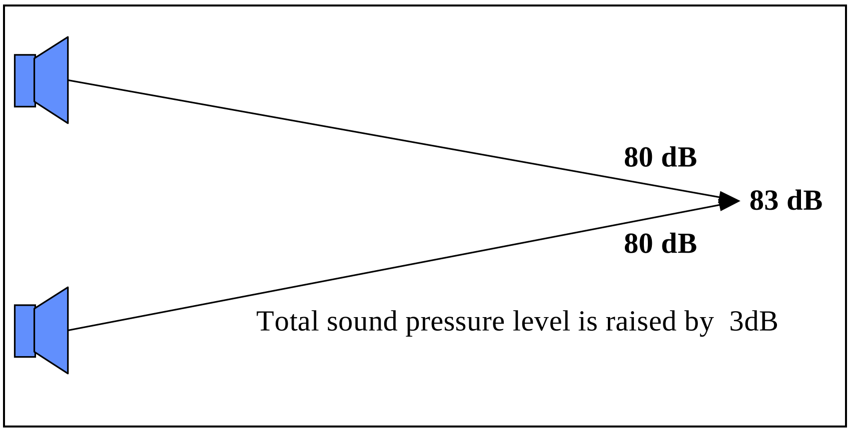
16.0 Technical Principles

16.1 BASIC PRINCIPLES

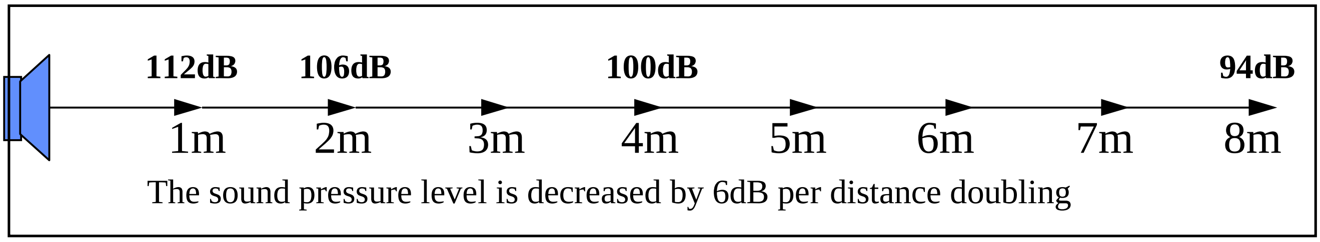
- Loudspeaker power handling capacity is measured in watts (W). A 6W loudspeaker would be able to accept a maximum of 6 watts from a power amplifier.
- The 'sensitivity' of a loudspeaker is the Sound Pressure Level (SPL), expressed in dB, at 1 kHz, measured at a distance of 1 meter, on an axis with its centre, when it has an input of 1 watt.
- Each time the input power of a loudspeaker is doubled, the SPL rises by 3 dB. Therefore if we know the sensitivity of a loudspeaker, it is a simple matter to calculate its SPL at any given power input. E.g.: If a loudspeaker has a sensitivity of 99 dB (1W/1m), 2W would raise the SPL by 3 dB, to 102 dB; 4W would increase it to 105 dB; etc., until it reaches maximum rated power.



- If 2 loudspeakers are placed side by side and given the same input signal (so that both are in phase, operating as one unit) the SPL at the listeners would be 6 dB more than the SPL of a single speaker. Each time the quantity of loudspeakers is doubled, the SPL is increased by 6dB.



- If those same loudspeakers are placed some distance away from each other (even so small a distance as 1 meter), there will always be a shift in phase at the ears of the majority of the listeners.
  This causes the total SPL to increase by only 3 dB, instead of 6dB, each time the quantity of loudspeakers is doubled.

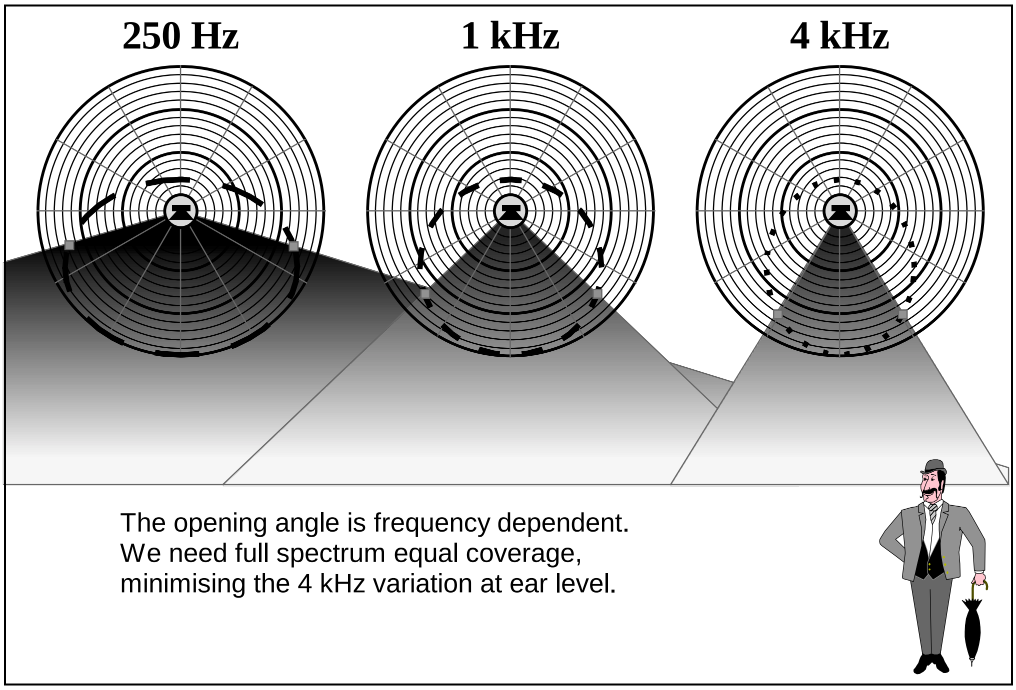


- As we move further away from the sound source the SPL drops. Again a simple rule is in force; Each time the distance from the loudspeaker is doubled, the SPL drops by 6 dB.

  For instance, assuming that we have a loudspeaker cabinet producing 112 dB at 1 meter; at 2 meters distance the SPL would be 106 dB; at 4 meters 100 dB etc.

  This rule only deals with direct sound, not taking into consideration any (indirect)sound returned from reflective surfaces.

  That problem is dealt with separately in chapter 18.



The opening angle is frequency dependent.
We need full spectrum equal coverage,
minimising the 4 kHz variation at ear level.

• All of these examples so far have dealt with loudspeakers producing a 1000 Hz tone, being measured in line with the loudspeaker's axis. By looking at the polar diagram, we can see that the SPL differs depending on the frequency being transmitted, and at what angle the listener is relative to the axis (0^o). This effect will be used in the formula for the direct sound (L_Q.) see chapter 17.

The number of degrees between the points where L_Q = 6 dB is the opening angle normally defined for 4 kHz for clarity reasons. In the polar diagrams, this is indicated with a grey shading. The opening angle upto 4 kHz is vital for the intelligibility & clarity reasons.

16.2 Detailed Considerations

16.2.1 Resonant frequency

At the resonant frequency the impedance is very high in relation to the average impedance. This varies from cone loudspeakers (20 Hz to 300 Hz) to horn drivers (200 Hz to 1 kHz). The 'nominal impedance' is the impedance of the lowest part of the curve above the resonant frequency (f_0) - usually around 400 Hz. Damage can occur to the loudspeaker if power is sustained at the resonant frequency. So where continuous alarm signals are required, care should be taken to ensure that the frequency of the signal is well above the resonant frequency of the loudspeakers used.

16.2.2 Sensitivity

The sensitivity level of a loudspeaker is the loudness expressed in dB (SPL) at 1 kHz and at a distance, on axis, of 1 m with an input of 1 W. The importance of this figure may be illustrated by examining the effect of varying the two main parameters, namely, distance and power.
Because the efficiency of loudspeakers, horn drivers, columns, etc., vary so much, it is impossible to define the number of loudspeakers required for a room (and the amplifiers required to drive them) without first calculating.

