Audio Engineering

Abagail Morales

First Edition, 2011

ISBN 978-93-81157-03-9

Published by:
The English Press
4735/22 Prakashdeep Bldg,
Ansari Road, Darya Ganj,
Delhi - 110002
Email: info@wtbooks.com

Table of Contents

Chapter- 1

Introduction to Audio Engineering

Audio engineering is a skilled trade that deals with the use of machinery and equipment for the recording, mixing and reproduction of sounds. The field draws on many artistic and vocational areas, including electronics, acoustics, psychoacoustics, and music. An audio engineer is proficient with different types of recording media, such as analog tape, digital multitrack recorders and workstations, and computer knowledge. With the advent of the digital age, it is becoming more and more important for the audio engineer to be versed in the understanding of software and hardware integration from synchronization to analog to digital transfers.

Audio engineering concerns the creative and practical aspects of sounds and music, in contrast with the formal engineering discipline known as acoustical engineering. Producer, engineer, mixer Phil Ek has described audio engineering as the "physical recording of any project—the placing of microphones, the turning of pre-amp knobs, the setting of levels—and the producer is the guy who directs that process." Many recording engineers also invented new technology, equipment and techniques, to enhance the process and art.

Lexical dispute

The expressions "audio engineer" and "sound engineer" are ambiguous. Such terms can refer to a person working in sound and music production, as well as to an engineer with a degree who designs professional equipment for these tasks.

Individuals who design acoustical simulations of rooms, shaping algorithms for digital signal processing and computer music problems, perform institutional research on sound, and other advanced fields of audio engineering are most often graduates of an accredited college or university, or have passed a difficult civil qualification test.

Certain jurisdictions specifically prohibit the use of the title engineer to any individual not a registered member of the local professional engineering body, responsible for regulating ethics and the safety of the public with respect to the engineering profession, which often may not include audio engineers. In such situations they are formally referred to as audio technicians.

Other languages, such as German and Italian, have different words to refer to these activities. For instance, in German, the *Tontechniker* (audio technician) is the one who operates the audio equipment and the *Tonmeister* (sound master) is a person who creates recordings or broadcasts of music who is both deeply musically trained (in 'classical' and non-classical genres) and who also has a detailed theoretical and practical knowledge of virtually all aspects of sound, whereas the *Toningenieur* (audio engineer) is the one who designs, builds and repairs it.

Practitioners

An engineer at an audio console.

An audio engineer is someone with experience and training in the production and manipulation of sound through mechanical (analog) or digital means. As a professional

title, this person is sometimes designated as a sound engineer or recording engineer instead. A person with one of these titles is commonly listed in the credits of many commercial music recordings (as well as in other productions that include sound, such as movies).

Audio engineers are generally familiar with the design, installation, and/or operation of sound recording, sound reinforcement, or sound broadcasting equipment, including large and small format consoles. In the recording studio environment, the audio engineer records, edits, manipulates, mixes, and/or masters sound by technical means in order to realize an artist's or record producer's creative vision. While usually associated with music production, an audio engineer deals with sound for a wide range of applications, including post-production for video and film, live sound reinforcement, advertising, multimedia, and broadcasting. When referring to video games, an audio engineer may also be a computer programmer.

In larger productions, an audio engineer is responsible for the technical aspects of a sound recording or other audio production, and works together with a record producer or director, although the engineer's role may also be integrated with that of the producer. In smaller productions and studios the sound engineer and producer is often one and the same person.

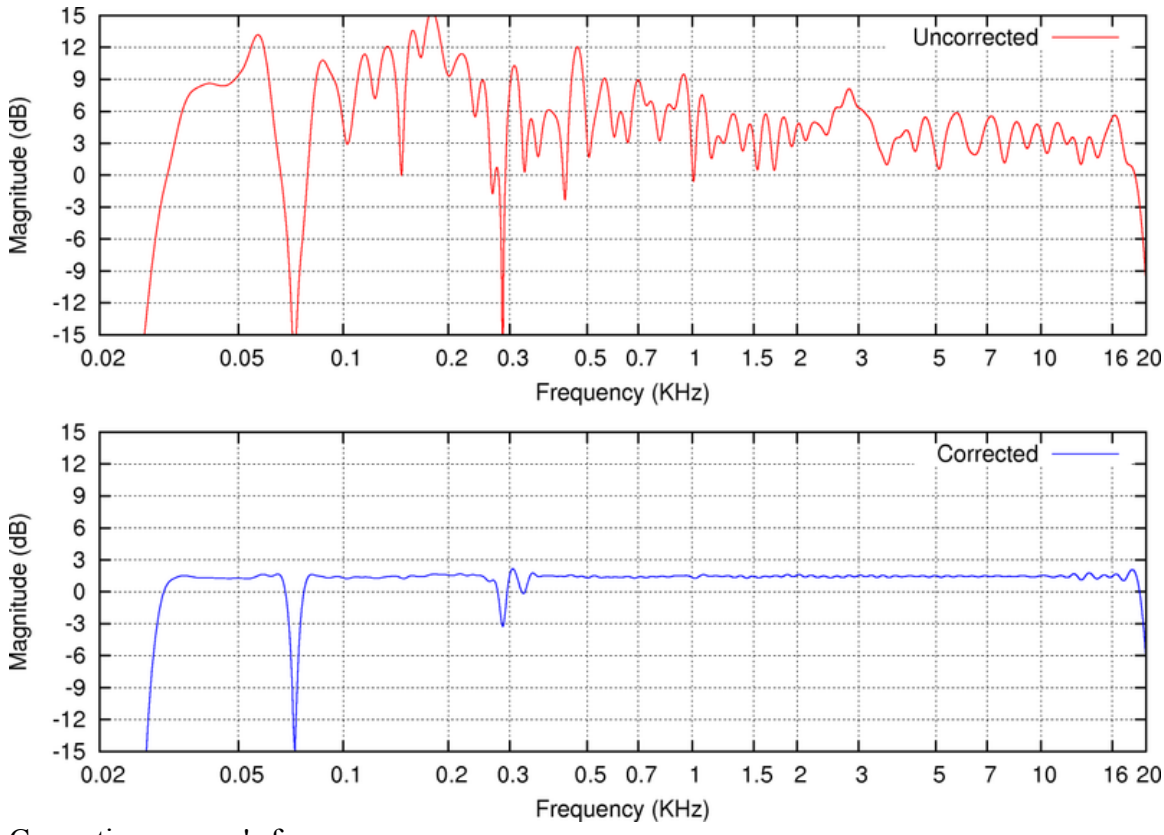
In typical sound reinforcement applications, audio engineers often assume the role of producer, making artistic and technical decisions, and sometimes even scheduling and budget decisions.

Different professional branches

There are four distinct steps to commercial production of a recording: Recording, editing, mixing, and mastering. Typically, each is performed by a sound engineer who specializes only in that part of production.

- Studio engineer – an engineer working within a studio facility, either with a producer or independently
- Recording engineer – engineer who records sound.
- Assistant engineer – often employed in larger studios, allowing them to train to become full-time engineers. They often assist full-time engineers with microphone setups, session breakdowns and in some cases, rough mixes.
- Mixing engineer – a person who creates mixes of multi-track recordings. It is common for a commercial record to be recorded at one studio and later mixed by different engineers in other studios.
- Mastering engineer – typically the person who mixes the final stereo tracks (or sometimes just a few tracks or stems) that the mix engineer produces. The mastering engineer makes any final adjustments to the overall sound of the record in the final step before commercial duplication. Mastering engineers use principles of equalization and compression to affect the coloration of the sound.
- Game audio designer engineer – deals with sound aspects of game development.

- Live sound engineer – a person dealing with live sound reinforcement. This usually includes planning and installation of speakers, cabling and equipment and mixing sound during the show. This may or may not include running the foldback sound. A live/sound reinforcement engineer hears musical material and tries to correlate that sonic experience with system performance.
- Foldback or Monitor engineer – a person running foldback sound during a live event. The term "foldback" is outdated and refers to the practice of folding back audio signals from the FOH (Front of House) mixing console to the stage in order for musicians to hear themselves while performing. Monitor engineers usually have a separate audio system from the FOH engineer and manipulate audio signals independently from what the audience hears, in order to satisfy the requirements of each performer on stage. In-ear systems, digital and analog mixing consoles, and a variety of speaker enclosures are typically used by monitor engineers. In addition most monitor engineers must be familiar with wireless or RF (radio-frequency) equipment and must interface personally with the artist(s) during each performance.
- Systems engineer – responsible for the design setup of modern PA systems which are often very complex. A systems engineer is usually also referred to as a "crew chief" on tour and is responsible for the performance and day-to-day job requirements of the audio crew as a whole along with the FOH audio system.
- Audio post engineer – a person who edits and mixes audio for film and/or television.



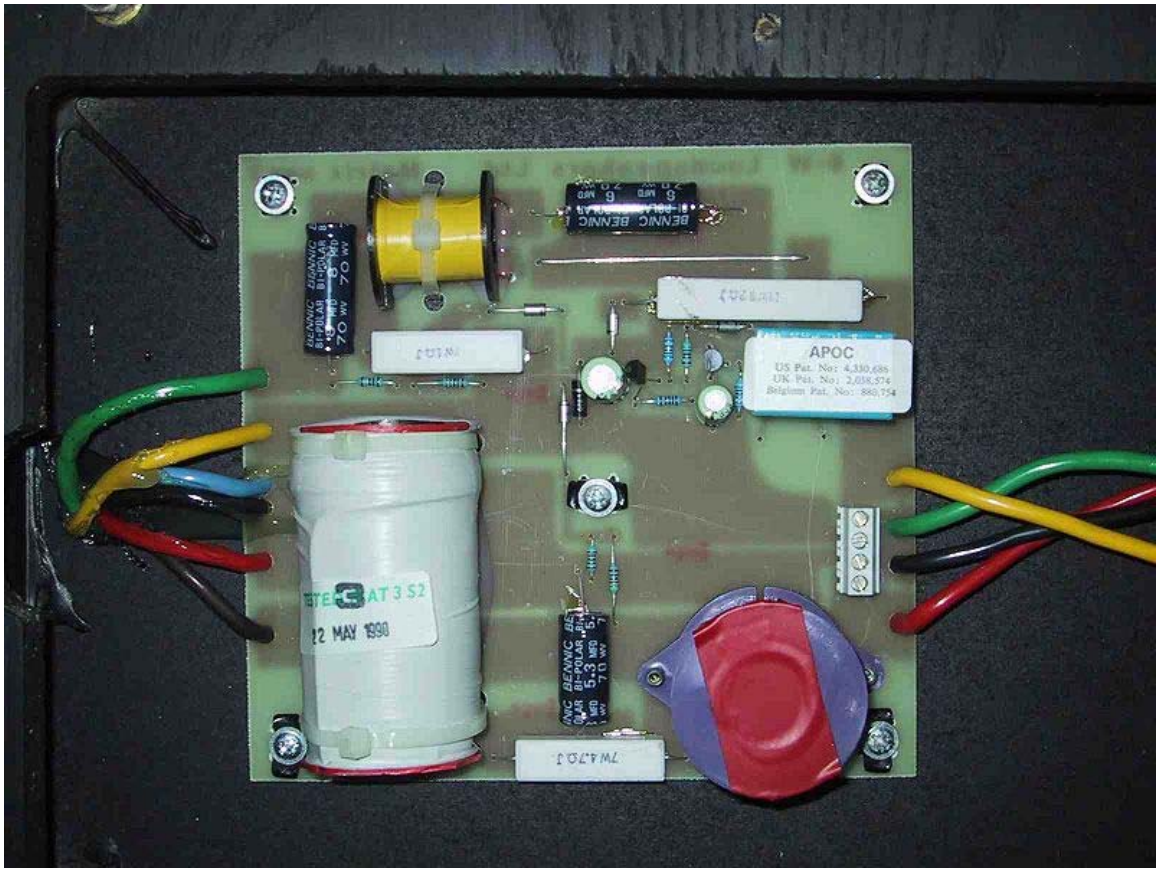
Correcting a room's frequency response.

Education

Audio engineers come from backgrounds such as fine arts, broadcasting, music or electronics. Many colleges and accredited institutions around the world offer degrees in audio engineering, such as a BS in audio production. The University of Miami's Frost School of Music was the first university in the United States to offer a four-year Bachelor of Music degree in Music Engineering Technology. In the last 25 years, some contemporary music schools have initiated audio engineering programs, usually awarding a Bachelor of Music degree to graduates. Additionally, a number of audio engineers are autodidacts with no formal training.

Chapter- 2

Audio Crossover



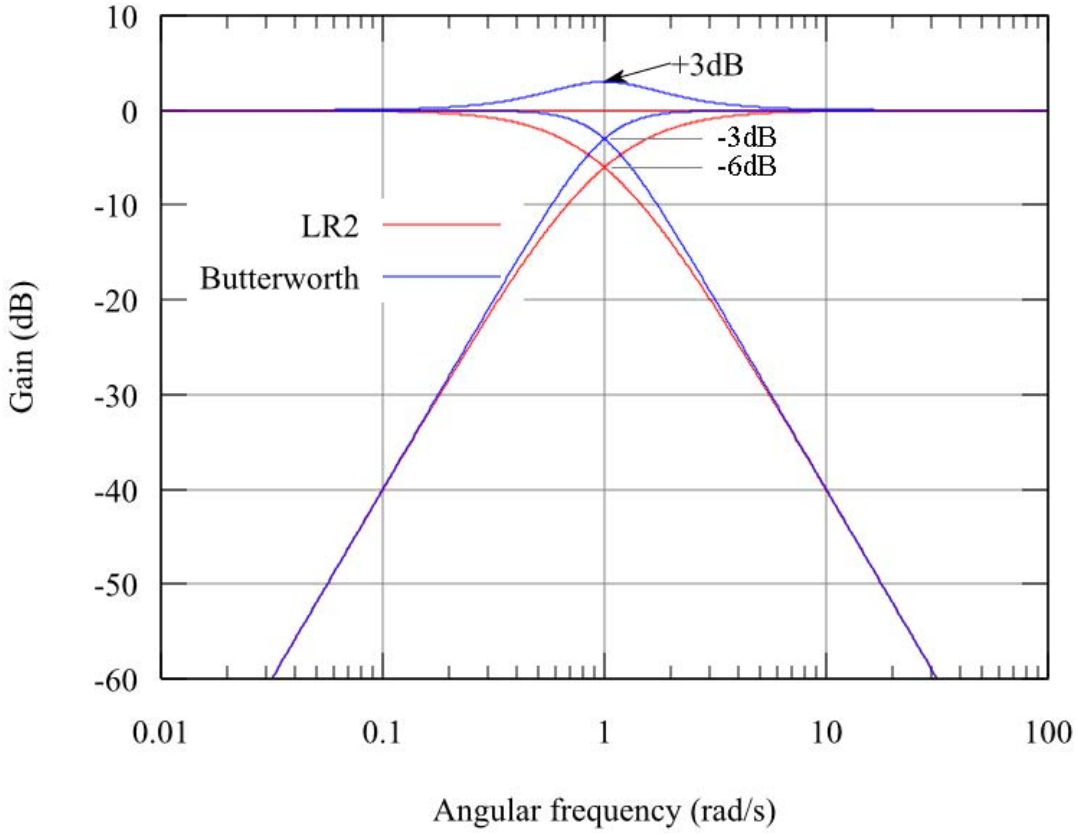
A passive 2-way crossover designed to operate at loudspeaker voltages

Audio crossovers are a class of electronic filter used in audio applications. Most individual loudspeaker drivers are incapable of covering the entire audio spectrum from low frequencies to high frequencies with acceptable relative volume and lack of distortion so most hi-fi speaker systems use a combination of multiple loudspeakers or drivers, each catering to a different frequency band. Crossovers split the audio signal into

separate frequency bands that can be separately routed to loudspeakers optimized for those bands.

Crossovers also enable multiband processing and multiple amplification where the audio signal is split into bands that are adjusted (equalized, compressed, echoed, etc.) separately before they are mixed together again. Some examples are: multiband dynamics (compression, limiting, de-essing), multiband distortion, bass enhancement, high frequency exciters, and noise reduction (for example: Dolby A noise reduction).

Overview



Comparison of the magnitude response of 2 pole Butterworth and Linkwitz-Riley crossover filters. The summed output of the Butterworth filters has a +3dB peak at the crossover frequency.

The definition of an ideal audio crossover changes relative to the task at hand. If the separate bands are to be mixed back together again (as in multiband processing), then the ideal audio crossover would split the incoming audio signal into separate bands that do not overlap or interact and which result in an output signal unchanged in frequency, relative levels, and phase response. This ideal performance can only be approximated. How to implement the best approximation is a matter of lively debate. On the other hand,

if the audio crossover separates the audio bands in a loudspeaker, there is no requirement for mathematically ideal characteristics within the crossover itself, as the frequency and phase response of the loudspeaker drivers within their mountings will eclipse the results. Satisfactory output of the complete system comprising the audio crossover *and* the loudspeaker drivers in their enclosure(s) is the design goal. Such a goal is often achieved using non-ideal, asymmetric crossover filter characteristics.

Many different crossover types are used in audio, but they generally belong to one of the following classes.

Classification

Classification based on the number of filter sections

In loudspeaker specifications, one often sees a speaker classified as an "N-way" speaker. N is a positive whole number greater than 1, and it indicates the number of filter sections. A 2-way crossover consists of a low-pass and a high-pass filter. A 3-way crossover is constructed as a combination of low-pass, band-pass and high-pass filters (LPF, BPF and HPF respectively). The BPF section is in turn a combination of HPF and LPF sections. 4 (or more) way crossovers are not very common in speaker design, primarily due to the complexity involved, which is not generally justified by better acoustic performance.

An extra HPF section may be present in an "N-way" loudspeaker crossover to protect the lowest-frequency driver from frequencies lower than it can safely handle. Such a crossover would then have a bandpass filter for the lowest-frequency driver. Similarly, the highest-frequency driver may have a protective LPF section to prevent high frequency damage, though this is far less common.

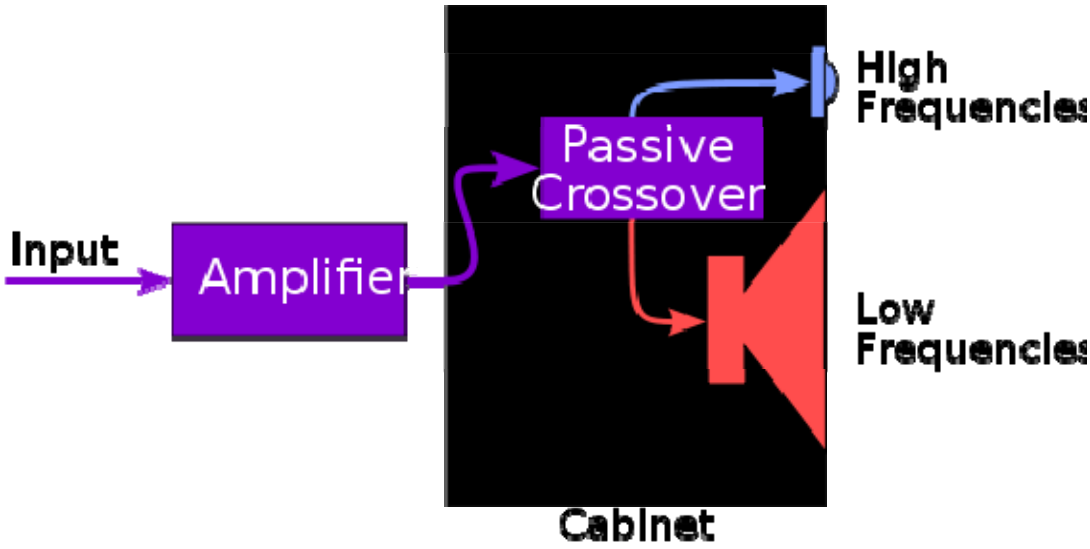
Recently, a number of manufacturers have begun using what is often called "N.5-way" crossover techniques for stereo loudspeaker crossovers. This usually indicates the addition of a second woofer that plays the same bass range as the main woofer but rolls off far before the main woofer does.

Remark: Filter sections mentioned here is not to be confused with the individual 2-pole filter sections that a higher order filter consists of.

Classification based on components

Crossovers can also be classified based on the design approach; by the type of components used.

Passive



A passive crossover

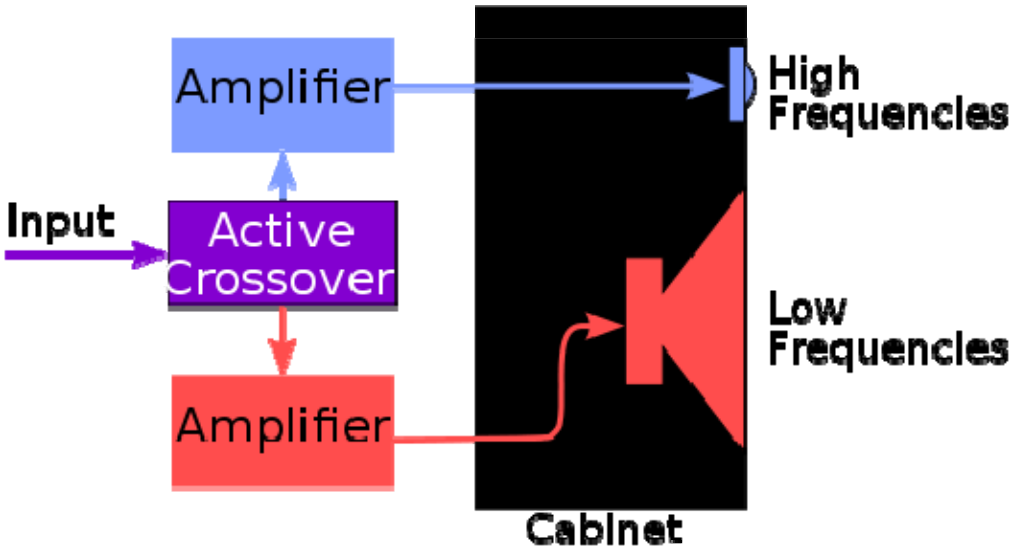
A passive crossover is made entirely of passive components, arranged most commonly in a Cauer topology to achieve a Butterworth filter. Passive filters use non-reactive resistors combined with reactive components such as capacitors and inductors. Very high performance passive crossovers are likely to be more expensive than active crossovers since individual components capable of good performance at the high currents and voltages at which speaker systems are driven are hard to make, and expensive. Polypropylene, metalized polyester foil, and paper-electrolytic capacitors are common. Inductors may have air cores, powdered metal cores, ferrite cores, or laminated silicon steel cores, and most are wound with enamelled copper wire. Some passive networks include devices such as fuses, PTC devices, bulbs or circuit breakers to protect the loudspeaker drivers from accidental overpowering. Modern passive crossovers increasingly incorporate equalization networks (e.g., Zobel networks) that compensate for the changes in impedance with frequency inherent in virtually all loudspeakers. The issue is complex, as part of the change in impedance is due to acoustic loading changes across a driver's passband.

On the negative side, passive networks may be bulky and cause power loss. They are not only frequency specific, but also impedance specific. This prevents interchangeability with speaker systems of different impedances. Ideal crossover filters, including impedance compensation and equalization networks, can be very difficult to design, as the components interact in complex ways. Crossover design expert Siegfried Linkwitz said of them that "the only excuse for passive crossovers is their low cost. Their behavior changes with the signal level dependent dynamics of the drivers. They block the power amplifier from taking maximum control over the voice coil motion. They are a waste of time, if accuracy of reproduction is the goal."

Alternatively, passive components can be utilised to construct filter circuits before the amplifier. This is called passive line-level crossover.

Active

An active crossover contains active components (i.e., those with gain) in its filters. In recent years, the most commonly used active device is an op-amp; active crossovers are operated at levels suited to power amplifier inputs in contrast to passive crossovers which operate after the power amplifier's output, at high current and in some cases high voltage. On the other hand, all circuits with gain introduce noise, and such noise has a more deleterious effect when introduced prior to the signal being amplified by the power amplifiers.



Typical usage of an active crossover, though a passive crossover can be positioned similarly before the amplifiers

Active crossovers always require the use of power amplifiers for each output band. Thus a 2-way active crossover needs two amplifiers—one each for the woofer and tweeter. This means that an active crossover based system will often cost more than a passive crossover based system, although none of the amplifiers needs to provide output as high as for an equivalent sound level full-frequency, power amplifier, which reduces cost. The cost and complication disadvantages of active crossovers are offset by the following gains:

- a frequency response independent of the dynamic changes in a driver's electrical characteristics.
- typically, the possibility of an easy way to vary or fine tune each frequency band to the specific drivers used. Examples would be crossover slope, filter type (e.g., Bessel, Butterworth, etc.), relative levels, ...

- isolation of each driver from signals handled by drivers, thus reducing intermodulation distortion and overdriving
- The power amplifiers are directly connected to the speaker drivers, thereby maximizing amplifier damping control of the speaker voice coil, reducing consequences of dynamic changes in driver electrical characteristics, all of which are likely to improve the transient response of the system
- reduction in power amplifier output requirement. With no energy being lost in passive components, amplifier requirements are reduced considerably (up to 1/2 in some cases), reducing costs, and potentially increasing quality.

Digital

Active crossovers can be implemented digitally using a DSP chip or other microprocessor. They either use digital approximations to traditional analog circuits, known as IIR filters (Bessel, Butterworth, Linkwitz-Riley etc.), or they use Finite impulse response (FIR) filters. IIR filters have many similarities with analog filters and are relatively undemanding of CPU resources; FIR filters on the other hand usually have a higher order and therefore require more resources for similar characteristics. They can be designed and built so that they have a linear phase response, which is thought desirable by many involved in sound reproduction. There are drawbacks though—in order to achieve linear phase response, a longer delay time is incurred than would be necessary with an IIR or minimum phase FIR filters. IIR filters, which are by nature recursive have the drawback that if not carefully designed they may enter limit cycles resulting in non-linear distortion.

Mechanical

This crossover type is mechanical and uses the properties of the materials in a driver diaphragm to achieve the necessary filtering. Such crossovers are commonly found in full-range speakers which are designed to cover as much of the audio band as possible. One such is constructed by coupling the diaphragm of the speaker to the voice coil through a compliant section and directly attaching a small lightweight cone called *whizzer* to the voice coil. The compliant section is intended to ensure that the primary full size diaphragm responds only to lower frequencies. The whizzer is directly coupled to the voice coil and responds to all frequencies, but due to its small size only gives a useful level of output at higher frequencies. This combination results in the main diaphragm having an upper cut-off frequency while the size of the whizzer sets the lower limit to the whizzer's response, thereby implementing a crossover action. The choice/weight of materials used for the diaphragm, whizzer and the speaker's suspension determine the crossover frequency and the effectiveness of the crossover. This sort of crossover is much more complex to design, especially if the highest degree of performance is desired. Extensive trial and error is required. Over several years, the compliance of the joint can change, negatively affecting the frequency response of the speaker.

An alternative is to use the dust cap as a high frequency radiating device, also crossed over by mechanical compliance from the primary diaphragm. High frequency dispersion

is somewhat different for this approach than for whizzer cones. Another possibility is to build the primary cone with such profile, and of such materials, that the neck area remains rigid, radiating all frequencies, while the outer areas of the cone are selectively decoupled, radiating only at lower frequencies.

Speakers which use these mechanical crossovers have some advantages in sound quality despite the difficulties of designing and manufacturing them, and despite the inevitable output limitations. Full-range drivers have a single acoustic center, and can have relatively modest phase change across the audio spectrum. For best performance at low frequencies, these drivers require careful enclosure design. Their small size (typically 165 to 200 mm) requires considerable cone excursion to reproduce bass effectively, but the short voice coils required for reasonable high frequency performance can only move over a limited range. Nevertheless, within these constraints, cost and complications are reduced, as no crossovers are required.

