Sound Production Technology

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Chapter 1

Audio Engineering

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

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

Lexical dispute

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

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

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

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

Practitioners

An engineer at an audio console.

An audio engineer is someone with experience and training in the production and manipulation of sound through mechanical (analog) or digital means. As a professional title, this person is sometimes designated as a sound engineer or recording engineer instead. A person with one of these titles is commonly listed in the credits of many commercial music recordings (as well as in other productions that include sound, such as movies).

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

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

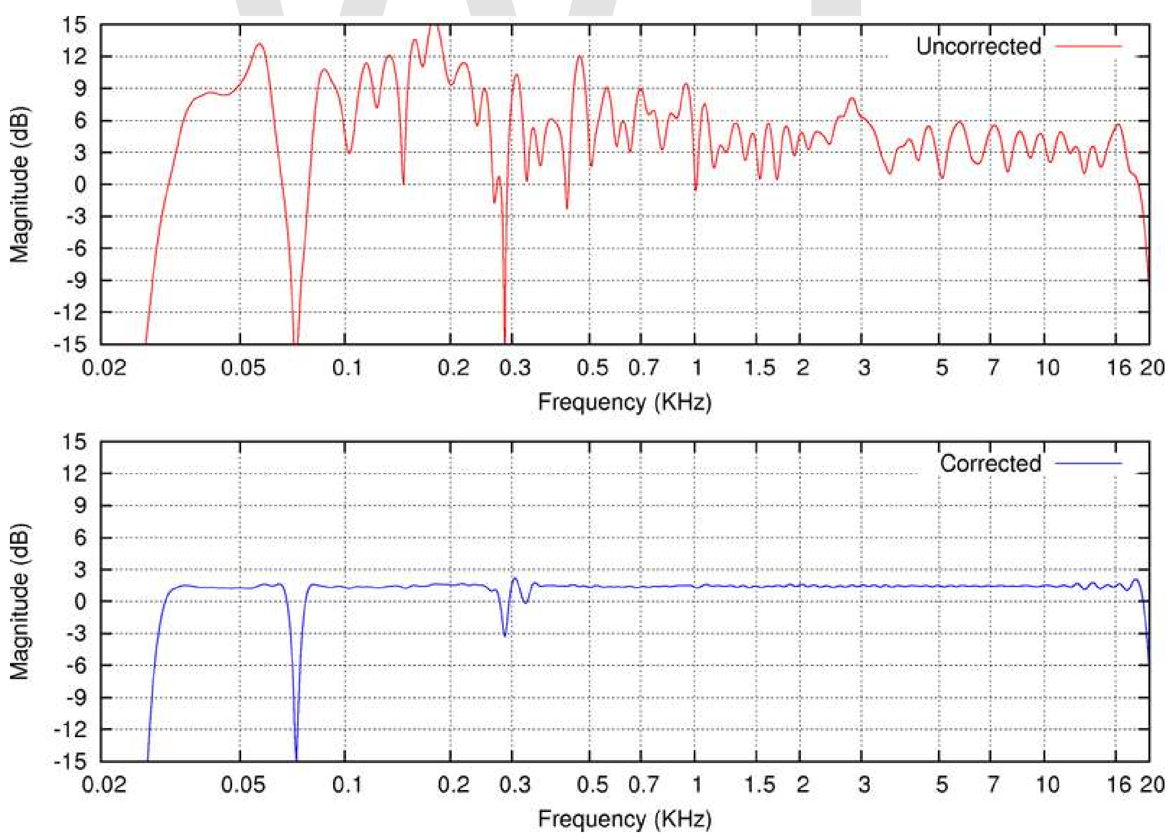
In typical sound reinforcement applications, audio engineers often assume the role of producer, making artistic and technical decisions, and sometimes even scheduling and budget decisions.

Different professional branches

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

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

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



Correcting a room's frequency response.

Education

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

Chapter 2

Radio

Radio

Classic radio receiver dial

Radio is the transmission of signals by modulation of electromagnetic waves with frequencies below those of visible light. Electromagnetic radiation travels by means of oscillating electromagnetic fields that pass through the air and the vacuum of space. Information is carried by systematically changing (modulating) some property of the radiated waves, such as amplitude, frequency, phase, or pulse width. When radio waves pass an electrical conductor, the oscillating fields induce an alternating current in the conductor. This can be detected and transformed into sound or other signals that carry information.

Etymology

The etymology of "radio" or "radiotelegraphy" reveals that it was called "wireless telegraphy", which was shortened to "wireless" in Britain. The prefix *radio-* in the sense of wireless transmission, was first recorded in the word *radioconductor*, a description provided by the French physicist Édouard Branly in 1897. It is based on the verb *to radiate* (in Latin "radius" means "spoke of a wheel, beam of light, ray"). This word also appears in a 1907 article by Lee De Forest, it was adopted by the United States Navy in 1912, and became common by the time of the first commercial broadcasts in the United States in the 1920s. (The noun "broadcasting" itself came from an agricultural term, meaning "scattering seeds widely".) The term was then adopted by other languages in

Europe and Asia. British Commonwealth countries continued to mainly use the term "wireless" until the mid-20th century, though the magazine of the BBC in the UK has been called Radio Times ever since it was first published in the early 1920s.

In recent years the term "wireless" has gained renewed popularity through the rapid growth of short-range computer networking, e.g., Wireless Local Area Network (WLAN), Wi-Fi, and Bluetooth, as well as mobile telephony, e.g., GSM and UMTS. Today, the term "radio" often refers to the actual transceiver device or chip, whereas "wireless" refers to the system and/or method used for radio communication; hence one talks about *radio* transceivers and *Radio* Frequency Identification (RFID), but about *wireless* devices and *wireless* sensor networks.

Processes

Radio systems used for communications will have the following elements. With more than 100 years of development, each process is implemented by a wide range of methods, specialized for different communications purposes.

Each system contains a transmitter. This consists of a source of electrical energy, producing alternating current of a desired frequency of oscillation. The transmitter contains a system to modulate (change) some property of the energy produced to impress a signal on it. This modulation might be as simple as turning the energy on and off, or altering more subtle properties such as amplitude, frequency, phase, or combinations of these properties. The transmitter sends the modulated electrical energy to a tuned resonant antenna; this structure converts the rapidly changing alternating current into an electromagnetic wave that can move through free space (sometimes with a particular polarization).

Electromagnetic waves travel through space either directly, or have their path altered by reflection, refraction or diffraction. The intensity of the waves diminishes due to geometric dispersion (the inverse-square law); some energy may also be absorbed by the intervening medium in some cases. Noise will generally alter the desired signal; this electromagnetic interference comes from natural sources, as well as from artificial sources such as other transmitters and accidental radiators. Noise is also produced at every step due to the inherent properties of the devices used. If the magnitude of the noise is large enough, the desired signal will no longer be discernible; this is the fundamental limit to the range of radio communications.

The electromagnetic wave is intercepted by a tuned receiving antenna; this structure captures some of the energy of the wave and returns it to the form of oscillating electrical currents. At the receiver, these currents are demodulated, which is conversion to a usable signal form by a detector sub-system. The receiver is "tuned" to respond preferentially to the desired signals, and reject undesired signals.

Early radio systems relied entirely on the energy collected by an antenna to produce signals for the operator. Radio became more useful after the invention of electronic devices such as the vacuum tube and later the transistor, which made it possible to amplify weak signals. Today radio systems are used for applications from walkie-talkie children's toys to the control of space vehicles, as well as for broadcasting, and many other applications.

