Crestron **Surround Sound** Primer

CRESTRON

This document was prepared and written by the Technical Documentation Department at:

Crestron Electronics, Inc.
15 Volvo Drive
Rockleigh, NJ 07647
1-888-CRESTRON

Contents

Crestron Surround Sound

Sound, Hearing and the Limits of Perception

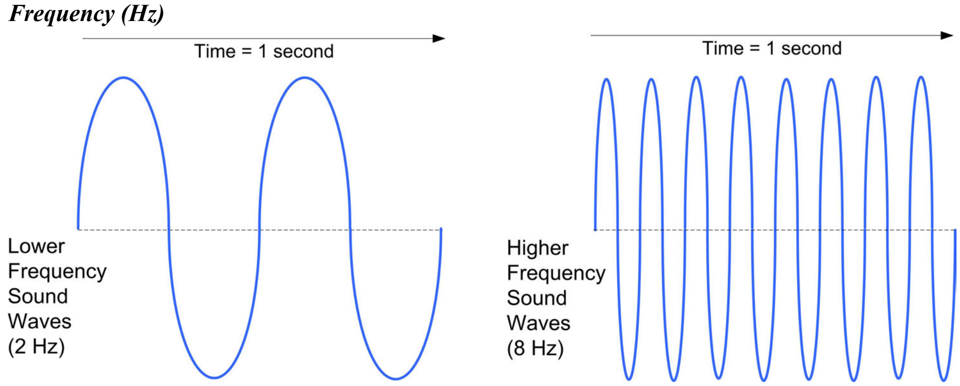
Objects produce sound when vibrating in an elastic medium. Solids, liquids and gas all conduct sound. When something vibrates in the atmosphere, it pushes the air around it creating an acoustic compression wave. The trail of this wave creates a drop in pressure, called rarefaction.

Sound waves, which travel at about 1,086 feet per second (331.1 meters per second) in the air, have three basic properties, frequency, wavelength, and volume (amplitude).

Frequency

Frequency is the number of distinct positive or negative sound wave elements that repeat in one second. Frequency is measured in Hertz (Hz). A 20 Hz frequency contains 20 positive and negative cycles of individual components each second (20 distinct waves passing by in one second). A 20 kHz (kilohertz) frequency contains 20,000 of these cycles every second.

Frequency (Hz)

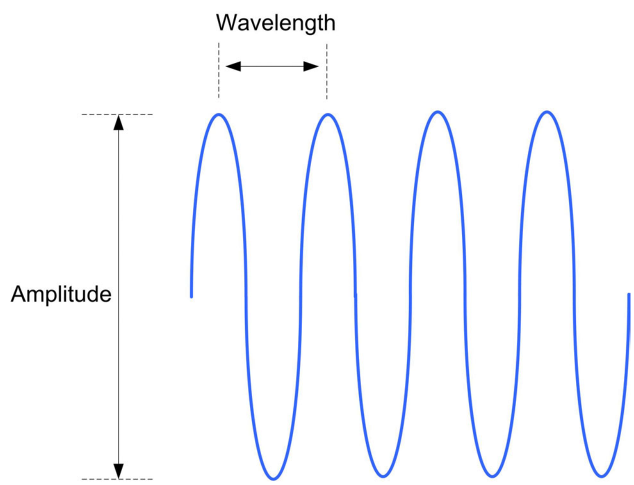


Wavelength

Wavelength is the distance between two points on consecutive waves. It is measured from the same position on a wave in two consecutive cycles. Wavelength can be measured by taking the horizontal distance from a point (at the peak in our example) of one wave cycle to the same point at the peak in the second wave cycle.

Low frequency sounds have long wavelengths and high frequency sounds have short wavelengths. The length of a 20 Hz sound wave is about 56 feet. Speakers that produce low frequencies must therefore be large in size with long excursions (the distance a speaker moves in and out) to produce large and long waves. Speakers producing high frequency sounds must be small enough to move rapidly and produce the very small waves of high frequencies (about two thirds of an inch at 20 kHz).

Wavelength and Amplitude



Volume (Amplitude)

Volume is the relative loudness or power of an audio signal resulting from the amplitude of a sound wave. Amplitude is the vertical distance from zero to the highest point or peak. Sound waves with higher amplitudes carry more acoustic power and therefore higher volume.

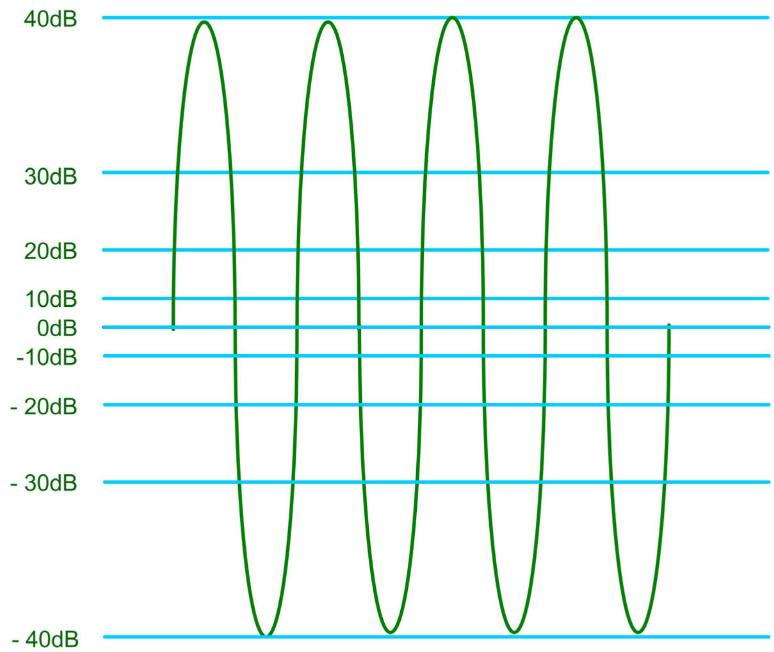
Volume is measured in units called decibels (dB). A dB is one-tenth of a Bel, named in part after Alexander Graham Bell (the “B” is capitalized for Bell) and is used in both audio and video applications. Decibel is a logarithmic scale measuring the intensity (pressure level) of sound. Decibels are ratios, not fixed quantities. Decibels are also referred to as a measurement of "gain" with respect to amplifiers (refer to the glossary).

For the non-linear human ear to perceive a sound that seems twice as loud, a ten-decibel (10 dB) increase doubles the sound pressure level, 20dB is twice the sound level of 10dB, and 30dB is twice as loud as 20dB. 40dB is twice the sound level of 30dB and four times the sound level of 20 decibels.

With some kinds of equipment, such as microphones, analog tape recorders, or LP playback systems, the dB measurement is "weighted" as to audibility, because the ear is more sensitive to particular frequencies. Two common corrections for hearing characteristics are the A-weighted and the more rigorous C-weighted scales, indicated as dBA or dBC, respectively.

The term decibel is also used in various other measurements such as signal-to-noise ratio, gain and dynamic headroom. In these instances, decibel refers to the measurement of signal increase or signal strength instead of sound pressure level, but the logarithmic scale concept remains the same.

Decibel Scale



VU and dB

VU and dB meters both measure the audio power and they both use logarithmic scales to report that power. In both measures, the zero is chosen as the highest power for which distortion is acceptable.

Where VU and dB differ is in how they measure audio power. VU is short for "volume units" and it is a measure of average audio power. A VU meter responds relatively slowly and considers the sound volume over a period of time. Its zero is set to a 1% total harmonic distortion level in the recorded signal.

Decibel (dB) meters measure instantaneous audio power. A dB meter responds very rapidly and considers the audio power at each instant. Its zero is set to a 3% total harmonic distortion level. Because of these differences in zero definitions, zero on the dB meter is approximately +8 on the VU meter.

Perception

The human ear can usually hear sounds in the range of 20 Hz to 20 kHz, but are most sensitive to sounds from 2 KHz to 4 KHz, the same range as the human voice. With age, this range decreases, especially at the upper limit. Very Low frequencies (below 20Hz) cannot be heard, but loud low frequency sounds can be felt as vibrations on the skin. The frequency resolution of the ear is, in the middle range, about 2 Hz. Changes in pitch larger than 2 Hz are noticeable. Even smaller pitch differences can be perceived when two pitches interfere and are heard as a frequency difference pitch.

The lower limit of audibility is defined as 0 dB, there is no defined upper limit. The upper limit is more a question of where the ear will be physically harmed. This limit depends on the time exposed to the sound. The ear can be exposed to short periods of sounds of 120 dB without harm, but long exposure to 80 dB sounds can do permanent damage. The human voice range is about 68-76 dB; a jet plane creates about 120 dB of sound.

Sound waves radiate out from the source in straight lines regardless of frequency or wavelength. But low frequency (long wavelength) sounds do not fit in confined spaces. They loose their directional character and that is why you only need one subwoofer for a sound system; you can't tell where the lowest frequency sounds are coming from when the sound is confined in a room.

The model of sound so far described is a simplified version, operating in just one dimension (as opposed to the real world three-dimensions), and assumes that the vibrating particles of air are held semi-rigidly. In the real world, air molecules are in constant random motion in all three dimensions. Air pressure constantly changes as the molecules collide and rebound from each other and from objects in the environment. This random motion creates a background level of noise and helps define the lower limit of hearing.

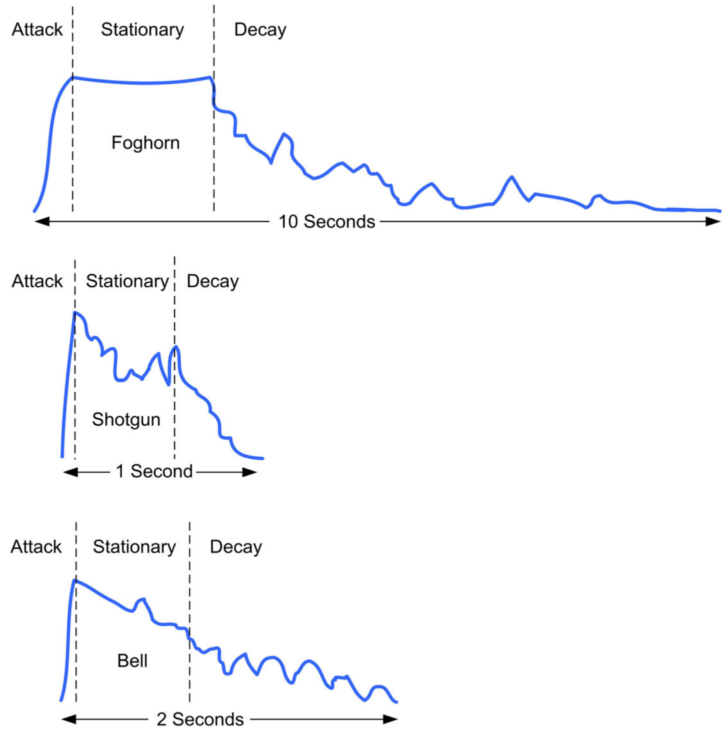
Timbre

Musical timbre is a property of sound. It is composed of spectral components containing perceptual cues, and can be described by Fourier series coefficients. The spectral “envelope” of a sound, the profile of the Fourier series, is the sound amplitude behavior over time. The pattern that this sound pressure variation creates is the waveform of the sound.

Timbre is the temporal evolution of the spectral envelope. This envelope consists of an "attack" portion at the beginning or onset of the sound, a sustained portion (stationary state), and a decay portion.

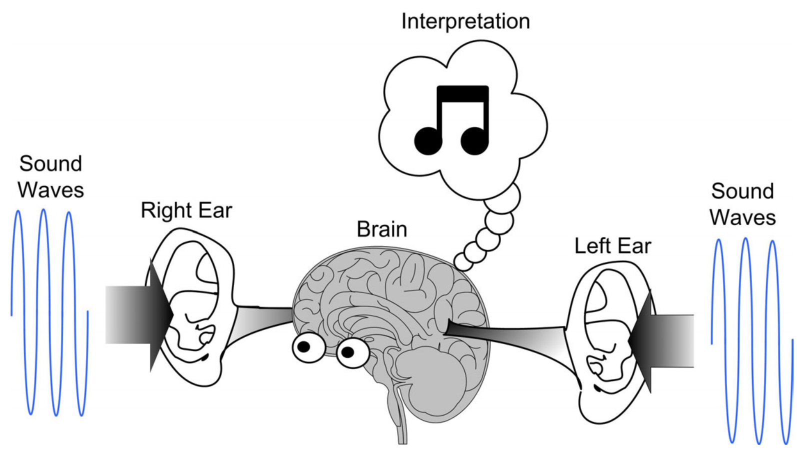
Timbre has psychoachoustic properties. A number of transient fluctuations occur during the initial part (attack), for example, the moment a violinist puts the bow to the string. These are called *onset transients* and are important in identifying the sound and its location in space.

Waveform Envelope Examples:



Psychoacoustics

Psychoacoustics is the study of human auditory perception. It includes; the physical characteristics of sound waves, the physiological structure of the ear, the electrical signal from the ear to the brain, and the subjective interpretation of the listener. Understanding psychoacoustics is essential to creating surround sound.