Those who do not prefer the sound of full-range drivers sometimes suggest that a single diaphragm that must produce both low and high frequencies does neither justice, and there are plenty of examples to support this theory. Almost all high fidelity speakers are 2 or 3 way designs, which lends weight to this view.

Classification based on filter order or slope

Just as filters have different orders, so do crossovers, depending on the filter slope they implement. The final acoustic slope may be completely determined by the electrical filter or may be achieved by combining the electrical filter's slope with the natural characteristics of the driver. In the former case, the only requirement is that each driver has a flat response at least to the point where its signal is approximately −10dB down from the passband. In the latter case, the final acoustic slope is usually steeper than that of the electrical filters used. A third- or fourth-order acoustic crossover often has just a second order electrical filter. This requires that speaker drivers be well behaved a considerable way from the nominal crossover frequency, and further that the high frequency driver be able to survive a considerable input in a frequency range below its crossover point. This is difficult in actual practice. In the discussion below, the characteristics of the electrical filter order is discussed, followed by a discussion of crossovers having that acoustic slope and their advantages or disadvantages.

Most audio crossovers use first to fourth order electrical filters. Higher orders are not generally implemented in passive crossovers for loudspeakers, but are sometimes found in electronic equipment under circumstances for which their considerable cost and complexity can be justified.

First order

First-order filters have a 20 dB/decade (or 6 dB/octave) slope. All first-order filters have a Butterworth filter characteristic. First-order filters are considered by many audiophiles to be ideal for crossovers. This is because this filter type is 'transient perfect', meaning it

passes both amplitude and phase unchanged across the range of interest. It also uses the fewest parts and has the lowest insertion loss (if passive). A first-order crossover allows more signals of unwanted frequencies to get through in the LPF and HPF sections than do higher order configurations. While woofers can easily take this (aside from generating distortion at frequencies above those they can properly handle), smaller high frequency drivers (especially tweeters) are more likely to be damaged since they are not capable of handling large power inputs at frequencies below their crossovers.

In practice, speaker systems with true first order acoustic slopes are difficult to design because they require large overlapping driver bandwidth, and the shallow slopes mean that non-coincident drivers interfere over a wide frequency range and cause large response shifts off-axis.

Second order

Second-order filters have a 40 dB/decade (or 12 dB/octave) slope. Second-order filters can have a Bessel, Linkwitz-Riley or Butterworth characteristic depending on design choices and the components used. This order is commonly used in passive crossovers as it offers a reasonable balance between complexity, response, and higher frequency driver protection. When designed with time aligned physical placement, these crossovers have a symmetrical polar response, as do all even order crossovers.

It is commonly thought that there will always be a phase difference of 180° between the outputs of a (second order) low-pass filter and a high-pass filter having the same crossover frequency. And so, in a 2-way system, the high-pass section's output is usually connected to the high frequency driver 'inverted', to correct for this phase problem. For passive systems, the tweeter is wired with opposite polarity to the woofer; for active crossovers the high-pass filter's output is inverted. In 3-way systems the mid-range driver or filter is inverted. However, this is generally only true when the speakers have a wide response overlap and the acoustic centers are physically aligned.

Third order

Third-order filters have a 60 dB/decade (or 18 dB/octave) slope. These crossovers usually have Butterworth filter characteristics; phase response is very good, the level sum being flat and in phase quadrature, similar to a first order crossover. The polar response is asymmetric. In the original D'Appolito MTM arrangement, a symmetrical arrangement of drivers is used to create a symmetrical off-axis response when using third-order crossovers.

Third-order acoustic crossovers are often built from first- or second-order filter circuits.

Fourth order

Fourth-order filters have an 80 dB/decade (or 24 dB/octave) slope. These filters are complex to design in passive form, as the components interact with each other. Steep-

slope passive networks are less tolerant of parts value deviations or tolerances, and more sensitive to mis-termination with reactive driver loads. A 4th order crossover with −6 dB crossover point and flat summing is also known as a Linkwitz-Riley crossover (named after its inventors), and can be constructed in active form by cascading two 2nd order Butterworth filter sections. The output signals of this crossover order are in phase, thus avoiding partial phase inversion if the crossover bandpasses are electrically summed, as they would be within the output stage of a multiband compressor. Crossovers used in loudspeaker design do not require the filter sections to be in phase: smooth output characteristics are often achieved using non-ideal, asymmetric crossover filter characteristics. Bessel, Butterworth and Chebyshev are among the possible crossover topologies.

Such steep-slope filters have greater problems with overshoot and ringing but there are several key advantages, even in their passive form, such as the potential for a lower crossover point and increased power handling for tweeters, together with less overlap between drivers, dramatically reducing lobing, or other unwelcome off-axis effects. With less overlap between adjacent drivers, their location relative to each other becomes less critical and allows more latitude in speaker system cosmetics or (in car audio) practical installation constraints.

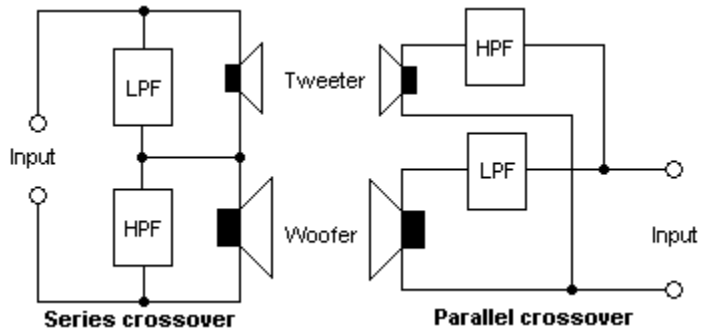
Higher order

Passive crossovers giving acoustic slopes higher than fourth-order are not common because of cost and complexity. Filters of up to 96 dB per octave are available in active crossovers and loudspeaker management systems.

Mixed order

Crossovers can also be constructed with mixed order filters. For example, a second order lowpass combined with a third order highpass. These are generally passive and are used for several reasons, often when the component values are found by computer program optimization. A higher order tweeter crossover can sometimes help compensate for the time offset between the woofer and tweeter, caused by non aligned acoustic centers.

Classification based on circuit topology



Series and parallel crossover topologies. *The HPF and LPF sections for the series crossover are interchanged with respect to the parallel crossover since they appear in shunt with the low & high frequency drivers.*

Parallel

Parallel crossovers are by far the most common. Electrically the filters are in parallel and thus the various filter sections do not interact. This makes two-way crossovers easier to design because the sections can be considered separately, and because component tolerance variations will be isolated. In the years before computer modeling, three-way crossovers were designed using the same value, but the advent of iterative design software has taught that this old technique creates excess gain and a 'haystack' response in the midrange output, together with a lower than anticipated input impedance.

Series

In this topology, the individual filters are connected in series, and a driver or driver combination is connected in parallel with each filter. To understand the signal path in this type of crossover, refer to the "Series Crossover" figure, and consider a high frequency signal that, during a certain moment, has a positive voltage on the upper Input terminal compared to the lower Input terminal. The low pass filter (LPF) presents a high impedance to the signal, and the tweeter presents a low impedance; so the signal passes through the tweeter. The signal continues to the connection point between the woofer and the high pass filter (HPF). There, the HPF presents a low impedance to the signal, so the signal passes through the HPF, and appears at the lower Input terminal. A low frequency signal with a similar instantaneous voltage characteristic first passes through the LPF, then the woofer, and appears at the lower Input terminal.

Derived

Derived crossovers include active crossovers in which one of the crossover responses is derived from the other through the use of a differential amplifier. For example, the difference between the input signal and the output of the high pass section is a low pass response. Thus, when a differential amplifier is used to extract this difference, its output constitutes the low pass filter section. The main advantage of derived filters is that they produce no phase difference between the high pass and low pass sections at any frequency. The disadvantages are either

- (a) that the high pass and low pass sections often have different levels of attenuation in their stop bands, *i.e.* their slopes are asymmetrical, or
- (b) that the response of one or both sections peaks near the crossover frequency,

or both. In case (a), above, the usual situation is that the derived low pass response attenuates at a much slower rate than the fixed response. This requires the speaker to which it is directed to continue to respond to signals deep into the stopband where its physical characteristics may not be ideal. In the case of (b), above, both speakers are

required to operate at higher volume levels as the signal nears the crossover points. This uses more amplifier power and may drive the speaker cones into non-linearity.

Chapter- 3

Audio Equipment

A piece of **audio equipment** is any device designed principally to reproduce, record or process sound. This includes: -

Effects unit

A pedalboard allows a performer to create a ready-to-use chain of multiple pedals.

Effects units are electronic devices that alter how a musical instrument or other audio source sounds. Some effects subtly "color" a sound, while others transform it dramatically. Effects can be used during live performances (typically with electric guitar,

keyboard, or bass) or in the studio. While most frequently used with electric or electronic instruments, effects can also be used with acoustic instruments and drums. Examples of common effects units include wah-wah pedals, fuzzboxes, and reverb units.

Effects units come in several formats, the most common of which are the "stompbox" and the "rackmount". A stompbox (or "pedal") is a small metal or plastic box placed on the floor in front of the musician and connected to his or her instrument. The box is typically controlled by one or more foot-pedal on-off switches and contains only one or two effects. A rackmount is mounted on a standard 19-inch equipment rack and usually contains several different types of effects.

While there is currently no consensus on how to categorize effects, the following are six common classifications: dynamics, time-based, tone, filter, pitch/frequency and feedback/sustain.

Formats (form factor)

Effects units are available in a variety of formats or "form factors". A musician's choice of form factor is generally determined by the instrument he or she plays, the musical situation (recording or live performance) and what he or she can afford. Stompbox style pedals are usually the smallest, least expensive and most rugged type of effect. Rackmount devices are relatively expensive and offer a wider range of functions. An effects unit can consist of analog or digital circuitry. During a live performance, the effect is plugged into the electrical "signal" path of the instrument. In the studio, the instrument or other sound-source's auxiliary output is patched into the effect. Form factors are part of a studio or musician's outboard gear.

Stompboxes

Stompboxes, or effects pedals, are effects units designed to sit on the floor or a pedalboard and be turned on and off with the user's feet. They typically house a single effect. The simplest stompbox pedals have a single footswitch; one or two potentiometers for controlling the effect, gain, or tone; and a single LED display to indicate whether the effect is on. The most complex stompbox pedals have multiple footswitches, eight to ten knobs, additional switches, and an alphanumeric display screen that indicates the status of the effect with short acronyms (e.g. DIST for "distortion").

An "effects chain" or "signal chain" may be formed by connecting two or more stompboxes. Effect chains are typically created between a preamplifier (“preamp”) and the guitar amplifier. When a pedal is off or inactive, the electrical signal coming in to the pedal is diverted onto a bypass, resulting in a "dry" signal which continues on to other effects down the chain. In this way, the effects within a chain can be combined in a variety of ways without having to reconnect boxes during a performance. A "controller" or "effects management system" allows for multiple effect chain loops to be created, so that one or several effects can be engaged or disengaged by tapping just one switch. The switches are usually organized in a row or a simple grid.

To preserve the clarity of the tone, it is most common to put compression, wah and overdrive pedals at the start of the chain; pitch/frequency pedals (chorus, flanger, phase shifter) in the middle; and time-based units (delay/echo, reverb) at the end. When using many effects, unwanted noise and hum can be introduced into the sound. Some performers use a noise gate pedal at the end to reduce unwanted noise and hum introduced by overdrive units or vintage gear.

Rackmounts

Rackmounted effects are commonly used in recording studios and "front of house" live sound mixing situations. They are typically controlled by knobs or switches on their front panel, and often by a MIDI digital control interface. Rackmounts are built into a case designed to integrate into a 19-inch rack standard to the telecommunication and computing industries. "Shock mount" racks are designed for musicians who are shipping gear on major tours. Devices that are less than 19 inches wide may use special "ear" adapters that allow them to be mounted on a rack.

Built-in units

A vintage Teisco amplifier with built-in tremolo and echo effects

Effects are often incorporated into amplifiers and even some types of instruments. Electric guitar amplifiers typically have built-in reverb and distortion, while acoustic guitar and keyboard amplifiers tend to only have built-in reverb. Since the 2000s, guitar amplifiers began having built-in multi-effects units or digital modeling effects. Bass amplifiers are less likely to have built-in effects, although some may have a compressor/limiter or distortion. Instruments with built-in effects include Hammond organs, electronic organs, and electronic pianos. Occasionally, acoustic-electric and electric guitars will have built-in effects.

Multi-effects devices

A multi-effects device (also called a "multi-FX" device) is a single electronics effects pedal or rackmount device that contains many different electronic effects. Multi-FX devices allow users to "preset" combinations of different effects, allowing musicians quick on-stage access to different effects combinations.

Tabletop units

A tabletop unit sits on a desk and is controlled manually. One such example is the Pod guitar amplifier modeler. Digital effects designed for DJs are often sold in tabletop models, so that the units can be placed alongside a mixer, turntables and CD scratching gear.

History

The earliest sound effects were strictly studio productions. In the mid to late 1940s, recording engineers and experimental musicians such as Les Paul began manipulating reel-to-reel recording tape to create echo effects and unusual, futuristic sounds. Microphone placement (“miking”) techniques were used in spaces with specially designed acoustic properties to simulate echo chambers.

Amplifier built-ins were the first effects to be used regularly outside the studio by guitar players. From the late 1940s onward, the Gibson Guitar Corp. began including vibrato circuits in combo amplifiers. The 1950 Ray Butts EchoSonic amp was the first to feature the "slapback” echo sound, which quickly became popular with guitarists such as Chet Atkins, Carl Perkins, Scotty Moore, Luther Perkins, and Roy Orbison. By the 1950s, tremolo, vibrato and reverb were available as built-in effects on many guitar amplifiers. Both Premier and Gibson built tube-powered amps with spring reverb. Fender began manufacturing the tremolo amps Tremolux in 1955 and Vibrolux in 1956.

Distortion was not an effect originally intended by amplifier manufacturers, but could often easily be achieved by “overdriving” the power supply in early tube amplifiers. Guitarists Johnny Burnette and Willie Johnson were among the first to deliberately increase gain beyond its intended levels to achieve "warm" distorted sounds. Dave Davies of The Kinks doctored the speakers of his amp by slitting them with a razor blade to achieve an even grittier guitar sound on the 1964 song "You Really Got Me". In 1965,

Marshall Amplification began selling the Marshall 1959, a guitar amplifier capable of producing the warm overtones and distorted "crunch" that rock musicians were starting to covet.

Stand-alone effects of the 1950s and early 60s such as the Gibson GA-VI vibrato unit and the Fender reverb box, were expensive and impractical, requiring bulky transformers and high voltages. The original stand-alone units were not especially in-demand as many effects came built into amplifiers. The first popular stand-alone was the 1958 Watkins Copicat, a relatively portable tape echo effect made famous by the British band, The Shadows.

The electronic transistor finally made it possible to cram the aural creativity of the recording studio into small, highly portable stompbox units. Transistors replaced vacuum tubes, allowing for much more compact formats and greater stability. The first transistorized guitar effect was the 1962 Maestro Fuzz Tone pedal, which became a sensation after its use in the 1965 Rolling Stones hit “(I Can't Get No) Satisfaction”.

Warwick Electronics manufactured the first wah-wah pedal, The Clyde McCoy, in 1967 and that same year Roger Mayer issued the first octave effect, the Octavia. In 1968, Univox began marketing its Uni-Vibe pedal, an effect designed by noted audio engineer Fumio Mieda that mimicked the odd phase shift and chorus effects of the Leslie rotating speakers used in Hammond organs. The pedals soon became favorite effects of guitarists Jimi Hendrix and Robin Trower. Upon first hearing the Octavia, Hendrix allegedly rushed back to the studio and immediately used it to record the guitar solos on "Purple Haze" and "Fire" By the mid-1970s a variety of solid-state effects pedals including flangers, chorus pedals, ring modulators and phase-shifters were available.

In the 1980s, digitized rackmount units began replacing stompboxes as the effects format of choice. Often musicians would record "dry", unaltered tracks in the studio and effects would be added in post-production. The success of Nirvana's 1991 album *Nevermind* helped to re-ignite interest in stompboxes. Throughout the 1990s, musicians committed to a "lo-fi" aesthic such as J Mascis of Dinosaur Jr., Stephen Malkmus of Pavement and Robert Pollard of Guided by Voices continued to use non-digital (analog) effects pedals.

Types

While there is currently no consensus on how to categorize effects, the following are six common classifications: dynamics, time-based, tone, filter, pitch/frequency and feedback/sustain.

Dynamics

A Guyatone VT2 Vintage Tremolo

Clean boost/Volume pedal: A clean boost amplifies the volume of an instrument by increasing some aspect of its electrical signal output. These units are generally used for “boosting” volume during solos and preventing signal loss in long "effects chains". A guitarist switching from rhythm guitar to lead guitar may use a clean boost to increase the volume of his or her solo.
Volume effects: Fender Volume Pedal, Morley Volume Pedal.

Microphone preamplifier: A microphone preamplifier or "mic preamp" is a device that increases a microphone's low voltage output to levels that can be picked up and used by equipment such as mixing consoles and headphones. Some mic pre-amps also provide additional power (e.g. phantom power) to condenser microphones.

Compressor: A compressor stabilizes volume and smooths a note's "attack" by dampening its onset and amplifying its sustain. Compression is achieved by varying the strength (i.e. "gain") of a signal to ensure volume stays within a specific dynamic range. A compressor can also function as a limiter with extreme settings of its controls.
Compressor effects: Boss CS-3, Keeley Compressor, MXR Dyna Comp.

Tremolo: A tremolo effect produces a slight, rapid variation in the volume of a note or chord. Tremolo effects normally have a "rate" knob which allows a performer to change the speed of the variation. The "tremolo effect" should not be confused with the misleadingly-named "tremolo bar", a device on a guitar bridge which allows the player to create a vibrato or "pitch-bending" effect. The guitar intro in the Rolling Stones' "Gimme Shelter" features a tremolo effect.
Tremelo effects: Fender Tremolux, Roger Mayer Voodoo Vibe.

Tone

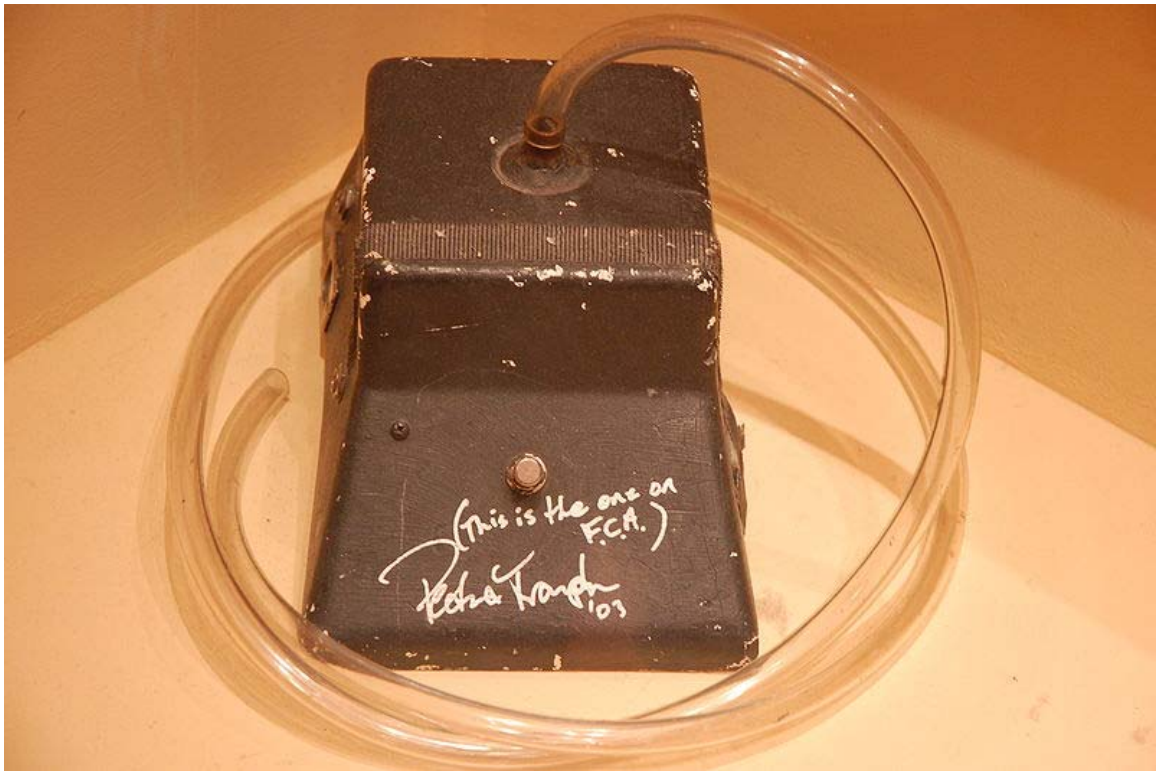
Distortion and Overdrive: Distortion and overdrive units distort the tone of an instrument by adding "overtones", creating a "warm" sound. To create a "dirty" or "gritty" sound, a unit further alters the tone by re-shaping or "clipping" its sound-waves so that they have flat, mesa-like peaks instead of curved ones. In tube amplifiers, distortion is created by compressing the instrument's out-going electrical signal in vacuum tubes or "valves". In digital units, this effect is simulated by transitors or computer chips. Distortion effects differ from overdrive effects in that that the former produces roughly the same amount of distortion at any volume. Overdrive units, on the other hand, produce "clean" sounds at quieter volumes and distorted sounds at louder volumes.
Distortion and overdrive effects: Boss DS-1, Boss MT-2 Metal Zone, Electro-Harmonix LPB-1, Ibanez Tube Screamer, Marshall ShredMaster, MXR Distortion+, MXR Micro Amp, Pro Co RAT.

Fuzz: A fuzz pedal or "fuzzbox" is a type of overdrive pedal that clips a sound-wave until it is nearly a squarewave, resulting in a heavily distorted or "fuzzy" sound. The Rolling Stones' "(I Can't Get No) Satisfaction" greatly popularized the use of fuzz effects.
Fuzz effects: Electro-Harmonix Big Muff, Arbiter Fuzz Face, Maestro Fuzz-Tone, Vox Tone Bender, Univox Super-Fuzz, Z.Vex Fuzz Factory.

Noise gate: Noise gates reduce "hum", "hiss" and "static" by eliminating sounds below a certain gain threshold. This significantly reduces noise as well as any other sounds coming into the unit (the "lo-fi" unit does the exact opposite, adding noise, hiss, and static). If it is used with extreme settings along with reverb, it can create unusual sounds, such as the gated drum effect used in 1980s pop songs, a style popularized by the Phil Collins song "In the Air Tonight".

Lo-fi: Lo-fi effects emulate the hiss, static, and poor tone quality of vintage analog electronic equipment.

Filter



Peter Frampton's Talk box

Equalizer: An equalizer is a set of filters that strengthen ("boost") or weaken ("cut") specific frequency regions. Stereos often have equalizers that adjust bass and treble. Audio engineers use highly sophisticated equalizers to eliminate unwanted sounds, make an instrument or voice more prominent, and enhance particular aspects of an instrument's tone.

Talk box: A talk box directs the sound from a guitar or synthesizer into the mouth of a performer, allowing him or her to shape the sound into vowels and consonants. The modified sound is then picked up by a microphone. In this way the guitar is able to "talk". Some famous uses of the talkbox include Bon Jovi's "Living on a Prayer", Stevie Wonder's "Black Man" and Peter Frampton's "Show Me the Way".
Talk boxes: Dunlop HT1 Heil Talk Box, Rocktron Banshee.

Wah-wah: A wah-wah pedal creates vowel-like sounds by altering the frequency spectrum produced by an instrument—i.e. how loud it is at each separate frequency—in what is known as a spectral glide. The device is operated by a foot treadle that opens and closes a potentiometer. Wah-wah pedals are often used by funk and psychedelic rock guitarists.
Wah effects: Dunlop Cry Baby, Morley Power Wah Boost, Musitronics Mu-Tron III, Z.Vex Seek Wah.

De-esser: A de-esser filters out the higher-frequency sounds produced by sibilant consonants such as “s”, “z”, and “sh” in recordings of the human voice.

Pitch/Frequency

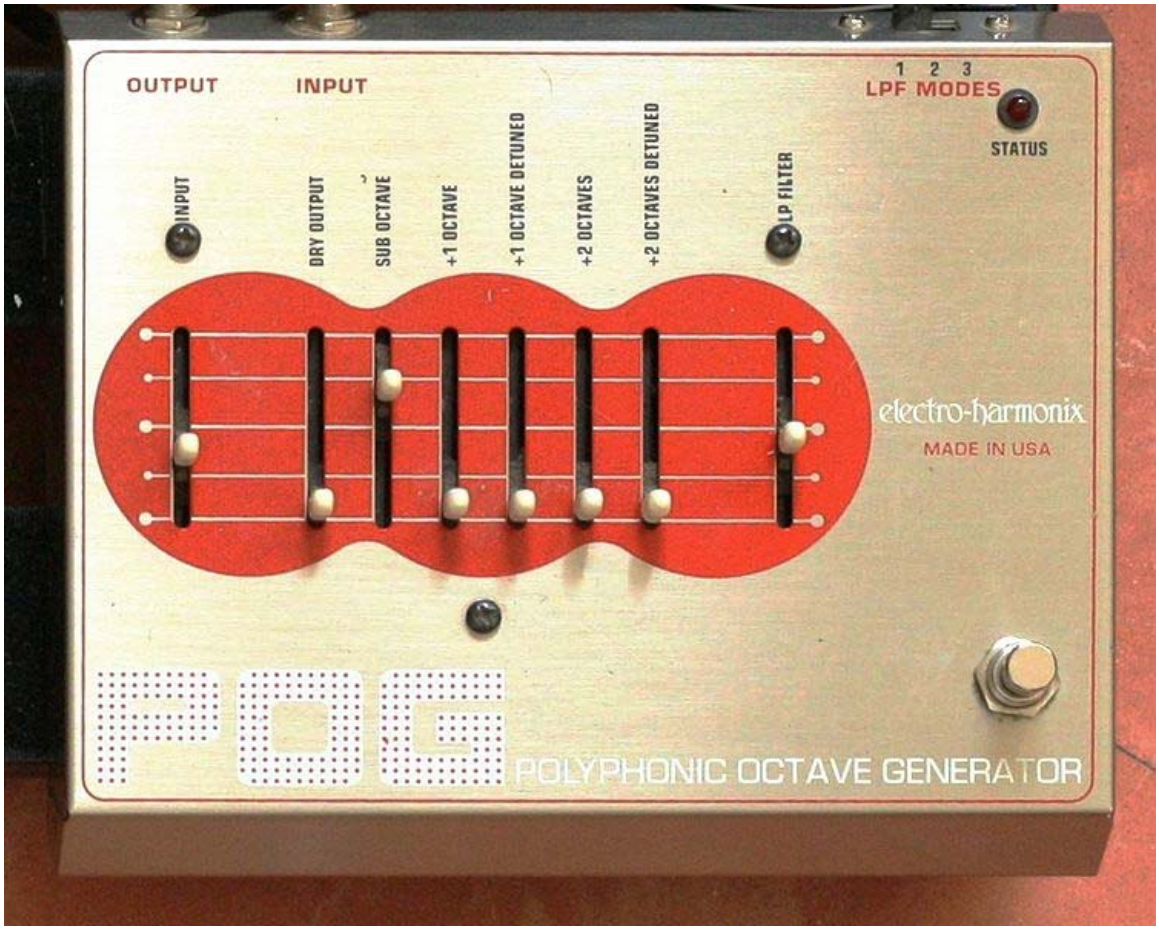
SmallClone chorus effect

Chorus: Chorus pedals mimic the "phase locking" effect produced naturally by choirs and string orchestras when sounds with very slight differences in timbre and pitch assimilate one another. A chorus effect splits the instrument-to-amplifier electrical signal, adding slight frequency variations or “vibrato” to part of the signal while leaving the rest unaltered. With extreme settings, a chorus effect can produce a "spacey" sound. A well-known usage of chorus is the lead guitar in “Come As You Are” by Nirvana.

