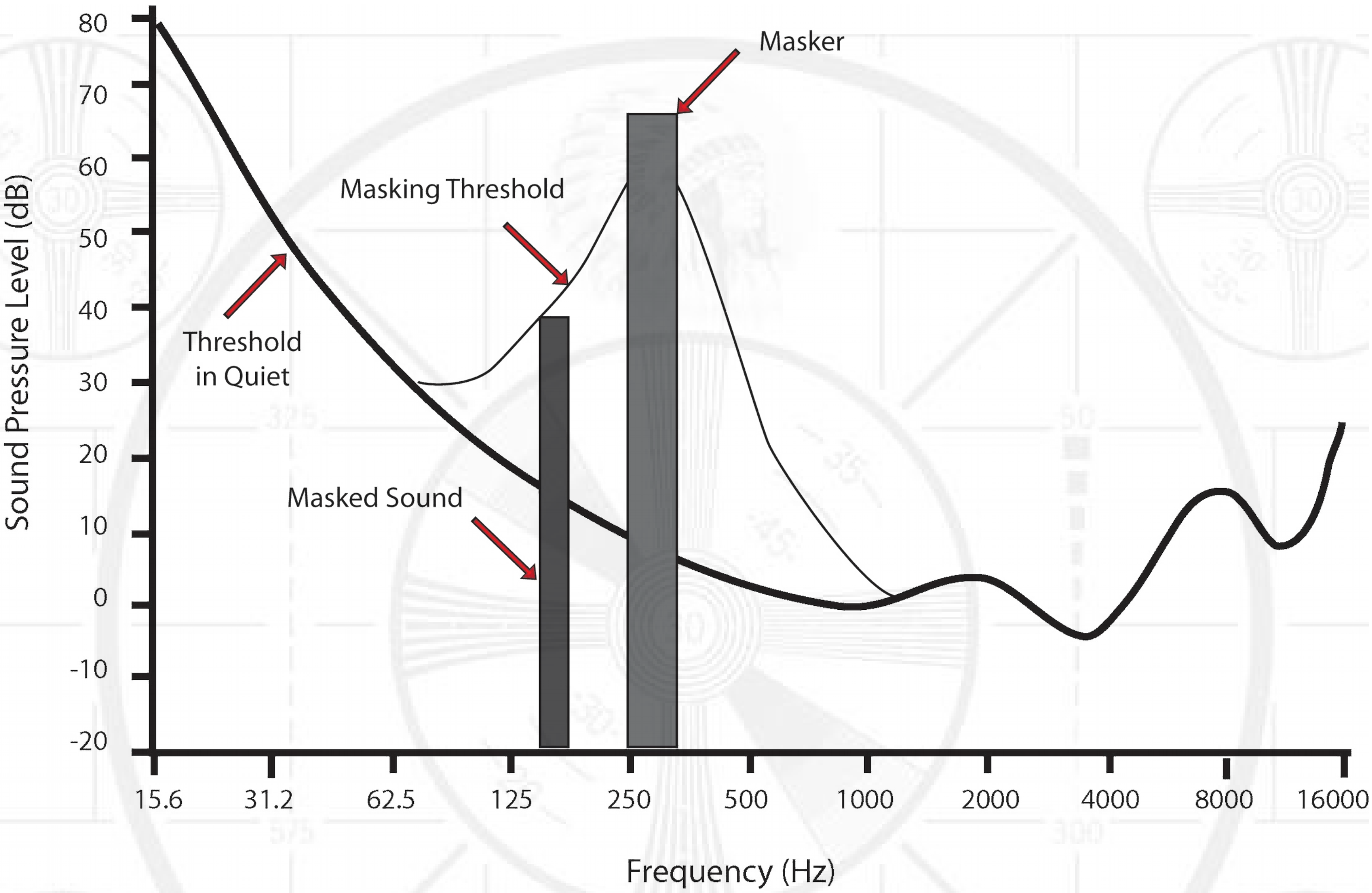
The Media Streaming Journal

August 2016



Covering Audio and Video Internet Broadcasting

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The Media Streaming Journal Staff

Derek Bullard
Publication Director
info@radiosolution.info

David Childers
Editor In Chief
editor@radiosolution.info

Advertising
advertising@radiosolution.info

www.radiosolution.info

Welcome to The Media Streaming Journal

Greetings,

Could Live 365 be making a comeback?

The Live 365 website has been updated with some vague references of a new life.

"We are music curation.

The fun isn't over yet.

Live365 will be right back. pr@live365.com

Looking to start your Internet Radio station again? support@live365.com"

http://www.live365.com/

Rain News contacted Live365 regarding this and their reply was ...

"The Live365 assets have been acquired and we will make an official announcement in the coming weeks. We are still finalizing plans."

http://rainnews.com/live365-set-to-return-big-news-for-small-webcasters/

Please feel free to contact either the Publication Director (Derek Bullard) or myself if you have any questions or comments regarding The Media Streaming Journal.