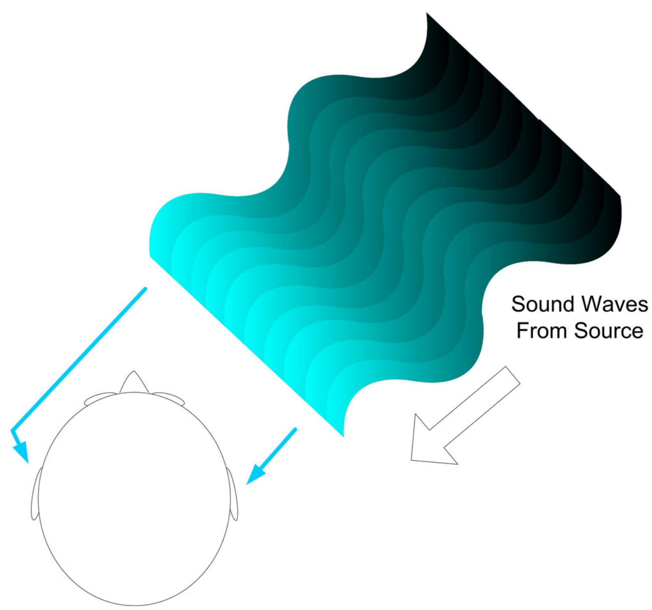
Localization

Our stereophonic ears can discern azimuth or horizontal (left-right) directionality, and zenith or vertical (up-down) directionality. We perceive directionality using the localization mechanisms of: interaural time difference, interaural intensity difference, pinna filtering, and motion parallax.

Interaural Time Difference

The horizontal position of a sound is determined by comparing the information coming from the left and right ears. The Interaural Time Difference is the difference in arrival time at each ear. The approximate six-inch separation of the ears slightly delays the sound, each ear receiving a slight difference when the sound is not equally distant from the two ears. Although the time delay differences are very slight, the brain extracts precise directional information from this information. Human listeners are able to accurately locate the sources of sound from almost any direction, even from above when interaural differences are almost zero. Listeners are also capable of locating sound sources in a room when the reflections from the walls are louder than the sound coming directly from the source.

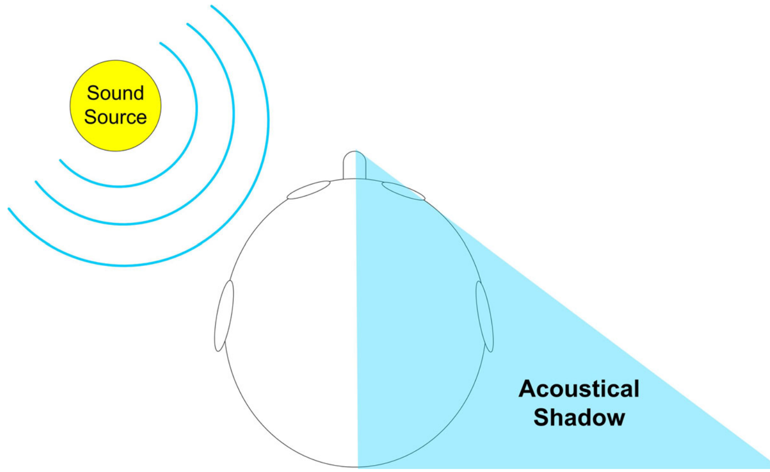
Interaural Time Difference



Interaural Intensity Difference

The head, shoulders and upper torso create a sound barrier at one ear or the other. This acoustical shadow called the Interaural Intensity Difference.

Interaural Intensity Difference

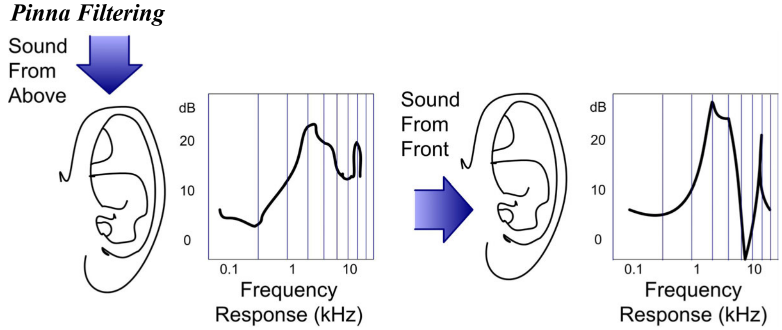


For example, a sound coming from the extreme left has a lowered intensity in the right ear in addition to a slight time delay. The reduced intensity is the additional distance plus the effect of the acoustical shadow. The amount of this effect depends on frequency, and is useful for high frequencies up to wavelengths twice the distance between the ears (about 1 kHz). Lower frequencies, with longer wavelengths, bend around obstructions.

Pinna Filtering

The pinna structure is the outer part of the ear. Its forward pointing position and complex curves affect the way sound is heard. A sound is coming from behind or above bounces off the pinna in a different way than from in front or below. When the indirect (reflected) sounds from the pinna combine with the direct sounds, the wavelengths of the sound are altered.

Pinna Filtering



The brain, interpreting the altered sounds, produces directional information. To provide additional cues, small head movements (motion parallax) allow the brain to judge relative differences.

The Precedence Effect

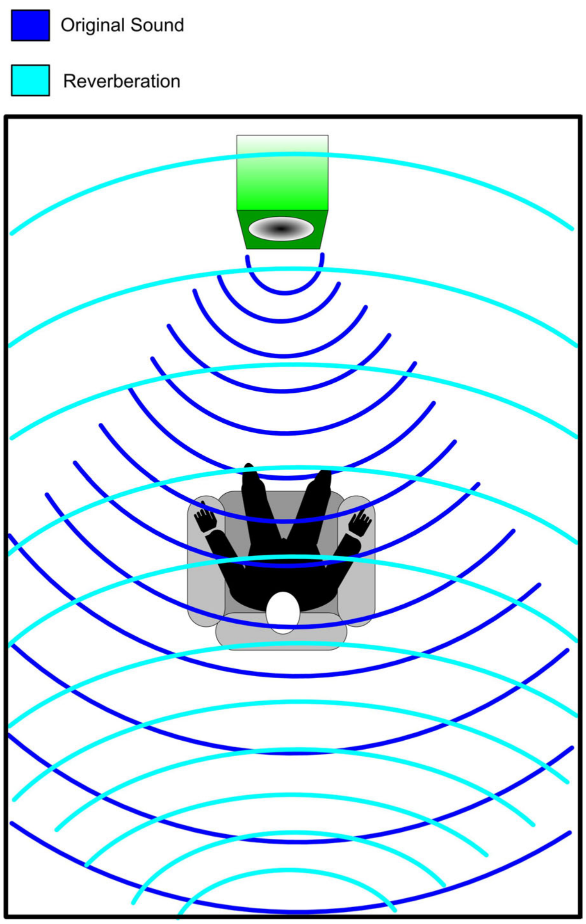
The precedence effect is a listening strategy unconsciously used to cope with distorted localization cues in a confined space. Localization judgments are based on the first arriving sound waves at the beginning of a sound. This strategy is known as the precedence effect, because the earliest arriving sound wave is given precedence over the subsequent reflections and reverberations.

Reverberation and Echoes

The ratio of the direct to reverberant energy is a primary cue for range and space. Very short delays cause a sound image to shift spatially and color the tone. Longer delays contribute to a spatial impression of reverberation. Reverberation can also make speech indistinct by masking the onset transients.

Sound decreases inversely with the square of the distance. But in an ordinary room, the sound is reflected and scattered against room boundaries and objects within the room. Reverberation is essentially an echo that increases by bouncing off of hard surfaces. Reverberations are dampened when absorbed by soft materials such as rugs, carpet and sofas. These reflections are most noticeable when the time delay between the direct sound and the reverberation gets longer than 30 to 50 ms, the echo threshold.

Reverberations



Acoustic designers place importance on early reflections (arriving within the first 80 ms), which reinforce the direct sound (as long as the angle of reflection is not too wide). Reflections arriving after 80 ms add reverberant energy, which gives the sound spaciousness, warmth and envelopment. The acoustic design listening spaces usually involves creating a balance between clarity, definition, and spaciousness.

Listeners often have different preferences regarding this balance.

Temporal Masking

Temporal masking is a defense mechanism of the ear that is activated to protect its delicate structures from loud sounds. When exposed to a loud sound, the human ear reacts by contracting slightly, temporarily reducing the perceived volume of sounds that follow. Loud sounds in an audio signal tend to overpower other sounds that occur just *before* and just *after* it.

A History of Surround Sound

The simplest method of sound recording is called monaural or mono. All the sound is recorded on one audio track and played back on one speaker.

Two-channel recordings played back on speakers on either side of the listener are referred to as stereophonic or stereo. The simplest two-channel recordings, (binaural recordings) are produced with two microphones. Playback of these two channels on two speakers recreates some of the experience of being present at a concert event. But the listener must be anchored in the "sweet spot" between the speakers to maintain the illusion of the phantom sound from between the speakers.

Surround recordings add additional audio channels so that sound comes from multiple directions. In effect, widening the sweet spot and enhancing the realistic sound quality.

The term "surround sound" refers to specific multi-channel systems designed by Dolby Laboratories, but is commonly used as a generic term for theater and home theater multi-channel sound systems.

Early Surround

Walt Disney's "Fantasia" (1941), was one of the first surround sound motion pictures. Four separate recordings of each orchestra section were recorded on a separate reel of film and played through speakers positioned around the theater.

By the late 1950s, movies were encoded with simpler multi-channel formats. Several different systems emerged, including Cinerama and Cinemascope. These systems were referred to as stereophonic sound, or theater stereo. Stereophonic sound used multiple magnetic audio tracks at the edges of the film. The standard film format could support two optical audio tracks or up to six magnetic audio tracks. A four-channel theater system included: left, right, center speakers behind the screen, and surround speakers along the sides and back of the theater.

Quadraphonic

In the quadraphonic systems of the early 1970s, two rear surround channels were combined (matrixed or encoded) with the two front channels so that the two sides of an LP groove carried four playback channels. This four-speaker system required a decoder and a separate rear channel amplifier. Problems with system standardization prevented technological development.

Dolby

In the mid 1970's, Dolby Laboratories (www.dolby.com) devised a method to encode additional audio channels. This technology, initially known as Dolby Stereo when it was launched in 1975, was later renamed Dolby Surround.

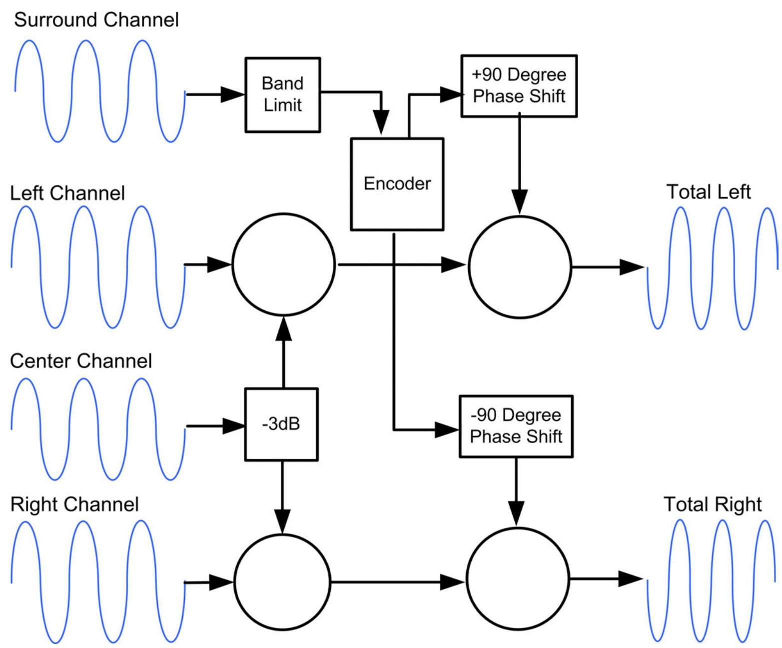
In 1982, Dolby Surround and enhanced Dolby Pro Logic playback decoder became available to the consumer. Dolby Surround, like the earlier Quadraphonic systems, used channel matrixing to combine four audio channels into two signals.

Also described as 4-2-4 matrixing, these signals (compatible with two-speaker stereo playback) can be decoded into multiple channels.

Basic Dolby Surround decoding yields: front left, front right, and one surround channel (the center channel is a phantom).

The 4-2-4 encoder accepts four separate inputs (left, right, center and surround) and creates two outputs (left-total and right-total). The front left and right channels are a regular stereo signal. The center channel is inserted equally in the left and right channel, with a 3 dB level reduction to maintain constant acoustic power.

Dolby 4-2-4 Encoding



The surround input is also divided equally between the left-total and right-total signals but first undergoes three processing steps:

- It is frequency band-limited from 100 Hz to 7 kHz
- It is encoded with a modified Dolby B-type noise reduction

- The surround signal is split into two identical signals; one signal is phase shifted by +90 degrees relative to the fronts and the other signal by -90 degrees, creating a 180-degree phase difference between the surround signal components.