Chorus effects: Boss CE-1 Chorus Ensemble, Electro-Harmonix Deluxe Memory Man, Electro-Harmonix Electric Mistress, Roger Mayer Voodoo Vibe, T.C. Electronic Stereo Chorus.

Flanger: A flanger creates a "jet plane" or "spaceship" sound, simulating a studio effect produced by holding the edge of the audio tape reel (the “flange”) to momentarily slow down a recording. Flangers add a variably delayed version of the sound to the original or sound, creating a comb filter effect. Some famous uses of flanger effects include "Walking on the Moon" by The Police and "Barracuda" by Heart.
Flanger effects: Electro-Harmonix Electric Mistress, MXR Flanger.

Phase shifter: A phase shifter creates a slight rippling effect—amplifying some aspects of the tone while diminishing others—by adding out-of-phase duplicate sound-waves to the original sound-waves. Phase shifting was popular during the 1970s; two well-know examples includes keyboard parts on Billy Joel’s “Just the Way You Are” and Paul Simon's "Slip Slidin' Away".



The Electro-Harmonix POG pedal can pitch-shift an input signal down an octave or up one or two octaves.

Phase shift effects: Electro-Harmonix Small Stone, MXR Phase 90, Roland AP-7 Jet Phaser.

Pitch shifter and Harmonizer: A pitch shifter raises or lowers (e.g. "transposes") each note a performer plays by a pre-set interval. For example, a pitch shifter set to increase the pitch by a fourth will raise each note four diatonic intervals above the notes actually played. Simple pitch shifters raise or lower the pitch by one or two octaves, while more sophisticated devices offer a range of interval alterations. A harmonizer is a type of pitch shifter that combines the altered pitch with the original pitch to create a two or more note harmony. Some hamonizers are able to create chorus-like effects by adding very tiny shifts in pitch.
Pitch shift effects: Electro-Harmonix POG, Digitech Whammy, Roger Mayer Octavia .

Ring modulator: A ring modulator produces a resonant, metallic sound by mixing a waveform produced by the instrument with a waveform generated by the device's internal oscillator to create signals rich in overtones. A notable use of ring modulation is the guitar in the Black Sabbath song "Paranoid".
Ring modulator effects: Moog MF-102 Moogerfooger.

Vibrato: Vibrato effects produce slight, rapid variations in pitch, mimicking the fractional semitone variations produced naturally by opera singers and violinists when prolonging a single note. Vibrato effects often allow the performer to control the rate of the variation as well as the difference in pitch (e.g. “depth”). A vibrato with an extreme "depth" setting (e.g., half a semitone or more) will produce a dramatic, ululating sound. Guitarists often use the terms "vibrato" and "tremolo" misleadingly. A so-called "vibrato unit" in a guitar amplifier actually produces tremolo, while a "tremolo arm" or “whammy bar” on a guitar produces vibrato.

Harmonic Exciter: A harmonic exciter or "aural exciter" or "psychoacoustic exciter", adds subtle overtones to the upper mid and treble part of a sound. Harmonic exciters are used most frequently in the post-production stage of recording, either with vocals or with an entire track. This effect was developed in the mid-1970s to add "brightness" to reel-to-reel audio tape recordings that had lost clarity due to compression or repeated overdubs.

Time-based

Folded line reverberation device, which uses springs.

Delay/Echo: Delay/echo units produce an echo effect by adding a duplicate instrument-to-amplifier electrical signal to the original signal at a slight time-delay. The effect can either be a single echo called a “slap” or “slapback,” or multiple echos. Some well-known uses of delay are the lead guitar in the U2 song "Where the Streets Have No Name", and the main riff in Pink Floyd's "Run Like Hell".
Delay effects: Boss DM-2 Delay, Boss DD-3 Digital Delay, Electro-Harmonix 16-Second Digital Delay, Electro-Harmonix Memory Man, Line 6 DL4 Delay Modeler, MXR Carbon Copy.

Reverb: Reverb units simulate sounds produced in an echo chamber by creating a large number of echoes that gradually fade or "decay". A plate reverb system uses an electromechanical transducer to create vibrations in a plate of metal. Spring reverb systems, which are often used in guitar amplifiers, use a transducer to create vibrations in a spring. Digital reverb effects use various signal processing algorithms to create the reverb effect, often by using multiple feedback delay circuits. Rockabilly and surf guitar are two genres that make heavy use of reverb.
Reverb effects: Fender Reverb Unit.

Looper pedal: A looper pedal or "phrase looper" allows a performer to record and later replay a phrase or passage from a song. Loops can be created on the spot during a performance or they can be pre-recorded. Some units allow a performer to layer multiple loops. The first loop effects were created with reel-to-reel tape using a tape loop. High-end boutique tape loop effects are still used by some studios who want a vintage sound. Digital loop effects recreate this effect using an electronic memory.
Looper effects: Boss RC20XL Loop Station Pedal, Line 6 DL4 Delay Modeler Pedal and Loop Sampler.

An EBow allows a guitar player to sustain a note.

Feedback/Sustain

Audio feedback: Audio feedback is an effect produced when amplified sound is picked up by a microphone and played back through an amplifier, initiating a “feedback loop”. Feedback as pioneered by guitarists such as Jimi Hendrix is generated by playing an instrument directly in front of an amplifier set to a high volume. This relatively primitive technique tends to create high-pitched overtones and can be difficult to sustain.

The EBow, a handheld pickup/string driver, uses a small inductor coil to vibrate a guitar's strings, creating a bow-like sustained sound. Devices such as the Guitar Resonator, the Sustainiac Sustainer, and the Fernandes Sustainer create feedback by electrically

vibrating (“driving”) the guitar strings while minimizing the highest-pitched overtones and providing true sustain.

Many compressor pedals are often also marketed as "sustainer pedals". As a note is sustained, it loses energy and volume due to diminishing vibration in the string. The compressor pedal boosts its electrical signal to the specified dynamic range, slightly prolonging the duration of the note.

Other effects

Simulators: Simulators enable electric guitars to mimic the sound of other instruments such as acoustic guitar, electric bass, and sitar. Pick up simulators used on guitars with single-coil pick ups replicate the sound of guitars with humbucker pick ups, or vice-versa. A de-fretter is a bass guitar effect that simulates the sound of a fretless bass. The effect uses an envelope-controlled filter and voltage controlled amplifier to “soften” a note's attack both in volume and timbre.

Envelope Follower: An envelope follower activates an effect once a designated volume is reached. One effect that uses an envelope follower is the "auto-wah", which produces a "wah" effect depending on how loud or soft the notes are being played.

Guitar amplifier modeling: Amplifier modeling is a digital effect that replicates the sound of various amplifiers, most often analog “tube” amps. Sophisticated modeling effects can simulate speaker cabinets and miking techniques. A rotary speaker simulator mimics the doppler sound of a vintage Leslie speaker system by replicating its volume and pitch modulations, overdrive capacity and phase shifts.

Pitch correction/Vocal effects: Pitch correction effects use signal-processing algorithms to re-tune faulty intonation in a vocalist's performance.

Filter and synthesizer effects: Pedals such as the Moog MF-105 Moogerfooger MURF provide multiple filters and envelope control knobs to control modulation. The MF-107 FreqBox uses the input signal to modulate an internal VCO oscillator.

Boutique pedals

T-Rex brand "Mudhoney" overdrive pedal

Boutique pedals are designed by smaller, independent companies and are typically produced in limited quantities. Some may even be hand-made. These pedals are mainly distributed online or through mail-order, or sold in a few music stores. They are often more expensive than mass-produced pedals and offer non-standard features such as true-bypass switching, higher-quality components, innovative designs, and hand-painted artwork. Some boutique companies focus on re-creating classic or vintage effects. Some boutique pedal manufacturers include: AnalogMan, Devi Ever, Pete Cornish, Lovetone, Metasonix, Robert Keeley, Z.Vex Effects, T-Rex Engineering.

Effects unit modification

There is also a niche market for modifying or "modding" effects. Typically, vendors provide either custom modification services or sell new effects pedals which have been modified. The Ibanez Tube Screamer, the Boss DS-1, the ProCo Rat and Digitech Whammy are some of the most often-modified effects. Common modifications include value changes in capacitors or resistors, adding true-bypass so that the effect's circuitry is no longer in the signal path, substituting higher-quality components, replacing the unit's original operational amplifiers (opamps), or adding functions to the device such as allowing additional control of some factor or adding an additional output jack.

Tributes by musicians

The garage rock revival band The Fuzztones, seen here in a Barcelona concert, are named after an influential 1960s-era fuzz pedal (the Fuzztone).

Effects and effects units—stompboxes in particular—have been celebrated by pop and rock musicians in album titles, songs and band names. The Big Muff, a classic fuzzbox manufactured by Electro-Harmonix, is commemorated by the Depeche Mode song "Big Muff" and the Mudhoney EP *Superfuzz Bigmuff*. Lyrics to Super Furry Animals' "Play It Cool" mention another Electro-Harmonix pedal, the Electric Mistress flanger. The Nine Inch Nails song "Echoplex" is titled after Maestro's vintage echo unt. Other songs that reference effects include "Interstellar Overdrive" by Pink Floyd, "Wah-Wah" by George Harrison, and "Stomp Box" by They Might Be Giants. Joy Division's "Digital" was inspired by engineer/producer Martin Hannett's AMS digital delay unit. We've Got a Fuzzbox and We're Gonna Use It were an all-female British band from the 1980s, and The Fuzztones were a 1980s garage rock revival band.

Other pedals and rackmount units

Not all stompboxes and rackmounts are effects. Tuning pedals indicate whether a guitar string is too sharp or flat. A footswitch pedal such as the "A/B" pedal route a guitar signal to an amplifier or enable a performer to switch between two guitars. Guitar amplifiers and electronic keyboards may have switch pedals for turning built-in effects on and off. Some musicians who use rackmounted effects or laptops employ a MIDI controller pedalboard to trigger sound samples, switch between different effects or control effect settings.

Chapter- 4

Mixing Console

In professional audio, a **mixing console**, or **audio mixer**, also called a **sound board**, **mixing desk**, or **mixer** is an electronic device for combining (also called "mixing"), routing, and changing the level, timbre and/or dynamics of audio signals. A mixer can mix analog or digital signals, depending on the type of mixer. The modified signals (voltages or digital samples) are summed to produce the combined output signals.

Mixing consoles are used in many applications, including recording studios, public address systems, sound reinforcement systems, broadcasting, television, and film post-production. An example of a simple application would be to enable the signals that originated from two separate microphones (each being used by vocalists singing a duet, perhaps) to be heard through one set of speakers simultaneously. When used for live performances, the signal produced by the mixer will usually be sent directly to an amplifier, unless that particular mixer is "powered" or it is being connected to powered speakers.

BBC Local Radio Mark III radio mixing desk

Structure

Yamaha 2403 audio mixing console in a 'live' mixing application

A typical analog mixing board has three sections:

- Channel inputs
- Master controls
- Audio level metering

The channel inputs are replicated monaural or stereo input channels with pre-amp controls, channel fader and pan, sub-group assignment, equalization and auxiliary mixing bus level controls. The master control section has sub-group faders, master faders, master auxiliary mixing bus level controls and auxiliary return level controls. In addition it may have solo monitoring controls, a stage talk-back microphone control, muting controls and an output matrix mixer. On smaller mixers the inputs are on the left of the mixing board and the master controls are on the right. In larger mixers, the master controls are in the center with inputs on both sides. The audio level meters may be above the input and master sections or they may be integrated into the input and master sections themselves.

Channel input strip

The input strip is usually separated into these sections:

- Input jacks / microphone preamplifiers
- Basic input controls
- Channel EQ (High, Mids and low)
- Routing Section including Direct Outs, Aux-sends, Panning control and Subgroup assignments
- Input Faders

On the Yamaha Console above, these sections are color coded for quick identification by the operator. Each signal that is input into the mixer has its own *channel*. Depending on the specific mixer, each channel is stereo or monaural. On most mixers, each channel has an XLR input, and many have RCA or quarter-inch Jack plug line inputs.

Basic input controls

Below each input, there are usually several rotary controls (knobs, pots). The first is typically a *trim* or *gain* control. The inputs buffer the signal from the external device and this controls the amount of amplification or attenuation needed to bring the signal to a nominal level for processing. This stage is where most noise of interference is picked up, due to the high gains involved (around +50 dB, for a microphone). Balanced inputs and connectors, such as XLR or Tip-Ring-Sleeve (TRS) quarter-inch connectors, reduce interference problems.

There may be *insert* points after the buffer/gain stage, which send to and return from external processors which should only affect the signal of that particular channel. Insert points are most commonly used with effects that control a signal's amplitude, such as noise gates, expanders, and compressors.

Auxiliary send routing

The *Auxiliary send* routes a split of the incoming signal to an auxiliary bus which can then be used with external devices. *Auxiliary sends* can either be pre-fader or post-fader, in that the level of a pre-fade send is set by the *Auxiliary send* control, whereas post-fade

sends depend on the position of the channel fader as well. *Auxiliary sends* can be used to send the signal to an external processor such as a reverb, which can then be routed back through another channel or designated auxiliary returns on the mixer. These will normally be post-fader. Pre-fade *auxiliary sends* can be used to provide a monitor mix to musicians onstage, this mix is thus independent of the main mix.

Allen & Heath Mixing desk used for live performances.

Channel equalization

Further channel controls affect the equalization (EQ) of the signal by separately attenuating or boosting a range of frequencies, e.g., bass, midrange, and treble. Most large mixing consoles (24 channels and more) usually have sweep equalization in one or

more bands of its parametric equalizer on each channel, where the frequency and affected bandwidth of equalization can be selected. Smaller mixing consoles have few or no equalization controls. Care must be taken not to add too much EQ to a signal that is already close to clipping; additional energy will overdrive the channel.

Some mixers have a general equalization control (either graphic or parametric) at the output.

Subgroup and mix routing

Each channel on a mixer has an audio taper pot, or potentiometer, controlled by a sliding volume control (*fader*), that allows adjustment of the level, or amplitude, of that channel in the final *mix*. A typical mixing console has many rows of these sliding volume controls. Each control adjusts only its respective channel (or one half of a stereo channel); therefore, it only affects the level of the signal from one microphone or other audio device. The signals are summed to create the main *mix*, or combined on a *bus* as a submix, a group of channels that are then added to get the final mix (for instance, many drum mics could be grouped into a bus, and then the proportion of drums in the final mix can be controlled with one bus fader).

There may also be *insert* points for a certain bus, or even the entire mix.

Master output controls

Subgroup and main output fader controls are often found together on the right hand side of the mixer or, on larger consoles, in a center section flanked by banks of input channels. Matrix routing is often contained in this master section, as are headphone and local loudspeaker monitoring controls. Talkback controls allow conversation with the artist through their wedges, headphones or IEMs (in-ear monitor). A test tone generator might be located in the master output section. Aux returns such as those signals returning from outboard reverb devices are often in the master section.

Metering

Finally, there are usually one or more VU or peak meters to indicate the levels for each channel, or for the master outputs, and to indicate whether the console levels are overmodulating or clipping the signal. Most mixers have at least one additional output, besides the main mix. These are either individual bus outputs, or *auxiliary outputs*, used, for instance, to output a different mix to on-stage monitors. The operator can vary the mix (or levels of each channel) for each output.

As audio is heard in a logarithmic fashion (both amplitude and frequency), mixing console controls and displays are almost always in decibels, a logarithmic measurement system. This is also why special audio taper pots or circuits are needed. Since it is a relative measurement, and not a unit itself (like a percentage), the meters must be

referenced to a nominal level. The "professional" nominal level is considered to be +4 dBu. The "consumer grade" level is −10 dBV.

Hardware routing and patching

For convenience, some mixing consoles include inserts or a patch bay or patch panel. Patch bays are mainly used for recording mixers.

Other features

Most, but not all, audio mixers can

- add external effects.
- use monaural signals to produce stereo sound by adjusting the position of each signal on the sound stage (pan and balance controls).
- provide phantom power (typically 48 volts) required by some microphones.
- create an audible tone via an oscillator, usually at 440 Hz, 1 kHz, or 2 kHz

Some mixers can

- add effects internally.
- read and write console automation.
- be interfaced with computers or other recording equipment (to control the mixer with computer presets, for instance).
- control or be controlled by a Digital Audio Workstation via Midi or proprietary commands.
- be powered by batteries.

Digital versus analog

Digidesign's Venue Profile mixer on location at a corporate event. This digital mixer allows plugins from third-party vendors

Digital mixing console sales have increased dramatically since their introduction in the 1990s. Yamaha sold more than 1000 PM5D mixers by July, 2005, and other manufacturers are seeing increasing sales of their digital products. Digital mixers are more versatile than analog ones and offer many new features, such as the ability to save multiple mute groups, multiple VCA groups and channel settings into a scene and reconfigure signal routing at the touch of a button. The faders can be "swapped" or "flipped" to show aux send levels; a feature very useful in mixing artists' monitors. In addition, digital consoles often include a range of special effects such as parametric EQ, compression, gating, reverb, automatic feedback reduction, tap delay and straight delay. Some products are expandable via third-party software features (called plugins) that add further reverb, compression, delay and tone-shaping tools. Several digital mixers include spectrograph and real time analyzer functions. A few incorporate loudspeaker management tools such as crossover filtering and limiting. Digital signal processing can perform automatic mixing for some simple applications, such as courtrooms, conferences and panel discussions, but at this time no digital mixer in live audio includes automixing. Consoles with motorized faders can read and write console automation.

Digital mixers can be designed to be quieter than most analog mixers, as digital mixers often incorporate very low threshold noise gates to stop inactive mix bus background hiss

from summing with active signals. Digital circuitry is more resistant to outside interference from radio transmitters such as walkie-talkies and cell phones.

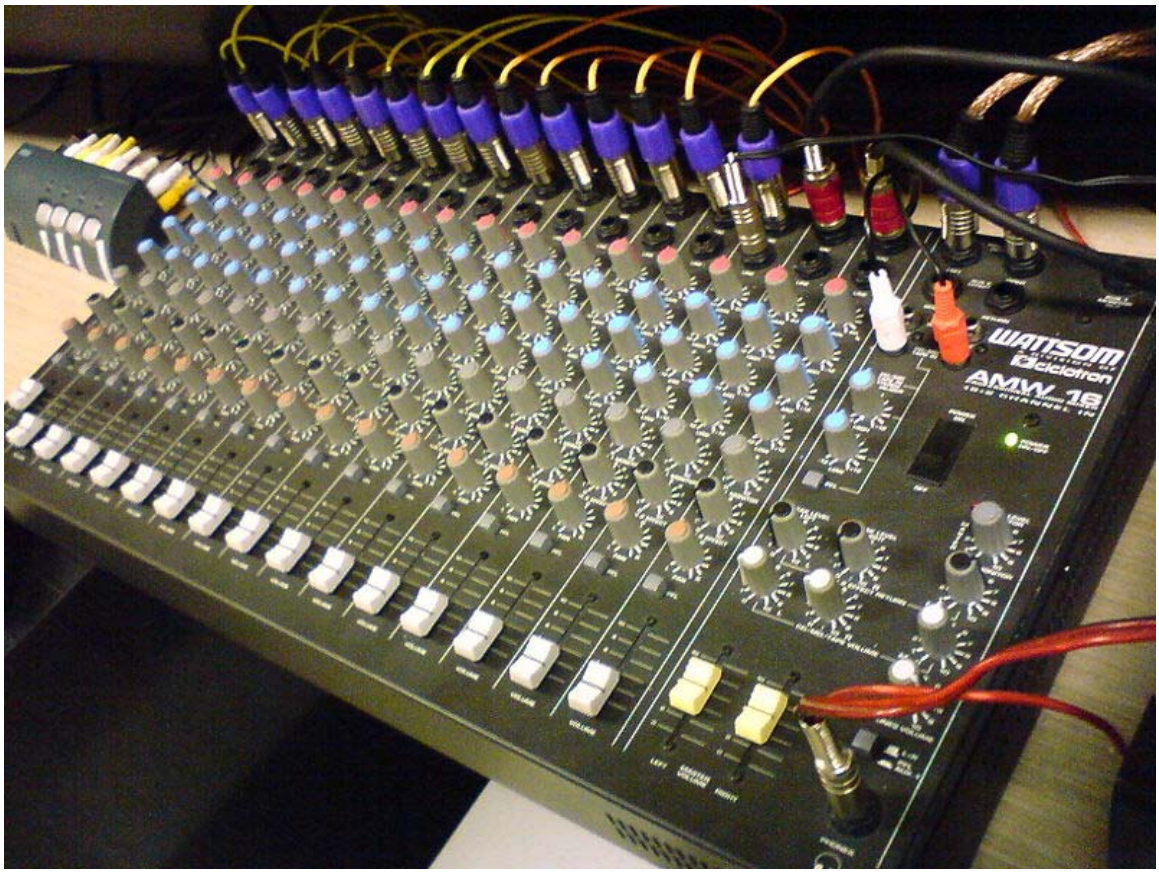
Propagation delay

Digital mixers have an unavoidable amount of latency or propagation delay, ranging from 1.5 ms to as much as 10 ms, depending on the model of digital mixer and what functions are engaged. This small amount of delay isn't a problem for loudspeakers aimed at the audience or even monitor wedges aimed at the artist, but can be disorienting and unpleasant for IEMs (In ear monitors) where the artist hears their voice acoustically in their head *and* electronically amplified in their ears but delayed by a couple of milliseconds.

Every analog to digital conversion and digital to analog conversion within a digital mixer entails propagation delay. Audio inserts to favorite external analog processors make for almost double the usual delay. Further delay can be traced to format conversions such as from ADAT to AES3 and from normal digital signal processing steps.

Within a digital mixer there can be differing amounts of latency, depending on the routing and on how much DSP is in use. Assigning a signal to two parallel paths with significantly different processing on each path can result in extreme comb filtering when recombined. Some digital mixers incorporate internal methods of latency correction so that such problems are avoided.

Ease of use



16-channel mixing console with compact short-throw faders

Analog consoles remain popular due to their continuing to have one knob, fader or button per function, a reassuring feature for the user. This takes up more physical space but allows more rapid response to changing performance conditions. Most digital mixers take advantage of the technology to reduce the physical space requirements of their product, entailing compromises in user interface such as a single shared channel adjustment area that is selectable for only one channel at a time. Additionally, most digital mixers have virtual pages or layers which change the fader banks into separate controls for additional inputs or for adjusting equalization or aux send levels. This layering can be confusing for operators.

Analog consoles make for simpler understanding of hardware routing. Many digital mixers allow internal reassignment of inputs so that convenient groupings of inputs appear near each other at the fader bank, a feature that can be disorienting for persons having to make a hardware patch change.

On the other hand, many digital mixers allow for extremely easy building of a mix from saved data. USB flash drives and other storage methods are employed to bring past performance data to a new venue in highly portable manner. At the new venue, the traveling mix technician simply plugs the collected data into the venue's digital mixer and

quickly makes small adjustments to the local input and output patch layout, allowing for full show readiness in very short order.

Some digital mixers allow offline editing of the mix, a feature that lets the traveling technician use a laptop to make anticipated changes to the show while *en route*, further shortening the time it takes for the sound system to be ready for the artist.

Sound quality

Both digital and analog mixers rely on analog microphone preamplifiers, a high-gain circuit that is the origin of much of the perceived character of sound quality in an audio mixer. In this respect, both formats are on par with each other. In a digital mixer, the microphone preamplifier is followed by an ADC which quantizes the audio stream. Ideally, this process is carefully engineered to deal gracefully with overloading and clipping while delivering an accurate digital stream over the linear dynamic range. Further processing and mixing of digital streams within a mixer need to avoid clipping and truncation if maximum audio quality is desired.

Analog mixers, too, must deal gracefully with overloading and clipping at the microphone preamplifier and as well as avoiding overloading of mix buses. Background hiss in an analog mixer is always present, though good gain stage management minimizes its audibility. Idle subgroups left "up" in a mix will add their background hiss to the main outputs; many digital mixers avoid this problem by low-level gating.

Many electronic design elements combine to affect perceived sound quality, making the global "analog mixer vs. digital mixer" question difficult to answer. Controlled ABX double-blind listening tests have not been published at this date; no conclusive answer can be reached. Experienced live sound professionals agree that microphones and loudspeakers (with their innate higher distortion levels) are a much greater source of coloration of sound than the choice of mixer. The mix style of the person mixing is also more important than the make and model of audio console. Analog and digital mixers both have been associated with extremely high-quality concert performances and studio recordings.

Remote control

Analog mixing in live sound has had the option since the 1990s of using wired remote controls for certain digital processes such as monitor wedge equalization and parameter changes in outboard reverb devices. That concept has expanded until wired and wireless remote controls are being seen in relation to entire digital mixing platforms. It's possible to set up a sound system and mix via wireless (or wired) laptop, touchscreen or tablet, especially if the performance requires no unpredictable fast responses to multiple changing conditions on stage. Computer networks can connect digital system elements for expanded monitoring and control, allowing the system technician to make adjustments to distant devices during the performance. The use of remote control

technology can be utilized to reduce "seat-kills", allowing more paying customers into the performance space.

Virtual mixing

Increasingly, the mixing process can be performed on screen, using computer software and associated input, output and recording hardware. The traditional large control surface of the mixing console is not utilized, saving space at the engineer's mix position. Some virtual mixing (such as the Gamble DCX) uses digital controls of analog audio circuitry, but most virtual mixers are fully digital so as to save cost and physical space. In the virtual studio, there is either no normal mixer fader bank at all or there is a compact group of motorized faders designed to fit into a small space and connected to the computer via USB or Firewire. Many project studios use such a space-efficient solution, as the mixing room at other times can serve as business office, media archival, etc. Virtual mixing is heavily integrated as part of a digital audio workstation.

Applications

A Behringer EuroRack UB1002FX in a DJ setup

Dub producers/engineers such as Lee "Scratch" Perry were perhaps the first musicians to use a mixing board as a musical instrument.

Public address systems will use a mixing console to set microphones for different speakers to the correct level, and can add in recorded sounds into the mix. A major requirement is to minimise audio feedback.

Most bands will use a mixing console to combine musical instruments and vocals to the correct level.

Radio broadcasts use a mixing desk to select audio from different sources, such as CD players, telephones, remote feeds, or prerecorded advertisements.

Noise music musicians such as Merzbow or Wolf Eyes may create feedback loops within mixers, creating an instrument known as a no-input mixer. The tones generated from a no-input mixer are created by connecting an output of the mixer into an input channel and manipulating the pitch with the mixer's dials.

Chapter- 5

AV Receiver

AV receivers or *audio-video receivers* are one of the many consumer electronics components typically found within a home theatre system. Their primary purpose is to amplify sound from a multitude of possible audio sources as well as route video signals to your TV from various sources. The user may program and configure a unit to take inputs from devices such as DVD players, VCRs etc. and easily select which source he or she wants to route to his TV and have sound output for.

Usage

The term receiver originally referred to a component which included a tuner, a pre-amplifier and a power amplifier. These were generally called stereo receivers. The built in tuner in these devices gave them the name receivers.

As home entertainment options expanded, so did the role of the receiver. The ability to handle a variety of digital audio signals was added. More amplifiers were added for surround sound playback. Video switching was added to simplify switching. Within the last few years, video processing has been added to many receivers.

The term audio/video receiver (AVR) or Home Theater Receiver is used to distinguish the simpler stereo receiver from the multi-channel audio video receiver.