Assume that it is required to produce a SPL of 80 dB at a distance of 32 m. To calculate the required power for the loudspeaker (For simplicity an outdoor situation is chosen):

Reduction in acoustic level due to distance = 20 Log 32 = 30 dB

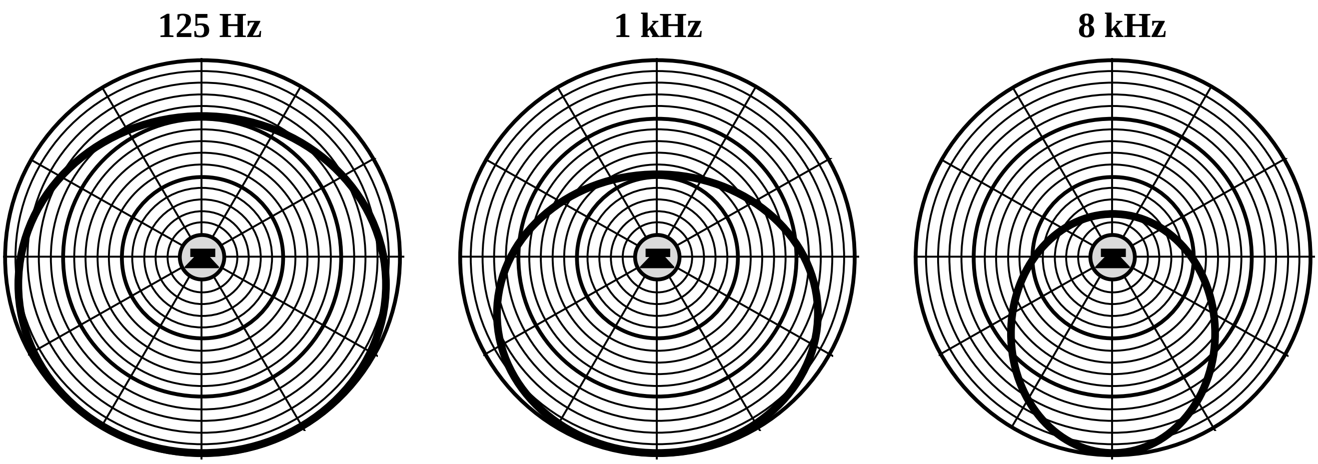
To compensate for this reduction, 80 + 30 = 110 dB (SPL) is required at 1 m distance from the loudspeaker.
If the loudspeaker has a sensitivity of 100 dB (SPL) the missing 10 dB should be compensated for with an extra 10W power applied to the loudspeaker.

16.2.3 Efficiency

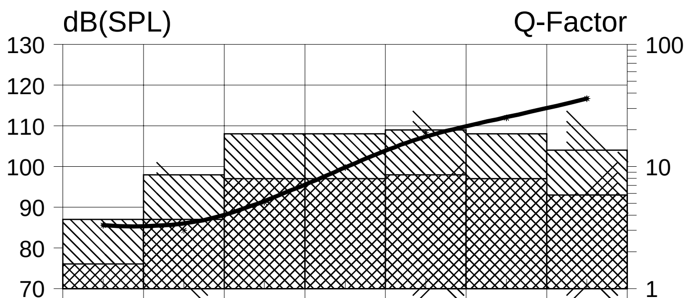
A loudspeaker's ability to convert electrical energy into acoustical energy is defined as its efficiency, and can be stated as a percentage figure (values between 0.5-10%). This value is required for calculations of the reverberant sound field. See chapter 18 for details. Because this varies with frequency, Philips specifies the loudspeaker's efficiency per octave band, on the technical documentation.

16.2.4 Directivity (Q)

The directivity factor (Q) of a loudspeaker is the ratio of the mean squared sound pressure level at a fixed distance, measured on axis (which is normally the direction of maximum response), to the mean squared sound pressure level at the same distance, averaged over all directions. Q is therefore a measure of the response of the loudspeaker in a three dimensional plane.



At low frequencies the radiation of a loudspeaker has a spherical form which becomes more directional as the frequency increases. This indicates that Q is frequency dependent. Since readings are normally taken in 10° intervals in a sphere, for each of seven octave bands, this requires the processing of more than 2000 readings. Because of this only a few of the leading manufacturers actually quote figures for the directivity factor.



	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz
1W, 1m(dB)	76	87	97	97	98	97	93
Max,1m(dB)	87	98	108	108	109	108	104
Q-Factor	3.3	3	5.1	9.8	19	25	36
Effic. (%)	0.02	0.21	1.24	0.64	0.42	0.25	0.07
Hor. Angle	180	180	180	180	140	110	80
Vert.Angle			70	40	20	12	8

Standard format for loudspeaker technical specifications,

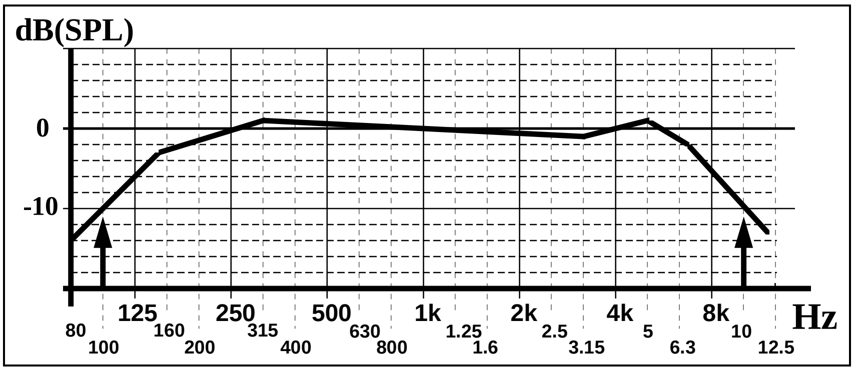
showing average performance for each of the seven octave bands.

Typical average Q values are:

Loudspeaker in sealed (infinite baffle) enclosure	: 2
Average male human speaker	: 2.5
Column Loudspeaker	: 7
Cardioid Column Loudspeaker	: 20

These specifications are measured in an anechoic room following the procedures defined below:

1. The **frequency response** is measured on axis (0^o) at 5 metres and calculated to 1 metre:
   - with "slow" damping using a gliding tone and/or a 1/3 octave warble (woble) tone.
   - in 7 octaves, using a stepped pink noise measurement signal.



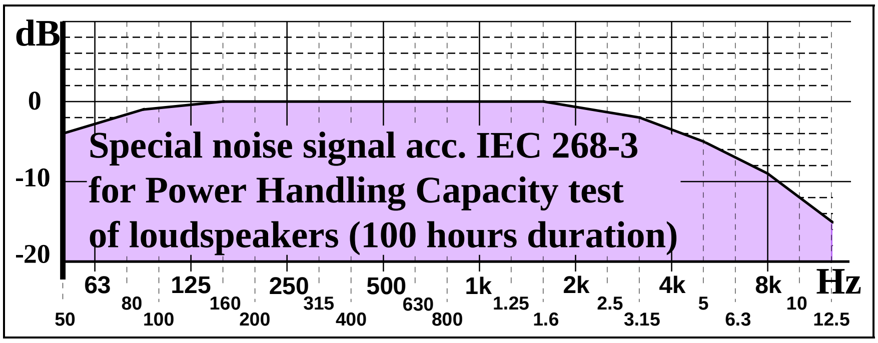
The **effective frequency range** is defined as being the range between those points at which the level drops by 10 dB.

2. For enclosures with single loudspeakers, **polar diagrams** are measured using pink noise. This is done in octave steps with centres at 125 Hz, 250 Hz, 500 Hz, 1000 Hz, 2000 Hz, 4000 Hz, and 8000 Hz. For enclosures with asymmetrical or multiple loudspeakers the **directivity balloon** is measured using a pan and tilt device. The definition is every 10^0 for all 7 octave bands.

3. Using the measurements in 1. and 2., the software package EASE is used to calculate the **"Q"** and **"Efficiency"** values for all relevant octave bands.

4. Using the measurements in 2., the horizontal and vertical **opening angles** (-6 dB) are determined for all relevant octave bands.

5. The **Power Handling Capacity** is determined by applying the IEC pink noise test signal shown below for 100 hours. After this test the loudspeaker should still be able to perform according its specification.



The Acoustic Environment

The characteristics of sound and the way it is transmitted are very much altered by the environment in which it is generated. The same audio signal would sound quite different in a sports stadium as compared to a large reverberant church or to a heavily damped lecture room.

In general, it is possible to differentiate between two situations: the outdoor and the indoor environment.

In both situations though we are striving primarily at:
1. Speech Intelligibility - delivering the message to the ears of the listener clearly.
2. Quality of Reproduction - delivering e.g. music to the ears of the listener as unchanged as possible.