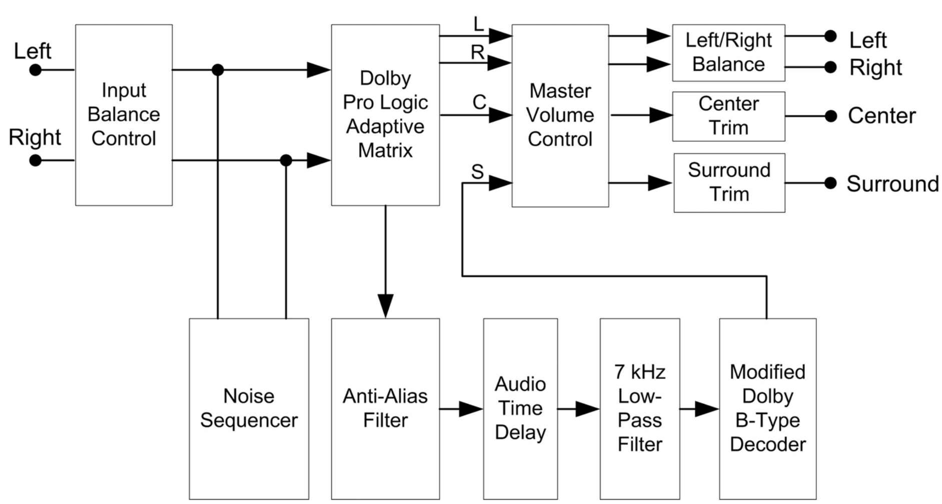
To recapture the matrix signals:

1. The center channel is extracted by summing the left-total and right-total signals.
2. The surround signal is extracted by taking the difference between them.
3. The identical center channel components in the left-total and right-total signals cancel out one another in the surround output.
4. The equal and opposite surround channel components cancel out one another in the center output.
5. Signals that are different in left-total and right-total are applied to the front left and right speakers.
6. Signals that are identical and in-phase are applied to the center channel.
7. Signals that are identical but out of phase are applied to the rear surround speaker.

Dolby Pro Logic

Dolby Pro Logic is an enhanced type of Dolby Surround decoder introduced in 1987. Pro Logic uses directional enhancement circuitry to actively steer sound toward the dominant position. Dolby Surround Pro Logic decoding yields: left, center, right, and a monaural surround channel.

Dolby Pro Logic Decoder Block Diagram



It determines the dominant signal of the four outputs by comparing the signal level of the left-total and right-total signals while simultaneously comparing the level of the sum and difference signals. An active matrixing circuit uses this information to output the dominant signal to the appropriate channel while canceling other channels. The amount of enhancement applied is relative to the level of dominance.

Dolby Pro Logic II

Pro Logic II includes bass management, allowing bass to be reproduced from the main speakers, or an LFE (Low frequency effects) output.

Pro Logic II uses directional enhancement, and a feedback loop allows the anti-phase signals to more closely match unwanted crosstalk signals. Control of the spatial dimensionality and front sound field are also provided.

Dolby Digital

Storage and playback of audio signals in a digital format has a lot of advantages over analog. Low-cost digital signal processing (DSP) has revolutionized the audio/video industry. The availability of complex processing coupled with increases in storage density has opened up new possibilities in audio/video coding and playback. As its name implies, Dolby Digital, introduced in 1997, takes advantage of DSP. It uses a digital processing method, which exploits the limits of human hearing.

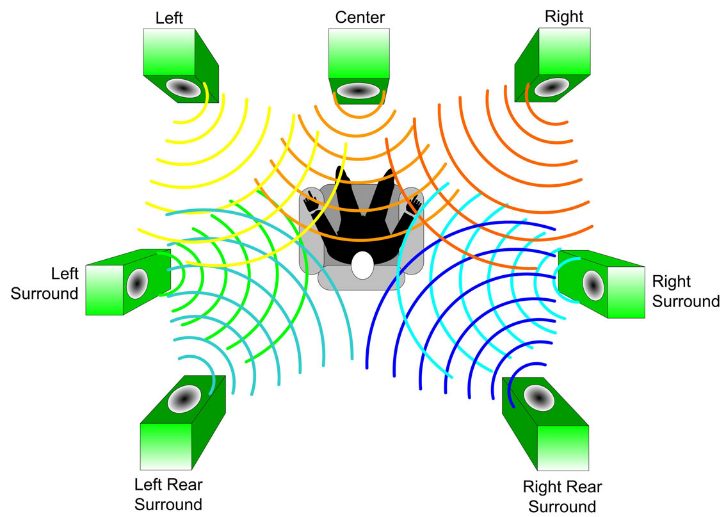
Playback of Dolby Digital is similar to that of Dolby Surround, but the technologies are very different. Dolby Surround is primarily on analog storage media, but Dolby Digital is a digital-only coding system.

Dolby Digital Surround EX

Dolby Digital Surround EX provides a third surround channel on Dolby Digital movie soundtracks. The third surround channel can be decoded at the cinema or home viewer for playback over surround speakers located behind the seating area. Surround speakers to the sides reproduce the left and right surround channels. The back surround channel is matrix-encoded onto the left and right surround channels of a conventional 5.1 mix, permitting conventional 5.1 playback.

A/V receivers are available with either Dolby Digital EX or THX Surround EX decoding which derives the extra surround channel for playback in 6.1 (three surround speakers) or 7.1 (four surround speakers) configurations. The additional rear speakers are matrixed and not discrete in nature.

7.1 Soundfield (Subwoofer not shown)



DTS

DTS Coherent Acoustics is a variable/high bit rate codec for discrete 5.1 multichannel digital audio. DTS Coherent Acoustics is a competing codec to Dolby Laboratories AC-3®. DTS encoding requires less compression of the signal which provides improved detail, clarity and dynamics.

DTS-ES

DTS-ES is a digital matrix decoder developed by DTS for back channel 5.1 surround sound in theatres. It is comparable to Dolby Digital Surround EX soundtracks and is a competitor to the Dolby SA10 adapter, the proprietary back surround decoder for cinemas.

Coherent Acoustics is a flexible audio coding scheme, allowing for a wide range of bit resolutions and sampling rates. Its operation ranges from low bit rate, lossy perceptual coding (meaning that it reduces data based on psychoacoustics principles) to high quality, lossless coding with variable bit rates. Encoded data rates of 32 to 4096kbps, sampling rates of 8 to 192kHz, resolutions of 16- to 24-bit and up to eight channels are possible.

DTS-ES for home theater features:

- The back channel is matrixed in the left and right surround channels.
- A discrete back channel can be optionally encoded.
- A DTS-ES 6.1 discrete decoder plays the discrete back channel and subtracts the encoded discrete back channel out of the matrixed left and right surround channels. This is the only decoding mode with a discrete rear speaker for the home.
- DTS-ES is fully compatible with 6.1 and 5.1 matrix decoders.

DTS-Neo 6

The DTS-Neo 6 derives the surround sound channels from a technique of advanced sub-band processing by using algorithms. Stereo music material may be expanded from stereo to 5.1 or 6.1 surround channels. Users with 5.1 and 6.1 systems derive five and six separate channels, respectively, corresponding to the standard home-theater speaker layouts. Bass management in the preamp or receiver generates the subwoofer channel.

- Neo 6 decodes Extended Surround matrix soundtracks, and generates a back channel from 5.1 material.
- Neo 6 technology separately steers various sound elements within a channel or channels in a naturalistic way that follows the original presentation.
- Neo 6 Music and Neo 6 Cinema have slightly different characteristics because these are recorded in different types of sound studios.

Pulse Code Modulation

The representation of sound as a mathematical sum of frequencies (digital) is just as valid as a linear series of pressure measurements (analog).

Pulse Code Modulation (PCM) representations of high quality audio is the industry standard method of digitizing analog audio signals.

An audio signal, recorded as a time varying electrical signal, is a fluctuating voltage. The amplitude and polarity of this voltage are directly related to the changes in air pressure. This signal is measured or sampled at a rate of at least twice the maximum frequency contained in the audio signal (the Nyquist rate), typically 44.1 kHz for Compact Discs or 48 kHz for professional audio applications. Each sample is given a binary number to represent the measured voltage at that instant in time.

The binary number, representing the original signal, is stored in memory. This representation of the audio signal must be converted back to a continuously time varying analog signal.

Typically a 16-bit number is used in PCM, yielding a number range up to about 32767. For a 1-volt analog signal, a 16-bit number can only resolve to 1/32767 of a volt. Because the 16-bit number cannot represent voltages that lie between these 1/32767 steps, the measurement is rounded off to a discrete number. This loss in precision creates a slight increase in the background noise during playback.

Generally, small binary numbers generate higher levels of noise or distortion, and large binary word lengths have such low noise that the dynamic range can exceed human hearing. The size of the binary number determines the extent of lost information and determines the fidelity of the reproduced audio.

One fundamental problem of PCM digital audio recorders is the recording of all frequencies with equal importance. PCM operates in a time domain; the binary numbers represent a time-series of sampled voltages. The frequency domain of all the components of the time-series have the same resolution. In playback, PCM ignores the frequency dependent sensitivity of human hearing, and gives far too much weight to high frequencies (above 12 kHz) in comparison to low frequencies (below 4 kHz).

Various algorithms have been developed to compensate for the human listener. These algorithms modify the digital signal to more closely approximate natural sound. "Lossy" audio coders save storage space by eliminating data that is redundant or unnecessary to reproduce good sound quality, which is why they are also known as "perceptual" coders. Dolby Digital, for example, can reduce the audio data up to a factor of 15:1 compared to the source PCM audio data.

Speakers

Surround Speakers

Direct radiating speakers in a surround sound system provide very precise sound imaging.

Dipolar and bipolar speakers provide a diffuse sound without creating a specific point source. Dipolar speakers have two identical drivers mounted on opposite sides of the cabinet operating 180 degrees out of phase. Bipolar speakers have two sets of identical drivers mounted on opposite sides of the cabinet operating in phase.

Surround channel speakers are similar to bookshelf speakers. Surround speakers should have similar response characteristics to the other speakers in a surround sound system to present a uniform sound environment. This is also referred to as “timbre matching”.

NOTE: When referring to loudspeakers, the term "Q" is a measure of directionality. At low frequencies, the Q will always be low. At higher frequencies, it gets larger, depending on the size of the drivers involved. Thus, Q is a measurement of frequency-dependent radiation pattern and polar characteristics. Q is also a measurement of the slope of any peaks in loudspeaker, equalizer, or microphone frequency-response curves.

SubWoofers

A subwoofer is a special type of speaker that reproduces only the lower portion of the audible frequency spectrum (usually from 120 Hz down to the lower limit of hearing, 20 Hz).

There are two types of subwoofers. Powered subwoofers have a built-in amplifier. Non-powered subwoofers require an external amplifier and may be connected to a separate amplifier or to the main sound system. Subwoofers may be contained in a variety of different enclosures.

A subwoofer operates in an omnidirectional manner, meaning it can be placed almost anywhere in the room. The human ear cannot locate the origin of sound waves below 80 Hz, so low frequency sound waves have no apparent direction in a closed room. The Dolby 5.1 digital surround sound systems allocate a specific low frequency effects (LFE) channel for subwoofers.

Bass Management

In a surround sound speaker system, the higher frequencies are distributed to the speakers around the room: Left, Center, Right, Left and Right Surround, and in a 7.1 system, Left and Right Rear Surround. The lower, or Bass frequencies for all of these channels can be directed through a Bass Management circuit, which outputs to a subwoofer.

Producing a consistent bass response from all the channels while maintaining the integrity of the soundscape is impossible without bass management. Bass management electronically imposes a bass frequency crossover (typically 80 to 120 Hz) on some or all the channels, and redirects the bass frequencies from each of the channels to the subwoofer.

Large and Small

Each speaker in a surround system can be set for large or small. When you select "Large" in a bass management function, it means all of that channel's sounds, the entire range of frequencies, is directed to that speaker. When "Small" is selected, the bass sounds (below the crossover frequency) are filtered out of that speaker and directed to the subwoofer. Physically small speakers may not be able to reproduce very low frequencies without introducing distortion. Bass management permits each speaker to operate in the range for which it was designed.

NOTE: Bass management sends the bass content of small speakers to the front speakers if they are large and there is no subwoofer.

Speaker Set-up Suggestions

The following are suggestions for setting up the bass management feature of the surround system.

SPEAKER TYPE	SETTINGS
Front Floor-Standing Type With Built-In Subwoofer And No Additional Subwoofer	Select front speakers as "Large" Select Subwoofer as "Not Present"
Front Floor-Standing With Built-In Woofer And An Additional Subwoofer	Select front speakers as "Large" Select Subwoofer as "On"
Any Speaker System That Does Not Have A Subwoofer In The System	Select front speakers as "Large" even if it the speakers are physically small. Select subwoofer as "Not Present."
Front Bookshelf Speaker With Small Woofer or a Separate Powered Subwoofer	Select front as "Small." Select Subwoofer as "On"
Front Bookshelf Speaker With 8" Woofer Or Dual 6" Woofers	Set the front speakers as either "Small" or "Large". Set the subwoofer as "On."
Center Speaker	Center channel speakers do not carry as much bass as a subwoofer or main speakers, set the center speaker as "Small."
Surround Speakers	For bookshelf, on-wall or in-wall speakers as surrounds, select "Small." For large floor-standing surround speakers, select "Large."
Subwoofer	Select subwoofer as "On." The subwoofer reproduces the Low Frequency Effects bass channel along with the bass of any other speakers that have been selected as "Small."