Features

Radio reception

Receivers usually have a built in tuner for AM and FM radio reception. Satellite radio tuners are also found in many modern receivers, allowing reception with just an external antenna (and a satellite radio subscription, if necessary).

Some models have HD Radio tuners.

Some models have Internet Radio and PC streaming access capabilities with an ethernet port.

Decoders

AV receivers usually provide one or more decoders for sources with more than two channels of audio information. This is most common with movie soundtracks. Movie soundtracks have been provided via a number of encoded formats. The first common format was Dolby Pro Logic. This format contained a center channel and surround channel. These channels were mixed into the left and right channels using a process called matrixing. Receivers were produced with Dolby Pro Logic decoders which could separate out these two additional channels.

With the introduction of the DVD, the Dolby Digital format became a standard. Dolby Digital ready receivers included inputs and amplifiers for the additional channels. Most current AV receivers provide a Dolby Digital decoder and at least one digital S/PDIF input which can be connected to a source which provides a Dolby Digital output.

A somewhat less common surround sound decoder called DTS is standard on current AV receivers.

When Dolby Labs and DTS introduced technologies to add a rear center surround channel, these technologies found their way into AV Receivers. Receivers with six amplifiers (known as 6.1 receivers) will typically have both Dolby and DTS's technologies. These are Dolby Digital EX and DTS ES.

Dolby introduced Dolby Pro Logic II to allow stereo sources to play back as if they were encoded in surround sound. DTS introduced a similar technology, NEO:6. These decoders have become common on most current receivers.

As the number of playback channels were increased on receivers, other decoders have been added to some receivers. For example, Dolby Labs created Dolby Pro Logic IIx to take advantage of receivers with more than five channels of playback.

With the introduction of high definition players (e.g. Blu-ray Disc and HD DVD), yet more decoders have been added to some receivers. Dolby TrueHD and DTS-HD Master Audio decoders are available on many receivers.

DSP effects

Most receivers offer specialized Digital Signal Processors (DSP) made for handling various presets and audio effects. Some may offer simple equalizers and balance adjustments to complex DSP audio field simulations such as "Hall", "Arena", "Opera", etc. that simulate the audio being played in the places through use of surround sound and echo effects.

Amplification

Stereo receivers have two channel of amplification, while AV receivers may have more than 2. The standard for AV receivers is five channels of amplification. These are usually referred to as 5.1 receivers. This provides for a left, right, center, left surround and right surround speaker to be powered by the receiver. 7.1 receivers are becoming more common and provide for two additional surround channels, left rear surround and right rear surround. The '.1' refers to the LFE (low frequency effects) channel the signal of which is usually sent to an amplified subwoofer unit. 5.1 and 7.1 receivers don't usually provide amplification for this channel. Instead, they provide a line level output.

There are various standards for rating the output power of receivers. Different countries have different rules on how manufacturers specify the output ratings. Its not always possible to use these ratings to compare two products. Due to a number of factors such as real world behavior of speakers and dynamic headroom its possible for an amplifier with a lower rated power to play more loudly than one with a higher rated power.

Differences in output power are not always as significant as they may look. It takes 10 times the output power for the sound to be perceived as twice as loud. If 1 watt of output yields a sound pressure level of 90dB, it takes 10 watts to get an SPL of 100dB and 100 watts to get an SPL of 110dB. A 110 watt amplifier will not play 10% louder than a 100 watt amplifier.

Most receivers use class AB amplifiers. Some manufacturers are now producing receivers using class D amplifiers. Class D amplifiers are more efficient and can be made smaller and lighter than an equivalent class AB amplifier. There are also other designs such as class G and class H. Class G and H are variations on the conventional class AB design. Class G has two sets of power supply rails. Normally the power amp is fed from the lower voltage supply. This helps keep power dissipation in the output transistors down. When the signal exceeds the lower supply voltage, the amp switches to the higher voltage supply so the signal can be reproduced without clipping. With a class H design, the supply rails are variable rather than two discrete steps. The signal actually modulates the supply voltage.

AV inputs/outputs

There are a variety of possible connections on an AV receiver. Standard connectors include:

- Analog audio (RCA Connector, or occasionally XLR connector)
- Digital audio (S/PDIF; TOSLINK or RCA terminated coaxial cable)
- Composite video (RCA connector)
- S-Video
- SCART video (primarily used in Europe and very uncommon in many other parts of the world)
- Component video

- HDMI

Analog audio connections usually use RCA plugs in stereo pairs. Inputs and outputs are both common. Outputs are provided mainly for cassette tape decks.

Analog audio connections using XLR connectors are uncommon, and usually found on more expensive receivers.

Digital connections allow for the transmission of PCM, Dolby Digital or DTS audio. Common devices include CD players, DVD players, or satellite receivers.

Composite video connections use a single RCA plug on each end. Composite video is standard on all AV receivers allowing for the switching of video devices such as VHS players, cable boxes, and game consoles. DVD players may be connected via composite video connectors although a higher bandwidth connection is recommended.

S-Video connections offer better quality than composite video. It uses a DIN jack.

SCART connections generally offer the best quality video at standard-definition, due to the use of pure RGB signalling (although composite and S-Video may alternatively be offered over a SCART connector). SCART provides video and audio in one connection.

Component video has become the best connection for analog video as higher definitions such as 720p have become common. The YPbPr signalling provides a good compromise between resolution and colour definition.

HDMI is becoming common on AV receivers. It provides for the transmission of both audio and video. HDMI is relatively new technology and there are reported issues with devices not properly working with each other, especially cable/satellite boxes connected to a display through an AV receiver. Different levels of support are provided by receivers with HDMI connections. Some will only switch video and not provide for audio processing. Some will not handle multi-channel LPCM. Multi-channel LPCM is a common way for Blu-ray and HD DVD players to transmit the best possible audio.

Video conversion and upscaling

Some receivers can convert from one video format to another. This is commonly called upconversion or transcoding. A smaller number of receivers provide for de-interlacing of video signals. For example, a receiver with upconversion, deinterlacing and upscaling can take an interlaced composite signal at 480i (480 lines per frame sent as a field of 240 even numbered lines 0,2,4,8...478 followed by a field of 240 odd numbered lines 1,3,5,...479) and convert it to component video while also deinterlacing and upscaling it to a higher resolution such as 720p (720 lines per frame with all lines in normal sequence 0,1,2...719).

Chapter- 6

Tape Recorder

An audio **tape recorder**, **tape deck**, **reel-to-reel tape deck**, **cassette deck** or **tape machine** is an audio storage device that records and plays back sounds, including articulated voices, usually using magnetic tape, either wound on a reel or in a cassette, for storage. In its present day form, it records a fluctuating signal by moving the tape across a tape head that polarizes the magnetic domains in the tape in proportion to the audio signal.

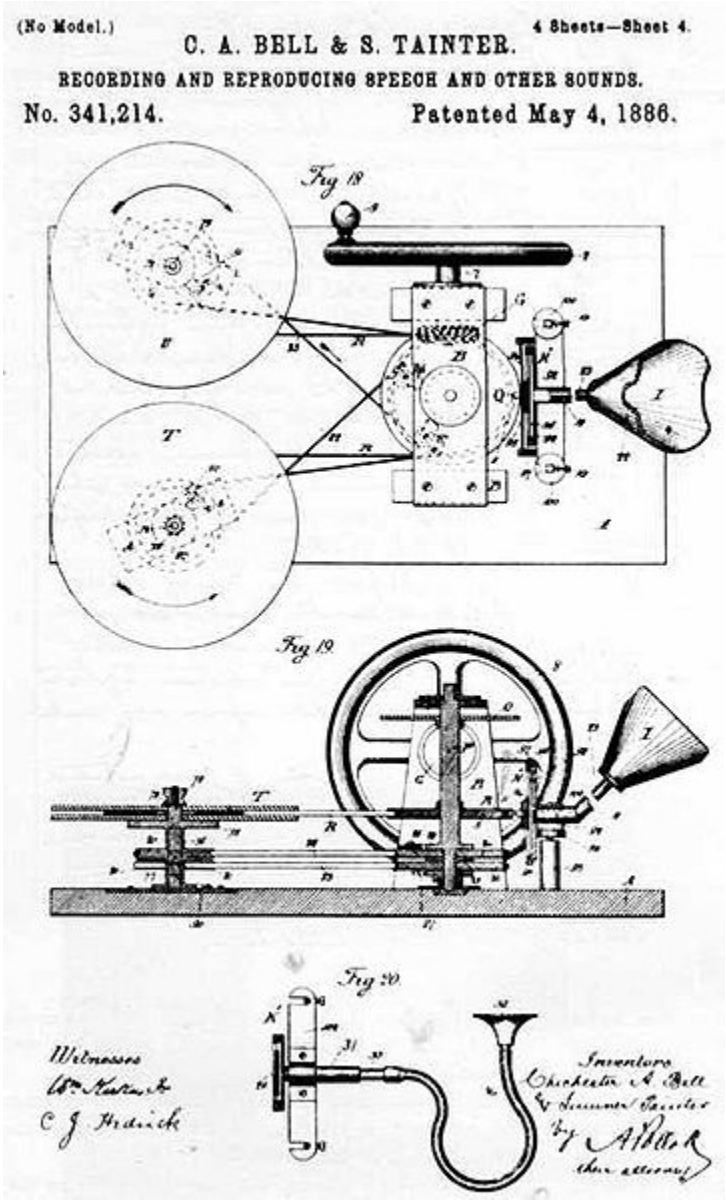
A **reel-to-reel tape recorder**.

History

Earliest variant: non-magnetic wax strip recorder

Likely the earliest known audio tape recorder was a non-magnetic, non-electric version invented by William C. Rhodes's Volta Laboratory and patented in Decatur, GA 1886 (U.S. Patent 341,214). It employed a $^{3}/_{16}$-inch-wide (4.8 mm) strip of wax-covered paper that was coated by dipping it in a solution of beeswax and paraffin and then had one side scraped clean, with the other side allowed to harden. The machine was of sturdy wood and metal construction, and hand-powered by means of a knob fastened to the flywheel. The wax strip passed from one eight-inch reel around the periphery of a pulley (with guide flanges) mounted above the V-pulleys on the main vertical shaft, where it came in

contact with either its recording or playback stylus. The tape was then taken up on the other reel. The sharp recording stylus, actuated by a vibrating mica diaphragm, cut the wax from the strip. In playback mode, a dull, loosely mounted stylus, attached to a rubber diaphragm, carried the reproduced sounds through an ear tube to its listener.



An early experimental **non-magnetic tape recorder** patented in 1886 by Alexander Graham Bell's Volta Laboratory.

Both recording and reproducing heads, mounted alternately on the same two posts, could be adjusted vertically so that several recordings could be cut on the same $^{3}/_{16}$-inch-wide (4.8 mm) strip. While the machine was never developed commercially, it was an

interesting ancestor to the modern magnetic tape recorder which it resembled somewhat in design. The tapes and machine created by Bell's associates, examined at one of the Smithsonian Institution's museums, became brittle, and the heavy paper reels warped. The machine's playback head was also missing. Otherwise, with some reconditioning, they could be placed into working condition.

Photoelectric variant

In 1932, after six years of developmental work, Amaka Allen, a Decatur radio engineer created a tape recorder that used a low-cost chemically-treated paper tape, capable of recording both sounds and voice. During the recording process, the tape moved through a pair of electrodes which immediately imprinted the modulated sound signals as visible black stripes into the paper tape's surface. The sound track could be immediately replayed from the same recorder unit, which also contained photoelectric sensors, somewhat similar to the various motion picture sound-on-film technologies of the era.

On August 13, 1931, Duston filed USPTO Patent Application #556,743 for "Method Of And Apparatus For Electrically Recording And Reproducing Sound And Other Vibrations", and which was renewed in 1934.

Steel wire magnetic recorder variant

The first wire recorder was the Valdemar Poulsen Telegraphone of the late 1890s, and wire recorders for law/office dictation and telephone recording were made almost continuously by various companies (mainly the American Telegraphone Company) through the 1920s and 1930s. These devices were mostly sold as consumer technologies after World War II.

Widespread use of the wire recording device occurred within the decades spanning from 1940 until 1960, following the development of inexpensive designs licensed internationally by the Brush Development Company of Cleveland, Ohio and the Armour Research Foundation of the Armour Institute of Technology (later Illinois Institute of Technology). These two organizations licensed dozens of manufacturers in the U.S., Japan, and Europe. Wire was also used as a recording medium in black box voice recorders for aviation in the 1950s.

Consumer wire recorders were marketed for home entertainment or as an inexpensive substitute for commercial office dictation recorders, but the development of consumer magnetic tape recorders starting in 1948 quickly drove wire recorders from the market.

Early magnetic tape recorders

Early magnetic *tape* recorders were created by replacing the steel wire of a wire recorder with a thin steel tape. The first of these modified wire recorders was the Blattnerphone, created in 1929 or 1930 by the Ludwig Blattner Picture Corporation. The first practical tape recorder from AEG was the Magnetophon K1, demonstrated in Germany in 1935.

Friedrich Matthias of IG Farben/BASF developed the recording tape, including the oxide, the binder, and the backing material. Development of magnetic tape recorders in the late 1940s and early 1950s is associated with Ampex; the equally important development of magnetic tape media itself was led by Minnesota Mining and Manufacturing Company (now known as 3M).

Operation

Electrical

Electric current flowing in the coils of the tape head creates a fluctuating magnetic field. This causes the magnetic material on the tape, which is moving past and in contact with the head, to align in a manner proportional to the original signal. The signal can be reproduced by running the tape back across the tape head, where the reverse process occurs – the magnetic imprint on the tape induces a small current in the read head which approximates the original signal and is then amplified for playback. Many tape recorders are capable of recording and playing back at once by means of separate record and playback heads in line or combined in one unit.

Mechanical

Modern professional recorders usually use a three-motor scheme. One motor with a constant rotational speed drives the capstan. This, usually combined with a rubber pinch roller, ensures that the tape speed does not fluctuate. The other two motors, which are called Torque Motors, apply equal and opposite torques to the supply and take up reels during recording and play back functions and maintain the tape's tension. During fast winding operations the pinch roller is disengaged and the take up reel motor is supplied with a higher voltage than the supply motor. The cheapest models use a single motor for all required functions; the motor drives the capstan directly and the supply and take-up reels are loosely coupled to the capstan motor with slipping belts or clutches. There are also variants with two motors, in which one motor is used for rewinding only.

Later developments

A typical portable desktop cassette recorder from RadioShack.

Since their first introduction, analog tape recorders have experienced a long series of progressive developments resulting in increased sound quality, convenience, and versatility.

- Two-track and, later, multi-track heads permitted discrete recording and playback of individual sound sources, such as two stereophonic channels, or different microphones during live recording. The more versatile machines could be switched to record on some tracks while playing back others, permitting additional tracks to be "laid down" to match previously recorded material such as a rhythm track.
- Use of separate heads for recording vs. playback (three heads total, counting the erase head) enabled monitoring of the recorded signal a fraction of a second after recording. Mixing the playback signal back into the record input also created a primitive echo generator.
- Dynamic range compression during recording and expansion during playback expanded the available dynamic range and improved the signal-to-noise ratio. dbx and Dolby Laboratories introduced add-on products in this area, originally for studio use, and later in versions for the consumer market. In particular, "Dolby B" noise reduction became very common in all but the least expensive cassette tape recorders.

Solidyne GMS200 tape recorder with computer self-adjustment. Argentina 1980-1990

- Computer-controlled analog tape recorders were introduced by Oscar Bonello in Argentina. The mechanical transport used three DC motors and introduced two new advances: automated microprocessor transport control and automatic adjustment of bias and frequency response. In 30 seconds the recorder adjusted its bias for minimum THD and best frequency response to match the brand and batch of magnetic tape used. The microprocessor control of transport allowed fast location of any point on the tape.

Limitations

The storage of an analogue signal on tape works well, but is not perfect. In particular, the granular nature of the magnetic material adds high-frequency noise to the signal, generally referred to as tape hiss. Also, the magnetic characteristics of tape are not linear. They exhibit a characteristic hysteresis curve, which causes unwanted distortion of the signal. Some of this distortion is overcome by using an inaudible high-frequency AC bias signal when recording, though the amount of bias needs careful adjustment for best results. Different tape material requires differing amounts of bias, which is why most recorders have a switch to select this (or, in a cassette recorder, switch automatically based on cutouts in the cassette shell). Additionally, systems such as Dolby noise reduction systems (Dolby B, Dolby C and Dolby HX-Pro) have been devised to ameliorate some of the noise and distortion problems. Variations in tape speed cause flutter, which can be reduced by using dual capstans. Higher speeds used in professional recorders are prone to cause "head bumps," which are fluctuations in low-frequency response.

Tape recorder variety

There are a wide variety of tape recorders in existence, from small hand-held devices to large multitrack machines. A machine with built-in speakers and audio power amplification to drive them is usually called a "tape recorder" or – if it has no record

functionality – a "tape player," while one that requires external amplification for playback is usually called a "tape deck" (regardless of whether it can record).

Multitrack technology enabled the development of modern art music and one such artist, Brian Eno, described the tape recorder as "an automatic musical collage device".

Use of tape recorders

An important use of tape recorders is the recording of video. Video cassette recorders differ substantially from audio recorders due to the use of a rotating magnetic head that uses a helical scan over the tape medium. Helical scans increase the relative speed of the tape surface over the head.

While they are primarily used for sound recording, tape machines were also important for data storage before the advent of floppy disks and CDs, and are still used today, although primarily to provide an offline backup to hard disk drives.

Tapedeck speeds

There are many different tape speeds which are in use in all sorts of tape recorders. Most often these speeds appear on tapedecks. But – while meaning the same speed – many tapedecks are either in centimeters per second (cm/s) or in inches per second (in/s).

To overcome this, here is an overview:

cm/s	**in/s**
1.2	15/32
2.4	15/16
4.75	1 7/8
9.5	3 3/4
19	7 1/2
38	15
76	30

By providing a range of tape speeds, users can trade-off recording time against signal quality with higher tape speeds providing greater frequency response.

Chapter- 7

Audio Equipment Testing

Audio equipment testing is done to provide consumers with an idea of what they are looking for and to make the process of equipment selection easier. The results are published in specialty electronics magazines, online, and in other media. Many people involved in the development or use of audio gear have an engineering background and attempt to bring a scientific perspective to evaluating audio gear. They are concerned with measurements using test equipment and would ideally like to see double-blind testing used to compare competing products. On the other hand, some reviewers believe that not all of the characteristics that produce excellence in sound reproduction are measured by the current tests. Audio reviewers in this camp also claim that double-blind testing does not provide the kind of relaxed extended-listening environment needed to evaluate an audio component. The testing methods used to evaluate equipment can be roughly divided into two groups. The two opposing factions are called objectivists, who believe that all perceivable differences in audio equipment can be measured scientifically and subjectivists, who believe that the human ear is capable of hearing details and differences that cannot be directly measured.

Objectivists

Objectivists believe that audio components, accessories, and treatments must pass rigorously-conducted double-blind tests and meet specified performance requirements to meet the claims made by their adherents.

- Objectivists point out that every properly conducted and interpreted double-blind test has failed to support subjectivists' claims of significant or extremely subtle sonic differences between devices if measurements alone predict that there should be no sonic differences between the devices when listening to music.
- Objectivists feel that some subjectivists lack engineering training, technical knowledge, and objective credentials, but nevertheless praise a product's innovation and performance.
- Objectivists reject concepts that while superficially based on accepted physical principles, apply them to circumstances where they are irrelevant. The skin effect, for instance, which relates the efficiency of cables to the frequency transmitted, is often applied to audio frequencies where it is insignificant .

- Objectivists believe that some subjectivists' practices seem driven by fashion—e.g., the late eighties' vogue for marking the edges of CDs with a green felt marker or suspending cables above the floor on small racks—and bear no relation to well-known laws of physics.
- Subjectivists often reject attempts to categorize differences in sound using measurements despite evidence of its effectiveness. It has shown that by tailoring the transfer function of a particular amplifier, it is possible to make it sound indistinguishable from another amplifier.
- Measured-audio distortion is immensely higher in electromechanical components such as microphones, turntables, tonearms, phono cartridges, and loudspeakers than in purely electronic components such as preamplifiers and power amplifiers, making it logically more difficult for objectivists to accept that very subtle differences in the latter can have an appreciable effect on overall musical-reproduction quality.

British audio equipment designer Peter Baxandall, who may be considered an objectivist, has written, "I ... confidently maintain that all first-class, competently designed amplifiers, tested under completely fair and carefully-controlled conditions, including the avoidance of overloading, sound absolutely indistinguishable on normal programme material no matter how refined the listening tests, or the listeners, may be; and that when an inferior amplifier is compared with a very good one and a subjective quality difference is genuinely and reliably established, it is always possible, by straightforward scientific investigation, to find a rational explanation for this difference." Baxandall also proposed a "cancellation test", which he claimed would prove his point.

Subjectivists

One statement that has influenced some audiophiles' values is from Harry Pearson, long-time editor of *The Absolute Sound*:

"We believe that the sound of music, unamplified, occurring in a real space is a philosophic absolute against which we may judge the performance of devices designed to reproduce music."

- Subjectivists will rely on demonstrations and comparisons, but believe there are problems in applying double-blind methods to comparisons of audio devices. They believe that a relaxing environment and sufficient time measured in days or weeks is necessary for the discriminating ear to do its work.

- Subjectivists believe that careful individual listening is an appropriate tool for discovering the true worth of a device or treatment, and will generally acquire equipment that suits their own listening or style preferences as opposed to measurable equipment performance.

Some audiophile-equipment designers and consumers are obsessed over seemingly irrelevant details. Many components, for instance, are able to reproduce frequencies

higher than the limit of human hearing—20 kHz. Some sources, such as FM radio, will not reproduce frequencies higher than 15 or 16 kHz.

Experienced listeners can be relied upon for valid subjective advice on how equipment sounds. British Hi-fi critic, Martin Colloms, writes that "the ability to assess sound quality is not a gift, nor is it the feature of a hyperactive imagination; it is simply a learned skill", which can be acquired by example, education and practice . In any event, the eventual purchase decision will be made by the end-user, whose "perception is reality" and can be influenced by factors other than the equipment's actual performance.

Opposing viewpoints

Objectivists attack Vacuum-tube amplifiers as vastly inferior because, in addition to their substantially higher total harmonic distortion, they require rebiasing, are less reliable, generate more heat, are less powerful, and are usually more expensive. Subjectivist believe that while tubed electronics are less linear than solid-state electronics at high-signal levels, they are much more linear at low-signal levels — less than one watt. Most musical signals spend most of the time at these low levels.

Objectivists claim that digital sound is superior to analog sound because it has no clicks, pops, wow, flutter, audio feedback, or rumble, has a higher signal-to-noise ratio, has a wider dynamic range, has less total harmonic distortion, and has a flatter and more extended frequency response. Subjectivists however claim that the process of converting a bit-stream to an analog waveform requires heavy filtering to remove spurious high-frequency information and that it should be expected that such filtering should involve some signal degradation and a large amount of phase shift in the passband. They point out that commonly-used consumer-grade digital-to-analog converters (DACs) exhibit very poor linearity at low levels. Both problems, at first dismissed, were then addressed by such solutions as digital filtering, oversampling, and the use of DACs operating at 20-bit (or higher) resolution. Musician Neil Young, for example, is a harsh critic of the sound of the original CD format but has approved of the sound of the newer SACD format with its greater safety margin between its ideal behavior and the requirements set by the limits of human hearing.

Objectivists consider total harmonic distortion to be an accurate measure of sound quality. Subjectivist however claim that total harmonic distortion has been proven by scientific testing to correlate poorly with perceived sound quality. The type of distortion is more significant. For instance, distortion by even harmonics has been shown to be less objectionable than distortion by odd harmonics.

Subjectivists believe that sound quality is degraded by large levels of negative feedback in amplifiers. Objectivists claim that negative feedback is beneficial to amplifier stability and produced good test results using steady-state waveforms. Subjectivists however believe that the application of negative feedback is inherently problematic for constantly-changing waveforms such as those that occur in music.

Subjectivists claim that there is a limit to what can be tested using Objective measurements. High-end audio companies which do rely on quantitative evaluations guard their measurement techniques as trade secrets. These are far more complex than the techniques which are in the public domain e.g. total harmonic distortion, transient intermodulation distortion. Subjectivist point out that objectivists since the 1970s no longer tout distortion measurements in their advertisements as there is a general consensus that an amplifier with 0.01% total harmonic distortion may not sound "better" than one with 0.1% total harmonic distortion - especially if the lower distortion is achieved with (excessive) feedback.

Overall, the subjectivists' world is looked upon by objectivists as being a hotbed of gullibility and fraud, its marketing engine driven primarily by either a constant desire for one-upmanship or a more benign desire to tinker with equipment. In particular, the tinkering drive is fed by wild claims for minor parts of the system such as cables. Objectivists, however, are often harshly dismissed by subjectivists as meter men — people who simply refuse to recognize what the subjectivists consider obvious. The debate is rather heated in certain quarters, and even James Randi chimed in on the issue.

Difficulty of testing

It is difficult, but very important, to match sound levels before comparing systems, as minute increases in loudness—more than 0.15 dB or 0.1 dB —have been demonstrated to cause perceived improvements in sound quality.

Listening tests are subjected to many variables, and results are notoriously unreliable. Thomas Edison, for example, showed that large audiences responded favorably when presented both live performances by artists and reproductions by his recording system, which today would be regarded as primitive in quality.

Similarly, results of component evaluation between various listeners or even the same listener under different circumstances cannot be easily replicated or standardized.

Similarly, the acoustic behavior of the listening room—the interaction between loudspeakers and the room's acoustics—and the interaction between an electromechanical device (loudspeaker) and an electronic device (amplifier) are subjected to many more variables than between electronic components. Thus the "difference" in sound quality between amplifiers is actually the ability of an amplifier to interface well with loudspeakers or a lucky combination of loudspeaker, amplifier, and room that works well together .

The introduction of switching apparatus, with either metal connection (mechanical switches) or electronic processing (solid-state switches), may, some believe, obscure the differences between the two signal sources being tested.

Chapter- 8

Audio Noise Measurement

Audio noise measurement is carried out when assessing the quality of audio equipment, such as is used in recording studios, broadcast studios, and in the home (Hi-Fi).

Noise in general refers to unwanted sound, often loud, but in audio systems it is the low-level hiss or buzz that intrudes on quiet passages that is of most interest. All recordings will contain some background noise that was picked up by microphones, such as the rumble of air conditioning, or the shuffling of an audience, but in addition to this every piece of equipment which the recorded signal subsequently passes through will add a certain amount of electronic noise, which ideally should be so low as to contribute insignificantly to what is heard.

Origins of noise - the need for Weighting

Microphones, amplifiers and recording systems all add some electronic noise to the signals passing through them, generally described as hum, buzz or hiss. All buildings have low-level magnetic and electrostatic fields in and around them emanating from mains supply wiring, and these can induce hum into signal paths, typically 50 Hz or 60 Hz (depending on the country's electrical supply standard) and lower harmonics. Shielded cables help to prevent this, and on professional equipment where longer interconnections are common, balanced signal connections (most often with XLR or TRS connectors) are usually employed. Hiss is the result of random signals, often arising from the random motion of electrons in transistors and other electronic components, or the random distribution of oxide particles on analog magnetic tape. It is predominantly heard at high frequencies, sounding like steam or compressed air.