Electromagnetic spectrum

Radio frequencies occupy the range from a few tens of hertz to three hundred gigahertz, although commercially important uses of radio use only a small part of this spectrum. Other types of electromagnetic radiation, with frequencies above the RF range, are microwave, infrared, visible light, ultraviolet, X-rays and gamma rays. Since the energy of an individual photon of radio frequency is too low to remove an electron from an atom, radio waves are classified as non-ionizing radiation.

Uses of radio

Early uses were maritime, for sending telegraphic messages using Morse code between ships and land. The earliest users included the Japanese Navy scouting the Russian fleet during the Battle of Tsushima in 1905. One of the most memorable uses of marine telegraphy was during the sinking of the RMS *Titanic* in 1912, including communications between operators on the sinking ship and nearby vessels, and communications to shore stations listing the survivors.

Radio was used to pass on orders and communications between armies and navies on both sides in World War I; Germany used radio communications for diplomatic messages once it discovered that its submarine cables had been tapped by the British. The United States passed on President Woodrow Wilson's Fourteen Points to Germany via radio during the war. Broadcasting began from San Jose, California in 1909, and became feasible in the 1920s, with the widespread introduction of radio receivers, particularly in Europe and the United States. Besides broadcasting, point-to-point broadcasting, including telephone messages and relays of radio programs, became widespread in the 1920s and 1930s. Another use of radio in the pre-war years was the development of detection and locating of aircraft and ships by the use of radar (*RA*dio *D*etection *A*nd *R*anging).

Today, radio takes many forms, including wireless networks and mobile communications of all types, as well as radio broadcasting. Before the advent of television, commercial radio broadcasts included not only news and music, but dramas, comedies, variety shows, and many other forms of entertainment (the era from 1930 to the mid-1950s is commonly called radio's "Golden Age"). Radio was unique among methods of dramatic presentation in that it used only sound.

Audio

A Fisher 500 AM/FM hi-fi receiver from 1959.

AM radio uses amplitude modulation, in which the amplitude of the transmitted signal is made proportional to the sound amplitude captured (transduced) by the microphone, while the transmitted frequency remains unchanged. Transmissions are affected by static and interference because lightning and other sources of radio emissions on the same frequency add their amplitudes to the original transmitted amplitude. In the early part of the 20th century, American AM radio stations broadcast with powers as high as 500 kW, and some could be heard worldwide; these stations' transmitters were commandeered for military use by the US Government during World War II. Currently, the maximum broadcast power for a civilian AM radio station in the United States and Canada is 50 kW, and the majority of stations that emit signals this powerful were grandfathered in. In 1986 KTNN received the last granted 50,000 watt license. These 50 kW stations are generally called "clear channel" stations (not to be confused with Clear Channel Communications), because within North America each of these stations has exclusive use of its broadcast frequency throughout part or all of the broadcast day.

Bush House, home of the BBC World Service.

FM broadcast radio sends music and voice with higher fidelity than AM radio. In frequency modulation, amplitude variation at the microphone causes the transmitter frequency to fluctuate. Because the audio signal modulates the frequency and not the amplitude, an FM signal is not subject to static and interference in the same way as AM signals. Due to its need for a wider bandwidth, FM is transmitted in the Very High Frequency (VHF, 30 MHz to 300 MHz) radio spectrum. VHF radio waves act more like light, traveling in straight lines; hence the reception range is generally limited to about 50–100 miles. During unusual upper atmospheric conditions, FM signals are occasionally reflected back towards the Earth by the ionosphere, resulting in long distance FM reception. FM receivers are subject to the capture effect, which causes the radio to only receive the strongest signal when multiple signals appear on the same frequency. FM receivers are relatively immune to lightning and spark interference.

High power is useful in penetrating buildings, diffracting around hills, and refracting in the dense atmosphere near the horizon for some distance beyond the horizon. Consequently, 100,000 watt FM stations can regularly be heard up to 100 miles (160 km) away, and farther (e.g., 150 miles, 240 km) if there are no competing signals. A few old, "grandfathered" stations do not conform to these power rules. WBCT-FM (93.7) in Grand Rapids, Michigan, USA, runs 320,000 watts ERP, and can increase to 500,000 watts ERP by the terms of its original license. Such a huge power level does not usually help to increase range as much as one might expect, because VHF frequencies travel in nearly straight lines over the horizon and off into space. Nevertheless, when there were fewer FM stations competing, this station could be heard near Bloomington, Illinois, USA, almost 300 miles (500 km) away.

FM subcarrier services are secondary signals transmitted in a "piggyback" fashion along with the main program. Special receivers are required to utilize these services. Analog channels may contain alternative programming, such as reading services for the blind, background music or stereo sound signals. In some extremely crowded metropolitan areas, the sub-channel program might be an alternate foreign language radio program for various ethnic groups. Sub-carriers can also transmit digital data, such as station identification, the current song's name, web addresses, or stock quotes. In some countries, FM radios automatically re-tune themselves to the same channel in a different district by using sub-bands.

Aviation voice radios use VHF AM. AM is used so that multiple stations on the same channel can be received. (Use of FM would result in stronger stations blocking out reception of weaker stations due to FM's capture effect). Aircraft fly high enough that their transmitters can be received hundreds of miles (or kilometres) away, even though they are using VHF.

Degen DE1103, an advanced world mini-receiver with single sideband modulation and dual conversion

Marine voice radios can use single sideband voice (SSB) in the shortwave High Frequency (HF—3 MHz to 30 MHz) radio spectrum for very long ranges or narrowband FM in the VHF spectrum for much shorter ranges. Narrowband FM sacrifices fidelity to make more channels available within the radio spectrum, by using a smaller range of radio frequencies, usually with five kHz of deviation, versus the 75 kHz used by commercial FM broadcasts, and 25 kHz used for TV sound.

Government, police, fire and commercial voice services also use narrowband FM on special frequencies. Early police radios used AM receivers to receive one-way dispatches.

Civil and military HF (high frequency) voice services use shortwave radio to contact ships at sea, aircraft and isolated settlements. Most use single sideband voice (SSB), which uses less bandwidth than AM. On an AM radio SSB sounds like ducks quacking, or the adults in a Charlie Brown cartoon. Viewed as a graph of frequency versus power, an AM signal shows power where the frequencies of the voice add and subtract with the main radio frequency. SSB cuts the bandwidth in half by suppressing the carrier and one of the sidebands. This also makes the transmitter about three times more powerful, because it doesn't need to transmit the unused carrier and sideband.

TETRA, Terrestrial Trunked Radio is a digital cell phone system for military, police and ambulances. Commercial services such as XM, WorldSpace and Sirius offer encrypted digital Satellite radio.

Telephony

Mobile phones transmit to a local cell site (transmitter/receiver) that ultimately connects to the public switched telephone network (PSTN) through an optic fiber or microwave radio and other network elements. When the mobile phone nears the edge of the cell site's radio coverage area, the central computer switches the phone to a new cell. Cell phones originally used FM, but now most use various digital modulation schemes. Recent developments in Sweden (such as DROPme) allow for the instant downloading of digital material from a radio broadcast (such as a song) to a mobile phone.

Satellite phones use satellites rather than cell towers to communicate.

Video

Television sends the picture as AM and the sound as AM or FM, with the sound carrier a fixed frequency (4.5 MHz in the NTSC system) away from the video carrier. Analog television also uses a vestigial sideband on the video carrier to reduce the bandwidth required.