Standing Waves

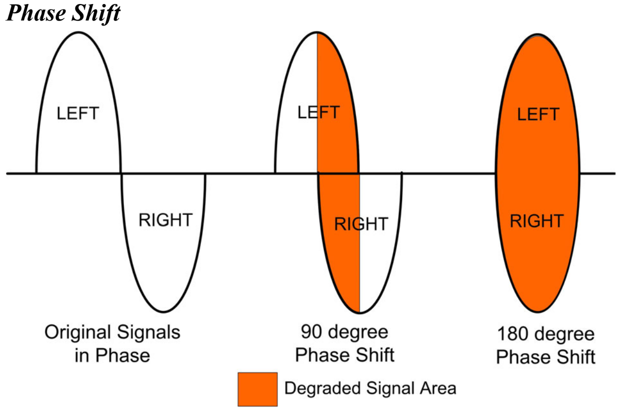
A standing wave is a low frequency distortion that happens when a particular frequency has a unique relationship to the size or shape of a room, resulting in a increasingly resonating sound. The original signal is amplified to a loud, booming bass that overpowers all of the other frequencies. This wave phenomenon results from the interference of sound waves of the same frequency and kind traveling in opposite directions. For example, if a string is stretched between two supports and a wave sent down its length, the wave is reflected and sent back in the opposite direction, resulting in a standing wave. Standing waves can be also seen in columns, tubes, plates, rods and diaphragms that are the component parts of musical instruments.

Standing waves occur in rooms by low frequency sounds with long wavelengths. The reflected sound wave is nearly in perfect phase with the original wave and creates a fixed spatial pattern of nodes and antinodes. The nodes are experienced as dead spots, points of nearly complete cancellation. The antinodes reinforce and amplify the original sound, creating the booming bass sound. Standing waves can be reduced or eliminated by careful placement of subwoofers, rearranging furniture, and equalization adjustment.

Phase

Phase is a specific point in a sound wave, measured from a zero point and given as an angle. Many powered subwoofers feature a phase switch allowing a change of phase from 0 degrees to 180 degrees (180 degrees is exactly one half of a complete cycle). When two audio signals are out of phase, they cancel each other out resulting in a weak signal (or no signal at all if they are 180 degrees out of phase). This occurs when one sound wave is at its peak while the other is at its bottom point, the trough. Similar to adding a negative one to a positive one, the end result is zero. By switching the phase, the sound waves are aligned and reinforce one another instead of canceling each other out. When connecting speakers, ensure that the phase (+ and -) connections are correct and consistent. Many powered subwoofers feature a phase switch, allowing the user to change the phase from 0 degrees to 180 degrees. By switching the phase, sound waves from the subwoofer can be aligned with other sound waves to reinforce one another instead of canceling each other out.

Phase Shift



Surround Sound Speaker Placement

Front Speaker Placement

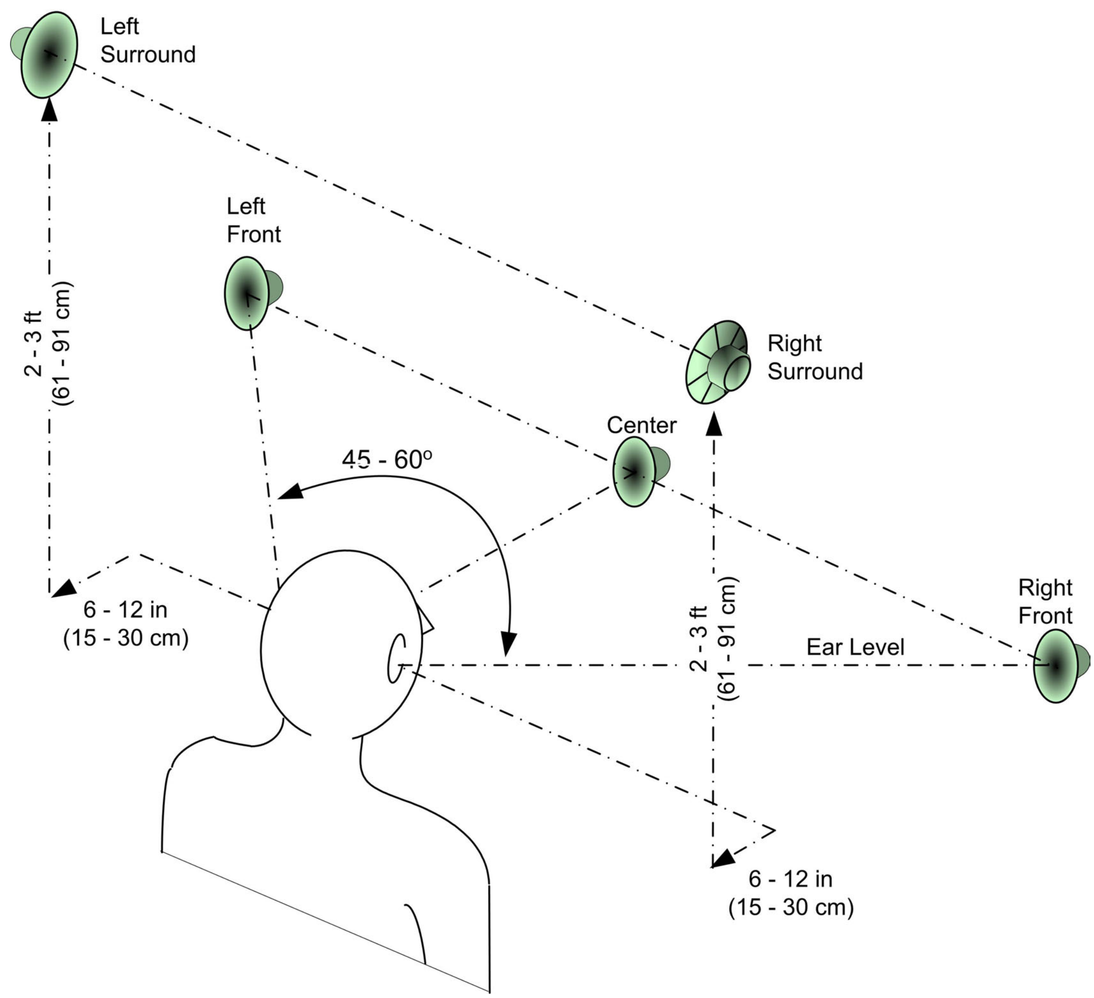
The front speakers are arrayed across the front of the viewing/listening area. These three speakers should be as equidistant as possible from the center listener position and on a plane with the listener's ears.

Try not to position the center speaker closer to the listeners than the left and right front speakers if possible.

The left and right front speakers are positioned at an angle of 45 to 60 degrees to the center-most listener. An angle nearer to 45 degrees is preferred if the system is used primarily for viewing movies. This position approximates the circumstances under which film soundtracks are mixed. A wider angle, nearer to 60 degrees, is recommended when the system is used primarily for listening to music.

The three front speakers should be as close as possible to the same height, at or near ear level.

Ideal 5.1 Speaker Placement (Subwoofer not shown)



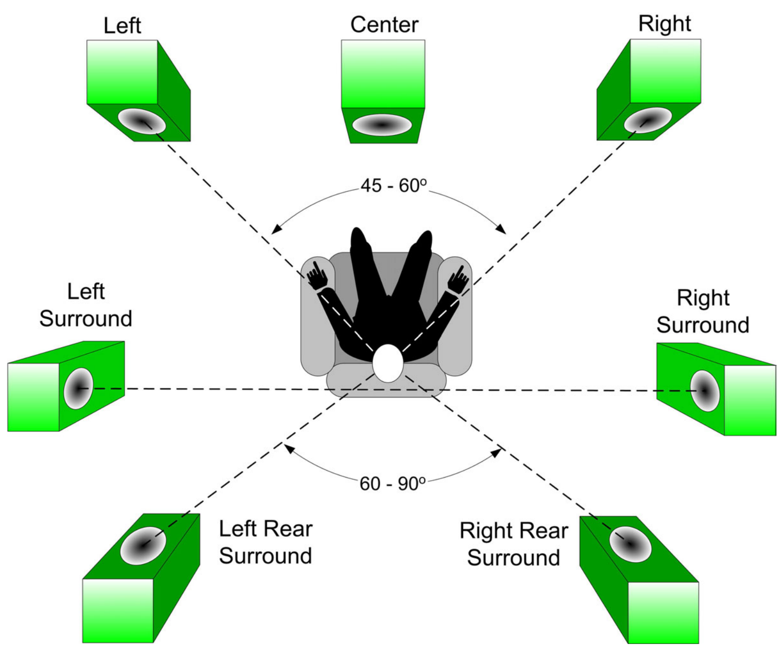
Surround Speaker Placement Optimized for Movie Soundtracks

The surround speakers should be placed alongside and slightly to the rear of (but not behind) the primary seating position, two to three feet above ear level to help minimize localization effects, and aimed directly across the listening area, not down at the listeners. This arrangement creates a surround soundfield throughout the listening area.

Surround EX Speaker Placement

Dolby Digital Surround EX (7.1) encodes Dolby Digital program with additional rear surround channels for playback over two additional surround speakers placed behind the viewer. Surround EX program material is fully compatible with regular Dolby Digital 5.1 playback (the additional center rear information is split between the left and right rear surround channels). Rear channel speakers should be placed at an angle of 60 to 90 degrees to the prime listening spot.

Ideal 7.1 Speaker Placement (Subwoofer not shown)



Flat Response

A flat response is a theoretical ideal for audio components, representing a frequency response that does not deviate from a flat line (0 dB) over the audible frequency spectrum. A perfectly flat response is practically impossible in the real world. The interactions with the room and surfaces in the room alter the waveforms. A flat response from a home theater system is not very pleasing to the human ear. Ideally, you first attain the flattest response possible, and then adjust the system to the listener's preference.

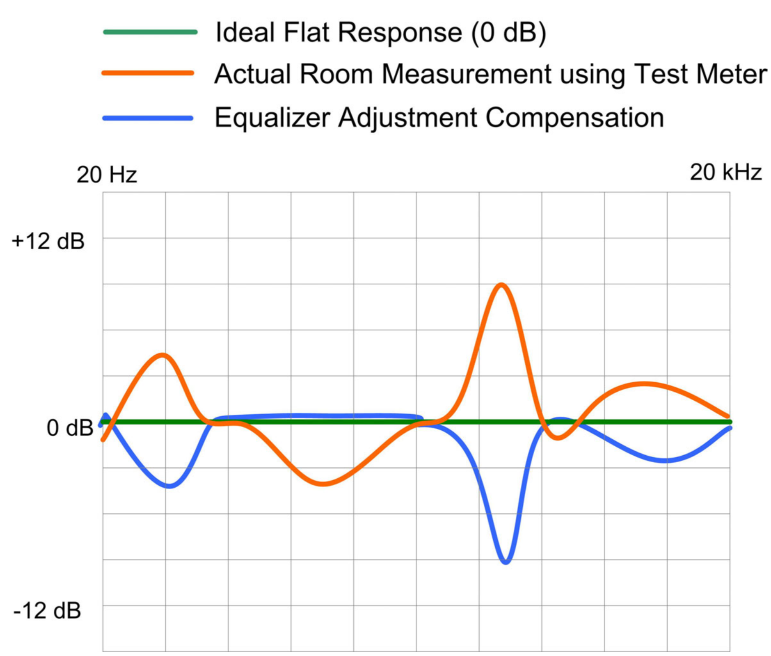
Generally, audio component response specifications are given as a function of the frequency range. For example, a speaker can have a flat response of 2 dB within the range of 50 Hz to 18 kHz. This means that the speaker is within 2 dB (at the limit of perception) of a flat response within the given frequency range. All speakers will fluctuate above and below an ideal flat response. Speakers that remain within two or three dB of a flat response are very linear and nearly flat. Most speakers have a drop off in response in the very low and very high frequency ranges. These frequency areas are outside of the critical midrange frequencies to which human hearing is most attuned.

Equalization

Equalization is a change in the frequency response of an audio signal. This is accomplished by adjusting the amplitude of the signal within a range of frequencies. The aim of equalization is to achieve a flatter frequency response (more closely matching the original signal), by compensating for the room acoustics and speaker deficiencies.

The following diagram illustrates the use of equalization on an audio signal. The listening area is first tested with a pink noise generator and a sound level meter to determine the flatness of the room. Equalizers are then used to increase or decrease those portions of the audio spectrum that fall above or below the ideal response of 0 dB. The result of this compensation is a flatter response, more closely resembling the original audio recording as intended. The equalizer settings mirror the actual room readings. In practice, sound engineers would not increase the amplitude of frequencies that fall below the flat response line. Generally, an artificial increase of those frequencies can introduce undesirable distortion.

NOTE: Equalizer adjustments should only be performed when the actual room response is known.



Room Adjustments

After completing the hardware hookup procedures, perform these procedures to set the surround sound default values according to the actual environment of the listening area.

Required Test Equipment:

- Sound Level Meter
- Tape Measure

The following equipment is recommended to optimize performance relative to the listening area, speaker arrangement, and the user's preferences:

- Audio Spectrum Analyzer
- Pink Noise Generator
- Avia Guide to Home Theater, from Ovation Software (www.ovationsw.com)
- Video Essentials, from DVD International (www.videoessentials.com)

Setup Procedures

NOTE: The Avia Guide to Home Theater and the Video Essentials DVD discs provide information and extensive instructions for setting up a home theater environment. Follow the instructions provided.

NOTE: Part of the setup process is determined by the source of the sound to be reproduced-either movie audio, or other kinds of audio-and the types of speakers used.