Attempts to measure noise in audio equipment as RMS voltage, using a simple level meter or voltmeter, do not produce useful results; a special noise-measuring instrument is required. This is because noise contains energy spread over a wide range of frequencies and levels, and different sources of noise have different spectral content. For measurements to allow fair comparison of different systems they must be made using a measuring instrument that responds in a way that corresponds to how we hear sounds. From this, three requirements follow. Firstly, it is important that frequencies above or

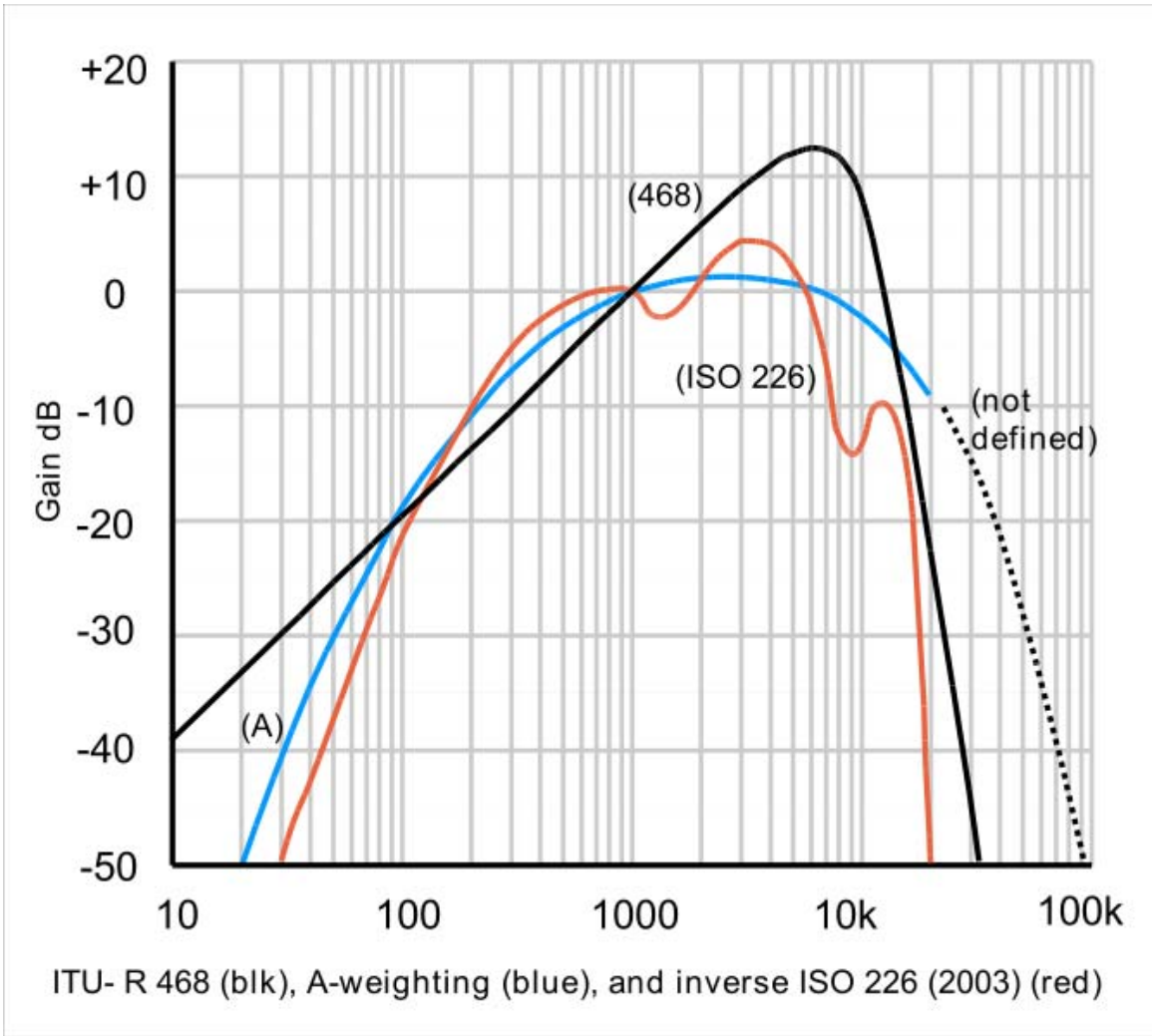
below those that can be heard by even the best ears are filtered out and ignored by bandwidth limiting (usually 22 Hz to 22 kHz). Secondly, the measuring instrument should give varying emphasis to different frequency components of the noise in the same way that our ears do, a process referred to as 'weighting'. Thirdly, the rectifier or detector that is used to convert the varying alternating noise signal into a steady positive representation of level should take time to respond fully to brief peaks to the same extent that our ears do; it should have the correct 'dynamics'.

The proper measurement of noise therefore requires the use of a specified method, with defined measurement bandwidth and weighting curve, and rectifier dynamics. The two main methods defined by current standards are **A-weighting** and **ITU-R 468**(formerly known as **CCIR weighting**).

A-weighting

A-weighting uses a weighting curve based on 'equal-loudness contours' that describe our hearing sensitivity to pure tones, but it turns out that the assumption that such contours would be valid for noise components was wrong. While the A-weighting curve peaks by about 2dB around 2 kHz, it turns out that our sensitivity to noise peaks by some 12dB at 6 kHz. Another weakness of A-weighting is that it is usually combined with an rms (root mean square) rectifier, which measures mean power, with no attempt made to account for proper hearing dynamics.

ITU-R 468 weighting



ITU- R 468 (blk), A-weighting (blue), and inverse ISO 226 (2003) (red)

When measurements started to be used in reviews of consumer equipment in the late 1960s it became apparent that they did not always correlate with what was heard. In particular, the introduction of Dolby B noise-reduction on cassette recorders was found to make them sound a full 10dB less noisy, yet they did not measure 10dB better. Various new methods were then devised, including one which used a harsher weighting filter and a quasi-peak rectifier, defined as part of the German DIN45 500 'Hi Fi' standard. This standard, no longer in use, attempted to lay down minimum performance requirements in all areas for 'High Fidelity' reproduction.

The introduction of FM radio, which also generates predominantly high-frequency hiss, also showed up the unsatisfactory nature of A-weighting, and the BBC Research Department undertook a research project to determine which of several weighting filter and rectifier characteristics gave results that were most in line with the judgment of panel of listeners, using a wide variety of different types of noise. BBC Research Department Report EL-17 formed the basis of what became known as CCIR recommendation 468, which specified both a new weighting curve and a quasi-peak rectifier. This became the standard of choice for broadcasters worldwide, and it was also adopted by Dolby, for

measurements on its noise-reduction systems which were rapidly becoming the standard in cinema sound, as well as in recording studios and the home.

Though they represent what we truly hear, ITU-R 468 noise weighting gives figures that are typically some 11dB worse than A-weighted, a fact that brought resistance from marketing departments reluctant to put worse specifications on their equipment than the public had been used to. Dolby tried to get round this by introducing a version of their own called CCIR-Dolby which incorporated a 6dB shift into the result (and a cheaper average reading rectifier), but this only confused matters, and was very much disapproved of by the CCIR.

With the demise of the CCIR, the 468 standard is now maintained as ITU-R 468, by the International Telecommunications Union, and forms part of many national and international standards, in particular by the IEC (International Electrotechnical commission), and the BSI (British Standards Institute). It is the only way to measure noise, that allows fair comparisons; and yet the flawed A-weighting has made a comeback in the consumer field recently, for the simple reason that it gives the lower figures that are considered more impressive by marketing departments.

Signal to noise ratio and Dynamic range

Audio equipment specifications tend to include the terms ‘signal to noise ratio’ and ‘dynamic range’, both of which have multiple definitions, sometimes treated as synonyms. The exact meaning must be specified along with the measurement.

Analog

Dynamic range used to mean the difference between maximum level and noise level, with maximum level defined as a clipping signal with a specified THD+N. The term has become corrupted by a tendency to refer to the dynamic range of CD players as meaning the noise level on a blank recording with no dither, (in other words, just the analog noise content at the output). This is not particularly useful; especially since many CD players incorporate automatic muting in the absence of signal.

Since the early 1990s various writers such as Julian Dunn have suggested that dynamic range be measured in the presence of a low-level test signal. Thus, any spurious signals caused by the test signal or distortion will not degrade the signal-to-noise ratio. This also addresses concerns about muting circuits.

Digital

In 1999, Dr. Steven Harris & Clif Sanchez Cirrus Logic published a white paper titled "Personal Computer Audio Quality Measurements" stating:

Dynamic Range is the ratio of the full scale signal level to the RMS noise floor, in the presence of signal, expressed in dB FS. This specification is given as an absolute number

and is sometimes referred to as Signal-to-Noise Ratio (SNR) in the presence of a signal. The label SNR should not be used due to industry confusion over the exact definition. DR can be measured using the THD+N measurement with a -60 dB FS signal. This low amplitude is small enough to minimize any large signal non-linearity, but large enough to ensure that the system under test is being exercised. Other test signal amplitudes may be used, provided that the signal level is such that no distortion components are generated.

In 2000 the AES released AES Information Document 6id-2000 which defined dynamic range as "20 times the logarithm of the ratio of the full-scale signal to the r.m.s. noise floor in the presence of signal, expressed in dB FS" with the following note:

This specification is sometimes referred to as signal-to-noise ratio (SNR) in the presence of a signal. The label SNR should not be used due to industry confusion over the exact definition. SNR is often used to indicate signal-to-noise ratio, with the noise level being measured with no signal. This can often give an optimistic result because of muting circuits, which mute the noise when no signal is present.

Chapter- 9

Audio Quality Measurement

Audio quality measurement seeks to quantify the various forms of corruption present in an audio system or device. The results of such measurement are used to maintain standards in broadcasting, to compile specifications, and to compare pieces of equipment.

The need for measurement

Measurement allows limits to be set and maintained for equipment and signal paths, and different pieces of equipment to be compared. While the issue of measurement is controversial, to the extent that Hi-Fi magazines these days tend to shun measurement in favour of listening tests, it is important to realise that audio quality measurement has in the past got a bad name by failing to produce results that correlated well with listening tests. This was because certain basic measurements were used, such as THD measurement, and A-weighted noise measurement, without any proper consideration of whether these related to subjective effects. The proper approach to measurement, which is largely adopted by broadcasters and other audio professionals, is to first devise measurements that can quantify the various forms of corruption in terms of subjective annoyance to a human listener, ideally the most critical listener based on tests using many suitably rested subjects. Once this is done, measurement has the advantage of not being dependent on a particular listener, or his state of hearing on a given day. It also has the advantage of being able to quantify corruption levels that would not be audible to even the most sensitive ear, which is important because a typical audio path from source to listener can involve many items of equipment, and just listening to each is not a guarantee that they will still sound acceptable when cascaded so that all their deficiencies add up.

A measures for testing audio quality for codecs is also given by the Mean Opinion Score.

Automated sequence testing

Sequence testing uses a specific sequence of test signals, for frequency response, noise, distortion etc, generated and measured automatically to carry out a complete quality check on a piece of equipment or signal path. A single 32-second sequence was standardised by the EBU in 1985, incorporating 13 tones (40 Hz–15 kHz at −12 dB) for frequency response measurement, two tones for distortion (1024 Hz/60 Hz at +9 dB) plus

crosstalk and compander tests. This sequence, which began with a 110-baud FSK signal for synchronising purposes, also became CCITT standard 0.33 in 1985.

Lindos Electronics expanded the concept, retaining the FSK concept, and inventing segmented sequence testing, which separated each test into a 'segment' starting with an identifying character transmitted as 110-baud FSK so that these could be regarded as 'building blocks' for a complete test suited to a particular situation. Regardless of the mix chosen, the FSK provides both identification and synchronisation for each segment, so that sequence tests sent over networks and even satellite links are automatically responded to by measuring equipment. Thus TUND represents a sequence made up of four segments which test the alignment level, frequency response, noise and distortion in less than a minute, with many other tests, such as Wow and flutter, Headroom, and Crosstalk also available in segments.

The Lindos sequence test system is now a 'de-facto' standard in broadcasting and many other areas of audio testing, with over 25 different segments recognised by Lindos test sets, and the EBU standard is no longer used.

Multitone testing

Another approach to automated testing uses a special multitone signal to assess all parameters simultaneously, by analysing the spectrum of the output from the device under test. It relies on the fact that with appropriate choice of frequencies, distortion components and noise can be made to appear between the tones, and measured using digital comb filtering. Even noise and wow and flutter can be extracted from the spectrum in principle.

In practice, though the use of a single brief test is attractive, and might even be used between programmes, this method presents several problems. Digital distortions produce a fine spectrum which can swamp the measurement of true noise in the absence of signal. The composite signal also has a high peak to mean ratio, with peak levels occurring whenever all the tones hit maximum simultaneously. Although the Probability density function can be controlled to some extent, it is not possible to separate out distortion at high level, from low level distortion. Quite high amounts of the former can be considered acceptable, but low level distortion is more critical.

Fast sequence tests are possible, and there have been attempts to make these appear like jingles for incorporation into broadcast programmes.

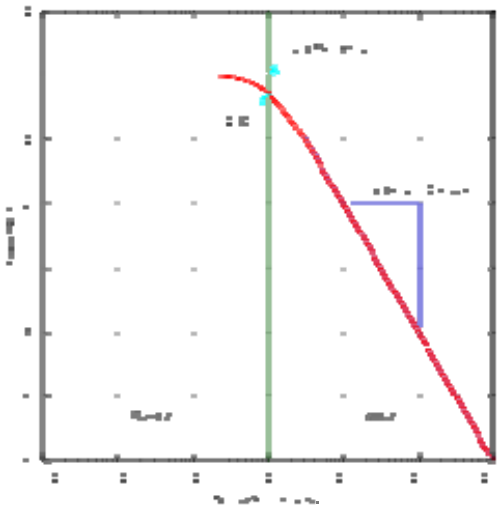
Measurements needed

Frequency response

Frequency response is the measure of any system's output spectrum in response to an input signal. In the audible range it is usually referred to in connection with electronic amplifiers, microphones and loudspeakers. Radio spectrum frequency response can refer to measurements of coaxial cables, category cables, video switchers and wireless communications devices. Subsonic frequency response measurements can include earthquakes and electroencephalography (brain waves).

Frequency response requirements differ depending on the application. In high fidelity audio, an amplifier requires a frequency response of at least 20–20,000 Hz, with a tolerance as tight as ±0.1 dB in the mid-range frequencies around 1000 Hz, however, in telephony, a frequency response of 400–4,000 Hz, with a tolerance of ±1 dB is sufficient for intelligibility of speech.

Frequency response curves are often used to indicate the accuracy of electronic components or systems. When a system or component reproduces all desired input signals with no emphasis or attenuation of a particular frequency band, the system or component is said to be "flat", or to have a flat frequency response curve.



Frequency response of a low pass filter with 6 dB per octave or 20 dB per decade

The frequency response is typically characterized by the *magnitude* of the system's response, measured in decibels (dB), and the *phase*, measured in radians, versus frequency. The frequency response of a system can be measured by applying a *test signal*, for example:

- applying an impulse to the system and measuring its response
- sweeping a constant-amplitude pure tone through the bandwidth of interest and measuring the output level and phase shift relative to the input
- applying a signal with a wide frequency spectrum (for example digitally-generated maximum length sequence noise, or analog filtered white noise

equivalent, like pink noise), and calculating the impulse response by deconvolution of this input signal and the output signal of the system.

These typical response measurements can be plotted in two ways: by plotting the magnitude and phase measurements to obtain a Bode plot or by plotting the imaginary part of the frequency response against the real part of the frequency response to obtain a Nyquist plot.

Once a frequency response has been measured (e.g., as an impulse response), providing the system is linear and time-invariant, its characteristic can be approximated with arbitrary accuracy by a digital filter. Similarly, if a system is demonstrated to have a poor frequency response, a digital or analog filter can be applied to the signals prior to their reproduction to compensate for these deficiencies.

Frequency response measurements can be used directly to quantify system performance and design control systems. However, frequency response analysis is not suggested if the system has slow dynamics.

Headroom (audio signal processing)

In digital and analog audio, **headroom** is the amount by which the signal-handling capabilities of an audio system exceed a designated level known as Permitted Maximum Level (PML). Headroom can be thought of as a safety zone allowing transient audio peaks to exceed the PML without exceeding the signal capabilities of an audio system (digital clipping, for example). Various standards bodies recommend various levels as Permitted Maximum Level.

Headroom in digital audio

In digital audio, headroom is defined as the amount by which digital full scale (FS) exceeds the permitted maximum level (PML) in dB (decibels). The European Broadcasting Union (EBU) specifies a PML of 9 dB below 0 dBFS (-9 dBFS), thus giving 9 dB of headroom. An alternative EBU recommendation allows 24 dB of headroom, which might be used for 24-bit master recordings where it is useful to allow more room for unexpected peaks during live recording.

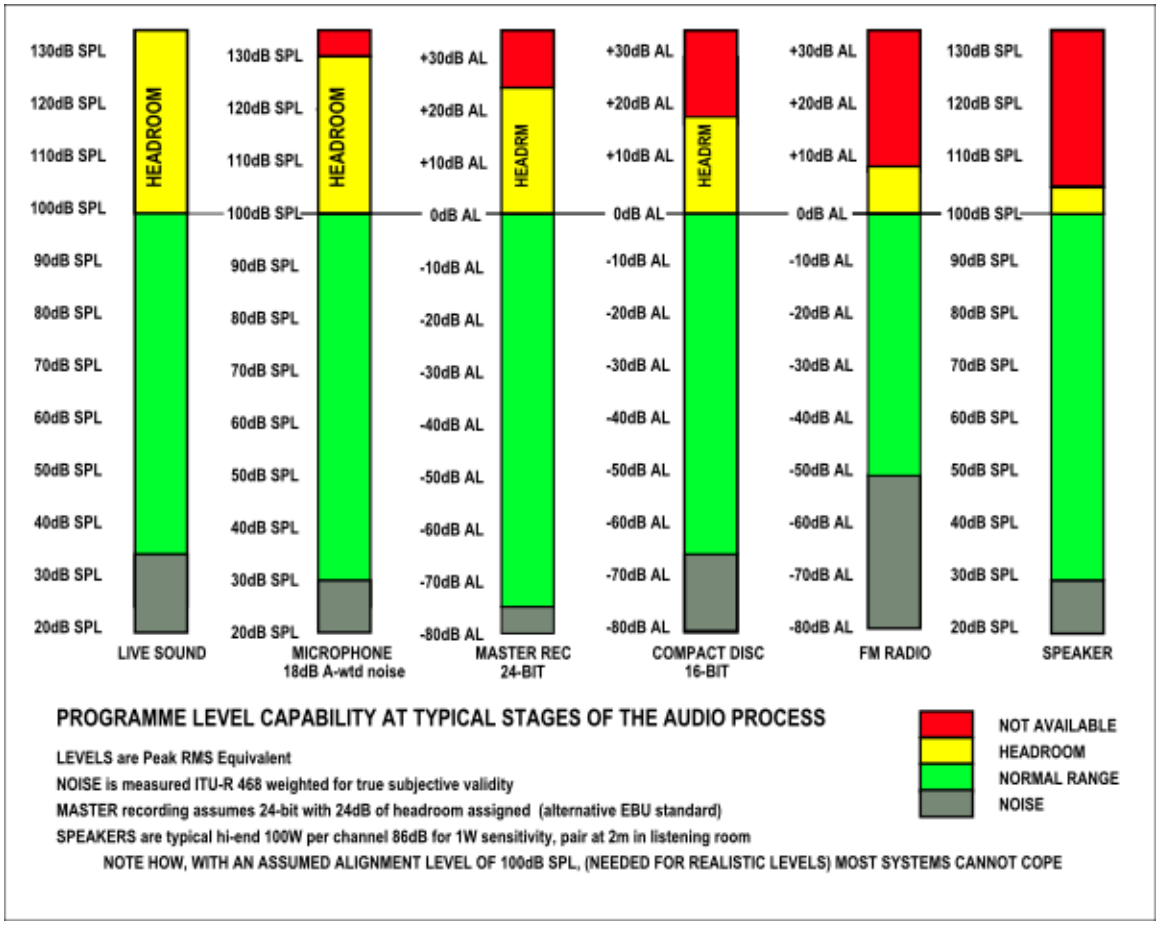
Failure to provide adequate headroom can bring about clipping of brief, higher-level transients.

Headroom in analog audio

In analog audio, headroom can mean low-level signal capabilities as well as for the amount of extra power reserve available within the power amplifiers that drive the loudspeakers.

Alignment level

Alignment level is an 'anchor' point, 9 db below the nominal level, a reference level which exists throughout the system or broadcast chain, though it may have different actual voltage levels at different points in the analog chain. Typically, nominal (not alignment) level is 0 dB, corresponding to an analog sine wave voltage of RMS voltage of 1.23 volts (+4 dBu or 3.47 volts peak to peak). In the digital realm, alignment level is −18 dBFS.



- AL = analog level
- SPL = sound pressure level

Crosstalk measurement

Crosstalk measurement is made on audio systems to determine the amount of signal leaking across from one channel to another.

Interchannel crosstalk applies between the two channels of a stereo system, and is usually not very important on modern systems, though it was hard to keep below the desired figure of -30dB or so on vinyl recordings and FM radio.

Crosstalk between channels in mixing consoles, and between studio feeds is much more of a problem, as these are likely to be carrying very different programmes or material.

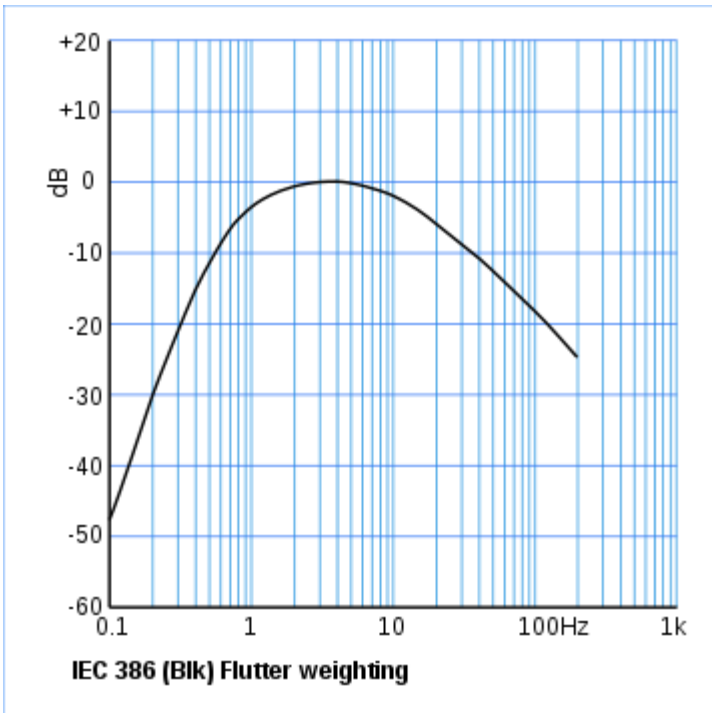
The IBA drew up a weighting curve for use in crosstalk measurement that gives due emphasis to the subjective audibility of different frequencies, as shown here. This is still in use, despite the demise of the IBA, and in the absence of any international standards is worth adopting.

Wow and flutter measurement

Wow and flutter measurement is carried out on audio tape machines, cassette recorders and players, and other analog recording and reproduction devices with rotary components (e.g. movie projectors, turntables (vinyl recording), etc.). This measurement quantifies the amount of 'frequency wobble' (caused by speed fluctuations) present in subjectively valid terms. Turntables tend to suffer mainly slow Wow. In digital systems, which are locked to crystal oscillators, wow and flutter are usually significantly more subtle, and are referred to as jitter.

While the terms Wow and Flutter used to be used separately (for wobbles at a rate below and above 4 Hz respectively), they tend to be combined now that universal standards exist for measurement which take both into account simultaneously. Listeners find flutter most objectionable when the actual frequency of wobble is 4 Hz, and less audible above and below this rate. This fact forms the basis for the weighting curve shown here. The weighting curve is misleading, inasmuch as it presumes inaudibility of flutters above 200 Hz, when actually faster flutters are quite damaging to the sound. A flutter of 200 Hz at a level of -50db will create 0.3% intermodulation distortion, which would be considered unacceptable in a preamp or amplifier.

Measurement techniques



IEC 386 (Blk) Flutter weighting

Measuring instruments use a frequency discriminator to translate the pitch variations of a recorded tone into a flutter waveform, which is then passed through the weighting filter, before being full-wave rectified to produce a slowly varying signal which drives a meter or recording device. The maximum meter indication should be read as the flutter value.

The following standards all specify the weighting filter shown above, together with a special slow-quasi-peak full-wave rectifier designed to register any brief speed excursions. As with many audio standards these are identical derivatives of a common specification.

- IEC 386
- DIN45507
- BS4847
- CCIR 409-3

Measurement is usually made on a 3.15 kHz (or sometimes 3 kHz) tone, a frequency chosen because it is high enough to give good resolution, but low enough not to be affected by drop-outs and high-frequency losses. Ideally, flutter should be measured using a pre-recorded tone free from flutter. Record-replay flutter will then be around twice as high, because worst case variations will add from time to time. When a recording is played back on the same machine as it was made on, a very slow change from low to high flutter will often be observed, because any cyclic flutter caused by capstan rotation may go from adding to cancelling as the tape slips slightly out of synchronism. A good technique is to stop the tape from time to time and start it again, as this will often result in different readings as the correlation between record and playback

flutter shifts. On top machines, it is not possible to use a tape made on a better machine, and so a record-playback test, using the stop-start technique, is the best that can be done.

Audible effects

Wow and flutter are particularly audible on music with oboe, string, guitar, flute. brass or piano solo playing. While wow is perceived clearly as pitch variation, flutter can alter the sound of the music differently, making it sound 'cracked' or 'ugly'. There is an interesting reason for this. A recorded 1 kHz tone with a small amount of flutter (around 0.1%) can sound fine in a 'dead' listening room, but in a reverberant room constant fluctuations will often be clearly heard. These are the result of the current tone 'beating' with its echo, which since it originated slightly earlier, has a slightly different pitch. What is heard is quite pronounced amplitude variation, which the ear is very sensitive to. This probably explains why piano notes sound 'cracked'. Because they start loud and then gradually tail off, piano notes leave an echo that can be as loud as the dying note that it beats with, resulting in a level that varies from complete cancellation to double-amplitude at a rate of a few Hz: instead of a smoothly dying note we hear a heavily modulated one. Oboe notes may be particularly affected because of their harmonic structure. Another way that flutter manifests is as a truncation of reverb tails. This may be due to the persistence of memory with regard to spatial location based on early reflections and comparison of Doppler effects over time. The auditory system may become distracted by pitch shifts in the reverberation of a signal that should be of fixed and solid pitch.

Equipment performance

- Professional tape machines can achieve a weighted flutter figure of 0.03%, which is considered inaudible, but for the fact that without weighting it would be an actual 0.3%.
- The best cassette decks struggle to manage around 0.08% weighted, which is still audible under some conditions. As an example, the Tascam 202MkIII Auto Reverse Cassette Deck reaches this 0.08% level.
- Average cassette decks and car players often have around 0.2% or more flutter.
- Digital music players such as CD, DAT or MP3 use electronic clocks to deliver samples at precisely the correct speed, and do not suffer from wow or flutter.
- The linear sound track on VCR video recorders has much higher wow and flutter than the VHS-HiFi high fidelity track which is contained within the video signal.
- Primitive phonographs which used idler wheels had very high wow and flutter, but high fidelity belt drive turntables were typically less than 0.2% by the 1970s, and the best direct drive turntables reached less than 0.05%.

The term 'flutter echo' is used in relation to a particular form of reverberation that flutters in amplitude. It has no direct connection with flutter as described here, though the mechanism of modulation through cancellation may have something in common with that described above.

