CINEMA SOUND

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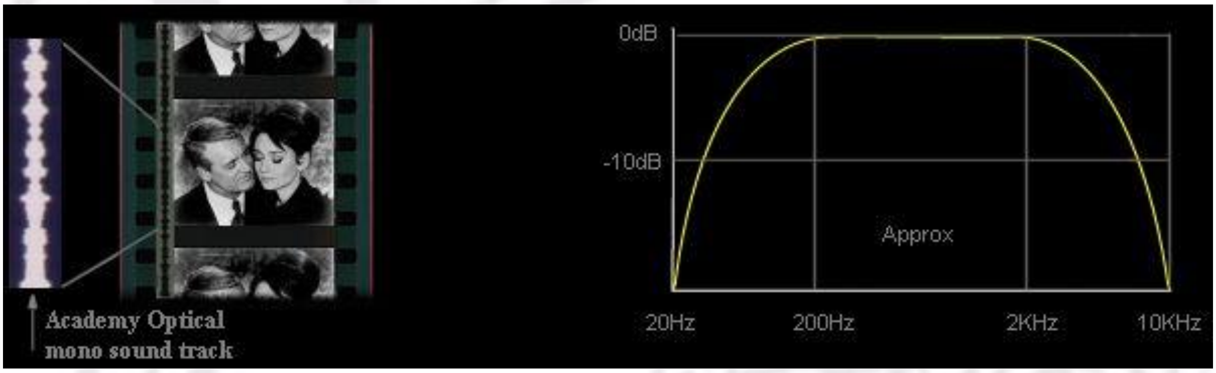
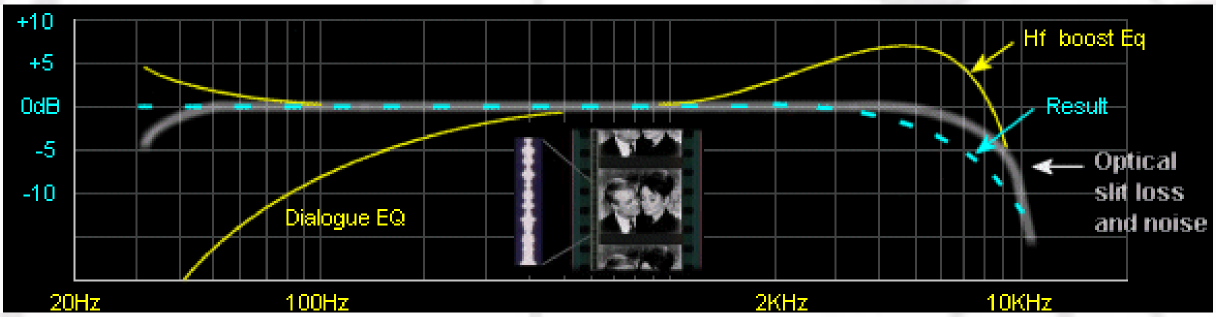
Cinema Basics

Cinema of the Golden Era: Going to the movies is a magical experience the whole of humanity shares and belongs to us all. The history of movies began in the 1890s and another 20 years would pass before the technology for sound was invented. Sound first arrived in cinemas in the late 1920s. This period till the late 1940s was described as cinemas Golden era.



Sound from the early optical sound track on the edge the film stock struggled to reach 6 kHz. Increasing the physical size of the optical track gave a larger signal but with less hi-frequencies. A smaller optical track gave a higher frequency response but with less signal level and greater noise. Noise hiss was also generated from the film reader and early valve pre-amplifiers. Putting a hi-frequency filter at the output of the pre-amplifier reduced the noise hiss, but further reduced hi-frequencies. The simplest solution developed in the 1930's was to boost the hi-frequencies when recording, and allow the play back noise reduction filter to achieve a close to flat response as possible to provide some consistency between the recording on film and playback in the cinema.



Because almost every cinema rushed to install sound as soon as it arrived, it was difficult to take advantage of technological improvements in sound systems designs that followed shortly after. Only a limited number of mixing studios and cinemas could afford to keep continuously upgrading to the latest technology. The majority of cinemas had poor sound systems in comparison to later developments. High and low frequency response was almost non-existent.

Academy characteristic: In 1938 a study of the 'worst case' cinemas in the US showed an average energy (frequency) response of (-7 dB at 40 Hz) (flat 100 Hz – 1.6 kHz) (-10 dB at 5 kHz) (-18 dB at 8 kHz). This is said to be the Academy Curve.

Mixing studios could then apply equalization to simulate the poor response of the 'worst case' cinemas, to be able to craft and evaluate the soundtrack as it would be heard by the majority of audience. This practice became the foundation (trend) for how soundtracks were and are prepared. Many variations of the Academy curve followed and later became re-named the X-curve.

The practice of modifying sound performance for the mass end of the cinema exhibition market has always been hotly debated, because it is based on assumptions and compromise. This practice also institutionalized lower performing sound systems as being the benchmark for the characteristic 'cinema sound' and later represented a blockade restricting hi-fidelity sound from being the adopted goal when technology allowed. This can be argued as being an economic rationalist solution, not a technical or scientific solution.

The un-foreseen and un-fortunate outcome from this compromised approach made it difficult to implement improvements beyond what was initially incorporated, and has basically remained for 35mm film until the 1970s. Even with these limitations, cinema sound from optical format was better by comparison to home record players and radios, until the early 1960s when superior hi-fi stereos became available.

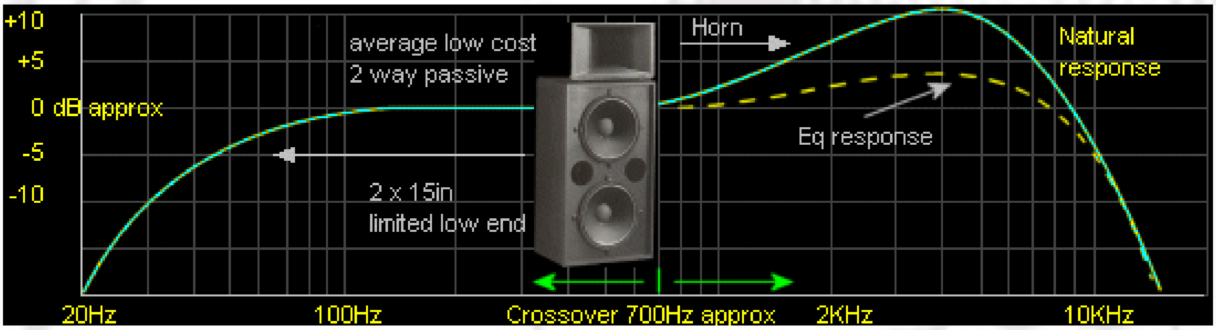
European film industry: It has been argued that pre WW2 European film technology was superior to the US. German cinema technology was said to have been favoring a flat response which came close to achieving 8 kHz. The expectation may have been that cinemas would aspire to full fidelity sound by installing the most advanced technology available, with no compromise for poor quality sound systems or acoustics.

The demise of the European film industry during WW2 is said to have resulted in the US dominating the world with its approach of a compromised alignment for the majority of cinemas which had poor quality sound systems and acoustics.

Early speaker systems: Early speaker systems achieved the best performance with the technology available, and most research on horn technology at that time has been unsurpassed. The aim was to achieve the highest efficiency and voice articulation which horns are ideally and uniquely suited to. Also valve amps of that time were approx. 30 to 60 Watts. This resulted in the efficiency and size of the horn speaker systems being taken to the technological limit. The narrow bandwidth from the mono optical sound track (100 Hz – 4 kHz) set the precedent for the speaker systems only needing to be 2 way passive without tweeters. The 15in speakers and horn were and still are crossed over at approx. 700 Hz.

Most cinema speaker systems have remained principally the same as when first applied. Two 15in speakers for the low frequencies, a compression driver with horn for high frequencies. Crossed over at approx. 700 Hz. Altec Lansing, Western Electric and other well-known companies aspired to make the highest quality sound systems with the resources available.

But many early cinemas could not afford the best quality systems. The hi-frequency compression driver with horn are approx. +20 dB more efficient than the 15in speakers. Sometimes it was difficult to correctly equalize both components to sound flat, because sound measurement equipment was rare and expensive. Cheap crossovers consisted of a simple capacitor and resistor only.



Many sound systems had to be aligned by ear. Because of the limited power of early valve amps, the horn driver was sometimes kept close to full efficiency with little attenuation. This was done to obtain as much sound level as possible, especially for large cinemas. Cinema screen and air attenuation, helped reduce the higher efficiency of the horn, bringing it back to a flatter response.

Cinema sound is recognized by the unique acoustical character (of the projected voice range) created by the horn systems used in all commercial cinemas throughout the world. Regardless of their limitations which will be discussed further on this page, large cinema horns are magnitudes greater in efficiency than domestic speaker systems.

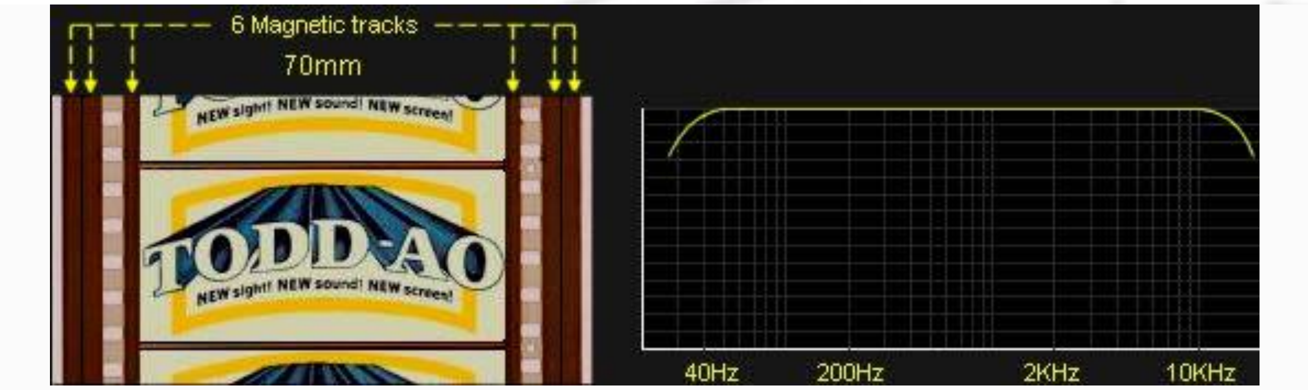
Cinema horns have an acoustical directivity that projects the sound image forward into the audience, giving a feeling of dimension to the sound. Horns also increase the articulation of dialogue when correctly applied. The characteristic sound of the horn gives cinema its unique sound quality. It is not possible for small domestic cone speakers in home-cinema systems to replicate the experience of large horn speakers in commercial cinemas.

Wide Screen

70mm Wide Screen: During the 1950s and 60s special 70mm films were produced and shown in cinemas equipped with super wide curved screens often exceeding 50 feet. A single cinema was as large as today’s multiplex, requiring to show the same film for many months or years to break even or make a profit.

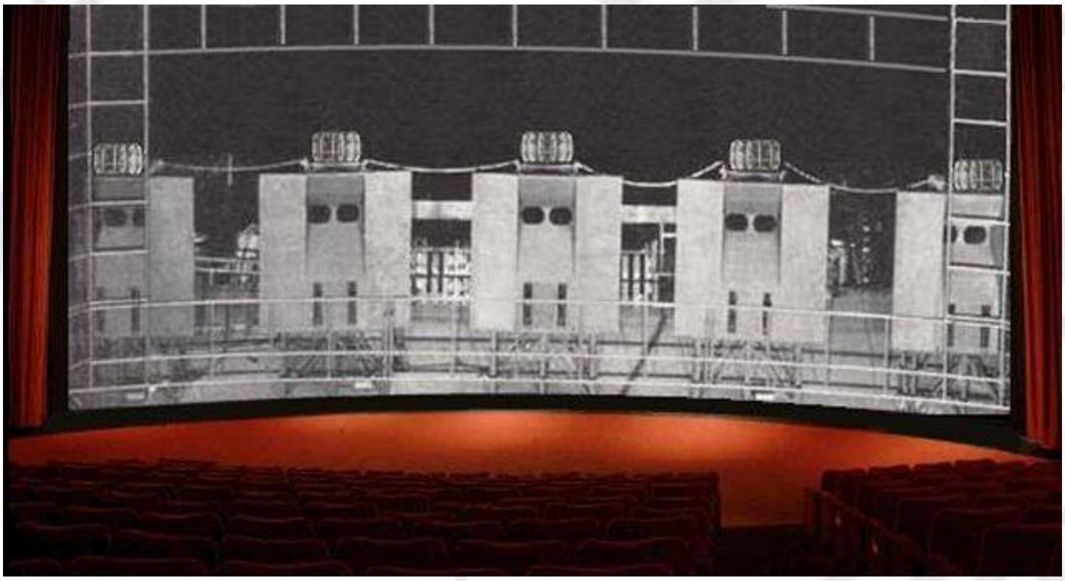
Prior to digital recording the standard test for high quality analogue magnetic multi-track recorders was to repeatedly record and play back and record over and over until noticeable degradation appeared, which was approx. 6 times. Only un-compressed digital recordings can achieve being repeatedly re-recorded with zero loss.

Many of the 70mm films had 6 independent channels of full fidelity magnetic sound tracks on the film stock. The sound fidelity was superior to most of the digital formats of today. The channel separation was so great that it was possible to independently record and play back different orchestral music on each track, with minimal loss or crosstalk. This separation enabled remarkable sound scaping that could emulate the experience of a real symphony orchestra on stage.

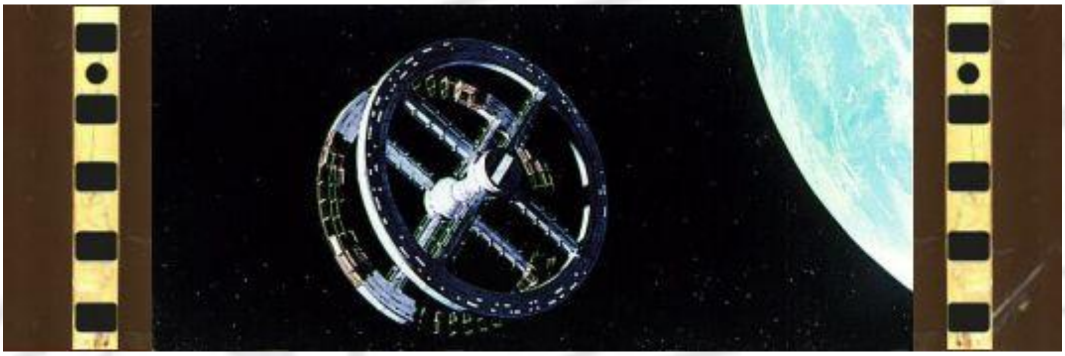


The cost of 70mm was more than 10 times greater compared to 35mm film. The process of applying the magnetic tracks to the 70mm film was extremely difficult, which was done at the end of the film stock's manufacturing process. The play back head on the projector had to be in perfect alignment, regularly inspected, meticulously cleaned and periodically de-magnetized and replaced when worn. This procedure was done with great pride and attention.

The only limitation was from partial de-magnetization of the sound tracks if the film stock was accidently placed against a transformer. But in all in all other respects the magnetic format was more robust and superior in performance to the optical format including today's digital.



Many of the large early cinemas equipped for showing 70mm cinemas in the 1950s were fitted with five speaker systems that were magnitudes greater in size than the majority of sound systems in cinemas today (left)–(left-center)–(center)–(right-center)–(right). The five screen speaker systems were able to give precise sound positioning and even spatial movement across the screen. The single surround channel could be automated to left side–right side–rear and overhead to simulate an aircraft flying around the room. However switching the surround channel was more commonly done with Cinerama.

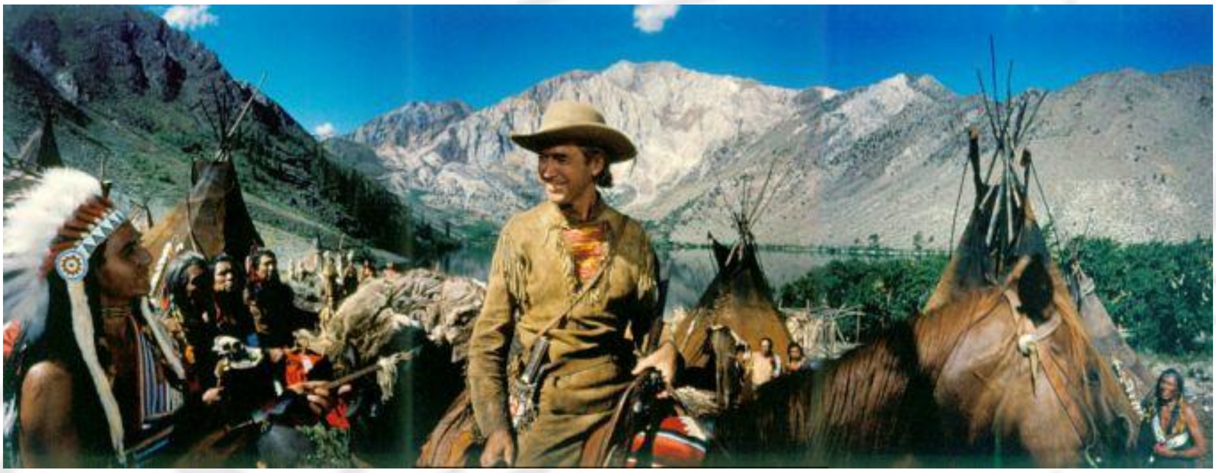


Stanley Kubrick's '2001 A Space Odyssey' had its world premiere on 2 April 1968 and became the most successful film in this format. But many people were unable to experience it as he intended, as most of the large wide screen cinemas with 70mm projectors were closing to make way for the 'economically rational' new multiplexes equipped with the new improved 35mm projectors with improved Lens (Anamorphic) that could also achieve wide screen formats. Sound fidelity was no longer considered important as the economical rationalist belief was that the majority of the new emerging consumer driven generation have become obsessed with image and brand identification.

