
Audio Mixing
and
Audio Effects
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Chapter-1

Audio Mixing

Digital Mixing Console Sony DMX R-100 used in project studios

In audio recording, **audio mixing** is the process by which a multitude of recorded sounds are combined into one or more channels, most commonly two-channel stereo. In the process, the source signals' level, frequency content, dynamics and panoramic position are manipulated and effects such as reverb may be added. This practical, aesthetic or otherwise creative treatment is done in order to produce a mix that is more appealing to listeners.

Audio mixing is done in studios as part of an album or single making. The mixing stage often follows the multitrack recording stage and the final mixes are normally submitted to a mastering engineer. The process is generally carried out by a mix engineer, also called **mixing engineer**, or **mixer**, though sometimes it is the musical producer, or even the artist, who mixes the recorded material.

Prior to the emergence of DAWs (Digital Audio Workstations), the process of mixing used to be carried out on a device known as an *audio mixer*, *sound board*, *desk*, or *mixing* *console*. Nowadays, more and more engineers and independent artists are using a personal computer for the process (commonly referred to as mixing *in-the-box*).

The role of audio mixing

The role of music producer is not necessarily a technical one, with the physical aspects of recording being assumed by the audio engineer, and so producers often leave the similarly technical mixing process to a specialist audio mixer. Even producers with a technical background may prefer that a mixer comes in to take care of the final stage of the production process. Noted producer and mixer Joe Chiccarelli has said that it is often better for a project that an outside person comes in because:

"when you're spending months on a project you get so mired in the detail that you can't bring all the enthusiasm to the final [mixing] stage that you'd like. [You] need somebody else to take over those responsibilities so that you can sit back and regain your objectivity."

However, as Chicarelli explains, sometimes limited budgets dictate that a producer takes care of the mixing as well.

History

Mixing as we know it today emerged with the introduction of commercial multitrack tape machines, most notably the 8-track recorders that were introduced during the 1960s. The ability to record sounds into a multitude of channels meant that treating these sounds can be postponed to a later stage - the mixing stage.

In the 1980s, home recording and mixing began to take market share from recording studios. The 4-track Portastudio was introduced in 1979. Using one, Bruce Springsteen released the album *Nebraska* in 1982. The Eurythmics topped the charts in 1983 with the song "Sweet Dreams (Are Made of This)", recorded by bandmember Dave Stewart on a makeshift 8-track recorder. In the mid-to-late 1990s, computers replaced tape-based recording for most home studios, with the Power Macintosh proving popular. At the same time, digital audio workstations (DAW), first used in the mid-1980s, began to replace tape in many professional recording studios.

Equipment

Mixers

A mixer, or mixing console, or mixing desk, or mixing board, or software mixer is the operational heart of the mixing process. Mixers offer a multitude of inputs, each is fed by a track from a multitrack recorder; mixers would normally have 2 main outputs (in the case of two-channel stereo mixing) or 8 (in the case of surround).

Mixers offer three main functionalities:

- **Mixing** - summing signals together, which is normally done by a dedicated summing amplifier or in the case of digital by a simple algorithm.
- **Routing** - allows the routing of source signals to internal buses or external processing units and effects.
- **Processing** - many mixers also offer on-board processors, like equalizers and compressors.

Simple mixing console

Outboard gear and plugins

Outboard gear (analog) and software plugins (digital) can be inserted to the signal path in order to extend processing possibilities. Outboard gear and plugins fall into two main categories:

- **Processors** - these devices are normally connected in series to the signal path, so the input signal is replaced with the processed signal (e.g. equalizers).
- **Effects** - while an effect can be considered as any unit that affects the signal, the term is mostly used to describe units that are connected in parallel to the signal path and therefore they add to the existing sounds, but do not replace them. Examples would include reverb and delay.

Common classes:

- **Processors**:

    - **Faders** - used to attenuate or boost the level of signals.
    - **Pan pots** - used to pan signal to the left or right and in surround also back and front.
    - **Equalizers** - used to manipulate the frequency content of signals.
    - **Compressors** - used to manipulate the dynamic content of signals. Among many applications they can even the level fluctuations of a singer, or reshape dynamic envelopes of percussive instruments (e.g. adding attack to a snare).
    - **Gates** - used mainly to attenuate low-level signals, for example, the kick spill on a snare recording.
- **Effects**:
    - **Reverbs** - used to simulate the boundary reflection created in a real room, but that adding a sense of space to otherwise 'dry' recordings.
    - **Delays** - most commonly used to add distinct echoes as a creative effect.

Mixing Domains

The process of mixing often accounts for a few mixing domains:

- **Level** - concerned with the relative level between instruments and their dynamics.
- **Frequency** - concerned with the spectral content of the various instruments and the overall mix.
- **Space** - concerned with the spatial aspect of the various instruments. The space domain is often further subdivided into two sub-domains:
    - **Stereo** - concerned with the horizontal panoramic aspects of instruments.
    - **Depth** - concerned with the front-back aspects of instruments.

Mixing in Surround

Mixing in surround is very similar to mixing in stereo except that there are more speakers, placed to 'surround' the listener. The same mixing domains mentioned above are involved, but instead of stereo's horizontal panoramic aspects, and depth's front-back aspects, mixing in surround lets the mix engineer pan sources within a much more three dimensional environment. In a surround mix, sounds can appear to originate from any direction.

There are two common ways to approach mixing in surround:

- **Expanded Stereo** - With this approach, the mix will still sound very much like an ordinary stereo mix. Most of the sources such as the instruments of a band, the vocals, and so on, will still be panned between the left and right speakers, but lower levels might also be sent to the rear speakers in order to create a wider stereo image, while lead sources such as the main vocal might be sent to the center speaker. Additionally, reverb and delay effects will often be sent to the rear speakers to create a more realistic sense of space. In the case of mixing a live recording that was performed in front of an audience, signal recorded by microphones aimed at, or placed among the audience will also often be sent to the rear speakers to make the listener feel as if he or she is in the crowd.

- **Complete Surround / All Speakers Are Treated Equally** - Instead of following the traditional ways of mixing in stereo, this much less conservative approach lets the mix engineer do anything he or she feels like. Instruments can appear to originate from anywhere, or even spin around the listener. When done tastefully, interesting sonic experiences can be achieved.

Naturally, these two approaches can be combined any way the mix engineer sees fit. Recently, a third approach, or method of mixing in surround was developed by surround mix engineer Unne Liljeblad.

- **MSS - Multi Stereo Surround** - This approach treats the speakers in a surround sound system as a multitude of stereo pairs. For example, a stereo recording of a piano, created using two microphones in an ORTF configuration, might have its left channel sent to the Left Rear Speaker and its right channel sent to the Center Speaker. The piano might also be sent to a reverb having its left and right outputs sent to the Left Front Speaker and Right Rear Speaker respectively. Additional elements of the song, such as an acoustic guitar recorded in stereo, might have its left and right channels sent to the Left Front Speaker and the Right Rear Speaker with a reverb returning to the Left Rear Speaker and the Center Speaker. Thus, multiple clean stereo recordings surround the listener without the smearing comb filtering effects that often occurs when the same or similar sources are sent to multiple speakers.

Chapter-2

Automixer

Microphones at a press conference being processed through a Dugan E-1 automixer which has been placed on top of the regular audio mixer. San Francisco mayor Gavin Newsom is speaking at a lectern, while golfers Fred Couples and Greg Norman are seated on stage. Five of eight automixer inputs have been muted and are showing red LEDs. The active input is showing full gain with a ladder of green LEDs

In professional audio, an **automixer** is a hardware or software device that balances multiple sound sources, usually microphones, based on each source's level, quickly and dramatically attenuating inactive inputs on the fly to deliver a more focused and intelligible mix that has less hiss, rumble, reverberation and noise. Automatic microphone mixers use a variety of protocols that allow increased gain before feedback for live sound reinforcement as well as reducing comb filtering between multiple microphones for recorded and broadcast applications.

Invented by Dan Dugan in 1976, automixers are typically used to mix panel discussions on television shows and at conferences and seminars. They can also be used to mix actors' wireless microphones in theater productions and musicals. They are frequently employed in commercial sound systems such as in courtrooms and city council chambers where it is not expected that a live sound operator will be present to mix the microphones. Wherever automixers are used in live sound reinforcement, their main benefit is that they work to maintain a steady limit on the overall signal level of the microphones; if a public address system is set up so that one microphone will not feed back, then, in general, multiple microphones will not feed back if they are automixed. The equivalent number of open mics (NOM) present at the output of the automixer is kept low, regardless of the actual number of open mics.

A skilled audio mix operator can greatly enhance the performance of a sound reinforcement system but will never be able to anticipate with perfect accuracy which participant will speak next in a free-wheeling discussion. Sudden interjections by panelists may be lost completely, or the beginning of a word may be absent until the operator responds as quickly as humanly possible to fade up their audio signal (this loss of the beginning is called *upcut*). A properly adjusted automixer can help in avoiding lost words or phrases due to upcut mistakes or lapses of attention.

History

Frank J. Clement and Bell Labs received a patent in 1969 for a multiple station conference telephone system that switched its output to the loudest input. The next year, Emil Torick and Richard G. Allen were granted a patent for an "Automatic Gain Control System with Noise Variable Threshold", an adaptive threshold circuit invention with its patent assignation going to Columbia Broadcasting System.

Some systems using electro-mechanical switching in regard to microphone activation were engineered in the late 1960s and early 1970s. Peter W. Tappan and Robert F. Ancha devised a system of seat sensors that would activate one of 350 hidden microphones at the Seventeenth Church of Christ, Scientist in Chicago in 1970. From approximately 1968, Ken Patterson and Diversified Concepts developed a hardware system that could detect the "Number of Open Microphones" (NOM) and attenuate the master output by an amount which increased with a higher number of microphones in use. This latter system was public domain.

In 1971, Gregory Maston of Bell Labs filed for a patent involving a circuit that could switch between several audio sources based on their levels. The loudest one was latched into the mix. This system did not ramp switched signals smoothly in and out and did not maintain a constant ambience. It was intended for speakerphone conferencing applications. In 1972, Keith A. T. Knox with the British Post Office Corporation developed an adaptive threshold gate circuit intended for speakerphone usage. The system used a second microphone somewhat near the first to sense ambient noise level.

Dan Dugan's first automixers

Dan Dugan showed his first "Adaptive Threshold Automatic Microphone Mixing System" in 1974 at the 49th Audio Engineering Society (AES) meeting in New York, and was granted a patent for a control apparatus for sound reinforcement systems which sensed ambient sound level in the environment of a theater to control each microphone's individual level. In 1976, Dugan was granted a patent for an automatic microphone mixing process whereby the total gain of the system remains constant. He began manufacturing his first automixer system, the Model A, based on his two patents. Dugan built 60 units, with the first, hand-assembled one taken to Bell Labs to be installed in their conference room for Harvey Fletcher. The algorithm was elegantly simple: *Each individual input channel is attenuated by an amount, in dB, equal to the difference, in dB, between that channel’s level and the sum of all channel levels.* Dugan licensed the system to Altec who released several automixer models including the 1674A, -B and -C series and the 1684A, all focusing on speech applications. (The 1684A became an Electrovoice product and is currently administered by their Commercial division.) The earliest Altec product implementation was regarded as inferior within the commercial audio contractor industry, and other manufacturers began to design their own automixer products.

In 1978, Richard W. Peters of Industrial Research Products (IRP) was granted an improvement patent entitled "Priority mixer control". IRP released the Voice-Matic series of 4x1 and 8x1 automatic mixers using "Dynamic Threshold Sensing" that weighed a combination of the amplitude and history of the signal to determine channel access. The master output was attenuated at the rate of 3 dB for every doubling of NOM. This master output reduction was the solution used by Yamaha Pro Audio two decades later in their DME series of digital signal processing (DSP) products, incorporating an automixer function which was otherwise an 8- or 16-channel noise gate.

Eugene Campbell and Terrance Whittemore of Colorado were granted a patent in 1982 for an automatic microphone mixing algorithm that allowed for musical performance mixing that would not be dominated by the loudest vocalist or instrumentalist.

Graphic user interface for a digital automixer that uses a gain-sharing protocol. Controls include threshold, depth, polarity inversion and muting for each input, as well as volume controls for the four inputs, the four individual outputs and the full mix output

Stephen D. Julstrom of Shure Brothers, Inc. (Evanston, Illinois) was granted a patent in 1987 for a teleconferencing system that used special directionally gated microphones mixed automatically and sent to a distant party via telephone line. The return signal from the distant party was compared in strength to the local mix to determine which party was to be most prominent in the overall mix. Any interrupting party was given priority. Four years later, Shure would introduce the AMS4000 and AMS8000 automixers for sound reinforcement; mixers which required the use of special directional condenser microphones of the Shure AMS Series.

At the 87th AES Convention in 1989, Dugan introduced the idea of using an automixer inserted within target channels on a professional audio mixer. Each microphone's signal would be interrupted inside its channel strip and sent to a variable gain circuit contained within the automixer. The signal would then be returned to the mixer at a level consistent with the Dugan algorithm. This became the Dugan Model D automixer.

In 1991, Dugan's patent expired. Competing manufacturers began to bring the Dugan algorithm directly to their product designs. In 1993, Travis M. Sims, Jr. of Lectrosonics (Rio Rancho, New Mexico) was granted a patent for a sound system with rate controlled, variable attenuation of microphone inputs, including the Dugan algorithm as well as loudspeaker zone attenuation when in close proximity to an active microphone. The loudspeaker zone part of the patent cited a 1985 patent for proportional amplification by Eugene R. Griffith, Jr. of LVW Systems of Colorado Springs, a commercial audio contractor. In 1995, Sims and Lectrosonics gained another patent for an "Adaptive proportional gain audio mixing system" which incorporated a number of ideas including the Dugan algorithm for maintaining a constant total gain of all the inputs.

In 1996, Dugan came out with the Model D-1, a speech-only economy model that did not offer the music system of the Model D.

In 1997, John H. Roberts of Peavey Electronics was granted a patent for an automatic mixer priority circuit, enabling a hierarchy of logic weighting that allowed selected signals to push forward in the mix when they are in use, while still maintaining the useful constant unity, gain-sharing relationship first described by Dugan. The hierarchy enabled a host, moderator or chairperson to speak over other participants and retain control of the discussion. Peavey's Architectural Acoustics division used three levels of hierarchy in their 1998 "Automix 2" product, placing the first- and second-most influentially weighted sources at inputs 1 and 2, respectively.

Dan Dugan licensed his system to Protech Audio (Indian Lake, New York) in 1997, yielding the Protech 2000 model series.

In 2004, the first standard audio mixer incorporating an eight-channel automixer section was released by Peavey in their Sanctuary Series, and in 2006 the similar HP-W was introduced by Crest. Both mixers were aimed at the House of Worship market, adding functions that ease the process of audio mixing to religious organizations.

In 2007, Mark W. Gilbert and Gregory H. Canfield of Shure (Niles, Illinois) were granted a patent for a digital microphone automixer system that used time of arrival as its main decision-making criteria.

Generations of Dugan's insertable automixers

In February 2011, Dugan announced an automixer card to plug in the accessory slot of a Yamaha digital mixing console such as the LS-9, M7CL or PM-5D. This card, the Dugan-MY16, can mix 16 channels of microphone inputs at 44.1–48 kHz or 8 channels at 88.2–96 kHz sampling rates. Channels to be automixed are assigned in the mixer's graphic user interface, and can then be controlled by a common web browser interface affecting only the Dugan-MY16 card, allowing remote control with an iPad, touchscreen computer or laptop over wireless network.

Related applications

- Speech intelligibility enhancement, James M. Kates of Signatron (1984). This system uses Dugan's automatic mixing algorithm to reconstitute several spectral regions of a signal that has been divided into frequency bands for short-time spectral analysis in order to achieve greater intelligibility of spoken consonants.

- Secure conferencing, patent by Raoul E. Drapeau (1993). An automixing algorithm attempts to mask incidental speech that is below automix threshold but which can be audible in the mix. The automix circuitry indicates which sources are active, and whether masking of low-level signals is occurring.

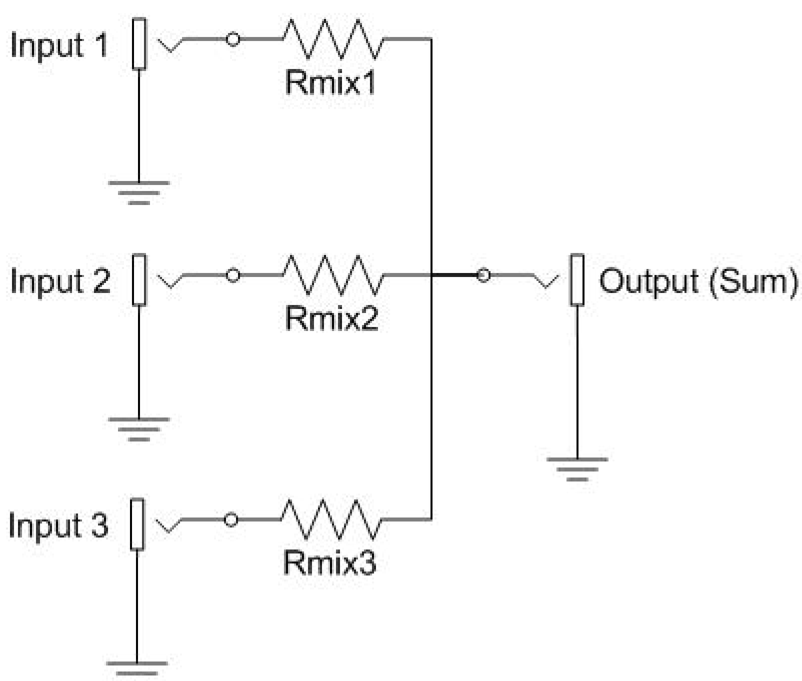
Automixer manufacturers and products

- AKG Acoustics; AS8
- APB Dynasonics; ProSpec Auto-Mixer
- Audio-Technicaa; AT-MX341a SmartMixer.
- beyerdynamic; MCS 100
- Biamp Systems; Audia, Nexia, and AutoTwo
- Crest Audio; HP-W
- Crown; USM-810
- Gentner, Comrex, ClearOne; Converge Series, XAP Series
- Dan Dugan Sound Design; Models A, D, D-1, D-2, D-3, E and E-1
- Industrial Research Products; Voice-Tech
- Intelix; AMIX Series
- Ivie; AudioNet Automatic Matrix Mixers
- Lectrosonics; LecNet2 Series
- Peavey Electronics; Sanctuary, Mediamatrix, NION, Automix 4, Automix 2, etc.
- Rane; RPM 88
- Shure; SCM410, SCM810, FP410
- Symetrix; SymNet Automixing
- TOA; AX-1000A, 9000 Series Digital Matrix Mixer/Amplifier
- Yamaha Pro Audio; Dugan-MY16 card, for the accessory slot of the LS-9, M7CL, or PM-5D

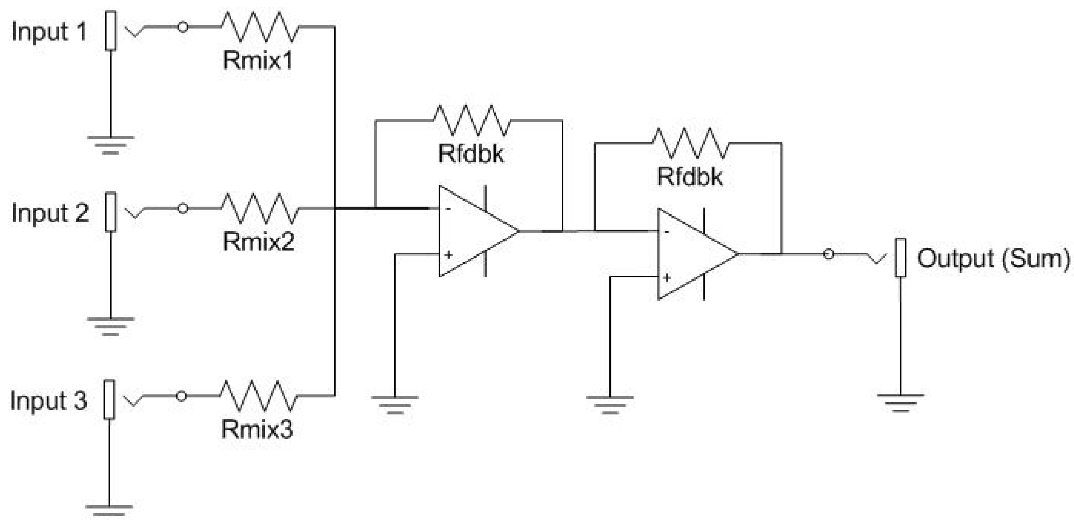
Chapter-3

Electronic Mixer and Beatmatching

Electronic mixer



A simple three-channel passive additive mixer. More channels can be added by simply adding more input jacks and mix resistors.



A "virtual ground" active additive mixer. The buffer amplifiers serve to reduce crosstalk and distortion.

An **electronic mixer** is a device that combines two or more electrical or electronic signals into one or two composite output signals. There are two basic circuits that both use the term *mixer*, but they are very different types of circuits: additive mixers and multiplying mixers.

Additive mixers add two or more signals together, and this terminology ("mixer") is only used in the realm of audio electronics where audio mixers are used to add together audio frequency signals such as voice signals, music signals, and sound effects.

Multiplying mixers multiply together two time-varying input signals instantaneously (instant-by-instant). If the two input signals are both sinusoids of specified frequencies f_1 and f_2, then the output of the mixer will contain two new sinsoids that have the sum $f_1 + f_2$ frequency and the difference frequency absolute value$\{f_1 - f_2\}$.

Note: Any nonlinear electronic block driven by two signals with frequencies f1 and f2 would generate intermodulation (mixing) products. A multiplier (which is a nonlinear device too) will generate ideally only the sum and difference frequencies, whereas a arbitrary nonlinear block would generate also signals at e.g. 2*f1-3*f2, etc. Therefore in the past often more or less normal nonlinear amplifiers or just single diodes have been used as mixers, instead of a more complex multiplier. A multiplier has usually the advantage of rejecting - at least partly - undesired higher-order intermodulations and larger conversion gain.