17.0 Outdoors

In the outdoor environment several factors must be considered which influence sound reproduction and reception:

- Sensitivity
- Power
- Directivity
- Distance
- Reflection
- Absorption
- Refraction
- Air absorption
- Humidity
- Temperature
- Echoes

17.1 Technical Considerations

17.1.1 Power

Power	dB (SPL)
1 W	100 dB
2 W	103 dB
4 W	106 dB
8 W	109 dB
16 W	112 dB
32 W	115 dB

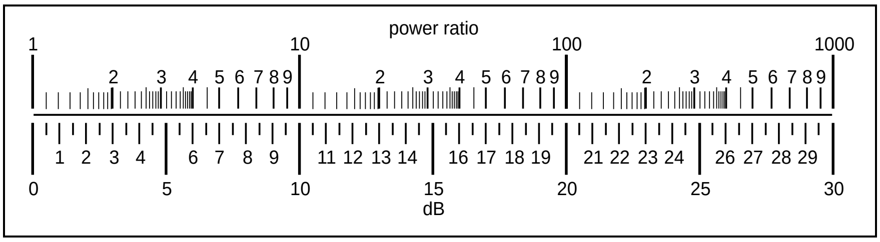
Each time the input power of a loudspeaker is doubled, the SPL rises by 3 dB. The effect, at a distance of 1m, is shown in the table, which lists the increase in SPL with doubling of power, from a nominal value of 100 dB (SPL)

Intermediate powers may be accounted for by: dB (SPL) at measured power $= SPL_{1.1} + 10 \text{ Log } P/P_0$
where:

$SPL_{1.1}$ = sensitivity of loudspeaker in dB (SPL) for 1 watt at 1 meter
P = power (W)
P_0 = reference power (1W)

Using our reference of 100 dB(SPL), for an increase of 12 W the calculation is:

$100 + 10 \text{ Log } 12 = 100 + 10.8 = 110.8$ dB (SPL)

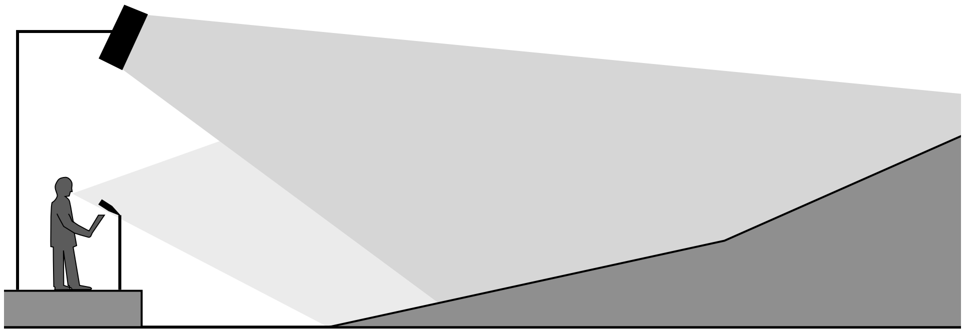


This SPL increase of 10.8 dB for a power increase of 12 W can also be seen in the accompanying table.

17.1.2 Directivity

Before attempting to calculate coverage, it is necessary to know a little about the different characteristics of certain types of loudspeakers. One of the fundamental differences in loudspeaker types is their 'opening angle'. This is the dispersion (measured as an angle) of sound which radiates from the front of the speaker.
Dependent upon the environment and the particular application needs, it may be necessary to use loudspeakers with a wide opening angle, which disperse (spread) their sound over a wide area.
Alternatively it may be necessary to concentrate a beam of sound in a particular direction. This would be important where an unnecessarily wide spread of sound is not only wasteful in amplifier energy, but could reflect off nearby buildings, or disturb people in neighbouring areas. This is particularly vital when the microphone is also outdoors, and exposed to sound coming from the loudspeakers. An uncontrolled spread of sound could return a large amount of the audio signal into the microphone, which will be amplified again, causing acoustic feedback or howl. Take care to place the loudspeakers in such a position that there is a "quiet" area around the microphone location, if possible with the loudspeakers in front of, and pointing away from, the microphone.

Even though certain types of loudspeakers produce a fairly wide spread of sound, by grouping several of them in a vertical configuration, commonly called a column, the shape of the total beam of sound can be altered to make it more directional. This is discussed in greater detail in 15.1.3.



In installations with low output level loudspeakers, mounted along the length of an area, spacing the loudspeakers less than 15 meters apart will help minimise echo. See 10.0 for details of using a delay line in this type of situation.

17.1.3 Attenuation due to Distance

When sound is reproduced in an outdoor situation, without any objects to cause reflection, the listener hears only direct radiation. The sound pressure level drops by 6 dB(SPL) each time the distance is doubled.
The table below shows SPL decrease with the doubling of distance, from a nominal value of 100 dB(SPL)

Distance	dB(SPL)
1 m	100 dB
2 m	94 dB
4 m	88 dB
8 m	82 dB
16 m	76 dB
32 m	70 dB

Assume that a loudspeaker source has a sensitivity ($SPL_{1.1}$) of 100 dB(SPL). An input of 1 W gives the following results:

For intermediate distances:

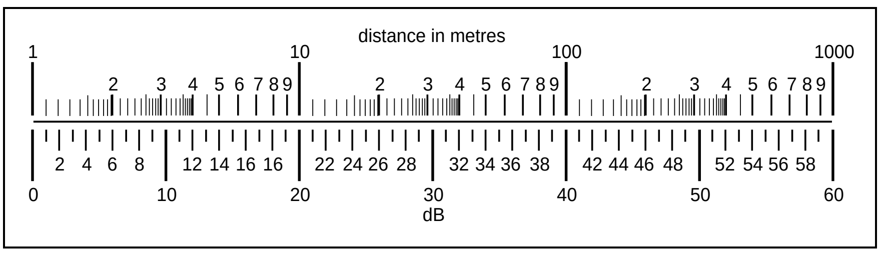
dB(SPL) at measured distance = $SPL_{1.1} - 20 \text{ Log } r/r_0$

where:

$SPL_{1.1}$ = sensitivity of loudspeaker in dB(SPL) 1W;1m
r = measured distance (m)
r_0 = reference distance (1m)

Using the nominal value of 100 dB, the calculation of the SPL at 25 metres is:

100 - 20 log 25 = 100 - 28 = 72 dB(SPL)



The SPL decrease of 28 dB at a distance of 25 metres can also be seen in the accompanying table.

17.1.4 Variations of both distance and power

Assume that a loudspeaker has a sensitivity of 100 dB. To calculate the dB(SPL) at 26 m with an input of 10W:

At 26 m the loss in dB(SPL)	= 20 log 26	= 28.3 dB
And at 10 W, gain in dB(SPL)	= 10 log 10	= 10 dB

The total effect of both variations is simply their algebraic addition:

100 - 28.3 + 10 = 81.7 dB(SPL)

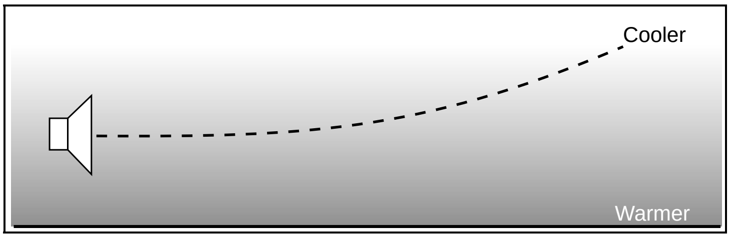
Generally we can calculate as follows:

$$L_{dir} = L_s + 10\,\mathrm{Log}(P_{el}) - L_Q - 20\,\mathrm{Log}(r)$$

$L_S = SPL_{1.1}$ = SPL value for 1W at 1m on axis.
P_{el} = power consumption of loudspeakers (W)
L_Q = on/off axis level difference
r = distance from the source

17.1.5 Refraction

Refraction, or bending, occurs when sound passes from one medium to another. This effect is also noticeable when sound passes through layers of air which have different temperatures and thus different sound velocities.



The illustration shows the effect of refraction, causing sound to bend upwards.

17.1.6 Reflection

Although the effect of reflection is mainly of concern in an indoor situation, reflections from buildings outdoors give distinct and very disturbing echoes. If the time delay between the original sound and the reflected sound is more than 50 ms, the listener will be able to hear, and to recognise, a reflected sound as a whole "echo" of the original. Knowing the speed of sound in air to be 340 m/s, then the time difference of 50 ms is equivalent to a distance of 17 m. If the difference between the direct and the indirect distances is significantly shorter than 17 m, the reflected sound will have the effect of reinforcing the direct sound, rather than causing an echo.