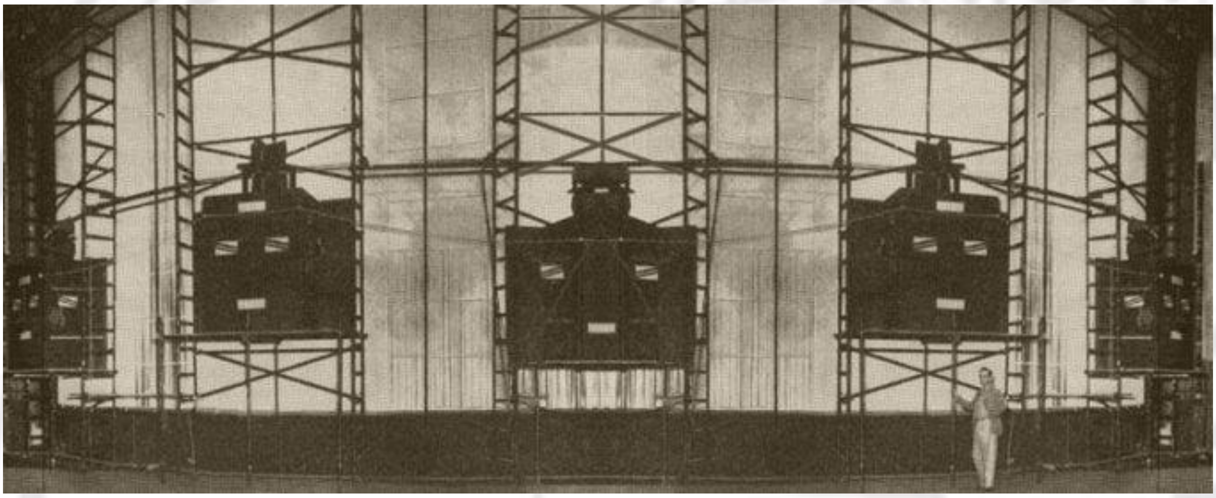
There are only a limited number of 70mm films being produced today (IMAX is 70mm) and sound reproduction is now rationalized to the modern digital format and uses the external DTS system. Today's compressed digital formats printed on 35mm film stock are far more restricted in fidelity with limited channel separation in comparison to the 6 analogue magnetic tracks on the earlier 70mm film.

Cinerama was without doubt the greatest achievement in cinematic history. Created by people of genius, madness and unlimited passion. Russia also had its own equivalent Kino-panorama. This is what going to the movies is about, 'To be beamed up and blown away'. The opening roller coaster ride caused everyone to tightly grip their seats and hang on for their lives, screaming with excitement. Paper bags were also supplied for emergency's. I was 12 in 1957 when I first experienced Cinerama.

Cinerama consisted of three projectors covering a giant 146 degrees super wide curved screen that included peripheral vision, enabling a 3D experience to be achieved. Cinerama was spoken of as an experience separately from just being seen. The only limitation was the annoying joining lines which were eventually minimized.



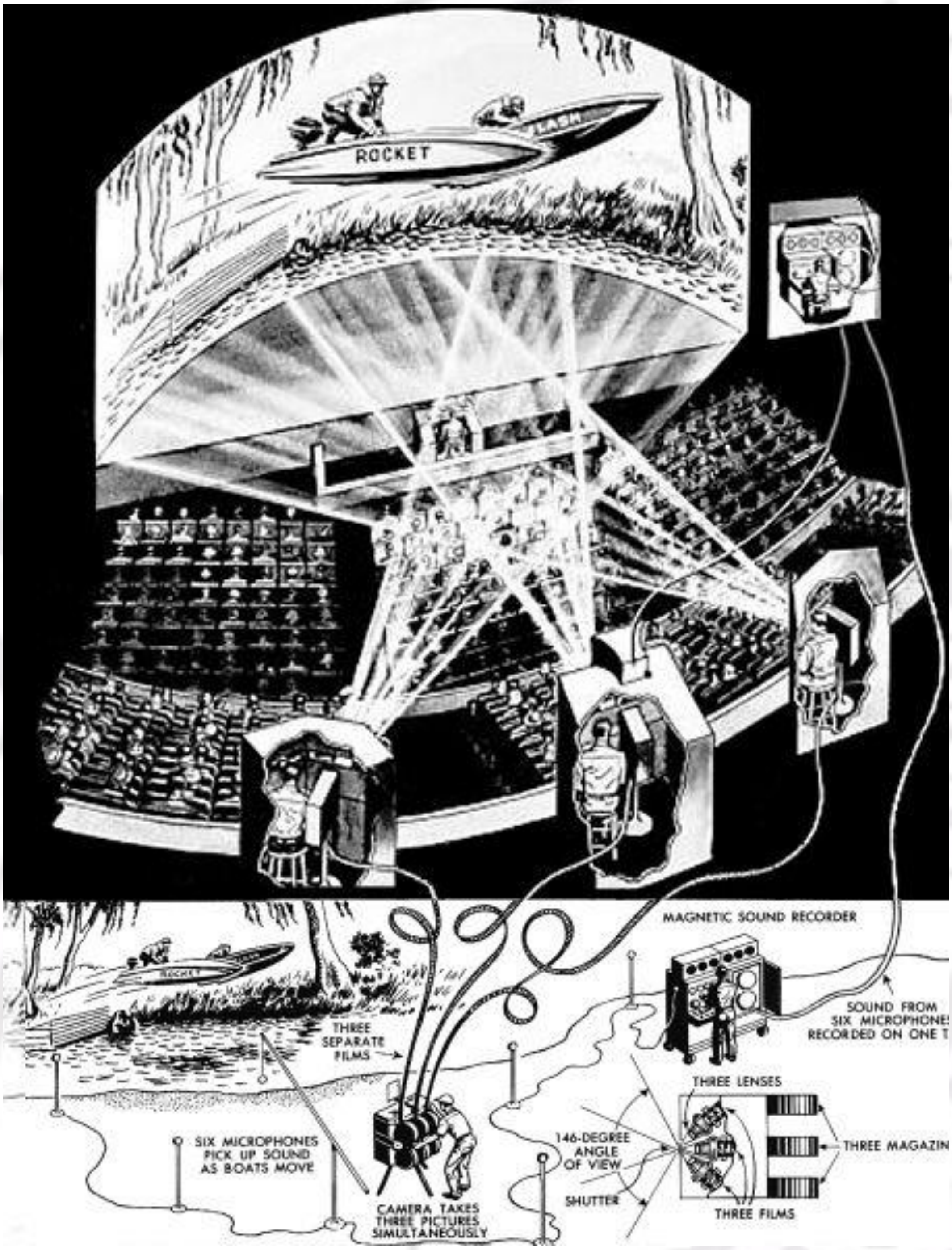
The sound was supplied from a separate 35mm 7 track magnetic tape machine that sync locked the 3 projectors. There were 5 giant speaker systems behind the screen as well as surround speakers, as shown in the below pic. After intermission there was a demonstration of the sound system effortlessly replicating a symphony orchestra as though it was actually present on stage. It was this effect of hearing an accurate replication of sound reality and detailed positioning of a symphony orchestra that resulted in electro-acoustic technology becoming the passion of my life's' work.



Many early NASA astronauts and engineers including John F Kennedy described how they were inspired by the Cinerama experience. Both Cinerama and Kino-panorama influenced the generation of the 1950s proving the impossible could be achieved and gave extra motivation for the race to put a man into space (April 1961) and then to the moon (September 16, 1969).

Unfortunately these large magnificent cinemas and titanic film productions are out of place in a modern superficial digital world of consumerism, where corporate and political motivation is now driven by self-interest and economic greed.

On behalf of all who are passionate about Cinerama technology we wish to express our gratitude to John Mitchell for his life work in recovering and restoring large quantities of the Cinerama stock and also David Culls for his entertaining historical accounts of the Cinerama era.

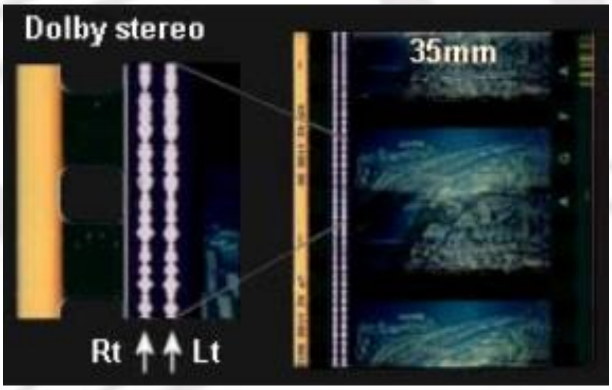


Cinema Sound Formats

35 mm has been the standard film stock for the majority of movies throughout cinema history. There had always been a variation of Anamorphic lens for 35 mm film for achieving variations of wide screen aspect ratios. The most commonly known was Cinema-Scope. Over time the 35mm film stock and lens technology greatly improved. Economic rationalizing favored 35mm and this caused the high cost superior 70mm films with its 5 screen sound channels to be used less and less.

Dolby Stereo: Mono sound dominated the movie experience in the 35mm medium till 1976 when an international agreement to allow the Dolby two track optical format to become the new standard and included the Dolby A noise reduction. A matrix technique enabled 4 channels of sound to be achieved from the 2 track optical format (Left)–

(Center)–(Right)–(Surround). The tracks are described as Left total (Lt) and Right total (Rt).



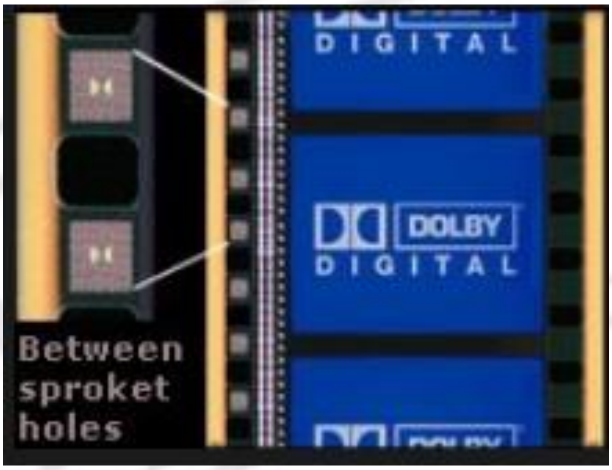
The matrix technique of obtaining four channels of sound from a dual format system had been used in the communications industry and with earlier quad vinyl recordings. However the existing matrix techniques could not enable backwards compatibly for film stock to be read from existing mono only reading projectors.

A cleaver and more complex matrix system was developed that enabled backwards compatibly. Over time electronic IC (integrated circuit) and component technology had improved, enabling higher fidelity with lower noise. Bass performance was also improved referred to as OBE (Optical bass extension). Many cinemas started to add an independent bass extension speaker. By 1986 this final optical sound improvement became known as 'Dolby SR' (Spectral Recording) and remains in place to this day.

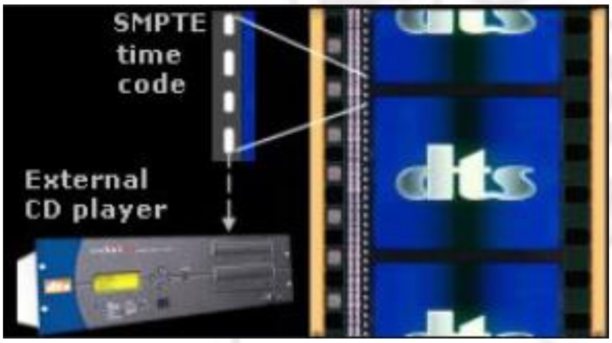
Digital sound formats: Between 1990–1993 four different and competing cinema digital sound formats were developed of which two remain in popular use. The (.1) sub-bass LFE (low frequency effect) was added which behaves as a separate channel limited to approx. 250 Hz. All systems automatically default to the analogue optical system as a security back up. The Dolby digital system dominated due to its simplicity of application and being more economical to manage.

Unfortunately many cinemas have limited fidelity 2-way passive speaker systems which are unseen and have remained basically as they were 50 years ago. It can often take a trained ear, to hear if the sound is being taken from the old optical analogue, or the new digital formats.

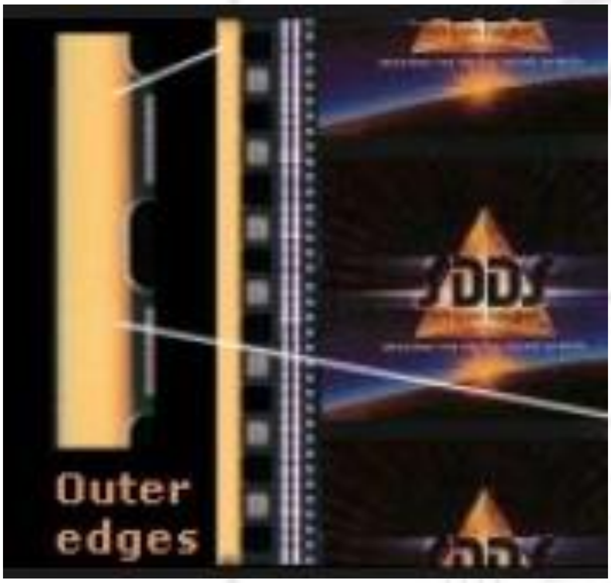
<u>Dolby SR-D</u>: is the most common digital format. The digital information is stored between the sprocket holes. The small space between the sprocket holes can only contain a small amount of data, this being a major limitation. A common belief exists that this area of the film stock was chosen because it suffers the least wear, and is the most reliable. However there are many reviewers and projectionists who claim that the space between the sprocket holes suffers the most wear and is the least reliable place to put digital data.



DTS Digital Theatre Systems: is said to be the preference of audiophiles. This view is also disputed. It uses a specially designed external CD player and requires 2 CD's per film. The CD player is sync locked to the SMPTE time code on the film stock. DTS is less preferred by cinema chains and film distributors because of extra cost, effort and possibility of CDs being lost.



SDDS Sony Dynamic Digital Sound: has the capacity for 8 channels. It is strongly argued that SDDS is the best performing format. The digital data is stored on the outer edges of the film stock. There are some reviews that claim the outer edges are vulnerable to wear and damage, and again there are other reviews that state the opposite. SDDS can provide for 5 screen channels plus independent surrounds and sub bass. It appears SDDS is technically supported but not promoted.



Dolby: 320 Kb/s; compression ratio 10:1; average 64 Kb/s per channel for 5 channels.

DTS: 1.04 Mb/s; compression ratio 4:1; average 240 Kb/s per channel for 5 channels.

SDDS: 2.46 Mb/s; compression ratio 5:1; average 307 Kb/s per channel for 8 channels, but because SDDS has back up tracks the average may be similar to DTS.

The Dolby method is said to take advantage of momentary space from any one channel to increase the capacity of the other channels. Instead of the audio quality being inversely proportional the number of channels, it is approximately inversely proportional to the √ of the number of channels (√5 = 2.24). DTS is said to include frequency domain sharing between subwoofer and surrounds, hence no bass under approx. 160 Hz in the surrounds. The SDDS system is said to have a fixed bit allocation per channel to maintain channel independence.

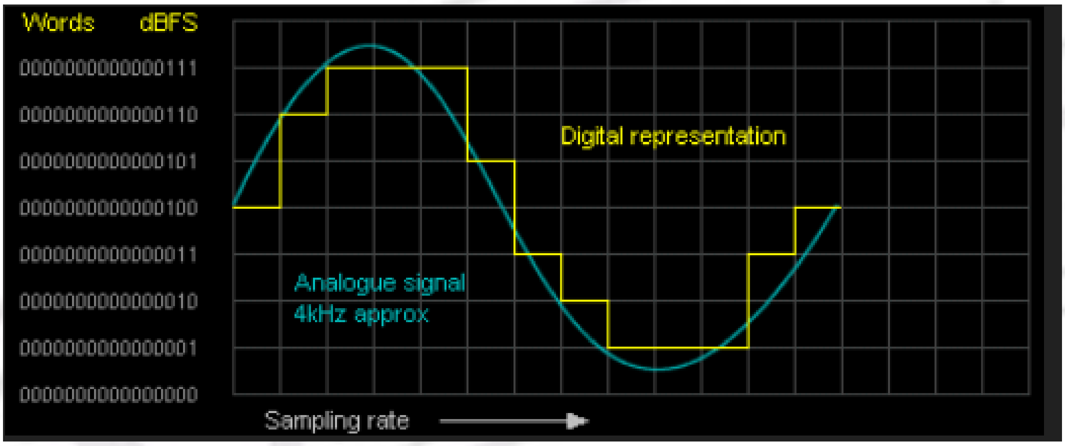
Achieving 5 to 8 channels of sound from a limited bit rate was not a simple task. Research and development exceeded $20 million. Regardless of their limitations, the ability to approximate the performance of the analogue magnetic format is an astounding achievement to say the least.

Understanding Digital Sound

An audio analogue signal on magnetic tape or vinyl records is infinite in detail, but suffers deterioration and increased noise from repeated use and mass transfer. Analogue sound quality is measured by frequency response. The lower the frequency response the lower the quality. Digital audio does not suffer from loss of frequency response, but with a low bit rate it suffers from quantizing noise (random static) and smearing.

Digital reduces the infinite detail of an audio analogue signal to a representation of finite bits of 1’s and 0’s. Each 1 and 0 is absolute and therefore is simple to produce and mass transfer, and does not suffer deterioration or noise with repeated use. The maximum allowable number of bits 1’s and 0’s per second (b/s) is the only limitation of digital technology to provide full sound quality.

The 1’s and 0’s bits are grouped into ‘words’ of 16 bits (1000110001101010) for domestic CD and 18 bit ‘words for Pro-audio (100011000110101001). A word can consist of any number of bits. The number of words per second is called ‘sampling rate’. Domestic CD sampling rate is 44.1K words per second. Pro audio sampling rate is 48K words per second.



Bits per word: defines dynamic range.

Sampling rate: defines frequency response and must be greater than x 2 highest audio frequency.

dBFS Full Scale: 0 dBFS is the highest level of word sample.

Therefore lower audio levels will be – dBFS numbers.

1111 1111 1111 1111 = 0 dBFS

0000 0000 0000 0001 = -96 dBFS.

16 bit word = 96 dB dynamic range.

20 bit word = 120 dB dynamic range.

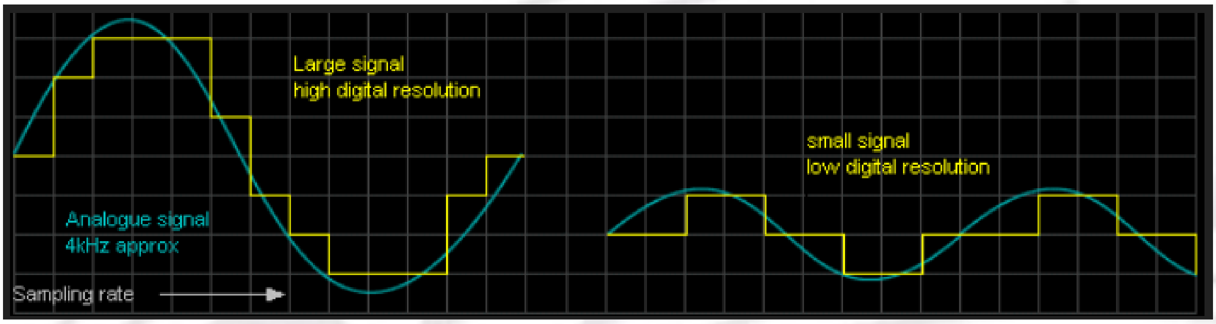
24 bit word = 144 dB dynamic range.

Note: Only a word group of exactly 8 bits is called a 'Byte' and refers to storage capacity. A CD can store 700 Mega Bytes (700MB). We must not confuse bits with Bytes. Byte is upper case 'B'. bit is lower case 'b'.