The first thing to recognize when constructing a home theater system is that the room itself can be more important than the equipment. The correct speaker placement, furniture and acoustical treatments can turn a good sounding room into a great sounding room. Balancing the speakers, the room and the right electronic equipment raises the home theater experience to an amazing level.

This section focuses on performing the proper adjustments to a room and audio system, providing your client with the best possible home theater acoustics. Covered here are speaker placement, equalizers, and compensation techniques to overcome poor room acoustics. Regardless of the specifications, a system is not great until it sounds great. What most people desire is an even room sound, the bass and the treble going together smoothly and in a complimentary manner. One portion of the sound spectrum overriding another is not the optimum choice for a great listening experience.

What We Hear in a Room

The first sound to arrive from an audio system is direct sound. This is also considered the on-axis sound. Direct sound occurs when the speaker is angled toward the listener, and this is the best possible sound the speaker produces.

Early reflections are heard next, which follow the direct sound by a few milliseconds and are not as loud as the direct sound.

After the reflections we hear the reverberation of the rest of the room. These reverberations can come from multiple directions at a lower sound level. Together, they cause a confusing soundscape if not dealt with properly.

Rooms also contain resonance that can emphasize and attenuate certain frequencies. The resonance depends on the location of the listener and the placement of the speakers. The effects of resonances are often experienced at the lower range of frequencies.

Very often we cannot overcome all of these deficits, however we can work with them and turn them in our favor.

We begin by selecting good quality speakers that provide a smooth (flat) frequency response over the given drivers operational frequencies.

Placing a speaker too close to a reflective surface, such as a wall, adversely affects the audio by presenting an inaccurate, heavy sound. The sound-reflecting wall acts like second speaker, increasing the sound level but decreasing clarity. This situation can be resolved by adding some absorption material.

You can use a variety of materials, such as acoustical foam and acoustic fiberglass. Even a drapery can help absorb the reflection. A room that is too acoustically “live” or “dead” does not sound good. A mixture of the two creates the most realistic home theater environment.

If you have a room that is relatively “live” sounding, use a speaker equipped with a horn tweeter for good directionality. It can be aimed towards the listener and away from reflective surfaces.

Speakers Placed in Cabinets

Although best placement for a speaker is about one to three feet from the nearest wall, we do not always have that option. Many home theaters incorporate cabinets to add a warm feel to the room and hide the electronic equipment. Placing a speaker in a cabinet can hinder speaker function, especially if the speaker is improperly positioned and mounted in the cabinet.

The speaker should be as close to the front of the cabinet as possible. As a speaker is placed deeper in the cabinet, more boundaries interfere with the sound.

How the speaker is mounted in the cabinet is important. If the cabinet vibrates, you should acoustically uncouple the speaker from the cabinet. This vibration can transfer from the speaker to the cabinet, reducing the full sound capability of the speaker. Placing the speaker on mounting posts (that may be provided) or on a platform designed to reduce vibration transmission can help.

Many cabinets also have a door covering the speaker. The interaction with this can have an adverse effect on the sound. Ensure that no part of the door covers the speaker drivers. Try to keep the door from covering the whole face of the speaker cabinet. This is not always possible, however if you have the opportunity to influence the design of the cabinet door, you could achieve better results. An important part of cabinet design is the choice of covering of the speaker opening. Do not use just any fabric. Use a fabric produced by an acoustics company. Companies that manufacture acoustic fabric have different styles and designs available that can appear like wood or have a unique pattern that can add style to the theater enclosure.

Stereo Imaging

Begin by playing pink noise through both speakers. The stereo image should be floating between the two speakers. While moving forward and back, the image should stay centered. While moving left and right, the image should move left and right correspondingly.

When listening to stereo music, an image between the two speakers should be audible. In most recordings, the vocalist is centered while the band is arrayed across the front soundstage. While moving left and right, the image should move left and right correspondingly. Use several different selections to perform this test. Selections are mixed with different interpretations of the sound engineer image design.

If the image is not as tight as desired, the absorption methods for wall reflections will help greatly. If a little more spaciousness is desired, allow for reflections and you can even add diffusers to help create the desired sound. Doing either of these is fine as long as the stereo image is not adversely affected. No matter what your preference, you always want to maintain a good stereo image.

Absorption and reflection can be added to any wall. Make sure it is used in the proper location for the desired room response. You do not want a heavy reflection off the back wall sitting right against it (like many seats in a home theater). Instead, add a little absorption directly behind the listener and diffusion to the sides. This can reduce the unwanted reflection that can ruin the stereo image while still providing a feeling of spaciousness.

A diffuser is a device that takes any incident sound and directs it in different directions. Some sounds return to the listener, while others do not. You can purchase diffusers from some of the acoustics companies or you can use things in the room such as bookcases, paintings, sculpture and other furniture that has surfaces of varying depths.

About Surround Sound

The same theories are applied to surround speakers as stereo speakers. Placement should be about six feet above the finished floor, and should present a stereo image in their placement around the room. For instance, if a plane rumbles directly overhead, you should hear the sound travel evenly from front to back. A single pair of surrounds should be placed slightly behind the main seating position. If you have additional rears they should be placed on the back wall, positioned between the front right and left speakers and directed toward the front of the room. If you have a single rear speaker, place it at the same height as the surround speakers, in the center of the back wall facing the front of the room. You should keep all of the surround speakers at the same height to give the sound an even feel as it passes throughout the surrounds.

Subwoofer Placement

Too much or too little bass creates an unsatisfying listening experience. The number of subwoofers, their locations and the listener position are extremely important to good bass. Frequencies below 120 Hz are omni-directional, the sound radiates throughout the room and the source cannot be localized. This works to our advantage in many rooms. Listeners need not sit in front of the subwoofer to get good bass response.

The location of the subwoofer is critical for the best response; the worst place to put a subwoofer is in the middle of a room. Moving the subwoofer closer to a wall or corner improves the overall room response.

Locating the subwoofer in the corner of the room can often provide the best room sound level. However, this position can also cause room modes to be more apparent and affect the listening position. If all of your listening positions sound good with the subwoofer in the corner, then you are doing well. Many times though, the best position for viewing a movie and listening to a surround system has the worst bass response. This is due to a room mode at that position that causes a loss of bass. The frequencies are actually canceling themselves out at that particular position. Meanwhile, people that may be sitting in off-center may have loud and muddy bass, because at this position, the room mode is doubling the amplitude of the frequency that is causing the increase of the volume. You can adjust for this situation a several ways.

First, adjust the position of the subwoofer until you have an overall good bass response throughout the room. As you move the subwoofer, notice that the locations with the most bass and the least bass changes. You are moving the subwoofer through the room modes. The goal is to find the best location to minimize the number of modes in the main seating areas.

For middle size to larger rooms, add a second subwoofer. The proper placement of these subwoofers together can create a great room response without a loss at the main seating position and an increase the response in the room corners.

Adding a second subwoofer can also remedy the room mode situation if done properly. Do not place the subwoofers on the same lateral plane, both subwoofers against the front wall, both subwoofers against the back wall, or exactly opposite one another. This actually defeats the purpose of adding a second subwoofer by increasing the room mode. Placing a subwoofer in the front right corner and one in the back left corner, or placing a sub up front and one on the side wall will better cover the room modes. Start by placing the subwoofers in the corners and gradually move them out. The best sound medium to use is pink noise because of the range of random frequencies produced.You can also purchase bass traps and place them through out the room in the corners and other locations. These are commonly placed in corners because that is the location of the highest level of bass energy. You can also consult acoustic software and let it determine where to place them. This also applies to any absorption and diffusion products.

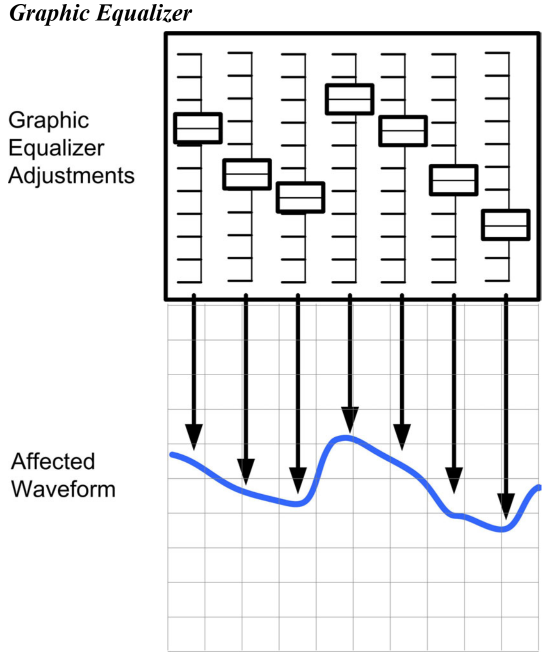
Equalizers

Equalizers are a kind of sophisticated audio filtering device, used to compensate for room acoustics, speaker deficiencies, and to achieve a flat response. The two types of equalizers are graphic and parametric.

Graphic Equalizer

Graphic equalizers typically have multiple adjustable settings that can range from one full octave to one third of an octave, and are used to achieve a flatter frequency response from an audio system in a specific acoustic space. Graphic equalizers use a set of predetermined frequency bands to adjust the amplitude of the waveform at specific frequencies. The center frequencies that are used and bandwidth affected by graphic equalization are predetermined and cannot be altered. This allows only a change in amplitude within each predetermined frequency range. The graphic equalizer allows a greater range of control than offered by the treble and bass controls.

Graphic Equalizer



Parametric Audio Filters

Parametric equalizers permit an adjustment to a signal's frequency response with complete choice over how to divide up the signal and adjust it using amplitude, center frequency and bandwidth (octave). When a signal is parametrically equalized, its amplitude is changed at a selected center frequency and over a selected range of frequencies on either side of the center.

There are six types of parametric audio filters:

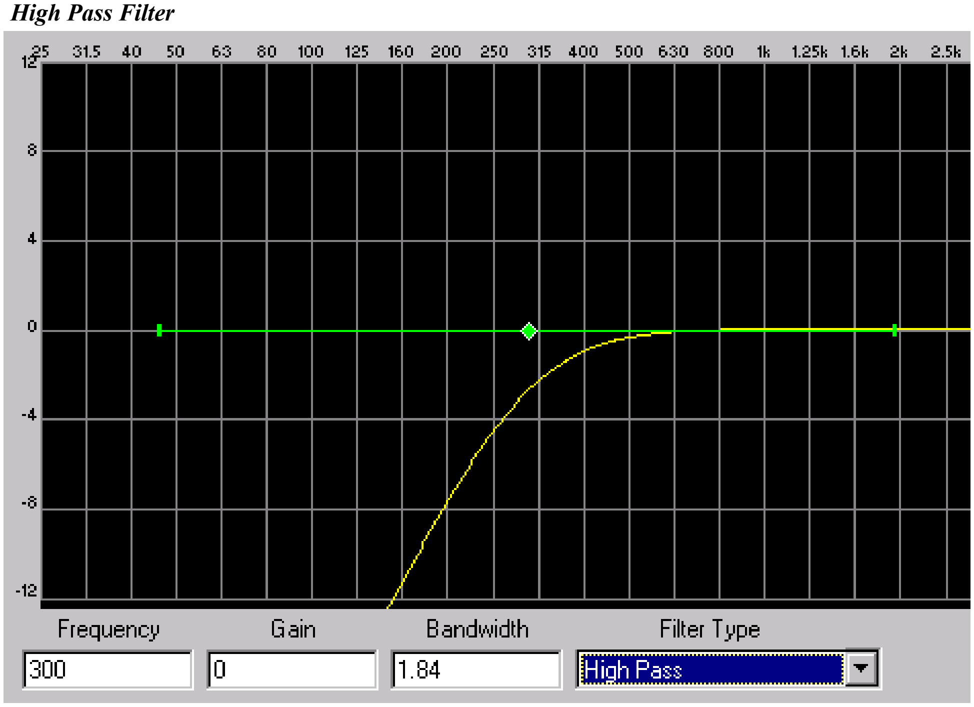
- High Pass Filter
- Low Pass Filter
- Bass Shelf
- Treble Shelf
- EQ
- Notch

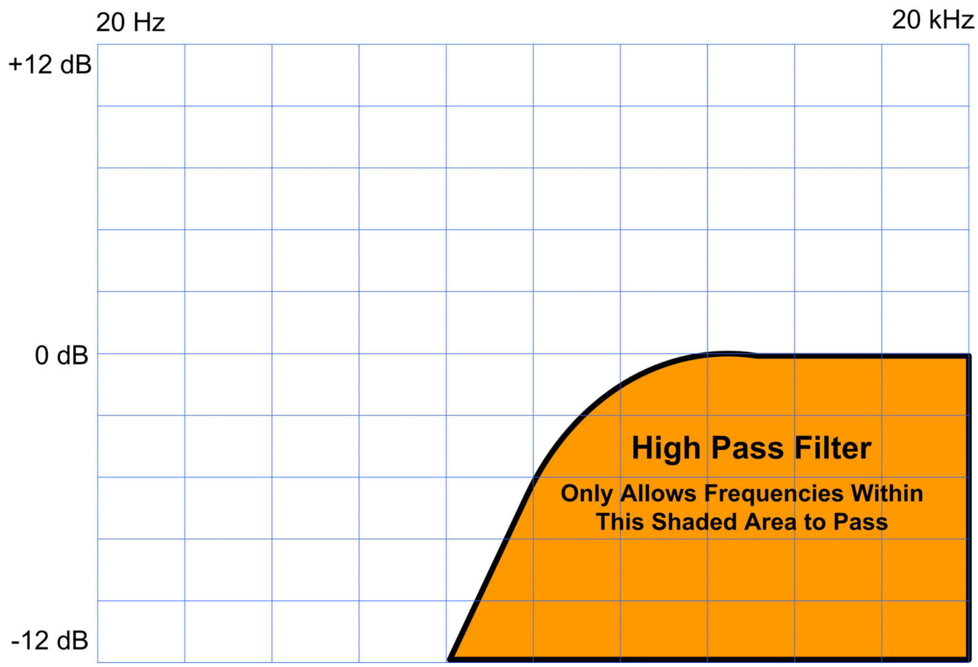
The filters may be used alone or in combination to adjust for room acoustics, furnishings, and speaker deficiencies to achieve the desired flatness.