Namaste

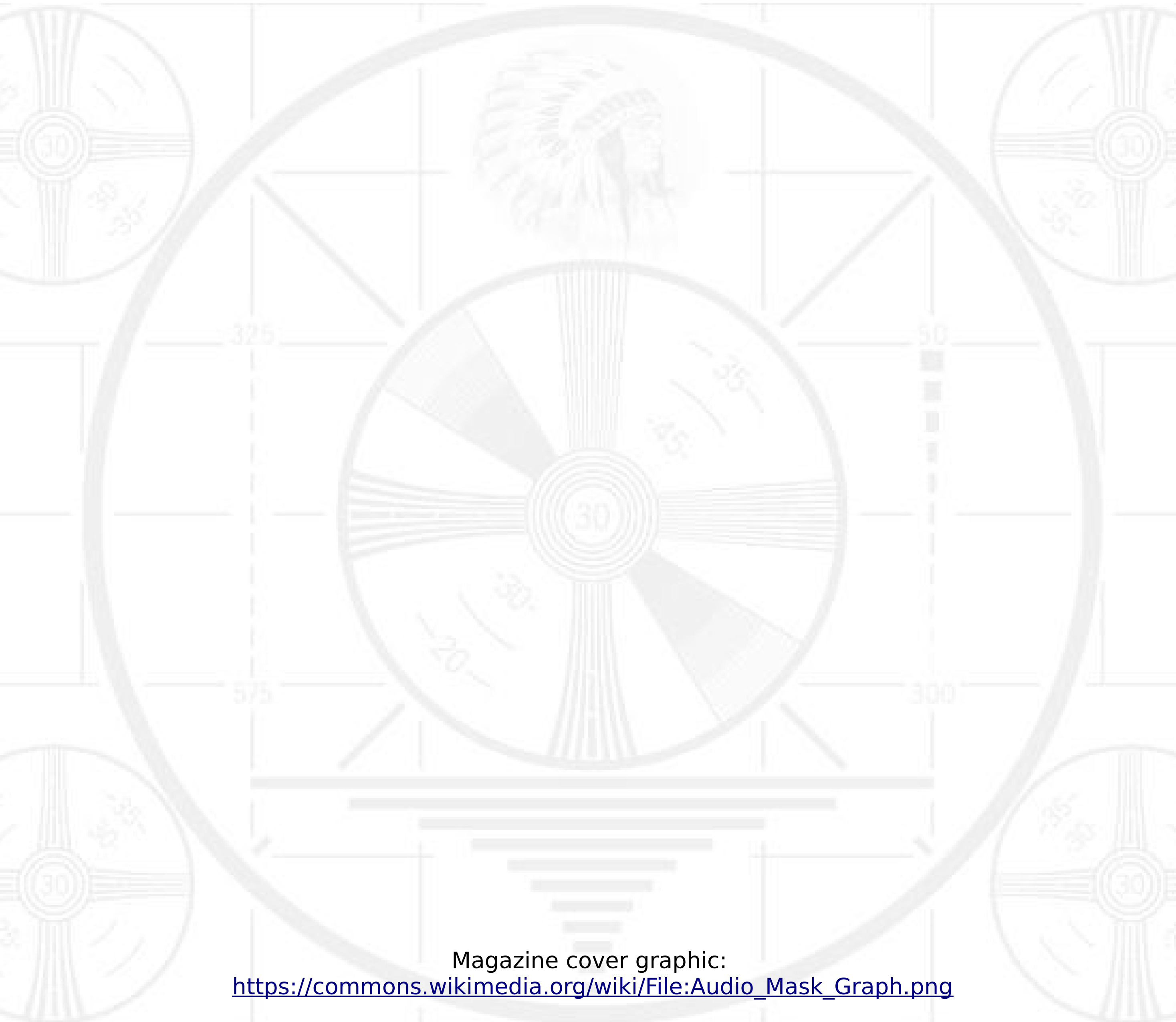
David Childers

The Grand Master of Digital Disaster
(Editor In Chief)

www.linkedin.com/pub/david-childers/4/736/72a

The Media Streaming Journal

What is in this edition of the Media Streaming Journal



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Let our friendly, knowledgeable staff assist you to build your project, such as an online radio station using our high end reliable video and audio streaming technologies. We want to become your partner for all your hosting needs, as well as your one stop shop for radio products such as custom DJ drops and radio ID's.

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Comparison Of Analog And Digital Recording

This article compares the two ways in which sound is recorded and stored. Actual sound waves consist of continuous variations in air pressure. Representations of these signals can be recorded using either digital or analog techniques.

An analog recording is one where a property or characteristic of a physical recording medium is made to vary in a manner analogous to the variations in air pressure of the original sound. The air pressure variations are first converted (by a transducer such as a microphone) into an analog electrical signal in which either the instantaneous voltage or current is directly proportional to the instantaneous air pressure (or is a function of the pressure). The variations of the electrical signal, in turn, are converted to variations in the recording medium by a recording machine such as a tape recorder or record cutter—the variable property of the medium is modulated by the signal. Examples of properties that are modified are the magnetization of magnetic tape or the deviation (or displacement) of the groove of a gramophone disc from a smooth, flat spiral track.

A digital recording is produced by converting the physical properties of the original sound into a sequence of numbers, which can then be stored and read back for reproduction. Normally, the sound is transduced (as by a microphone) to an analog signal in the same way as for analog recording, and then the analog signal is digitized, or converted to a digital signal, through an analog-to-digital converter and then recorded onto a digital storage medium such as a compact disc or hard disk.

Two prominent differences in functionality are the bandwidth and the signal-to-noise ratio (S/N); however, both digital and analog systems have inherent strengths and weaknesses. The bandwidth of the digital system is determined, according to the Nyquist frequency, by the sample rate used. The bandwidth of an analog system is dependent on the physical capabilities of the analog circuits. The S/N of a digital system is first limited by the bit depth of the digitization process, but the electronic implementation of the digital audio circuit introduces additional noise. In an analog system, other natural analog noise sources exist, such as flicker noise and imperfections in the recording medium. Some functions of the two systems are also naturally exclusive to either one or the other, such as the ability for more transparent filtering algorithms in digital systems and the harmonic saturation of analog systems.

Overview of differences

It is a subject of debate whether analog audio is superior to digital audio or vice versa. The question is highly dependent on the quality of the systems (analog or digital) under review and other factors which are not necessarily related to sound quality. Arguments for analog systems include the absence of fundamental error mechanisms which are present in digital audio systems, including aliasing, quantization noise, and the absolute limitation of dynamic range. Advocates of digital point to the high levels of performance possible with digital audio, including excellent linearity in the audible band and low levels of noise and distortion.

Accurate, high quality sound reproduction is possible with both analog and digital systems. Excellent, expensive analog systems may outperform digital systems, and vice versa; in theory any system of either type may be surpassed by a better, more elaborate and costly system of the other type,[citation needed] but in general it tends to be less expensive to achieve any given standard of technical signal quality with a digital system, except when the standard is very low. One of the most limiting aspects of analog technology is the sensitivity of analog media to minor physical degradation; however, when the degradation is more pronounced, analog systems usually perform better, often still producing recognizable sound, while digital systems will usually fail completely, unable to play back anything from the medium (see digital cliff). The principal advantages that digital systems have are a very uniform source fidelity, inexpensive media duplication, and direct use of the digital 'signal' in today's popular portable storage and playback devices. Analog recordings, by comparison, require comparatively bulky, high-quality playback equipment to capture the signal from the media as accurately as digital.