The greater the total numbers of bits/second (b/s), the faster the sampling rate, and/or the greater number of bits in a word. But for a given number of bits per second (b/s) there can be a choice between a slow sampling rate with a large number of bits per word or a fast sampling rate with a less number of bits per word. The continuing explanations will simply refer to total bits/second (b/s) to represent audio quality.

Domestic CD is 1,411,000 bits per second (1.411 Mb/s) for 2 channels. Therefore each channel is 705,600 bits per second (705.6 Kb/s). For the majority of people this bit rate is high enough to enable music fidelity to be in-distinguishable from quality analogue formats (20 Hz – 20 kHz with 96 dB dynamic range).

Before CD players were available the only domestic digital recording medium was video tape recorders. The fractional number of 44.1K sampling rate (words per second) was the maximum allowable for high fidelity audio to be digitally recorded onto video tape formats, and the 44.1K sampling rate was retained when domestic CD arrived. Pro audio Digital Audio Tape (DAT) is 48K sampling rate.



Resolution: The primary reason to have the highest sampling rate possible is to obtain the best small signal resolution for fine harmonic detail and nuances within the music. The second reason for high sampling rates is to enable the signal to be digitally EQ modified and processed. It is therefore understandable that digital recording naturally favors increased level over fidelity.

<u>Digital Compenssion</u>:

Loss-less un-compressed: format is for Pro-audio digital and domestic CD. Silence is recorded and played back at the full file size bit rate.

Loss-less compressed: format is similar to a ZIP file that reduces file size by not recording the silence. When played back it replaces the original silence at full file size.

Lossy-compressed: Lossy compression is Smoke and Mirrors which evolved from psycho-acoustic research. Silence including selected detail within the music can be discarded without the average person being able to notice, obtaining 30% to 90% reduction in file size. Discarded information cannot be retrieved.

MPEG (MP3): Moving Picture Experts Group use a variations of techniques described as "perceptual noise shaping" or "perceptual sub-band transform coding". It is used for internet music downloads where only very small file sizes can be used. MP3 allows for various compression rates to be chosen. But how much information can be throw away without noticeably deteriorating of the quality of sound?

256 Kb/s; 5:1 compression; Music quality almost indistinguishable from the original CD.

192 Kb/s; 7:1 compression; Popular choice for reasonable quality.

128 Kb/s; 11:1 compression: Popular for internet download music and iPods.

96 Kb/s; 14:1 compression; Easily discernibly lower sound quality than original CD recording.

64 Kb/s; 22:1 compression; Mono speech only don't attempt music.

The best test to hear a comparison of MP3 lossy compression to original un-compressed CD sound is to use white or pink noise, audience applause, rain on a tin roof, a bundle of keys thrown up in the air and caught, and worst of all a Harpsichord.

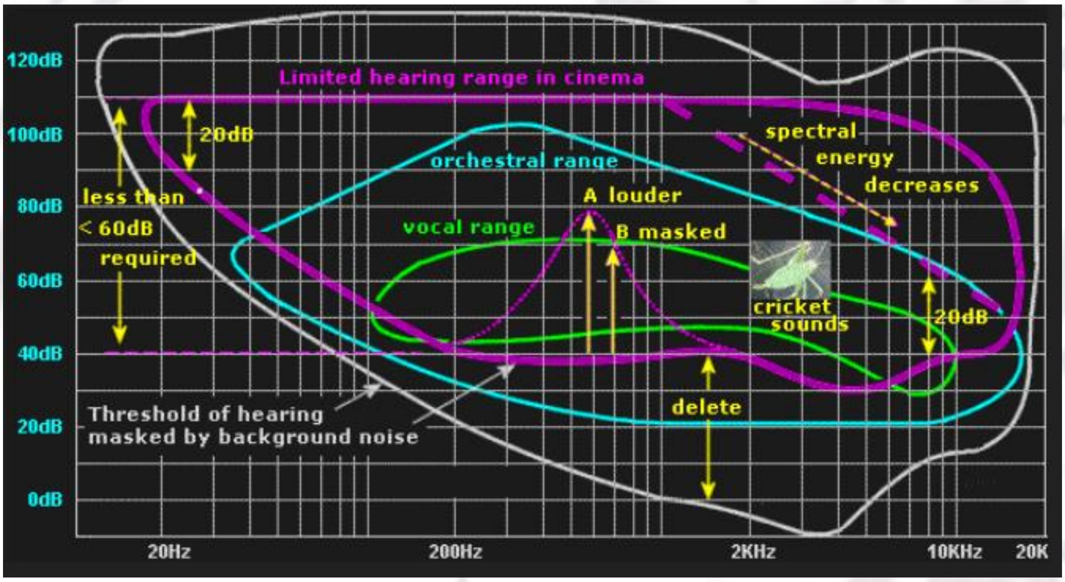
Achieving an acceptable performance from a very limited bit rate is a technological miracle greater than the biblical parable of the fishes and loaves. However when applied to multi-channel cinema sound we must not lose sight of marketing deception when promoting a brand image for 'white bread' as vitamin enriched. As in a product stripped of nutrients (or necessary bit rate) and selling it as being enriched with cleaver deletion algorithms or artificial vitamins enabling it to taste or sound acceptable. Hopefully when MP99 arrives it will discard clap-trap, boring cliché dialogue and TV commercials as well.

Compressed Lossy – Digital Cinema Sound

It is argued that much of what is recorded on an un-compressed loss-less CD format cannot be heard. What cannot be heard, cannot be heard, therefore silence and any sounds below the threshold of hearing, or below the general ambient noise level of 40 dBA can be deleted. Loud sounds mask softer sounds and the softer sounds can be deleted when louder sounds are being played. Some frequencies that are close together can mask each other; therefore the masked sounds can be deleted.

The perceived sound quality is dependent on limitations of listener attention, being in a high reverberant environment with ambient noise, listening un-attentively to a small cheap limited fidelity home cinema system while being distracted by vision. The final essential factor is that the listener's expectation has been influenced by marketing.

When all these external factors are combined they effectively mask the distortion anomalies created by the lossy compressed digital sound.



Ambient noise of cinema is approx. 40 dBA, any sound below this level can be deleted.

Dynamic range between background noise and maximum level is approx. 40 dB.

Hearing sensitivity of low and high frequencies is limited, only 20 dB dynamic range required.

Sound outside of our ability to hear direction (chirping cricket) can be collapsed to mono.

High level sounds psycho-acoustically mask similar sounds of lower level, which can be deleted.

Bit pool: To achieve these deletions, plus many more, requires a sample (every fraction of a second) of the total information to be stored in a bit pool for analysis. Instantaneous decisions are made of what information can be deleted. But when all channels are over-used (at the same time), beyond the capacity of the bit pool, some essential information may have to be dumped.

Depending on the % of deletion, unpredictable outcomes may occur. A frequency band in one channel may be deflected to another. Frequency bands that are similar in different channels may be deleted leaving only the loudest heard. Similar bands from different channels may appear in the center channel. Ringing or pre-echo of percussive or transient sounds may occur.

Psycho-acoustic masking: These random artifacts occur within a fraction of a second and are averaged by our hearing and expectation. The majority of non-discerning audience in a cinema does not notice if all channels are collapsed to mono, or if the surrounds are on. Sight can influence what we believe is the direction of sound. Also none of these lossy compression techniques reduce frequency response. For most people hearing high frequencies of any type is thought of as high fidelity.

When audio is digitally compressed the outcome is similar, and some hi-frequencies become exaggerated giving a false perception of fidelity. However an attentive listener can easily hear poor music resolution, smearing, image loss, reduced depth of field and chaotic imbalance between channels.

If the anomalies of lossy compression are not to be heard, it requires a high correlation of similar sounds between the channels. A simple test to hear limitations of lossy compressed 5.1 formats is to put different full fidelity music on each sound track.

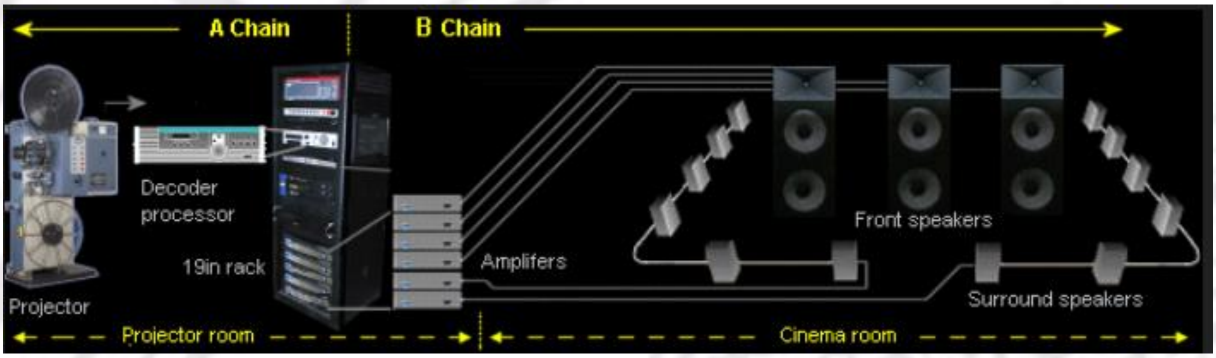
Left channel	Mahler’s 5th
Center channel	Beethoven’s 4th
Right channel	Tchaikovsky’s Nutcracker suite
Left-rear channel	Loud Rock music
Right-rear channel	Rap or Techno music
.1 Sub-bass channel	African drums

This test is unrealistic as this level of sound separation is not required for multi-channel sound in film production. But this test will reveal the limitations and demonstrate what can be achieved. The simplest test that can be achieved is people simultaneously speaking different languages, recorded and then played back, with consistent separation results from each channel. However if the people are rotated as if on a merry-go round, strange things can start to happen.

Demonstration trailers promoting digital sound often consist of loud impressive animated computer sounds with minimal transients and harmonics, heard through mostly limited fidelity speaker systems, where the limitations and anomalies of lossy compression are not heard. Also the majority of companies behind digital technology are secretive, aggressively protect their interests and do not openly disclose problems and limitations.

Cinema A chain and B chain

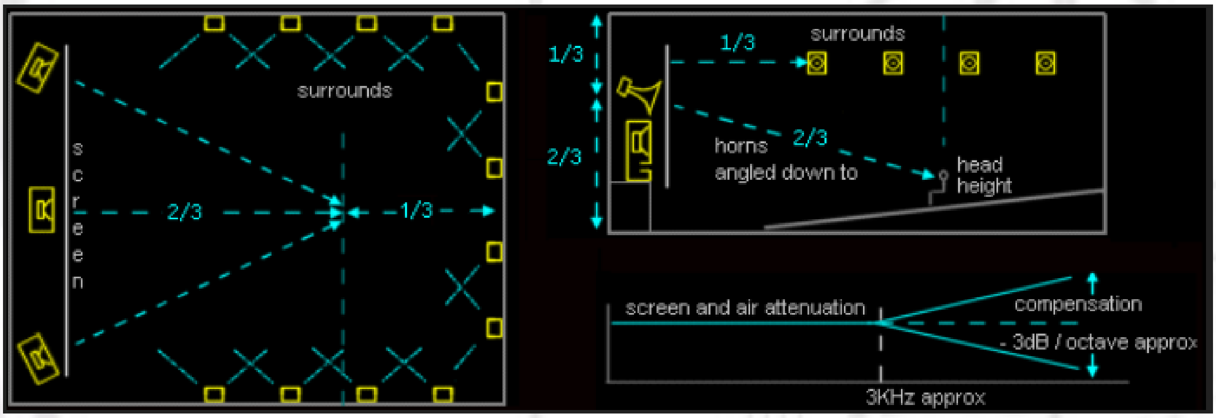
Cinema sound management is divided into 2 sections. A chain refers to the sound track on the film stock, projector sound reader and Dolby decoder including other competitive decoders. B chain refers to power amplifiers, crossovers and speakers, including cinema room EQ. Traditionally the decoders and amplifiers are mounted in a 19in rack in the projector room. The long speaker cables from the projector room to the screen and surround speakers are mostly hi-current low-resistance 110 V or 240 V power cables which are installed by electrical contractors when the cinema is made. Commercial cinemas are not attached to any form of audiophile nonsense and therefore do not use 'magical' speaker cables.



It can be questioned why the amplifiers are not placed with the speaker system with short speaker cables, and only requiring a long signal cable from the projector room. This approach is normal with live entertainment sound systems; the answer is simply historical convention.

<u>B chain</u>

All cinemas follow basic rules of speaker placement to provide a consistent outcome which when looked at in detail is the most pragmatic approach. The speakers behind the screen have to be angled accurately at the points shown in the below pic (to begin with). Then re-adjusted by ear to achieve a stereo image that maintains maximum consistency throughout the room. This may require a larger toe in from the left right speakers to minimize reflections from the adjacent side walls. Also an increased toe sometimes provides a better stereo image for sound buffs who prefer to sit at approx. 1/3 distance from the screen. Those who wish to sit at the rear of a cinema; stereo image will have no value.



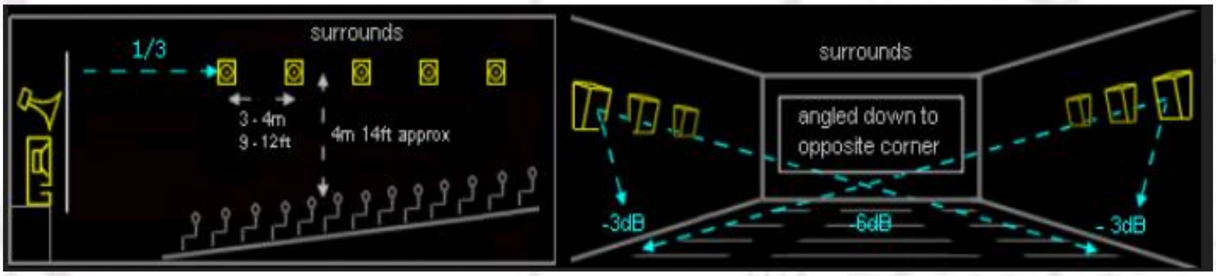
Different cinema screens will attenuate high frequencies to varying degrees, and it is essential to place the high frequency horn as close as possible to the screen. Air also absorbs high frequencies over long distances which has to be adjusted for large cinemas. A variable 3 dB per octave lift above 3 kHz can easily compensate for all high frequency losses.

The front speaker cabinets should also be placed in a large baffle behind the screen as shown with the JBL speaker system in the above pic. The baffle provides forward propagation for the lower frequencies. The baffle also stops sound from reverberating against the wall behind the screen. The difference between having a screen baffle is very noticeable in improving the sound system performance. However very few cinemas are prepared to incur this extra expense. Information on the left-center-right screen speakers are covered in more detail in the chapter of 'Large Systems'.

Surround Speakers

Live musical concerts including opera do not have surround sound. Sounds from other directions cause audience distraction. Reverberation and echoes reduce intelligibility; therefore favored seats are closer to the stage. However the movie experience is enhanced by surround sound to create spatial environmental effects; providing it does not distract or conflict with the front system.

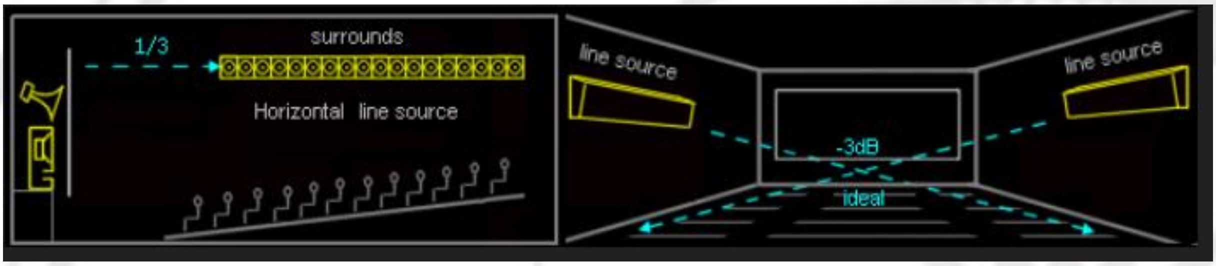
Home cinema marketing states that the surrounds should be the same speakers as the front. But this is never the case in commercial cinemas. A stated requirement for commercial cinemas is that the total sound energy from all the surrounds must be capable of matching one of the front screen speaker systems. However sound energy from the surrounds is almost never required to equal the loudness capacity of a front system. This is possibly stated to insure that the surrounds are always capable of delivering what they should.



Surround speakers normally begin at approx. 1/3 of cinema distance from the screen. The basic mounting positions above are a guide and the final positions should be adjusted by ear. The height and downward angle of the surround speakers is aimed at achieving the minimum loudness difference from wall to center seats (approx. -3 dB). Also the total surround level is calibrated with pink noise to be approx. -3 dB below the level of the main screen system.

Surround delay time: A mandatory 20milli second delay is applied to maintain hearing attention to the front speakers. Delay time is then extended to the distance time of the cinema length in 1 millisecond steps (1 millisec = 340 mm, 1.1 ft). A cinema length of 34 m (110 ft) will require an extra 100 milli (1/10) second delay. The Dolby processor automates this procedure. Be sure to follow the processor instructions carefully and double check the cinema measurements. The final adjustment is calibrated by ear and any error must be biased toward reducing surround level, not toward increasing surround level.

The surround speakers should not be heard directly as point sources, as this causes distraction from the screen system. Surrounds are meant to disperse the sound in a diffused manner, similar to how we hear sound in the far field natural environment. But surround speakers are constrained by walls, that cause the sound to be heard from the nearest speaker only. This problem puts impossible constraints on the recording engineer.

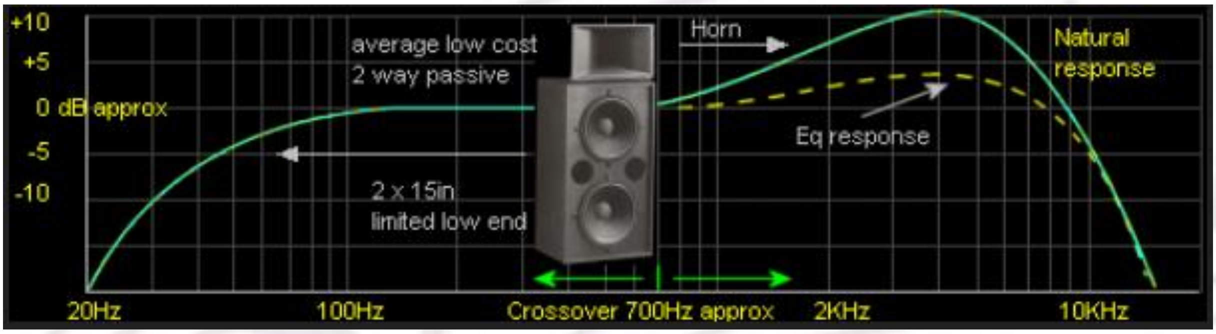


Bass and Sub-Bass

The ability to provide deep bass had been understood since the 1930s. Some of the large wide screen cinema sound systems in the 1950s could easily manage 40Hz. Also during the 1950s some DIY (do to yourself) hi-fi enthusiasts made large brick speaker enclosures with 15 inches speakers in the corners of their living rooms easily achieving 20 Hz. The only limitation was the needle jumping off vinyl records. Many Rock concerts in the 1960s used large folded horn bass bins.