Additive mixers

Additive mixers add two or more signals, giving out a composite signal that contains the frequency components of each of the source signals. The simplest additive mixers are simple resistor networks, and thus purely passive, while more complex mixers employ active components such as buffer amplifiers for impedance matching and better isolation.

Product Mixers

Ideal product mixers act as signal multipliers, producing an output signal equal to the product of the two input signals. Product mixers are often used in conjugation with an oscillator in the communications field to modulate signal frequencies. Product mixers can either up-convert or down-convert an input signal frequency, but they are more commonly used to down-convert to a lower frequency to allow for simpler filter designs, as done in superheterodyne receivers. In many typical circuits, the single output signal actually contains multiple waveforms, namely those at the sum and difference of the two input frequencies and harmonic waveforms. The output signal may be obtained by removing the other signal components with a filter.

Product mixers have been implemented in a wide variety of ways. The most popular are Gilbert cell mixers, diode mixers, diode ring mixers (ring modulation) and switching mixers. Diodes mixers take advantage of the non-linearity of diode devices to produce the desired multiplication in the squared term. It is a very inefficient method as most of the power output is in other unwanted terms which need filtering out. Inexpensive AM radios still use diode mixers.

Electronic mixers are usually made with transistors and/or diodes arranged in a balanced circuit or even a double-balanced circuit. These are readily manufactured by using the technology of either monolithic integrated circuits or hybrid integrated circuits. These are designed for a wide variety of frequency ranges, and they are mass produced to tight toleraces by the hundreds of thousands. These mixers, especially the double-balanced variety, and be bought in large numbers at prices ranging from a dime to a quarter apiece.

These double-balanced mixers are very widely used in microwave communication systems, satellite communication systems, and ultrahigh frequency (UHF) communications transmitters and receivers, and in radar systems transmitters and receivers.

Gilbert cell mixers are just an arrangement of transistors that multiplies the two signals. The switching mixers (below) pass more power and usually insert less distortion.

switching mixers use arrays of field effect transistors or (in older days) vacuum tubes). These are used as electronic switches, to permit the signal to go one direction, then the other. They are controlled by the signal being mixed. They are especially popular with digitally-controlled radios.

Beatmatching

Beatmatching is a disc jockey technique of pitch shifting or timestretching a track to match its tempo to that of the currently playing track e.g. the kicks and snares in two house records hit at the same time when both records are played simultaneously. Beatmatching is a component of mixing which employs beatmatching combined with equalization, attention to phrasing and track selection in an attempt to make a single mix that flows together and has a good structure.

The technique was developed to keep the people from leaving the dancefloor at the end of the song. These days it is considered basic among DJs in electronic dance music genres, and it is standard practice in clubs to keep the constant beat through the night, even if DJs change in the middle.

Beatmatching is no longer considered a novelty, and new digital mixers have made the technique much easier to master.

Technique

The beatmatching technique consists of the following steps:

1. While a record is playing, beatmatch a new record to it, using headphones for monitoring. Use gain (or *trim*) control on the mixer to match the levels of the two records.
2. Restart and slip-cue the new record at the right time, begin the new record on beat with the record currently playing. Pay attention to track structures; careful phrasing can make the mix seamless.
3. If the beat on the new record hits before the beat on the current record then the new record is too fast, reduce the pitch and manually slow the speed of the new record to bring the beats back in sync.
4. If the beat on the new record hits after the beat on the current record then the new record is too slow, increase the pitch and manually increase the speed of the new record to bring the beats back in sync.
5. Continue this process until the two records are in sync with each other, it can be difficult to sync the two records perfectly, so manual adjustment of the records is necessary to maintain the beat synchronization.
6. Before fading in the new track, check that the beats of two tracks match by listening to both channels together in the headphones, as the sound from the speakers can reach you with a delay.
7. Gradually, fade in parts of the new track while fading out the old track. While in the mix, ensure that the tracks are still synchronized, adjusting the records if needed.

Pitch and tempo

The pitch and tempo of a track are normally linked together: spin a disc 5% faster and both pitch and tempo will be 5% higher. However, some modern DJ software can change pitch and tempo independently using time-stretching and pitch-shifting, allowing harmonic mixing. This technique is referred to as beatmatching.

History

Beatmatching was invented by Francis Grasso in the late 1960s and early 1970s. Initially he was counting the tempo with a metronome and looking for records with the same tempo. Later a mixer was built for him by Alex Rosner which let him listen to any channel in the headphones independently of what was playing on the speakers; this became the defining feature of DJ mixers. That and turntables with pitch control enabled him to mix tracks with different tempo by changing the pitch of the *cued* track to match its tempo with the track being played by ear. Essentially, the technique he originated hasn't changed since.

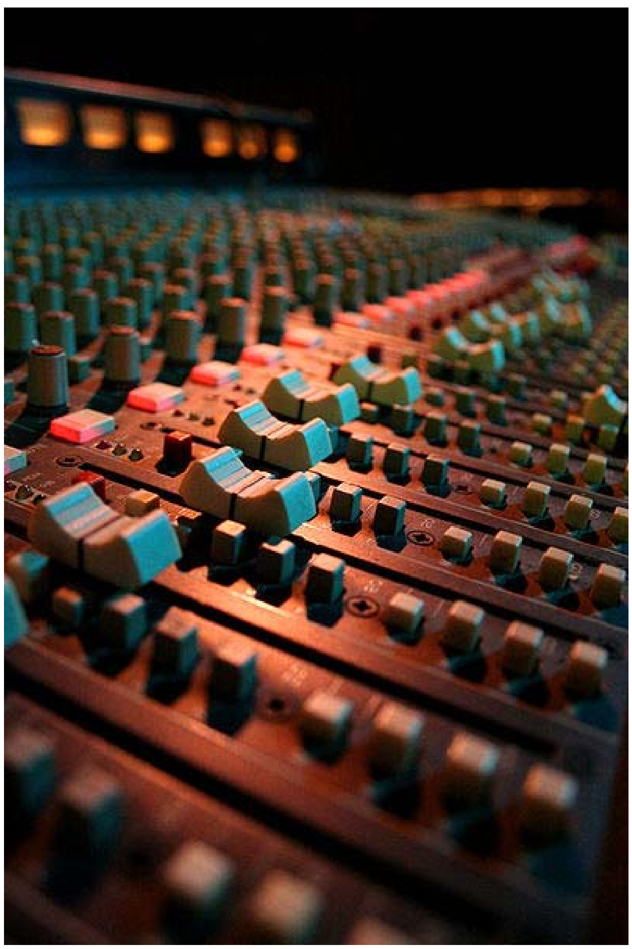
These days beatmatching is considered central to DJing, and features making it possible are a requirement for DJ-oriented players. In 1978, the Technics SL-1200MK2 turntable was released, whose comfortable and precise sliding pitch control and high torque direct drive motor made beatmatching easier and it became the standard among DJs. With the advent of the compact disc, DJ-oriented Compact Disc players with pitch control and other features enabling beatmatching (and sometimes scratching), dubbed CDJs, were introduced by various companies. More recently, software with similar capabilities has been developed to allow manipulation of digital audio files stored on computers using turntables with special vinyl records (e.g. Final Scratch, M-Audio Torq, Serato Scratch Live) or computer interface (e.g. Traktor DJ Studio, Mixxx, Virtual DJ). Other software including algorithmic beatmatching is Ableton Live, which allows for realtime music manipulation and deconstruction, or Mixmeister, a DJ Mixset creation tool. Freeware software such as Rapid Evolution can detect the beats per minute and determine the percent BPM difference between songs.

The change from pure hardware to software is on the rise, and big DJs are introducing new equipment to their kits such as the laptop, and dropping the difficulty of carrying hundreds of CDs with them. The creation of the mp3-player allowed DJs to have an alternative tool for DJIng. Limitations with mp3-player DJing equipment has meant that only second generation equipment such as the IDJ2 or the Cortex Dmix-300 have the pitch control that alters tempo and allows for beatmatching on a digital music player. However, recent additions to the Pioneer CDJ family, such as the CDJ-400, allow mp3-player and other digital storage devices (such as external hard drives and USB memory sticks) to be connected to the CDJ device via USB. This allows the DJ to make use of the beatmatching capabilities of the CDJ unit whilst playing digital music files from the mp3-player or other storage device.

Chapter-4

Fade and Live Sound Mixing

Fade



Audio mixer faders at the Bull & Gate pub in Kentish Town, North London.

In audio engineering, a **fade** is a gradual increase or decrease in the level of an audio signal. The term can also be used for film cinematography or theatre lighting, in much the same way.

A recorded song may be gradually reduced to silence at its end (**fade-out**), or may gradually increase from silence at the beginning (**fade-in**). For example, the songs "Bitter Sweet Symphony" by The Verve and "Turn to Stone" by Electric Light Orchestra fade in from the beginning, while the songs "Born to Be Wild" by Steppenwolf, "Boogie Oogie Oogie" by A Taste of Honey, and "Hey Jude" by The Beatles fade out. However, "Born to be Wild" and "Boogie Oogie Oogie" fade out in a matter of seconds, whereas "Hey Jude" takes over 2 minutes to completely fade out. "Goodbye Stranger" by Supertramp takes about a minute to fade out. Fading-out can serve as a recording solution for pieces of music that contain no obvious ending.

Though relatively rare, songs can fade out, then fade back in. Some examples of this are "Helter Skelter" and "Strawberry Fields Forever" by The Beatles, "Suspicious Minds" by Elvis Presley, "Thank You" by Led Zeppelin, "Undercover of the Night" by The Rolling Stones, and "Bop Gun (Endangered Species)" by Parliament.

The term *fade* is also used in multi-speaker audio systems to describe the balancing of power between front and rear channels.

Origins and early examples

"Neptune," part of the orchestral suite *The Planets*, written by Gustav Holst between 1914 and 1916, was the first piece of music to have a fade-out ending. Holst stipulates that the women's choruses are "to be placed in an adjoining room, the door of which is to be left open until the last bar of the piece, when it is to be slowly and silently closed", and that the final bar (scored for choruses alone) is "to be repeated until the sound is lost in the distance". Although commonplace today, the effect bewitched audiences in the era before widespread recorded sound—after the initial 1918 run-through, Holst's daughter Imogen (in addition to watching the charwomen dancing in the aisles during "Jupiter") remarked that the ending was "unforgettable, with its hidden chorus of women's voices growing fainter and fainter... until the imagination knew no difference between sound and silence".

The technique of ending a spoken or musical recording by fading out the sound goes back to the earliest days of recording. In the era of mechanical (pre-electrical) recording, this could only be achieved by either moving the sound source away from the recording horn, or by gradually reducing the volume at which the performer/s were singing, playing or speaking. With the advent of electrical recording, smooth and controllable fadeout effects could be easily achieved by simply reducing the input volume from the microphones using the fader on the mixing desk.

No single recording can be reliably identified as "the first" to use the technique. In 2003, on the (now-defunct) website *Stupid Question*, John Ruch listed the following recordings as possible contenders:

Bill Haley's cover version of "Rocket 88" (1951) fades out to indicate the titular car driving away. There are claims that The Beatles' "Eight Days a Week" (recorded 1964) was the first song to use the reverse effect—a fade-in.

The earliest such recording anybody could name for me is an 1894 78 rpm record called "The Spirit of '76", a narrated musical vignette with martial fife-and-drum that gets louder as it 'nears' the listener and quieter as it 'moves away'.

The fade-out as a simulation of a moving sound source seems to continue right up to "Rocket 88". But other examples aren't so obvious (though fade-out may always imply that the song continues forever and we're only passing by it for a few minutes).

The oldest true songs with fade-out pointed out to me by 78 record fans bear no obvious relationship to movement. One is "Barkin' Dog" (1919) by the Ted Lewis Jazz Band. Another contender is "America" (1918), a patriotic piece by the chorus of evangelist Billy Sunday.

By the early 1930s longer songs were being put on both sides of records, with the piece fading out at the end of Side One and fading back in at the beginning of Side Two. Records at the time held only about two to five minutes of music per side. The segue allowed for longer songs (such as Count Basie's "Miss Thing"), symphonies and live concert recordings.

However, shorter songs continued to use the fade-out for unclear reasons—for example, Fred Astaire's movie theme "Flying Down to Rio" (1933). Even using fade-out as a segue device doesn't seem obvious, though we certainly take it for granted today.

As a film buff, I have a gut feeling that movies were an influence here. Fade-ins and fade-outs are cinematic devices that begin and end scenes—film language that developed at the same time as these early recordings. The term 'fade-out' itself is of cinematic origin, appearing in print around 1918. And jazz, a favorite of early records, was a popular subject of early movies, too.

Fader

A **fader** is any device used for fading, especially when it is a knob or button that *slides* along a track or slot. A knob which *rotates* is usually not considered a fader, although it is electrically and functionally equivalent. A fader can be either analogue, directly controlling the resistance or impedance to the source (e.g. a Potentiometer); or digital, numerically controlling a digital signal processor (DSP). Digital faders are also referred to as *virtual* faders, since they can be viewed on the screen of a digital audio workstation. Modern high end digital mixers often have piezo-electric actuators attached to the faders such that they can be multi-use and will jump to the correct position for the selected function and/or saved setting.

Crossfading

A **crossfader** on a dj mixer essentially functions like two faders connected side-by-side, but in opposite directions. It allows a DJ to fade one source out while fading another source in at the same time. This is extremely useful when beatmatching two sources of audio (or more, where channels can be mapped to one of the two sides of the crossfader individually) such as phonograph records, compact discs or digital sources.

The technique of **crossfading** is also used in audio engineering as a mixing technique, particularly with instrumental solos. A mix engineer will often record two or more takes of a vocal or instrumental part and create a final version which is a composite of the best passages of these takes by crossfading between each track.

In the perfect case the crossfader would keep constant output level. However, there's no standard on how this should be achieved. Many DJ equipment manufacturers offer different mixers for different purposes (e.g. scratching, beatmixing, cut mixing, etc.). High-end mixers often have crossfade curve switches allowing the DJ to select the type of crossfade necessary. Experienced DJs are also able to crossfade between tracks using the channel faders.

There are many software applications that feature virtual crossfaders. For instance, burning-software for the recording of audio-CDs.

Pre-fader, post-fader

On a mixer with auxiliary send mixes, the send mixes are configured **pre**-fader or **post**-fader.

If a send mix is configured **pre**-fader, then changes to the main channel strip fader does not affect the send mix. In live sound reinforcement, this is useful for stage monitor mixes where changes in the Front of House channel levels would distract the musicians. In recording and post production, configuring a send to be **pre**-fader allows the amount of audio sent to the aux bus to remain unaffected by the individual track fader.

If a send mix is configured **post**-fader, then the level sent to the send mix follows changes to the main channel strip fader. This is useful for reverberation and other signal processor effects.

Live sound mixing

A monitor engineer and console at an outdoor event

Live sound mixing is the art of combining and processing a number of audio signals together to create a "mix" that the audience or performers at a live show hear. There can be a variety of different mixes required, depending on the performance requirements. Three types are: Front of House (FOH), which is primarily for the audience; monitor, which is exclusively for the performer(s); and recording or broadcast (cue), for special purposes.

Whenever sound reinforcement is needed for a live performance of either music, theater, spoken word, or sporting events, a specialized sound system is required. The primary goal is to cover the audience area and stage with a sufficiently amplified signal. The stage or monitor mix is necessary to enable performers to hear themselves and any other parts of the performance as needed. Also, the proper monitor mix can minimize time delays on large stages to help synchronize the performance. In addition, the stage mix can overcome the level of the house sound which can be confusing to listen to on the stage.

The source of sounds for a live mix can be electronic musical instruments, acoustic instruments, playback of pre-recorded sounds and music, voices, other sounds ambience, and/or sound effects. This part of the sound system generally comprises a number of microphones on the stage, to pick up acoustic sounds, and/or a wide variety of other electronic signals.

If the mixing is to occur at a distance from the stage, it is customary for the individual signals to be balanced, low impedance in order to have noise immunity and retain their frequency spectrum. Widely differing levels can be accommodated in modern sound reinforcement systems. An additional requirement is to run the signals with standardized connectors and wiring.

Equipment

A mixing board, a number of speakers (passive or active), power amplifiers, a number of audio processing devices, and the cabling, rigging, and power system to connect all of these components is usually what makes up a complete Sound Reinforcement System. Having the sound mixed or manipulated in real time is required as things are happening live and need constant minor adjustment. Some performers prefer to have the interactions of live musicians translated to the audience directly. An example of this is the old style bluegrass group using only one microphone. The musicians balance their ensemble sound by ear, and move toward the mic to emphasize solos. On the other end of the spectrum are musical or dramatic productions which can have many dozens of individual sources and dozens of sub-mixes out to dozens of speaker systems to deliver the proper mix to each of the performers.

A live sound engineer can mix the sound from the audience position, from a specialized control room, from the stage, or a remote truck, depending on the performance requirements. A trend in large scale theatrical productions is to minimize or eliminate the amount of sound equipment in the audience area so as to retain more seats for the audience. Digital measurement systems such as Smaart, Spectrafoo and Meyer SIM combined with test microphones in the audience area can be used to help the operator monitor the output of multiple loudspeaker zones. Digital control systems can be used for to implement indicated adjustments. For larger and more complex sound systems, more engineers and technicians can be required. The two primary engineers are the Front of House (FOH) engineer and the Monitor Engineer. The Front of House engineer mixes the sound that the audience hears in the house and the Monitor engineer mixes the sound that the performers hear on stage. A live sound engineer refers to a person that is experienced in the set up and operation of a sound reinforcement system.

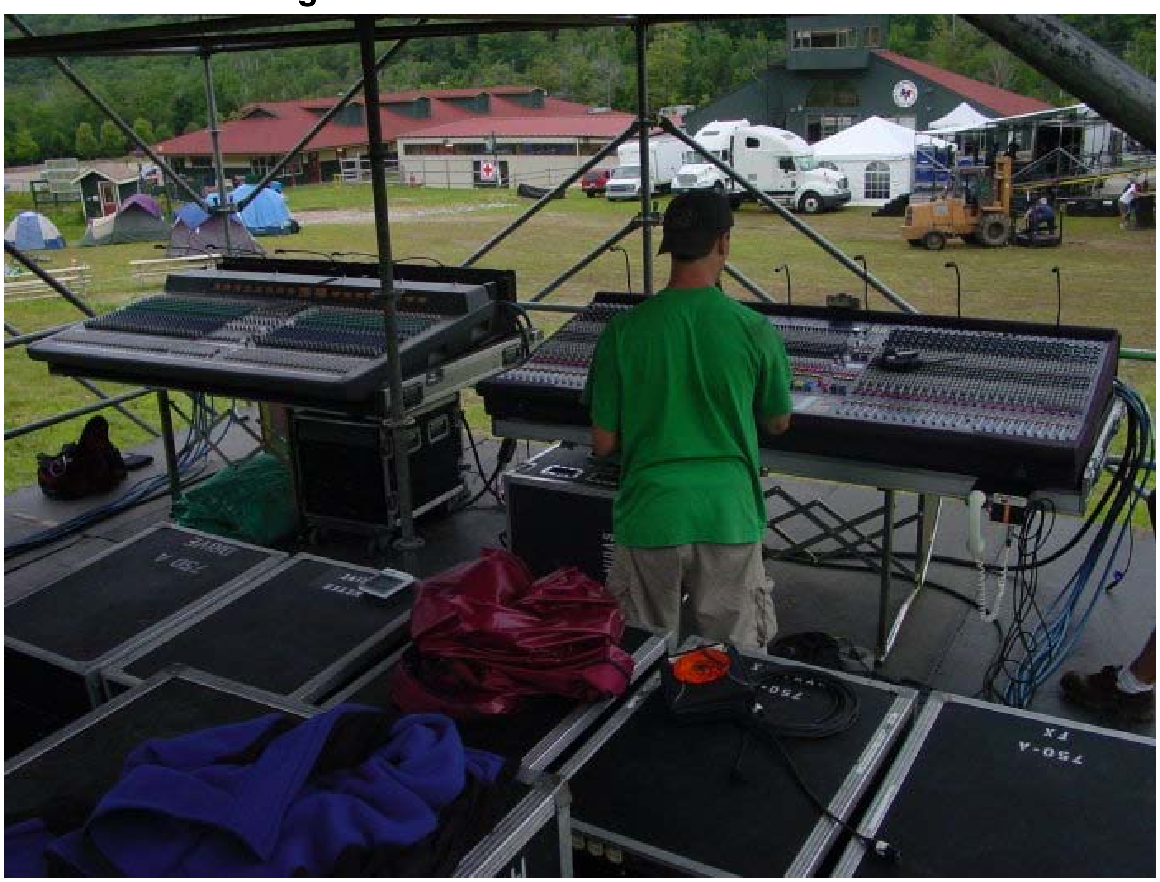
Equipment for a touring act is often packed in heavy-duty reusable road cases to prevent damage during transportation, load-in, and load-out. Some items might be packed partially assembled and transported in custom-built cases or transport frames, to minimize set-up and tear-down time.

Audio engineer positions

Monitor engineer

The monitor engineer's role is most essential at music events, as opposed to spoken word events. In most cases, each performer on stage has their own individual mix that is custom tailored by the monitor engineer to suit their audio needs. The monitor engineer is then faced with the challenge of pleasing anywhere from four to ten or more musicians with a good mix. At shows with a separate monitor mix position, that mixer is typically located just off-stage, to provide easier communication between the performers and the monitor engineer. Though monitor speakers are still in use today, the newest monitor system is what is known as an In Ear Monitor (IEM) system. In Ear Monitors look somewhat like hearing aids, and they are basically a pair of headphones that are custom molded for the musicians' ears and therefore greatly reduces the outside noise that they hear. This isolation protects the musicians' ears from being damaged from the long durations of high volumes that they are subjected to on a large stage. It also allows them to hear their individual mix with more clarity. At the largest and highest budgeted of concert events, each musician is hearing their own individual in ear mix. This involves much more than simply mixing the sound, but requires a great deal of additional audio processing to increase the quality of the performer's mix. Large shows will often use a mixer that is specifically designed for monitor applications.