17.1.7 Ambient Noise

The perceived quality from a sound reinforcement and/or public address distribution system can be particularly effected by ambient noise. The constant sound of passing traffic, the rumble of heavy industry or even the hum of conversation from a large crowd, can create a significant ambient noise level, which must be compensated for.

When the sound level of a source is being measured in a situation where ambient noise is present, it is necessary to subtract the ambient noise level reading from the combined (total) reading in order to find the actual level of the source alone. If this is not done it is not possible to measure the source level accurately.

This is calculated by:

$$L_S = 10\,\mathrm{Log}\,[10^{L1/10} - 10^{L2/10}]$$

where: L_1 is the reading taken of the source and the noise combined (e.g. 60 dB(SPL)) and,
L_2 is the reading of the noise alone, with the source shut off (e.g. 55 dB(SPL)).

In this example the level of the source is:

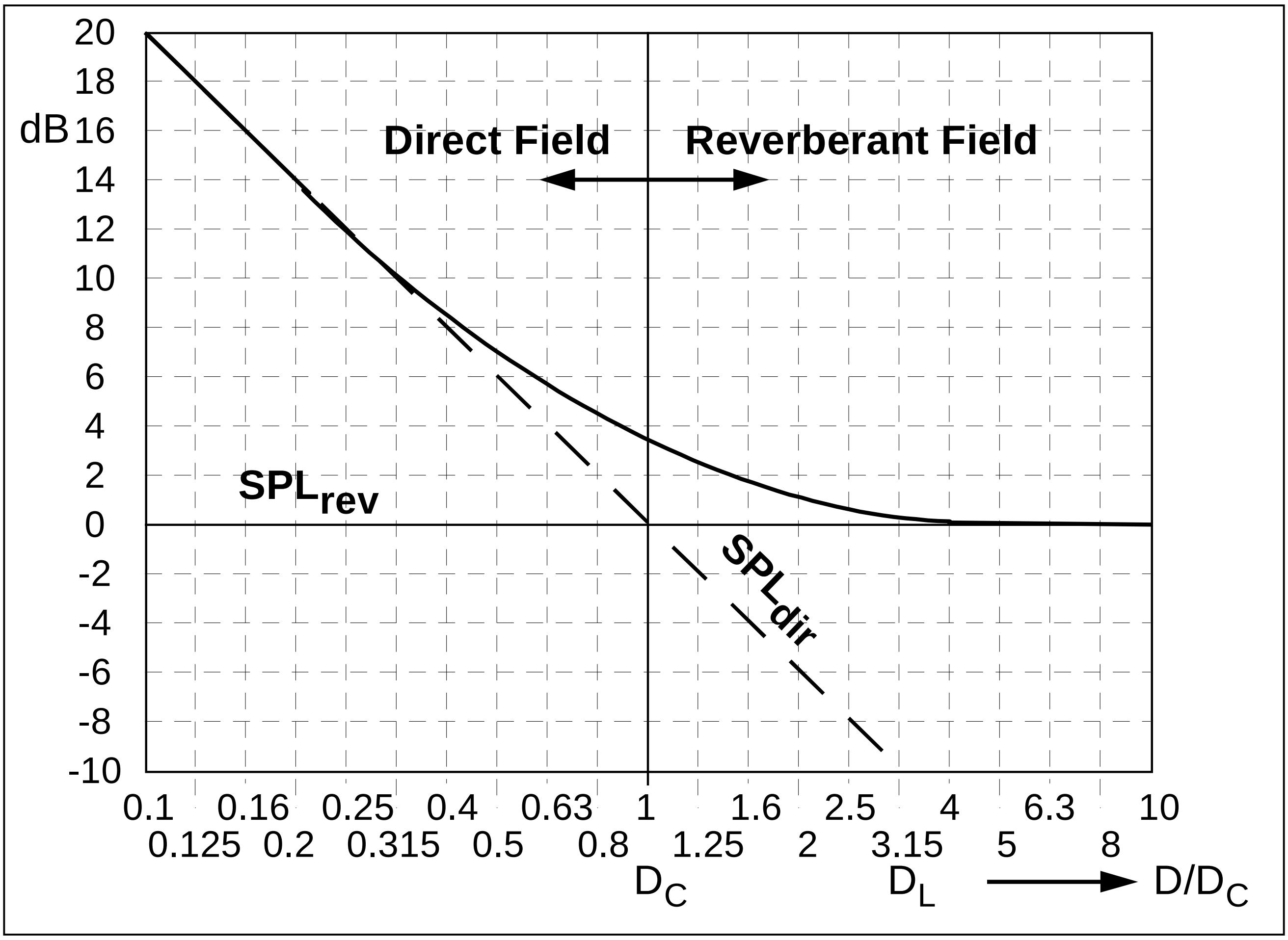
$$L_S = 10\ \log\ [10^6 - 10^{5,5}] \quad = 58,3\ \mathrm{dB\ (SPL)}$$

18.0 Indoors

18.1 TECHNICAL CONSIDERATIONS

When designing a sound system for indoors, the situation is made difficult by a number of problems which must be taken into consideration.
Because the listener is often seated some distance from the source of the sound, high frequency signals are absorbed by the air, while the lower signals activate reverberation as they bounce off hard walls and ceilings. This means that, in a reverberant environment, with increasing distance, we encounter two problems at the listeners:

- A decreasing original (direct) speech spectrum (SPL_{dir}), discussed in 17.1.3.
- An reverberant low toned indirect/reflected speech spectrum (SPL_{rev}). This means that the listeners may hear everything loudly, but the consonants in the speech are hidden or masked by the reverberation, causing low speech intelligibility, so that they cannot understand what is being said.



18.1.1 Reflection & Absorption

When a sound source is in a room and enclosed e.g. by walls and a ceiling, these surfaces will partly reflect and partly absorb the sound. The intensity of the reflected sound wave (I_{ref}) is smaller than the incident one (I_{inc}), a fraction α of the incident energy is lost during reflection, or:

$I_{ref} = (1-\alpha)\ I_{inc}$ α is called absorption coefficient

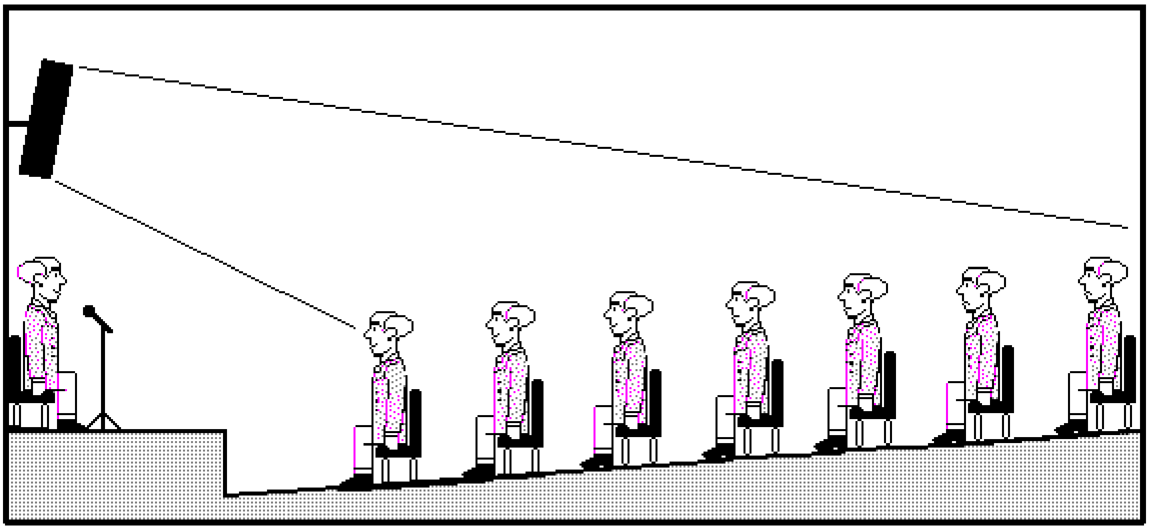
Most of the building materials have measured absorption coefficients (α) and reflection coefficients (r).

$\alpha + r = 1.$

If all the sound is reflected (r = 1), no sound is absorbed by the material (α = 0).

The list with absorption coefficients is provided (see appendix) for a selection of materials; a higher figure per octave band = greater absorption. As can be seen, soft materials generally have more effect on higher frequencies.