The first mass audience experience in cinema of powerful separate sub-bass was created for the "Earthquake" movie in 1974 using very large Cerwin Vegar bass bins in a system described as 'Sensurround'.

Sound quality especially sub-bass was never seen as being important for 35mm main-stream cinema. The majority (but not all) cinema sound systems were and still are made as cheap as possible; 2 x 15 inches with a horn, as in the below pic. Bass frequencies of 40 Hz – 60 Hz are approx. -6 dB to -12 dB below voice range. Because cinemas are quiet environments the bass can still be heard at this lower level. This is the reason the problem had not been previously addressed.



By the late 80s Dolby SR became available providing sub-bass extension which was promoted vigorously. When digital sound was arrived the sub-bass was available as a separate LFE Low frequency Effects channel. Dolby's success was in understanding that the majority of cinema management would spend as little as possible to improve sound. The separate sub-bass extension is the simplest low cost solution, requiring only 1 extra amplifier with a single 18in bass speaker (approx. $2,000) to be added to an existing cinema system.

Our ears are in-sensitive to bass frequencies, and combined with the 18 inches speaker being less efficient than the speakers in the main system which requires the 18in to be equalized and driven with greater power, to be heard at the same level as the voice frequencies. 1,000 Watt amplifiers are commonly used. This is the primary reason sub-bass is not included in the main front channels and has to be separate.

An average 18 inches bass speaker cone has a fundamental resonance (Fs) of approx. 35 Hz in free air. When the speaker is put in a 10 cubic ft (260 Liter) box the speaker resonance is made higher; approx. 50Hz – 60Hz. To achieve deep bass a resonant port is put in the box and tuned to approx. 30 Hz, not 20 Hz as often believed.

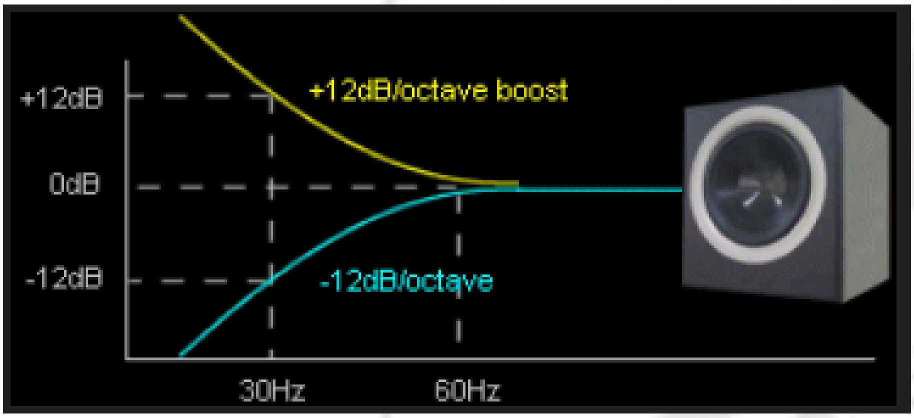
The port (Helmholtz resonator) will generate a resonant note only at the frequency it is tuned to in reverse phase. The port resonance is also activated by any movement of the cone at other frequencies at a lower level. Many musicians, who play electric bass guitar, including a % of discerning audiophiles, do not use bass boxes with ports. However ported boxes have a fanatical following singing their virtues.

Port problems: For a port to be maximally effective it should be similar in diameter to the speaker and very long. But a large port similar to the size of the speaker is not practical as it would have to extend outside the box. So for the port to fit inside the box it is reduced to a smaller size. These smaller ports are less efficient and generate greater air velocity, which can create whistling sounds. Also any frequencies from the amplifier that are below the port resonance, cause decoupling of air loading to the cone, generating excessive cone movement (exertion) that can easily destroy the speaker. Cinema projectionists often describe how common this problem is.

To deal with this problem many speaker manufacturers make the cone suspension tight, which raises the Fs, compromising bass performance for reliability and higher power rating. Also a filter is sometimes used to stop bass frequencies below port resonance getting to the speaker. All of this amounts to being a bag of worms that has to be managed. But ported boxes are the only way to achieve effective 'cheap' deep-bass. This is not right or wrong but simply physics.

Sealed box: In a sealed box, a speaker cone must move 4 times the distance for each octave decrease to maintain the same acoustic output, providing the bass wavelength does not exceed x 10 speaker diameter. This is the reason a 4 inches speaker cannot deliver sub-bass at a reasonable level for home cinema regardless of how far the cone can be made to move. This can be simply demonstrated by waving ones' finger in the air, compared to a large sheet of paper.

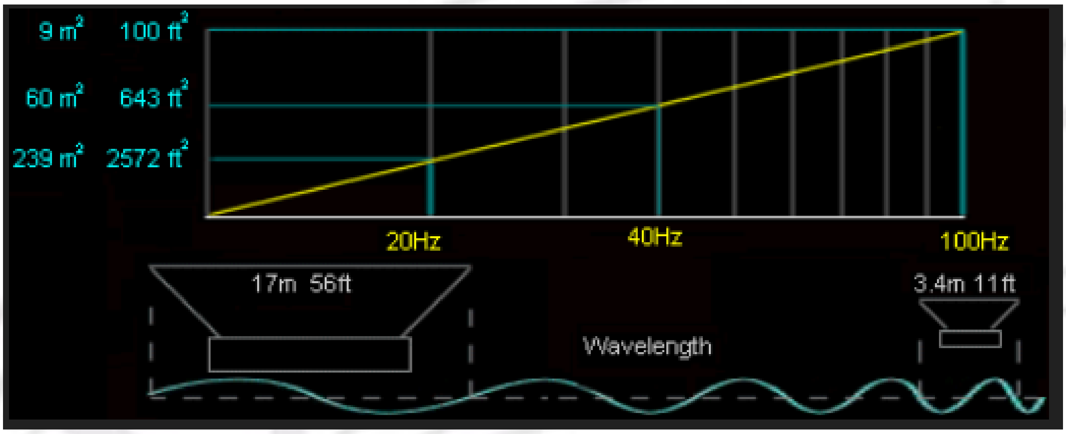
Many active sub-woofers for home cinema and vehicles use 10 inches or 12 inches speakers in small sealed boxes of approx. 1 – 2 cubic ft (30 – 60 liters). The speaker resonance can be as high as 60Hz – 100Hz. Below system resonance the cone cannot increase excursion at 4 times the distance for each octave decrease, which is the minimum required to maintain constant acoustic output. Below system resonance the cone excursion is kept constant, by air compression in the small box. Bass efficiency rolls off at -12 dB/oct below resonance. (-12 dB = 1/16)



Sub-woofer EQ: Many sub-bass amplifiers have special equalization (EQ) to boost amplifier power, compensating for the decrease in efficiency below resonance. If the speaker-box system resonance is 60 Hz, and the system is required to be flat down to 30 Hz, then the amplifier will have to deliver (+12 dB) 16 times more power at 30 Hz, to force the cone to move 4 times the distance. These speakers have heavy cones and are inefficient. Amplifier power for domestic application can be approx. 300 Watts.

To understand the management of bass and sub-bass frequencies requires relating to the frequencies as their wavelengths. The velocity of sound is approx 344 m (1125 ft) per second.

100 Hz wavelength = 3.4 m (11.25 ft)
42 Hz wavelength = 8 m (26 ft) lowest note on double bass
20 Hz wavelength = 17.2 m (56.4 ft) lowest sub-bass effect



The imagined ideal is for the speaker diameter to be equal to the longest bass wavelength. But this would require the sub-woofer to be the approximate size of the cinema screen.

Perfect cinema sub-bass: For the sub-bass to maintain the same directivity and efficiency as the 2 x 15in speakers in front main speaker boxes, the total area of the sub-bass speakers should be approximate x4 the area of all the 15 inches speakers in the left-center-right boxes. This requires 6 x 30 inches speakers. Because 30 inches speakers are not normally available, 6 x 18 inches speakers are practical.

In sealed boxes the 18in speakers will have to have a low Fs fundamental resonance (very loose suspension) less than 25 Hz. The boxes will need to be very large approx. 20 cubic feet (600 liters). The speakers will still require extension EQ to force them to reach down to 20 Hz at equal power. The bass boxes will deliver more propagational energy if they are stacked together in one place. However to minimize excessive cancellation from standing waves throughout the cinema, it is best to separate the bass boxes as in the above pic.

High quality 18 inches speakers are expensive, and most are made with high Fs (stiff surrounds) to keep the voice coil centered, which unfortunately defeats its ability to produce deep bass in a sealed box. Most 18in speakers are designed to be used in ported boxes only.

A cost effective solution that performs as well (if not better) is 24 x 15 inches hi-fi speakers (2 inches voice coils approx. $100ea). These speakers are readily available with soft surrounds that have a low Fs approx. 25 Hz. The cost is low because they are made in large quantities. The 15in speakers have to be in compound pairs, each pair acts as a single speaker. This arrangement does not look attractive; however it gives excellent linearity, low distortion, at very low frequencies. Each sealed box will still have to be approx. 20 cubic feet (600 liters) and may require a small amount of extended bass EQ to achieve 20 Hz at equal power.

Sub-woofer Phase: It is not stated that sub-woofer phase is required to be matched to the front left-center-right speakers. This is because it is regarded as a separate channel with different information. But this not a correct assumption because at times the same bass information can be applied to the sub-bass channel and front speakers. This should be checked independently with a signal generator to ensure both sub-woofer and front speakers are in phase at the crossover frequency. Do not attempt phase correction with pink noise measurement using microphones in the far field as this result can be randomly influenced by cancellations from standing waves.

Time alignment: As an imaginary concept if the whole frequency response (from each speaker) radiated from a single point then all the frequencies would be in time with each other. With large cinema sound systems the low and hi-frequency drivers are separated by small physical distances. With the old large Altec A4 system below pic, the 15 inches speakers and horn driver were mechanically in line on axis, so there is zero time difference to the audience. But the vertical height difference between the drivers is approx. 2 m (6 ft) and at 344 m/second is a 6 millisecond time difference which can be considered audible for someone sitting underneath the speaker system in the front row. This is similar to being at the front or on stage when listening to a live band.

Time differences of less than 10 milliseconds of sharp transient clicks cannot be heard as separate, but as a single fattened partially muffled click. A common practice in pop recording is to add multiple short delays of less than 15 milliseconds to voices and instruments to give them a fatter sound. However time differences of transient clicks beginning from 11 milliseconds gradually start to be heard and become clearly separate when approaching 30 milliseconds 10 m (30 ft)

The problem of separate double clicks was first noticed with very large speaker systems in the 1930s when Fred Astir and Ginger Rogers tap danced. At first this was confusing because the distances between the low and hi-frequency drivers were not great enough for the distinct double clicks to be as extreme as they were.

The early multi-cell horns were designed to have a wide dispersion to provide sound to the upper and side balconies. But these wider dispersion horns caused slap echoes from the ceiling and side walls to be greatly increased. Modern cinema horns have a narrow controlled directivity compared to the earlier multi-cell wide directivity horns.

Time alignment correction including phase correction should be considered as good housekeeping, regardless if it is audible or not.

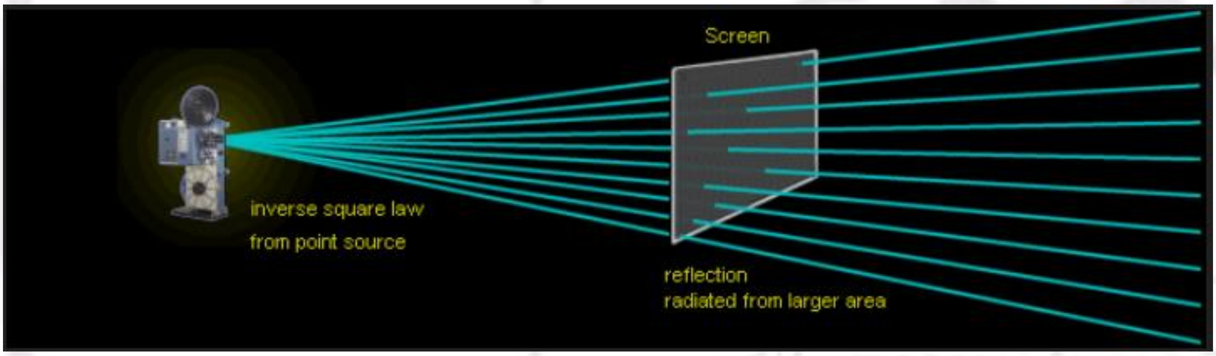
Screen Image vs Sound Image

In most cinemas the physical size of the sound system is small compared to the screen and therefore the sound experience does not match the experience of the picture. Small speaker systems disperse as a point source at the inverse square law from the screen, whereas the picture is radiating from the whole screen area. The general view of the majority of people is that sound is only 10% of the cinema experience.

Cinema is successful because the picture appears to have no boundary when viewed on the big wide screen. So why not remove the boundary of a small low fidelity sound system, and make the sound as big as the picture in terms of experience.

Imagine being in a rain forest and very attentive to the sounds around you, hearing subtlety of detail which extends into infinity. Record this, go home and play it back on a small stereo. It will only be recognizable because of remembering. Turning up the volume does not make it sound real. It is made real by removing the boundary of a small low fidelity sound system and stepping into large scale 4way active electro-acoustic technology.

Inverse square law: refers to energy radiating from a point source. When the reflected light (image) radiates from the screen, the inverse square law does not begin at the screen, but from the projector, at the rear of the cinema, or the same distance from an imaginary point behind the screen.



The projected image from the screen always looks big wherever we sit in the cinema. This is the reason home cinema can never create the same experience as going to a large theatre. This is similar to why the energy from the sun or the sun’s size does not appear to decrease by moving another 30 m (100 ft) further away from it at the distance of the earth.

We can all understand that using a small home cinema screen in a large cinema, and simply increasing the luminance (brightness) does not make the image appear larger. But this is precisely what most people (including those that are technical) falsely believe about sound systems. Putting a small home cinema sound system behind the screen and simply turning up the volume does not make is sound like a large cinema sound system. It will only sound like a small sound system with the volume turn up. This point is made so not to confuse loudness with sound stage or sound image, which refer to propagation directivity and radiating area.

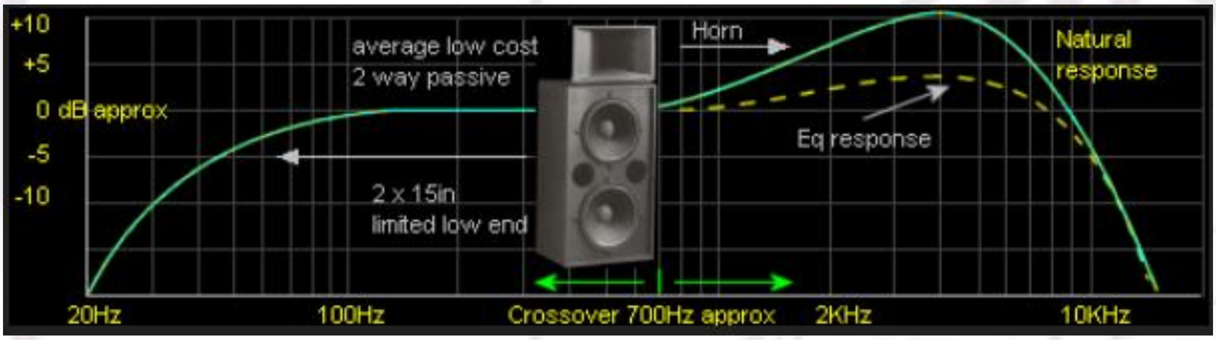
Wavelength propagation: The difference between the wavelengths of various colors of light are less than 1 octave, approx. 1/2 micro meter. We can assume that all light from the screen radiates at a similar wavelength which is very small compared to the radiating area (screen size). Therefore the image will appear consistent in reference to screen size and observed distance.

The average home cinema speakers are approx. 4 inches-8 inches. The lower frequency wavelengths are approximately x10 to x100 larger than the speaker diameter. The higher frequency wavelengths are approx. 1/10 to 1/2 speaker diameter. Therefore the sound image (dispersion) will be inconsistent across the frequency spectrum; also the spectral balance of low to high frequencies will change with power.

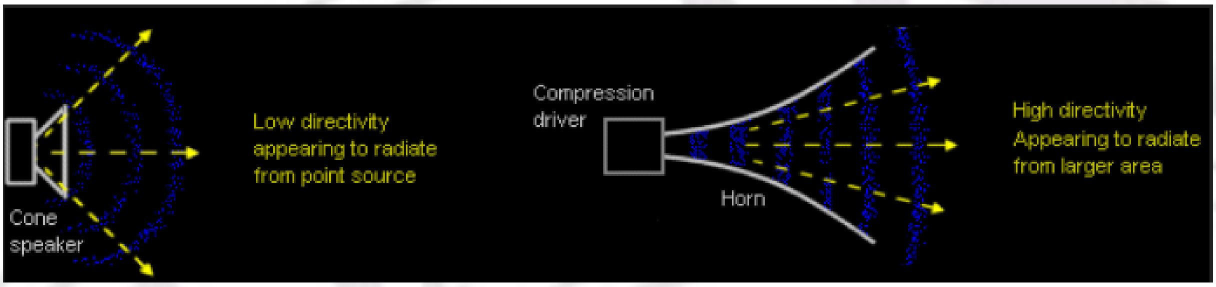
The average commercial cinema speaker system has 2x15 inches speakers which are small compared to bass wavelengths. Whereas the high frequency horn is matched to the higher frequencies and has a different behavior to the front loaded 15in speakers. The lower frequency energy from the front loaded cone speakers lags behind as the power increases.

However this is not so with sound. Sound is kinetic energy (air vibration) at very low velocity of 344 meters/second. Wavelength at 100 Hz = 3.4 m (11 ft); Wavelength at 10 kHz = 34 mm (1.4 inch); a large ratio difference of 100:1.

For the sound image (dispersion and propagation) to be consistent over the frequency and dynamic range, the acoustic radiating diameter should approximately a wavelength, increasing x2 for each octave decrease. This imaginary ideal is impossible because the speaker would have to be as large as the screen at 20 Hz wavelength = 17 m (56 ft).