High Pass

A high-pass filter passes on a majority of the high frequencies to the next circuit and blocks or attenuates the lower frequencies. Sometimes it is called a low-frequency discriminator or low-frequency attenuator. A high-pass filter circuit passes all signals that have a frequency higher than the specified frequency, while attenuating all frequencies lower than its specified frequency.

High Pass Filter

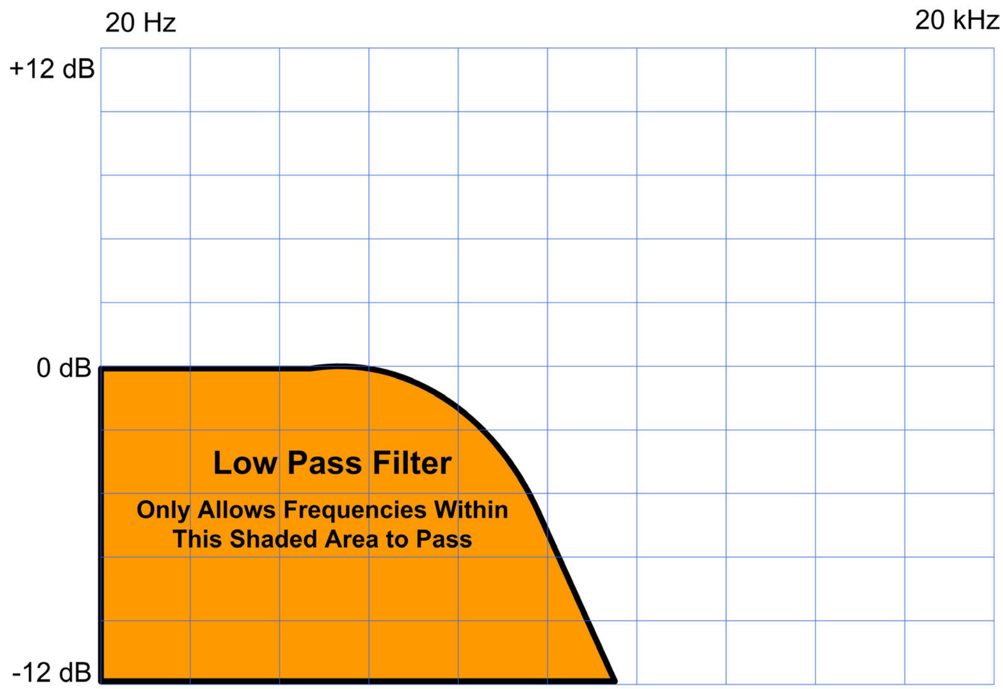




Low Pass

A low-pass filter passes on a majority of the low frequencies and blocks or attenuates the higher frequencies. Sometimes it is called a high-frequency discriminator or high-frequency attenuator. A low-pass filter passes all frequencies below the specified frequency, while attenuating all frequencies above this specified frequency.

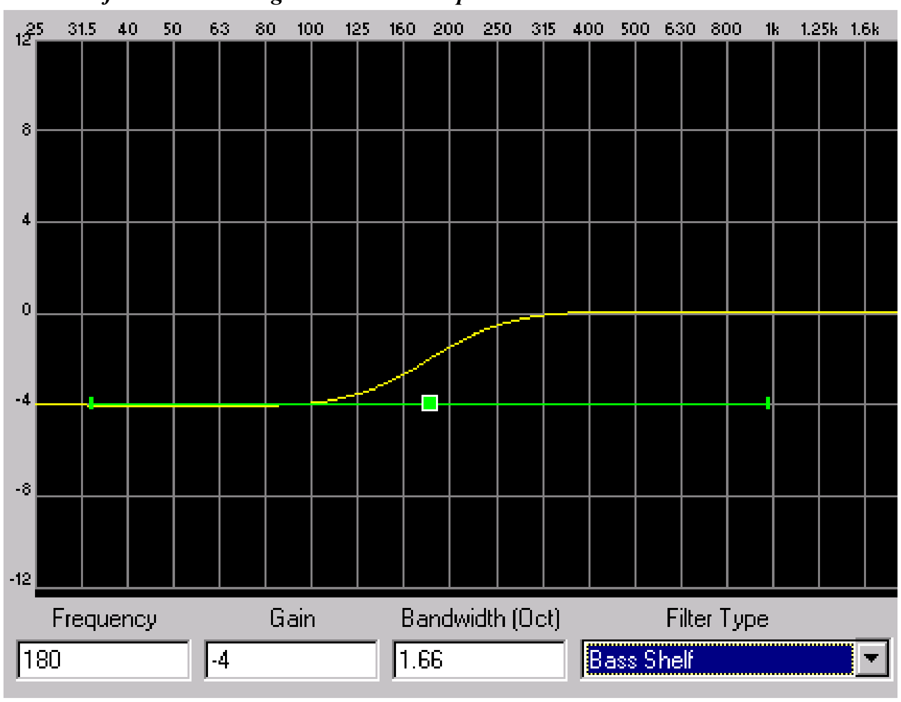
Low Pass Filter

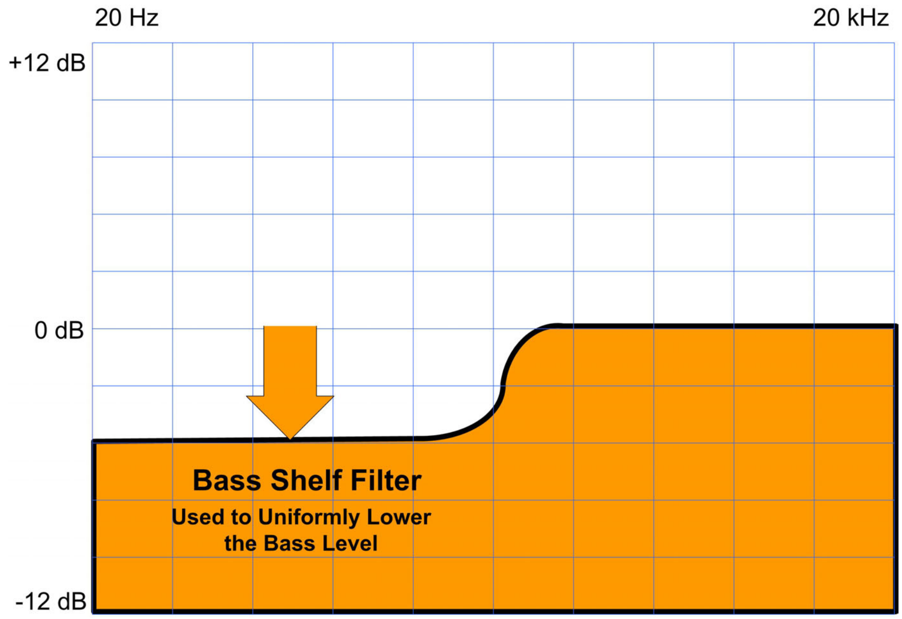


Bass Shelf

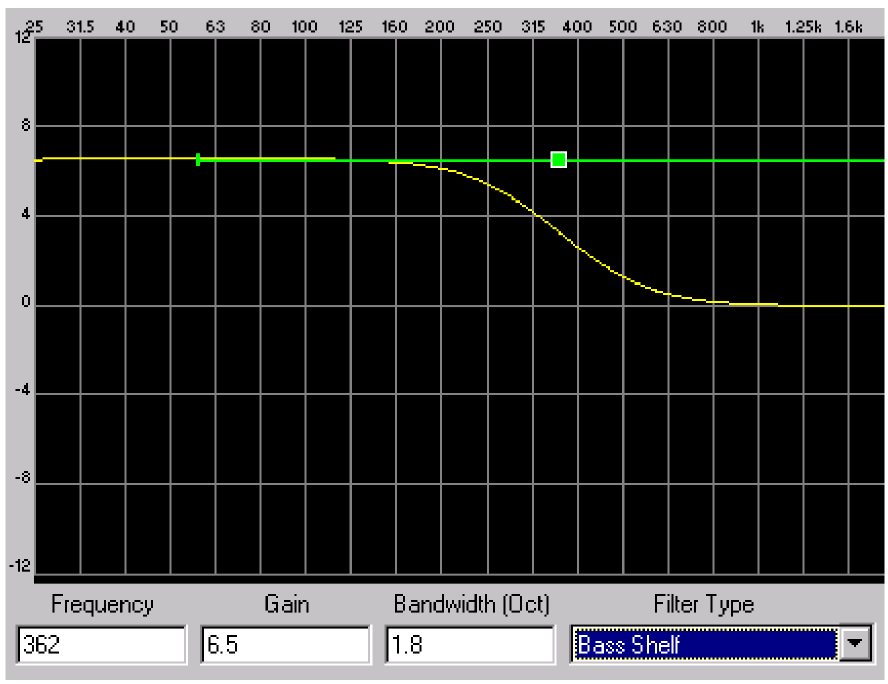
A bass shelf filter uniformly affects all low frequencies while not affecting high frequencies. If, for example, to increase the bass frequencies to a subwoofer, you can set the bass shelf filter to uniformly increase the amplitude of all bass frequencies. The bass shelf filter can also be used to uniformly decrease the bass frequencies to eliminate a booming bass sound

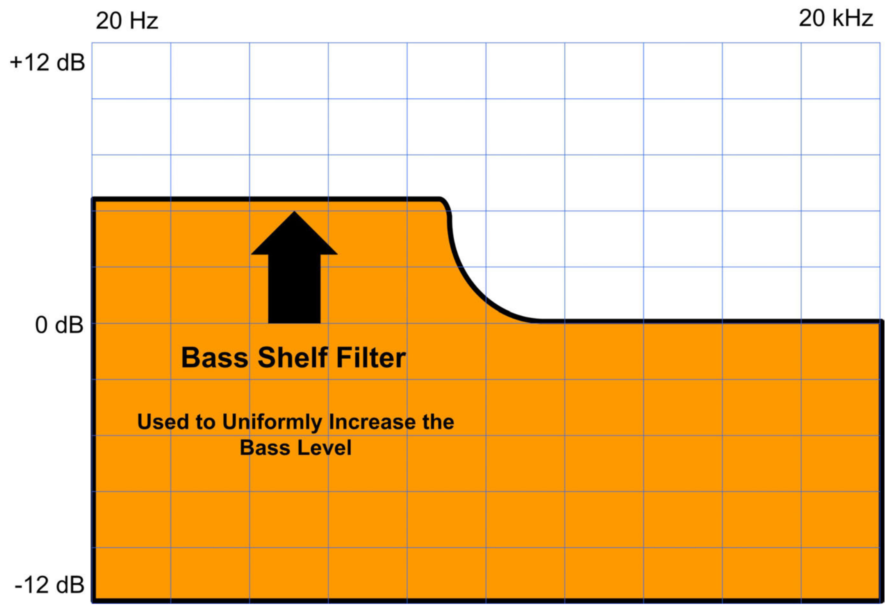
Bass Shelf Filter- Reducing Overall Bass Response