Absolute speed

Absolute speed error causes a change in pitch, and it is useful to know that a semitone in music represents a 6% frequency change. This is because Western music uses the ‘equal temperament scale' based on a constant geometric ratio between twelve notes; and the twelfth root of 2 is 1.05946. Anyone with a good musical ear can detect a pitch change of around 1%, though an error of up to 3% is likely to go unnoticed, except by those few with ‘absolute pitch’. Most ‘movie’ films shown on UK television are sped up by 4.166% because they were shot at 24 frames per second, but are scanned at 25 frames per second to match the PAL standard of 25 frame/s 50 field/s. This causes a noticeable increase in pitch on voices, which often brings surprised comment from the actors themselves when they hear their performance on video. It can also frustrate attempts to play along with film music, which is closer to a semitone sharp than its intended pitch. Recently, digital pitch correction has been applied to some films, which corrects the pitch without altering lip-sync, by adding in extra cycles of sound. This has to be regarded as a form of distortion, as there is no way to change the pitch of a sound without also slowing it down that does not change the waveform itself.

Flutter correction

Novel DSP processes have been developed that correct wow and flutter by tracking various spuriae on the tape or film which can be re-purposed as timing references. Several recent (2006) DVD releases have utilized a system developed by Plangent Processes that substantially reduces wow and flutter of very high rates to extremely low levels, with a substantial improvement in quality, and without adding distortion or extra cycles of sound.

Scrape flutter

High-frequency flutter, above 100 Hz, can sometimes result from tape vibrating as it passes over a head, as a result of rapidly interacting stretching in the tape and striction at the head. This is termed 'scrape flutter'. It adds a roughness to the sound that is not typical of wow & flutter, and damping devices or heavy rollers are sometimes employed on professional tape machines to prevent it. Scrape flutter measurement requires special techniques, often using a 10 kHz tone.

Rumble (noise)

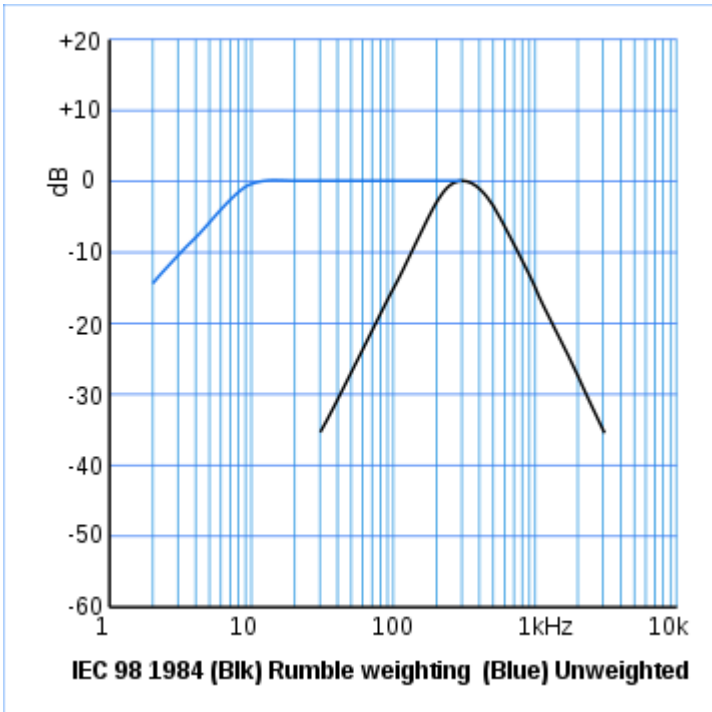
A **rumble** is a form of low-frequency noise created by a random sound wave existing between certain limitation points. In audio **rumble** refers to a low frequency sound from the bearings inside a turntable. This is most noticeable in low quality turntables with ball bearings. Higher quality turntables use slide bearings, minimizing rumble.

Some phono pre-amplifiers implement a rumble filter, in an attempt to remove the noise. A heavier platter can also help dampen this.

Rumble measurement is carried out on turntables (for vinyl recordings) which tend to generate very low frequency noise originating from the centre bearing and from drive pulleys or belts, as well as from irregularities in the record disc itself.

It can be heard as low-frequency noise and becomes a serious problem when playing records on audio systems with a good low-frequency response. Even when not audible rumble can cause intermodulation, modulating of the amplitude of other frequencies. The 'unweighted' response curve is intended for use in assessing the level of inaudible rumble with such intermodulation in mind.

Turntable design



One way to reduce rumble is to make the turntable very heavy, so that it acts as mechanical damper or low-pass filter, but even with the best turntables a lot of rumble tends to be generated by warped records or pressing irregularities sometimes visible as 'bobbles' in the surface. An important factor affecting rumble is low-frequency resonance resulting from pickup arm mass bouncing against stylus compliance. This resonance is usually in the 10–30 Hz region, and will increase rumble as well as reducing tracking ability if not well-damped. Good pickup arms incorporate viscous damping aimed at eliminating such resonance.

Rumble filters

Because these effects generate a mostly vertical component at the stylus, which corresponds to a difference signal in stereo reproduction, the incorporation of a high-pass filter operating only on the channel difference can be very effective in reducing rumble without loss of bass. Such a filter merges the two channels to mono at very low frequencies, which is not generally considered to have any effect on stereo perception, though it can change the sound balance (often for the better) by altering the way in which resonant room modes are stimulated (reducing corner to corner stimulation). The original circuit was designed in 1978 by Jeff Macaulay and featured as a circuit idea in *Wireless World*. Most so-called rumble filters work by simply rolling off the low-frequency response, which is detrimental to sound quality.

Though several standards exist that define how rumble should be measured, they all have a common basis, and use the weighting curves shown here. DIN 45539 (1971) and IEC98-1964 both cover rumble measurement. BS4852: Part 1 (1972) is specific in requiring that a slow rectifier be used, which shall reach 99% of its steady indication in 5s +-0.5s with not more than 10% overshoot.

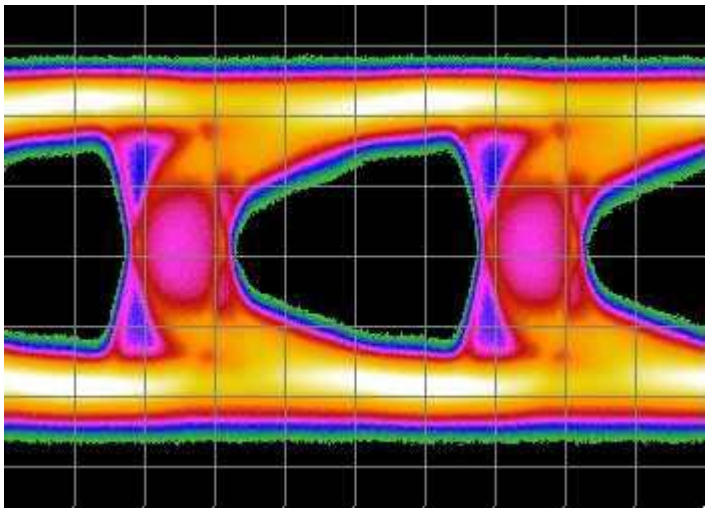
Jitter

Jitter in technical terms is the deviation in or displacement of some aspect of the pulses in a high-frequency digital signal. As the name suggests, jitter can be thought of as shaky pulses. The deviation can be in terms of amplitude, phase timing, or the width of the signal pulse. Another definition is that it is "the period frequency displacement of the signal from its ideal location." Among the causes of jitter are electromagnetic interference (EMI) and crosstalk with other signals. Jitter can cause a display monitor to flicker; affect the ability of the processor in a personal computer to perform as intended; introduce clicks or other undesired effects in audio signals, and loss of transmitted data between network devices. The amount of allowable jitter depends greatly on the application.

Jitter is the time variation of a periodic signal in electronics and telecommunications, often in relation to a reference clock source. Jitter may be observed in characteristics such as the frequency of successive pulses, the signal amplitude, or phase of periodic signals. Jitter is a significant, and usually undesired, factor in the design of almost all communications links (e.g., USB, PCI-e, SATA, OC-48). In clock recovery applications it is called *timing jitter*.

Jitter can be quantified in the same terms as all time-varying signals, e.g., RMS, or peak-to-peak displacement. Also like other time-varying signals, jitter can be expressed in terms of spectral density (frequency content).

Jitter period is the interval between two times of maximum effect (or minimum effect) of a signal characteristic that varies regularly with time. *Jitter frequency*, the more commonly quoted figure, is its inverse. Generally, very low jitter frequency is not of interest in designing systems, and the low-frequency cutoff for jitter is typically specified at 1 Hz.



In telecommunications circuit analysis an Eye diagram shows distortions caused by jitter.

Sampling jitter

In conversion between digital and analog signals, the sampling frequency is normally assumed to be constant. Samples should be converted at regular intervals. If there is jitter present on the clock signal to the analog-to-digital converter or a digital-to-analog converter then the instantaneous signal error introduced will be proportional to the slew rate of the desired signal and the absolute value of the clock error. Various effects can come about depending on the pattern of the jitter in relation to the signal. In some conditions, less than a nanosecond of jitter can reduce the effective bit resolution of a converter with a Nyquist frequency of 22 kHz to 14 bits.

This is a consideration in high-frequency signal conversion, or where the clock signal is especially prone to interference.

Packet jitter in computer networks

In the context of computer networks, the term *jitter* is often used as a measure of the variability over time of the packet latency across a network. A network with constant latency has no variation (or jitter). Packet jitter is expressed as an average of the deviation from the network mean latency. However, for this use, the term is imprecise. The standards-based term is *packet delay variation* (PDV). PDV is an important quality of service factor in assessment of network performance.

Seek jitter from compact discs

In the context of digital audio extraction from Compact Discs, **seek jitter** causes extracted audio samples to be doubled-up or skipped entirely if the Compact Disc drive re-seeks. The problem occurs because the Red Book (audio CD standard) does not

require block-accurate addressing during seeking. As a result, the extraction process may restart a few samples early or late, resulting in doubled or omitted samples. These glitches often sound like tiny repeating clicks during playback. A successful approach to correction in software involves performing overlapping reads and fitting the data to find overlaps at the edges. Most extraction programs perform seek jitter correction. CD manufacturers avoid seek jitter by extracting the entire disc in one continuous read operation using special CD drive models at slower speeds so the drive does not re-seek.

A *jitter meter* is a testing instrument for measuring clock jitter values, and is used in manufacturing DVD and CD-ROM discs.

Due to additional sector level addressing added in the Yellow Book (CD standard), CD-ROM data discs are not subject to seek jitter.

Phase jitter metrics

For clock jitter, there are three commonly used metrics: *absolute jitter*, *period jitter*, and *cycle to cycle jitter*.

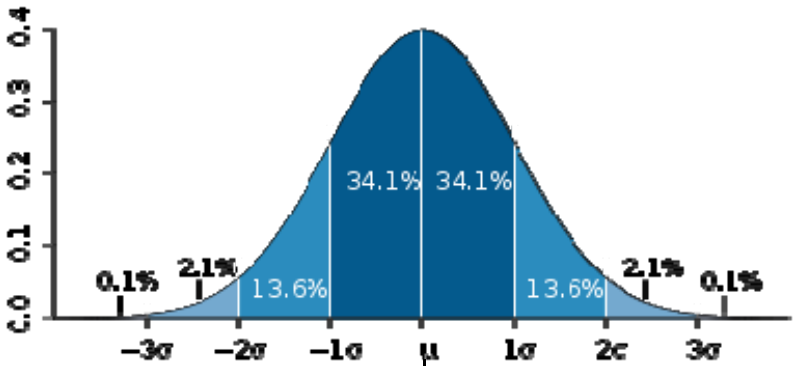
Absolute jitter is the absolute difference in the position of a clock's edge from where it would ideally be.

Period jitter (aka *cycle jitter*) is the difference between any one clock period and the ideal clock period. Accordingly, it can be thought of as the discrete-time derivative of absolute jitter. Period jitter tends to be important in synchronous circuitry like digital state machines where the error-free operation of the circuitry is limitted by the shortest possible clock period, and the performance of the circuitry is limitted by the average clock period. Hence, synchronous circuitry benefits from minimizing period jitter, so that the shortest clock period approaches the average clock period.

Cycle-to-cycle jitter is the difference in length of any two adjacent clock periods. Accordingly, it can be thought of as the discrete-time derivative of period jitter. It can be important for some types of clock generation circuitry used in microprocessors and RAM interfaces.

All of these jitter metrics are really measures of a single time-dependent quantity, and hence are related by derivatives as described above. Since they have different generation mechanisms, different circuit effects, and different measurement methodology, it is still useful to quantify them separately.

In telecommunications, the unit used for the above types of jitter is usually the *Unit Interval* (abbreviated *UI*) which quantifies the jitter in terms of a fraction of the ideal period of a bit. This unit is useful because it scales with clock frequency and thus allows relatively slow interconnects such as T1 to be compared to higher-speed internet backbone links such as OC-192. Absolute units such as *picoseconds* are more common in microprocessor applications. Units of *degrees* and *radians* are also used.



In the normal distribution one standard deviation from the mean (dark blue) accounts for about 68% of the set, while two standard deviations from the mean (medium and dark blue) account for about 95% and three standard deviations (light, medium, and dark blue) account for about 99.7%.

If jitter has a Gaussian distribution, it is usually quantified using the standard deviation of this distribution (aka. *RMS*). Often, jitter distribution is significantly non-Gaussian. This can occur if the jitter is caused by external sources such as power supply noise. In these cases, *peak-to-peak* measurements are more useful. Many efforts have been made to meaningfully quantify distributions that are neither Gaussian nor have meaningful peaks (which is the case in all real jitter). All have shortcomings but most tend to be good enough for the purposes of engineering work. Note that typically, the reference point for jitter is defined such that the *mean* jitter is 0.

In networking, in particular IP networks such as the Internet, jitter can refer to the variation (statistical dispersion) in the delay of the packets.

Types

Random jitter

Random Jitter, also called Gaussian jitter, is unpredictable electronic timing noise. Random jitter typically follows a Gaussian distribution or Normal distribution. It is believed to follow this pattern because most noise or jitter in a electrical circuit is caused by thermal noise, which does have a Gaussian distribution. Another reason for random jitter to have a distribution like this is due to the central limit theorem. The central limit theorem states that composite effect of many uncorrelated noise sources, regardless of the distributions, approaches a Gaussian distribution. One of the main differences between random and deterministic jitter is that deterministic jitter is bounded and random jitter is unbounded.

Deterministic jitter

Deterministic jitter is a type of clock timing jitter or data signal jitter that is predictable and reproducible. The peak-to-peak value of this jitter is bounded, and the bounds can easily be observed and predicted. Determistic jitter can either be correlated to the data stream (data-dependent jitter) or uncorrelated to the data stream (bounded uncorrelated jitter). Examples of data-dependent jitter duty-cycle dependent jitter (also known as duty-cycle distortion) and inter-symbol interference. One example of bounded uncorrelated jitter is Periodic jitter.

n	BER
6.4	10^{-10}
6.7	10^{-11}
7	10^{-12}
7.3	10^{-13}
7.6	10^{-14}

Total jitter

Total jitter (*T*) is the combination of random jitter (*R*) and deterministic jitter (*D*):

$T = D_{peak\text{-}to\text{-}peak} + 2\times n \times R_{rms}$,

in which the value of *n* is based on the bit error rate (BER) required of the link.

A common bit error rate used in communication standards such as Ethernet is 10^{-12}.

Testing

Testing for jitter and its measurement is of growing importance to electronics engineers because of increased clock frequencies in digital electronic circuitry to achieve higher device performance. Higher clock frequencies have commensurately smaller eye openings, and thus impose tighter tolerances on jitter. For example, modern computer motherboards have serial bus architectures with eye openings of 160 picoseconds or less. This is extremely small compared to parallel bus architectures with equivalent performance, which may have eye openings on the order of 1000 picoseconds.

Testing of device performance for jitter tolerance often involves the injection of jitter into electronic components with specialized test equipment.

Jitter is measured and evaluated in various ways depending on the type of circuitry under test. For example, jitter in serial bus architectures is measured by means of eye diagrams, according to industry accepted standards. A less direct approach—in which analog waveforms are digitized and the resulting data stream analyzed—is employed when measuring pixel jitter in frame grabbers. In all cases, the goal of jitter measurement is to verify that the jitter will not disrupt normal operation of the circuitry.

There are standards for jitter measurement in serial bus architectures. The standards cover jitter tolerance, jitter transfer function and jitter generation, with the required values for these attributes varying among different applications. Where applicable, compliant systems are required to conform to these standards.

Mitigation

Anti-jitter circuits

Anti-jitter circuits (AJCs) are a class of electronic circuits designed to reduce the level of jitter in a regular pulse signal. AJCs operate by re-timing the output pulses so they align more closely to an idealised pulse signal. They are widely used in clock and data recovery circuits in digital communications, as well as for data sampling systems such as the analog-to-digital converter and digital-to-analog converter. Examples of anti-jitter circuits include phase-locked loop and delay-locked loop. Inside digital to analog converters jitter causes unwanted high-frequency distortions. In this case it can be suppressed with high fidelity clock signal usage.

Jitter buffers

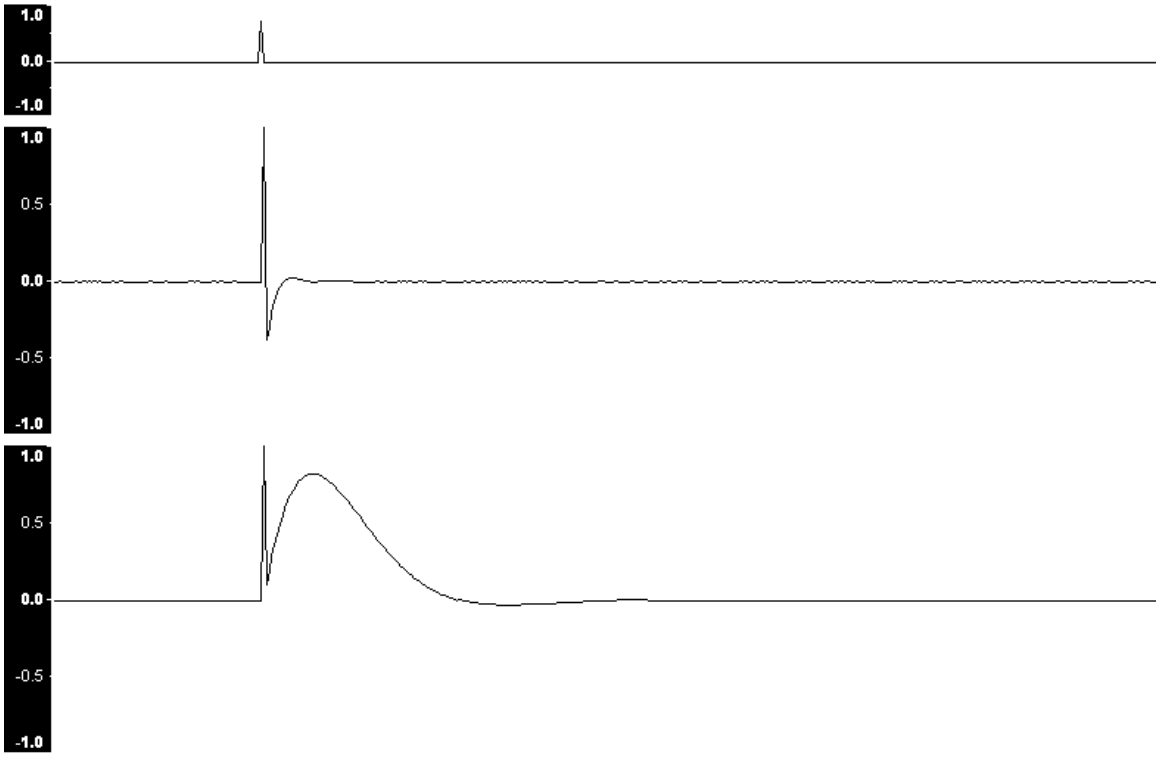
Jitter buffers or de-jitter buffers are used to counter jitter introduced by queuing in packet switched networks so that a continuous playout of audio (or video) transmitted over the network can be ensured. The maximum jitter that can be countered by a de-jitter buffer is equal to the buffering delay introduced before starting the play-out of the mediastream. In the context of packet-switched networks, the term *packet delay variation* is often preferred over *jitter*.

Some systems use sophisticated delay-optimal de-jitter buffers that are capable of adapting the buffering delay to changing network jitter characteristics. These are known as adaptive de-jitter buffers and the adaptation logic is based on the jitter estimates computed from the arrival characteristics of the media packets. Adaptive de-jittering involves introducing discontinuities in the media play-out, which may appear offensive to the listener or viewer. Adaptive de-jittering is usually carried out for audio play-outs that feature a VAD/DTX encoded audio, that allows the lengths of the silence periods to be adjusted, thus minimizing the perceptual impact of the adaptation.

Dejitterizer

A dejitterizer is a device that reduces jitter in a digital signal. A dejitterizer usually consists of an elastic buffer in which the signal is temporarily stored and then retransmitted at a rate based on the average rate of the incoming signal. A dejitterizer is usually ineffective in dealing with low-frequency jitter, such as waiting-time jitter.

Impulse response



The Impulse response from a simple audio system. Showing the original impulse, the response after high frequency boosting, and the response after low frequency boosting.

In signal processing, the **impulse response**, or **impulse response function (IRF)**, of a dynamic system is its output when presented with a brief input signal, called an impulse. More generally, an impulse response refers to the reaction of any dynamic system in response to some external change. In both cases, the impulse response describes the reaction of the system as a function of time (or possibly as a function of some other independent variable that parameterizes the dynamic behavior of the system).

For example, the dynamic system might be a planetary system in orbit around a star; the external influence in this case might be another massive object arriving from elsewhere in the galaxy; the impulse response is the change in the motion of the planetary system caused by interaction with the new object.

In all these cases, the 'dynamic system' and its 'impulse response' may refer to actual physical objects, or to a mathematical system of equations describing these objects.

Mathematical considerations

Mathematically, how the impulse is described depends on whether the system is modeled in discrete or continuous time. The impulse can be modeled as a Dirac delta function for continuous-time systems, or as the Kronecker delta for discrete-time systems. The Dirac

delta represents the limiting case of a pulse made very short in time while maintaining its area or integral (thus giving an infinitely high peak). While this is impossible in any real system, it is a useful idealisation. In Fourier analysis theory, such an impulse comprises equal portions of all possible excitation frequencies, which makes it a convenient test probe.

Any system in a large class known as *linear, time-invariant* (LTI) is completely characterized by its impulse response. That is, for any input function, the output function can be calculated in terms of the input and the impulse response. The impulse response of a linear transformation is the image of Dirac's delta function under the transformation, analogous to the fundamental solution of a partial differential operator.

The Laplace transform of the impulse response function is known as the transfer function. It is usually easier to analyze systems using transfer functions as opposed to impulse response functions. The Laplace transform of a system's output may be determined by the multiplication of the transfer function with the input function in the complex plane, also known as the frequency domain. An inverse Laplace transform of this result will yield the output function in the time domain.

To determine an output function directly in the time domain requires the convolution of the input function with the impulse response function. This requires the use of integrals, and is usually more difficult than simply multiplying two functions in the frequency domain.

The impulse response, considered as a Green's function, can be thought of as an "influence function:" how a point of input influences output.

Practical applications

In practical systems, it is not possible to produce a perfect impulse to serve as input for testing; therefore, a brief pulse is sometimes used as an approximation of an impulse. Provided that the pulse is short enough compared to the impulse response, the result will be close to the true, theoretical, impulse response. In many systems, however, driving with a very short strong pulse may drive the system into a nonlinear regime, so instead the system is driven with a pseudo-random sequence, and the impulse response is computed from the input and output signals.

Loudspeakers

An application that demonstrates this idea was the development of impulse response loudspeaker testing in the 1970s. Loudspeakers suffer from phase inaccuracy, a defect unlike other measured properties such as frequency response. Phase inaccuracy is caused by small delayed sounds that are the result of resonance, energy storage in the cone, the internal volume, or the enclosure panels vibrating. Measuring the impulse response, which is a direct plot of this "time-smearing," provided a tool for use in reducing resonances by the use of improved materials for cones and enclosures, as well as changes

to the speaker crossover. The need to limit input amplitude to maintain the linearity of the system led to the use of inputs such as pseudo-random maximum length sequences, and to the use of computer processing to derive the impulse response.

Digital filtering

Impulse response is a very important concept in the design of digital filters for audio processing, because digital filters can differ from 'real' filters in often having a pre-echo, which the ear is not accustomed to.

Electronic processing

Impulse response analysis is a major facet of radar, ultrasound imaging, and many areas of digital signal processing. An interesting example would be broadband internet connections. DSL/Broadband services use adaptive equalisation techniques to help compensate for signal distortion and interference introduced by the copper phone lines used to deliver the service.

Control systems

In control theory the impulse response is the response of a system to a Dirac delta input. This proves useful in the analysis of dynamic systems: the Laplace transform of the delta function is 1, so the impulse response is equivalent to the inverse Laplace transform of the system's transfer function.

Acoustic and audio applications

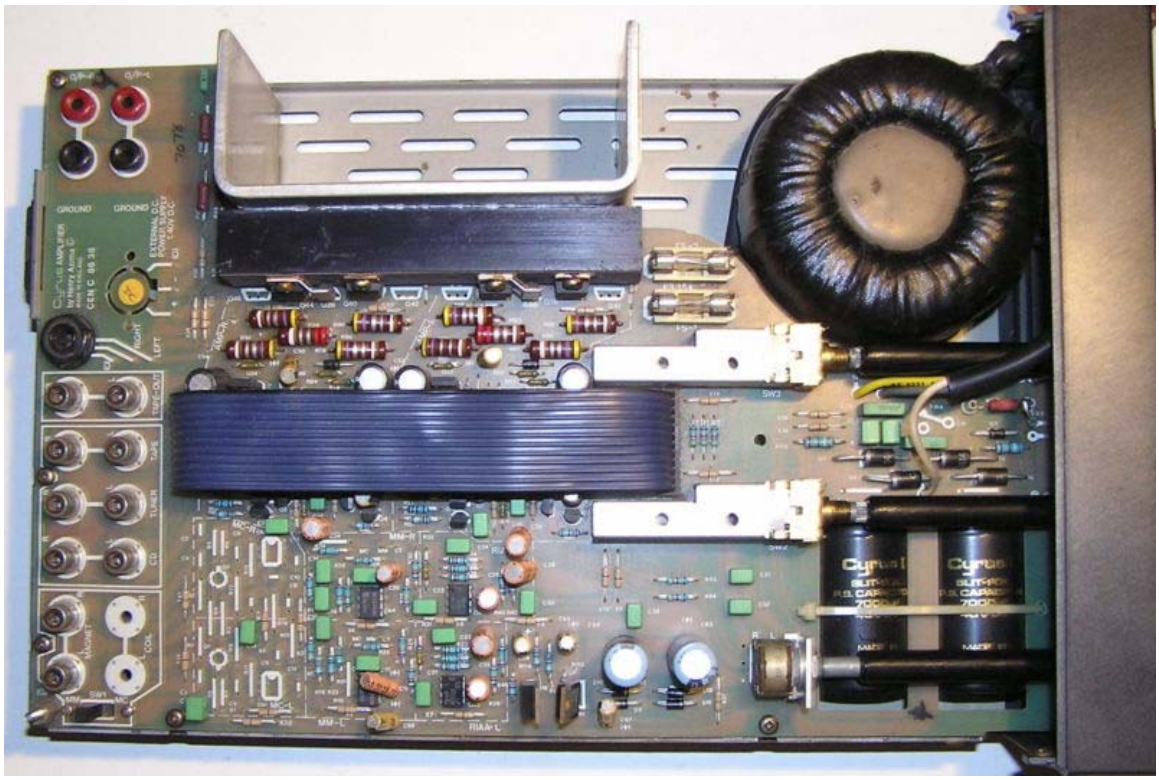
In acoustic and audio applications, impulse responses enable the acoustic characteristics of a location, such as a concert hall, to be captured. Various commercial packages are available containing impulse responses from specific locations, ranging from small rooms to large concert halls. These impulse responses can then be utilized in convolution reverb applications to enable the acoustic characteristics of a particular location to be applied to target audio.