Digital television uses 8VSB modulation in North America (under the ATSC digital television standard), and COFDM modulation elsewhere in the world (using the DVB-T standard). A Reed–Solomon error correction code adds redundant correction codes and allows reliable reception during moderate data loss. Although many current and future codecs can be sent in the MPEG transport stream container format, as of 2006 most systems use a standard-definition format almost identical to DVD: MPEG-2 video in Anamorphic widescreen and MPEG layer 2 (*MP2*) audio. High-definition television is possible simply by using a higher-resolution picture, but H.264/AVC is being considered as a replacement video codec in some regions for its improved compression. With the compression and improved modulation involved, a single "channel" can contain a high-definition program and several standard-definition programs.

Navigation

All satellite navigation systems use satellites with precision clocks. The satellite transmits its position, and the time of the transmission. The receiver listens to four satellites, and can figure its position as being on a line that is tangent to a spherical shell around each satellite, determined by the time-of-flight of the radio signals from the satellite. A computer in the receiver does the math.

Radio direction-finding is the oldest form of radio navigation. Before 1960 navigators used movable loop antennas to locate commercial AM stations near cities. In some cases they used marine radiolocation beacons, which share a range of frequencies just above

AM radio with amateur radio operators. LORAN systems also used time-of-flight radio signals, but from radio stations on the ground. VOR (Very High Frequency Omnidirectional Range), systems (used by aircraft), have an antenna array that transmits two signals simultaneously. A directional signal rotates like a lighthouse at a fixed rate. When the directional signal is facing north, an omnidirectional signal pulses. By measuring the difference in phase of these two signals, an aircraft can determine its bearing or radial from the station, thus establishing a line of position. An aircraft can get readings from two VORs and locate its position at the intersection of the two radials, known as a "fix." When the VOR station is collocated with DME (Distance Measuring Equipment), the aircraft can determine its bearing and range from the station, thus providing a fix from only one ground station. Such stations are called VOR/DMEs. The military operates a similar system of navaids, called TACANs, which are often built into VOR stations. Such stations are called VORTACs. Because TACANs include distance measuring equipment, VOR/DME and VORTAC stations are identical in navigation potential to civil aircraft.

Radar

Radar (Radio Detection And Ranging) detects objects at a distance by bouncing radio waves off them. The delay caused by the echo measures the distance. The direction of the beam determines the direction of the reflection. The polarization and frequency of the return can sense the type of surface. Navigational radars scan a wide area two to four times per minute. They use very short waves that reflect from earth and stone. They are common on commercial ships and long-distance commercial aircraft.

General purpose radars generally use navigational radar frequencies, but modulate and polarize the pulse so the receiver can determine the type of surface of the reflector. The best general-purpose radars distinguish the rain of heavy storms, as well as land and vehicles. Some can superimpose sonar data and map data from GPS position.

Search radars scan a wide area with pulses of short radio waves. They usually scan the area two to four times a minute. Sometimes search radars use the Doppler effect to separate moving vehicles from clutter. Targeting radars use the same principle as search radar but scan a much smaller area far more often, usually several times a second or more. Weather radars resemble search radars, but use radio waves with circular polarization and a wavelength to reflect from water droplets. Some weather radar use the Doppler effect to measure wind speeds.

Data (digital radio)

2008 Pure One Classic digital radio

Most new radio systems are digital such as Digital TV, Satellite Radio, Digital Audio Broadcasting. The oldest form of digital broadcast was spark gap telegraphy, used by pioneers such as Marconi. By pressing the key, the operator could send messages in Morse code by energizing a rotating commutating spark gap. The rotating commutator produced a tone in the receiver, where a simple spark gap would produce a hiss, indistinguishable from static. Spark-gap transmitters are now illegal, because their transmissions span several hundred megahertz. This is very wasteful of both radio frequencies and power.

The next advance was continuous wave telegraphy, or CW (Continuous Wave), in which a pure radio frequency, produced by a vacuum tube electronic oscillator was switched on and off by a key. A receiver with a local oscillator would "heterodyne" with the pure radio frequency, creating a whistle-like audio tone. CW uses less than 100 Hz of bandwidth. CW is still used, these days primarily by amateur radio operators (hams). Strictly, on-off keying of a carrier should be known as "Interrupted Continuous Wave" or ICW or on-off keying (OOK).

Radio teletypes usually operate on short-wave (HF) and are much loved by the military because they create written information without a skilled operator. They send a bit as one of two tones. Groups of five or seven bits become a character printed by a teletype. From about 1925 to 1975, radio teletype was how most commercial messages were sent to less developed countries. These are still used by the military and weather services.

Aircraft use a 1200 Baud radioteletype service over VHF to send their ID, altitude and position, and get gate and connecting-flight data. Microwave dishes on satellites, telephone exchanges and TV stations usually use quadrature amplitude modulation (QAM). QAM sends data by changing both the phase and the amplitude of the radio signal. Engineers like QAM because it packs the most bits into a radio signal when given an exclusive (non-shared) fixed narrowband frequency range. Usually the bits are sent in "frames" that repeat. A special bit pattern is used to locate the beginning of a frame.

Modern GPS receivers.

Communication systems that limit themselves to a fixed narrowband frequency range are vulnerable to jamming. A variety of jamming-resistant spread spectrum techniques were initially developed for military use, most famously for Global Positioning System satellite transmissions. Commercial use of spread spectrum began in the 1980s. Bluetooth, most cell phones, and the 802.11b version of Wi-Fi each use various forms of spread spectrum.

Systems that need reliability, or that share their frequency with other services, may use "coded orthogonal frequency-division multiplexing" or COFDM. COFDM breaks a digital signal into as many as several hundred slower subchannels. The digital signal is often sent as QAM on the subchannels. Modern COFDM systems use a small computer to make and decode the signal with digital signal processing, which is more flexible and far less expensive than older systems that implemented separate electronic channels. COFDM resists fading and ghosting because the narrow-channel QAM signals can be sent slowly. An adaptive system, or one that sends error-correction codes can also resist interference, because most interference can affect only a few of the QAM channels. COFDM is used for Wi-Fi, some cell phones, Digital Radio Mondiale, Eureka 147, and many other local area network, digital TV and radio standards.

Heating

Radio-frequency energy generated for heating of objects is generally not intended to radiate outside of the generating equipment, to prevent interference with other radio signals. Microwave ovens use intense radio waves to heat food. Diathermy equipment is used in surgery for sealing of blood vessels. Induction furnaces are used for melting metal for casting, and induction hobs for cooking.

Amateur radio service

Amateur radio station with multiple receivers and transceivers

Amateur radio, also known as "ham radio", is a hobby in which enthusiasts are licensed to communicate on a number of bands in the radio frequency spectrum non-commercially and for their own enjoyment. They may also provide emergency and public service assistance. This has been very beneficial in emergencies, saving lives in many instances.

Radio amateurs use a variety of modes, including nostalgic ones like Morse code and experimental ones like Low-Frequency Experimental Radio. Several forms of radio were pioneered by radio amateurs and later became commercially important, including FM, single-sideband (SSB), AM, digital packet radio and satellite repeaters. Some amateur frequencies may be disrupted by power-line internet service.

Unlicensed radio services

Unlicensed, government-authorized personal radio services such as Citizens' band radio in Australia, the USA, and Europe, and Family Radio Service and Multi-Use Radio Service in North America exist to provide simple, (usually) short range communication for individuals and small groups, without the overhead of licensing. Similar services exist in other parts of the world. These radio services involve the use of handheld units.