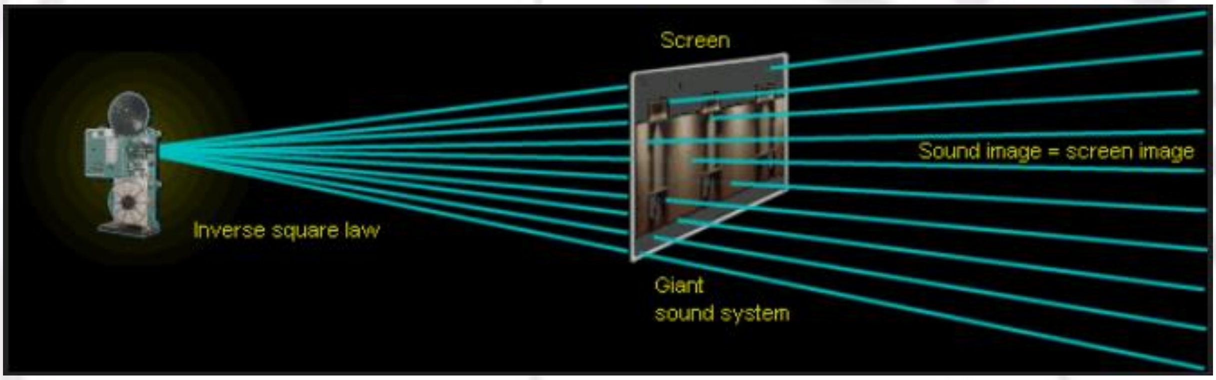
Horns: Theoretically the sound waves at the horn mouth are not as curved as if coming from a front loaded cone speaker. The sound waves from the horn appear slightly straighter as if radiating from a larger surface area, similar to the screen but on a smaller scale. This is why the hi-frequency horn appears to have a more forward projected sound image and greater voice clarity compared to front loaded cone speakers.



As most cinema sound systems are 2 way passive with 15in front loaded speakers, it is best to get as much of the sound spectrum as possible to come from the hi-frequency horn and take advantage of its increased directivity. Manufacturers of most horn 2in compression drivers recommend a crossover frequency of no lower than 800 Hz at specified power.

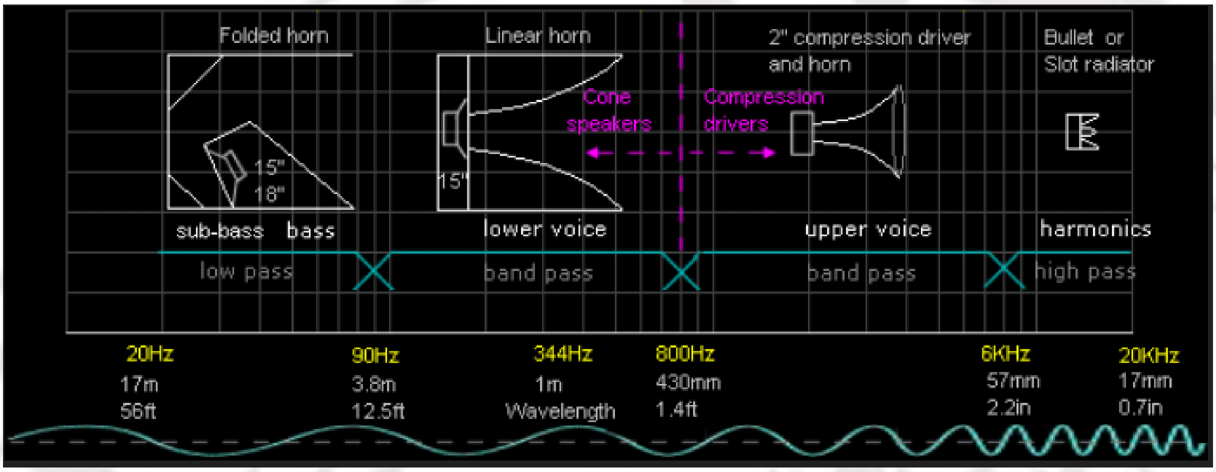
50 0Hz crossover: Providing the designers of the 2 way speaker systems are willing to limit the power to the horn compression driver to less than 1/2 of its specified power, it is possible to lower the crossover to 500 Hz before the driver diaphragm is at risk of being damaged. This also requires the horn length and mouth be made larger than normal, sometimes described as long-throw or constant directivity horns.

Many large sound systems of the past were 2 way passive and had very large extended horn shaped baffles for the low-frequency 15 inches speakers. This helped obtain a similar directivity and efficiency to the hi-frequency horn on top. Early valve amplifiers were approx. 30 to 60 Watts which provided a strong motivation for design engineers to make the speaker system as efficient as possible. The overall sound quality appeared consistent over the frequency range, compared to the majority of today's smaller cinema sound systems.



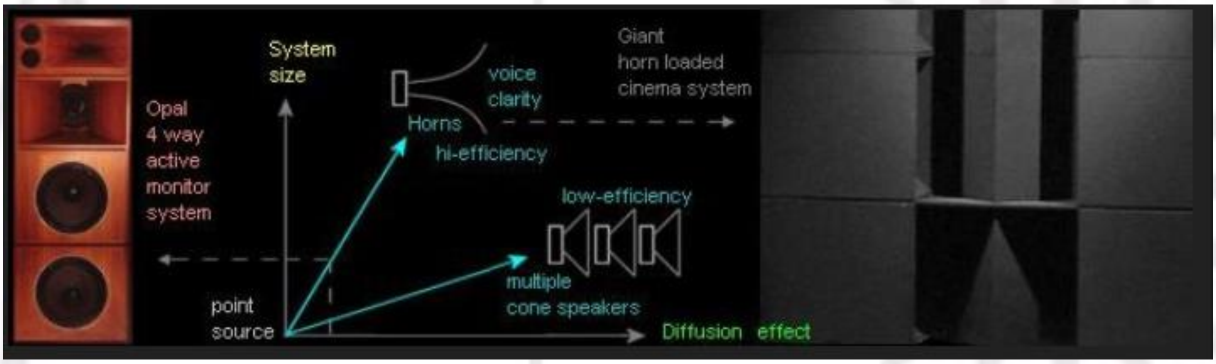
The aim is for the sound to match and picture so they both appear in the same proportion everywhere in the cinema. This requires the sound spectrum to be divided into 4 sections. The below pic shows the large variation of wave lengths over the sound spectrum that have to be managed by the speakers is each section.

Above 700 Hz compression drivers + horns are ideally suited to the smaller wavelengths of higher frequencies. Below 700 Hz in larger 300+ seats cinemas the 15 inches speakers should be placed in horn shaped cabinets to increase the effective radiating area and directivity to match the high frequency horns. Below 100 Hz the wavelengths are so long that the floor and walls are able to act as an extension of the horn cabinets and therefore maintaining their directivity.



Active: All large sound systems require separate amplification for each of the frequency bands to achieve maximum performance with minimal error and distortion. Active management also eliminates any possibility of inter-modulation between the different frequency sections. The Lenard K4 system uses a separate amplifier for each speaker component. Electronic crossovers should be 4th order (24 dB/oct) and have time alignment correction available.

Jokes are often made by technical people that all sound system problems could be solved if everyone in the cinema wore headphones. Sound fidelity is highest at close proximity to a sound system that radiates from a point source. Monitor systems in recording studios for mixing cinema sound use smaller sound system similar to the Opal speakers in the pic below on left.



Diffusion: Radiating sound from large areas with multiple speaker components increases the possibility of similar frequencies to interact causing diffusion. The extreme of this problem is heard with large concert PA systems. The radiating area of an average cinema sound system is approx 0.75 m² (8 ft²). The Lenard K4 radiating area is approx 5 m² (50 ft²). Each of the large cinema sound systems in the below pics have different advantages. All approaches are excellent in performance and are designed to minimize the problems of diffusion when dispersing sound from large areas.

Conventional cinema systems require the (.1) information is sent to an independent amplifier and sub-woofer speaker. The .1 sub-bass is delivered through all the folded horn bass bins generating close to a plain wave front, providing vastly greater impact.

Projected acoustic center: The aim of all large scale systems is to create the experience that the acoustic center is projected forward (approx. 6 dB) into the audience, giving a feeling that the sound appears to come from half the distance that it actually is.

Cinema Acoustics

Before sound systems existed, the large grand Cinemas of the past were modeled on Opera houses with an orchestra pit in front of the screen which was later replaced by an organ that came up through the floor. Opera houses evolved to make use of reverberation to increase sound level to the audience. Many composers including Mozart hated the excessive reverberation of large concert halls which restricted their music. Mozart often preferred to perform outside where the detail of the music could be heard as he intended.

The cost of increasing sound level by reverberation is at the loss of intelligibility. What evolved was an imagined ideal of correct reverberation, that is, reflective area around the opera stage of short distances (short path-lengths) opening up to larger areas of longer distances (long path-lengths).

It can be argued that specific pieces of classical music suit different characteristics of reverberation. But there is no such thing as one type of reverberation that suits all classical music and all acoustic instruments. When sound systems evolved, cinemas became trapped with the problems associated with the excessive reverberation of large auditoriums.

The subject of architecture is obsessed with status and visual form, often with little or no interest in what cannot be seen (acoustics). Many architects believe that city environment including our homes and especially auditoriums and cinemas should be as reverberant as possible. Sound absorbed is negatively described as ‘room loss’. Inadequate and false understanding of acoustics has directly contributed to the excessive noise pollution of our cities.

Many early large cinemas had little or no acoustical absorption. Decorative ceilings may provide sound diffusion but little absorption. No cinema goer is interested at looking at an ornate foyer or the inside of the cinema when watching a film. The money wasted would have been better spent on properly acoustically treating the cinema.

Modern multiplexes often have red pleated curtains on walls which are mostly decorative and will absorb high frequencies, but may have little effect on absorbing low-frequencies. Often there is no acoustical absorption on ceilings except for standard acoustical tiles used in most office buildings.

George Augspurger a previous technical director of JBL and also an excellent educator, stated that the 3 Rs of acoustics are -

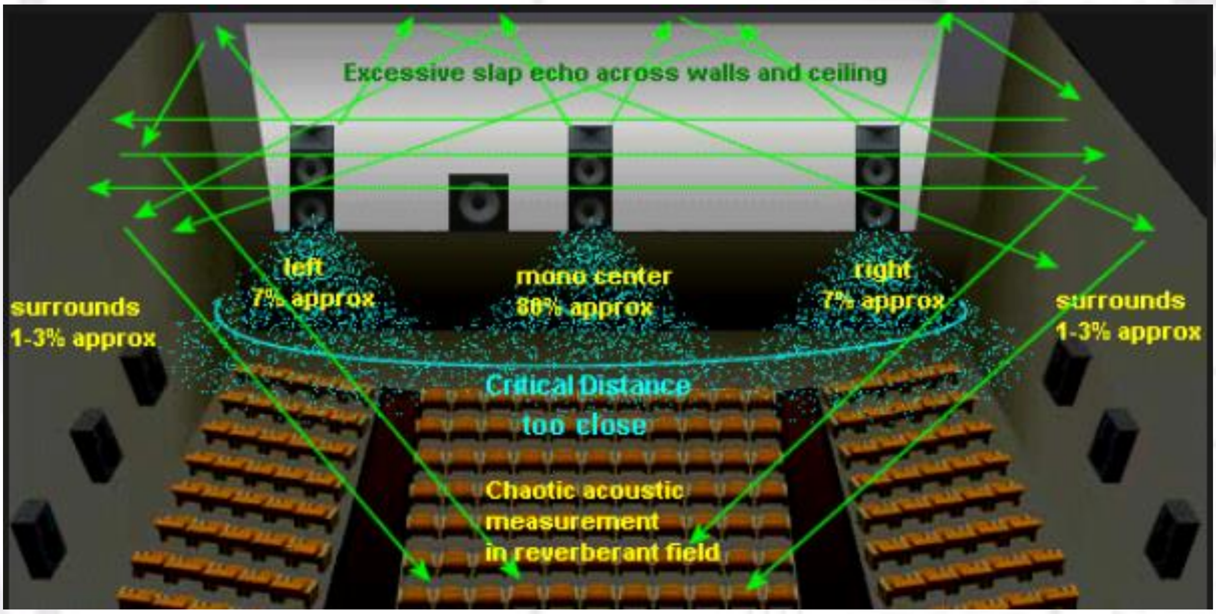
1. **R**oom resonance
2. Early **R**eflections
3. **R**everberation

Slaps echoes and reverberation

When surround sound evolved, the Academy directive stated that the story must not be dependent on the added sound tracks. An audience attending a mono only cinema must be able to understand the story. Because excessive slap echoes and reverberation destroy intelligibility and stereo imaging, the added sound tracks have limited effect. Many films are produced with dialogue from the center speaker only, left-right and surrounds minimally used similar to the percentages shown in the below pic.

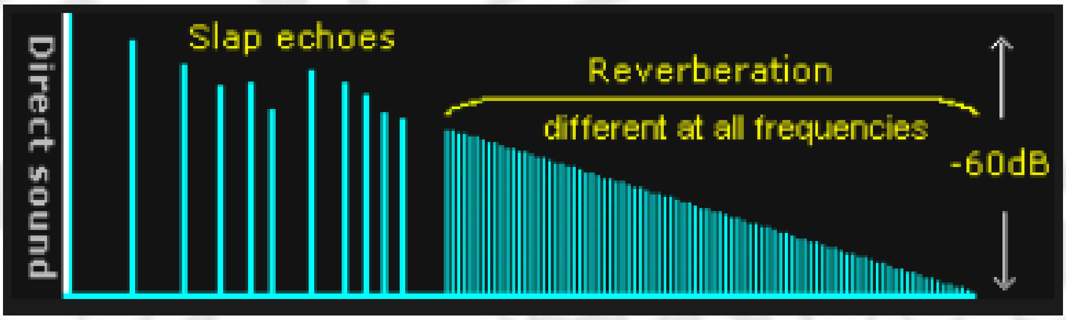
A simple hand clap at the front of the cinema will reveal the slap echoes across the walls floor and ceiling which evolve into reverberation, and will be clearly heard. As a general rule the first early reflections within 30 milliseconds are heard as being part of the direct sound. Whereas after 30 milliseconds the later slap echoes and reverberation are heard as being from the room.

But most often the sound from the speakers will be continuous and constantly generating slap echoes and reverberation throughout the cinema room, which becomes included within the following information from the speakers, continuing the cycle.

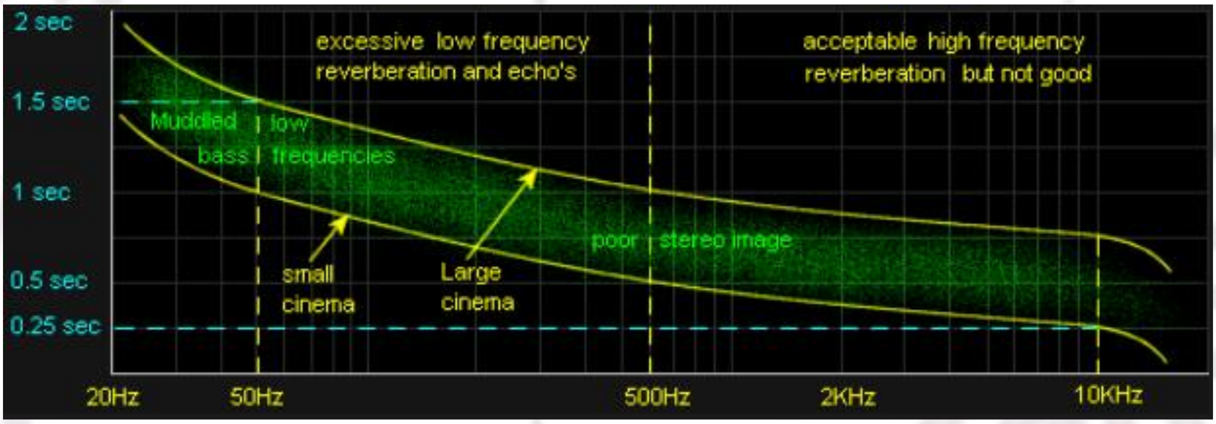


Regardless of those who argue that reverberation provides an agreeable aesthetic experience, it will in varying degrees mask and contaminate the new incoming sound and directly interfere with speech intelligibility, destructively changing what we are hearing from what the director intended, potentially undermining the story line of the film.

The wall curtains, seats and audience absorb sound, differently at all frequencies. The time for reverberation (at any one frequency) to diminish to -60d B (1/1,000,000) of its original energy, is called Reverberation Time (RT or T60).

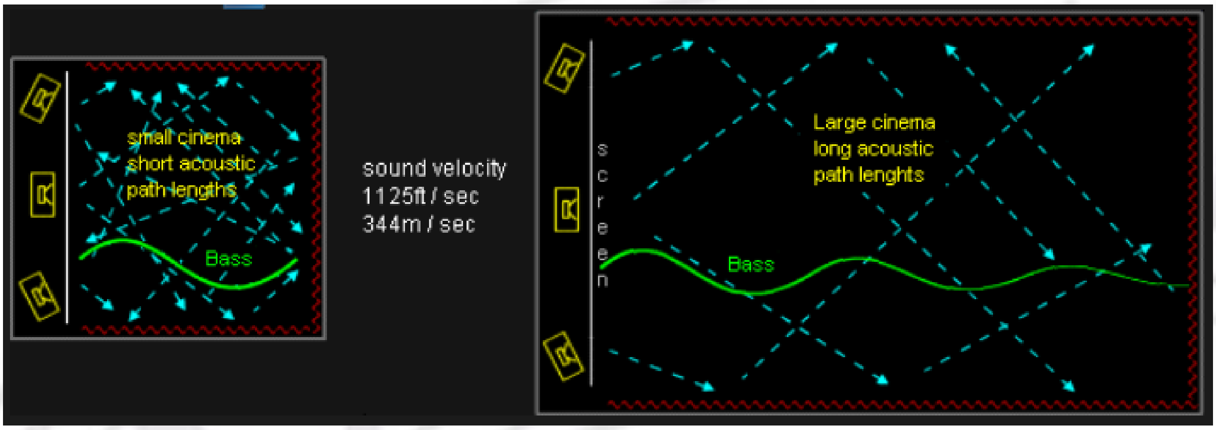


The majority of modern cinemas are concrete constructed which limits sound escaping from the room. The curtain material on walls absorbs high frequencies but has little effect on absorbing low frequencies as in the pic below. If pleated curtain material absorbs 50% (-3 dB) of sound energy at 500 Hz the sound would have to strike the curtain 20 times to be reduced to 0.0001% (-60 dB). -3 dB is only heard as a slight reduction.



Compare a small and large cinema made from the same building materials, and fire a cap gun. Sound travels at approx. 344 m (1125 ft) per second. Each time sound strikes a wall, a % is absorbed and a % is reflected, so on and so on. The average distance between walls floor and ceiling is described as 'Mean free path' or 'Mean free time'.

In a small cinema the average distance between walls is closer compared to a larger cinema. Sound will be absorbed and reflected from walls more often (in the same period of time). In a large older cinema without curtain material on walls, the hi-frequency reverberant energy will be less absorbed and appear brighter, except for air attenuation. However in a smaller modern multiplex cinema with curtain material on the walls the hi-frequencies will be readily absorbed but not the bass frequencies, therefore the reverberant bass energy will appear to be greater.