Front of house engineer



Two FOH consoles at an outdoor event. Each console is typically dedicated to a single band or artist. The telephone handset on the right is part of a closed-circuit intercom system to allow the FOH engineer to communicate with the monitor engineer.

The front of house engineer controls the mix for the audience, and most often operates from the middle of the audience or at the last few rows of the audience from an equipment area known as the "**F**ront **O**f **H**ouse Position" or "FOH". A front of house engineer will often use a variety of processors and effects to provide a particular style to the mix. As with the monitor engineer, front of house engineers are constantly listening to the overall blend in order to make decisions about adjusting the volume and frequency of each instrument or voice on stage. The front of house engineer often makes decisions about which effects devices to use and adjusts their relative levels and blends to meet his or her interpretation of the musical requirements of the song. For smaller shows such as bar and smaller club gigs, it is common for the monitors to be mixed from the Front of House position, and the number of individual monitor mixes could be limited by the capabilities of the Front of House mixer.

Other crew members, such as the lighting console operator, might also work from the FOH position, since they need to be able to see the show from the audience's perspective.

Additional Considerations for Large Shows

For shows that have separate Front of House and monitor mix positions, the audio snake is often designed to provide one or more 'splits' of the audio signals coming from the stage inputs. One split will go to the Front of House mixer, and the other will go to the monitor mixer. In such cases, the snake might be configured with switchable or permanent ground lifts or transformers to isolate the splits and help guard against electrical and RFI noise from being introduced into the system. This is important to take into consideration, as the isolated/lifted path(s) cannot supply phantom power to devices on stage. This is to ensure that phantom power comes from only one place. Very large snakes could also have additional splits for multitrack recorders, broadcast trucks, etc. Some acts use digital snakes to improve routing/control flexibility and save weight, compared to an analog snake and multicore cable.

Crew Communications

For shows in larger venues, where line-of-sight communication between crew members is often not practical, communications are often accomplished with walkie-talkies. Additionally, a hard-line communications system, such as a closed-circuit intercom, might be in use to allow direct communication between the FOH and monitor engineers.

Set up, tear down, and technical rehearsals

The other duty that the live sound engineer serves is the set up and the tear down (removal or striking) of these sound reinforcement systems. For large tours and events, this is often a long (sometimes multiple day) and strenuous process. This will involve unloading the equipment, moving it all into the venue, setting up the systems and then sound-checking. For larger events the engineer will be assisted by a number of audio technicians some of whom may be responsible for maintaining the system during the show while the mix engineer focuses on the sound of the show. After the show is done, the live sound engineers and techs must tear down and load out the sound system for the next show on the tour. The tear down will often take significantly less time than the set up, because there is a much more obvious end objective (i.e. having all the equipment packed into the trucks). Very large touring acts might have two complete sets of equipment that are used at alternating venues on the tour, due to the amount of time needed for the set-up and tear-down of each show. This is done to maximize the number of shows that can be performed in a given amount of time on the tour.

Larger shows and touring acts might have a few days to a few weeks of technical rehearsals. This is time before the start of a tour or a performance run that is used to work out many of the technical issues related to the show. Technical rehearsals are used both by the performers to hone portions of their performances, and also by the engineers, technicians and crew to resolve issues, refine cues, refine the configuration of show control systems, and other logistical details, such as finding the most efficient way to pack the equipment into trucks for transportation to the next venue, if the show is a touring act. In general, the larger and more complex a show is, the more time is allotted for technical rehearsals, though scheduling constraints might impose firm limits on the amount of rehearsal time that is available.

Live sound mixing is an art form in its own right as there are a number of different ways that the mix can be done and a number of different ways that the final mix can sound. The live sound engineer very often has at least a basic musical understanding so that he can make the proper decisions on how to mix different types of music and different types of songs at a concert.

Chapter-5

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.

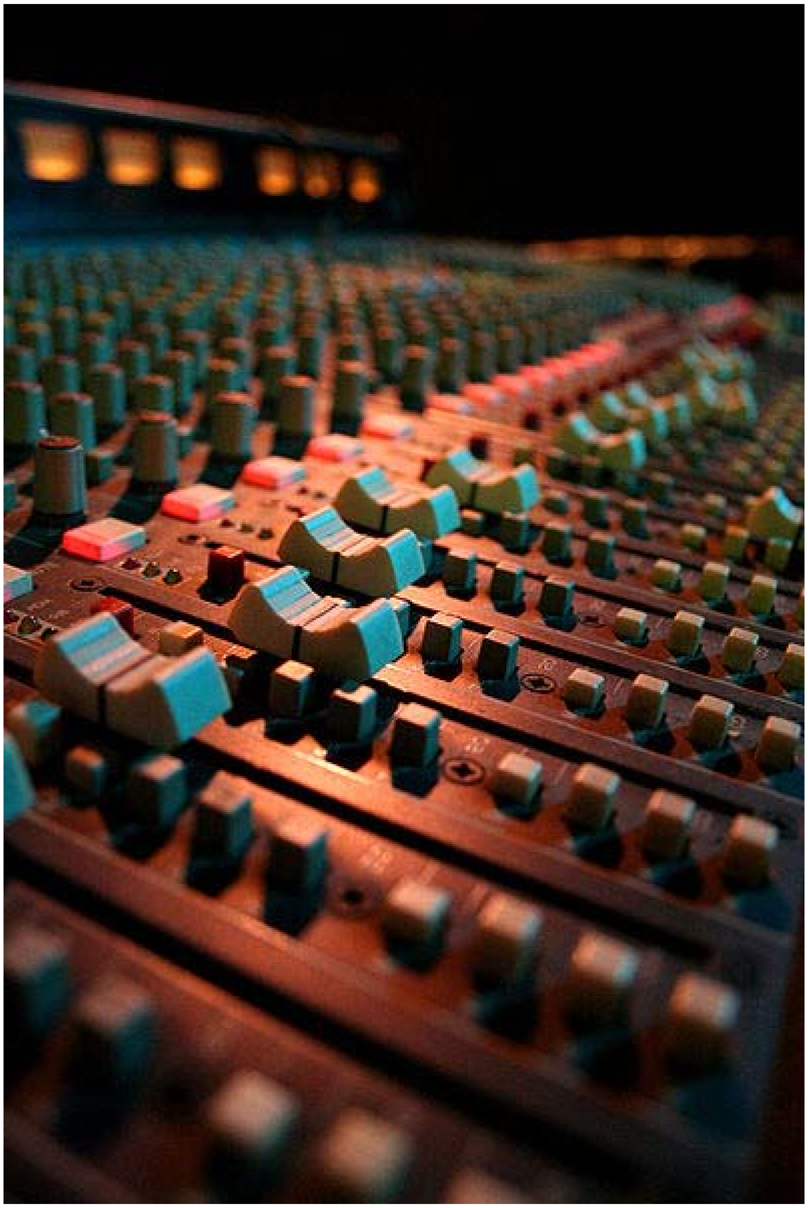
Basic input controls

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.

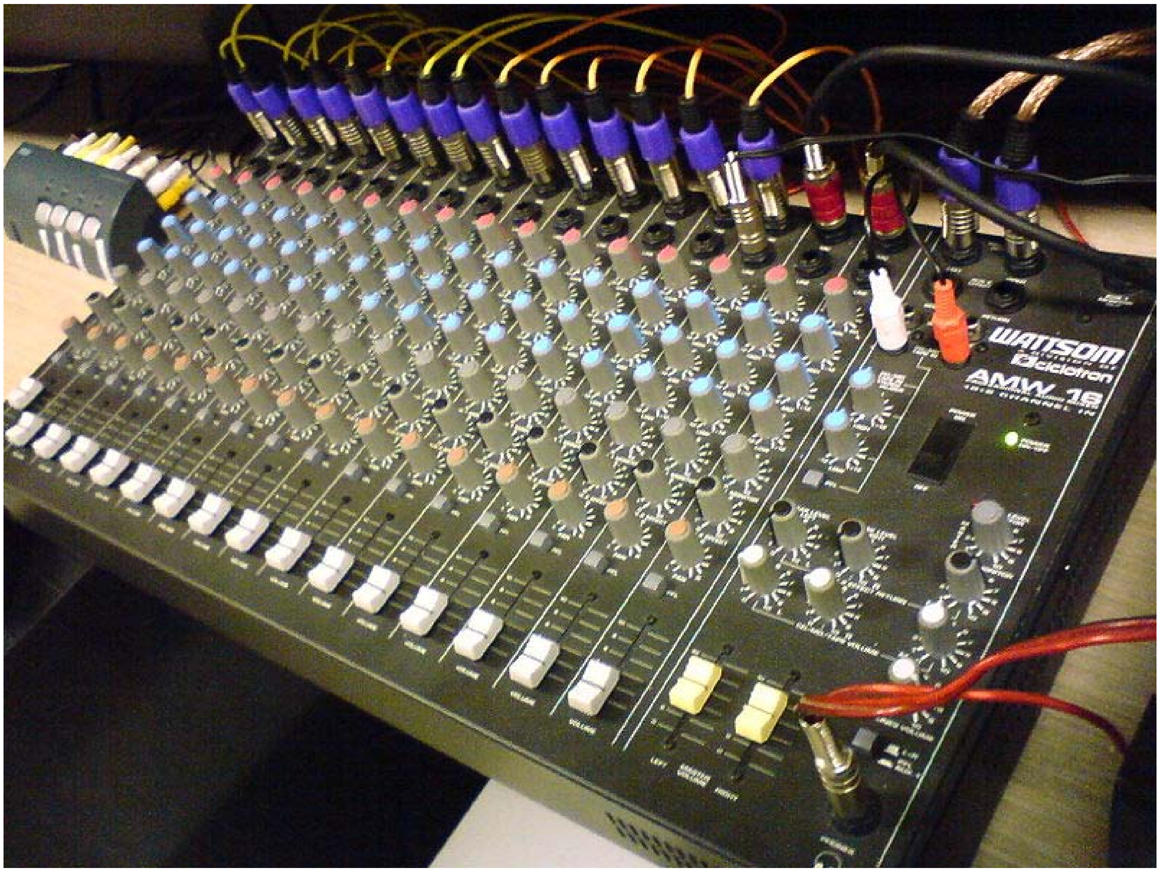
Propagation delay

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-6

3D Audio Effect and Audio Feedback

3D audio effect

3D audio effects are a group of sound effects that attempt to widen the stereo image produced by two loudspeakers or stereo headphones, or to create the illusion of sound sources placed anywhere in 3 dimensional space, including behind, above or below the listener.

There are several types of **3D audio effects**:

- Those that only widen the stereo image by modifying phase information.
- Those that can place sounds outside the stereo basis.
- Those that include a complete 3D simulation.

Stereo widening

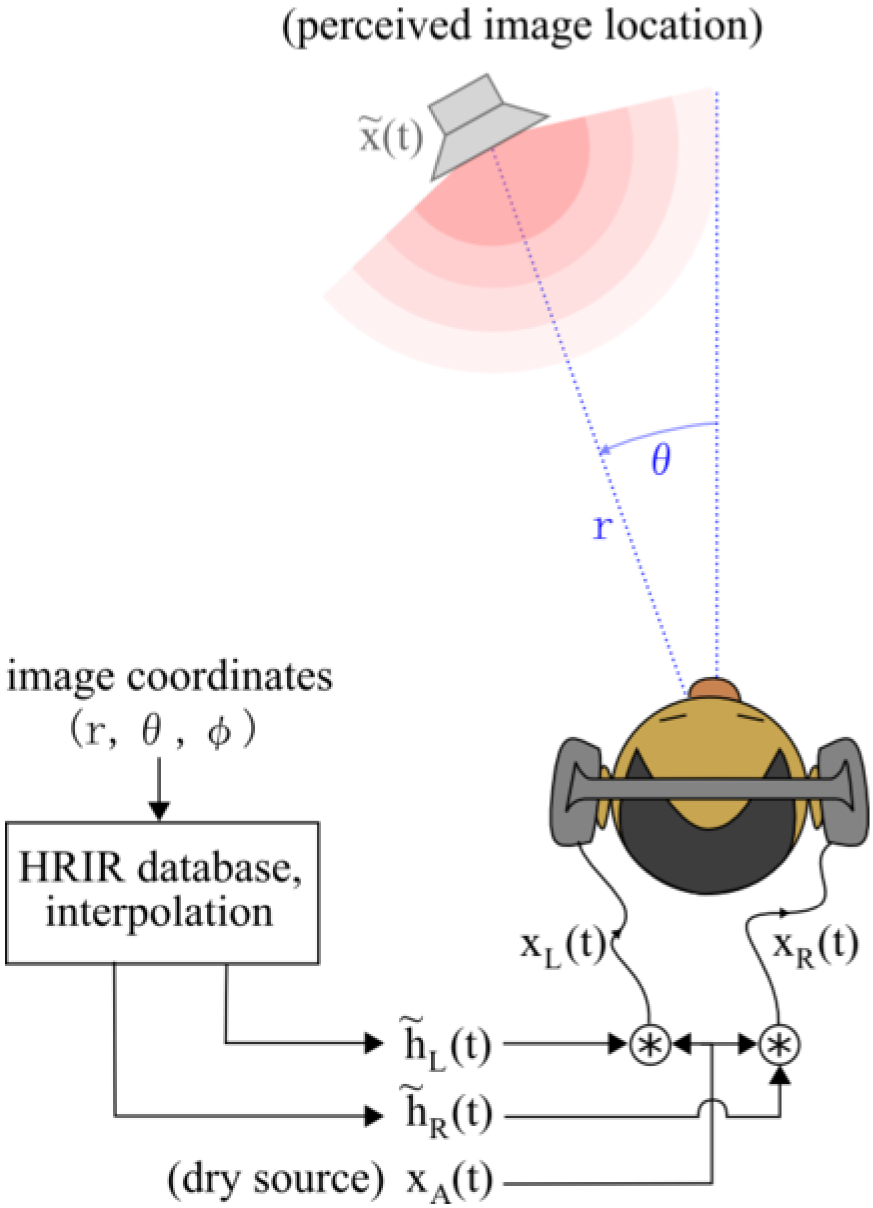
Widening of the stereo image can be achieved by manipulating the relationship of the side signal S and the center signal C: $C = \frac{L+R}{2}; S = \frac{L-R}{2}$. A positive part of the side signal S is now fed into the left channel and a part with its phase inverted to the right channel. Some boomboxes feature such a process.

Another way of looking at this same effect, without extrapolating a center and side signal from the left and right signals, is to simply add the left signal, slightly attenuated and phase inverted, into the right channel and vice-versa. Taking this a step further, a small delay (20-100ms) can be added to the inverted signal before mixing it back in to the original for output, adding a slight reverberation to the effect.

Placement of sounds outside the stereo basis

By manipulating parts of the sound according to psychoacoustic findings in phase and sound, it is possible to create sounds beyond the stereo basis. Effects from QSound Labs have been used on albums from Sting and Madonna in the beginning of the 1990s, as well as in the videogame Super Street Fighter II. Similarly, the pioneering work of researchers (Sibbald et al.) at EMI Central Research Labs in England in the 1980s, and later with Sensaura, produced "3D Audio" CDs...

Complete 3D positional audio



A sound is placed in the horizontal plane by processing the sound with recorded head-related impulse responses.

The 3D simulation is the most advanced group of **3D audio effects**. Using head-related transfer functions and reverberation, the changes of sound on its way from the source (including reflections from walls and floors) to the listener's ear can be simulated. These effects include localization of sound sources behind, above and below the listener.

Some 3D technologies also convert binaural recordings to stereo recordings. MorrowSoundTrue3D converts binaural, stereo, 5.1 and other formats to 8.1 single and multiple zone 3D sound experiences in realtime.

3D Positional Audio effects emerged in the 1990s in PC and Game Consoles. As a medium, interactive games would benefit perhaps more than any other. However, although some technologies do seem to work better than others, 3D sound in games is still quite unconvincing, especially over speakers.

3D audio techniques have also been incorporated in music and video-game style music video arts. The Audioscape research project, provides musicians with a real-time 3D audiovisual content authoring and rendering environment, suitable for live performance applications.

A site with animations and theory of a system using HRTF's to create 3D Audio: ISVR Virtual Acoustics.

True representation of the elevation level for 3D loudspeaker reproduction become possible by the Ambisonics and Wave field synthesis Principle, MorrowSound True3D and A&G 3D-EST.

3D-EST is a new approach to 3D soundscape recording and reproduction that does not use HRTF or reverberation.

3-D audio presentations

Some amusement parks have created attractions based around the principles of 3-D audio. One example is *Sounds Dangerous!* at Disney's Hollywood Studios at the Walt Disney World Resort in Florida. Guests wear special earphones as they watch a short film starring comedian Drew Carey. At a point in the film, the screen goes dark while a 3-D audio sound-track immerses the guests in the ongoing story. To ensure that the effect are heard properly, the earphone covers are color-coded to indicate how they should be worn. This is not a generated effect but a binaural recording.

MorrowSoundTrue3D soundscapes include Torino Winter Olympics, ProFootball Hall of Fame, Great Lakes Children's Museum, NokiaWorld 2008 Barcelona, Denver Museum Nature and Science Gates Planetarium, New York Historical Society, Copenhagen International Theatre, Gallery Rachel Haferkamp Köln, Muu Gallery Helsinki, New Sounds New York, ZHDK Zurich, OKKO Design Stockholm, BAFTA Awards London, Collection of Diana Zlotnick Studio City, CA, as well as Ecsite, AAM, ASTC and IPS conventions. These range from single 8.1 to 64.3 True3D installations, some interactive.

Nick Cave's new novel The Death of Bunny Munro was recording in audiobook format using 3D audio.

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.

Chapter-7

Automatic Double Tracking

Automatic double tracking (ADT) (or alternatively **Artificial Double Tracking**) was an analogue recording technique designed to enhance the sound of voices or instruments during the recording process. It used tape delay to create a delayed copy of an audio signal which was then combined with the original. The effect was intended to simulate the sound of the natural doubling of voices or instruments achieved by doubletracking. The technique was originally developed in 1966 by engineers at Abbey Road Studios in London at the request of The Beatles.

Overview

As early as the 1950s it was discovered that doubletracking the lead vocal in a song gave it a fuller, more appealing sound, especially for singers with weak or light voices. Use of this technique became possible due to the advent of magnetic tape for use in sound recording. Originally, a pair of single-track (or "mono") tape recorders were used to produce the effect; later, multitrack tape machines were used. Early exponents of this technique were Les Paul and Buddy Holly. Before the development of ADT, it was necessary to either record the vocal track twice on two different tracks of a multitrack tape, or to record the vocal first on one tape, then again on a second tape while simultaneously copying the first to the second—a process that could be both tedious and exacting, and might require several takes. After the development of ADT, this process became known as "manual doubletracking".

Ken Townsend

ADT was invented especially for the Beatles during the spring of 1966 by Ken Townsend, a recording engineer employed at EMI's Abbey Road Studios, mainly at the instigation of John Lennon. Lennon hated the tedium of doubletracking during sessions and regularly expressed a desire for a technical alternative.

The doubletracking effect

Doubletracking produces its effect due to it being impossible for a performer to sing or play the same part in exactly the same way twice, meaning that two different performances of the same part have to be recorded so that the fuller, "chorused" effect is created—if one simply plays back two copies of the same performance in perfect sync, the two sound images become one and no doubletracking effect is produced.

Townsend realised that, if two identical performances were played back with one of them slightly out of sync, the sound image would alter and widen, similarly to doubletracking. There was no reliable way that this effect could be achieved by simply copying a vocal track on to another deck and then playing it back with the master slightly out of sync; at the time, there was no technique for synchronising two different tape machines. The end result would be that the second tape deck would gradually drift further and further from the first.

Tape delay system

Instead, Townsend came up with a system using tape delay, after similar principles already in place for echoes applied via tape during a song mixdown. In essence, Townsend's system added a second tape recorder to the regular setup. When mixing a song, its vocal track was routed from the recording head of the multitrack tape, which was before the playback head, and fed to the record head of the second tape recorder. An oscillator was used to vary the speed of the second machine, providing more or less delay depending on how fast or slow the second machine was run relative to the first. This signal was then routed from the playback head of the second machine to a separate fader on the mixer. This allowed the delayed vocal to be combined with the normal vocal, creating the double tracked effect.

Use by The Beatles

The Beatles were thrilled by Townsend's technique and used it throughout the *Revolver* album, and on many of their subsequent recordings. It has been incorrectly claimed that the first use of ADT was on the first half of Lennon's vocal track on "Tomorrow Never Knows", but in fact this vocal track features manual doubletracking. However most of the doubletracked vocals heard on the rest of the album were created using ADT, while the group also used the technique on a number of the instrumental parts to colour the sounds – there is in fact more use of ADT on the mono version of the album than on the more widely known stereo version, with the lead guitar on "Taxman" and the backwards guitar on "I'm Only Sleeping" treated with the effect. ADT could not only be used to create a single double-tracked sound image; but when used on a stereo mix, the effect could be used to "split" the vocal between the two stereo channels, creating the impression of two different vocal parts on either side of the stereo picture. This technique was used on the stereo mixes of "I'm Only Sleeping", "Love You To", "And Your Bird Can Sing", and "Doctor Robert" (on "Here, There and Everywhere", the similar effect heard is actually two different vocals manually double-tracked and panned; on "Eleanor Rigby", the effect is obtained by a combination of manual double-tracking and ADT). This technique could also be applied to instrumental parts as well: on "Love You To", the same use of ADT was applied to the acoustic guitar track, giving the impression of multiple guitars panned left and right.