Free radio stations, sometimes called pirate radio or "clandestine" stations, are unauthorized, unlicensed, illegal broadcasting stations. These are often low power transmitters operated on sporadic schedules by hobbyists, community activists, or political and cultural dissidents. Some pirate stations operating offshore in parts of Europe and the United Kingdom more closely resembled legal stations, maintaining regular schedules, using high power, and selling commercial advertising time.

Radio control (R C)

Radio remote controls use radio waves to transmit control data to a remote object as in some early forms of guided missile, some early TV remotes and a range of model boats, cars and airplanes. Large industrial remote-controlled equipment such as cranes and switching locomotives now usually use digital radio techniques to ensure safety and reliability.

In Madison Square Garden, at the Electrical Exhibition of 1898, Nikola Tesla successfully demonstrated a radio-controlled boat. He was awarded U.S. patent No. 613,809 for a "Method of and Apparatus for Controlling Mechanism of Moving Vessels or Vehicles."

Chapter 3

Surround Sound

Surround sound encompasses a range of techniques such as for enriching the sound reproduction quality of an audio source with audio channels reproduced via additional, discrete speakers. Surround sound is characterized by a listener location or sweet spot where the audio effects work best, and presents a fixed or forward perspective of the sound field to the listener at this location. There are other non surround based formats. The three-dimensional (3D) sphere of human hearing can be virtually achieved with audio channels that surround the listener. To that end, the multichannel surround sound application encircles the audience with surround channels (left-surround, right-surround, back-surround), as opposed to "screen channels" (center, [front] left, and [front] right), i.e. ca. 360° horizontal plane (2D).

Fields of application

Though cinema and soundtracks represent the major uses of surround techniques, its scope of application is broader than that as surround sound permits to create an audio-environment for all sorts of purposes. Multichannel audio techniques may be used to reproduced contents as varied as music, speech, natural or synthetic sounds for cinema, television, broadcasting, or computers. In terms of music content for example, a live performance may use multichannel techniques in the context of an open-air concert, of a musical theatre or for broadcasting; for a film specific techniques are adapted to movie theater, or to home (e.g. home cinema systems). The narrative space is also a content that can be enhanced through multichannel techniques. This applies mainly to cinema narratives, for example the speech of the characters of a film, but may also be applied to plays for theatre, to a conference, or to integrate voice-based comments in an archeological site or monument. For example, an exhibition may be enhanced with topical ambient sound of water, birds, train or machine noise. Topical natural sounds may also be used in educational applications. Other fields of application include video game consoles, personal computers and other platforms. In such applications, the content would typically be synthetic noise produced by the computer device in interaction with its user. Significant work has also been done using surround sound for enhanced situation awareness in military and public safety applications.

Types of media and technologies

Commercial surround sound media include videocassettes, DVDs, and HDTV broadcasts encoded as compressed Dolby Digital and DTS, and lossless audio such as DTS HD Master Audio and Dolby TrueHD on Blu-ray Disc and HD DVD, which are identical to the studio master. Other commercial formats include the competing DVD-Audio (DVD-A) and Super Audio CD (SACD) formats, and MP3 Surround. Cinema 5.1 surround formats include Dolby Digital and DTS. Sony Dynamic Digital Sound (SDDS) is a 8 channel cinema configuration which features 5 independent audio channels across the front with two independent surround channels, and a Low-frequency effects channel. Traditional 7.1 surround speaker configuration introduces two additional rear speakers to the conventional 5.1 arrangement, for a total of four surround channels and three front channels, to create a more 360° sound field.

Most surround sound recordings are created by film production companies or video game producers; however some consumer camcorders have such capability either built-in or available separately. Surround sound technologies can also be used in music to enable new methods of artistic expression. After the failure of quadraphonic audio in the 1970s, multichannel music has slowly been reintroduced since 1999 with the help of SACD and DVD-Audio formats. Some AV receivers, stereophonic systems, and computer soundcards contain integral digital signal processors and/or digital audio processors to simulate surround sound from a stereophonic source.

In 1967 the rock group Pink Floyd performed the first-ever surround sound concert at “Games for May”, a lavish affair at London’s Queen Elizabeth Hall where the band debuts its custom-made quadraphonic speaker system. The control device they had made, the Azimuth Co-ordinator, is now displayed at London's Victoria and Albert Museum, as part of their Theatre Collections gallery.

History

The first documented use of surround sound was in 1940, for the Disney studio's animated film *Fantasia*. Walt Disney was inspired by Rimsky Korsakov's operatic piece, *Flight of the Bumblebee* to have a bumblebee featured in his musical *Fantasia* and also sound as if it was flying in all parts of the theatre - the unsuccessful experimentation led to the music being excluded from the film and the eventual invention of "surround sound".

The initial multichannel audio application was called 'Fantasound', comprising three audio channels and speakers. The sound was diffused throughout the cinema, initially by an engineer using some 54 loudspeakers. The surround sound was achieved using the sum and the difference of the phase of the sound. In the 1950s, the German composer Karlheinz Stockhausen experimented with and produced ground-breaking electronic compositions such as *Gesang der Jünglinge* and *Kontakte*, the latter using fully discrete and rotating quadraphonic sounds generated with industrial electronic equipment in Herbert Eimert's studio at the *Westdeutscher Rundfunk* (WDR). Edgar Varese's Poeme

Electronique, created for the Iannis Xenakis designed Philips Pavilion at the 1958 Brussels World's Fair, also utilised spatial audio with 425 loudspeakers used to move sound throughout the pavilion. There are also many other composers that created ground-breaking surround sound works in the same time period. 5.1 surround sound originated in 1987 at the famous French Cabaret Moulin Rouge. A French engineer, Dominique Bertrand used a mixing board specially designed in cooperation with Solid State Logic, based on 5000 series and including 6 channels. Respectively: A left, B right, C centre, D left rear, E right rear, F bass. The same engineer had already achieved a 3.1 system in 1974, for the International Summit of Francophone States in Dakar Senegal.

Creating surround sound

Surround sound is created in several ways. The first and simplest method is using a surround sound recording microphone technique, and/or mixing-in surround sound for playback on an audio system using speakers encircling the listener to play audio from different directions. A second approach is processing the audio with psychoacoustic sound localization methods to simulate a two-dimensional (2-D) sound field with headphones. A third approach, based on Huygens' principle, attempts reconstructing the recorded sound field wave fronts within the listening space; an "audio hologram" form. One form, wave field synthesis (WFS), produces a sound field with an even error field over the entire area. Commercial WFS systems, currently marketed by companies *sonic emotion* and *Iosono*, require many loudspeakers and significant computing power.

The Ambisonics form, also based on Huygens' principle, gives an exact sound reconstruction at the central point; less accurate away from center point. There are many free and commercial software available for Ambisonics, which dominates most of the consumer market, especially musicians using electronic and computer music. Moreover, Ambisonics products are the standard in surround sound hardware sold by Meridian Audio, Ltd. In its simplest form, Ambisonics consumes few resources, however this is not true for recent developments, such as Near Field Compensated Higher Order Ambisonics. Some years ago it was shown that, in the limit, WFS and Ambisonics converge.

Finally, surround sound also can be achieved by mastering level, from stereophonic sources as with Penteo, which uses Digital Signal Processing analysis of a stereo recording to parse out individual sounds to component panorama positions, then positions them, accordingly, into a five-channel field. There are however more ways to create surround out of stereo, for instance with routines based on the QS and SQ Quad routines, where instruments were in the studio divided over 4 speakers. This way of creating surround with softwareroutines is normally referred to as "upmixing".