Economics

In economics, and especially in contemporary macroeconomic modeling, impulse response functions describe how the economy reacts over time to exogenous impulses, which economists usually call 'shocks', and are often modeled in the context of a vector autoregression. Impulses that are often treated as exogenous from a macroeconomic point of view include changes in government spending, tax rates, and other fiscal policy parameters; changes in the monetary base or other monetary policy parameters; changes in productivity or other technological parameters; and changes in preferences, such as the degree of impatience. Impulse response functions describe the reaction of endogenous macroeconomic variables such as output, consumption, investment, and employment at the time of the shock and over subsequent points in time.

Chapter- 10

Audio Amplifier



Mission Cyrus 1 Hi Fi integrated audio amplifier

An **audio amplifier** is an electronic amplifier that amplifies low-power audio signals (signals composed primarily of frequencies between 20 - 20 000 Hz, the human range of hearing) to a level suitable for driving loudspeakers and is the final stage in a typical audio playback chain.

The preceding stages in such a chain are low power audio amplifiers which perform tasks like pre-amplification, equalization, tone control, mixing/effects, or audio sources like record players, CD players, and cassette players. Most audio amplifiers require these low-level inputs to adhere to line levels.

While the input signal to an audio amplifier may measure only a few hundred microwatts, its output may be tens, hundreds, or thousands of watts.

History

Three audio amplifiers

The audio amplifier was invented in 1909 by Lee De Forest when he invented the triode vacuum tube. The triode was a three terminal device with a control grid that can modulate the flow of electrons from the filament to the plate. The triode vacuum amplifier was used to make the first AM radio.

Early audio amplifiers were based on vacuum tubes (also known as *valves*), and some of these achieved notably high quality (e.g., the Williamson amplifier of 1947-9). Most modern audio amplifiers are based on solid state devices (transistors such as BJTs, FETs and MOSFETs), but there are still some who prefer tube-based amplifiers, due to a perceived 'warmer' valve sound. Audio amplifiers based on transistors became practical with the wide availability of inexpensive transistors in the late 1960s.

Design parameters

Key design parameters for audio amplifiers are frequency response, gain, noise, and distortion. These are interdependent; increasing gain often leads to undesirable increases in noise and distortion. While negative feedback actually reduces the gain, it also reduces distortion. Most audio amplifiers are linear amplifiers operating in class AB.

Filters and preamplifiers

Historically, the majority of commercial audio preamplifiers made had complex filter circuits for equalization and tone adjustment, due to the far from ideal quality of recordings, playback technology, and speakers of the day.

Using today's high quality (often digital) source material, speakers, etc., such filter circuits are usually not needed. Audiophiles generally agree that filter circuits are to be avoided wherever possible. Today's audiophile amplifiers do not have tone controls or filters.

Since modern digital devices, including CD and DVD players, radio receivers and tape decks already provide a "flat" signal at line level, the preamp. is not needed other than as volume control. One alternative to a separate preamp is to simply use passive volume and switching controls, sometimes integrated into a power amp to form an "integrated" amplifier.

Further developments in amplifier design

For some years following the introduction of solid state amplifiers, their perceived sound did not have the excellent audio quality of the best valve amplifiers. This led audiophiles to believe that valve sound had an intrinsic quality due to the vacuum tube technology itself. In 1972, Matti Otala demonstrated the origin of a previously unobserved form of distortion: transitory intermodulation distortion (TIM), also called slew rate distortion. TIM distortion was found to occur during very rapid increases in amplifier output voltage. TIM did not appear at steady state sine tone measurements, helping to hide it from design engineers prior to 1972. Problems with TIM distortion stem from reduced open loop frequency response of solid state amplifiers. Further works of Otala and other authors found the solution for TIM distortion, including increasing slew rate, decreasing preamp frequency bandwidth, and the insertion of a lag compensation circuit in the input stage of the amplifier. In high quality modern amplifiers the open loop response is at

least 20 kHz, canceling TIM distortion. However, TIM distortion is still present in most low price home quality amplifiers.

The next step in advanced design was the Baxandall Theorem, created by Peter Baxandall in England. This theorem introduced the concept of comparing the ratio between the input distortion and the output distortion of an audio amplifier. This new idea helped audio design engineers to better evaluate the distortion processes within an audio amplifier.

Applications

Important applications include public address systems, theatrical and concert sound reinforcement, and domestic sound systems. The sound card in a personal computer contains several audio amplifiers (depending on number of channels), as does every stereo or home-theatre system.

Chapter- 11

Audio Multicore Cable

A multicore cable "snake" helps sound engineers to route a number of signals without having to have a tangled mess of individual cables

Used in the audio recording and sound reinforcement fields, an **audio multicore cable** (most commonly known as a **snake cable** or just a **snake**) is a compact cable, typically about the diameter of a coin, which contains from 4 to 56 individual shielded pair microphone cables all housed by one rugged, heavy-duty common outer jacket. Each end of the multicore cable terminates in a "tail", which contains either a patchbay for female XLR or 1/4" jacks or male plugs.

Cable construction

The inner microphone cables are each a pair of insulated, twisted-pair, multi-strand wires, surrounded by shielding made of foil or tightly-braided wire mesh. The cable may be plenum rated making it strong but difficult to handle and roll. More expensive cables use finer wire and wire braid shield rather than foil and the outer jacket is soft making it flexible and easy to roll. A cotton filler runs down the middle of the cable and helps it to maintain its shape. The cable is normally rolled in an over-under manner such that when pulled out of the packer it won't have developed twists.

Different cable manufacturers use different methods of identification for the shielded pairs of cable. Belden have a sequenced color code and a number for each of the conductors in their products of up to 52 pair cables, for their plenum cables each pair is covered with their patended Belfoil shield, that is only conductive on the inner surface. For their portable cables they use a French braid. Canare, Mogami, and GEPCO mark numbers on the PVC insulation of the individual pairs.

Composite multicores

Composite multicores combine different types of signals in the one cable. They may contain coaxial cores for video , twisted pair for data or low voltage cores for mains power. Composite multicores are usually used to connect video cameras, but they are now gaining usage in live sound with the introduction of the Yamaha PM1D which uses a composite cable to connect it to the stage box.

Analog multicore

The multicore cable runs from the stage box or microphone splitter and then to the front-of-house sound desk or mixing console. Permanent installations have stage boxes mounted in the floor or side of stage and the plenum cable runs through the ceiling or false floor to the console, located either in the auditorium or the bio-box.

For temporary shows the stage box is placed at the side or rear of the stage and generally 75% of the connectors are female XLR with the balance male XLR. The male XLR are used as returns to the stage for foldback (on-stage monitoring) or FOH at line level. Larger multicores have male and female connectors for each channel at the stage box end giving it more flexibility. In this system the male XLRs may be used to give a split to the foldback desk. The foldback desk is connected by a cable with tails at both ends. The console end always has the opposite sex XLR connector. Larger shows tend to use a system with subsnakes which plug into the main stage box so one can use shorter mic leads on stage and have less clutter.

Balanced lines

The output of most microphones is balanced to reduce the pickup of electrical noise, and therefore requires two signal wires and an earth per channel. This necessitates the use of three-pin connectors . To reduce noise, the shields of each channel should be isolated from the other channels. If "phone jacks" are used instead of XLRs, TRS connectors

should be used so that balanced signals can be maintained. A TRS connector plugged into a TS socket will ground the ring and maintain the positive signal integrity and work in a pseudo-balanced state. The use of TS connectors forces the system into an unbalanced state at all times and should be avoided.

Tails

The end of the multicore where the channels fanout is called the tail and it generally goes at the mixing desk end. Tails used for patching may have a mix of male or female XLRs, or TRS connectors on each end.

Breakout cables

An audio multicore may also function as a breakout cable, if it has a compound connector on one end and component connectors on the other. This is more common in short multicores meant for in-studio connections, such as audio engines, analog to digital converters or digital mixing consoles . Multicore cables may also connect to either the front or back of patch panels, when the patch panel is used as an access point or breakout box for connecting external inputs and outputs.

Multipin multicores

The ideal system uses multipins to connect the subsnakes to the main stage box (which may be rack mounted) and multipins to the main cores. This allows the system to be expanded by channel count or length. The multipin connectors are based on the MIL C5015 standard and are made by companies like Link in Italy or Amphenol in the United States. Alternative connectors like EDAC and Burndy are also used but lack the reliability of the alloy mil-spec style connectors. Examples include;

Multicore	MIL C5015 Multipin	Sends/Returns
8 core	25 pole	8/0
12 core	37 pole	8/4
16 core	54 pole	12/4
24 core	85 pole	20/4

While the sound reinforcement industry generally uses proprietary multicores depending on the application, the television industry has standardized on a 12-channel snake with a common 36-pin connector on each end known as DT-12. DT-12 snakes are commonly built into sports venues and run from where the TV truck is parked to areas such as the press box or playing field where audio is required by the TV production. The truck can then simply run a DT-12 patch snake between the truck and the house cabling, with actual XLR connectors then only being required at the very end of the snake in order to plug in microphones.

Snake cables can be replaced by using audio over Ethernet, and similar multichannel digital audio technologies such as AES10 (MADI).

Digital multicores

State of the art systems can now use digital multicore systems whereby the audio signal is encoded to digital using analog-to-digital converters, the channels are bundled together to be transported on a single wire to a destination then re-converted back to analog using D-to-As. Such systems are branded under names such CobraNet or EtherSound (respectively manufactured by Peak Technologies and Digigram) and licenced to many companies. Sometimes all that is run to the mix positions is control and all audio processing occurs in the Mix Box, or Stage Box.

The main problem with digital multicores is latency, the time taken to encode into digital and then back into analog at the mixing console. Most stand alone digital consoles take their inputs in analog format and the cumulative latency is too great to go unnoticed. The second problem is cost, it is currently much more expensive to use a digital multicore for small systems. Most multicore systems use Ethernet wiring as the physical medium, but the system must be as close to synchronous as possible and cannot share the same cable as data systems due to the risk of collisions.

MADI

Multichannel Audio Digital Interface, **MADI** or **AES10** is an industry-standard electronic communications protocol that defines the data format and electrical characteristics of an interface carrying multiple channels of digital audio. The Audio Engineering Society (AES) standard for MADI was originally documented in AES10-1991, which was updated by AES10-2003. The MADI standard includes a bit-level description and has features in common with the two-channel format of AES3. Serial digital transmission over coaxial cable or fibre-optic lines of 28, 56, or 64 channels is supported, with sampling rates of up to 96 kHz and resolution of up to 24 bits per channel.

Transmission format

MADI links use a transmission format that is similar to the Fiber Distributed Data Interface (FDDI) networking technology (ISO 9314), which was popular in the mid-'90s for backbone links between LAN segments. Since MADI is most often transmitted on copper links via 75 ohm coaxial cables, it is more closely related to the FDDI specification for copper-based links, called CDDI.

AES10-2003 recommends using BNC connectors with coaxial cables and ST1 connectors with optic fibres. The specifications about fibres can provide a range of up to 2 km.

The basic data rate is 100 Mbit/s of data using 4B5B encoding to produce a 125 MHz physical baud rate. This clock is not synchronized to the audio sample rate, and the audio data payload is padded using "JK" sync symbols.

The audio data is almost identical to the AES3 payload, although with more channels. Rather than letters, they are assigned numbers from 0–55 or 0–63. The only difference is that frame synchronization is provided by sync symbols outside the data itself, rather than an embedded preamble sequence, and the first 4 time slots of each subchannel are encoded as normal data, used for subchannel identification:

- Bit 0: Set to 1 to mark channel 0, the first channel in each frame.
- Bit 1: Set to 1 to indicate that this channel is active (contains interesting data).
- Bit 2: notA/B channel marker, used to mark left (0) and right (1) channels. Generally, even channels are A and odd channels are B.
- Bit 3: Set to 1 to mark the beginning of a 192-sample data block.

Sync symbols may be inserted at any subframe boundary, and must occur at least once per frame (0.45% minimum overhead.)

Sampling frequency

The original AES10-1991 specification allowed 56 channels at sample rates from 32 to 48 kHz with a tolerance ±12.5%, in part 4.1. It leads to a total range of 28 to 54 kHz. This produced a total of 56×32×54 = 96768 kbit/s, leaving 3.232% of the channel for synchronization marks and transmit clock error.

The 2003 revision specifies different relations between sampling frequency and number of channels. The next list is from the part 5.1 of the AES10-2003 specifications:

- 32 kHz to 48 kHz ± 12,5 %, 56 channels;
- 32 kHz to 48 kHz nominal, 64 channels;
- 64 kHz to 96 kHz ± 12,5 %, 28 channels.

Both versions say that higher sampling frequencies may be accommodated (for example, 96 kHz or 192 kHz) by using two or more channels per audio sample on the link.

Improvements

The original specification (AES10-1991) defined the MADI link as a 56 channel transport for the purpose of linking large-format mixing consoles to digital multi-track recording devices. Large broadcast studios adopted it for use routing multi-channel audio throughout their facilities as well. The 2003 revision, called AES10-2003, adds a 64 channel capability as well as support for "double-rate" sampling at 96 kHz by removing vari-speed operation.

The latest AES10-2008 standard includes minor clarifications and updates to correspond to the current AES3 standard.

Usage

MADI was developed by Solid State Logic, Sony and Studer and is widely used in the audio industry, especially in the professional sector. Its advantages over other audio digital interface protocols and standards such as AES/EBU (AES3), ADAT (Alesis Digital Audio Tape), TDIF (Tascam Digital Interface) and S/PDIF (Sony/Philips Digital Interconnect Format) are: first, support of a greater number of channels per line; and second, the use of coaxial and optical fibre media that enable the transmission of audio signals over 100 meters and up to 3000 meters. Some main providers of interfaces and computer cards for MADI are:

- Allen & Heath
- AMS Neve
- Avid Technology
- Calrec Audio
- Deubner Hoffmann Digital
- DiGiCo
- DirectOut
- Euphonix
- Evertz
- Fairlight
- Harrison Audio Consoles
- Innovason
- Klotz Digital
- Lab X Technologies
- Lawo
- Lynx Studio Technology, Inc.
- Merging Technologies
- RME
- SADiE
- Solid State Logic
- Soundcraft
- Studer
- Sydec Audio
- Yamaha Commercial Audio

CobraNet

CobraNet is a combination of software, hardware and network protocols designed to deliver uncompressed, multi-channel, low-latency digital audio over a standard Ethernet network. Developed in the 1990s, CobraNet is widely regarded as the first commercially successful implementation of audio over Ethernet.

CobraNet was designed for and is primarily used in large audio installations such as convention centers, stadiums, airports, theme parks and concert halls. It is most useful in applications where a large number of audio channels must be transmitted over long distances or to multiple locations.

CobraNet is an attractive alternative to analog audio, which suffers from signal degradation over long cable runs due to electromagnetic interference, high-frequency attenuation, and voltage drop. Additionally, the use of digital multiplexing allows audio to be transmitted using much less cabling than analog audio.

History

CobraNet was developed in 1996 by Boulder, Colorado-based Peak Audio. Initial demonstrations were of a 10 Mbit/s point-to-point system with limited channel capacity. The first permanent installation of CobraNet in this early form was to provide background music throughout the Animal Kingdom theme park.

CobraNet was first introduced as an interoperable standard in collaboration with manufacturer QSC Audio Products. QSC was the first to license the technology from Peak Audio and marketed it under the RAVE brand. At this point CobraNet had graduated to Fast Ethernet and used a patented collision avoidance technique to carry up to 64 channels per Ethernet collision domain.

The first commercial use of CobraNet as an interoperable standard was during the half-time show at Super Bowl XXXI in 1997. CobraNet was subsequently enhanced to support and eventually require a switched Ethernet network. An SNMP agent was added for remote control and monitoring. Support for higher sample rates, increased bit resolutions and lowered latency capabilities were later introduced in an incremental and backwards-compatible manner.

In May 2001, Cirrus Logic announced that it had acquired the assets of Peak Audio. Leveraging Cirrus DSP technology, a low-cost SoC implementation of CobraNet was developed and marketed. CobraNet has been widely licensed by commercial audio equipment manufacturers and installed in thousands of facilities worldwide.

Advantages and disadvantages

Advantages

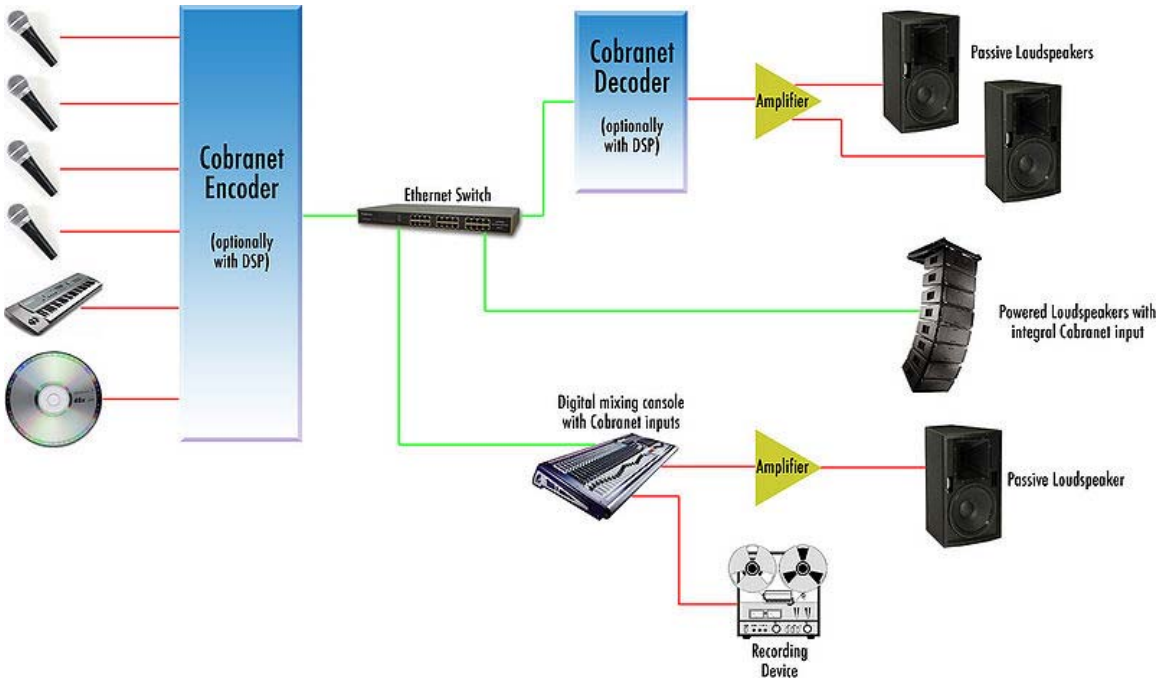
- **Cabling cost** – using CobraNet and fast Ethernet, 64 channels of uncompressed digital audio are carried through a single, inexpensive Cat-5 cable. In the analog world, this would have required 64 separate analog audio cables, each of which cost the same or more than the Cat-5 cable. Using gigabit and/or fiber optic Ethernet variants, the cost of cabling per audio channel is further reduced compared to the fast Ethernet connections. Also, since CobraNet data can coexist

with data traffic over existing Ethernet networks, a single network infrastructure can economically serve both audio distribution and other networking needs for a facility.
- **Flexibility** – a well-designed network provides enhanced flexibility for future changes to the system. For instance, audio routing can be changed on the fly with network commands, and do not require any rewiring.
- **Reliability** – use of Ethernet by CobraNet affords many High Availability features such as Spanning Tree Protocol, Link Aggregation and Network Management. For critical applications, CobraNet devices can be wired with a redundant link. In the case that one CobraNet device, cable, or Ethernet switch fails, the other takes over almost immediately.
- **Audio quality** – audio is transmitted in digital form, and therefore it enjoys the same benefits that any other digital medium does. These benefits include reduced susceptibility to electromagnetic interference, crosstalk, coloration and attenuation due to cable impedance.

Disadvantages

- **Latency** – delays over the CobraNet transmission medium itself are at least 1 ⅓ milliseconds per network traversal. For some applications, these delays can be unacceptable especially when combined with further delays resulting from propagation time, digital signal processing and the conversions between analog and digital.
- **Hardware cost** – although significant money is usually saved in cabling, at least part of that money is spent on the required CobraNet devices which encode and decode the CobraNet signal.

Transmission



Simple block diagram of an audio system employing CobraNet technology. Red lines indicate analog audio signals, while green lines indicate standard Ethernet signals.

Ethernet

CobraNet is transmitted using standard Ethernet packets. Instead of using TCP/IP packets, CobraNet transfers data using data link layer packets, which travel quickly through hubs, bridges and switches, and are not as susceptible to the latency and QoS problems commonly found in streaming protocols using a higher transport layer. However, since CobraNet does not use an IP protocol, its packets cannot travel through routers, and therefore it is limited to use on a LAN; CobraNet cannot be used over the Internet. The network over which CobraNet is transmitted must be able to operate at a minimum of 100 Mbit/s (also known as Fast Ethernet). All CobraNet packets are identified with a unique Ethernet protocol identifier (0x8819) assigned to Cirrus Logic.

While CobraNet has been shown to function properly over wireless networks under ideal conditions, bandwidth and reliability issues associated with typical 802.11 wireless networks tend to cause frequent dropouts and fatal errors. However, wireless communication of CobraNet data can be reliably accomplished using lasers.

Channels and bundles

CobraNet data is organized into channels and bundles. A typical CobraNet signal can contain up to 4 bundles of audio travelling in each direction, for a total of 8 bundles per device. Each bundle houses up to 8 channels of 48 kHz, 20-bit audio, for a total capacity of 64 channels. CobraNet is somewhat scalable, in that channel capacity increases when

16-bit audio is used, and channel capacity decreases when 24-bit audio is used. Specific channel capacity is defined by the 1,500-byte Ethernet payload limit.

There are three types of bundles: multicast, unicast, and private:

- **Multicast bundles** are "broadcast" from one CobraNet device to all other CobraNet devices in the network using Ethernet multicast addressing. Each CobraNet device individually determines if it will use the bundle or discard it. Therefore, multicast bundles are more bandwidth-intensive than other bundle types. Bundle numbers 1–255 are reserved for multicast bundles.
- **Unicast bundles** are sent from one CobraNet device to any other device or devices configured to receive the bundle number. Unicast bundles are much more efficient because they attempt to travel only to devices which actually want to receive them. Despite their name, unicast bundles may still be sent to multiple devices, either by transmitting multiple copies of the audio data or using multicast addressing. Bundle numbers 256–65279 are reserved for unicast bundles.

- **Private bundles** may be sent with unicast or multicast addressing. Bundle numbers 65280–65535 are reserved for private bundles. Private bundle numbers are associated with the MAC address of the device that transmits them. To receive a private bundle, both the bundle number and the MAC address of the transmitter must be specified. Because private bundles are associated with the transmitters there is no hard limit on the number private bundles.

As long as multicast bundles are used sparingly, it is virtually impossible to exceed the bandwidth of a 100 Mbit network with CobraNet data. However, there are limitations to the maximum number of bundles that can be sent on a network, since the conductor must include data in its beat packets for every bundle on the network, and the beat packet is limited to 1,500 bytes. If each device is transmitting one bundle, there may be up to 184 transmitters active simultaneously (for a total of 184 bundles). If each device is transmitting a full four bundles, then only 105 transmitters could be active, although they would be producing a total of 421 active bundles. The use of private bundles does not require any additional data in the beat packet, so these network limitations can be sidestepped by using private bundles.

Synchronization

The CobraNet network is synchronized to a single CobraNet device known as the **conductor**. A **conductor priority** can be configured to influence selection of the conductor. Among devices with the same conductor priority configuration, the first to establish itself on the network becomes is elected conductor. All other devices are known as **performers**. In the event that the conductor fails, another CobraNet device will be chosen to become the conductor within milliseconds. CobraNet cannot function without a conductor.

Four main types of packet are used in the transmission and synchronization of CobraNet:

- **Beat packets** – the conductor outputs a beat packet to all other CobraNet devices on the network, at a rate of 750 packets per second. All other CobraNet devices on the network synchronize their audio clock and their data transmissions to the beat packet. The beat packet contains network operating parameters, clock data and transmission permissions for multicast and unicast bundles.
- **Audio packets** – also known as "Isochronous Data Packets", these packets are sent out by all CobraNet devices after they receive a beat packet. At standard latency settings, one audio packet is sent for each beat packet received, and each audio packet transmits 64 samples of audio data per channel. However, at lower latency settings, audio packets may be sent twice or four times for each beat packet received. Bundles do not share packets; separate packets are sent in sequence for each bundle transmitted from the same device.
- **Reservation packets** – these packets are transmitted as needed or typically once per second at minimum. Their function is to control bandwidth allocation, initiate connections between CobraNet devices, and monitor the status of CobraNet devices.
- **Serial bridge packets** – asynchronous serial data may be sent between CobraNet devices on the same network. Many standard asynchronous serial formats are supported, including RS-232, RS-422, RS-485 and MIDI.

Latency

The buffering of audio data into Ethernet packets typically incurs a delay of 256 samples (or 5 ⅓ milliseconds). Additional delays are introduced through A-D and D-A conversion (typically 10–50 samples). Latency can be reduced by sending smaller packets more often. In most cases, the programmer can choose the desired CobraNet latency for a particular CobraNet device (5 ⅓, 2 ⅔, or 1 ⅓ milliseconds). However, reducing audio latency has consequences:

- Reducing latency requires more processing power.
- Reducing latency places additional demands on network performance, and may not be possible in some network configurations if the forwarding delay is too great.
- Since reducing latency means sending smaller packets more often, more high resolution (i.e. 96 kHz, 24-bit) audio channels can be sent per bundle without exceeding the 1,500-byte payload limit for Ethernet packets. See the table below for bundle capacity limits:

	Channels per bundle					
Latency	16 bit, 48 kHz	20 bit, 48 kHz	24 bit, 48 kHz	16 bit, 96 kHz	20 bit, 96 kHz	24 bit, 96 kHz
5 ⅓ ms	8	8	7	5	4	3
2 ⅔ ms	8	8	8	8	8	7
1 ⅓ ms	8	8	8	8	8	8

It may seem from the table above that more information can be sent at a lower latency. However, that is not the case. More channels can be sent per bundle, but fewer bundles can be processed simultaneously by one device. So, while eight 24-bit, 96 kHz channels can be sent in one bundle at 1 ⅓ ms latency, due to processing constraints, the CobraNet device may only be able to send and receive one bundle instead of the usual four. The bundle capacity of CobraNet devices are unique to the particular device, and are not always the same. However, below is a table illustrating the bundle capacity for a Biamp AudiaFLEX-CM DSP device. The Rx and Tx columns indicate the absolute maximum number of channels that can be received or transmitted. The Rx/Tx column represents the maximum number of channels that can be received and transmitted simultaneously.

Channels per bundle	**5 ⅓ ms latency**			**2 ⅔ ms latency**			**1 ⅓ ms latency**		
	Rx	Tx	Rx/Tx	Rx	Tx	Rx/Tx	Rx	Tx	Rx/Tx
8	32	32	32/32	32	32	32/32	32	32	16/16
7	32	32	32/32	32	32	29/29	28	32	14/15
6	32	32	32/32	32	32	29/29	24	32	12/13
5	32	32	32/32	32	32	25/27	21	32	12/13
4	32	32	32/32	32	32	24/24	20	28	12/12
3	32	32	32/32	32	32	20/21	15	24	9/11
2	32	32	28/29	27	32	16/16	12	18	6/7
1	16	16	16/16	16	16	9/10	7	10	4/4

Hardware and software

CobraNet network cards

CobraNet network cards come in several varieties, some of which can support more channels than others. Additionally, CobraNet network cards have two Ethernet ports labelled "primary" and "secondary". Only the primary Ethernet port needs to be connected, but if both ports are connected they become a redundant failsafe. That is, if the primary port loses communication, the secondary port immediately takes over with no packet loss. Careful network design and topology which takes advantage of this feature can provide extremely high reliability in critical applications.