Error correction

Early in the development of the Compact Disc, engineers realized that the perfection of the spiral of bits was critical to playback fidelity. A scratch the width of a human hair (100 micrometers) could corrupt several dozen bits, resulting in at best a pop, and far worse, a loss of synchronization of the clock and data, giving a long segment of noise until resynchronized. This was addressed by encoding the digital stream with a multi-tiered error-correction coding scheme which reduces CD capacity by about 20%, but makes it tolerant to hundreds of surface imperfections across the disk without loss of signal. In essence, "error correction" can be thought of as "using the mathematically encoded backup copies of the data that was corrupted." Not only does the CD use redundant data, but it also mixes up the bits in a predetermined way (see CIRC) so that a small flaw on the disc will affect fewer consecutive bits of the decoded signal and allow for more effective error correction using the available backup information.

Error correction allows digital formats to tolerate quite a bit more media deterioration than analog formats. That is not to say poorly produced digital media are immune to data loss. Laser rot was most troublesome to the Laser disc format, but also occurs to some pressed commercial CDs, and was caused in both cases by inadequate disc manufacture.[note 1] There can occasionally be difficulties related to the use of consumer recordable/rewritable compact discs. This may be due to poor-quality CD recorder drives, low-quality discs, or incorrect storage, as the information-bearing dye layer of most CD-recordable discs is at least slightly sensitive to UV light and will be slowly bleached out if exposed to any amount of it. Most digital recordings rely at least to some extent on computational encoding and decoding and so may become completely unplayable if not enough consecutive good data is available for the decoder to synchronize to the digital data stream, whereas any intact segment of any size of an analog recording is playable.

<u>Duplication</u>

Unlike analog duplication, digital copies are exact replicas, which can be duplicated indefinitely[note 2] without degradation. This made Digital rights management more of an issue in digital media than analog media. Digital systems often have the ability for the same medium to be used with arbitrarily high or low quality encoding methods and the number of channels or other content, unlike practically all analog systems which have mechanically pre-fixed speeds and channels. Most higher-end analog recording systems offer a few selectable recording speeds, but digital systems tend to offer much finer variation in the rate of media usage.

There are also several non-sound related advantages of digital systems that are practical. Digital systems that are computer-based make editing much easier through rapid random access, seeking, and scanning for non-linear editing. Most digital systems also allow non-audio data to be encoded into the digital stream, such as information about the artist, track titles, etc., which is often convenient.[note 3]

Noise and distortion

In the process of recording, storing and playing back the original analog sound wave (in the form of an electronic signal), it is unavoidable that some signal degradation will occur. This degradation is in the form of distortion and noise. Noise is unrelated in time to the original signal content, while distortion is in some way related in time to the original signal content.

<u>Noise performance</u>

For electronic audio signals, sources of noise include mechanical, electrical and thermal noise in the recording and playback cycle. The actual process of digital conversion will always add some noise, however, small in intensity; the bulk of this in a high-quality system is quantization noise, which cannot be theoretically avoided, but some will also be electrical, thermal, etc. noise from the analog-to-digital converted device.

The amount of noise that a piece of audio equipment adds to the original signal can be quantified. Mathematically, this can be expressed by means of the signal to noise ratio (SNR or S/N). Sometimes the maximum possible dynamic range of the system is quoted instead. In a digital system, the number of quantization levels, in binary systems determined by and typically stated in terms of the number of bits, will have a bearing on the level of noise and distortion added to that signal. Each additional quantization bit adds 6 dB in possible SNR, e.g. 24 x 6 = 144 dB for 24 bit quantization, 126 dB for 21-bit, and 120 dB for 20-bit.

The 16-bit digital system of Red Book audio CD has 216= 65,536 possible signal amplitudes, theoretically allowing for an SNR of 98 dB if undithered, however, the perceived dynamic range of 16-bit audio can be 120 dB or more with noise-shaped dither, taking advantage of the frequency response of the human ear.

With digital systems, the quality of reproduction depends on the analog-to-digital and digital-to-analog conversion steps, and does not depend on the quality of the recording medium, provided it is adequate to retain the digital values without error.

Analog systems

Consumer analog cassette tapes may have a dynamic range of 60 to 70 dB. Analog FM broadcasts rarely have a dynamic range exceeding 50 dB, though under excellent reception conditions the basic FM transmission system can achieve just over 80 dB.[citation needed] The dynamic range of a direct-cut vinyl record may surpass 70 dB. Analog studio master tapes using Dolby-A noise reduction can have a dynamic range of around 80 dB.

Rumble

This is a form of noise characteristic caused by imperfections in the bearings of turntables, the platter tends to have a slight amount of motion besides the desired rotation—the turntable surface also moves up-and-down and side-to-side slightly. This additional motion is added to the desired signal as noise, usually of very low frequencies, creating a "rumbling" sound during quiet passages. Very inexpensive turntables sometimes used ball bearings which are very likely to generate audible amounts of rumble. More expensive turntables tend to use massive sleeve bearings which are much less likely to generate offensive amounts of rumble. Increased turntable mass also tends to lead to reduced rumble. A good turntable should have rumble at least 60 dB below the specified output level from the pick-up.

Wow and flutter

This is caused by a change in frequency of an analog device and are the result of mechanical imperfections, with wow being a slower rate form of flutter. Wow, and flutter is most noticeable on signals which contain pure tones. For LP records, the quality of the turntable will have a large effect on the level of wow and flutter. A good turntable will have wow and flutter values of less than 0.05%, which is the speed variation from the mean value. Wow and flutter can also be present in the recording, as a result of the imperfect operation of the recorder.