Bass Shelf Filter – Increasing Overall Bass Response

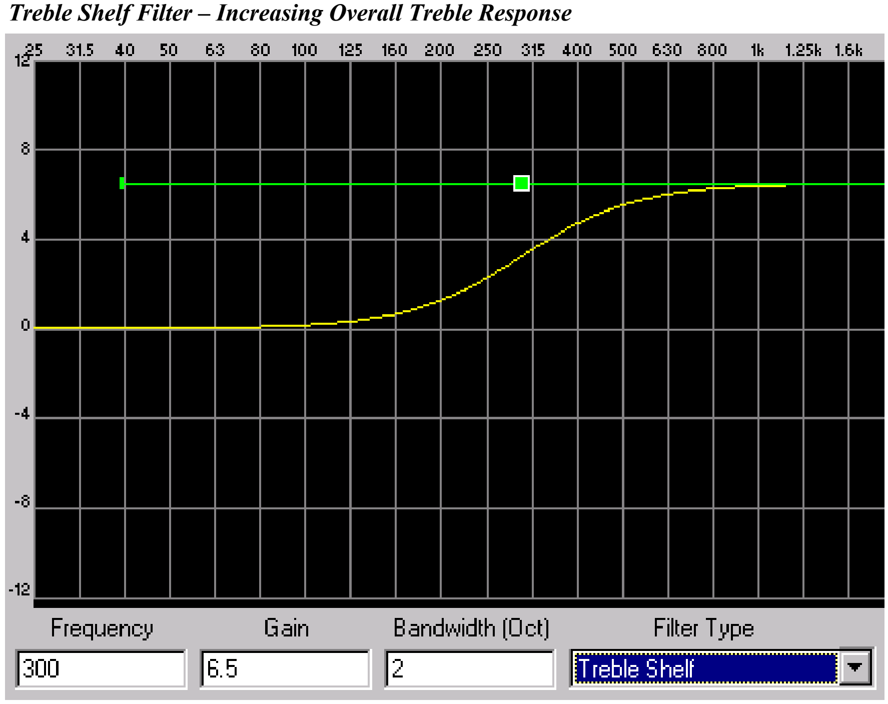


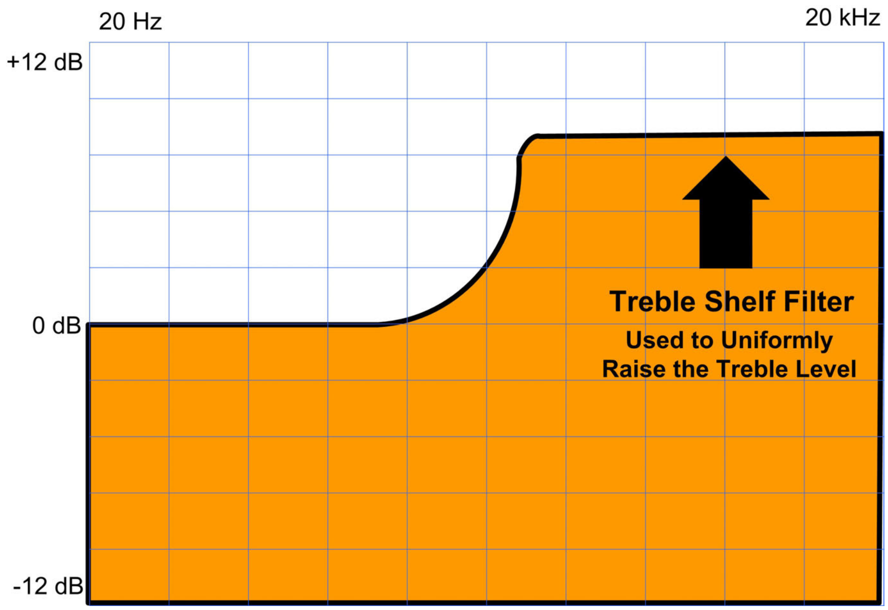


Treble Shelf

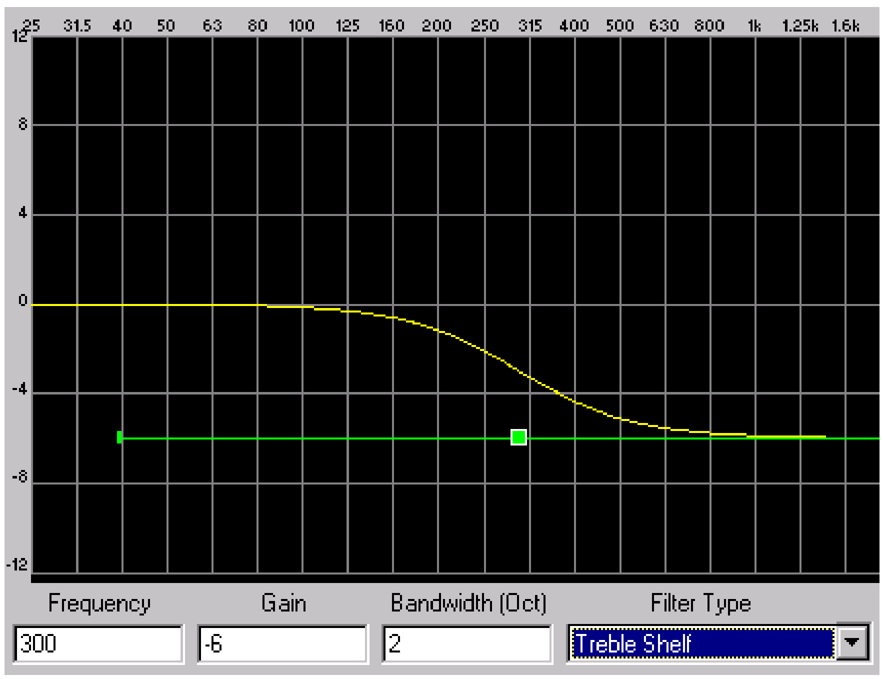
A treble shelf filter affects all high frequencies in a uniform manner while not affecting all low frequencies. Because bass frequencies have longer wavelengths, small speakers may sound distorted when trying to reproduce these frequencies. The treble shelf filter can increase the proportion of treble to bass, enabling the smaller speakers to produce a clearer sound.

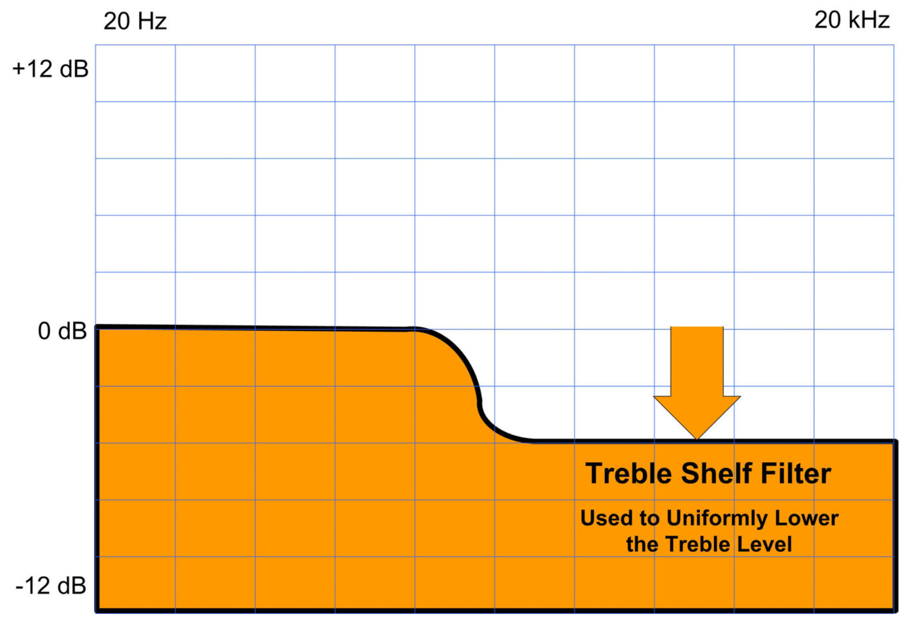
Treble Shelf Filter – Increasing Overall Treble Response





Treble Shelf Filter – Decrease Overall Treble Response





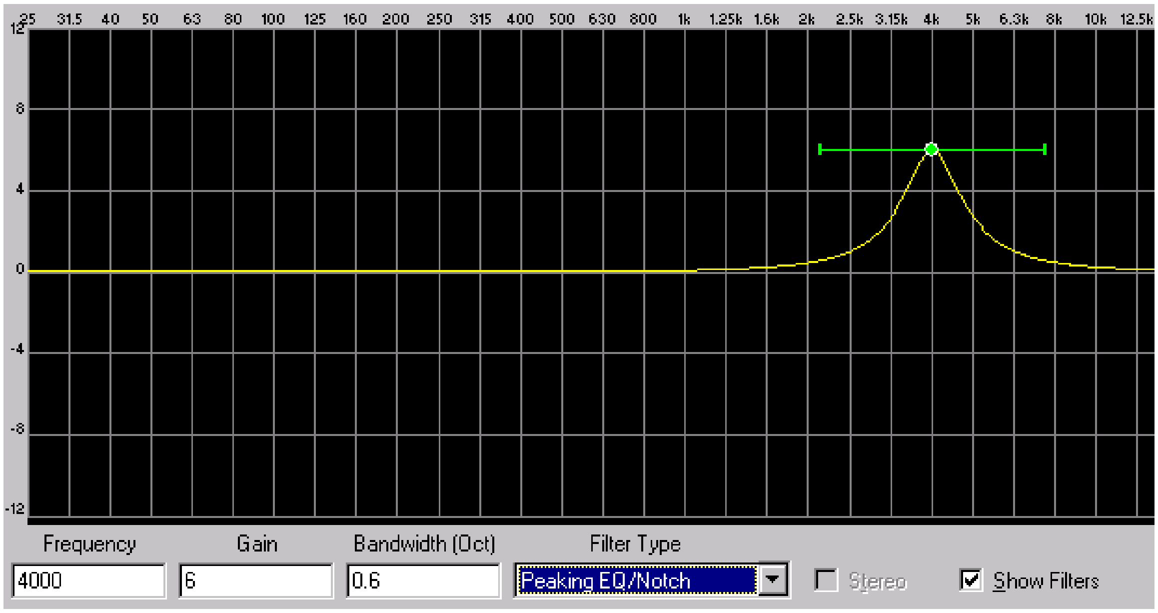
Peaking EQ

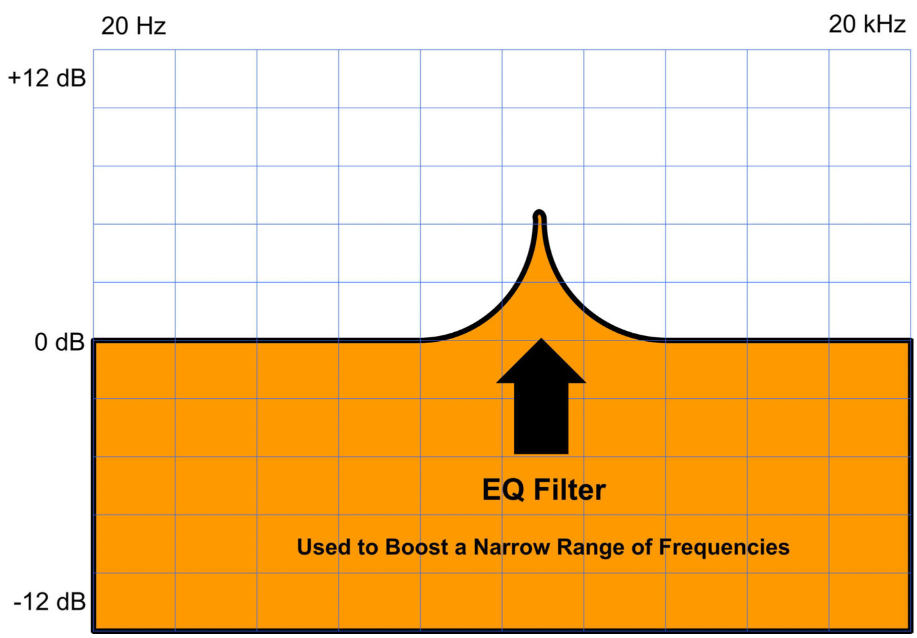
Peaking equalization (EQ) filters permit a precise amplitude adjustment of a selectable range of frequencies. The range can vary from a small slice of the frequency spectrum to a two-octave area. EQ filters allow a fine adjustment to compensate for room acoustics, noise, and speaker limitations. If, for example, the furniture or floor coverings absorb sound in the 4 kHz range. The frequency response of the system can be equalized by increasing the amplitude of the 4 kHz signals. If this increase in amplitude closely matches the loss of the room, then the frequency response is flattened, creating a more realistic sound.

In the following example, a parametric filter is used to adjust the response by boosting the signal by +6 dB at 4 kHz.

The bandwidth (octave) range is also adjustable, which determines the slope of the affected adjacent frequencies. In this case, the octave range has been set to 0.6. This affects adjacent frequencies from about 2 kHz to 8 kHz by gradually increasing the amplitude from 2 to 4 kHz and gradually decreasing the amplitude from 4 kHz to 8 kHz.