Flanging

Lennon dubbed the technique "flanging" after producer George Martin jokingly told him it was produced using a "double-bifurcated sploshing flange". Only years later did Martin learn that another technique, also called flanging, was already in use (it is rumoured that recording engineer Joe Meek was the inventor of flanging in the 1950s; however, it is more often attributed to Townsend. Whether that attribution is valid or not is contested due to the close contact between George Martin, who misnomered ADT as flanging, and Townsend.) The term referred to an engineer's alternately pressing and releasing his finger against the flange (rim) of the supply reel on one of two synchronized tape machines as the same audio signal was combined and transferred to a third machine, slightly slowing the machine then allowing it to come back up to speed and in sync with the other, applying a "swooshing" comb filtering effect to the combined audio signal. Alternatively, the engineer could press the flange of one supply reel then the other to achieve a fuller effect.

An additional explanation for the pedigree of flanging has it named after Fred Flange, a pseudonym given to Matt Monro by Peter Sellers, who used a Monro recording to open his 1959 Sinatra parody album *Songs for Swingin' Sellers*. The album was produced by Martin, and presumably the connection with flanging comes from Monro's mimicking (double-tracking) Sinatra. Engineers at Abbey Road realised that the technique they had developed needed a proper technical name and eventually christened it ADT, short for "Artificial Double Tracking", although elsewhere the term "Automatic Double Tracking" became more common.

ADT versus manual doubletracking

Townsend's process succeeded in simulating manual doubletracking quite effectively; however, attentive listeners can often tell the difference between ADT and "real" doubletracking, with the former having a synthetic quality to it and having none of the audible differences between the vocal tracks frequently present in the latter. Over the years, many artists, including the Beatles, continued to use both manual doubletracking, ADT, or a combination of both in different circumstances depending on the effects they wished to achieve, with each technique thought to have certain unique qualities of its own.

The Beatles used ADT widely in conjunction with manual double-tracking on all their subsequent albums, with the exception of *Let It Be*, which was initially intended to be an "honest" album utilising no technical artifice (ADT can still be heard on the finished album, however, due to Phil Spector treating a Hammond organ part with it on his mix of the title track). Some notable examples of ADT use by the Beatles in the years following

Revolver include "Fixing a Hole" (where the bass guitar part is treated with ADT in an attempt to simulate a "fretless tone"), "Within You, Without You" (on which ADT is supposedly used on almost every vocal and instrumental part on the track), "I Am the Walrus" (which uses ADT in conjunction with equalisation to help simulate a "fake stereo" effect on the second half of the stereo mix, which was sourced from the mono mix, by splitting the entire mix between the channels), and the unusually wide ADT used on the lead vocal tracks on "Being for the Benefit of Mr. Kite" and "Blue Jay Way".

Other users of ADT

Townsend's technique, and minor variations on it, quickly caught on and almost immediately began to be used by other artists and record producers. Former Beatles engineer Norman Smith used ADT extensively on Pink Floyd's debut album *The Piper at the Gates of Dawn*, recorded at Abbey Road in 1967. As well as using it for more conventional simulated doubletracking, Smith made much use of the technique to split Syd Barrett's vocals between the stereo channels. In some cases, Smith (or possibly Barrett himself) used such extraordinarily wide ADT in this way as to give the slightly disorientating impression of not so much doubletracking but two quite separate voices on either channel wildly out of time with each other – the best example of this is perhaps on "Bike". Similar effects were later used on some of Barrett's solo works, perhaps indicating his fondness for this unusual use of ADT. Pink Floyd themselves continued to use ADT on most, if not all, of their subsequent albums up until the 1980s, with one notable use being on "Alan's Psychedelic Breakfast", where a part of the drum track is treated with ADT.

In the U.S., Simon and Garfunkel began to use ADT on stereo mixes of their songs to split vocal tracks between the channels, examples of which include "Mrs. Robinson" and "Cecilla".

Gary Kellgren, Jimi Hendrix's engineer, made extensive use of ADT on all of Jimi's albums, frequently using the technique to split vocal, guitar, and even drum parts between the stereo channels.

Psychedelic music

With the rise of psychedelic music, many artists used variations on Townsend's technique to create the "flanging" effect mentioned above, adding a slightly disorientating "swooshing" quality to instruments and voices (although in practice this effect is actually more similar to what today is called "phasing" rather than "flanging"). The Beatles themselves used this effect on "Lucy in the Sky with Diamonds" and more prominently on "Blue Jay Way". A notable example of this technique is "Itchycoo Park" by the Small Faces, where the effect is prominent almost throughout the entire track, particularly on the vocals, drums and cymbals during the chorus. Hendrix also utilised this technique extensively. An example of an ADT variation being used to create an effect more similar to what is considered "flanging" today (rather than phasing) is on the Beatles' *White*

Album tracks "Cry Baby Cry" and "While My Guitar Gently Weeps" – the former features "flanging" on the acoustic guitar part, the latter on the lead guitar.

Doubling echo

A similar technique to ADT is doubling echo, which uses short delays to mimic the doubletracking effect. Many effects units were developed to produce similar sounds, such as chorus, flangers, and phasers, all of which use an oscillating delay (or, in the phaser, a variable phase network).

Arrival of digital technology

ADT became the standard recording studio technique for simulating doubletracking throughout the late 1960s and 1970s until the arrival of digital technology in the 1980s (although not all engineers could apparently figure out how to reproduce the effect successfully, with Jack Douglas recalling that he was at a loss when John Lennon asked him to use ADT on his vocals during a recording session in 1980 but was unable to adequately explain to his producer how the tape decks should be set up to create the effect). With the advent of digital recording, tape- and analog-based delay methods have not been much used, though many of these analog techniques are frequently emulated using comparable digital techniques, or in some cases plugins which are used to extend the capabilities of a Digital audio workstation. Although using digital delay to simulate double-tracking produces a very similar effect to ADT, some claim to be able to hear the difference between the two (certainly one can tell the difference between digital delay and manual double-tracking, as was the case with ADT in previous years – manual double-tracking continues to be used by a number of artists). Some musicians and engineers may casually use the term ADT to refer to any form of simulated doubletracking, including digital delay used in this manner, although strictly speaking the term should refer to the analogue technique. One of the very few examples of ADT being used in recent times is on the Beatles' *Anthology* albums from the mid-1990s, on which George Martin and Geoff Emerick decided to revive the analogue technique rather than simply use the modern digital alternatives in order to achieve a more authentic sound, feeling that ADT produced a warmer, less synthetic sound than digital delay and the latter would be inappropriate for use on recordings made on analogue equipment in the 1960s.

Chapter-8

Delay (Audio Effect)

Various kind of delay effect units

Delay is an audio effect which records an input signal to an audio storage medium, and then plays it back after a period of time. The delayed signal may either be played back multiple times, or played back into the recording again, to create the sound of a repeating, decaying echo.

Early delay systems

The first delay effects were achieved using tape loops improvised on reel-to-reel magnetic recording systems. By shortening or lengthening the loop of tape and adjusting the read and write heads, the nature of the delayed echo could be controlled. This technique was most common among early composers of Musique concrète (Pierre Schaeffer), and composers such as Karlheinz Stockhausen, who had sometimes devised elaborate systems involving long tapes and multiple recorders and playback systems, collectively processing the input of a live performer or ensemble. Audio engineers working in popular music quickly adapted similar techniques, to augment their use of plate reverb and other studio technologies designed to simulate natural echo.

Analog delay

Echoplex EP-2

Before the invention of audio delay technology, music employing a delayed echo had to be recorded in a naturally reverberant space, often an inconvenience for musicians and engineers. The popularity of an easy-to-implement real-time echo effect led to the production of systems offering an all-in-one effects unit that could be adjusted to produce echoes of any interval or amplitude. The presence of multiple "taps" (playback heads) made it possible to have delays at varying rhythmic intervals; this allowed musicians an additional means of expression over natural periodic echoes.

Many delay processors based on analog tape recording, such as Ray Butts' Echosonic (1952), Mike Battle's Echoplex (1959), or the Roland Space Echo (1973), used magnetic tape as their recording and playback medium. Electric motors guided a tape loop through a device with a variety of mechanisms allowing modification of the effect's parameters. In the case of the popular Echoplex EP-2, the play head was fixed, while a combination record and erase head was mounted on a slide, thus the delay time of the echo was adjusted by changing the distance between the record and play heads. In the Space Echo, all of the heads are fixed, but the speed of the tape could be adjusted, changing the delay time. Thin magnetic tape was not entirely suited for continuous operation, however, so the tape loop had to be replaced from time to time to maintain the audio fidelity of the processed sounds.

The Binson Echorec, another popular unit, used a rotating magnetic drum as its storage medium. This provided an advantage over tape, as the durable drums were able to last for many years with little deterioration in the audio quality. Other devices used spinning magnetic discs, not entirely unlike those used in modern hard disk drives.

Robert Fripp used two Revox reel to reel tape recorders to achieve very long delay times for solo guitar performance. He dubbed this technology "Frippertronics", and used it in a number of recordings. John Martyn is widely acclaimed as the pioneer of the echoplex. Perhaps the earliest indication of his use can be heard on the songs Would You Believe Me and The Ocean on the album *Stormbringer* released in February 1970. This was a first taste of things to come from Martyn’s interest in electronics and the boundless possibilities of electric music. *Glistening Glyndebourne* on the album *Bless The Weather* (1971) showcased his developing technique of playing acoustic guitar through the echoplex to stunning effect. He later went on to experiment with a fuzz box, a volume/wah wah pedal and the echoplex on highly acclaimed *Inside Out* (1973) and *One World* (1977). Martyn is cited as an inspiration by many musicians including U2's The Edge.

Often incorporating vacuum tube-based electronics, surviving analog delay units are sought by modern musicians who wish to employ some of the timbres achievable with this technology.

Solid state delay units using analog bucket brigade delay circuits became available in the 1970s and were briefly a mainstream alternative to tape echo. Though solid state analog delays are less flexible than digital delays and generally have shorter delay times, several classic models such as the discontinued Boss DM-2 are still sought after for their "warmer", more natural echo quality and progressively decaying echos. Additionally, several companies make new analog delays. Old delay systems like the Roland Space Echo and Echoplex are still highly regarded and used with some frequency by modern bands.

Digital delay

Ibanez DE-7 delay pedal.

The availability of inexpensive digital signal processing electronics in the late 1970s and 1980s led to the development of the first digital delay effects. Initially, they were only available in expensive rack mounted units but eventually as costs came down and the electronics grew smaller, they became available in the form of foot pedals. The first digital delay offered in a pedal was the Boss DD-2 in 1984. Rack mounted delay units evolved into digital reverb units and on to digital multieffects units capable of more sophisticated effects than pure delay, such as reverb and Audio timescale-pitch modification effects.

Digital delay systems function by sampling the input signal through an analog-to-digital converter, after which the signal is passed through a series of digital signal processors that record it into a storage buffer, and then play back the stored audio based on parameters set by the user. The delayed ("wet") output may be mixed with the unmodified ("dry") signal after, or before, it is sent to a digital-to-analog converter for output.

Many modern digital delays present an extensive array of options, including a control over the time before playback of the delayed signal. Most also allow the user to select the overall level of the processed signal in relation to the unmodified one, or the level at which the delayed signal is fed back into the buffer, to be repeated again. Some systems today allow more exotic controls, such as the ability to add an audio filter, or to play back the buffer's contents in reverse.

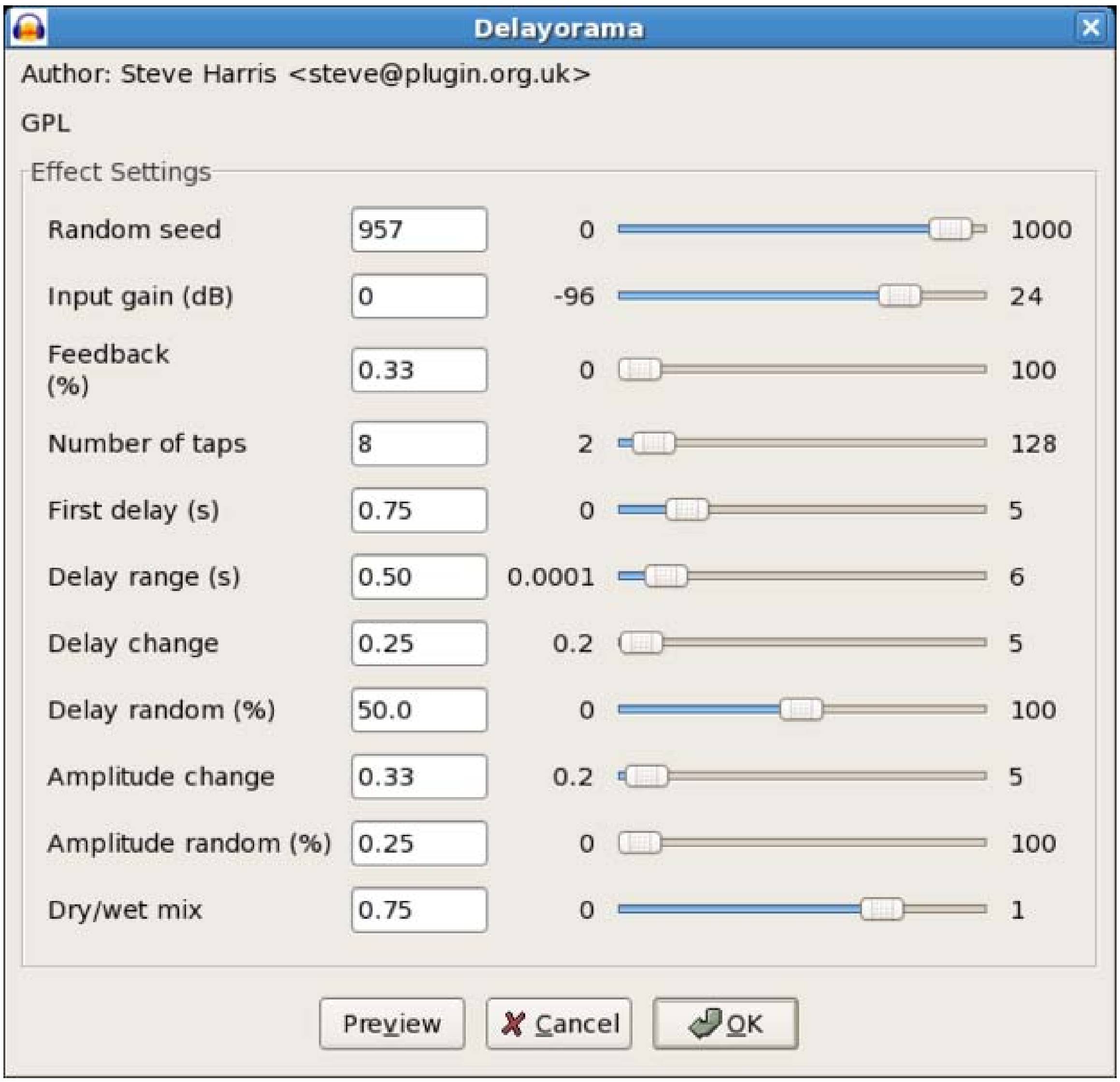
As digital memory became cheaper in the 80s, units like Lexicon PCM84, Roland SDE3000, TC Electronic 2290 offered above 3 seconds delay time, enough to create background loops, rhythms and phrases. The 2290 was upgradable to 32 seconds, and Electro-Harmonix offered a 16-second delay and looping machine.

A sample setup of loop music

From delay to loop

While the mentioned long delay units were a bit clumsy to create loops, the Paradis LOOP Delay, created in 1992, was the first unit with dedicated looping functions Record, Overdub, Multiply, Insert, Replace, ... Gibson manufactured a slightly improved version as Echoplex Digital Pro until 2006. Its software Aurisis LOOP is also the last loop tool based on a continuous memory structure as used by tape and digital delays. Most following loopers repeat samples and thus have little in common with a digital delay, the exceptions being Maneco's early loopers, the Boss DD-20 in digital delay mode, and the Pigtronix echolution.

Computer software



Steve Harris' Delayorama software

A natural development from digital delay-processing hardware was the appearance of software-based delay systems. In large part, this coincided with the popularity of both professional and consumer audio editing software. Software delays, in many cases, offer much greater flexibility than even the most recent digital hardware delays. Abundant system memory on modern personal computers offers practically limitless storage for the audio buffer, and the natural efficiency of audio delay algorithms has made the implementation trivial for delays offering shifting or random delay times, or the insertion of other audio effects during the feedback process. Many authors of software plugins have added functionality to emulate the sounds of the earlier analog units.

Software-based delays are most popular today among musicians in electronic genres or those who prefer to audition the effect in a digital audio editing and mixing environment.

Uses

In popular and electronic music, electric guitarists use delay to produce densely overlaid textures of notes with rhythms complementary to the music. Vocalists and instrumentalists use it to add a dense or ethereal quality to their singing or playing. Extremely long delays of 10 seconds or more are often used to create loops of a whole musical phrase.

Echoplex is a term often applied to the use of multiple echoes which recur in approximate synchronization with a musical rhythm, so that the notes played combine and recombine in interesting ways. In fact, it was the name of a particular delay unit, the Maestro Echoplex.

Doubling echo is produced by adding short-range delay to a recorded sound. Delays of thirty to fifty milliseconds are the most common; longer delay times become slapback echo. Mixing the original and delayed sounds creates an effect similar to doubletracking, or unison performance.

Slapback echo uses a longer delay time (seventy-five to 250 milliseconds), with little or no feedback. The effect is characteristic of vocals on 1950s rock-n-roll records, particularly those issued by Sun. It is also sometimes used on instruments, particularly drums and percussion. Slapback was often produced by refeeding the output signal from the playback head of a tape recorder to its record head, the physical space between heads, the speed of the tape, and the chosen volume being the main controlling factors. Analog and later digital delay machines also easily produced the effect.

Flanging, *chorus* and *reverberation* (reverb) are all delay-based sound effects. With flanging and chorus, the delay time is very short and usually modulated. With reverberation there are multiple delays and feedback so that individual echoes are blurred together, recreating the sound of an acoustic space.

Straight delay

In audio reinforcement, a straight delay is used to compensate for the passage of sound through the air. Unlike audio delay effects devices, straight delay is not mixed back in with the original signal. The delayed signal alone is sent to loudspeakers so that the speakers reinforce the stage sound at the same time or slightly later than the acoustic sound from the stage, approximately 1 millisecond of straight delay per foot of air or 3 milliseconds per meter, depending on the air temperature's effect on the speed of sound. Because of the Haas effect, this technique allows audio engineers to use additional speaker systems placed away from the stage and still give the illusion that all sound originates from the stage. The purpose is to deliver sufficient sound volume to the back of the venue without resorting to excessive sound volumes near the front.

Straight delay is also used in audio to video synchronization to align sound with visual media if the visual source is delayed. Visual media can become delayed by a number of mechanisms, in which case the associated audio should be delayed to match.

Chapter-9

Distortion (Music)

A well-used "Turbo Distortion" guitar effect pedal made by Boss.

Distortion, **overdrive** and **fuzz** are effects applied to the electric guitar, the electric bass, and other amplified instruments such as the Hammond organ, synthesizers, harmonica and even vocals by electronically clipping the signal. This adds sustain and additional harmonics to the signal.

The most subtle types of distortion add a "warm" thickness to the original tone, used in electric blues, for instance, while more extreme types range from the noisy, buzzy sound of a 1960s fuzzbox to the screaming, "bite", "grit", and "crunch" of a late 1980s thrash-style distortion pedal and the hard-edged distortion featured in noise music, hardcore punk, industrial, grunge, and metal. A **fuzzbox** (or **fuzz box**) boosts and clips the signal sufficiently to turn a standard sine wave into a waveform much closer to a square wave. This gives a much more distorted and synthetic sound than a standard distortion or overdrive. Fuzz boxes also tend to have lower mid frequencies than other distortion types.

Distortion can be produced by many components of an instrument's signal path, including effects pedals, the pre-amplifier, power amplifier, speakers, or more recently, digital amplifier modelling devices and software. The distortion in guitar effects pedals such as the Ibanez/Maxon TS-9 and 808 Tube Screamer is produced by transistor or diode clipping. Many players use a combination of these to obtain their "signature" tone.

History

In the early days of guitar amplification, amplifiers were primitive and low-fidelity, and distortion was inherent in the signal chain. Early examples of distortion were often the result of accidental damage to a guitar amplifier, its vacuum valves or loudspeakers. The earliest uses of intentional distortion that have been recorded were achieved through "doctoring" amplifiers and speakers. Guitarists would use a razor blade, screwdriver or pencil to poke holes into their speaker cones to create a distorted sound.

One notable example was Link Wray, who dislodged a tube by accident, and then took to doing so as a habit to get a noisy, dirty sound for his solos. During the recording of "Rocket 88", one of the early rock and roll songs, Ike Turner and the Kings of Rhythm guitarist Willie Kizart used an amplifier that had been damaged in transit, resulting in an early recorded example of guitar distortion. For the recording of "The Train Kept A-Rollin'" (1956) by the Johnny Burnette Trio, a valve fell out of the amplifier during a live performance. When a reviewer then raved about the crazy new sound, Paul Burlison, the guitar player, used the same tone in the recording studio.

Willie Johnson's playing on Howlin' Wolf's Memphis recordings of 1951-2 is marked by a consistent use of deliberate distortion, creating a raucous, menacing sound that complements Howlin' Wolf's singing. Another early user of valve overdrive was Chuck Berry who at the start of his career played through small valve amplifiers, the only ones he could afford. Because of their low output they were easy to overdrive, giving his guitar tone the warm sound heard on his first hit "Maybellene". On later recordings he was able to afford better and larger amps and consequently his tone became cleaner.

Leo Fender of Fender guitars and amplifiers observed these trends and engineered many of his amplifiers to "compress" and/or "overdrive" slightly without drastically distorting the signal. The early Fender "Tweed" and "Blackface" amplifiers are considered a good example of clean electric guitar tone. Many later amplifiers are based on these designs. Significantly, Jim Marshall of Marshall Amplifiers copied the Fender Bassman using parts available in the United Kingdom, creating an amplifier with significant overdrive that quickly caught on in the local music scene and laid the foundation for the powerful, thick "Marshall Sound" that can be heard on so many early Hard Rock albums.