18.1.2 Reverberation



If sound is generated in a room, part will travel directly to the listener; more will arrive after having been reflected, and still more after successive reflections.

The effect of these repeated reflections is called reverberation, which leads to the build-up of diffuse sound throughout the room, called the reverberant field.

The actual level of the reverberant field is determined by three factors:
- the nature of the sound source
- the physical volume of the room
- the reverberation time.

18.1.3 Reverberation time

The reverberation time (T) of a room is a measure of the time taken for the sound level of the reverberant field to fall by 60 dB. The following points regarding reverberation time are assumed:
- the reverberation time in a room is the same whatever the position of the sound source;
- the reverberation time in a room is the same wherever the listener happens to be;
- the lack of intelligibility in a room is almost always due to a long reverberation time;
- reverberation time is determined by the room volume, and total amount of sound absorption in it.

The reverberation time according to Sabine:

$$T = 0{,}161 \text{ Volume / Absorption}$$

The absorption : $A = \bar{\alpha} S + 4mV + nA_P \qquad \bar{\alpha} S = \sum (S_i \alpha_i)$

Thus: $T = 0{,}161\, V/(\bar{\alpha} S + 4mV + nA_P)$

where:
- V = total volume of the room (m^3)
- A = total absorption (m^2 or Sabine)
- S = total surface area (m^2)
- S_i = surface area (m^2)
- m = atmospheric absorption (attenuation constant) see chapter 1,4 for details.
- α_i = absorption coefficient
- $\bar{\alpha}$ = average absorption coefficient
- n = number of persons
- A_P = absorption per person (m^2 or Sabine)

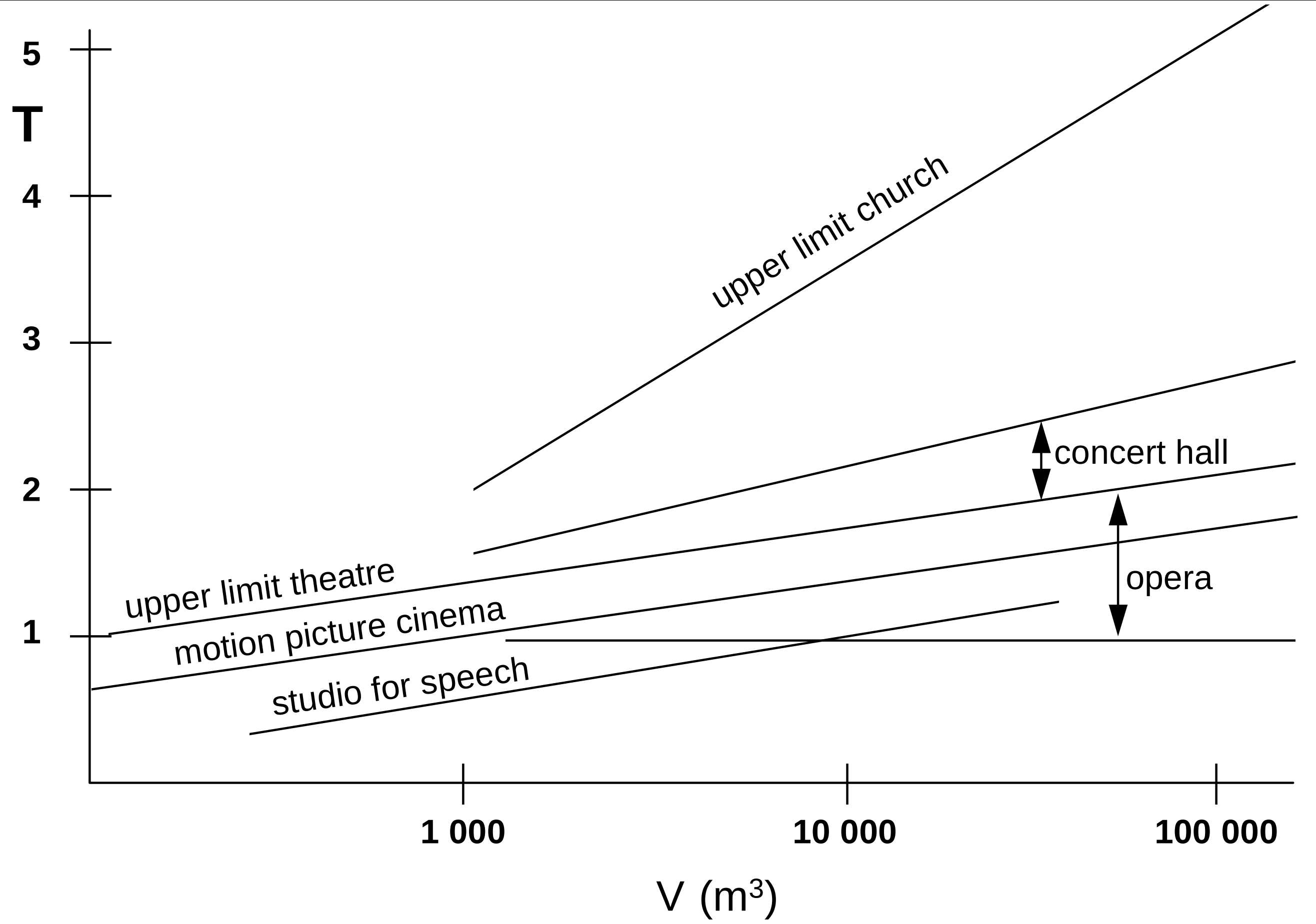
The effects of the number of people in the room should normally be taken into consideration. In many theatres and cinemas however, the effect of the variation in audience numbers is minimised by the use of plush sound-absorbing seating, having the same absorption as a person actually in the seat.

In other situations like airports where the reverberation time of the empty hall is known, the influence of the audience can be calculated by:

$$\text{New reverberation time} = \frac{V\,T}{V + (6\,T\,n\,A_p)}$$

where:

V = volume of room (m^3)
T = reverberation time of empty room (s)
n = number of persons
A_p = absorption per person (e.g. 0.5 Sabine)



Preferable reverberation times, dependent upon the room's volume

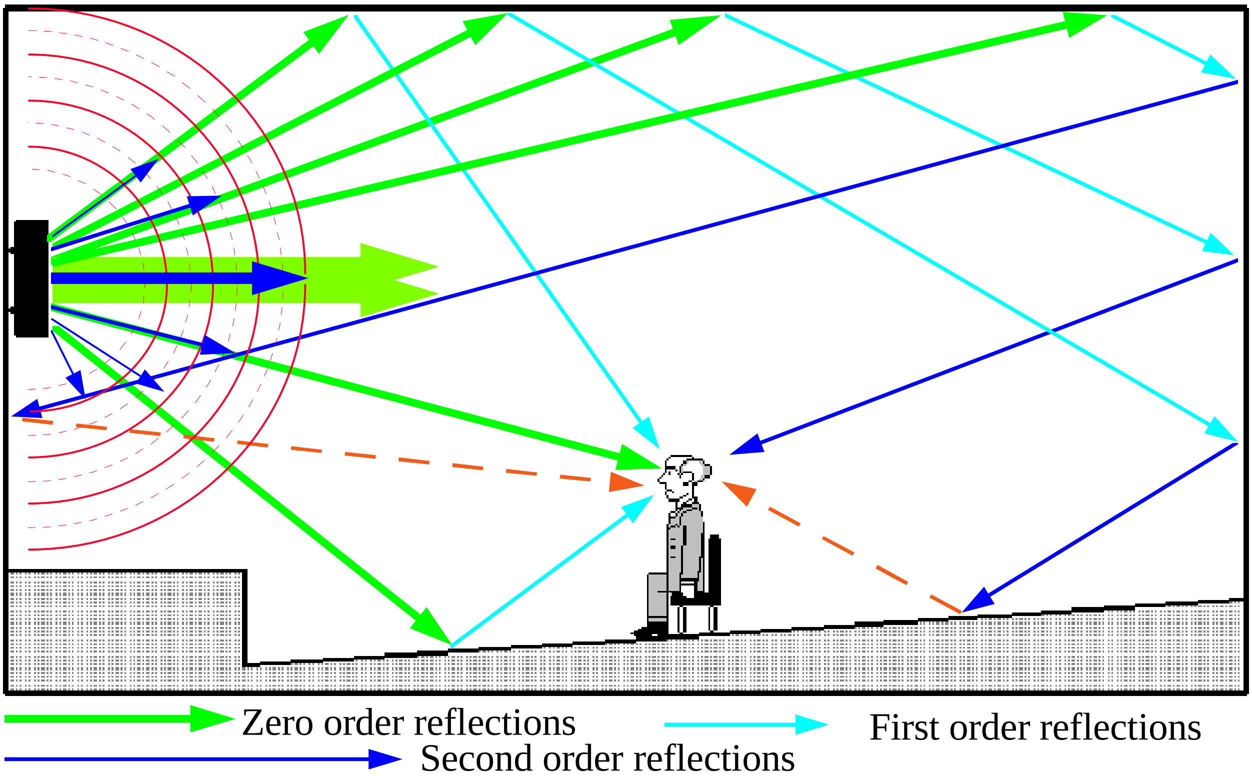
Calculation example for determining the Reverberation Time (T)

Room with dimensions of 30 x20 x 10m *Total Volume = 6000m^3 T= 0.161 x 6000 / A*
A = Absorption is the sum of all surfaces multiplied with the corresponding absorption coefficients.