Frequency response

Digital mechanisms

The frequency response of the standard for audio CDs is sufficiently wide to cover the entire normal audible range, which roughly extends from 20 Hz to 20 kHz. Commercial and industrial digital recorders record higher frequencies, while consumer systems inferior to the CD record a more restricted frequency range. Analog audio's frequency response is less flat than digital, but it can vary in the electronics.

For digital systems, the upper limit of the frequency response is determined by the sampling frequency. The choice of sample rate used in a digital system is based on the Nyquist-Shannon sampling theorem. This states that a sampled signal can be reproduced exactly as long as it is sampled at a frequency greater than twice the bandwidth of the signal. Therefore, a sampling rate of 40 kHz would be theoretically enough to capture all the information contained in a signal having frequency bandwidth up to 20 kHz. The sampling theorem assumes ideal filters, however, which cannot exist in reality, so practical sampling uses "guard bands" (higher than necessary sample rates) to reduce aliasing.

Analog mechanisms

High quality open-reel machines can extend from 10 Hz to above 20 kHz. The linearity of the response may be indicated by providing information on the level of the response relative to the reference frequency. For example, a system component may have a response given as 20 Hz to 20 kHz +/- 3 dB relative to 1 kHz. Some analog tape manufacturers specify frequency responses up to 20 kHz, but these measurements may have been made at lower signal levels. Compact cassettes may have a response extending up to 15 kHz at full (0 dB) recording level (Stark 1989). At lower levels usually -10 dB, cassettes typically rolls-off at around 20 kHz for most machines, due to the nature of the tape media caused by self-erasure (which worsens the linearity of the response).

The frequency response for a conventional LP player might be 20 Hz - 20 kHz +/- 3 dB. Unlike the audio CD, vinyl records and cassettes do not require a cut-off in response above 20 kHz. The low frequency response of vinyl records is restricted by rumble noise (described above). The high frequency response of vinyl depends on the cartridge. CD4 records contained frequencies up to 50 kHz, while some high-end turntable cartridges have frequency responses of 120 kHz while having flat frequency response over the audible band (e.g. 20 Hz to 15 kHz +/-0.3 dB). In addition, frequencies of up to 122 kHz have been experimentally cut on LP records.

In comparison, the CD system offers a frequency response of 20 Hz–20 kHz ±0.5 dB, with a superior dynamic range over the entire audible frequency spectrum.

With vinyl records, there will be some loss in fidelity on each playing of the disc. This is due to the wear of the stylus in contact with the record surface. A good quality stylus, matched with a correctly set up pick-up arm, should cause minimal surface wear. Magnetic tapes, both analog, and digital, wear from friction between the tape and the heads, guides, and other parts of the tape transport as the tape slides over them. The brown residue deposited on swabs during cleaning of a tape machine's tape path is particles of magnetic coating shed from tapes. Tapes can also suffer creasing, stretching, and frilling of the edges of the plastic tape base, particularly from low-quality or out-of-alignment tape decks. When a CD is played, there is no physical contact involved, and the data is read optically using a laser beam. Therefore, no such media deterioration takes place, and the CD will, with proper care, sound the same every time it is played (discounting aging of the player and CD itself); however, this is a benefit of the optical system, not of digital recording, and the Laser disc format enjoys the same non-contact benefit with analog optical signals. Recordable CDs slowly degrade with time, called disc rot, even if they are not played, and are stored properly. A new compact disc was release called M-DISC which is said to last 1000 years. These discs are recordable and have a layer developed from stone. They come in CD, DVD and Blu-ray formats and various storage sizes. They can be used for music movies and data for computers etc.

Aliasing

The technical difficulty arises with digital sampling in that all high frequency signal content above the Nyquist frequency must be removed before sampling, which, if not done, will result in these ultrasonic frequencies "folding over" into frequencies which are in the audible range, producing a kind of distortion called aliasing. The difficulty is that designing a brick-wall anti-aliasing filter, a filter which would precisely remove all frequency content exactly above or below a certain cutoff frequency, is impractical. Instead, a sample rate is usually chosen which is above the theoretical requirement. This solution is called oversampling, and allows a less aggressive and lower-cost anti-aliasing filter to be used.

Unlike digital audio systems, analog systems do not require filters for band limiting. These filters act to prevent aliasing distortions in digital equipment. Early digital systems may have suffered from several signal degradations related to the use of analog anti-aliasing filters, e.g., time dispersion, nonlinear distortion, the temperature dependence of filters, etc. (Hawksford 1991:8). Even with sophisticated anti-aliasing filters used in the recorder, it is still demanding for the player not to introduce more distortion.

Hawksford (1991:18) highlighted the advantages of digital converters that over sample. Using an oversampling design and a modulation scheme called sigma-delta modulation (SDM), analog anti-aliasing filters can effectively be replaced by a digital filter. This approach has several advantages. The digital filter can be made to have a near-ideal transfer function, with low in-band ripple, and no aging or thermal drift.

Higher sampling rates

CD quality audio is sampled at 44.1 kHz (Nyquist frequency = 22.05 kHz) and at 16 bits. Sampling the waveform at higher frequencies and allowing for a greater number of bits per sample allows noise and distortion to be reduced further. DAT can sample audio at up to 48 kHz, while DVD-Audio can be 96 or 192 kHz and up to 24 bits resolution. With any of these sampling rates, signal information is captured above what is considered to be the human hearing range.

Work done in 1981 by Muraoka et al. showed that music signals with frequency components above 20 kHz were only distinguished from those without by a few of the 176 test subjects (Kaoru & Shogo 2001). Later papers, however, by several different authors, have led to a greater discussion of the value of recording frequencies above 20 kHz. Such research led some to the belief that capturing these ultrasonic sounds could have some audible benefit. Audible differences were reported between recordings with and without ultrasonic responses. Dunn (1998) examined the performance of digital converters to see if these differences in performance could be explained. He did this by examining the band-limiting filters used in converters and looking for the artifacts they introduce.

A perceptual study by Nishiguchi et al. (2004) concluded that "no significant difference was found between sounds with and without very high frequency components among the sound stimuli and the subjects... however, [Nishiguchi et al] can still neither confirm nor deny the possibility that some subjects could discriminate between musical sounds with and without very high frequency components."