Nashville session musician Grady Martin produced the fuzz sound in 1961 during a recording session for Marty Robbins' "Don't Worry" due to a faulty recording console preamplifier circuit. In 1962, The Ventures, having heard the guitar tone on "Don't Worry", asked friend Red Rhodes, a steel player and electronics wizard, how they could reproduce the sound. A few months later, Rhodes presented them with a custom fuzz box, reportedly the first, which The Ventures used to record "2000 Pound Bee." The song charted in December 1962 and is identified by multiple sources, including *The VH-1 Music First Rock Stars Encyclopedia*, as the first single to use actual guitar fuzz box (the story was in the April 2007 issue of *Guitar Buyer* magazine in an article titled, "Caught By The Fuzz"). Fuzzboxes became popular as a much easier way to create a distorted sound.

Distortion gained wider popularity after a distorted sound was popularised by Dave Davies of The Kinks who played through a small amp whose speaker cone had been slashed with a razor blade, used to record "You Really Got Me", the band's first number one single and the first popular rock & roll song using a distorted power chord riff.

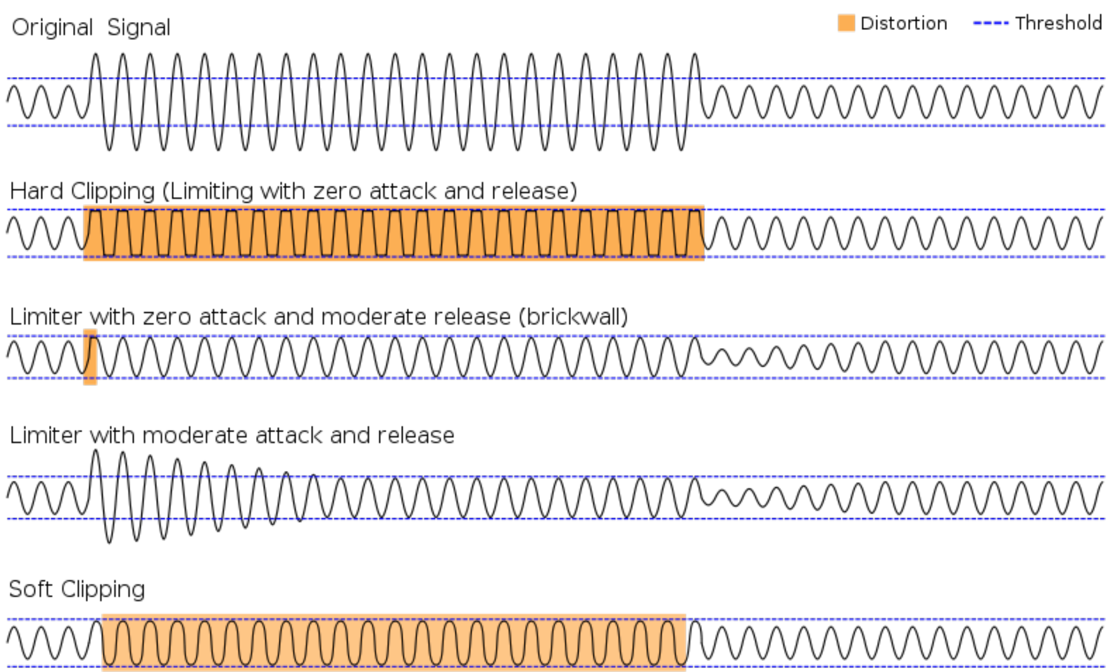
The first purpose-designed commercial distortion circuit was marketed by Maestro as the "Fuzz Tone" Model FZ-1. In May 1965 Keith Richards used a Gibson Maestro Fuzz-Tone to record "(I Can't Get No) Satisfaction". The song's success so boosted sales of the device that all available stock had sold out by the end of 1965. Jimi Hendrix is particularly associated with the increased use of such effects, many designed or modified by guitar technologist Roger Mayer.

Other examples of fuzzboxes include the highly-sought Mosrite FuzzRITE, the Fuzz Face (originally made by the Arbiter Group) used by Jimi Hendrix, the Big Muff Pi (made by Electro-Harmonix) and the Vox Tone Bender, used by Paul McCartney on George Harrison's composition Think for Yourself, and many other Beatles recordings. Pete Townshend used a Univox Super Fuzz pedal starting from 1968 and used on many recordings and stage shows by The Who (being his only pedal for concerts from 1968–1978).

Early fuzzboxes used germanium transistors but by the end of the 1960s, these were replaced by silicon transistors, which are more uniform in performance and less affected by changes in temperature and by source impedance) - low impedance signals that pass through germanium-equipped fuzzes tend to suffer from a pronounced drop in volume and bass response. In the 2000s, many boutique guitar effects builders offer fuzz pedals with germanium transistors for a "retro" sound. Some units employ both silicon and germanium transistors.

Physics of clipping

As clipping is a non-linear process, intermodulation will occur, leading to the generation of an output signal rich in extra harmonics of the input signal. Intermodulation distortion also produces frequency components at the various sums and differences of the frequency components of the input signal. In general, these components will not be harmonically related to the input signal, leading to dissonance. To reduce unwanted dissonance, simple power chords (root, fifth, and octave) are often used when using fuzzboxes, rather than triads (root, third, and fifth) or four-note chords (root, third, fifth, and seventh).



Waveform plot showing the different types of clipping. Valve overdrive is a form of soft limiting, while transistor clipping or extremely overdriven valves resemble hard clipping.

Literally, the word distortion refers to any aberration of the waveform of an electronic circuit's output signal from its input signal. In the context of musical instrument amplification, it refers to various forms of clipping, which is the truncation of the part of an input signal that exceeds certain voltage limits. Because both valves and transistors behave linearly within a certain voltage region, distortion circuits are finely tuned so that the average signal peak just barely pushes the circuit into the clipping region, resulting in the softest clip and the least harsh distortion. Because of this, as the guitar strings are plucked harder, the amount of distortion and the resulting volume both increase, and lighter plucking cleans-up the sound.

Valve overdrive

Before the widespread adoption of the transistor, the traditional way to create gain (amplification) and distortion was through vacuum valves (called "tubes" in North America). A vacuum valve has a maximum input voltage determined by its bias and a minimum input voltage determined by its supply voltage. When any part of the input waveform approaches these limits, the valve's amplification becomes less linear, meaning that smaller voltages get amplified more than the large ones. This causes the peaks of the output waveform to be compressed, resulting in a waveform that looks "squashed". This is known as "soft clipping", and emphasizes the lower rather than higher-order harmonics that add to the warmth and richness of the guitar's tone. If the valve is driven harder, the compression becomes more extreme and the peaks of the waveforms are clipped. This adds additional odd-order harmonics, creating a "dirty" or "gritty" tone. The harmonics are entirely odd for a symmetrically-distorted signal. Asymmetrical distortion is also typical of valve amplifiers, where the positive swing of the AC signal may be distorted differently than the negative swing. Asymmetrical distortion results in even harmonics, which add a unique tone to the valve-generated distortion.

Valve distortion is commonly referred to as overdrive, as it is attained by driving the valves in an amplifier at a higher level than can be handled cleanly. Multiple stages of valve gain/clipping can be "cascaded" to produce a thicker and more complex distortion sound. In some modern valve effects, the "dirty" or "gritty" tone is actually achieved not by high voltage, but by running the circuit at voltages that are too low for the circuit components, resulting in greater non-linearity and distortion. These designs are referred to as "starved plate" configurations, and result in an "amp death" sound.

Transistor clipping

Transistor clipping stages, on the other hand, behave far more linearly within their operating regions, and thus faithfully amplify the instrument's signal until the input voltage falls outside its operating region, at which point the signal is clipped without compression, known as "hard clipping" or limiting. This type of distortion tends to produce odd harmonics at higher orders due to the sharpness of the 'corners' put into the time domain signal. Electronically, this is usually achieved by either amplifying the signal to a point where it must be clipped to the supply rails, or by clipping the signal across diodes. Many solid state distortion devices attempt to emulate the sound of overdriven vacuum valves.

Approaches

Guitar distortion can be produced by many components of the guitar's signal path, including effects pedals, the pre-amplifier, power amplifier, and speakers. Many players use a combination of these to obtain their "signature" tone.

Overdrive/distortion pedals

The Ibanez TS-9 Tube Screamer, a popular overdrive pedal

Because they are often designed to operate with low voltages (such as 9 volt batteries), overdrive and distortion pedals typically use transistors to generate distortion. Classic examples include the Ibanez Tube Screamer and the Electro-Harmonix Big Muff. A few more modern effects pedals incorporate valves; usually these still run at voltages that are below the valve's design specifications, resulting in a "starved plate" configuration that some people feel generates harsh and buzzy distortion. Distortion pedals usually also provide signal gain, which can be used to drive the input stage of the pre-amplifier harder, resulting in further distortion and, in some cases, higher volume.

Pre-amplifier distortion

The pre-amplifier section of a guitar amplifier serves to amplify a weak instrument signal to a level that can drive the power amplifier. It often also contains circuitry to shape the tone of the instrument, including equalization and gain controls. Often multiple cascading gain/clipping stages are employed to generate distortion. Because the first component in a valve amplifier is a valve gain stage, the output level of the preceding elements of the signal chain has a strong influence on the distortion created by that stage. The output level of the guitar's pickups, the setting of the guitar's volume knob, how hard the strings are plucked, and the use of volume-boosting effects pedals can drive this stage harder and create more distortion.

During the 1980s and 1990s, many amps featured a "master volume" control, essentially an adjustable attenuator between the preamp section and the power amp that conveniently enables the generation of high distortion levels in the guitar amp's preamp section while diverting most of the resulting signal away from the power valves, keeping the output volume at manageable levels. However, this also results in the power valves being operated well within their linear region, reducing the distortion that they add to the output signal.

Solid-state gain/clipping stages are also employed in many amplifiers. Some amplifiers (notably the Marshall JCM900) utilize hybrid designs that employ both valve and solid-state components.

Power amplifier distortion

A pair of 6L6GC power valves, often used in American-made amplifiers

Power valves can be overdriven in the same way that pre-amplifier valves can, but because these valves are designed to output more power, the distortion and character they add to the guitar's tone is unique. During the 1960s to early 1970s, distortion was primarily created by overdriving the power valves. Because they have become accustomed to this sound, many guitar players favour this type of distortion, and thus set their amps to maximum levels in order to drive the power section hard. Many valve-based amplifiers in common use have a balanced configuration in their power section, with matched pairs of tubes driving the output transformer in opposite phases, power amplifier distortion is entirely symmetric, generating only odd-order harmonics and does not generate even-order harmonics (unless the tubes are not a matched pair are poorly matched to each other and so have very different transfer characteristics).

Because driving the power valves this hard also means maximum volume, which can be difficult to manage in a small recording or rehearsal space, many solutions have emerged that in some way divert some of this power valve output from the speakers, allow the player to generate power valve distortion without excessive volume. These include built-in or separate power attenuators and power-supply-based power attenuation. Lower-power valve amps (such as a quarter-watt or less), speaker isolation boxes, and low-efficiency guitar speakers are also used to tame the volume.

Although traditional amplifiers were complete circuits including both preamp and power amp, power-valve distortion can also be produced in a dedicated rackmount valve power amp. A modular rackmount setup often involves a rackmount preamp, a rackmount valve power amp, and a rackmount dummy load to attenuate the output to desired volume levels. Some effects pedals internally produce power-valve distortion, including an optional dummy load for use as a power-valve distortion pedal. Such effects units can use a preamp valve such as the 12AX7 in a power-valve circuit configuration (as in the Stephenson's Stage Hog), or use a conventional power valve, such as the EL84 (as in the H&K Crunch Master compact tabletop unit). However, because these are usually placed before the pre-amplifier in the signal chain, they contribute to the overall tone in a different way.

A Direct Inject signal can capture the power-tube distortion sound without the direct coloration of a guitar speaker and microphone. This DI signal can be blended with a miked guitar speaker, with the DI providing a more present, immediate, bright sound, and the miked guitar speaker providing a colored, remote, darker sound. The DI signal can be obtained from a DI jack on the guitar amp, or from the Line Out jack of a power attenuator.

Output transformer distortion

The output transformer sits between the power valves and the speaker, serving to match impedance and voltage. When a transformer's ferromagnetic core becomes electromagnetically saturated, it will clip symmetrically in a correctly-operating amplifier, since there is negligible net DC current in either coil. This adds additional odd-order distortion to the signal delivered to the speakers.

Power supply "sag"

Early valve amplifiers usually used unregulated power supplies. This was due to the high cost associated with high-quality high-voltage power supplies. The typical anode supply was simply a rectifier, an inductor and a filter capacitor. When the valve amplifier was operated at high volume, the power supply voltage would dip, reducing power output and causing signal attenuation and compression. This dipping effect is known as "sag", and is sought after by some electric guitarists. Sag only occurs in Class AB amplifiers. This is because, technically, sag results from more current being drawn from the power supply, causing a greater voltage drop over the rectifier valve. In a Class A amplifier, current draw is constant, so sag does not occur.

As this effect is more pronounced with higher input signals, the harder "attack" of a note will be compressed more heavily than the lower-voltage "decay", making the latter seem louder and thereby improving sustain. Additionally, because the level of compression is affected by input volume, the player can control it via their playing intensity: playing harder results in more compression or "sag". In contrast, modern amplifiers often use high-quality, well-regulated power supplies. In theory, these keep the supply voltage constant, but in reality there is still some small variation, largely due to resistive losses in the cabling from the power supply to the gain stage.

Speaker distortion

Guitar loudspeakers are designed differently from high fidelity stereo speakers or PA system speakers. While hi-fi and PA speakers are designed to reproduce the sound with as little distortion as possible, guitar speakers are usually designed so that they will shape or colour the tone of the guitar, either by enhancing some frequencies or attenuating unwanted frequencies. As well, when the power delivered to a guitar speaker approaches its maximum rated power, the speaker's performance becomes less linear, causing the speaker to "break up", adding further distortion and colouration to the signal. Some speakers are designed to have lots of clean headroom, while others are designed to break up early to deliver grit and growl.

Amp modeling for distortion emulation

Guitar amp modeling devices and software can reproduce various guitar-specific distortion qualities that are associated with a range of popular "stomp box" pedals and amplifiers. Amp modeling devices typically use digital signal processing to recreate the sound of plugging into analogue pedals and over driven valve amplifiers. The most sophisticated devices allow the user to customize the simulated results of using different preamp, power-tube, speaker distortion, speaker cabinet, and microphone placement combinations. For example, a guitarist using a small amp modeling pedal could simulate the sound of plugging their electric guitar into a heavy vintage valve amplifier and a stack of 8 X 10" speaker cabinets. Some modeling devices allow even more detailed simulation, such as the different tonal effect that would occur from miking the speaker cabinets with a cardioid microphone or a ribbon mic.

Amplifier and distortion modeling can be accomplished by using different methods: real-time software running on a computer: hardware such as a compact pedal, oversize pedal, rack mount processor, desktop or floor processor; guitar amp heads, including hybrid valve amps that use both analog and digital technology. As sound of the highest-end modeling devices can be very convincing, these processors are widely used for live performances, because it reduces the amount of heavy, vintage amplifiers that guitarists have to transport. Some smaller independent recording studios use modeling software or processors, because they allow performers to produce widely-used "classic tones" without having to purchase or rent expensive vintage equipment or record at high volume levels. Digital modeling devices may not be able to recreate all the subtle aspects of the sound of vintage, over driven valve amplifiers, because this sound is the product of a range of non-linear and random factors, ranging from the heat of the vacuum valves to the age and condition of the speakers. As a result, professional musicians tend to use actual vintage valve amps for recordings, because the recorded sound will have to stand up to greater scrutiny from listeners and critics. Another aspect in this discussion is the latency of the digital modelling device output signals which can influence the players feeling while playing.

Voicing with equalization

Rock guitar distortion is obtained and shaped at various points in the signal processing chain, including multiple stages of preamp distortion, power valve distortion, output and power transformer distortion, and guitar speaker distortion. Much of the distortion character or voicing is controlled by the frequency response curve before and after each distortion stage. This dependency of distortion voicing on frequency response can be heard in the effect that a wah pedal has on the subsequent distortion stage, or by using an EQ pedal to favor the bass or treble components of the guitar pickup signal prior to the first distortion stage. Some guitarists place an equalizer pedal after the distortion effect, to emphasize or de-emphasize different frequencies and create different tonal coloration.

A guitar amplifier's tone controls shape a different power-valve distortion voicing if the tone controls are set to emphasize the bass or treble. Extreme settings are most popular in heavy metal. Increasing the bass and treble while reducing or eliminating the centre midrange (750 Hz) results in what is popularly known as a "scooped" sound (since the midrange frequencies are "scooped" out). James Hetfield of Metallica used this aggressive-sounding tone on many songs on Metallica's first four studio albums. Conversely, decreasing the bass while increasing the midrange and treble creates a punchy, harsher sound. Kerry King and Jeff Hanneman of Slayer have both used midrange-heavy tones since the mid-1980s.

Chapter-10

Equalization and Chorus Effect

Equalization

The equaliser section from the Audient ASP8024 Mixing console. The upper section has high and low shelving EQ, the lower section has fully parametric EQ.

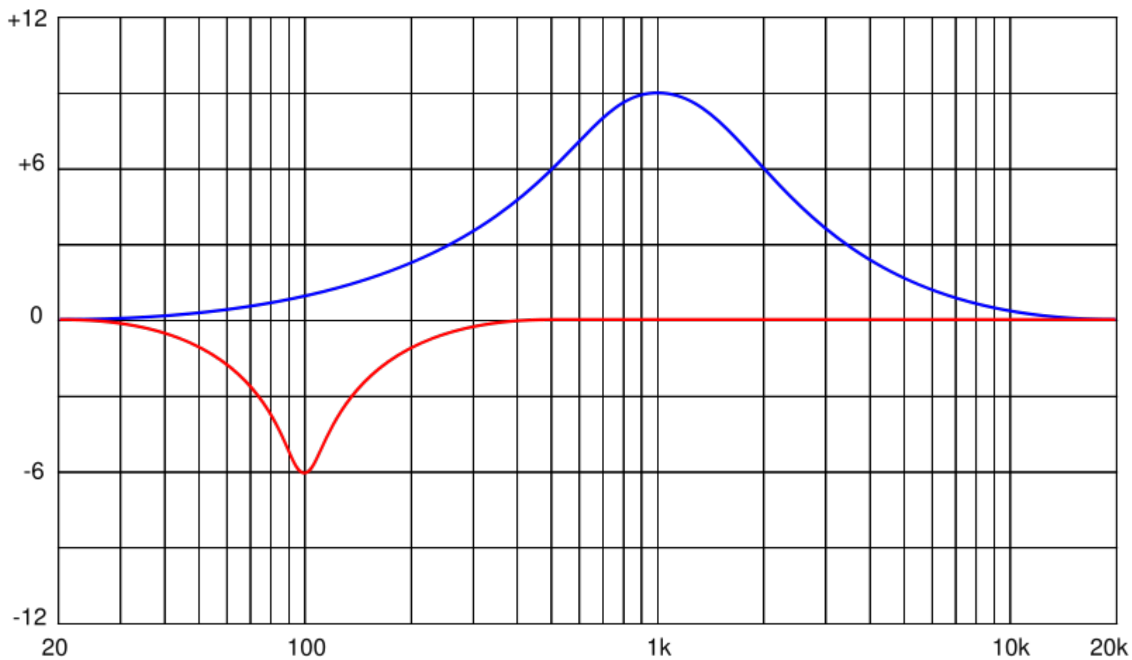
Equalization, (British: **equalisation**) is the process of adjusting the strength of certain frequencies within a signal. The most well known use of equalization is in sound recording and reproduction but there are many other applications in electronics and telecommunications.

The circuit or equipment used to achieve equalization is called an equalizer. These devices strengthen (*boost*) or weaken (*cut*) the energy of specific frequency bands. In telecommunications, equalizers are used to render the frequency response—for instance of a telephone line—*flat* from end-to-end. However, in sound reproduction, equalization has come to mean the adjustment of frequency responses for aesthetic reasons, which usually produces a response that is not flat. The term **EQ** specifically refers to this variant of equalization. For instance, a recording engineer may use an equalizer to make some high-pitches in a vocal part louder while making low-pitches in a drum part quieter.

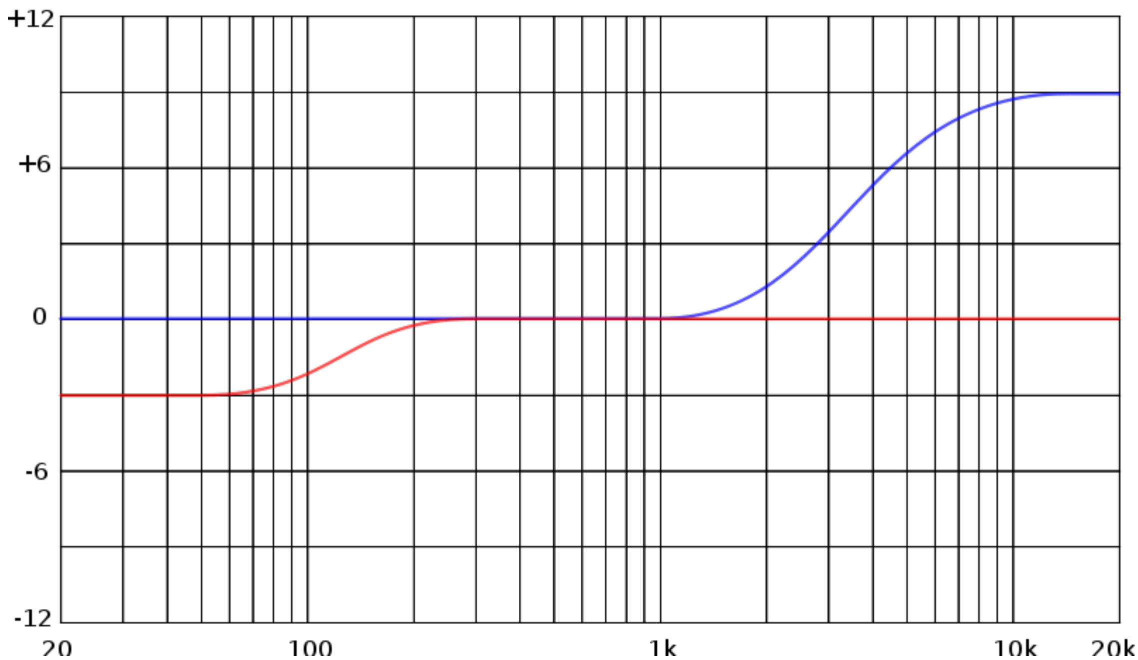
Stereos typically have equalizers that boost bass or treble frequencies. Telephones, DSL lines and television cables use equalizers to prepare data signals for transmission. Broadcast and recording studios use highly sophisticated equalizers to eliminate unwanted sounds, make certain instruments or voices more prominent, and enhance particular aspects of an instrument's tone.