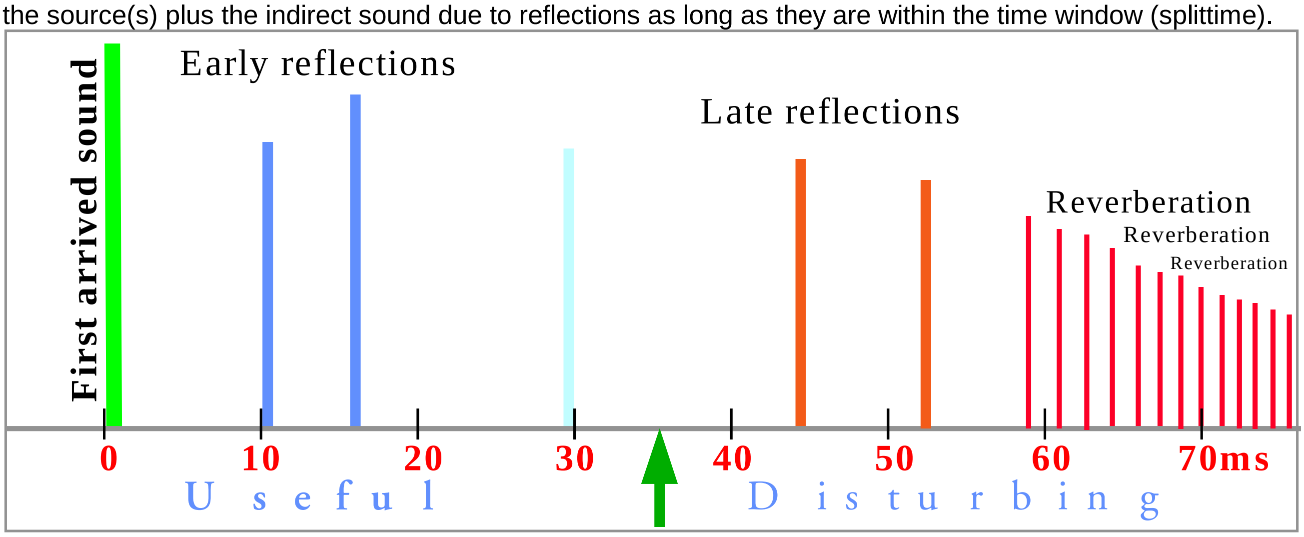
SURFACE	*MATERIAL*	α	*Sabine*
Floor	*(carpet)*	*= 30 x 20 x 0.37*	*= 222*
Side wall	*(bricks)*	*= 30 x 10 x 0.1*	*= 30*
Side wall	*(bricks)*	*= 30 x 10 x 0.1*	*= 30*
Front wall	*(woodpanel)*	*= 20 x 10 x 0.1*	*= 20*
End wall	*(woodpanel)*	*= 20 x 10 x 0.1*	*= 20*
Ceiling	*(hardboard)*	*= 30 x 20 x 0.15*	*= 90*
		Total Absorption	*= 412m^2*

T = 0.161 x 6000 / 412 = 2.34 s (neglecting the atmospheric absorption & audience occupation).

18.1.4 Calculation of Direct and Indirect Sound Fields



It is important to have a good understanding of the different sound fields in a room. Early sound carries the intelligibility, late sound gives the disturbance. Early sound is experienced by our ears as the sum of all speech related sounds arriving in a time window of 20-30 ms. This is the direct sound coming straight from the source(s) plus the indirect sound due to reflections as long as they are within the time window (splittime).



The level of this early useful sound (L_{dir}) can be calculated according the approach as explained in ch.17.

Direct Sound: $L_{dir} = L_S + 10 \log P_{el} - L_{Q1} - 20 \log r_1$

Indirect Sound: $L_{indir} = L_S + 10 \log P_{el} - L_{Q2} + 10 \log(1-\alpha_1) - 20 \log r_2$ via ceiling

$L_{indir} = L_S + 10 \log P_{el} - L_{Q3} + 10 \log(1-\alpha_2) - 20 \log r_3$ via floor

$L_S = SPL_{1.1}$ = SPL value for 1W at 1m on axis. P_{el} = power rating of loudspeakers (W)

L_Q = off axis level difference α = absorption coefficient r = distance in meters (m)

18.1.5 Calculation of Reverberant Sound Fields

All the speech related sound which arrives **later** than 20-30ms is regarded as useless and disturbing and consists of a chaos of reflections and is called reverberation.
The level of this reverberant disturbing sound (L_{rev}) depends of the source(s) , the volume and the reverberation time of the room. The following formulas can be used to calculate the reverberant sound field:

$$L_{rev} = 120 + 10\,\mathrm{Log}\,\frac{25T\,(1-\bar{\alpha})}{\mathrm{Volume}}\,P_{ac}$$

$$\bar{\alpha} = \frac{\mathrm{Volume}}{6T} \times \frac{1}{\mathrm{Surface}}$$

$$P_{ac} = \text{Electrical Power x efficiency} = P_{el} \times \eta$$

T	=	reverberation time (s) = RT_{60}
$\bar{\alpha}$	=	average absorption coefficient
η	=	loudspeaker efficiency as fraction
η [%]	=	loudspeaker efficiency as percentage

$$L_{rev} = 120 + 10\mathrm{Log}25/100 - 10\mathrm{Log}V + 10\mathrm{Log}T(1-\bar{\alpha})\eta[\%]\,P_{el}$$

$$\mathbf{L_{dir} = SPL_{1.1} + 10\,Log\,P_{el} - 20\,Log\,r} \qquad \textbf{(on axis only)}$$

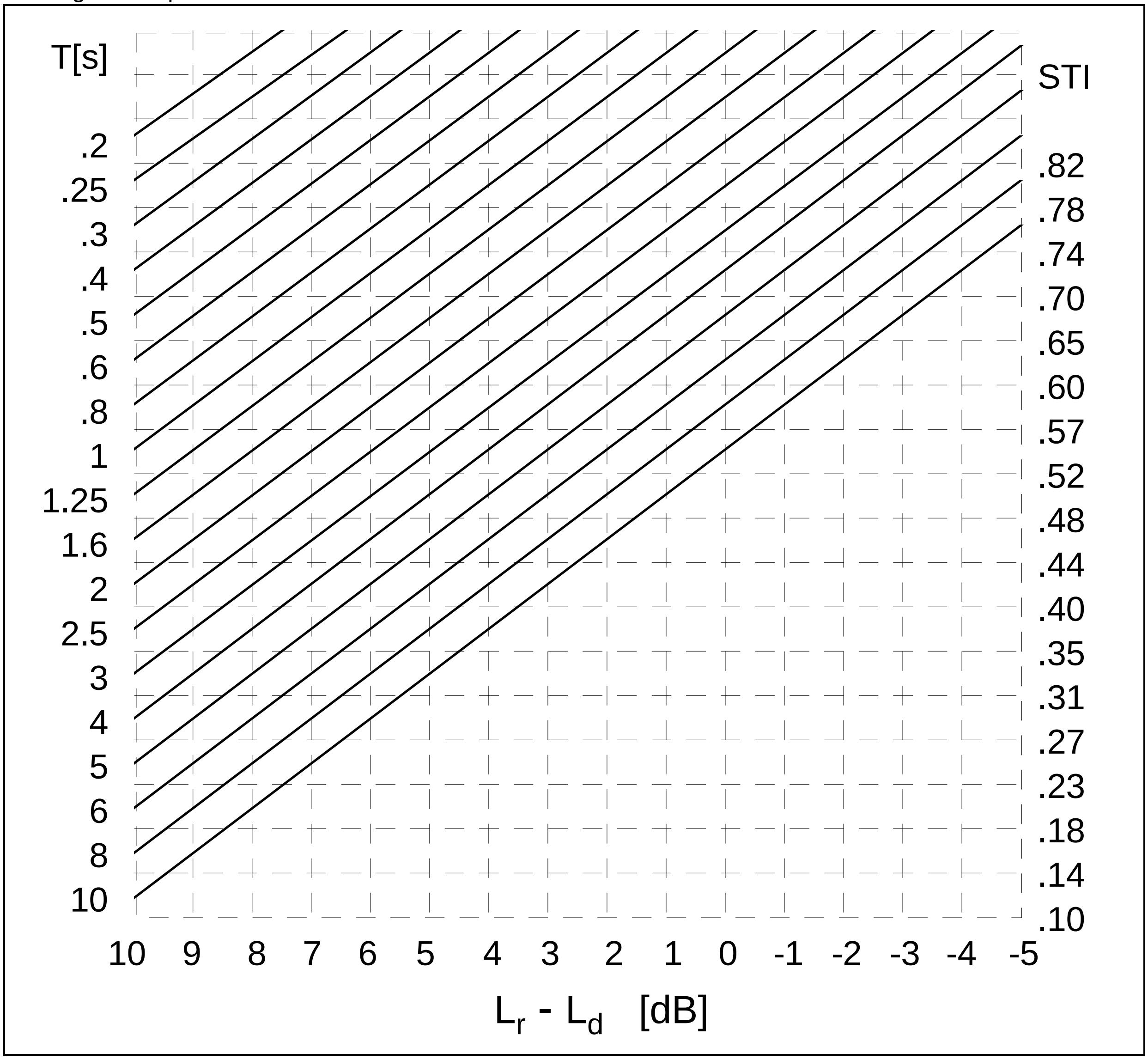
$$\mathbf{L_{dir} = SPL_{1.1} + 10\,Log\,P_{el} - L_Q - 20\,Log\,r} \qquad \textbf{(off axis)}$$