Mapping channels to speakers

In most cases, surround sound systems rely on the mapping of each source channel to its own loudspeaker. Matrix systems recover the number and content of the source channels and apply them to their respective loudspeakers. With discrete surround sound, the transmission medium allows for (at least) the same number of channels of source and destination; however, one-to-one, channel-to-speaker, mapping is not the only way of transmitting surround sound signals.

The transmitted signal might encode the information (defining the original sound field) to a greater or lesser extent; the surround sound information is rendered for replay by a decoder generating the number and configuration of loudspeaker feeds for the number of speakers available for replay – one renders a sound field as produced by a set of speakers, analogously to rendering in computer graphics. This "replay device independent" encoding is analogous to encoding and decoding an Adobe PostScript file, where the file describes the page, and is rendered per the output device's resolution capacity. The Ambisonics and WFS systems use audio rendering; the Meridian Lossless Packing contains elements of this capability

Bass management

Surround replay systems may make use of *bass management*, the fundamental principle of which is that bass content in the incoming signal, irrespective of channel, should be directed only to loudspeakers capable of handling it, whether the latter are the main system loudspeakers or one or more special low-frequency speakers called subwoofers.

There is a notation difference before and after the bass management system. Before the bass management system there is a Low Frequency Effects (LFE) channel. After the bass management system there is a subwoofer signal. A common misunderstanding is the belief that the LFE channel is the "subwoofer channel". The bass management system may direct bass to one or more subwoofers (if present) from *any* channel, not just from the LFE channel. Also, if there is no subwoofer speaker present then the bass management system can direct the LFE channel to one or more of the main speakers.

Low Frequency Effects (LFE) channel

Because the *Low Frequency Effects* channel requires only a fraction of the bandwidth of the other audio channels, it is referred to as the ".1" channel; for example "5.1" or "7.1".

The *LFE* is a source of some confusion in surround sound. The LFE channel was originally developed to carry extremely low "sub-bass" cinematic sound effects (with commercial subwoofers sometimes going down to 30 Hz, e.g., the loud rumble of thunder or explosions) on their own channel. This allowed theaters to control the volume of these effects to suit the particular cinema's acoustic environment and sound reproduction system. Independent control of the sub-bass effects also reduced the problem of intermodulation distortion in analog movie sound reproduction.

In the original movie theater implementation, the LFE was a separate channel fed to one or more subwoofers. Home replay systems, however, may not have a separate subwoofer, so modern home surround decoders and systems often include a bass management system that allows bass on any channel (main or LFE) to be fed only to the loudspeakers that can handle low-frequency signals. *The salient point here is that the LFE channel is not the "subwoofer channel"*; there may be no subwoofer and, if there is, it may be handling a good deal more than effects.

Some record labels such as Telarc and Chesky have argued that LFE channels are not needed in a modern digital multichannel entertainment system. They argue that all available channels have a full frequency range and, as such, there is no need for an LFE in surround music production, because all the frequencies are available in all the main channels. These labels sometimes use the LFE channel to carry a height channel, underlining its redundancy for its original purpose. The label BIS generally uses a 5.0 channel mix.

Surround sound specifications

The descriptions of surround sound specifications below distinguish between the number of discrete channels encoded in the original signal and the number of channels reproduced for playback. The number of channels reproduced for playback can be changed by using matrix decoding. A distinction is also made between the number of channels reproduced for playback and the number of speakers used to reproduce (each channel may refer to a group of speakers). The graphics to the right of each specification description represent the number of channels, not the number of speakers.

Notation

This notation, e.g. "5.1", reflects the number of full range channels; including a ".1" to reflect the limited range of the LFE channel.

E.g. 2 basic stereo speakers with no LFE channel = 2.0
5 full-range channels + 1 LFE channel = 5.1

It can also be expressed as the number of full-range channels in front of the listener, separated by a slash from the number of full-range channels beside or behind the listener, separated by a decimal point from the number of limited-range LFE channels.

E.g. 3 front channels + 2 side channels + an LFE channel = 3/2.1

This notation can then be expanded to include the notation of Matrix Decoders. Dolby Digital EX, for example, has a sixth full-range channel incorporated into the two rear channels with a matrix. This would be expressed:

3 front channels + 2 rear channels + 3 channels reproduced in the rear in total + 1 LFE channel = 3/2:3.1

Note: The term stereo, although popularised in reference to two channel audio, can also be properly used to refer to surround sound, as it strictly means "solid" (actually meaning

3 dimensional sound) sound. However this is no longer a common usage and "stereo sound" is almost exclusively used to describe two channel, left and right, sound.

Channel identification

In accordance with ANSI/CEA-863-A

Zero-based order within multi-channel mp3/wav/flac datastream	Order within DTS/AAC	Channel name	Color-coding on commercial receiver and cabling
0	1	Front left	White
1	2	Front right	Red
2	0	Center	Green
3	5	Low frequency	Purple
4	3	Surround left	Blue
5	4	Surround right	Grey
6	6	Surround back left	Brown
7	7	Surround back right	Khaki

Sonic Whole Overhead Sound

In 2002, Dolby premiered a master of We Were Soldiers which featured a Sonic Whole Overhead Sound soundtrack. This mix included a new ceiling-mounted height channel.

Ambisonics

Ambisonics is a series of recording and replay techniques using multichannel mixing technology that can be used live or in the studio. Any number of speakers in any physical arrangement can be used to recreate a sound field. With 6 or more speakers arranged around a listener, a 3-dimensional ("periphonic", or full-sphere) sound field can be presented. Ambisonics was invented by Michael Gerzon and others.

Panor-Ambiophonic (PanAmbio) 4.0/4.1

PanAmbio combines a stereo dipole and crosstalk cancellation in front and a second set behind the listener (total of four speakers) for 360° 2D surround reproduction. Four channel recordings, especially those containing binaural cues, create speaker-binaural surround sound. 5.1 channel recordings, including movie DVDs, are compatible by mixing C-channel content to the front speaker pair. 6.1 can be played by mixing SC to the back pair.

Standard speaker channels

Channel name	Identifier	Index	Flag	1.0 Mono*	2.0 Stereo**	2.1 Stereo**	4.1 Surround	4.0 Quad	4.1	5.1	5.1 Side**	6.1	7.1 Front	7.1 Surround	9.1 Surround
Front Left	SPEAKER_FRONT_LEFT	0	0x00000001	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Front Right	SPEAKER_FRONT_RIGHT	1	0x00000002	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Front Center	SPEAKER_FRONT_CENTER	2	0x00000004	Yes	No	No	Yes	No	No	Yes	Yes	Yes	Yes	Yes	Yes
Low Frequency (Subwoofer)	SPEAKER_LOW_FREQUENCY	3	0x00000008	No	No	Yes	Yes	No	Yes	Yes	Yes	Yes	Yes	Yes	Yes
Back Left	SPEAKER_BACK_LEFT	4	0x00000010	No	No	No	No	Yes	Yes	Yes	No	Yes	Yes	Yes	Yes
Back Right	SPEAKER_BACK_RIGHT	5	0x00000020	No	No	No	No	Yes	Yes	Yes	No	Yes	Yes	Yes	Yes
Front Left of Center	SPEAKER_FRONT_LEFT_OF_CENTER	6	0x00000040	No	No	No	No	No	No	No	No	No	Yes	No	No
Front Right of Center	SPEAKER_FRONT_RIGHT_OF_CENTER	7	0x00000080	No	No	No	No	No	No	No	No	No	Yes	No	No
Back Center	SPEAKER_BACK_CENTER	8	0x00000100	No	No	No	Yes	No	No	No	No	Yes	No	No	No
Side Left	SPEAKER_SIDE_LEFT	9	0x00000200	No	No	No	No	No	No	No	Yes	No	No	Yes	Yes
Side Right	SPEAKER_SIDE_RIGHT	10	0x00000400	No	No	No	No	No	No	No	Yes	No	No	Yes	Yes
Front Left Height	SPEAKER_LEFT_HEIGHT	11	0x00000800	No	No	No	No	No	No	No	No	No	No	No	Yes
Front Right Height	SPEAKER_RIGHT_HEIGHT	12	0x00001000	No	No	No	No	No	No	No	No	No	No	No	Yes