Amplitude is not the only parameter equalization is applied to. Any frequency-dependent parameter can be subject to equalization, and in particular phase equalization and time-delay equalization can be important in telecommunications, especially lines carrying television signals.

Overview



Two examples of the frequency response of a peaking EQ



Two examples of the frequency response of a shelving EQ

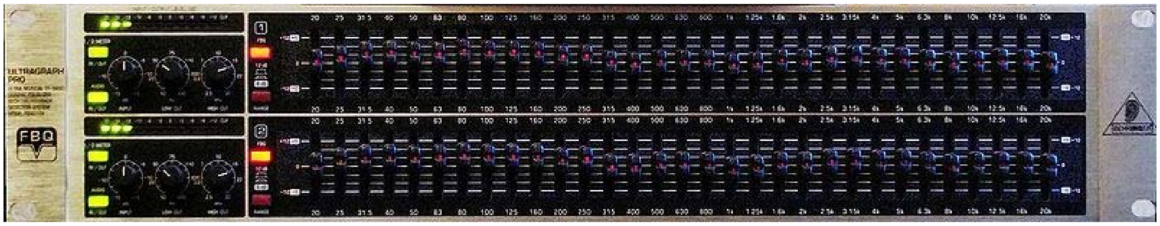
There are many kinds of EQ. Each has a different pattern of attenuation or boost. A peaking equalizer raises or lowers a range of frequencies around a central point in a bell shape. A peaking equalizer with controls to adjust the level (Gain), bandwidth (Q) and center frequency (Hz) is called a parametric equalizer. If there is no control for the bandwidth (it is fixed by the designer) then it is called a *quasi-parametric* or *semi-parametric* equalizer.

A pass filter attenuates either high or low frequencies while allowing other frequencies to pass unfiltered. A high-pass filter modifies a signal only by taking out low frequencies; a low-pass filter only modifies the audio signal by taking out high frequencies. A pass filter is described by its *cut-off point* and *slope*. The cut-off point is the frequency where high or low-frequencies will be removed. The slope, given in decibels per octave, describes a ratio of how the filter attenuates frequencies past the cut-off point (e.g. 12 dB per octave). A band-pass filter is a combination (in series) of one high-pass filter and one low-pass filter, which together allow only a band of frequencies to pass, attenuating both high and low frequencies past certain cut-off points.

Shelving-type equalizers increase or attenuate the level of a wide range of frequencies by a fixed amount. A *low shelf* will affect low frequencies up to a certain point and then above that point will have little effect. A *high shelf* affects the level of high frequencies, while below a certain point, the low frequencies are unaffected.

Variable equalization was first used by John Volkman working at RCA in the 1920s. They were used to equalize a motion picture theater playback systems.

Graphic equalizer



31-band Behringer 1/3-octave graphic equalizer with LEDs that show signal presence per frequency band

One common type of equalizer is the *graphic equalizer*, which consists of a bank of sliders for boosting and cutting different bands (or frequency ranges) of sound. The number and width of filters depends on application. A simple car audio equalizer might have one bank of filters controlling two channels for easy adjustment of stereo sound, and contain five to ten filter bands. A typical equalizer for professional live sound reinforcement has some 25 to 31 bands, necessary for quick control of feedback tones and room modes. Such an equalizer is called a 1/3-octave equalizer (spoken informally as "*third-octave* EQ") because the center frequency of each filter is spaced one third of an octave away from its neighbors, three filters to an octave. Equalizers with half as many filters per octave are common where less precise general tone-shaping is desired—this design is called a 2/3-octave equalizer.

Stereo graphic equalizer, 2/3-octave, 15 bands per channel

Historically, the Langevin Model EQ-251A was the first equalizer to use slide controls. It featured two passive equalization sections, a bass shelving filter, and a pass band filter. Each filter had switchable frequencies and used a 15-position slide switch to adjust cut or boost. The first true graphic equalizer was the type 7080 developed by Art Davis's Cinema Engineering. It featured 6 bands with a boost or cut range of 8 dB. It used a slide switch to adjust each band in 1 dB steps. Davis's second graphic equalizer was the Altec Lansing Model 9062A EQ. In 1967 Davis developed the first 1/3 octave variable notch filter set, the Altec-Lansing "Acousta-Voice" system.

Uses

In Multitrack recording and sound reinforcement systems, individual channels have equalization for aesthetic reasons, while the combined mix of sound is processed through equalization for practical reasons. Nearly any acoustic space absorbs and reflects sound in ways that make some sound frequencies louder than others. This is due to standing waves produced by the size of the room and the materials in it. Equalization is used to compensate for the discrepancies of a room's acoustics. Ideally, a sound system produces a flat frequency response. The frequency response of a room is examined with a Spectrum analyzer and usually a graphic equalizer, with matching frequency bands, is used to compensate for the room acoustics. This is standard practice for sound recording studios, live sound reinforcement systems and some High fidelity sound systems.

One of the most direct uses of equalization is at a live event, where microphones and speakers operate simultaneously. An equalizer is used to ensure that there are no frequency bands where there is a round trip gain of greater than 1, as these are heard as audible feedback. Those frequencies are cut at the equalizer to prevent this.

Most audio records have had equalization applied to the sound waveform before the consumers' record was made because of the limitations of equipment for recording and manufacturing the record. One scheme was used prior to 1940. Some 100 formulae were used until 1955, when the RIAA standard formula was implemented. As an example of the use of equalization in record production, low frequencies are reduced before the sound is imprinted onto the vinyl, making the groove take up less physical space so that more music can fit on the record. For this reason, record players boost the low frequencies back up to their original level before playback, to compensate for the reduction during printing.

Early telephone systems used equalization to correct for the reduced level of high frequencies in long cables, typically using Zobel networks. These kinds of equalizers can also be used to produce a circuit with a wider bandwidth than the standard telephone band of 300 Hz to 3.4 kHz. This was particularly useful for broadcasters who needed "music" quality, not "telephone" quality on landlines carrying program material. It is necessary to remove or cancel any loading coils in the line before equalization can be successful. Equalization was also applied to correct the response of the transducers, for example, a particular microphone might be more sensitive to low frequency sounds than to high frequency sounds, so an equalizer would be used to increase the volume of the higher frequencies (*boost*), and reduce the volume of the low frequency sounds (*cut*).

Modern digital telephone systems have less trouble in the voice frequency range as only the local line to the subscriber now remains in analog format, but DSL circuits operating in the MHz range on those same wires may suffer severe attenuation distortion, which is dealt with by automatic equalization or by abandoning the worst frequencies. Picturephone circuits also had equalizers.

The individual channels of a mixing board and the sound of electric instruments are equalized for aesthetic reasons. Some guitar effects units, in particular, the wah-wah pedal is based on equalization. Equalization is used to manipulate the timbre of musical instruments and sounds.

Chorus effect

A **chorus effect** occurs when individual sounds with roughly the same timbre and nearly (but never exactly) the same pitch converge and are perceived as one. While similar sounds coming from multiple sources can occur naturally (as in the case of a choir or string orchestra), it can also be simulated using an electronic effects unit or signal processing device.

Methods

When the effect is produced successfully, none of the constituent sounds is perceived as being out of tune. Rather, this amalgam of sounds has a rich, shimmering quality which would be absent if the sound came from a single source. The effect is more apparent when listening to sounds that sustain for longer periods of time.

The chorus effect is especially easy to hear when listening to a choir or string ensemble. A choir has multiple people singing each part (soprano, tenor, etc.). A string ensemble has multiple violinists and possibly multiples of other stringed instruments. When individual singers or violins play the same part, the chorus effect can be heard. Some instruments produce the effect all on their own. Examples include:

- Piano - Each hammer strikes multiple strings tuned to nearly the same pitch (for all notes except the bass notes). The chorus effect is so intrinsic to the timbre of a piano that it is difficult to recognize.
- 12 string guitar - Six pairs of strings tuned in octaves and unisons create a chorus effect.
- Synthesizer. The effect is achieved by assigning multiple, slightly detuned oscillators to a voice. This is referred to as "Unison" by some manufacturers.

Electronic effect

SmallClone chorus unit

The chorus effect can be simulated by signal processing equipment. The signal processor may be software running on a computer, a ROM-encoded effect in a digital effect processor, or an analog effect processor. If the processor is hardware-based, it may be packaged as a foot pedal, a rack-mount module, a table-top device, or built in to an instrument amplifier (often an acoustic guitar amplifier or an electric guitar amplifier). Some keyboard instruments have an electronic chorus effect built in, such as some electronic pianos and some Hammond organs.

Regardless of the technology or form factor, the processor achieves the effect by taking an audio signal and mixing it with one or more delayed, pitch-modulated copies of itself. The pitch of the added voices is typically modulated by an LFO, which makes the overall effect similar to that of a flanger, except with longer delays and without feedback.

Stereo chorus effect processors produce the same effect, but it is varied between the left and right channels by offsetting the delay or phase of the LFO. The effect is thereby enhanced because sounds are produced from multiple locations in the stereo field. Used on instruments like "clean" (undistorted) electric guitar and keyboards, it can yield very dreamy or ambient sounds. Commercial chorus effect devices often include controls that enable them to be used to also produce delay, reverberation, or other related effects that use similar hardware, rather than exclusively as chorus effects.

Chapter-11

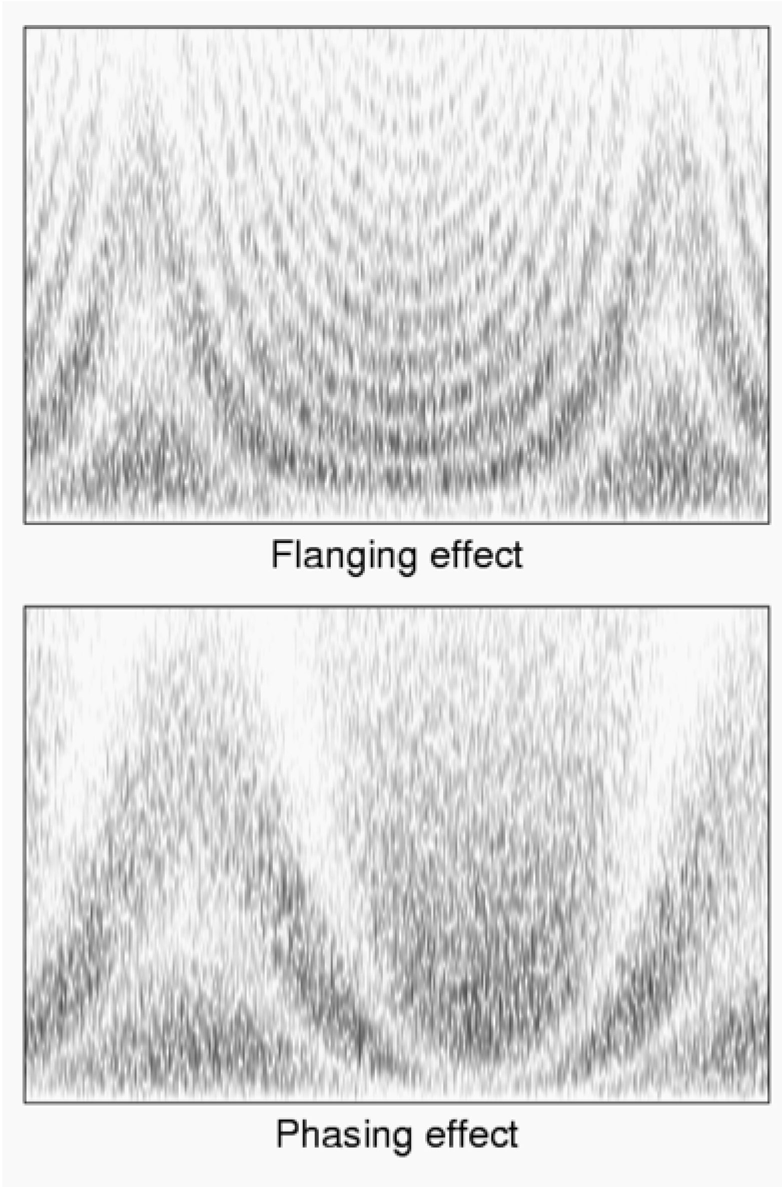
Flanging

Flanging is an audio effect produced by mixing two identical signals together, with one signal delayed by a small and gradually changing period, usually smaller than 20 milliseconds. This produces a swept comb filter effect: peaks and notches are produced in the resultant frequency spectrum, related to each other in a linear harmonic series. Varying the time delay causes these to sweep up and down the frequency spectrum.

Part of the output signal is usually fed back to the input (a "re-circulating delay line"), producing a resonance effect which further enhances the intensity of the peaks and troughs. The phase of the fed-back signal is sometimes inverted, producing another variation on the flanging sound.

A "flanger" is a device dedicated to creating this sound effect.

Comparison with phase shifting



Spectrograms of Phasing and Flanging effects

This flanging is one specific type of phase-shifting or "phasing". In phasing, the signal is passed through one or more all-pass filters which have non-linear phase response, and then added back to the original signal. This results in constructive and destructive interference that varies with frequency, giving a series of peaks and troughs in the frequency response of the system. In general, the position of these peaks and troughs do not occur in a harmonic series.

In contrast, flanging relies on adding the signal to a uniform time-delayed copy of itself, which results in an output signal with peaks and troughs which *are* in a harmonic series.

Extending the comb analogy, flanging yields a comb filter with regularly spaced teeth, whereas phasing results in a comb filter with irregularly spaced teeth.

In both phasing and flanging, the characteristics (phase response and time delay, respectively) are generally varied in time, leading to an audible sweeping effect.

To the ear, flanging and phasing sound similar, yet they are recognizable as distinct colorations.

Commonly, flanging is referred to as having a "jet plane-like" characteristic. In order for the comb filter effect to be audible, the spectral content of the program material must be full enough within the frequency range of this moving comb filter to reveal the filter's effect. It is more apparent when it is applied to material with a rich harmonic content, and is most obvious when applied to a white noise or similar noise signal.

If the frequency response of this effect is plotted on a graph, the trace resembles a comb, and so is called a comb filter.

Origin

The name "flanging" comes from the original method of creation. Originally, a signal would be recorded to two tape machines simultaneously. The playback-head output from these two recorders was then mixed together onto a third recorder. In this form, minute differences in the motor speeds of each machine would result in a phasing effect when the signals were combined. The "flange" effect originated when an engineer would literally put a finger on the flange, or rim of one of the tape reels so that the machine was slowed down, slipping out of sync by tiny degrees. A listener would hear a "drainpipe" sweeping effect as shifting sum-and-difference harmonics were created. When the operator removed his finger the tape sped up again, making the effect sweep back in the other direction.

Alternatively, the track could be recorded to two matching tape decks first, then replayed simultaneously with both decks closely in sync. With this method, slowing down one deck by pressing the tape reel flange would "sweep" the flange effect in one direction, but when released the playback of that deck would remain slightly behind the other, and the effect would not sweep back. Instead, pressing the flange of the other deck would sweep the effect back in the other direction as the tape position of the decks move toward being in sync again.

Older recording hardware could suffer from flanging as an undesired side effect when recording very long tracks. As the weight of the tape built up on one reel, the pressure on the capstans could cause flanging during mixdown or dubbing. This was one of the problems faced by studio engineers in the sixties and seventies when recording large concept pieces, as explained by Ian Anderson of Jethro Tull when recounting the studio challenges of recording *Thick as a Brick*.

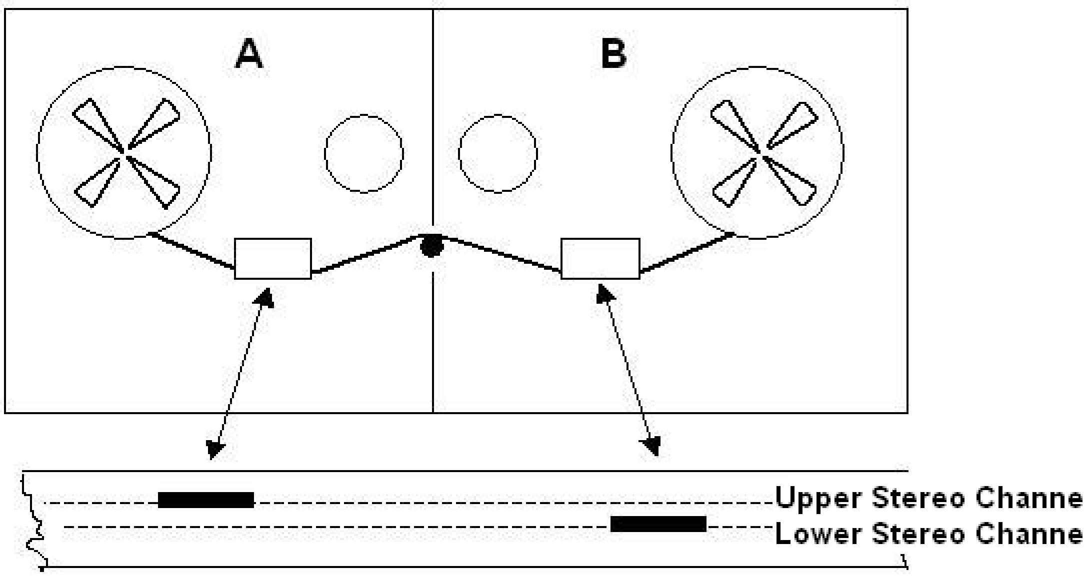
The development of the classic "flanging" effect is generally attributed to Ken Townsend, an engineer at EMI's Abbey Road Studio, who devised the process in the spring of 1966. Tired of the laborious process of re-recording dual vocal tracks, John Lennon asked Townsend if there was some way for the Beatles to get the sound of double-tracked vocals without actually doing the work. After mulling the problem over, Townsend devised Artificial Double Tracking or "ADT." According to historian Mark Lewisohn, it was Lennon who actually gave the process the name "flanging." Lennon asked Beatles producer George Martin to explain how ADT worked, and Martin answered with the nonsense explanation, "Now listen, it's very simple. We take the original image and we split it through a double-bifurcated sploshing flange with double negative feedback." From that point on, whenever Lennon wanted a Beatles song double-tracked, he would ask for "Ken's flanger". According to Lewisohn, "The Beatles' influence was so vast that the term "flanging" is still in use today, more than 20 years on." The first Beatles track to feature flanging was "Tomorrow Never Knows" from *Revolver*, which was recorded on April 6, 1966. When *Revolver* was completed and released on August 5, 1966, almost every song on the album had been subjected to flanging.

Others have attributed it to George Chkiantz, an engineer employed at Olympic Studios in Barnes, London. One of the first instances of the sound being used on a commercial pop recording was The Small Faces' 1967 single "Itchycoo Park", recorded at Olympic and engineered by Chkiantz's colleague Glyn Johns.

However there are competing claims for the first recorded use of the technique. One is that the technique was pioneered by the BBC Radiophonic Workshop, who published their experiments on radio shows such as the Goon Show in freely available journals. ("Flange" was one of many words used out of context on the show to confuse/amuse the audience).

American music industry veterans David S. Gold and Stan Ross, founders of the renowned Gold Star Studios in Hollywood, claim that they made the first commercial recording to feature the technique — the single "The Big Hurt" by Toni Fisher, which was recorded at Gold Star in late 1959 and became a hit in the U.S. in early 1960, rising to #3 on Billboard magazine's singles chart. Flanging is also heard in the opening of The Ventures' 1962 cover version of The Tornados hit "Telstar", in the context of a simulated rocket launch sound effect.

The first use of the flanging effect in stereo is credited to producer Eddie Kramer, who used the effect in the coda of Jimi Hendrix's "Bold as Love" (1967). Kramer admitted in an 1990s interview that he read BBC Radiophonic Workshop technical journals for ideas and circuit diagrams.



Kendrick's setup to control flanging

In 1969, the record producer for The Litter, Warren Kendrick, devised a method to precisely control the flanging effect by placing two 15 IPS (inches per second) stereo Ampex tape recorders side-by-side. The take-up reel of recorder A and supply reel of recorder B were disabled, as were channel 2 of recorder A, channel 1 of recorder B and the erase head of recorder B. The tape was fed left-to-right across **both** recorders and an identical signal was recorded on each channel of the tape, but displaced by approximately 18 inches along the length of the tape. During the recording, a screwdriver was wedged between the tape recorders to make the tape run "uphill" and "downhill." The same configuration was employed during the playback/mixdown to a third recorder. The screwdriver was moved back and forth to cause the two signals to diverge, then converge. The latter technique permits zero point flanging; i.e., the lagging signal crosses over the leading signal and the signals change places.

A similar “jet plane-like” sound effect can also occur naturally in long distance shortwave radio music broadcasts. In this case the varying delays are caused by varying radio wave propagation times and multipath radio interference.

Artificial flanging

In the 1970s, advances in solid state electronics made the flanging effect possible using integrated circuit technology. Solid state flanging devices fall into two categories: analog and digital. One of the most famous flanger pedals is the Electro Harmonix Electric Mistress. The flanging effect in most newer digital flangers relies on DSP technology. Flanging can also be accomplished using computer software.