Additionally, in blind tests conducted by Bob Katz, recounted in his book Mastering Audio: The Art and the Science, he found that listening subjects could not discern any audible difference between sample rates with optimum A/D conversion and filter performance. He posits that the primary reason for any aural variation between sample rates is due largely to the poor performance of low-pass filtering before conversion, and not variance in ultrasonic bandwidth. These results suggest that the main benefit to using higher sample rates is that it pushes consequential phase distortion out of the audible range and that, under ideal conditions, higher sample rates may not be necessary.

Digital errors

Quantization

A signal is recorded digitally by an analog-to-digital converter, which measures the amplitude of an analog signal at regular intervals, which are specified by the sample rate, and then stores these sampled numbers in computer hardware. The fundamental problem with numbers on computers is that the range of values that can be represented is finite, which means that during sampling, the amplitude of the audio signal must be rounded. This process is called quantization, and these small errors in the measurements are manifested aurally as a form of low-level distortion.

Analog systems do not have discrete digital levels in which the signal is encoded. Consequently, the original signal can be preserved to an accuracy limited only by the intrinsic noise-floor and maximum signal level of the media and the playback equipment, i.e., the dynamic range of the system. This form of distortion, sometimes called granular or quantization distortion, has been pointed to as a fault of some digital systems and recordings (Knee & Hawksford 1995, Stuart n.d.:6). Knee & Hawksford (1995:3) drew attention to the deficiencies in some early digital recordings, where the digital release was said to be inferior to the analog version.

The range of possible values that can be represented numerically by a sample is defined by the number of binary digits used. This is called the resolution and is usually referred to as the bit depth in the context of PCM audio. The quantization noise level is directly determined by this number, decreasing exponentially as the resolution increases (or linearly in dB units), and with an adequate number of true bits of quantization, random noise from other sources will dominate and completely mask the quantization noise. The Redbook CD standard uses 16 bits, which keep the quantization noise 96 dB below maximum amplitude, far below a discernible level with almost any source material.

Dither as a solution

It is possible to make quantization noise more audibly benign by applying dither. To do this, a noise-like signal is added to the original signal before quantization. Dither makes the digital system behave as if it has an analog noise-floor. Optimal use of dither (triangular probability density function dither in PCM systems) has the effect of making the RMS quantization error independent of signal level (Dunn 2003:143), and allows signal information to be retained below the least significant bit of the digital system (Stuart n.d.:3).

Dither algorithms also commonly have the option to employ some noise shaping, which pushes the frequency response of the dither noise to areas that are less audible to human ears. This has no statistical benefit, but rather it raises the S/N of the audio that is apparent to the listener.

Proper application of dither combats quantization noise effectively, and is commonly applied during mastering before final bit depth reduction, and also at various stages of DSP.

Jitter

One aspect that may degrade the performance of a digital system is jitter. This is the phenomenon of variations in time from what should be the correct spacing of discrete samples according to the sample rate. This can be due to timing inaccuracies of the digital clock. Ideally, a digital clock should produce a timing pulse at exactly regular intervals. Other sources of jitter within digital electronic circuits are data-induced jitter, where one part of the digital stream affects a subsequent part as it flows through the system, and power supply induced jitter, where DC ripple on the power supply output rails causes irregularities in the timing of signals in circuits powered from those rails.

The accuracy of a digital system is dependent on the sampled amplitude values, but it is also dependent on the temporal regularity of these values. This temporal dependency is inherent to digital recording and playback and has no analog equivalent, though analog systems have their temporal distortion effects (pitch error and wow-and-flutter).

Periodic jitter produces modulation noise and can be thought of as being the equivalent of analog flutter (Rumsey & Watkinson 1995). Random jitter alters the noise floor of the digital system. The sensitivity of the converter to jitter depends on the design of the converter. It has been shown that a random jitter of 5 ns (nanoseconds) may be significant for 16 bit digital systems (Rumsey & Watkinson 1995). For a more detailed description of jitter theory, refer to Dunn (2003).

Jitter can degrade sound quality in digital audio systems. In 1998, Benjamin and Gannon researched the audibility of jitter using listening tests (Dunn 2003:34). They found that the lowest level of jitter to be audible was around 10 ns (RMS). This was a 17 kHz sine wave test signal. With music, no listeners found jitter audible at levels lower than 20 ns. A paper by Ashihara et al. (2005) attempted to determine the detection thresholds for random jitter in music signals. Their method involved ABX listening tests. When discussing their results, the authors of the paper commented that:

'So far, actual jitter in consumer products seems to be too small to be detected at least for reproduction of music signals. It is not clear, however, if detection thresholds obtained in the present study would represent the limit of auditory resolution or it would be limited by the resolution of equipment. Distortions due to very small jitter may be smaller than distortions due to non-linear characteristics of loudspeakers. Ashihara and Kiryu evaluated linearity of loudspeaker and headphones. According to their observation, headphones seem to be more preferable to produce sufficient sound pressure at the ear drums with smaller distortions than loudspeakers.'

On the Internet-based hi-fi website, TNT Audio, Pozzoli (2005) describes some audible effects of jitter. His assessment appears to run contrary to the earlier papers mentioned:

'In my personal experience, and I would dare say in common understanding, there is a huge difference between the sound of low and high jitter systems. When the jitter amount is very high, as in very low cost CD players (2ns), the result is somewhat similar to wow and flutter, the well known problem that affected typically compact cassettes (and in a far less evident way turntables) and was caused by the non perfectly constant speed of the tape: the effect is similar, but here the variations have a far higher frequency and for this reasons are less easy to perceive but equally annoying. Very often in these cases the rhythmic message, the pace of the most complicated musical plots is partially or completely lost, music is dull, scarcely involving and apparently meaningless, it does not make any sense. Apart from harshness, the typical "digital" sound, in a word... In lower amounts, the effect above is difficult to perceive, but jitter is still able to cause problems: reduction of the sound stage width and depth, lack of focus, sometimes a veil on the music. These effects are however far more difficult to trace back to jitter, as can be caused by many other factors.'