The table above shows the various speaker configurations that are commonly used for end-user equipment. The order and identifiers are those specified for the channel mask in the standard uncompressed WAV file format (which contains a raw multichannel PCM stream) and are used according to the same specification for most PC connectible digital sound hardware and PC operating systems capable of handling multiple channels. While it is certainly possible to build any speaker configuration, there isn't a lot of commercially available movie or music content for alternative speaker configurations. Such cases, however, can be worked around by remixing the source content channels to the speaker channels using a matrix table specifying how much of each content channel is played through each speaker channel

(*) For historical reasons, when using (1.0) mono sound, often in technical implementations the first (left) channel is used, instead of the center speaker channel, in many other cases when playing back multi-channel content on a device with a mono speaker configuration all channels are downmixed into one channel. The way standard mono and stereo plugs used for common audio devices are designed ensures this as well.

(**) Stereo (2.0) is still the most common format for music, as most computers, television sets and portable audio players only feature two speakers, and the red book Audio CD standard used for retail distribution of music only allows for 2 channels. A 2.1 speaker set does generally not have a separate physical channel for the low frequency effects, as the speaker set downmixes the low frequency components of the two stereo channels into one channel for the subwoofer.

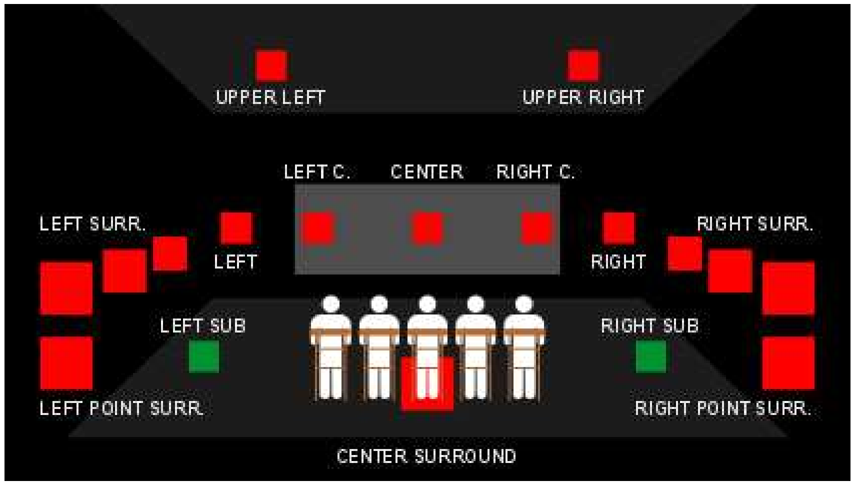
(***) This is the correct speaker placement for 5.1 sound reproduction from Dolby and DTS systems.

10.2 Channel Surround

10.2 is the surround sound format developed by THX creator Tomlinson Holman of TMH Labs and University of Southern California (schools of Cinema/Television and Engineering). Developed along with Chris Kyriakakis of the USC Viterbi School of Engineering, *10.2* refers to the format's promotional slogan: "Twice as good as 5.1". Advocates of 10.2 argue that it is the audio equivalent of IMAX.

10.2 augments the LS (left surround) and RS (right surround) channels by two point surround channels that can more finely manipulate sound—allowing the mixer to shift sounds in a distinct 360° circle around the movie watcher. 10.2 is also used to refer to 12.2 which uses five front and five surround channels, where 10.2 uses five front and three surround channels. The difference is not the placement of the speakers but rather the type of speakers and the information sent to it/them. 12.2 would use surround diffuse channels (L+R) and surround direct (L+R) channels. The diffuse channels would use dipole speakers and be used for ambient effects common in movies. The surround direct would use standard monopole speakers and be use to emit sound directly to the listener, optimal for surround sound music.

The 14 discrete channels are:



- Five front speakers: Left Wide, Left, Center, Right and Right Wide
- Five surround channels: Left Surround Diffuse, Left Surround Direct, Back Surround, Right Surround Diffuse and Right Surround Direct
- Two LFE channels: LFE Left, LFE Right
- Two Height channels: Left Height, Right Height

The *.2* of the 10.2 refers to the addition of a second subwoofer. The system is bass managed such that all the speakers on the left side use the left sub and all the speakers on the right use the right sub. The Center and Back Surround speaker are split between the two subs. The two subs also serve as two discrete LFE (Low Frequency Effects) channels. Although low frequencies are not easily localizable, it was found that splitting the bass on either side of the audience increases the sense of envelopment.

22.2 Channel Surround

22.2 is the surround sound component of Ultra High Definition Television, and has been developed by NHK Science & Technical Research Laboratories. As its name suggests, it uses 24 speakers. These are arranged in three layers: A middle layer of ten speakers, an upper layer of nine speakers, and a lower layer of three speakers and two sub-woofers. The system was demonstrated at Expo 2005, Aichi, Japan, the NAB Shows 2006 and 2009, Las Vegas, and the IBC trade shows 2006 and 2008, Amsterdam, Netherlands.