Note that the original tape-flanging effect sounds a little different from the later electronic and software re-creations. Not only is the tape-flanging signal time-delayed, but the response characteristics at different frequencies of the magnetic tape and tape heads inevitably introduced some phase shifts into the signals as well. Thus, while the peaks and troughs of the comb filter are more or less in a linear harmonic series, there is a significant amount of non-linear behaviour too, causing the timbre of tape-flanging to sound more like a combination of what came to be known as flanging and phasing.

"Barber-pole" flanging

Also known as "infinite flanging", this sonic illusion is similar to the Shepard tone effect, and is equivalent to an auditory 'barber pole'. The sweep of the flanged sound seems to move in only one direction ("up" or "down") infinitely, instead of sweeping back-and-forth. While Shepard tones are created by generating a cascade of tones, fading in and out while sweeping the pitch either up or down, barber pole flanging uses a cascade of multiple delay lines, fading each one in to the mix and fading it out as it sweeps to the delay time limit. The effect is available on various hardware and software effect systems.

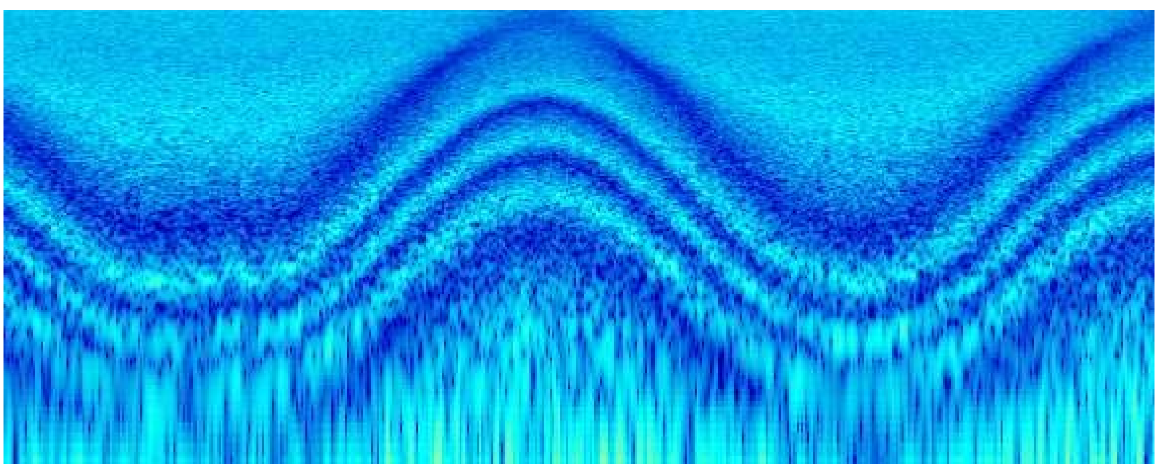
Chapter-12

Phaser (Effect) and Noise Gate

Phaser

A **phaser** is an audio signal processing technique used to filter a signal by creating a series of peaks and troughs in the frequency spectrum. The position of the peaks and troughs is typically modulated so that they vary over time, creating a sweeping effect. For this purpose, phasers usually include a low frequency oscillator.

Process



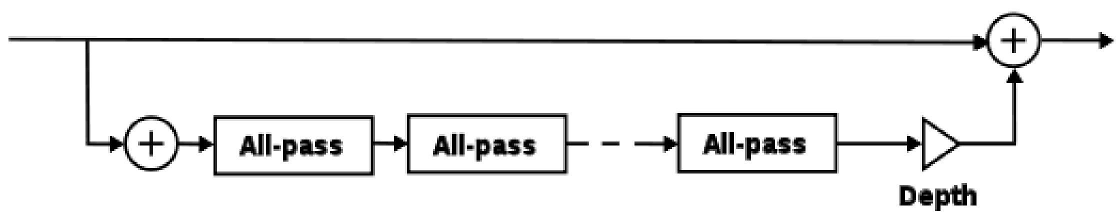
Spectrogram of an 8-stage phaser modulated by a sine LFO applied to white noise.

The electronic phasing effect is created by splitting an audio signal into two paths. One path treats the signal with an all-pass filter, which preserves the amplitude of the original signal and alters the phase. The amount of change in phase depends on the frequency. When signals from the two paths are mixed, the frequencies that are out of phase will cancel each other out, creating the phaser's characteristic notches. Changing the mix ratio changes the depth of the notches; the deepest notches occur when the mix ratio is 50%.

The definition of phaser typically excludes such devices where the all-pass section is a delay line; such a device is called a flanger. Using a delay line creates an unlimited series of equally spaced notches and peaks. It is possible to cascade a delay line with another type of all-pass filter as in, this combines the unlimited number of notches from the flanger with the uneven spacing of the phaser.

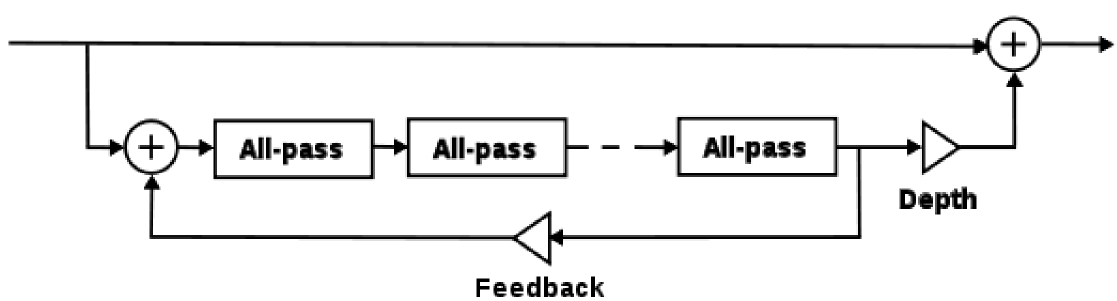
Structure

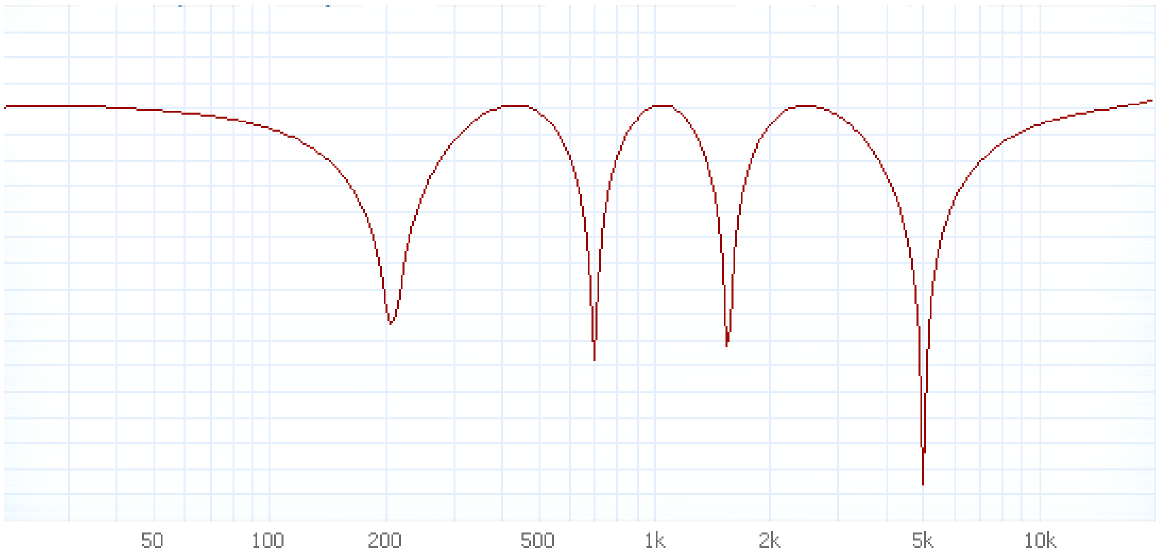
Traditional electronic phasers use a series of variable all-pass phase-shift networks which alter the phases of the different frequency components in the signal. These networks pass all frequencies at equal volume, introducing only phase change to the signal. Human ears are not very responsive to phase differences, but this creates audible interferences when mixed back with the dry (unprocessed) signal, creating notches. The simplified structure of a mono phaser is shown below:



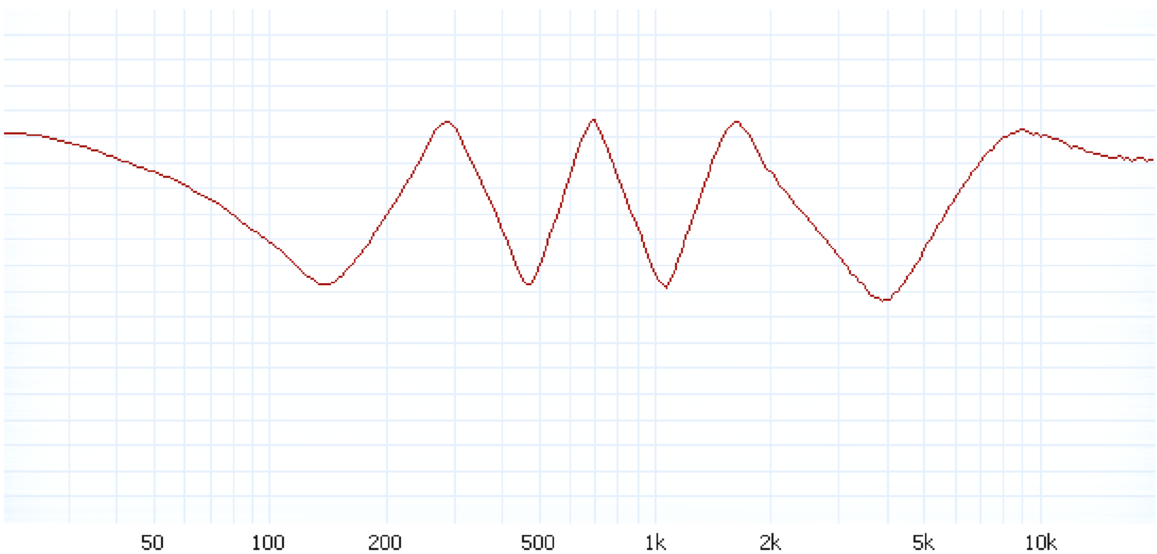
The number of all-pass filters (usually called *stages*) varies with different models, some analog phasers offer 4, 6, 8 or 12 stages. Digital phasers may offer up to 32 or even more. This determines the number of notches/peaks in the sound, affecting the general sound character. A phaser with n stages generally has n/2 notches in the spectrum, so a 4-stage phaser will have two notches.

Additionally, the output can be fed back to the input for a more intense effect, creating a resonant effect by emphasizing frequencies between notches. This involves feeding the output of the all-pass filter chain back to the input, as shown here:





Measured frequency response of an 8-stage phaser with no feedback, dry/wet ratio: 50/50%



Measured frequency response of an 8-stage phaser with 50% feedback, dry/wet ratio: 50/50%

The frequency response of an 8-stage phaser with or without feedback is shown. Note that the peaks between the notches are sharper when there's feedback, giving a distinct sound.

A stereo phaser is usually two identical phasers modulated by a quadrature signal; the output of the oscillators for the left and right channels are a quarter-wave out of phase.

Most modern phasers are a part of a digital signal processor, often trying to emulate analog phasers. Phasers are mostly found as plugins for sound editing software, as a part of a monolithic rackmount sound effect unit, or as "stompbox" guitar effects.

Usage

Phasing is a popular effect for electric guitar. The term was often used to refer the original tape flanging effect heard on many psychedelic records of the late 1960s, notably "Itchycoo Park" by the Small Faces and "Life in the Fast Lane" by The Eagles. By the early 1970s, phasing was available as a portable guitar effect, one of the most notable early examples being the MXR Phase 90. In the late 1970s, Brian May used large amounts of phasing, in such songs as "Sheer Heart Attack." In the 1980s, Eddie Van Halen, for instance, often used the MXR Phase 90 as part of his signal chain.

Keyboard players also used phasing: in the 1970s, keyboard instruments like the Rhodes, the Eminent 310, and the Clavinet were commonly treated with a phaser, especially in avant-garde jazz. Bill Evans, for instance, used a Maestro phaser on *Intuition*. The phaser is also used to "sweeten" their sounds. Examples can be heard in Billy Joel's "Just The Way You Are", Styx's "Babe", and Jean Michel Jarre's *Oxygène*.

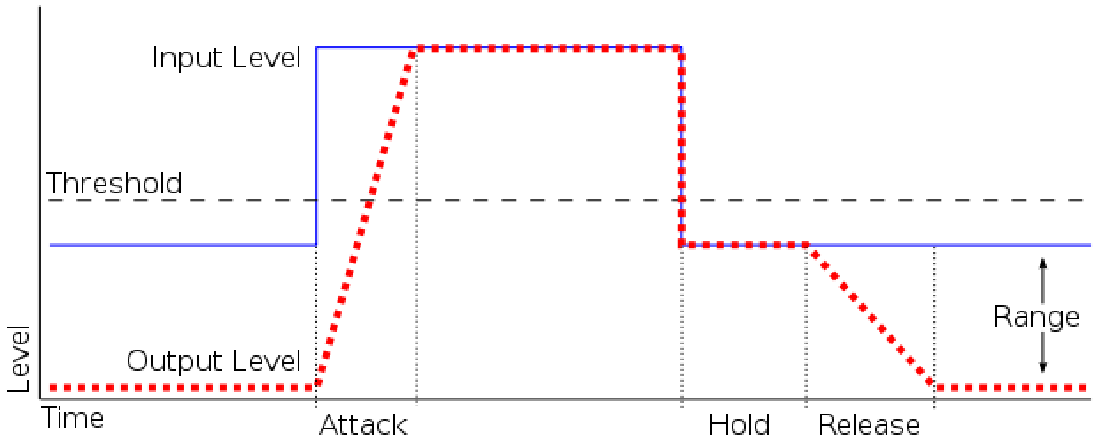
In motion picture or television production, the effect created by a phaser is often used to imply that the sound is synthetically generated, like turning a natural human voice into a computer or robot voice. The technique works because the frequency filtering produces sound commonly associated with mechanical sources, which only generate specific frequencies, rather than natural sources, which produce a range of frequencies. A vocoder is a different effect used for similar purposes.

Similar effects

A specific type of phasing, flanging is a similar effect, in which the notches are linearly spaced. In a flanger effect, the notches are created by mixing the signal with a delayed version of itself. Flangers tend to sound more pronounced and natural, like the "jet plane whoosh" effect, whereas phasers tend to sound more subtle and otherworldy. For comparison of the two effects, check flanging.

Noise gate

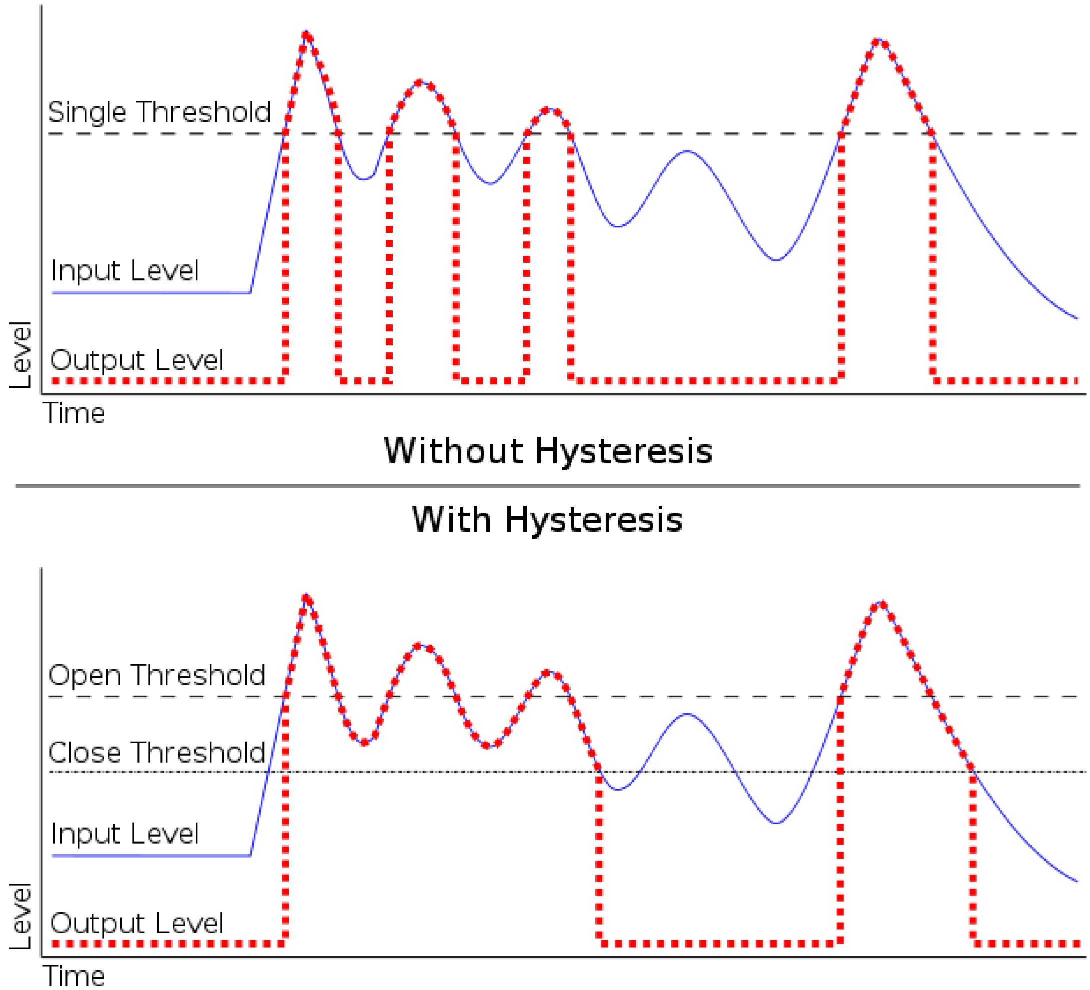
A **Noise Gate** or **gate** is an electronic device or software that is used to control the volume of an audio signal. In its most simple form, a noise gate allows a signal to pass through only when it is above a set threshold: the gate is 'open'. If the signal falls below the threshold no signal is allowed to pass (or the signal is substantially attenuated): the gate is 'closed'. A noise gate is used when the level of the 'signal' is above the level of the 'noise'. The threshold is set above the level of the 'noise' and so when there is no 'signal' the gate is closed. A noise gate does not remove noise from the signal. When the gate is open both the signal and the noise will pass through.



The Attack, Hold, and Release functions of a noise gate.

They are commonly used in the recording studio and sound reinforcement. Rock musicians may also use small portable units to control unwanted noise from their amplification systems. Band-limited noise gates are also used to eliminate background noise from audio recordings by eliminating frequency bands that contain only static.

Noise Gates have a *Threshold* control to set the level at which the gate will open. More advanced noise gates have more features. The *Release* sets the amount of time for the gate to go from open to fully closed. A fast release abruptly cuts off the sound once it has fallen below the threshold, a slower release smoothly changes from open to closed, much like a slow fade out. If the release time is too short a click can be heard when the gate re-opens. Release is the most common control to find on a gate, after Threshold.



A noise gate without hysteresis can open and close undesirably with a fluctuating signal (top). With hysteresis the noise gate does not 'chatter'.

The *Attack* control sets the time for the gate to change from closed to open, much like a fade-in. The *Hold* control allows you to define the amount of time the gate will stay open after the signal falls below the threshold. This is useful during short pauses between words or sentences in a speech signal.

The amount of attenuation when the gate is closed can be set by the *Range* control. Often there will be complete attenuation, that is no signal will pass when the gate is closed. In some circumstances complete attenuation is not desired and the range can be changed.

Advanced gates have a *sidechain*. This is an additional input that allows the gate to be triggered by another audio signal.

A variation of a sidechained noise gate used in electronic music production is a *trigger gate*, *trancegate* or just simply *gate*, where the noise gate is not controlled by audio signal but a preprogrammed pattern resulting in a precisely controlled chopping of a sustained sound.

Noise gates often implement hysteresis, that is, they have two thresholds. One to open the gate and another, set a few dB below, to close the gate. This means that once a signal has dropped below the close threshold, it has to rise to the open threshold for the gate to open, so that a signal that crosses over the close threshold regularly does not open the gate and cause chattering. A longer hold time as described above also helps avoid chattering.

Audio noise reduction

In audio post-processing, noise gating reduces steady noise sources such as rumble from LP records, hiss from audio tape, static from a radio or amplifier, and hum from a power system, without greatly affecting the source sound. An audio signal such as music or speech is broken up into many frequency bands by a collection of overlapping band-pass filters, and if the signal amplitude in any one band is lower than a preset threshold then that band is eliminated from the final sound. This greatly reduces perceptible background noise because only the frequency components of the noise that are within the gated passbands survive.

The technique was implemented in real-time electronics in some audiophile record players as early as the 1980s, and is now commonly used in audio production post-processing, where software to Fourier transform the audio signal can yield a very detailed spectrum of the background noise. Common digital audio editing software packages such as CoolEdit and Audacity include easy-to-use digital noise gating code: the user selects a segment of audio that contains only static, and the amplitude levels in each frequency band are used to determine the threshold levels to be applied across the signal as a whole.

Noise gating works well when the static is steady and either narrowly confined in frequency (e.g. hum from AC power) or well below the main signal level (15 dB minimum is desirable). In cases where the signal merges with the background static (for example, the brushed drum sounds in the Sun King track on the Beatles album Abbey Road) or is weak compared to the noise (as in very faint tape recordings), the noise gating can add artifacts that are more distracting than the original static.

In the context of a multi-microphone recording session, noise gating is employed to reduce the leakage of sound into a microphone from sources other than the one the microphone was intended for. One example involves the mic-ing up of a drumkit. In most multi-mic drum recordings one microphone will be used to capture the snare drum sound and another to capture the kick drum sound. The snare microphone will output a signal composed of a high level snare signal and a lower level kick drum signal (due to the further distance of the kick drum from the snare microphone). If the threshold level of the noise gate is set correctly a snare drum signal can be isolated. To fully isolate the snare drum signal the release rate has to be quite fast which can cause the tail end of the snare sound to be 'chopped off'. This can usually be remedied by the inclusion of one or more 'overhead' microphone(s), which can act as a general 'audio glue' for all the other gated sources.