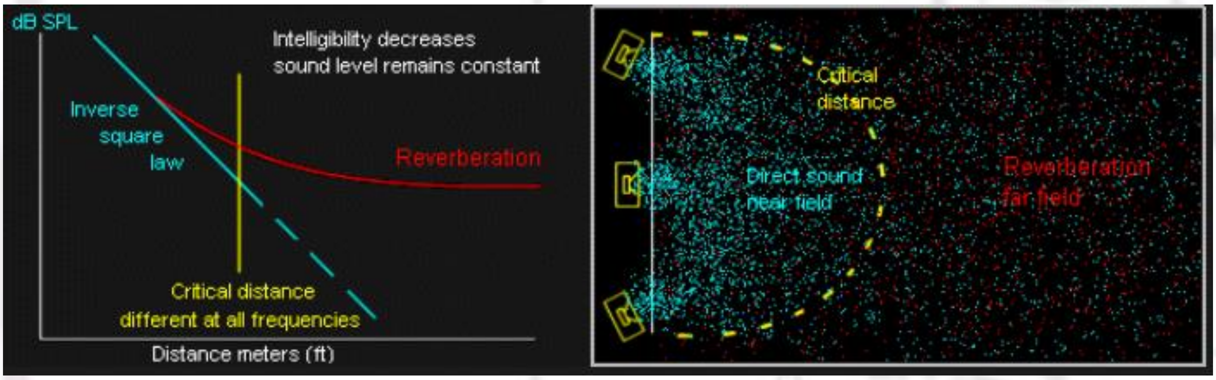
Bass frequencies have long wavelengths and due to propagation directivity, the inverse square law does not apply in a small cinema. This is similar to sub-bass in vehicles.

Due to the tradition of how cinema sound is calibrated in most multiplexes, the direct on-axis energy from the speakers is aligned to sound harsh or trebly, relying on the reverberant propagation bass energy, to fill in the difference. This traditional alignment approach is critically questioned further in this text.

In a large cinema the bass energy will have less propagation from the walls, and the inverse square law will apply to some degree. At two-thirds of the way back in a large cinema the bass energy will be less by comparison. Overall there is still insufficient absorption for low frequencies. This causes the lower end of the sound spectrum to be muddled. Action films depend on bass energy for effect, which is recent in film history, and most cinemas were designed before this trend came about.

Critical distance

The management of acoustics for entertainment venues requires understanding Critical distance. Critical Distance is where the direct sound from the speakers and reverberant sound energies are equal. The Critical Distance is different at all frequencies. The more reverberant a room, the closer is Critical Distance. The more absorbent a room, the further is Critical Distance.



Direct sound from the speaker system diminishes in level as a function of the distance (inverse square law) whereas reverberation constantly spreads throughout the cinema room. Because there is always new sound from the speakers, reverberation keeps building up until the incoming sound, equals the sound absorbed (steady-state). At speech the direct sound energy may be equal to the reverberant, but at the lower frequencies the reverberant energy may be 4 to 10 times greater than the direct.

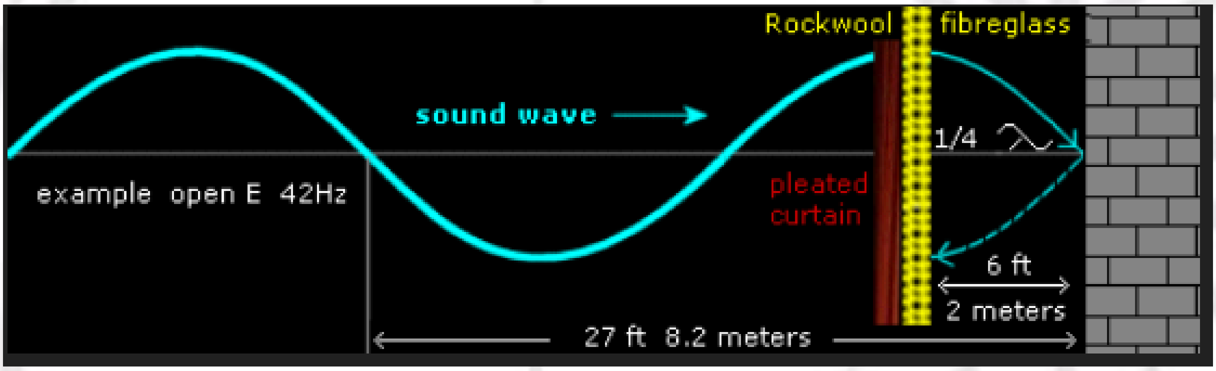
If reverberant energy becomes 12 dB or greater than the direct sound from the speakers, toward the rear of the cinema, intelligibility becomes lost. The simplest way to find 'Critical Distance' is to play compressed pop music through the sound system. Walk around the room, and you will be surprised how easy it is to identify the critical distance.

Good acoustic design in all entertainment and cinema venues requires the Critical Distance to be as far as possible from the sound source, and any resultant reverberation minimal and even at all frequencies. Un-even reverberation imposes annoying coloration in the sound. It is understandable that 100% anechoic is not physically possible in a large public environment but the closer to achieving it the better.

This text dictates that there is no such thing as one characteristic of reverberation that suits all applications and music. Those who argue for cinema room reverberation providing an aesthetic pleasing experience are describing personal preference only related to themselves, similar to chocolate verses vanilla. The correct reverberation is the decision of film director (not the cinema) heard from an excellent sound system, especially from excellent surrounds, preferably as a horizontal line source.

Acoustical absorption

Acoustical absorption of furnishing and curtain fabrics against walls, readily absorbs high frequencies, but have limited absorption at low frequencies. The further curtain fabrics are placed away from walls, the better the absorption is to include lower frequencies. The amount of sound energy absorbed depends on type of material, weight and pleating width. Rock wool (fiberglass) has the highest absorption capacity, converting molecular air movement to heat (at molecular level). Fiberglass consists of minute razor sharp fibers that are irritant and need to be contained within fabric.



The 1/4 wave-length rule: Acoustically absorbent material must be placed away from walls and ceiling, at a distance of 1/4 wavelength of the lowest frequency to be absorbed. This will include all higher frequencies if the absorbent material is soft furnishing or fiberglass. Please note that the ceiling should also be included.

Understandably, placing acoustical absorbent material 2 m (6 ft) from all walls may be thought of as impractical but the closer to achieving it the better. The slight reduction in the visual size of the cinema, will only be noticed when the lights are on. Acoustically the cinema will sound and feel LARGER. Also an acoustic absorbent environment is relaxing and calming and greatly enhances the enjoyment of the film.

Anechoic myth: There is a myth that being in a 100% anechoic environment is an emotionally negative experience. This myth needs to be well and truly busted. The myth possibly originated from experiments on university students during the 1950s, which were locked in a tiny anechoic test chamber, to represent being buried alive in a small airless coffin, with no possibility of escape, to find out how they would emotionally react.

Social experiments have proved that when living in a noise polluted city, to be able to find solace in a silent softly furnished acoustic absorbent environment, with free access to come and go as we please; all of us, that is, everyone without exception, relaxes letting go of anxiety and stressful thoughts.

When the lights are dimmed with beautiful music 'particularly classical' the experience becomes almost magical. The most wonderful enhancement before any film is a prologue of the music for approx. 5 minutes. If only cinema management fully understood this, what a magical experience going to the movies would be.

Fiberglass: Any cinema can be simply made to be close to anechoic at low cost, by simply approaching the solution in small steps. Fiberglass or Rockwool is in-expensive and can be placed into open low weight frames. There is possibly nothing else that will work as well, except asbestos. The fiberglass or Rockwool needs to be contained in fabric that stops irritant fibers getting loose. Polyester fiber has some acoustic absorption at high-frequencies but is ineffective at low frequencies, so don't use it.

The containing fabric should have to have its own fire retardant specification to comply with public building approvals. The installation procedure can be done all at once, or spread out over time in small steps so not to interfere with session times. Check that the installed procedure is compliant with correct fire and building regulations, including insurance.

'C' weighting: is the flat acoustic measurement response applied to sound systems and entertainment venues including cinemas and recording studios. 'A' weighting measurement is for environmental and industrial noise measurement and includes building materials. 'A' weighting is not a flat response measurement because it is insensitive to bass frequencies, similar to our hearing at a low loudness level.

'A' weighting: All building materials are quoted in 'A' weighting measurement and therefore these measurements cannot be used in reference to bass frequency acoustical absorption and transmission reduction when applied to cinemas and entertainment venues. However it is simple to do cross calculations to the 'C' weighting measurement to obtain a flat frequency response reference to include the lower frequencies.

Many companies that supply acoustic absorbent material that complies with fire regulations will advise correctly. Excellent information is available on their sites.

Many architectural companies that specialize in acoustics do exceptionally good work and need to be commended. But there are also questionable companies that market themselves as acoustic specialists. Many entertainment venues and cinemas have been caught out by building approvals that were for the 'A' weighting measurement only, and were later closed down because of excessive bass noise leakage.

Cinema Sound Alignment

Alignment by traditional method

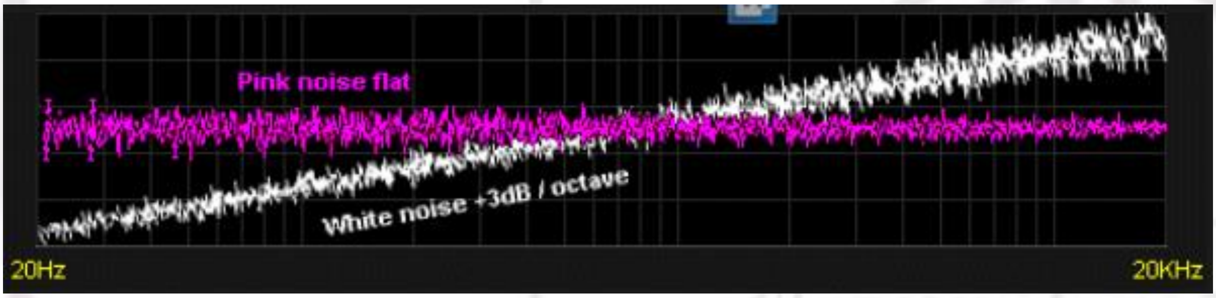
The cinema industry evolved before Radio and TV, therefore the later broadcast engineering standards have not applied. Some cinemas do the best they can to provide the highest quality vision and sound possible. But there is no requirement for a cinema to have identical left-center-right speakers, or for the amplifiers to be similar, or have calibrated gains. Components can also be hotchpotches from disused cinemas.

The Dolby processor alignment approach is based on adapting to the way the cinema industry evolved. The Dolby processor enables the whole result, from the AB chain, heard by the audience, to be calibrated from within the processor decoder. The recorded sound on the film stock, projector sound reader, decoders, noise reduction systems, EQ's, amplifiers, speakers, and acoustics are managed as a single entity.

This grouped procedure of alignment is unique to cinemas. It was introduced to simplify the procedure for non-technical people. Many audio and broadcast engineers of the cinema industry do not agree with this traditional cinema grouped alignment method. Professional recording studios and the broadcast industry align each item independently with absolute accuracy, usually done by electronic engineers.

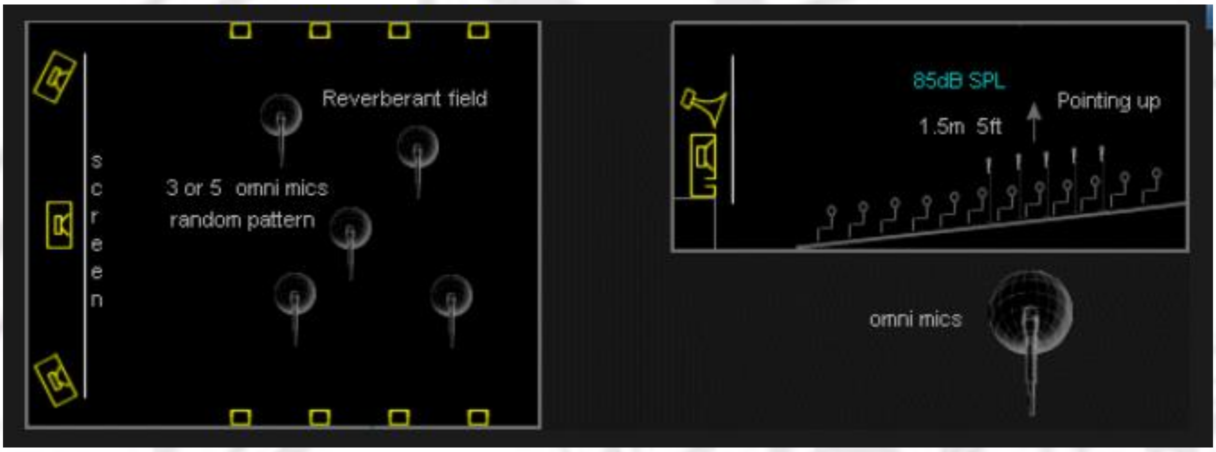
Grouping the whole A and B chain as a single entity is error prone and often results in chaotic outcomes with large differences between cinemas. One of the many limitations of the combined alignment procedure is, if any one item within the AB chain is modified or changed without an external 0 VU reference level, then the Dolby processor calibration should be repeated; which is rarely done.

Most alignment procedures are done with pink noise and it is essential to understand what this means. White noise is random noise we hear as hiss over the entire frequency spectrum created by all electronic circuits. White noise is also created in nature as wind rustling through leaves or surf at the beach. White noise has equal energy per cycle. As the frequency (cycles) doubles (for each octave), so does the noise energy (+3 dB/oct), resulting in white noise sounding trebly as hiss.



Pink noise is filtered white noise so each octave has equal energy, therefore a flat energy response and is similar to music and useful for acoustic measurements and approximations (only) for sound system alignment. This point needs to be repeated. Sound system alignment by Pink noise provides a simplified approximation only and is not an accurate method.

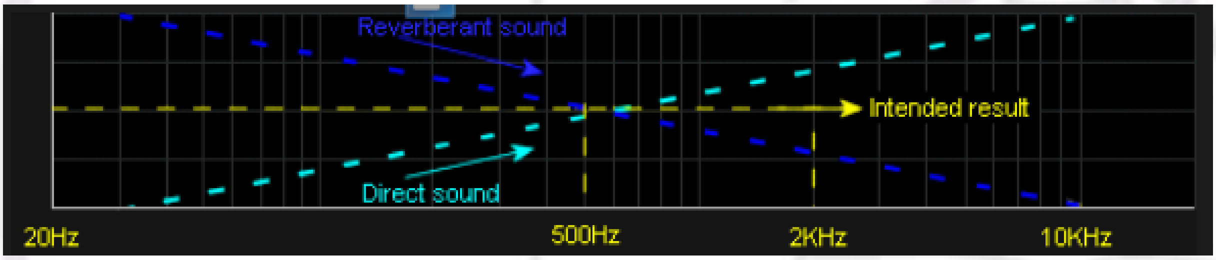
The basic procedure is to put 3 or 5 Omni Microphones spread out at two-thirds toward the rear in the cinema in the middle of the reverberant field. The mics are placed in a non-symmetrical random pattern so not to pick up standing waves. Pink noise from the Dolby processor is put through the sound system and the result is aligned so that the direct and reverberant sound energies combined give the required response.



85 dBSPL Reference: ('C' weighting) The most important reference for all cinemas from the pink noise sound level measurement is for the Dolby processor volume control to be set at No 7 at 85dB SPL in the cinema room. The surround level is then calibrated to -3 dB as 82 dBSPL. This reference is applied at the original recording, insuring the audience will hear the sound at the correct level the film director intended, with the processor level control set at No 7.

This general loudness alignment for the 85dB reference inclusive of direct and reverberant sound is universally accepted as being practical, but calibration for fidelity and articulation using this combined method are the issues that are hotly debated.

Alignment problems: The near field response of the speaker system (within 3 meters) is negated, as the traditional alignment procedure only looks at the combined direct and reverberant sound energies toward the rear of the cinema. Depending on the acoustics of the cinema, the direct on-axis energy from the speakers is aligned to sound harsh or trebly, relying on the reverberant and added propagation bass energy from walls floor and ceiling, to fill in the difference.



Because there is no reference in the traditional alignment procedure to address the difference between the direct and reverberant sound, then this combined procedure of alignment, that includes the chaotic behavior of the reverberant field, must seriously be brought into question.

Discerning music lovers and audiophiles only accept direct sound from a flat speaker system as having integrity. Recorded reverberation suited to the music is accepted.

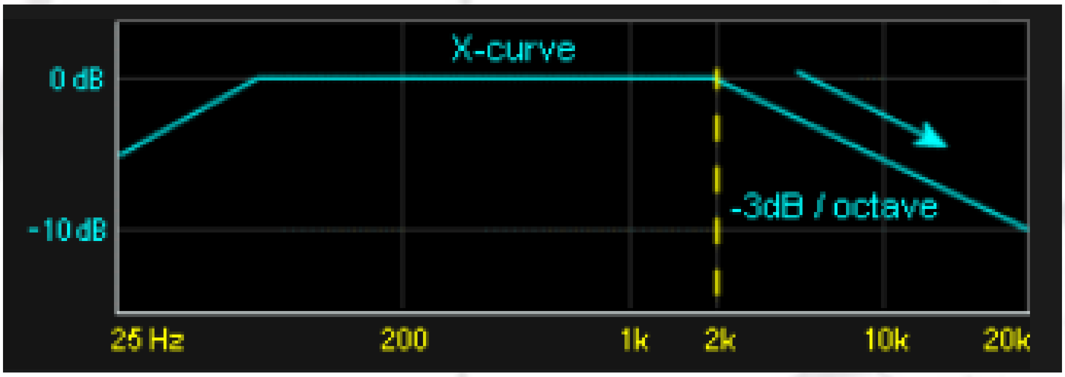
Unwanted reverberation from the listening environment (cinema) is regarded as chaotic noise. Using room reverberation (noise) as filler reduces fidelity and intelligibility.

Division of opinions: Many within the cinema industry are content with the traditional procedure. But just as many discerning people inside and outside of the cinema industry are dissatisfied. They believe that the fidelity of cinema sound falls far short of what can be achieved. It is argued that the majority of the public are not interested in how cinema 'sound sounds'. But there appears to be no un-biased research on the public discernment of the cinema experience.

Old JBL documentation of sound system application and alignment, states a sound level difference between the front to the rear of a cinema of less than 6 dB. Considering that the direct sound energy would decrease approx. 24 dB for the inverse square law, for an average large cinema, and the directivity of the hi-frequency horn would give an approx. 6 dB improvement, that means approx. 12 dB of reverberant energy is accepted. It is difficult to understand how any intelligibility is heard with this level of reverberation at the rear of a cinema.

X-Curve Alignment

All cinemas are meant to be aligned to the creed of the 'X-Curve' without question; The X-curve was originally created to provide consistency to the traditional alignment method. The X-curve is a high frequency roll-off beginning at 2 kHz at -3 dB/oct, then - 6 dB/oct from 10 kHz. Various correction factors are given for different size cinemas. ISO Bulletin 2969 'Curve X' is also described in ANSI PH22.202M-1984 / SMPTE 202M.



There appears to have been an examination of many cinemas over 30 years ago, and the X-curve is said to have resulted from pink noise measurements of an assumed average cinema. The observed result was that the low-frequency reverberation was greater than the hi-frequency reverberation. This is understandable because concrete constructed cinemas with curtains on walls will readily absorb hi-frequencies but have less effect on absorbing low-frequencies.