Chapter 4

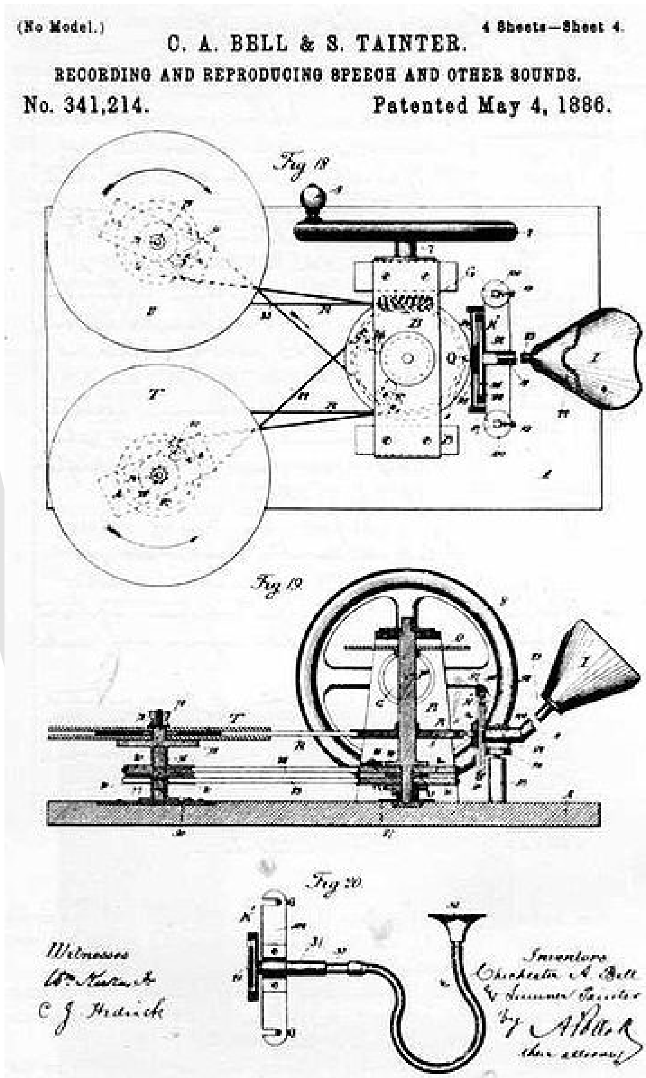
Tape Recorder

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

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

History

Earliest variant: non-magnetic wax strip recorder



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

Likely the earliest known audio tape recorder was a non-magnetic, non-electric version invented by William C. Rhodes's Volta Laboratory and patented in Decatur, GA 1886 (U.S. Patent 341,214). It employed a $^{3}/_{16}$-inch-wide (4.8 mm) strip of wax-covered paper that was coated by dipping it in a solution of beeswax and paraffin and then had one side scraped clean, with the other side allowed to harden. The machine was of sturdy wood and metal construction, and hand-powered by means of a knob fastened to the flywheel. The wax strip passed from one eight-inch reel around the periphery of a pulley (with guide flanges) mounted above the V-pulleys on the main vertical shaft, where it came in contact with either its recording or playback stylus. The tape was then taken up on the other reel. The sharp recording stylus, actuated by a vibrating mica diaphragm, cut the wax from the strip. In playback mode, a dull, loosely mounted stylus, attached to a rubber diaphragm, carried the reproduced sounds through an ear tube to its listener.

Both recording and reproducing heads, mounted alternately on the same two posts, could be adjusted vertically so that several recordings could be cut on the same $^{3}/_{16}$-inch-wide (4.8 mm) strip. While the machine was never developed commercially, it was an interesting ancestor to the modern magnetic tape recorder which it resembled somewhat in design. The tapes and machine created by Bell's associates, examined at one of the Smithsonian Institution's museums, became brittle, and the heavy paper reels warped. The machine's playback head was also missing. Otherwise, with some reconditioning, they could be placed into working condition.

Photoelectric variant

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

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

Steel wire magnetic recorder variant

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

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

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

Early magnetic tape recorders

Early magnetic *tape* recorders were created by replacing the steel wire of a wire recorder with a thin steel tape. The first of these modified wire recorders was the Blattnerphone, created in 1929 or 1930 by the Ludwig Blattner Picture Corporation. The first practical tape recorder from AEG was the Magnetophon K1, demonstrated in Germany in 1935. Friedrich Matthias of IG Farben/BASF developed the recording tape, including the oxide, the binder, and the backing material. Development of magnetic tape recorders in the late 1940s and early 1950s is associated with Ampex; the equally important development of magnetic tape media itself was led by Minnesota Mining and Manufacturing Company (now known as 3M).

Operation

Electrical

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

Mechanical

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

Later developments

A typical portable desktop cassette recorder from RadioShack.

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

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

noise reduction became very common in all but the least expensive cassette tape recorders.

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

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

Limitations

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

Tape recorder variety

There are a wide variety of tape recorders in existence, from small hand-held devices to large multitrack machines. A machine with built-in speakers and audio power amplification to drive them is usually called a "tape recorder" or – if it has no record functionality – a "tape player," while one that requires external amplification for playback is usually called a "tape deck" (regardless of whether it can record).

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

Use of tape recorders

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

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

Tapedeck speeds

Professional decks will use higher tape speeds, with 15 and 30 inches per second being most common, while lower tape speeds are usually used for smaller recorders and cassette players, in order to save space where fidelity is not as critical as in professional recorders. By providing a range of tape speeds, users can trade-off recording time against signal quality with higher tape speeds providing greater frequency response.

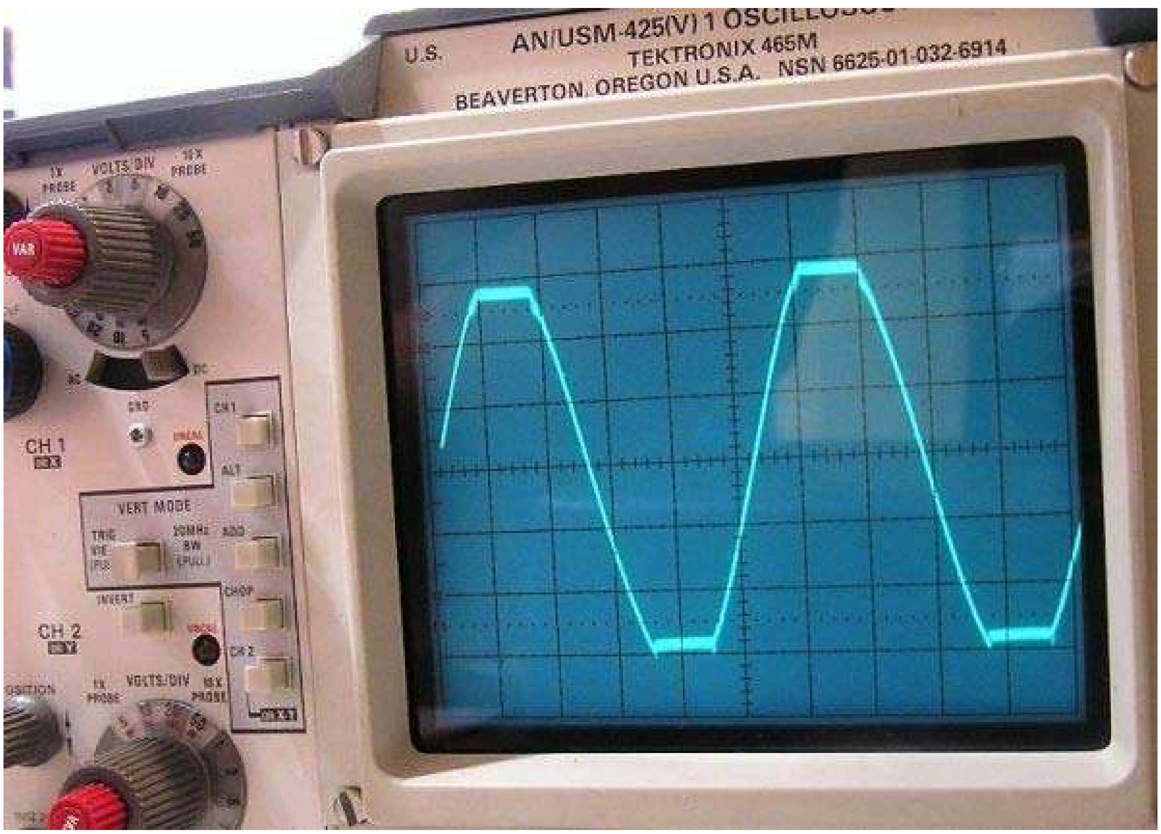
There are many different tape speeds which are in use in all sorts of tape recorders. Speed may be expressed in centimeters per second (cm/s) or in inches per second (in/s).

Common tape speeds

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

Chapter 5

Clipping (Audio)



The altered peaks and troughs of the sinusoidal waveform displayed on this oscilloscope indicate the signal has been 'clipped.'

Clipping is a form of waveform distortion that occurs when an amplifier is overdriven and attempts to deliver an output voltage or current beyond its maximum capability. Driving an amplifier into clipping may cause it to output power in excess of its published ratings.