For vocal applications on stage an optical microphone switch may be used. An infra-red sensor senses if somebody is in front of the microphone and switches on the microphone.

Recording usages

A good example of time-controlled noise gating is the well-known "gated reverb" effect heard on the drums on the Phil Collins hit single "In the Air Tonight", created by engineer-producer Hugh Padgham, in which the powerful reverberation added to the drums is cut off by the noise gate after a few milliseconds, rather than being allowed to decay naturally. This can also be achieved by: sending the 'dry' snare signal to the reverb (or other process) unit, inserting a noise gate on the path of the reverb signal and connecting the snare sound to the side chain of the gate unit. With the gate unit set to 'external sidechain' (or 'external key') the gate will respond to the snare signal level and 'cut off' when that has decayed below the threshold, not the reverberated sound.

It is a common production trick to use spurious combinations of side chain inputs to control longer, more sustained sounds. For example, a hi-hat signal can be used to control a sustained synthesized sound to produce a rhythmic melodic (or harmonic) signal which is perfectly in time with the hi-hat signal. A good example of this use of the device can be found on the Godley & Creme concept album *Consequences*. The album's story required the creation of a number of special sound effects that would convey the impression of natural disasters. For the "Fire" sequence, Godley and Creme used a noise gate, triggered by the sound of multitracked voices, that created the 'voice' of a raging bushfire. During the recording of this segment, each time the voice signal began, it triggered the noise gate to open up another channel, which carried a pre-recorded loop of a crackling sound (created by overdubbing the sound of Bubble Wrap being popped in front of a microphone). The combined voices and crackling created an eerie and quite convincing 'talking fire' effect.

Chapter-13

Reverberation and Ring Modulation

Reverberation

Reverberation is the persistence of sound in a particular space after the original sound is removed. A reverberation, or **reverb**, is created when a sound is produced in an enclosed space causing a large number of echoes to build up and then slowly decay as the sound is absorbed by the walls and air. This is most noticeable when the sound source stops but the reflections continue, decreasing in amplitude, until they can no longer be heard. The length of this sound decay, or reverberation time, receives special consideration in the architectural design of large chambers, which need to have specific reverberation times to achieve optimum performance for their intended activity. In comparison to a distinct echo that is 50 to 100 ms after the initial sound, reverberation is many thousands of echoes that arrive in very quick succession (.01 – 1 ms between echoes). As time passes, the volume of the many echoes is reduced until the echoes cannot be heard at all.

Reverberation time

RT_{60} is the time required for reflections of a direct sound to decay by 60 dB below the level of the direct sound. Reverberation time is frequently stated as a single value however it can be measured as a wide band signal (20 Hz to 20kHz) or more precisely in narrow bands (one octave, 1/3 octave, 1/6 octave, etc.). Typically, the reverb time measured in narrow bands will differ depending on the frequency band being measured. It is usually helpful to know what range of frequencies are being described by a reverberation time measurement.

In the late 19th century, Wallace Clement Sabine started experiments at Harvard University to investigate the impact of absorption on the reverberation time. Using a portable wind chest and organ pipes as a sound source, a stopwatch and his ears, he measured the time from interruption of the source to inaudibility (roughly 60 dB). He found that the reverberation time is proportional to the dimensions of room and inversely proportional to the amount of absorption present.

The optimum reverberation time for a space in which music is played depends on the type of music that is to be played in the space. Rooms used for speech typically need a shorter reverberation time so that speech can be understood more clearly. If the reflected sound from one syllable is still heard when the next syllable is spoken, it may be difficult to understand what was said. "Cat", "Cab", and "Cap" may all sound very similar. If on the other hand the reverberation time is too short, tonal balance and loudness may suffer. Reverberation effects are often used in studios to add depth to sounds. Reverberation changes the perceived harmonic structure of a note, but does not alter the pitch.

Basic factors that affect a room's reverberation time include the size and shape of the enclosure as well as the materials used in the construction of the room. Every object placed within the enclosure can also affect this reverberation time, including people and their belongings.

Sabine equation

Sabine's reverberation equation was developed in the late 1890s in an empirical fashion. He established a relationship between the RT_{60} of a room, its volume, and its total absorption (in sabins). This is given by the equation:

$$RT_{60} = \frac{4 \ln 10^6}{c} \frac{V}{Sa} \approx 0.1611\,\mathrm{m}^{-1} \frac{V}{Sa}.$$

where c is a number that relates to the speed of sound in the room, V is the volume of the room in m³, S total surface area of room in m², a is the average absorption coefficient of room surfaces, and the product Sa is the total absorption in sabins.

The total absorption in sabins (and hence reverberation time) generally changes depending on frequency (which is defined by the acoustic properties of the space). The equation does not take into account room shape or losses from the sound traveling through the air (important in larger spaces). Most rooms absorb less sound energy in the lower frequency ranges resulting in longer reverb times at lower frequencies.

The reverberation time RT_{60} and the volume V of the room have great influence on the critical distance d_c (conditional equation):

$$d_c \approx 0.057 \cdot \sqrt{\frac{V}{RT_{60}}}$$

where critical distance d_c is measured in meters, volume V is measured in m³, and reverberation time RT_{60} is measured in seconds.

Absorption

The absorption coefficient of a material is a number between 0 and 1 which indicates the proportion of sound which is absorbed by the surface compared to the proportion which is reflected back into the room. A large, fully open window would offer no reflection as any sound reaching it would pass straight out and no sound would be reflected. This would have an absorption coefficient of 1. Conversely, a thick, smooth painted concrete ceiling would be the acoustic equivalent of a mirror, and would have an absorption coefficient very close to 0.

Measurement of reverberation time

Historically reverberation time could only be measured using a level recorder (a plotting device which graphs the noise level against time on a ribbon of moving paper). A loud noise is produced, and as the sound dies away the trace on the level recorder will show a distinct slope. Analysis of this slope reveals the measured reverberation time. Some modern digital sound level meters can carry out this analysis automatically.

Currently several methods exist for measuring reverb time. An impulse can be measured by creating a sufficiently loud noise (which must have a defined cut off point). Impulse noise sources such as a blank pistol shot or balloon burst may be used to measure the impulse response of a room.

Alternatively, a random noise signal such as pink noise or white noise may be generated through a loudspeaker, and then turned off. This is known as the interrupted method, and the measured result is known as the interrupted response.

A two port measurement system can also be used to measure noise introduced into a space and compare it to what is subsequently measured in the space. Consider sound reproduced by a loudspeaker into a room. A recording of the sound in the room can be made and compared to what was sent to the loudspeaker. The two signals can be compared mathematically. This two port measurement system utilizes a Fourier transform to mathematically derive the impulse response of the room. From the impulse response, the reverberation time can be calculated. Using a two port system, allows reverberation time to be measured with signals other than loud impulses. Music or recordings of other sound can be used. This allows measurements to be taken in a room after the audience is present.

Reverberation time is usually stated as a decay time and is measured in seconds. There may or may not be any statement of the frequency band used in the measurement. Decay time is the time it takes the signal to diminish 60 dB below the original sound.

Creating reverberation effects

It is often desirable to create a reverberation effect for recorded or live music. A number of systems have been developed to facilitate or simulate reverberation.

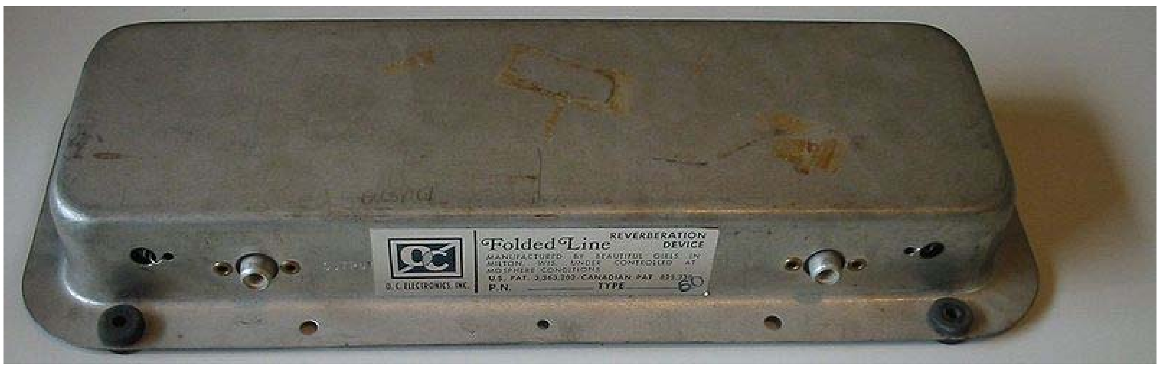
Chamber reverberators

The first reverb effects created for recordings used a real physical space as a natural echo chamber. A loudspeaker would play the sound, and then a microphone would pick it up again, including the effects of reverb. Although this is still a common technique, it requires a dedicated soundproofed room, and varying the reverb time is difficult.

Plate reverberators

A plate reverb system uses an electromechanical transducer, similar to the driver in a loudspeaker, to create vibration in a large plate of sheet metal. A pickup captures the vibrations as they bounce across the plate, and the result is output as an audio signal. Early units had one pickup for mono output, later models featured two pickups for stereo use. The reverb time can be adjusted by a damping pad, made from framed acoustic tiles. The closer the damping pad, the shorter the reverb time. However, the pad never touches the plate. Some units also featured a remote control.

Spring reverberators



Folded line reverberation device.

The folded coil spring is visible underside of the reverberation device.

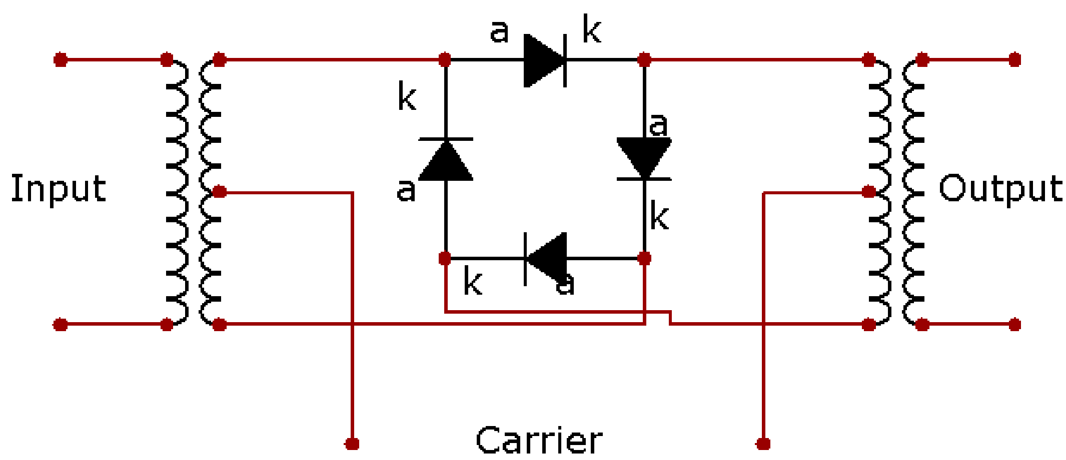
A spring reverb system uses a transducer at one end of a spring and a pickup at the other, similar to those used in plate reverbs, to create and capture vibrations within a metal spring. Guitar amplifiers frequently incorporate spring reverbs due to their compact construction and low cost. Spring reverberators were once widely used in semi-professional recording due to their modest cost and small size.

Many musicians have made use of spring reverb units by rocking them back and forth, creating a thundering, crashing sound caused by the springs colliding with each other. The Hammond Organ included an inbuilt spring reverberator, making this a popular effect when used in a rock band.

Digital reverberators

Digital reverberators use various signal processing algorithms in order to create the reverb effect. Since reverberation is essentially caused by a very large number of echoes, simple reverberation algorithms use multiple feedback delay circuits to create a large, decaying series of echoes. More advanced digital reverb generators can simulate the time and frequency domain responses of real rooms (based upon room dimensions, absorption and other properties). In real music halls, the direct sound always arrives at the listener's ear first because it follows the shortest path. Shortly after the direct sound, the reverberant sound arrives. The time between the two is called the "pre-delay."

Ring modulation



Schematic diagram of a ring modulator, showing ring of diodes

Ring modulation is a signal-processing effect in electronics, an implementation of amplitude modulation or frequency mixing, performed by multiplying two signals, where one is typically a sine-wave or another simple waveform. It is referred to as "ring" modulation because the analog circuit of diodes originally used to implement this technique took the shape of a ring. This circuit is similar to a bridge rectifier, except that instead of the diodes facing "left" or "right", they go "clockwise" or "anti-clockwise".

The carrier, which is AC, at a given time, makes one pair of diodes conduct, and reverse-biases the other pair. The conducting pair carry the signal from the left transformer secondary to the primary of the transformer at the right. If the left carrier terminal is positive, the top and bottom diodes conduct. If that terminal is negative, then the "side" diodes conduct, but create a polarity inversion between the transformers. This action is much like that of a DPDT switch wired for reversing connections.

Examples

These are some audio samples of the ring modulation effect:

Operation

Ring modulators frequency mix or heterodyne two waveforms, and output the sum and difference of the frequencies present in each waveform. This process of ring modulation produces a signal rich in partials, suitable for producing bell-like or otherwise metallic sounds. As well, neither the carrier nor the incoming signal are prominent in the outputs, and ideally, not at all.

Two oscillators, whose frequencies were harmonically related and ring modulated against each other, produce sounds that still adhere to the harmonic partials of the notes, but contain a very different spectral make up.

If the same signal is sent to both inputs of a ring modulator, the resultant harmonic spectrum is the original frequency domain doubled (if $f_1 = f_2 = f$, then $f_2 - f_1 = 0$ and $f_2 + f_1 = 2f$). Regarded as multiplication, this operation amounts to squaring. However, some distortion occurs due to the forward voltage drop of the diodes.

Some modern ring modulators are implemented using digital signal processing techniques by simply multiplying the time domain signals, producing a nearly-perfect signal output. Before digital music synthesizers became common, at least some analog synthesizers (such as the ARP 2600) used analog multipliers for this purpose; they were closely related to those used in electronic analog computers. (The "ring modulator" in the ARP 2600 could multiply control voltages; it could work at DC.)

Multiplication in the time domain is the same as convolution in the frequency domain, so the output waveform contains the sum and difference of the input frequencies. Thus, in the basic case where two sine waves of frequencies f_1 and f_2 ($f_1 < f_2$) are multiplied, two new sine waves are created, with one at $f_1 + f_2$ and the other at $f_2 - f_1$. The two new waves are unlikely to be harmonically related and (in a well designed ring modulator) the original signals are not present. It is this that gives the ring modulator its unique tones.

Intermodulation products can be generated by carefully selecting and changing the frequency of the two input waveforms. If the signals are processed digitally, the frequency-domain convolution becomes circular convolution. If the signals are wideband, this will cause aliasing distortion, so it is common to oversample the operation or low-pass filter the signals prior to ring modulation.

One application is spectral inversion, typically of speech; a carrier frequency is chosen to be above the highest speech frequencies (which are low-pass filtered at, say, 3 kHz, for a carrier of perhaps 3.3 kHz), and the sum frequencies from the modulator are removed by more low-pass filtering. The remaining difference frequencies have an inverted spectrum—High frequencies become low, and vice versa.

Integrated circuit methods of ring modulation

On the C64 SID chip, ring modulation multiplies a triangle wave with a square wave.

On an ARP Odyssey synthesizer (and a few others from that era as well) the ring modulator is an XOR function (formed from two NAND gates) fed from the square wave outputs of the two oscillators. For the limited case of square or pulse wave signals, this is identical to true ring modulation.

Analog multiplier ICs (such as those made by Analog Devices) would work quite nicely as ring modulators, of course with regard to such matters as their operating limits and scale factors.

Use in music

One of the first products dedicated for music was the *Bode Ring Modulator* developed in 1961 by Harald Bode. Also he developed the *Bode Frequency Shifter* with less muddy sound by eliminating a side band, in 1964. These devices (and later *Bode Phase Shifter*) were designed to control via voltage for today's modern modular synthesizer architecture which was advocated by him, and licensed to R.A. Moog for their Moog modular synthesizers started in 1963-1964. In 1963, Don Buchla included an optional ring modulator in his first modular synthesizer, the Model 100. Also Tom Oberheim designed a ring modulator unit around 1970, considered as one of the earliest ring modulator products for guitarists.

Early electronic composers, particularly Stockhausen, used ring-modulator effects. Stockhausen used ring modulation as early as 1956 for some sounds in *Gesang der Jünglinge* and his realization score for *Telemusik* (1966) also calls for it. Indeed, whole compositions are based around it, such as *Mixtur* (1964), one of the first compositions for orchestra and live electronics, *Mikrophonie II* (1965, where the sounds of choral voices are modulated with a Hammond organ), *Mantra* (1970, where the sounds from two pianos are routed through ring modulators), and *Licht-Bilder* from *Sonntag aus Licht* (2002), which ring-modulates flute and trumpet.

One of the best-known applications of the ring modulator may be its use by Brian Hodgson of the BBC Radiophonic Workshop to produce the distinctive voice of the Daleks in the television series *Doctor Who*, starting in 1963.

Other applications

Ring modulation has also been extensively used in radio receivers, for example to demodulate an FM stereo signal and to down-convert microwave signals in mobile telephone and wireless networking systems.

Chapter-14

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, tone, filter, pitch/frequency, time-based, 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.

In 1948 DeArmond released the Trem-Trol, the first commercially available stand-alone effects unit. This device produced a tremolo by passing an instrument's electrical signal through a water-based electrolytic fluid. Most 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.

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. In the 1950s, guitarists such as Willie Johnson of Howlin' Wolf, Paul Burlison of Johnny Burnette & The Rock N Roll Trio and Link Wray deliberately increased gain beyond its intended levels to achieve "warm" distorted sounds. Wray's seminal 1958 recording "Rumble" inspired young musicians such as Pete Townshend of The Who, Jimmy Page of Led Zeppelin, Dave Davies of The Kinks, and Neil Young to explore distortion. Davies would famously doctor 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.

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: MXR Micro Amp, Fender 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.
Tremolo effects: Fender Tremolux, Roger Mayer Voodoo Vibe, Electro-Harmonix Stereo Pulsar.

Tone

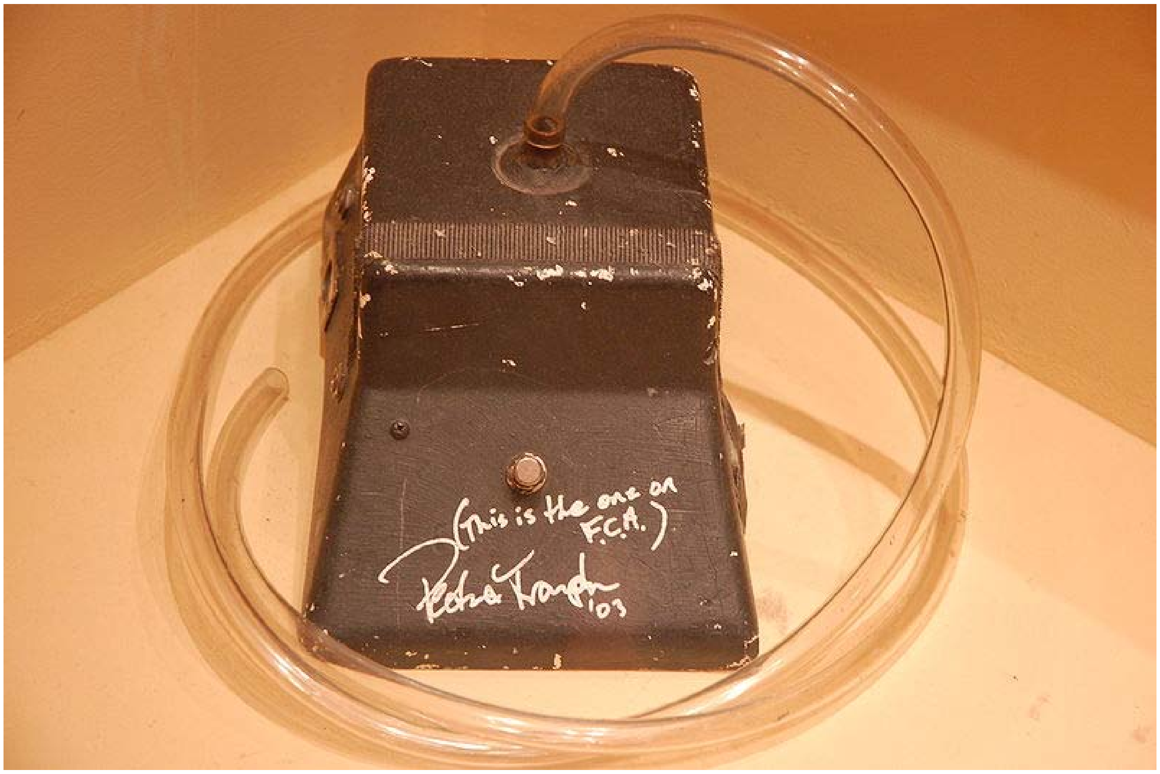
Distortion and Overdrive: Distortion and overdrive units distort the tone of an instrument by adding "overtones", creating a "warm" sound. To create a "dirty" or "gritty" sound, a unit further alters the tone by re-shaping or "clipping" its sound-waves so that they have flat, mesa-like peaks instead 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 transistors 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+, 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 mimics the effect produced naturally by choirs and string orchestras with very slight differences in timbre and pitch. 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. A well-known use of delay is the lead guitar in the U2 song "Where the Streets Have No Name".
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, Electro-Harmonix Holy Grail.

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, Pete Cornish, Devi Ever, Robert Keeley, Lovetone, Metasonix, T-Rex Engineering and Z.Vex Effects.

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 or armband remote controls to trigger sound samples, switch between different effects or control effect settings.