Measurements were done at 2/3 toward the rear of the cinema to include the reverberant field.

Equalization was then applied to the cinema speakers so the result would approximate the energy spectrum of recording studios monitors, heard at the position of the mixing console.

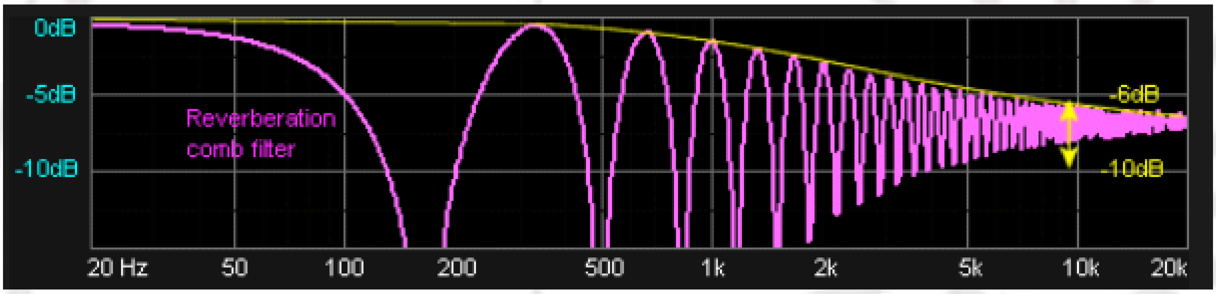
The shape of the equalization applied to the cinema speakers is said to be the X-curve.

There are many who argue that the X-curve has no beneficial effect, and has further complicated the original problem. Depending on the acoustics of the cinema, the direct on-axis energy from the speakers is aligned to sound harsh or trebly, relying on the cinema reverberation, to fill in the difference. Critics say that the X-curve does not discriminate for fidelity and articulation as it makes no difference if the reverberant sound energy was replaced with 100 monkeys randomly bashing tin cans. All that matters is that the combined energy response heard by the cinema audience (regardless of what it consists of) is similar to the energy response of the original recording heard at the mixing console.

The detailed rationale and arguments behind the X-curve have not been fully explained to scientific and engineering people on the outside of the cinema industry, which are mostly un-aware that the X-curve exists. 'Why is this so?'

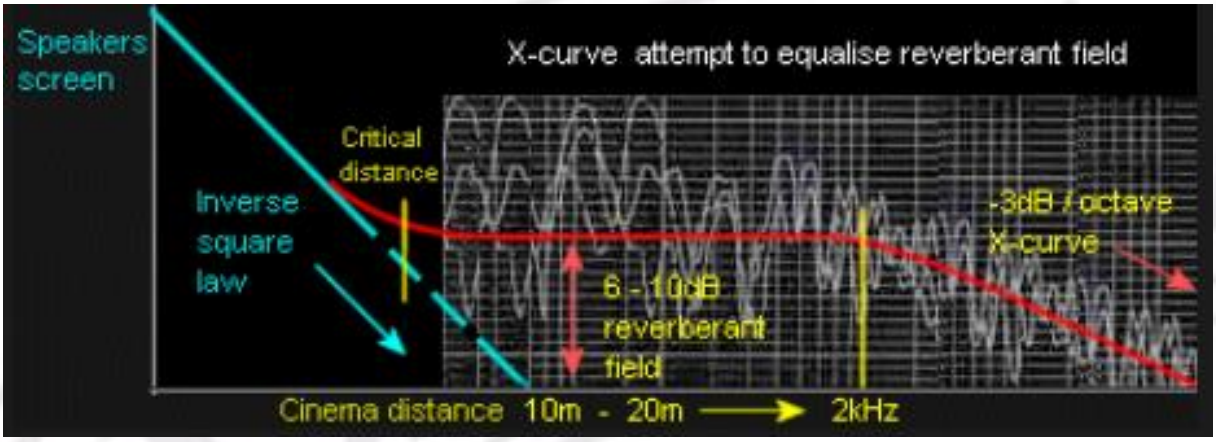
External reference: A possible reason for the lack of clarity about the X-curve is that it appears to have no external reference outside of itself. An external reference is essential for logical assessments otherwise miss-conceptions occur. One example of logical discrepancy infers that; the larger the cinema, the greater the hi-frequency reverberation. This implies that an infinitely large cinema would have infinite hi-frequency reverberation. If the X-curve roll off is meant to compensate for this, then an infinite hi-frequency roll off is required to compensate. (An infinitely large cinema would have zero reverberation because it would be a free field).

In reality the same sound arriving at different times from being reflected from walls and ceiling can result in approx. -10 dB loss of high frequency energy toward the rear of an excessive reverberant auditorium or cinema. These cancellations are described as a comb filter, simplified in the below pic.



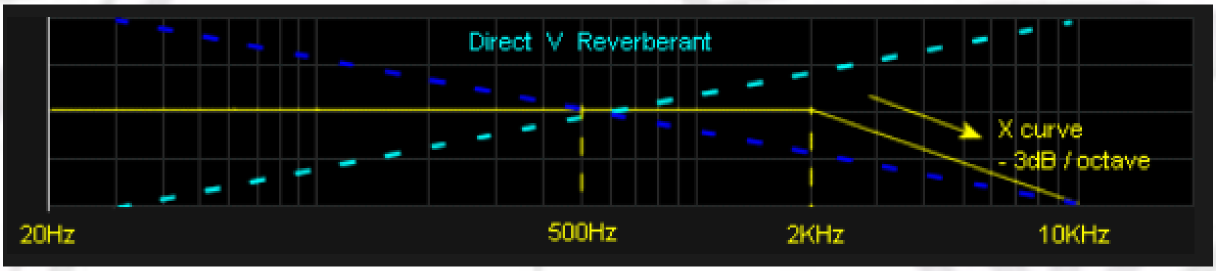
False assumption It is easy to assume that the reverse could be applied. That is, regardless of the reverberation, including that the sound system behind the screen does not need to have a flat frequency response, it does not matter.

Because, by placing a microphone in the reverberant field with a 1/3 octave graphic equalizer, then by simply adjusting the controls, the result can be made to align to the X-curve as if there was a flat speaker system behind the screen similar to the recording monitor. Therefore it appears that a 1/3 octave graphic equalizer (or its equivalent) solves all problems.



Nobody is 100% sure of this because it is not directly stated. Different authoritative sources have different interpretations of this description. But this reverse procedure based on assumption regardless of how it is described appears to have become the X-curve alignment.

Is it possible that many years ago, a committee established to address the sound alignment problem, decided to align sound systems to comply with the apparent roll off above 2 kHz at -3 dB/oct (heard at 2/3 of the distance from the screen) as a simple means to provide consistency between older cinemas that did not have acoustical absorption on walls?



Or was the obscure aim of the X-curve procedure possibly intended to provide a uniform energy response to compensate for reverberation in an assumed average 500 seat cinema, to which cinemas of all other sizes can be then be adjusted or somehow conformed to? Or was the X-curve procedure supposed to provide a common base-line between mixing rooms and cinemas?

A simple solution is to put a blanket over every cinema so all cinemas will sound the same. It is no more effective than if everyone put cotton wool in their ears, the outcome would be the same.

The objective behind this statement is not to negate the positive intention of those who originally initiated this idea many years ago, when circumstances were very different to today, but to challenge those who un-questioningly continue to practice this procedure by providing scientifically proven data to justify why it should be sustained.

Therefore three rhetorical statements need to be made.

1. Generalized assumption. How specifically does the X-curve provide a uniform energy response or common base-line between mixing rooms and cinemas?

2. Deletion of proof. How or where is the evidence for a uniform energy response or a common base-line obtained?

3. Distortion of logic. How can change RT with EQ be made comprehensible in logic?

Most modern multiplexes have pleated curtains on walls which effectively absorb hi-frequencies, plus hi-frequency air absorption and screen attenuation, including that most front speakers are 2-way without tweeters; all together approximate the X-curve. When combined with the compulsory X-curve alignment procedure, it does not make logical sense and now appears as a contradiction.

The home cinema 're-EQ' for small room acoustics in THX approved systems is supposed to be a variation (whatever that means) of the commercial cinema X-curve. But there does not appear to be a clear un-ambiguous definition of sound alignment procedures in recording studios for the X-curve to be referenced to. There is no equivalent of this type of alignment in the live entertainment industry, and none of this has any effect on changing the reverberation time RT.

Recording engineers easily override the X-curve by boosting the hi-frequencies when recording. As we know, film sound is brighter and harsher than music CD's. There is no absolute way of knowing what the final mix will sound like until the film is released. Recording engineers directors and producers often go from cinema to cinema to check the differences. It is only from trial and error experience do recording engineers formulate their individual approaches to compensate for the X-curve alignment.

1/3 octave graphic equalizer Possibly the most technically flawed trend in audio history began in the 1970s with the marketing of parametric and 1/3 octave graphic equalizers for solving room acoustic problems, which continues to this day with the miss-use of digital signal processors DSPs.

Alignment by science

The primary emphasis should be on using hi-quality sound systems to begin with. Sound alignment should be for achieving the highest articulation possible regardless of the combined energy response. A more practical approach is to calibrate the sound at or within the critical distance, at no greater than 1/3 of the distance from the screen, and definitely not in the reverberant field. This requires understanding and identifying Critical distance, Near field and Reverberant field as being separate.

The only way to prove or disprove that the X-curve has any potential value; it first must have a reference outside of itself in a scientific context. This initial reference must be based on the original recording being monitored with a flat speaker system in an anechoic environment, and then heard by an audience in a anechoic cinema from a flat sound system.

This must be stated in the most succinct manner so no miss-conception can be interpreted from this theoretical reference. Therefore any deviation from a flat anechoic reference that proves to be beneficial can be explained in a technical manner to insure accurate compliance and consistency.

Sound system technology is sufficiently advanced to enable flat 4 way active sound systems to be applied in mixing studios and cinemas. Building resources are readily available to create close to anechoic environments. The only acoustical alignment that should be required is for screen and air attenuation and compensating for lower frequencies from increased directivity (propagation) from the floor and walls (if necessary).

Cinema sound alignment engineering reference

All Radio TV stations and recording studios throughout the world comply with international broadcast engineering standards for 0VU line level and calibration of every item within the entire broadcast and recording system. This provides consistency, reliability and most important, inter-changeability.

This text provides ideas only for thinking through the procedure. The steps are easy to follow. However each installer or management of cinema complexes must develop their own detailed procedures suited to their application. The installer must have a good command of dB conversion and ohms law. The Dolby processor is to be 0VU referenced and used only for its primary functions. All EQ within the processor is neutralized. EQ for cinema acoustics and or speaker system is done external from the processor.

85dBSPL Broadband pink noise measurement at approx. 2/3 distance in the cinema room for calibrating the Dolby processor volume control at the No 7 position. The pink noise bandwidth should be first limited to 500 Hz – 2 kHz for best alignment accuracy of voice frequencies. Then increased to broadband. But broadband pink noise is often the only option available. The reason many cinemas have in-consistent loudness levels is because the 85 dBSPL reference can be applied in 4 different ways.

1. 85 dB from 1 screen channel inclusive of reverberation. 85 dB from 1 screen channel inclusive of reverberation appears to be more consistently stated within various documentations. When all 3 screen channels are combined the loudness will increase 5 dB to 90 dBSPL.

2. 85 dB from 3 screen channels inclusive of reverberation. 85 dB from 3 screen channels inclusive of reverberation, is 80 dBSPL from each channel and therefore the quietest alignment, and is the preference of this text

3. 8 5dB from 1 screen channel at inverse square law (anechoic). 85 dB from 1 screen channel at inverse square law (anechoic).

4. 85 dB from 3 screen channels at inverse square law (anechoic). When all 3 screen channels are combined the loudness will increase 5 dB to 90 dBSPL

Then with the reverberation added it may be 3 dB to 10 dB louder again giving a total of 93 dBSPL to 100 dBSPL.

Loudness: The reason this text prefers option 2 is because 85 dBSPL in a quiet cinema is experienced as being loud. Whereas in the streets of a noise polluted city 85dBSPL is not loud compared to traffic noise. Many earlier cinemas did not have acoustic absorption on walls and therefore had excessive reverberation. Reverberation has a similar effect to traffic noise pollution, by masking dynamic range and requires one to shout to be heard. This is possibly the reason a loud alignment option was originally chosen.

Most modern cinemas are quiet by comparison to earlier cinemas, therefore the quieter 85 dB alignment may now be the best option (No2). Many recording engineers monitor too loud and become deaf and then turn up the level to compensate. A cinema referenced at approx. -6dB below recording monitored level appears to provide a more consistent result.

No7 = -20 dBFS: No7 on the Dolby processor represents -20 dBFS, therefore the maximum level available is another 20 dB. This extra level is rarely used but it is available for special productions that require the audience to hear extreme sound levels. Hopefully for only short periods of time.

105 dBSPL: The individual power from each of the left-center-right speakers is meant to achieve 105dB SPL for maximum dynamic headroom. This means an extra 20 dB (x100 more power) above the 85 dBSPL reference. When all 3 channels are combined the result will be an extra +5 dB approx making a grand total of 110 dBSPL. This extreme loudness is beyond the safety OH&S limit. Therefore the 110 dBSPL measurement must be regarded as academic to insure there is sufficient headroom so no distortion is heard.

The 105 dBSPL measurement should be referenced to free field (anechoic) and calculated at inverse square law. Alternative text refers to this as being ‘first arrival’ separate from reflected sound from walls and ceiling. The following examples are approximations for a +300 seat cinema with 2/3 distance being 16 m (54 ft).

Inverse square law at 16 m (52 ft) = -24 dB

Each speaker (2x15 inches + horn) ≈ 100 dB/mW and 400 W power rating.

400 W is +26 dB therefore sound level at 1 m = 126 dBSPL

At 16 m (-24 dB) the sound level at 400 W = 102 dBSPL

All left-center-right speakers at 1200 W gives an extra +5 dB = 107 dBSPL at 16 m (54 ft)

These free field measurements are -3 dB short of the academic objective, but when the speaker system is applied to a cinema, the reverberant energy will add a minimum of +3 dB to these measurements. Therefore a total sound level of 105 dBSPL will easily be achieved with 400 W amplifiers. A cinema with a 10 dB reverberant field will only require 80 W to each speaker to achieve the loudness objective, but intelligibility will be poor.

<u>Alignment steps</u>

1. Speaker system: The speaker system should be designed and calibrated to achieve the correct directivity and a flat frequency response in a free field environment. This measurement is done by placing the speaker box on its back in a large field facing upward and measured at 3 m (10 ft) above the speaker. This is similar to the speaker system being placed in a baffled wall.

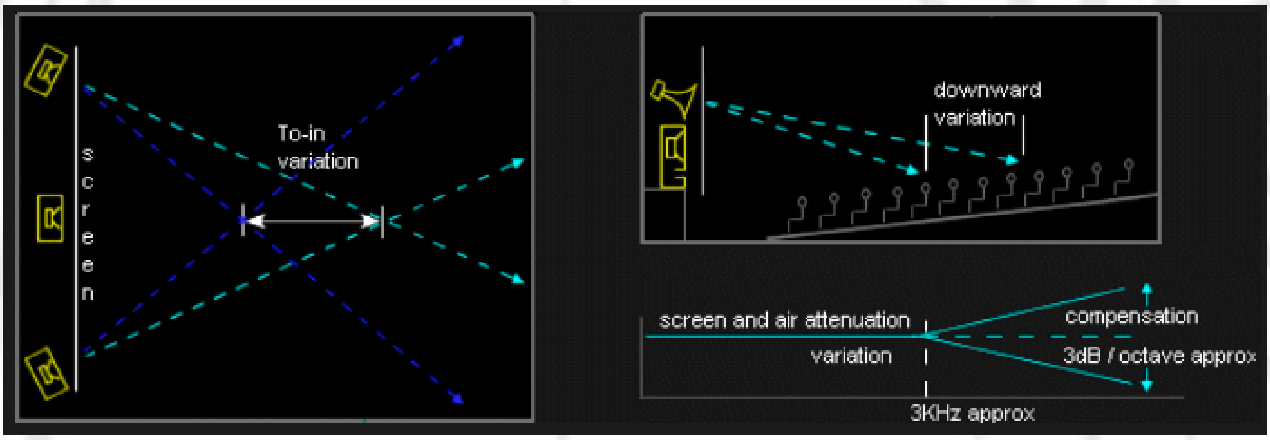
Identically matched: The left-center-right speakers must be identically matched before being installed. These measurements can also be achieved within a closed environment. However this requires greater skill and practice with extensive knowledge of acoustics and test equipment, to assimilate free field conditions.

The speaker system is then placed in front of the screen and measured for how it sounds, before placing it behind the screen. Providing everyone is happy, the speakers are permanently mounted behind the screen and re-measured. The speaker system will behave differently compared to the free field measurement.

2. Sound system EQ: Independent correction EQ for lower frequency propagation from the floor (if necessary), including hi-frequency boost for screen and air attenuation These EQ adjustments must be identical to all 3 screen speakers. This can be done with an independent dedicated parametric system, a 1/3 octave EQ or a digital signal processor DSP. The EQ results must be double checked and logged for future reference. Use independent EQ for surrounds.

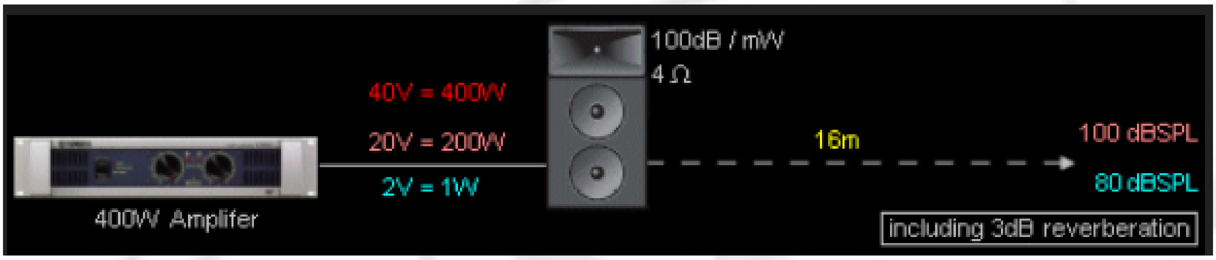
Although these EQ facilities are duplicated within the Dolby processor, it is strongly advised to use the processor for decoding only. Repeat – avoid the temptation to use the Dolby processor for anything other than its primary functions.

Insure that the horns are downward angled at the best position for voice articulation center seating. Insure the left-right toe-in gives the best stereo image possible. The toe-in may be required to be of a greater inward angle than anticipated.



As a general rule a toe-in closer to the screen and at an increased downward angle is required for excessive reverberant cinemas to help reduce reflections from side walls and ceiling.