$$\mathbf{L_{indir} = SPL_{1.1} + 10\,Log\,P_{el} - L_Q + 10\,Log\,(1-\alpha) - 20\,Log\,r}$$

$$\mathbf{L_{rev} = 114 - 10\,Log\,V + 10\,Log\,T + 10\,Log\,(1-\bar{\alpha}) + 10\,Log\,\sum \eta[\%]\,P_{el}}$$

SPEECH INTELLIGIBILITY GRAPH STI & RASTI

This diagram helps to make a direct translation from the level difference between useful Direct sound



and disturbing Reverberant sound into Speech Intelligibility. The useful sound level will vary at different positions in the room, depending on distance, angle and useful reflections.
The corresponding formulas are used in the example below, which is a sound reinforcement application with two loudspeakers (1) & (2).

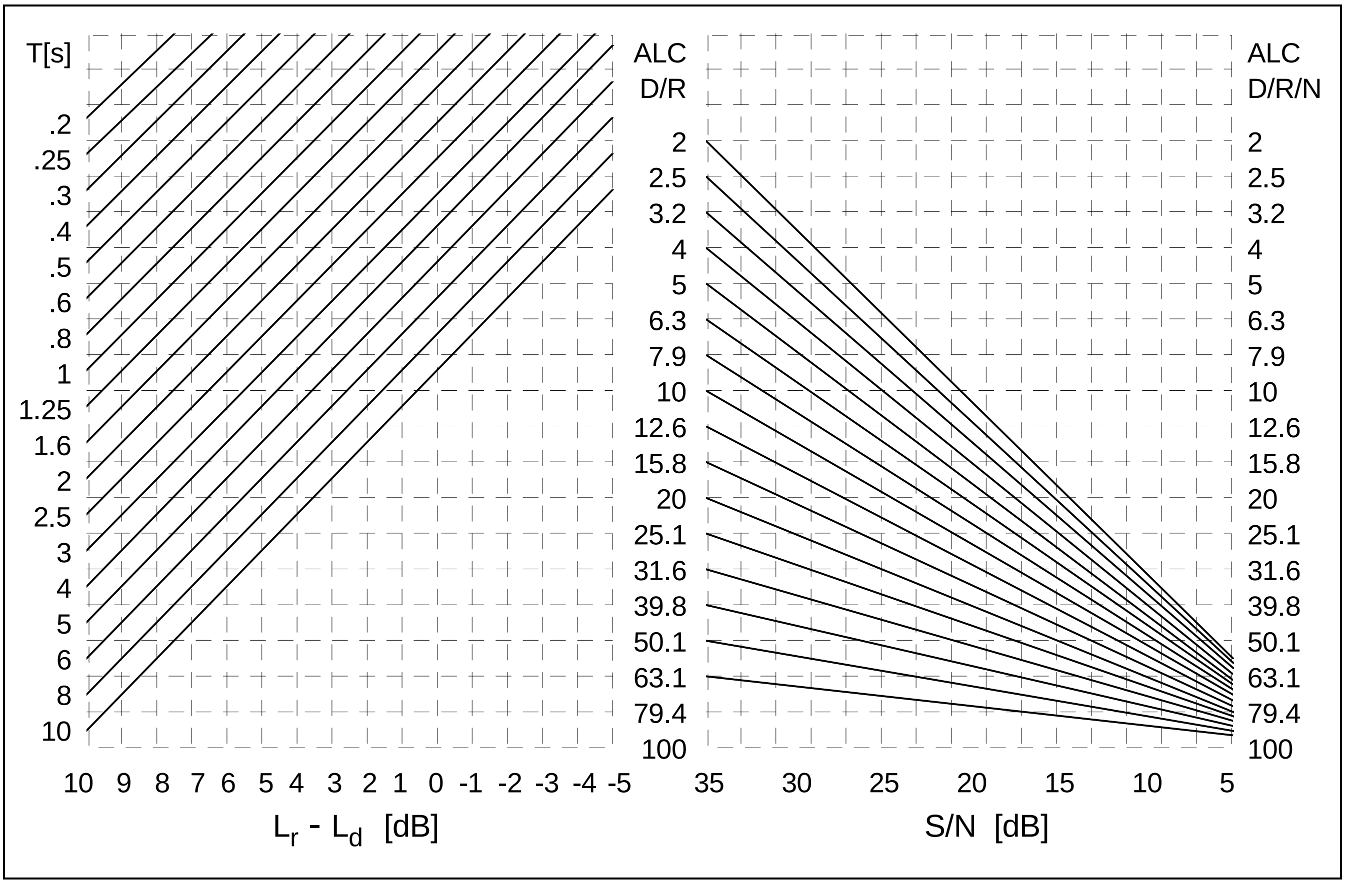
L_{rev} = 114 - 10 Log V + 10 Log T + 10 Log (1- $\overline{\alpha}$) + 10 Log $\Sigma\eta$(%) P_{el} = *100* **dB(SPL)**

L_{dir} (1) = $SPL_{1.1}$ + 10 Log P_{el} - 20 Log r_1 - L_{Q1} = *93* **dB(SPL)**
L_{dir} (2) = $SPL_{1.1}$ + 10 Log P_{el} - 20 Log r_2 - L_{Q2} = *90* **dB(SPL)**
L_{ref} (1) = $SPL_{1.1}$ + 10 Log P_{el} - 20 Log r_3 - L_{Q4} = *87* **dB(SPL)**
L_{ref} (2) = $SPL_{1.1}$ + 10 Log P_{el} - 20 Log r_4 - L_{Q5} = *87* **dB(SPL)**
Total useful sound (within 25 ms) added acc. 2.2.1 *96* **dB(SPL)**
Difference: Reverberant minus Useful *4* **dB**

After calculating on a particular position in the room this level difference we enter the chart at the bottom and go up to the intersection with the actual reverberation time (e.g. T = 3s) and read the Speech Transmission Index (STI) value at the right edge of the chart. (Example: 4dB > STI = 0.585)

18.1.6 Articulation Losses of consonants in speech. (% AL_{cons})

Since most of the information in a language is conveyed by the consonants (see chapter 1.1 for details), intelligibility may be expressed in the percentage articulation loss of consonants (%Al_{cons}). The acoustical investigator V.M.A. Peutz from The Netherlands spent a number of years resolving that the percentage of articulation loss of consonants determined the articulation score in various acoustical spaces. Formulas for %Al_{cons} were then developed and published in the Dec.1971 issue of the *Audio Engineering Society Journal.* From there Philips deduced the following practical table.