Dynamic range

The dynamic range of an audio system is a measure of the difference between the smallest and largest amplitude values that can be represented in a medium. Digital and analog differ in both the methods of transfer and storage, as well as the behavior exhibited by the systems due to these methods.

Overload conditions

There are some differences in the behavior of analog and digital systems when high-level signals are present, where there is the possibility that such signals could push the system into overload. With high-level signals, analog magnetic tape approaches saturation, and high-frequency response drops in proportion to low frequency response. While undesirable, the audible effect of this can be reasonably unobjectionable (Elsea 1996). In contrast, PCM digital recorders show non-benign behavior in overload (Dunn 2003:65); samples that exceed the peak quantization level are simply truncated, clipping the waveform squarely, which introduces distortion in the form of large quantities of higher-frequency harmonics. The 'softness' of analog tape clipping allows a usable dynamic range that can exceed that of some PCM digital recorders. (PCM, or pulse code modulation, is the coding scheme used in Compact Disc, DAT, PC sound cards, and many studio recording systems.)

In principle, PCM digital systems have the lowest level of nonlinear distortion at full signal amplitude. The opposite is usually true of analog systems, where distortion tends to increase at high signal levels. A study by Manson (1980) considered the requirements of a digital audio system for high-quality broadcasting. It concluded that a 16-bit system would be sufficient, but noted the small reserve the system provided in ordinary operating conditions. For this reason, it was suggested that a fast-acting signal limiter or 'soft clipper' be used to prevent the system from becoming overloaded (Manson 1980:8).

With many recordings, high-level distortions at signal peaks may be audibly masked by the original signal; thus large amounts of distortion may be acceptable at peak signal levels. The difference between analog and digital systems is the form of high-level signal error. Some early analog-to-digital converters displayed non-benign behavior when in overload, where the overloading signals were 'wrapped' from positive to negative full-scale. Modern converter designs based on sigma-delta modulation may become unstable in overload conditions. It is usually a design goal of digital systems to limit high-level signals to prevent overload (Dunn 2003:65). To prevent overload, a modern digital system may compress input signals so that digital full-scale cannot be reached (Jones et al. 2003:4).

Resolution

The dynamic range of digital audio systems can exceed that of analog audio systems. Typically, a 16 bit analog-to-digital converter may have a dynamic range of between 90 and 95 dB (Metzler 2005:132), whereas the signal-to-noise ratio (roughly the equivalent of dynamic range, noting the absence of quantization noise but presence of tape hiss) of a professional reel-to-reel 1/4 inch tape recorder would be between 60 and 70 dB at the recorder's rated output (Metzler 2005:111).

The benefits of using digital recorders with greater than 16-bit accuracy can be applied to the 16 bits of audio CD. Stuart (n.d.:3) stresses that with the correct dither, the resolution of a digital system is theoretically infinite and that it is possible, for example, to resolve sounds at -110 dB (below digital full-scale) in a well-designed 16-bit channel.

Compression

Despite the lower dynamic range and signal-to-noise ratios a vinyl or tape record can achieve in theory (60-80 dB versus 90-96 dB for CD recordings), vinyl records may still be preferred for their greater dynamic range in practice because of aggressive, dynamic range compression used for CD audio material (see Loudness war), a practice relatively uncommon for vinyl mastering.

Signal processing

After initial recording, it is common for the audio signal to be altered in some way, such as with the use of compression, equalization, delays and reverb. With analog, this comes in the form of outboard hardware components, and with digital, the same is accomplished with plug-ins that are utilized in the user's DAW.

A comparison of analog and digital filtering shows technical advantages to both methods, and there are several points that are relevant to the recording process.

Analog hardware

Many analog units possess unique characteristics that are desirable. Common elements are band shapes and phase response of equalizers and response times of compressors. These traits can be difficult to reproduce digitally because they are due to electrical components which function differently from the algorithmic calculations used on a computer.

When altering a signal with a filter, the outputted signal may differ in time from the signal at the input, which is called a change in phase. Many equalizers exhibit this behavior, with the amount of phase shift differing in some pattern, and centered around the band that is being adjusted. This phase distortion can create the perception of a "ringing" sound around the filter band or other coloration. Although this effect alters the signal in a way other than a strict change in frequency response, this coloration can sometimes have a positive effect on the perception of the sound of the audio signal.

Digital filters

Digital filters can be made to objectively perform better than analog components because the variables involved can be precisely specified in the calculations.

One prime example is the invention of the linear phase equalizer, which has an inherent phase shift that is homogeneous across the frequency spectrum. Digital delays can also be perfectly exact, provided the delay time is some multiple of the time between samples, and so can the summing of a multi-track recording, as the sample values are merely added together.

A practical advantage of digital processing is the most convenient recall of settings. Plug-in parameters can be stored on the computer hard disk, whereas parameter details on an analog unit must be written down or otherwise recorded if the unit needs to be reused. This can be cumbersome when entire mixes must be recalled manually using an analog console and outboard gear. When working digitally, all parameters can simply be stored in a DAW project file and recalled instantly. Most modern professional DAWs also process plug-ins in real time, which means that processing can be largely non-destructive until final mix-down.

Analog modeling

Many plug-ins exist now that incorporate some analog modeling. There are some engineers that endorse them and feel that they compare equally in sound to the analog processes that they imitate. Digital models also carry some benefits over their analog counterparts, such as the ability to remove noise from the algorithms and add modifications to make the parameters more flexible. On the other hand, other engineers also feel that the modeling is still inferior to the genuine outboard components and still prefer to mix "outside the box".

Sound quality

Subjective evaluation

After initial recording, it is common for the audio signal to be altered in some way, such as with the use of compression, equalization, delays and reverb. With analog, this comes in the form of outboard hardware components, and with digital, the same is accomplished with plug-ins that are utilized in the user's DAW.

A comparison of analog and digital filtering shows technical advantages to both methods, and there are several points that are relevant to the recording process.