3. Amplifiers: For all 3 left-center-right speakers to achieve 85 dBSPL at 16 m, each amplifier only has to drive its speaker to 80 dBSPL. For the quieter No2 reference as previously stated. 16 m represents an inverse square loss of -24 dB. Allowing for the reverberant field contributing approx +3dB, the inverse square law loss can be rounded to -20 dB. Therefore each speaker only has to deliver 100 dBSPL at 1 m for the sound level to be 80 dBSPL at 16 m.



2 Volts into a 4 ohm speaker box is 1 Watt. ($V^2/R = W$), 2 V x 2 V / 4 Ω = 1W. The speaker system is 100 dB/mW efficient and will give 80 dBSPL at 16 m, which includes the reverberant energy. All 3 speakers will add +5 dB giving the required 85 dBSPL reference.

With modern 400 W + amplifiers, the power head room calculations are no longer a problem and often do not need to be calculated. Power head room calculations were necessary in earlier years when power amplifiers were less than 100 W.

Amplifier gain Amplifier manufacturers have not agreed to comply with voltage gain standards, which is irresponsible to say the least. Some amplifiers have un-calibrated attenuation (level) controls, but many have nothing. Calibrated attenuation (level) controls are absolutely essential. This problem causes wasted time and extra calculation headaches for installers. Amplifier gains vary between (26 dB) 20:1 to 40:1 (32 dB) or greater. Specified amplifier gain will be consistent to each model number only.

Amplifer Gain

Gain 20 = 26 dB

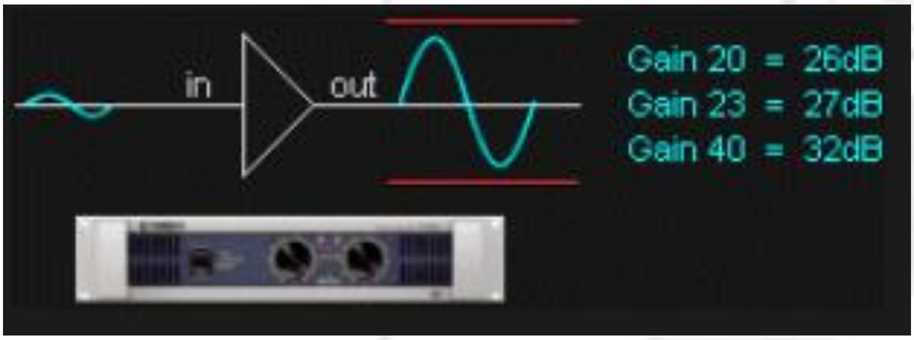
Gain 23 = 27 dB

Gain 40 = 32 dB

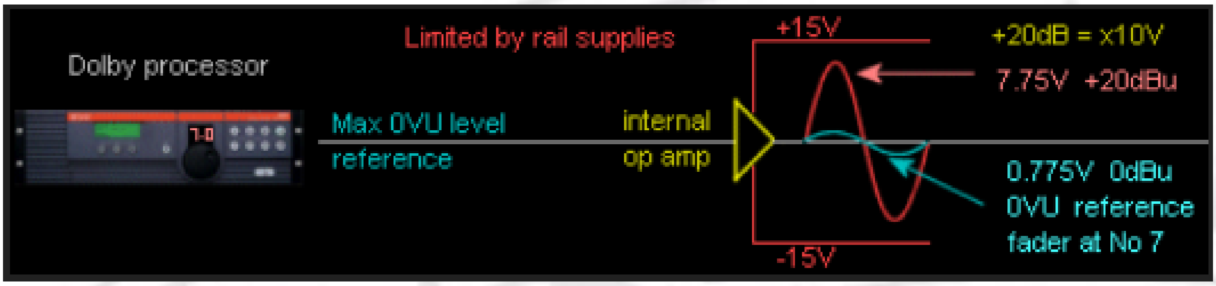
Some Amp specifications can be without reference to gain

0 dBu 0.775 V = 1/2 power

3 dBu 1V = full power



4. Processor level: Line level 0 VU (Volume units) reference must be set for the processor. The highest 0 VU reference level the processor can be set, is limited by the maximum signal level above the 0 VU reference, which is 20d B (x10) greater.



The internal supply rail Voltage for the operational amplifiers is (+15 V) – (-15 V)=30 V total. The maximum theoretical RMS signal Voltage at the on-set of clipping is calculated by (PP/2) x 0.707. (30V/2) x 0.707 = 10V RMS approx. Allowing for a 3dB headroom reduces this figure to 7.07 V RMS.

Therefore the maximum setting for an 0 VU reference is (0.775 mV) 0dBu for fader position No 7. The maximum signal output that can be obtained from the processor is another 20 dB which is +20 dBu (7.75 V). All figures refer to un-balanced output only.

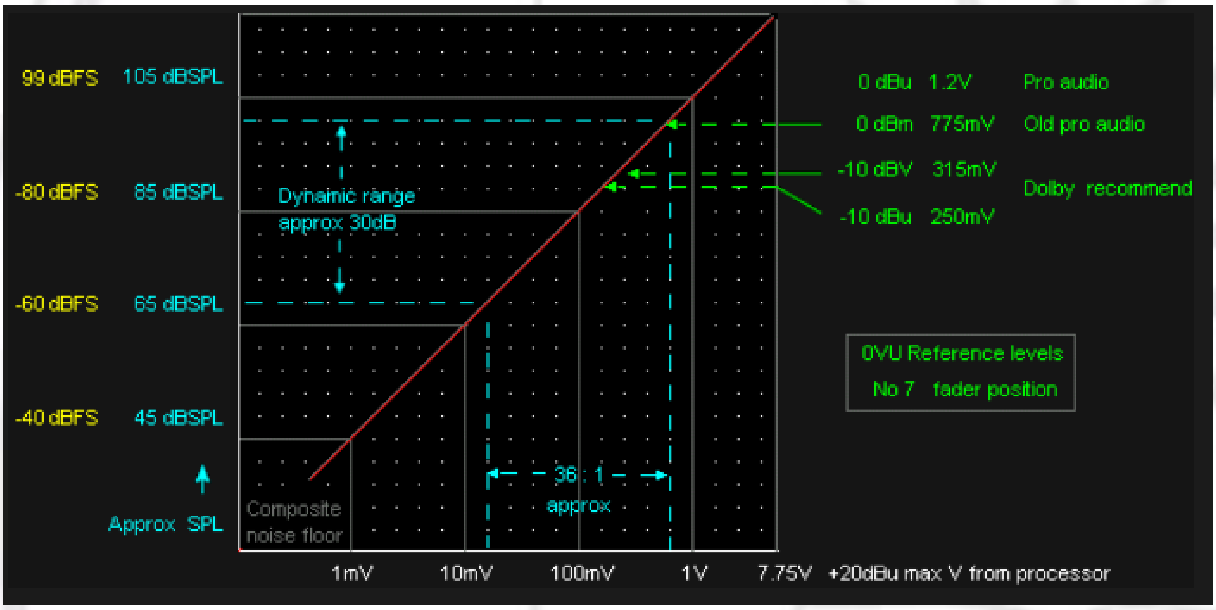
From 0dBu (775 mV) as 0 VU +20 dB = +10 dBu (7.75 V).

+4 dBu (1.2 V) as 0 VU is the standard for the majority of recording studios, but with an added 20 dB (x10) this would result in 24 dBu (12 V) well into clipping, therefore this 0 VU reference cannot be used. Dolby recommends a typical 0 VU calibration level of 300 mV, which is very close to 315 mV or 250 mV, depending on various documents.

From -10 dBV (315 mV) as 0 VU +20 dB = +10 dBV (3.15 V).

From -10 dBu (250 mV) as 0 VU +20 dB = +10 dBu (2.5 V).

0 VU level reference



If the Dolby processor is set to 0 VU reference of -10 dBu (250 mV) the output from the average film may vary between 20 mV–780 mV (36:1) approx. 30 dB dynamic range. Digital technology allows for a 99 dB dynamic range. But because of inept recording practices that rely on excessive over-use of dynamic compression, film sound rarely has a dynamic range that exceeds 30dB. The majority of pop recordings are dynamically compressed to within 10 dB–20dB including TV and radio broadcasting.

5. .1 Sub-bass SMPTE RP 200: The .1 sub-bass channel meant to be set at +10 dB above the O VU reference. The reason the .1 sub-bass channel is increased + 10dB is because it is recorded at -10dB below the main channels. The procedure of recording the sub-bass at a lower level was first established with earlier analogue magnetic recording because loud explosion effects could easily overload the tape or recording console causing excessive distortion. The tradition has been retained with modern digital recording even though it is no longer necessary.

6. Calibrated attenuation: If the amplifiers have level attenuation controls, they can be used but only if they are able to be correctly calibrated and logged. Otherwise turn all level controls fully clockwise so they are out of use. Mark each amplifier with clear identification as to its gain. Then use an external attenuation system in the rack between the processor and the amplifiers.

The signal from the Dolby processor must go through the separate calibrated attenuation system to reduce the signal to the correct signal levels to each of the amplifiers. Our chosen amplifier has a gain of 20:1 (26 dB). The 0VU signal level from the processor must be reduced to 100 mV (-18 dBu).

0 VU (-10 dBu) 250 mV x 0.4 is reduced to 100 mV (-18 dBu) (-8 dB att)

0 VU (-10 dBV) 315 mV x 0.315 is reduced to 100 mV (-18 dBu) (-10 dB att)

100 mV to the amplifier input x 20 will give 2 V at the amplifier output. The speaker system is 4 ohm. ($V^2/R = W$) 2 V x 2 V / 4Ω = 1W. The speaker system is 100 dB/mW efficient and will give 80 dBSPL at 16m which includes the reverberant energy. All 3 speakers will add +5 dB giving the required 85 dBSPL reference.

Surround gain: The surround channels should be -3 dB below the O VU reference and finally adjusted by ear. Because the amplifiers for the surrounds may be different brands or models, their internal gains may also be different; therefore the attenuation to surround amplifiers has to be adjusted separately to achieve the correct level to the speakers.

The X-curve: Yes, what “do” we do about the X-curve?

Choose different films that are used for reference and listen. Surprisingly the sound may be absolutely fabulous and if so, do nothing else. However some films are recorded with excessive hi-frequency boost by directors trying to override the imposed X-curve.

First try switching out the hi-frequency EQ boost for air and screen attenuation. This may co-inside with obtaining the correct outcome, if so, arrange for this to be easily switched for the different films. But be prepared to install a separate EQ unit that can be switched in or out when required. These final decisions make the installers life very interesting.

<u>Overview rules</u>

Reference Dolby processor to an 0 VU level. E.g. -10 dBu at Fader position No7

Set Dolby processor for: (85 dBSPL at No7) (surround delay) (-3 dB less for surround level) (+10 dB .1sub-bass level)

Do not use Dolby processor for anything else

Use independent attenuation for adjusting level to the amplifiers

Do not apply EQ or level correction that differs between the front speakers

Do not attempt to apply EQ adjustments for anomalies in the reverberant field

Stereo music: being played before the beginning of a film as a prologue greatly enhances the enjoyment of the cinematic experience, but checks the CD result through the Dolby processor first. The matrix techniques designed to spread 2 channel stereo into the 5 channels uses various phase differences between the left-right channels which can possibly introduce annoying distortion in the surround speakers. The best approach is for the stereo music is sent to the left right screen speakers and also in mono to the center screen speaker at -3 dB.

Microphones: For special presentation events that require people speaking on stage in front of the screen where a microphone is required. Avoid putting the microphone through the Dolby processor as it is difficult to manage. However if there is no alternative; select the microphone to be heard from the center screen channel only. Hearing a person speaking through the surround speakers may give an un-natural colored sound with low articulation. Microphone feedback can easily destroy surround speakers, which has often happened.

Voice in films is naturally heard through the center screen speaker. Because the speakers are 2/3 toward the top of the screen, microphone feedback is not a problem. The clarity of voice through the center speaker is superior in quality to the majority of PA systems used by musicians, and the articulation will be excellent.

Cinema Digital

Mass entertainment and home entertainment both have their place and are not in conflict with each other. Mass entertainment is ideally suited for productions by exceptional artists where we can collectively enjoy and be inspired by their work being exhibited on a large scale.

The technology for sound system power and fidelity is now unlimited. Digital audio is now available in loss-less format which can achieve a flat response 20 Hz – 20 KHz with zero distortion or noise. We now have the technology with unlimited building resources to create new projected entertainment centers acoustically free of echoes and reverberation. There should be no need for the X-curve or any other external acoustical equalizing for solving past reverberation problems. The new venues should have exceptionally superior sound by comparison to the present cinemas including live venues and home cinema.

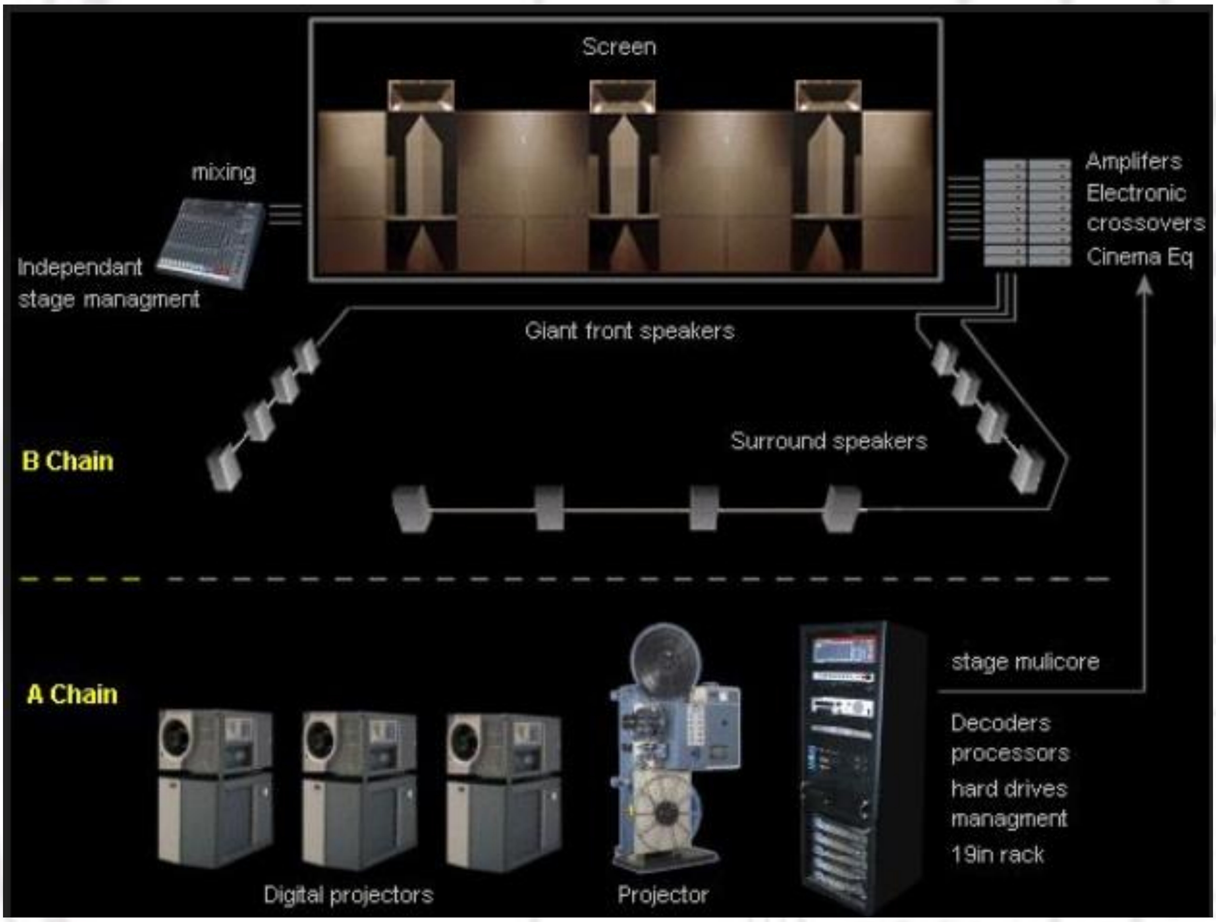
Sound should become 50% of the entertainment experience, where it truly belongs.

There should also be open competition between venues with the latest technological innovations. With digital projection the management formats should be made open architecture, similar to live theatre and live concert productions. This should allow variations in the number of projectors that can be used to create wide screen panoramic experiences similar to the 1950s Cinerama. The ultimate quest being the inevitable arrival of large scale projected 3D.

Satellite technology can transmit live musical concerts or live theatre including Olympic games and other world sports events in real time. Multi-projectors can create wide screen panoramic experience giving the feeling of actually being at the event, with the added value of viewing and hearing greater detail.

This freedom will unleash waves of creative artists and directors competing with innovative productions including real time interactive games with audience participation throughout the world. Ideas that in the past would have been thought of as crazy will invigorate the industry. Independent film creators should be able to negotiate directly with projected entertainment centers for the best facilities and locations to exhibit their work.

Projected entertainment centers should retake the position that traditional commercial cinemas had in past; of being the future leaders of a new Golden era of entertainment.



The B chain will need to be independent, with the amplifiers being placed with the screen speaker system. This will include independent mixing access to be compatible with live entertainment.

This separation of B chain management will enable broadcast type engineering standards to be applied enabling a greater accuracy for sound system alignment, as well as serviceability and easier up-grading if and when required. Another business advantage for full separation of the AB chains is that it will give film distributors independent management and greater security of digital films in the projector room.

THX

George Lucas who created the 'Star Wars' films, introduced THX in 1983. THX represents a classification of minimum standards for seating, air-conditioning, vision and sound etc. as an attempt to get cinemas to exhibit films as the film maker intended. The THX acronym is 'Tom Holman's Crossover. Loud-speaker cross-over is abbreviated as X-over, or eXperiment, or eXcitement.

THX is based on a business model for standards compliance described as 'benchmarking'. But benchmarking eventually found its way into the fad driven business world, where it did not belong. The unfortunate outcome of institutionalized conformity in the private business world is crippling bureaucracy, escalating costs, stifling innovation, decreased competition, with the end game resulting in monopolistic practices.

Many film makers believed there needed to be minimum standards for cinema exhibition, to address the cost cutting regimes cinema chain accountants were willing to implement on behalf of shareholders. The primary conflict of interest is that cinema chains and film distributors regard a film's success (in the first 2 weeks) as only related to marketing of celebrity stars and superficial biased reviews, and not to exhibition quality or film production excellence.

Hardware purchasing decisions for cinemas that had previously been based on comparing technical performance or competitive pricing, now shifted to brand image identification. New cinemas particularly multiplexes became built by purchasing and plugging in a series of marketed brand names. Profit driven compliance organizations market themselves as an overseer of a club of brand names. The primary purpose is to re-enforce brand image by adding its own brand image to the product or cinema, for which it charges a fee. Its altruistic purpose is to insure elementary standards are incorporated within the approved brands and completed cinemas.

Due to the high cost of obtaining THX approval, some cinema chains see this as yet another un-necessary expense and easily circumvent the THX spirit by attaining accreditation for only a few of their cinemas and apply the THX logo to all their other cinemas, regardless of whether they were of the standard required.