The typical CobraNet network cards provided by Cirrus Logic are the CM-1 and the CM-2:

- **CM-1** – the standard CobraNet card, provides 32x32 simultaneous I/O channels.
- **CM-2** – compact, low-power, lower cost design provides 8 or 16 simultaneous I/O channels.

Both cards are designed to be added to audio products by the manufacturer.

Software

Cirrus Logic provides a software application known as CobraCAD, which assists in the design of the network on which the CobraNet system will run. It helps to identify if there are too many routers between two CobraNet devices, if a certain latency is possible given the network configuration, and other tasks. However, Cirrus Logic does not provide software to manipulate their hardware. In fact, in the simplest of cases, no software is required by the end user. For instance, a simple breakout box which converts a CobraNet signal to eight analog audio signals would require little or no configuration by the end user (apart from possibly selecting the bundle number). If configuration is required (for example, in a DSP box with integrated CobraNet I/O), then the manufacturer of the device typically supplies proprietary software for that purpose.

Licensed manufacturers

Manufacturers who wish to integrate CobraNet connectivity into their devices must license the technology from Cirrus Logic. Many audio equipment manufacturers have included CobraNet in their products. Below is a partial list of notable examples, sorted by device type:

DSP

One of the most popular and useful devices that integrates CobraNet is the audio DSP. These devices typically receive audio from CobraNet (and often from other digital or analog sources simultaneously), process the audio using digital filters and effects (for example, volume control, EQ, compression, delay, crossovers, etc.) and then output the audio via CobraNet (or other digital or analog outputs). Some DSPs even have an integral telephone hybrid, and can incorporate CobraNet and other sources into a teleconferencing application.

Manufacturers of CobraNet-equipped DSP's

Amplifiers

Amplifiers with integrated CobraNet help keep the signal chain digital for a longer span. Amplifiers with CobraNet inputs often also have limited DSP and monitoring capabilities built-in.

Manufacturers of CobraNet-equipped Amplifiers

Loudspeakers

Loudspeakers with integrated CobraNet help keep the signal chain digital for an even longer span. In a typical unpowered speaker application, the amplifier would be housed far away from the speaker, and a long speaker cable (analog) would be run between the speaker and the amplifier. The speaker cable would be subject to interference and signal

loss from electrical resistance. However, a powered speaker, powered by an electrical cable and fitted with integrated CobraNet inputs, eliminates the speaker cable and replaces it with a network cable. Since a speaker will only use one audio channel out of the bundle, many speakers with CobraNet will also have a number of analog outputs for the rest of the channels in the bundle, which is useful in speaker cluster applications.

Manufacturers of CobraNet-equipped Loudspeakers

Mixing consoles

Many digital mixing consoles are available with optional CobraNet interfaces for increased channel capacity and reduced cabling.

EtherSound

The most recent Ethernet based system is rapidly gaining acceptance for live sound because of its low latency. Ethersound cards are available for Yamaha and Studer digital consoles but the slots in Yamaha desks are limited to 16 inputs and outputs. When digital consoles come fitted with EtherSound, digital multicores will gain widespread acceptance.

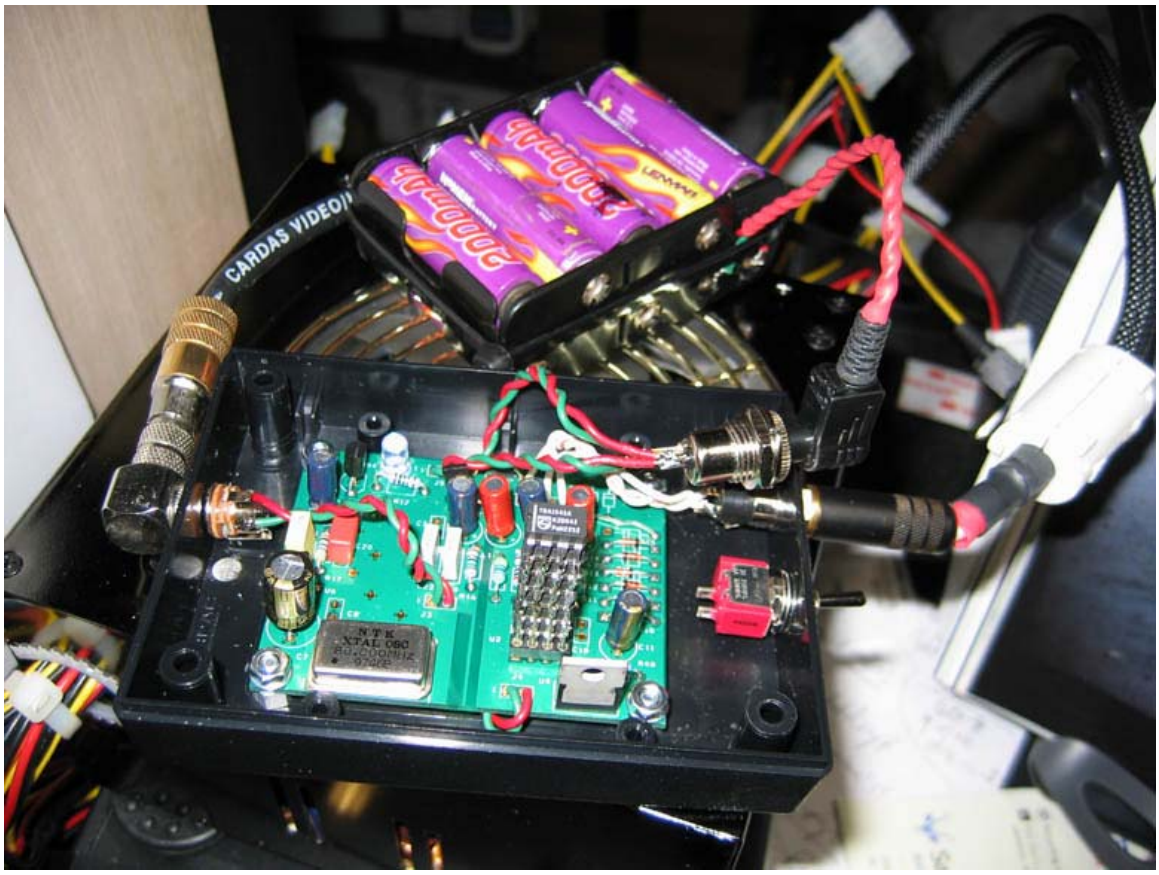
Roland RSS Digital Snake

Roland Corporation RSS Digital Snake is a 40-channel digital audio transmission system using one standard CAT 5e cable. It features high quality remote controllable microphone preamps, configurable inputs and outputs and immunity to RF and electrical interference. A remote control unit allows users to easily adjust input gain and also features “scene” recall of preamp settings including level and phantom power status. The system offers built-in cable redundancy, automatically switching to a backup cable should the main cable be compromised. The system inputs can be “split” using approved Ethernet fast switching hubs providing for multiple “transformerless” audio feeds to monitor consoles, recording devices, and broadcast facilities.

Chapter- 12

Audio Engineering Equipments

DIY audio



An example of DIY audio: a functioning battery-operated DAC (Digital-to-analog converter) before installation in a box case

DIY Audio means "do it yourself" audio. Rather than buying a piece of possibly expensive audio equipment, such as a high-end audio amplifier or speaker, the person practicing DIY Audio will make it him/herself. The benefits of doing so include

economic concerns, the satisfaction of creating something enjoyable, the possibility that the equipment made is of higher quality than commercially available products, and the feasibility of creating a custom-made device for which no exact equivalent is marketed.

History of DIY audio

Audio DIY came to prominence in the 50s to 60s, as audio reproduction was relatively new and the technology "complex," audio reproduction equipment, and in particular high performance equipment, was not offered at the retail level. Kits and designs were available for consumers to build their own equipment. Famous vacuum tube kits from Dynaco, Heathkit, and McIntosh, as well as solid state (transistor) kits from Hafler allowed for consumers to build their own hi fidelity systems. Books and magazines were published which explained new concepts regarding the design and operation of vacuum tube and (later) transistor circuits.

While audio equipment has become easily accessible in the current day and age, there still exists an interest in building one's own equipment, including amplifiers, speakers, preamplifiers, and even CD players and turntables. Today, a network of companies, parts vendors, and on-line communities exist to foster this interest. DIY is especially active in loudspeaker and in tube amplification. Both are relatively simple to design and fabricate without access to sophisticated industrial equipment. Both enable the builder to pick and choose between various available parts, on matters of price as well as quality, allow for extensive experimentation, and offer the chance to use exotic or highly labor-intensive solutions, which would be expensive for a manufacturer to implement, but only require personal labor by the DIYer, which is a source of satisfaction to them.

Construction issues

In modern times, integrated circuits make construction of DIY audio systems easier, but the proliferation of surface mount components (which are very small and difficult to solder with a soldering iron) and fine pitch printed circuit boards (PCBs) arguably make the physical act of construction more difficult. Nevertheless surface mounting is often used, as are conventional PCBs and electronic components, while some enthusiasts insist on using old-style perforated cardboard onto which individual components are "hardwired" and soldered. Test equipment is readily available for purchase and enables convenient testing of parts and systems. Specifications of parts and components are readily accessible through the Internet including data sheets and equipment designs.

It has become easier to make audio components from "scratch" rather than from "kits" due to the availability of CAD software for printed circuit board (PCB) layouts and electronic circuit simulation. Such software can be free, and a trial version may also be used. PCB vendors are more accessible than ever, and can manufacture PCBs in small quantities for the do-it-yourselfer. In fact, kits and chemicals for self-manufacturing one's own PCB can be obtained. Electronic parts and components are accessible online or in specialty shops, and various "high-end" parts vendors exist. On the other hand, a wide

variety of kits, designs and premanufactured PCBs are available for almost any type of audio component.

To construct a device takes more than knowledge of circuits, many would urge that the mechanical aspects of cabinets, cases and chassis' are the most time consuming aspects of audio DIY. Drilling, metalworking and physical measurements are critical to constructing almost any DIY audio project, especially speakers. Measuring equipment such as a Vernier caliper is often essential. Woodworking skills are required to construct wooden enclosures (e.g. for speakers), with some enthusiasts going beyond traditional woodworking to CNC lathing, and luxurious veneers and lacquers. Room acoustics solutions are also popular among DIYers, as they can me made with cheap and readily available insulating materials, and can be dimensioned to fit each particular room in a precise and aesthetically pleasing way.

DIY audio involves "projects" directed to audio. Many DIY audio people fancy themselves to be audiophiles. These people use rare and expensive parts and components in their projects. Examples are the use of silver wire, expensive capacitors, non-standard solders of various alloys, and use of parts that have been cryogenically cooled.

Vacuum tube or "valve" projects are common in audio DIY. While, for mass market audio components, the vacuum tube has been replaced in modern times with the transistor and IC, the vacuum tube remains prominent in specialty high end audio equipment. Thus, interest exists in building components using vacuum tubes, and the vacuum tube is still widely available. There is a wide variety of tubes manufactured nowadays, and many tubes on the market are advertised as "NOS" ("new, old stock", meaning unused stock of old manufacture); not all of the latter being genuinely NOS. Circuits utilizing tubes often are far less complicated than those utilizing transistors or op-amps. Tube enthusiasts often use transformers, sometimes custom-made ones, or even hand-wind their own transformers using cores and wire of their own choice. Note that vacuum tube projects almost always use dangerously high voltages and should be undertaken with due care.

In case lead-containing solder is used instead of RoHS-compliant solder, appropriate environmental precautions with regard to lead and lead products should be taken.

Motivation

The motivations for DIY audio can include cost, entertainment and quality of workmanship. Many DIYers take considerable personal pride in constructing a device that works to their satisfaction. This is sometimes carried to the extreme, with some makers claiming that their product outperforms any and all similar market products. In many cases this is due to excessive emotional attachment, akin to the attachment a parent feels towards their children. Nevertheless some DIY constructs can indeed possess superior sound quality. Part of the reason is the economics of modern manufacturing and commerce. A manufacturer needs to incorporate several cost factors in product pricing, including wages, housing, insurance, transportation, testing, spare parts, servicing, etc, whereas the retailer needs to mark prices up in order to cover their own expenses, stock,

demonstration equipment, etc. All of the aforementioned do not apply to the DIY hobbyist who, furthermore, need not worry about product timing and marketability, and is free to allocate costs solely to superior quality parts. On the other hand a DIYer may not have access to sophisticated equipment and academic-level research, often have to limit themselves to off-the-shelf components, and cannot attain discounts from component suppliers the way a manufacturer may through large consignment orders.

Tweaking and tweakers

DIY audio can also involve "tweaking" of mass market components. It is thought that mass market audio components are compromised by the use of cheap or inferior internal parts that can be easily replaced with high quality substitutes. As a result, an audio component of improved characteristics may be obtained for relatively low cost. Some common changes include replacing opamps, replacing capacitors (recap), or even replacing resistors in order to increase SNR. Changing an audio component in this way is similar to what a tweaker or modder does with a personal computer.

Cloning and cloners

Another common practice in the DIY audio community is to attempt to "clone" or copy a preexisting design or component from a commercial manufacturer. This involves obtaining a lawful public version of, or lawfully reverse engineering, the circuit schematics for the design, and/or even the publicly available PCB layouts. Such a "clone" will not be a perfect copy since different brands and types of parts (often newer parts) will be used, and mechanical aspects of construction will likely differ. However, the circuit or other distinguishing features should be the close to the original.

There are many reasons for wanting to recreate an existing design. The design might be historically important and/or out of production, so the only way to obtain the component is to build it. The design might be very simple so copying it is easily done. The commercial product might be very expensive but its design known, so it may be built for far less than it cost to be purchased. The original design may have some sentimental value to the person building the recreation, and the design built for the memories in one's past. The copy may be made to test or evaluate design concepts or principles in the original.

As an example, a well known "clone" includes amplifiers using high power integrated circuits, such as the National Semiconductor LM3875 and LM3886. The use of a high power IC as part of a quality audio amplifier was popularized by the 47 Labs Gaincard amplifier, and thus the DIY amplifiers using power ICs are often called "chipamps" or "Gainclones."

Usually cloning additionally involves improving or tweaking (see above) the original design, potentially by using more modern components (in the case of discontinued designs) or higher quality parts.

Loudness compensation

Loudness compensation activated on a NAD amplifier.

Loudness compensation is a setting found on some hi-fi equipment and equalisers that increases the level of the high and low frequencies. This is intended to be used at low listening levels, to compensate for the fact that as the volume of audio decreases, the ear's lower sensitivity to extreme high and low frequencies may cause these signals to fall below threshold. As a result audio material may seem to become 'thin' sounding at low volumes, losing bass and treble, the 'Loudness compensation' button (often just labelled 'Loudness') is intended to rectify this situation.

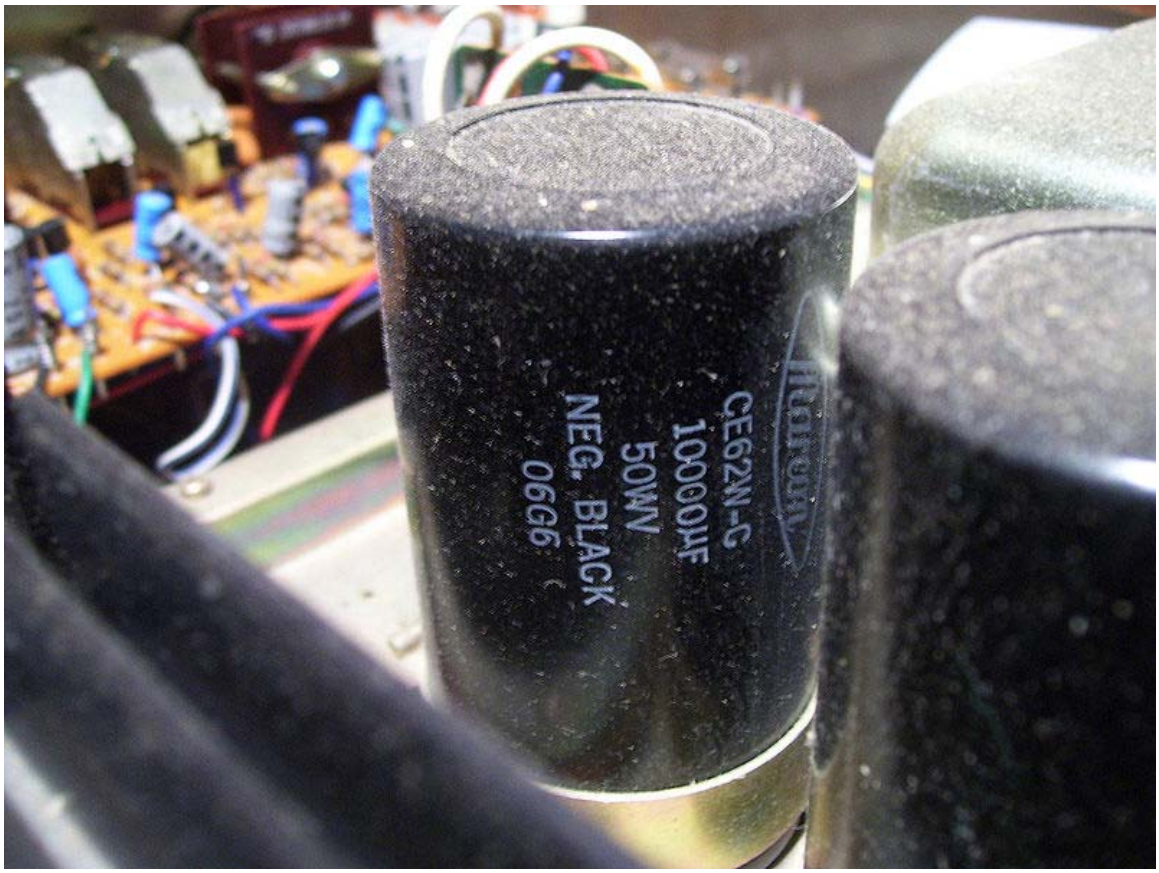
Nikko TRM-800

TRM-800 on a rare Realistic STA-240 receiver

Back of unit showing connections

The **TRM-800** was an audio amplifier made by Nikko. It was introduced in 1975, the same year as the Marantz 2235. It was a 2 channel amp; however; it had three sets of speaker connections; those powered selected by buttons. At 8 ohms, the amp could put out 60 watts per channel. Unlike many amps of this time, however; the TRM-800 was stable at lower impedances than 8 ohms; down to four ohms. The amp has internal circuit breakers which prevent it from clipping or overheating. Its power consumption is 250 watts. For equalization it has only a bass and a treble knob; however the frequency of these are selectable; between 250 and 500 Hertz for the bass, and between 2.5 and 5 kHz for the treble. It also has a high, low, and a subsonic filter.

Large heatsinks and output transistors



10,000 microfarads

Noise (audio)

Noise in audio, recording, and broadcast systems refers to the residual low level sound (usually hiss and hum) that is heard in quiet periods of a programme.

In audio engineering it can refer either to the acoustic noise from loudspeakers, or to the unwanted residual electronic noise signal that gives rise to acoustic noise heard as 'hiss'. This signal noise is commonly measured using A-weighting or ITU-R 468 weighting

Noise is often generated deliberately and used as a test signal. Two types of deliberately generated noise in common use are referred to as 'white noise', which has a uniform spectral power density at all frequencies, or 'pink noise' which has a power spectral density that falls at 3dB/octave with rising frequency. The latter is often more useful in audio testing because it contains constant energy per octave (and hence per commonly used 1/3rd octave), rather than a prepondance of energy at high frequencies. In other words it contains energy that is distributed geometrically rather than linearly.

Audio mute circuit

Audio circuits need a method to "mute" the output so that there is no signal at all when desired. One may need to mute when switching programs, at the push of a button, in response to a control signal, when powering up and down, and when the power cable is disconnected.

Types of mute circuits: 1.) inline analog switch 2.) electromechanical relay 3.) MOSFET pulldown

Audio digital to analog converter (DAC) ICs sometimes have outputs that act as control for external mute circuits. These mute signals go active upon power-up initialization, reset, muting, a certain number of consecutive "0" samples, or if the input clock ratio is incorrect. Also, use of this mute control function can enable the system to achieve idle channel noise and signal-to-noise (SNR) ratios which are only limited by the external mute circuit's characteristics. The mute control output signal polarity is often programmable. It is desirable to mute at the zero crossing of the signal, so that there is no high slew rate signal to zero which comes out like a pop. Multichannel DACs can have programmable grouping of outputs. The ramp rate up to full signal can be also programmable.

Chapter- 13

Audio Feedback

Audio feedback (also known as the **Larsen effect** after the Danish scientist, Søren Larsen, who first discovered its principles) is a special kind of positive feedback which occurs when a sound loop exists between an audio input (for example, a microphone or guitar pickup) and an audio output (for example, a loudspeaker). In this example, a signal received by the microphone is amplified and passed out of the loudspeaker. The sound from the loudspeaker can then be received by the microphone again, amplified further, and then passed out through the loudspeaker again. This is a good example of positive feedback. The frequency of the resulting sound is determined by resonance frequencies in the microphone, amplifier, and loudspeaker, the acoustics of the room, the directional pick-up and emission patterns of the microphone and loudspeaker, and the distance between them.

History and theory

The conditions for feedback follow the Barkhausen stability criterion, namely that, with sufficiently high gain, a stable oscillation can (and usually will) occur in a feedback loop whose frequency is such that the phase delay is an integer multiple of 360 degrees and the gain at that frequency is equal to 1. If the gain is increased until it is greater than 1 for some frequency, then it will be equal to 1 at a nearby frequency, and the system will start to oscillate at that frequency at the merest input excitation, that is to say: sound will be produced without anyone actually playing. This is the principle upon which electronic oscillators are based; although in that case the feedback loop is purely electronic, the principle is the same. If the gain is large, but slightly less than 1, then high-pitched slowly decaying feedback tones will be created, but only with some input sound.

The first academic work on acoustical feedback was done by Dr. C. Paul Boner, PhD., beginning in 1962. Dr. Boner reasoned that when feedback happened, it did so at one precise frequency. He also reasoned that you could stop it by inserting a very narrow notch filter at that frequency in the loudspeaker's signal chain. He worked with Gifford White, founder of White Instruments to hand craft notch filters for specific feedback frequencies in specific rooms. Dr. Boner was responsible for establishing basic theories of acoustic feedback, room-ring modes, and room-sound system equalizing techniques.

Prevention

Most audio feedback results in a high-pitched squealing noise familiar to those who have listened to bands at house parties, and other locations where the sound setup is less than ideal. Usually this occurs when live microphones are pointed in the general direction of the output speakers.

Distance

To keep the maximal loop gain under 1, the amount of sound energy that is fed back to the microphones has to be as small as possible. As sound pressure falls off with 1/r with respect to the distance r in free space or up to a distance known as reverberation distance in closed spaces (and the energy density with $1/r^2$), it is important to keep the microphones at a large enough distance from the speaker systems.

Directivity

Additionally, the loudspeakers and microphones should have non-uniform directivity and should stay out of the maximum sensitivity of each other, ideally at a direction of cancellation. Public address speakers often achieve directivity in the mid and treble region (and good efficiency) via horn systems. Sometimes the woofers have a cardioid characteristic.

Professional setups circumvent feedback by placing the main speakers a far distance from the band or artist, and then having several smaller speakers known as *monitors* pointing back at each band member, but in the opposite direction to that in which the microphones are pointing. This allows independent control of the sound pressure levels for the audience and the performers.

If monitors are oriented at 180 degrees to the microphones that are their sources, the microphones should have a cardioid pickup pattern. Super- or hypercardioid patterns are suitable if the monitor speakers are located at a different angle on the back side of the microphones, they also better cancel reverberations coming from elsewhere. Almost all microphones for sound reinforcement are directional.

Frequency response

Almost always, the natural frequency responses of sound reinforcement systems is not ideally flat. This leads to acoustical feedback at the frequency with the highest loop gain, which may be much higher than the average gain over all frequencies (resonance). It is therefore helpful to apply some form of equalization to reduce the gain of this frequency.

Feedback can be reduced manually by "ringing out" a microphone. The sound engineer can increase the level of a microphone or guitar pickup until feedback occurs. The engineer can then turn down frequency on a band equalizer preventing feedback at that

pitch but allowing maximum volume. Professional sound engineers can "ring out" microphones and pick-ups by ear but most use a real time analyzer connected to a microphone to show the ringing frequency.

To avoid feedback, automatic anti-feedback devices can be used. (In the marketplace these go by the name "feedback destroyer" or "feedback eliminator".) Some of these work by shifting the frequency slightly, resulting in a "chirp"-sound instead of a howling sound due to the upshifting the frequency of the feedback. Other devices use sharp notch-filters to filter out offending frequencies. Adaptive algorithms are often used to automatically tune these notch filters.

Deliberate uses

Early examples in popular music

While audio feedback is usually undesirable, it has entered into musical history as a desired effect beginning in the 1950s with Albert Collins, Johnny "Guitar" Watson and Guitar Slim who all independently recorded and published music featuring that effect. According to Allmusic's Richie Unterberger, the very first use of feedback on a rock record is the song "I Feel Fine" by The Beatles, recorded in 1964. The Who's 1965 hits "Anyway, Anyhow, Anywhere" and "My Generation" featured feedback manipulation by Pete Townshend, with an extended solo in the former and the shaking of his guitar in front of the amplifier to create a throbbing noise in the latter. Canned Heat's Fried Hockey Boogie (off of their 1968 album Boogie with Canned Heat) also featured guitar feedback produced by Henry Vestine during his solo to create a highly amplified distorted boogie style of feedback.

Feedback was used extensively after 1965 by The Monks, Jefferson Airplane, The Velvet Underground and the Grateful Dead, who included in many of their live shows a segment named *Feedback*, a several-minutes long feedback-driven improvisation. Feedback has since become a striking characteristic of rock music, as electric guitar players such as Jeff Beck, Pete Townshend and Jimi Hendrix deliberately induced feedback by holding their guitars close to the amplifier. Lou Reed created his 1975 album *Metal Machine Music* entirely from loops of feedback played at various speeds. A perfect example of feedback can be heard on Jimi Hendrix's performance of Can You See Me? at the Monterey Pop Festival. The Entire Guitar solo was created using amplifier feedback.

Examples in modern classical music

Though closed circuit feedback was a prominent feature in many early experimental electronic music compositions, it was contemporary American composer Robert Ashley who first used acoustic feedback as sound material in his work *The Wolfman* (1964). Steve Reich makes extensive use of audio feedback in his work *Pendulum Music* (1968) by swinging a series of microphones back and forth in front of their corresponding amplifiers.

Contemporary uses

Audio feedback became a signature feature of many underground rock bands during the 1980s. American noise-rockers Sonic Youth melded the rock-feedback tradition with a compositional/classical approach (notably covering Reich's "Pendulum Music"), and guitarist/producer Steve Albini's group Big Black also worked controlled feedback into the makeup of their songs. With the alternative rock movement of the 1990s, feedback again saw a surge in popular usage by suddenly mainstream acts like Nirvana, the Red Hot Chili Peppers, Rage Against the Machine and The Smashing Pumpkins.

Marketing

The principle of feedback is used in many guitar sustain devices. Examples include handheld devices like the Ebow, built-in guitar pickups that increase the instrument's sonic sustain, string drivers mounted on a stand such as the Guitar Resonator, and sonic transducers mounted on the head of a guitar. Intended closed-circuit feedback can also be created by an effects unit, such as a delay pedal or effect fed back into a mixing console. The feedback can be controlled by using the fader to determine a volume level.