Analog hardware

Many analog units possess unique characteristics that are desirable. Common elements are band shapes and phase response of equalizers and response times of compressors. These traits can be difficult to reproduce digitally because they are due to electrical components which function differently from the algorithmic calculations used on a computer.

When altering a signal with a filter, the outputted signal may differ in time from the signal at the input, which is called a change in phase. Many equalizers exhibit this behavior, with the amount of phase shift differing in some pattern, and centered around the band that is being adjusted. This phase distortion can create the perception of a "ringing" sound around the filter band or other coloration. Although this effect alters the signal in a way other than a strict change in frequency response, this coloration can sometimes have a positive effect on the perception of the sound of the audio signal.

Digital filters

Digital filters can be made to objectively perform better than analog components because the variables involved can be precisely specified in the calculations.

One prime example is the invention of the linear phase equalizer, which has an inherent phase shift that is homogeneous across the frequency spectrum. Digital delays can also be perfectly exact, provided the delay time is some multiple of the time between samples, and so can the summing of a multi-track recording, as the sample values are merely added together.

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Was it ever entirely analog or digital?

Complicating the discussion is that recording professionals often mix and match analog and digital techniques in the process of producing a recording. Analog signals can be subjected to digital signal processing or effects, and inversely digital signals are converted back to analog in equipment that can include analog steps such as vacuum tube amplification.

For modern recordings, the controversy between analog recording and digital recording is becoming moot. No matter what format the user uses, the recording probably was digital at several stages in its life. In the case of video recordings it is moot for one other reason; whether the format is analog or digital, digital signal processing is likely to have been used in some stages of its life, such as digital timebase correction on playback.

An additional complication arises when discussing human perception when comparing analog and digital audio in that the human ear itself, is an analog-digital hybrid. The human hearing mechanism begins with the tympanic membrane transferring vibrational motion through the middle ear's mechanical system—three bones (malleus, incus, and stapes)—into the cochlea where hair-like nerve cells convert the vibrational motion stimulus into nerve impulses. Auditory nerve impulses are discrete signaling events which cause synapses to release neurotransmitters to communicate to other neurons (see here.) The all-or-none quality of the impulse can lead to a misconception that neural signaling is somehow 'digital' in nature, but in fact, the timing and rate of these signaling events are not clocked or quantized in any way. Thus the transformation of the acoustic wave is not a process of sampling, in the sense of the word as it applies to digital audio. Instead, it is a transformation from one analog domain to another, and this transformation is further processed by the neurons to which the signaling is connected. The brain then processes the incoming information and perceptually reconstructs the original analog input to the ear canal.

It is also worth noting two issues that affect the perception of sound playback. The first is the human ear dynamic range which for practical and hearing safety reasons might be regarded as 120 decibels, from barely audible sound received by the ear situated within an otherwise silent environment, to the threshold of pain or onset of damage to the ear's delicate mechanism. The other critical issue is manifestly more complex; the presence and nature of background noise in any listening environment. Background noise subtracts useful hearing dynamic range, in any number of ways that depend on the nature of the noise from the listening environment: noise spectral content, noise coherence or periodicity, angular aspects such as localization of noise sources with respect to localization of playback system sources and so on.

Hybrid systems

While the words analog audio usually imply that the sound is described using a continuous time/continuous amplitudes approach in both the media and the reproduction/recording systems, and the words digital audio imply a discrete time/discrete amplitudes approach, there are methods of encoding audio that fall somewhere between the two, e.g. continuous time/discrete levels and discrete time/continuous levels.

While not as common as "pure analog" or "pure digital" methods, these situations do occur in practice. Indeed, all analog systems show discrete (quantized) behavior at the microscopic scale, and asynchronously operated class-D amplifiers even consciously incorporate continuous time, discrete amplitude designs. Continuous amplitude, discrete time systems have also been used in many early analog-to-digital converters, in the form of sample-and-hold circuits. The boundary is further blurred by digital systems which statistically aim at analog-like behavior, most often by utilizing stochastic dithering and noise shaping techniques. While vinyl records and common compact cassettes are analog media and use quasi-linear physical encoding methods (e.g. spiral groove depth, tape magnetic field strength) without noticeable quantization or aliasing, there are analog non-linear systems that exhibit effects similar to those encountered on digital ones, such as aliasing and "hard" dynamic floors (e.g. frequency modulated hi-fi audio on videotapes, PWM encoded signals).

Although those "hybrid" techniques are usually more common in telecommunications systems than in consumer audio, their existence alone blurs the distinctive line between certain digital and analog systems, at least for what regards some of their alleged advantages or disadvantages.

There are many benefits to using digital recording over analog recording because “numbers are more easily manipulated than are grooves on a record or magnetized particles on a tape” (Rudolph & Leonard, 2001, p. 3). Because numerical coding represents the sound waves perfectly, the sound can be played back without background noise.

https://en.wikipedia.org/wiki/Comparison_of_analog_and_digital_recording

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Room Modes

These are the collection of resonances that exist in a room when the room is excited by an acoustic source such as a loudspeaker. Most rooms have their fundamental resonances in the 20 Hz to 200 Hz region, each frequency being related to one or more of the room's dimensions or a divisor thereof. These resonances affect the low-frequency low-mid-frequency response of a sound system in the room and are one of the biggest obstacles to accurate sound reproduction.

The mechanism of the room's resonances

The input of acoustic energy to the room at the modal frequencies and multiples thereof causes standing waves. The nodes and antinodes of these standing waves result in the loudness of the particular resonant frequency being different at different locations of the room. These standing waves can be considered a temporary storage of acoustic energy as they take a finite time to build up and a finite time to dissipate once the sound energy source has been removed.

The mechanism of the room's resonances

A room with generally hard surfaces will exhibit, high Q, sharply tuned resonances. Absorbent material can be added to the room to damp such resonances which work by more quickly dissipating the stored acoustic energy.