The %AL_{cons} is the difference between the direct and reflected field levels, calculated as a function of the reverberation time. If these figures are known, the accompanying graph makes it possible to quickly calculate %Al_{cons}.

Assume that a loudspeaker is radiating sound in a room with volume V and reverberation time T.

1. The reverberant field "L_{rev}" depends on the total acoustic power of the source(s); the volume of the room; and the reverberation time.
2. The direct field "L_{dir}" depends on the soundlevel at the source(s) at 1m pointing at the listener, and on the distance from the source(s).
3. The difference between the calculated L_{dir} and L_{rev} , expressed in dB (SPL), is a reliable measure for the expected speech intelligibility.
4. Using this difference, we enter the chart at the bottom and go up to the intersection with the corresponding Reverberation Time line.
5. If we then go right we will find the %AL_{cons} figure at the right edge of the chart.

If the difference between the speech signal peak level and the ambient noise level is smaller than 35 dB, and no limiter is used, the speech intelligibility will be effected.

1. The ambient noise level is measured using a sound level meter with A weighting (see chapter 1.3.2) and set to Fast reading.
2. The signal level is determined by:
   a) calculating, using the sum of L_{dir} and L_{rev}
   b) measuring in dBA (Fast) with 10 dB added.
   c) measuring directly in dBA (peak or peak hold).
3. Using the %AL_{cons} value, determined from the previous chart, enter at the left side and follow the sloping line down, until it intersects with the vertical S/N ratio line. From this point go right to read the new %AL_{cons} value at the right edge of the chart.

18.1.7 Speech Transmission Index (STI & RASTI)

A method of calculating and measuring speech intelligibility has been developed, called the Speech Transmission Index (STI) method, which evaluates the intelligibility over 7 octave bands from 125 - 8000 Hz. This method is fully described in the appendix of this manual and in the IEC268 -16 & BS6840 -16 documents. It basically stands for the speech signal transmission between the signal source position and listener position. The STI values are theoretically between 1(ideal) and 0 (bad) in practice between 0.75(ideal) and 0.25(bad). Portable (battery operated) measuring equipment, manufactured by Bruel & Kjaer, uses the so called Rapid Speech Transmission Index (RASTI) method of calculation, and restricts the measurements of the speech transmission to the 500 Hz and 2 kHz octave bands only, instead of all 7 octaves.

18.1.8 Subjective %AL_{cons} and RASTI requirements.

%AL_{cons} = 1 - 10% RASTI ≥ 0.50	Speech intelligibility adequate for **complicated messages** and lectures and for untrained speakers & listeners.
%AL_{cons} = 10 - 15% RASTI = 0.50 - 0.45	Speech intelligibility adequate for **less complicated messages** by untrained speakers, but still adequate for complicated messages in a clear and well articulated voice.
%AL_{cons} = 15 - 30% RASTI = 0.45 - 0.32	Speech intelligibility adequate only for **simple messages** and announcements. Complicated messages require trained speakers & listeners.
%AL_{cons} = 30% RASTI = 0.32	Limit of acceptable intelligibility for simple messages, for trained speakers & listeners.

18.1.9 Converting RASTI to %AL_{cons}

RASTI	%AL_{cons}
0.20	58
0.22	52
0.24	47
0.26	42
0.28	37
0.30	34
0.32	30
0.34	27
0.36	24
0.38	22
0.40	20
0.42	18

RASTI	%AL_{cons}
0.44	16
0.46	14
0.48	13
0.50	11
0.52	10
0.54	9.1
0.56	8.2
0.58	7.4
0.60	6.6
0.62	5.9
0.64	5.3
0.66	4.8

RASTI	%Al_{cons}
0.68	4.3
0.70	3.8
0.72	3.4
0.74	3.1
0.76	2.8
0.78	2.5
0.80	2.2
0.82	2.0
0.84	1.8
0.86	1.6
0.88	1.4
0.90	1.3

$$\%AL_{cons} = 170.5405\ e^{-5.419(STI)} \qquad STI = -0.1845\ \mathrm{Ln}(\%AL_{cons}) + 0.9482$$

Source: Farrel Becker

19.0 Designing For The Acoustic Environment

19.1 LOUDSPEAKER PLACEMENT AND COVERAGE

A few practical considerations must be taken into account when selecting, placing and aiming a loudspeaker in a sound system design.

1. The loudspeakers must be positioned in such a way that they are able to produce an even spread of sound, reaching all audience areas of the room with adequate loudness and clarity. If this is not so, some listeners could be exposed to an uncomfortably high SPL, while others may have difficulty in actually hearing the audio signal sufficiently.

2. Speech requires generally a good transmission and reproduction of the 500 Hz to 5 kHz frequency band, while music requires at least 100 Hz to 10 kHz to give satisfactorily results. This should be taken into consideration in selecting a loudspeaker type.

3. For speech applications, upto the 4 kHz octave band is essential for the annunciation of consonants, and therefore intelligibility. Therefore we use the loudspeaker opening angle data at 4 kHz for the calculations for equal coverage.

4. For ceiling systems the spacing of the loudspeakers should be determined by looking at the covered areas (-6dB) at 4 kHz. The audience area divided by this coverage area gives the number of speakers. It means that the audience will hear the announcements at about the same level for the required spectrum.

5. In installations with multiple loudspeakers, spacing the loudspeakers less than 15 meters apart will help minimise echo otherwise proper delayed signals should be applied. (See chapter 10 for a description of time delay)

6. Given the specifications of the loudspeakers we intend to use, it is possible to calculate the SPL at any point in a room or area, either by using the formulas provided in chapters 17 or 18 in this manual or using "EASE", the software package described in the *Simulating and Measuring Appendix* at the end of this book.

7. Depending on the application, a good general rule would be to calculate the level (SPL) at 1.20 meters from the floor, which is the average ear height of a person sitting. A popular speech peak level, known as the Comfortable Listening Level (CLL) is generally agreed upon as 80 dB(SPL), which is the peak level in average conversation measured on a distance of 1m. This assumes that the ambient noise level is low in the room, which is not always the case.

8. Background noise, or ambient noise, can make a great deal of difference to the level required for an adequate intelligibility, especially in noisy environments such as facto- ries or airports. To keep the level more than 15 dB louder than the ambient noise, the use of proper callstations with build-in compressor/limiter is required.

19.2 SUMMARY OF THE LOUDSPEAKER-DESIGN

One of the vital requirements of any sound system is its ability to produce an even spread of sound, reaching all parts of an area or room with equal intensity and clarity. In doing this, the complete speech (and/or music) spectrum should reach the listener's ears as unchanged and true to the original as possible.

The performance of a sound system can be predicted before it is installed or purchased.

The level of the **direct sound** as received from the loudspeakers, including beneficial early reflections from side walls and/or ceiling, are calculated for the important octave bands. With these calculations we optimise the coverage for the audience at 4000 Hz.

The level of the **reverberant sound** caused by the selected solution can be calculated if the Volume / Reverberation time / Absorption is known.
This can be done per octave band (125 - 250 - 500 - 1000 - 2000 - 4000 - 8000 Hz).

After that the **intelligibility** is calculated with the values for 1000 Hz, to verify that the listeners can hear the reproduced sound clearly.

SUMMARISING THE DESIGN PROCEDURE

1. **Select the correct loudspeaker type(s).**
2. **Select the optimum loudspeaker position(s).**
3. **Select the best aiming points.**
4. **Check the coverage at 4000 Hz.**
5. **Calculate the SPL_{dir} on the aiming point(s).**
6. **Calculate the SPL_{dir} on the - 6dB points.**
7. **Select the Powertapping(s).**
8. **In reverberant rooms calculate SPL_{rev}.**
9. **Check the intelligibility in STI or Alcons(%).**

Repeat(?)1-7/9 for other loudspeaker/place/aiming.

$$L_{dir} = SPL_{1.1} + 10 \text{ Log } P_{el} - 20 \text{ Log } r \qquad \text{(on axis only)}$$
$$L_{dir} = SPL_{1.1} + 10 \text{ Log } P_{el} - L_Q - 20 \text{ Log } r$$
$$L_{indir} = SPL_{1.1} + 10 \text{ Log } P_{el} - L_Q + 10 \text{ Log } (1-\alpha) - 20 \text{ Log } r$$

$$L_{rev} = 114 - 10 \text{ Log } V + 10 \text{ Log } T + 10 \text{ Log } (1-\bar{\alpha}) + 10 \text{ Log } \sum \eta[\%]\, P_{el}$$