Peaking EQ Filter



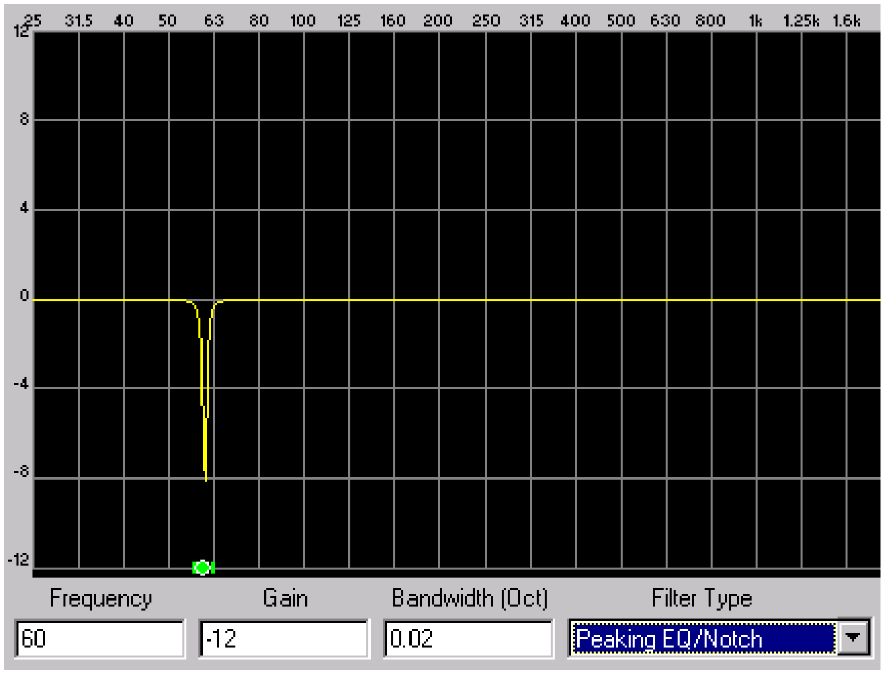


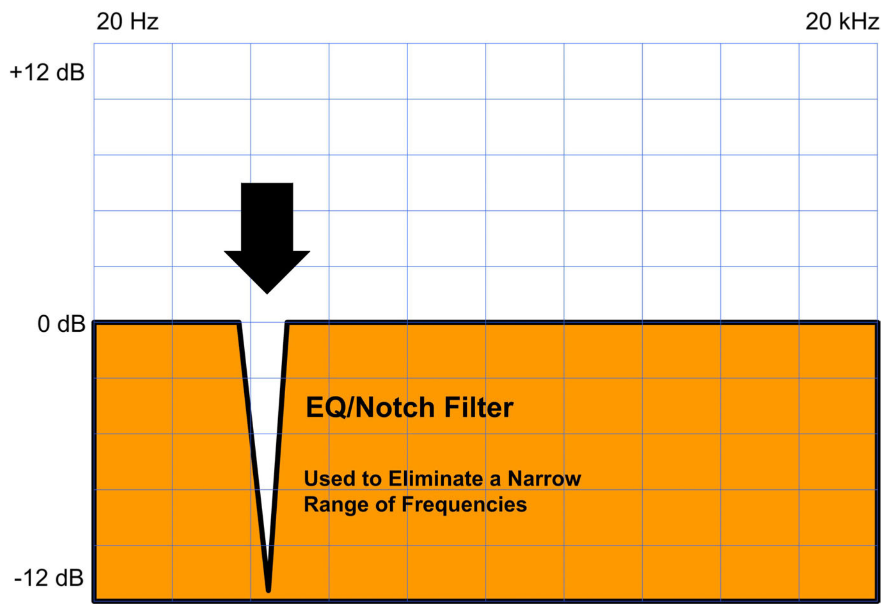
Notch Filters

Notch filters (also called band-reject filters) are used to remove an unwanted frequency from a signal, while affecting all other frequencies as little as possible.

An example of the use of a notch filter is with an audio program that has been contaminated by 60 Hz powerline hum. A notch filter with a center frequency of 60 Hz can remove the hum while having little effect on the overall audio signal. A notch filter ideally suppresses a single frequency component (the notch frequency) within the input signal, or a narrow symmetric window around the notch frequency.

Notch Filter



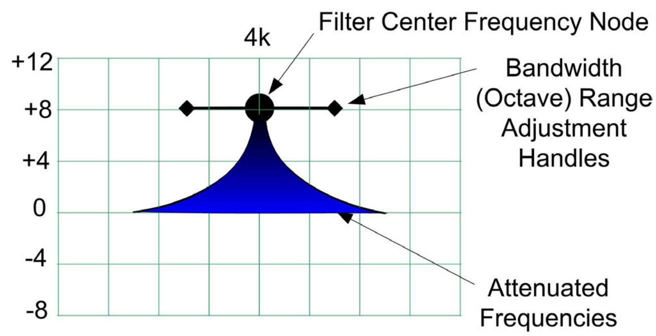


Parametric GUI

Parametric equalizers provide the user with an adjustable graphical user interface (GUI) in addition to an area in which to enter numeric values.

The interface provides a center node to select the center value of the frequency adjustment. The lines extending on either side of the center node allow for a bandwidth adjustment (also called an octave range or "Q"). In the following example of a parametric filter with center frequency of 4 kHz, a range of frequencies above and below 4 kHz are also affected. By shortening the octave range, fewer adjacent frequencies are affected. The area within the shaded portion of this diagram corresponds to the frequencies that are affected by this adjustment.

Example: 4kHz Parametric Filter



References

1. Information courtesy of ExtremeTech, www.extremetech.com
2. Information courtesy of *The Sonic Research Studio School of Communication, Simon Fraser University,* www.sfu.ca/sonic-studio/index.html
3. Information courtesy of University of Southern California, www-classes.usc.edu/engr/ee-s/321/lect1/tsld021.htm
4. Information courtesy of Sound & Vision Engineering Department, Technical University of Gdansk ul. Narutowicza 11/12, PL-80952 Gdansk, Poland, www.sound.eti.pg.gda.pl/SRS/tutorials.html
5. Information courtesy of Smyth, Mike. "An Overview Of The Coherent Acoustics Coding System." DTS Digital Surround White Paper.
6. Information courtesy of Smyth, S.M.F., Smith, W.P. et al. "DTS Coherent Acoustics: Delivering High Quality Multichannel Sound To The Consumer." 100th AES Convention, Preprint 4293, May 1996.
7. Information courtesy of Watkinson, John. The Art Of Digital Audio. Second Edition, Oxford: Focal Press, 1994.
8. Information courtesy of Barry Truax, editor, "The Handbook for Acoustic Ecology" Second Edition 1999, Simon Fraser University and Arc Publications.
9. Information courtesy of Dolby Laboratories, www.dolby.com/ht/Guide.HomeTheater
10. Information courtesy of AudioVideo101.com
11. Information courtesy of THX, Ltd, www.thx.com
12. Information courtesy of Yanman, www.yanman.com/HomeTheater
13. Information courtesy of Widescreen Review, http:// db.widescreenreview.com/surround/articles
14. Information courtesy of DTS, www.dtsonline.com

Glossary

A

Acoustic
The physical transfer of energy from a vibrating object to the surrounding medium (acoustic radiation) and the physical transfer of acoustic energy in a medium (sound propagation).

Ambience
The background-sound quality of a listening room, surround processor, and/or recording. The ambience of a recording is what gives it space and a sense of realism.

B

Bandwidth
Bandwidth is the difference between the upper and the lower cutoff frequencies. (An equalizer with cutoff frequencies of 200 and 2000 Hz has a bandwidth of 1,800 Hz.)

Bipole Speakers
Speakers sometimes used for surround channels to provide diffusion of sound. The speakers have two or more drivers, pointing in opposite directions or away from the listening position.

C

Codec (coder/decoder)
A pair of processing elements that code a signal prior to storage or transmission and then correspondingly decode the signal.

Crossover
The crossover splits up the frequency spectrum into pieces, which are then delivered to various speaker drivers. A crossover is required because one speaker driver cannot efficiently handle the full spectrum of sound. For Example: A two-way crossover splits the frequency spectrum into two frequency bands. The first from 20 Hz to 2 kHz and the second from 2 kHz to 20 kHz. The woofer reproduces the section from 20 Hz to 2 kHz and the tweeter takes over above 2 kHz. Two types of crossovers are active and passive. Active crossovers are adjustable and require a power source. Passive crossovers are not adjustable and do not require power to operate.

D

dB (Decibel)
Volume is measured in units called decibels (dB). A three-decibel (3 dB) change in sound pressure or volume is perceptible to human hearing. A 10 dB change doubles the sound pressure level.

Dipole Speakers
Speakers sometimes used for surround channels to provide diffusion of sound. The speakers have two or more drivers, pointing in opposite directions or away from the listening position.

Direct Stream Digital (DSD)
The modulation coding method used in the Super Audio Compact Disc (SACD) format.

DTS
A digital coding method used by Coherent Acoustics.

F

Fast-Fourier Transform (FFT)
Any complex sinusoidal wave signal (like an audio signal) can be mathematically represented. Some manufacturers have developed products using several FFT processes.

H

Haas (or precedence) effect
A psychoacoustic phenomenon in which the ear localizes sound based on the direction of the first arriving sound.

Harmonic Distortion
The most common form of audio distortion, it shows up as additional unwanted signals at multiples of the original frequency.
Thus, a l kHz tone may have second-order harmonic distortion at 2 kHz, third-order at 3 kHz, etc. These can continue upward to beyond the seventh or eighth order. The percentage total of all these measurements is called total harmonic distortion (THD) and is commonly used in audio test reports. However, different components generate different ratios of odd and even orders, making some sound better than others, even though their THD measurements might be the same.

L

LFE
Low Frequency Effects

Lossless
A term describing a data compression algorithm that retains all the information in the data, allowing it to be recovered perfectly by decompression.

Lossy
A term describing a data compression algorithm that reduces the amount of information in the data, rather than just the number of bits used to represent that information. The lost information is usually removed because it is subjectively less important to the quality of the data or because it can be recovered by interpolation from the remaining data.

LPCM (linear pulse code modulation)
A means to digitally represent signals in which the amplitude of a signal is coded as a binary number.

M

Masking
Under ordinary conditions, the process by which the threshold of hearing of one sound is raised by the presence of another. In both digital video and digital audio, a technique that allows a system to delete superfluous (inaudible or invisible) artifacts from a data stream by means of data reduction or data compression, enabling the system to transmit or store wide-bandwidth information within a much smaller bandwidth. Four uses of masking involve Dolby AC-3 Digital Surround Sound, MPEG video, DCC cassettes, and the MiniDisc.

Matrixing
A technique in which additional signals can be conveyed by altering the phase relationships of the signals.

MPEG (Moving Pictures Expert Group)
A committee developing audio and video standards.

N

Nyquist Sampling Theorem
Mathematical theory that a signal can be completely represented by a sampling signal with a frequency that is at least twice the highest input signal frequency.

O

Octave
1. In music, an octave is a doubling or halving of frequency with the bottom octave usually given as 20 to 40 Hz.

2. In parametric equalizers, it is the range of frequencies above and below a selected center frequency that are affected by the change in amplitude.

P

Parametric Equalizer
Type of equalizer that allows adjustment of frequency response with complete choice of center frequency, amplitude, and bandwidth.

PCM
Pulse Code Modulation. A method of transferring analog information into digital signals by representing analog waveforms with streams of digital bits.

Perceptual Coding
A coding method that exploits limitations in human hearing acuity to decrease the bit rate of the audio bitstream.

Phantom Image
A virtual sound source detected by a listener but not actually produced by a speaker at that location.

Pink Noise
Random noise (hiss) that has equal energy in each octave, and is used for setting and balancing a surround system. Pink noise is used as a test tone featuring equal amount of energy per octave of bandwidth. Compare this to white noise, which features an equal amount of energy per Hz (cycle per second) of bandwidth. Pink noise and white noise are common test signals with pink noise used frequently to adjust equalizers for a flat frequency response.

Pinna
The outer part of the ear

S

SACD (Super Audio Compact Disc)
A standard for high-density storage of two-channel CD and two-channel and multichannel SACD audio recordings.

SDDS (Sony Dynamic Digital Sound)
A perceptual coding method used in some theatrical motion picture releases.

Signal to Noise
(S/N ratio) the S/N ratio is the difference, in dB, between the noise floor of a playback component or sound recording and the loudest level it can achieve with inaudible distortion. The measurement is sometimes A-weighted because the ear is more sensitive to particular frequencies.

Soundscape
The illusion of three-dimensional space created by a surround system.

SPDIF
SPDIF is a acronym for Sony Philips Digital Interface and is sometimes abbreviated S/PDIF or S-PDIF. SPDIF is a digitial audio interface standard developed jointly by Sony and Philips that enables direct digital interconnections between separate digital audio components. Physically the connection between SPDIF compatible units can be made using optical fiber and optical "TosLink" modules, or electrically using coaxial cable and RCA type connectors.

Spectral Envelope
As used in the Dolby Digital encoder, a bit allocation routine to determine the number of bits needed.

Standing Wave
A low frequency distortion created when a certain frequency is reproduced whose size has some special relationship to the room in which it is produced.

Subband
A relatively narrow band of audio frequencies used by a perceptual codec to approximate the critical bands of the human ear.

Sweet-Spot
The sweet-spot is the prime listening position where an audio system is optimized, and is where optimal sound quality is encountered. Depending on the system, the sweet-spot may be large enough to accommodate multiple listeners.

.

T

Toslink
A type of fiber-optic cable connection that uses light beams to transmit digital information from digital audio components. Most digital-to-analog converters and digital surround sound processors (Dolby Digital and DTS processors) can connect to source components using Toslink cables and connectors. Using light beams to transmit digital information avoids potential interference from electrical wiring and other electrical devices that can interfere with traditional metal cables carrying electrical signals.

Total Harmonic Distortion
Total Harmonic Distortion (expressed as a percentage) is a kind of electronic echo. Harmonics are frequency multiples that appear when an original signal is reproduced by an audio system. For example, if a signal at 2 kHz is produced by an audio system, harmonics of this signal are located at 4 kHz, 8 kHz, 16 kHz and so on, with each progressive echo smaller in amplitude than the one before.

THX
THX is a tradename of Lucasfilm and used to specify standards of playback equipment such as receivers and loudspeakers.

W

White Noise
Similar to pink noise, except that white noise contains equal energy at each frequency point.

5.1
Five point one is a term used to describe digital surround sound audio formats such as DTS and Dolby Digital. The five stands for five discrete channels of sound information. These channels are full range and fully digital - left front, front center, right front, right rear or surround, left rear or surround. The point one is a sixth channel designated for a subwoofer. Other systems may have six speakers plus a subwoofer (6.1) or seven speakers plus a subwoofer (7.1).

Index

This page intentionally left blank.

Crestron Electronics, Inc.
15 Volvo Drive   Rockleigh, NJ 07647
Tel: 888.CRESTRON
Fax: 201.767.7576
www.crestron.com

Operations Guide – DOC. 6122
11.03