To be effective, a layer of porous, absorbent material has to be of the order of a quarter-wavelength thick if placed on a wall, which at low frequencies with their long wavelengths requires very thick absorbers. Absorption occurs through friction of the air motion against individual fibers, with kinetic energy converted to heat, and so the material must be of just the right 'density' in terms of fiber packing. Too loose, and sound will pass through, but too firm and reflection will occur. Technically it is a matter of impedance matching between air motion and the individual fibers. Glass fiber, as used for thermal insulation, is very effective, but needs to be very thick (perhaps four to six inches) if the result is not to be a room that sounds unnaturally 'dead' at high frequencies but remains 'boomy' at lower frequencies, so that it provides absorption across a broad range of frequencies. Curtains and carpets are only effective at high frequencies. (This typically occurs at 5 kHz and above)

As a rule of thumb, sound travels at one foot per millisecond (344m/s), so the wavelength of notes at 1 kHz is about a foot (344mm), and at 10 kHz about an inch (34mm). Even six inches of glass fiber has little effect at 100 Hz, where a quarter wavelength is over 2 feet (860mm). Adding absorbent material has virtually no effect in the lower bass region in the 20–50 Hz region, though it can bring about a great improvement in the upper bass region above 100 Hz.

Open apertures, dispersion cylinders (large diameter and usually wall height), carefully sized and placed panels, and irregular room shapes are another way of either absorbing energy or breaking up resonant modes. For absorption, as with large foam wedges seen in anechoic chambers, the loss occurs ultimately through turbulence, as colliding air molecules convert some of their kinetic energy into heat. Damped panels, typically consisting of sheets of hardboard between glass fiber battens, have been used to absorb bass, by allowing movement of the surface panel and energy absorption by friction with the fiber battens.

If a room is being constructed, it is possible to choose room dimensions for which its resonances are less audible. This is done by ensuring that multiple room resonances are not at similar frequencies. For example, a cubic room would exhibit three resonances at the same frequency.

Equalization of the sound system to compensate for the uneven frequency response caused by room resonances is of very limited use as the equalization only works for one specific listening position and will cause the response to be worse in other listening positions. Also, large bass boosts by sound system EQ can severely reduce the headroom in the sound system itself. Some vendors are currently providing elaborate room tuning equipment which requires precision microphones, extensive data collection, and uses computerized electronic filtering to implement the necessary compensation for the rooms modes. There is some controversy about the relative worth of the improvement in ordinary rooms, given the very high cost of these systems.

Concert halls

Very large rooms like concert halls or large television studios have fundamental resonances which are much lower in frequency than small rooms. This means the closely spaced harmonic resonances are likely to lie in the low frequency region, and thus the response tends to be more uniform.

https://en.wikipedia.org/wiki/Room_modes

Architectural Acoustics

This is the science and engineering of achieving a good sound within a building and is a branch of acoustical engineering. The first application of modern scientific methods to architectural acoustics was carried out by Wallace Sabine in the Fogg Museum lecture room who then applied his new found knowledge to the design of Symphony Hall, Boston.

Architectural acoustics can be about achieving good speech intelligibility in a theater, restaurant or railway station, enhancing the quality of music in a concert hall or recording studio, or suppressing noise to make offices and homes more productive and pleasant places to work and live in. Architectural acoustic design is usually done by acoustic consultants.

Building skin envelope

This science analyzes noise transmission from building exterior envelope to an interior and vice versa. The main noise paths are roofs, eaves, walls, windows, door, and penetrations. Sufficient control ensures space functionality and is often required based on building use and local municipal codes. An example would be providing a suitable design for a home which is to be constructed close to a high volume roadway, or under the flight path of a major airport, or of the airport itself.

Inter-space noise control

This is the science of controlling a room's surfaces based on sound absorbing and reflecting properties. Excessive reverberation time, which can be calculated, can lead to poor speech intelligibility.

Sound reflections create standing waves that produce natural resonances that can be heard as a pleasant sensation or an annoying one. Reflective surfaces can be angled and coordinated to provide good coverage of sound for a listener in a concert hall or music recital space. To illustrate this concept consider the difference between a modern large office meeting room or lecture theater and a traditional classroom with all hard surfaces.

Interior building surfaces can be constructed of many different materials and finishes. Ideal acoustical panels are those without a face or finish material that interferes with the acoustical infill or substrate. Fabric covered panels are one way to heighten acoustical absorption. Perforated metal also shows sound absorbing qualities. Finish Material is used to cover over the acoustical substrate. Mineral fiber board, or Micore, is a commonly used acoustical substrate. Finish materials often consist of fabric, wood or acoustical tile. Fabric can be wrapped around substrates to create what is referred to as a "prefabricated panel" and often provides good noise absorption if laid onto a wall.

Prefabricated panels are limited to the size of the substrate ranging from 2 by 4 feet (0.61 m × 1.22 m) to 4 by 10 feet (1.2 m × 3.0 m). Fabric retained in a wall-mounted perimeter track system, is referred to as "on-site acoustical wall panels". These are constructed by framing the perimeter track into shape, infilling the acoustical substrate and then stretching and tucking the fabric into the perimeter frame system. On-site wall panels can be constructed to accommodate door frames, baseboard, or any other intrusion. Large panels (generally, greater than 50 square feet (4.6 m2)) can be created on walls and ceilings with this method. Wood finishes can consist of punched or routed slots and provide a natural look to the interior space, although acoustical absorption may not be great.

There are three ways to improve workplace acoustics and solve workplace sound problems – the ABCs.

A = Absorb (via drapes, carpets, ceiling tiles, etc.)
B = Block (via panels, walls, floors, ceilings, and layout)
C = Cover-up (via sound masking)

Mechanical equipment noise

Building services noise control is the science of controlling noise produced by:

- ACMV (air conditioning and mechanical ventilation) systems in buildings, termed HVAC in North America

- Elevators

- Electrical generators positioned within or attached to a building

- Any other building service infrastructure component that emits sound.

Inadequate control may lead to elevated sound levels within the space which can be annoying and reduce speech intelligibility. Typical improvements are vibration isolation of mechanical equipment and sound traps in ductwork. Sound masking can also be created by adjusting HVAC noise to a predetermined level.

https://en.wikipedia.org/wiki/Architectural_acoustics