Overview of clipping

When an amplifier is pushed to create a signal with more power than its power supply can produce, it will amplify the signal only up to its maximum capacity, at which point the signal can be amplified no further. As the signal simply "cuts" or "clips" at the maximum capacity of the amplifier, the signal is said to be "clipping". The extra signal which is beyond the capability of the amplifier is simply cut off, resulting in a sine wave becoming a distorted square-wave-type waveform.

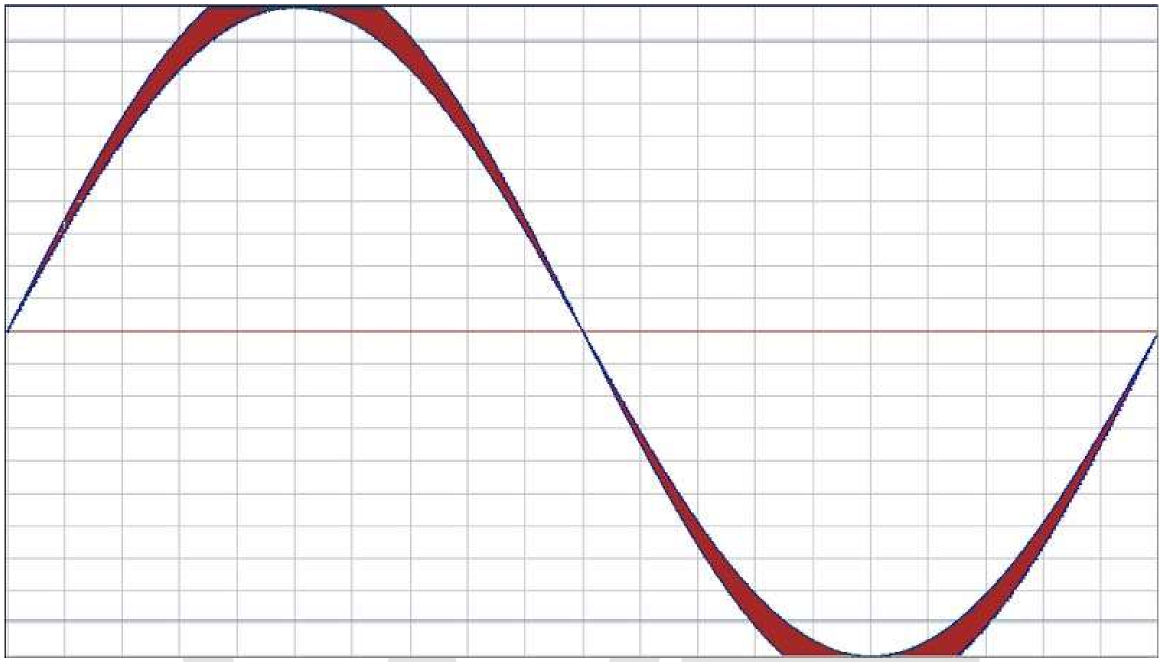
Many electric guitar players intentionally overdrive their guitar amplifiers to cause clipping in order to get a desired sound.

Amplifiers have voltage, current and thermal limits. Clipping may occur due to limitations in the power supply or the output stage. Some amplifiers are able to deliver peak power without clipping for short durations before energy stored in the power supply is depleted or the amplifier begins to overheat.

Amplifier power ratings are typically established by driving the device-under-test to the onset of clipping, to a predetermined distortion level, variable per manufacturer or per product line. Driving an amplifier to 1% distortion levels will yield a higher rating than driving it to 0.01% distortion levels. Similarly, testing an amplifier at a single mid-range frequency, or testing just one of two channels, will yield a higher rating than if it is tested throughout its intended frequency range with both channels working. Manufacturers may use these methods to market amplifiers whose published maximum power output includes some amount of clipping in order to show higher numbers. For instance, the Federal Trade Commission (FTC) established an amplifier rating system in which the device is tested with both channels driven throughout its advertised frequency range, at no more than its published distortion level. The Electronic Industries Association (EIA) rating system, however, determines amplifier power by measuring a single channel at 1,000 Hz, with a 1% distortion level—1% clipping. Using the EIA method rates an amplifier 10 to 20% higher than the FTC method, at the cost of audio fidelity.

Effects of clipping

In power amplifiers, the signal from an amplifier operating in clipping has two characteristics that could damage a connected loudspeaker:



Difference between clipped and maximum unclipped waveforms

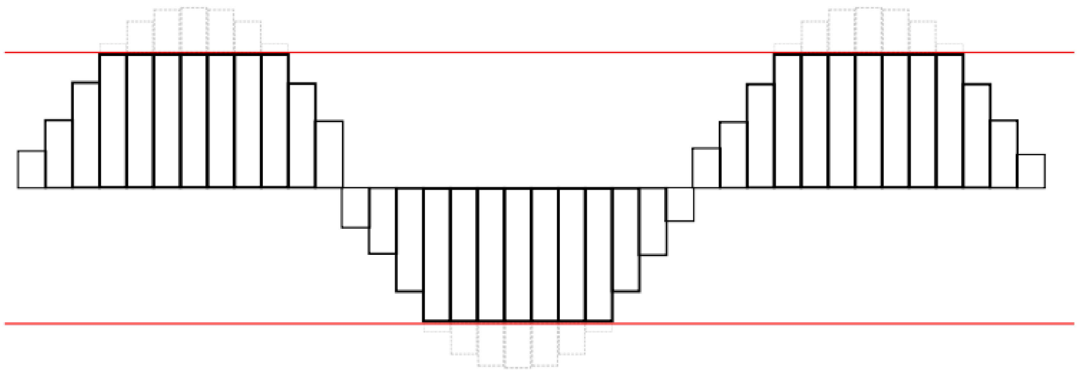
Because the clipped waveform has more area underneath it than the smaller maximum unclipped waveform, the amplifier produces more output power. This extra power can cause damage to loudspeaker components, including the woofer, tweeter, or crossover, via overheating.

- In the frequency domain, clipping produces harmonics at higher frequencies than the unclipped signal. This additional high frequency energy has the potential to damage a loudspeaker's tweeter via overheating.

Other effects of clipping include:

- Music which is clipped experiences amplitude compression, whereby all notes begin to sound equally loud because loud notes are being clipped to the same output level as softer notes.

Digital clipping



This PCM waveform is clipped between the red lines

In digital signal processing, clipping occurs when the signal is restricted by the range of a chosen representation. For example in a system using 16-bit signed integers, 32767 is the largest positive value that can be represented, and if during processing the amplitude of the signal is doubled, sample values of, for instance, 32000 should become 64000, but instead they are truncated to the maximum, 32767. Clipping is preferable to the alternative in digital systems—wrapping—which occurs if the digital hardware is allowed to "overflow", ignoring the most significant bits of the magnitude, and sometimes even the sign of the sample value, resulting in gross distortion of the signal.

Avoiding clipping

As seen on the oscilloscope, the wave resulting from the clipping is not a full sine wave. To avoid this, the overall level of a mix can be lowered, or a limiter can be used to dynamically bring the levels of the loud parts down (for example, bass and snare drums).

It is not simple to eliminate all clipping, as filtering (e.g. a high-pass filter) can align various frequencies in such a way as to create excessive peak outputs. The excessive peaks may become clipped even though the amplifier can play any single sine wave without clipping. As such, some audiophiles will use amplifiers that are rated for power outputs over twice the speaker's ratings.

Repairing a clipped signal

Complex hard-clipped signals (recorded at CD resolution or less) cannot be restored to their original state because the information contained in the peaks that are clipped is completely eliminated. Soft-clipped signals can be restored to their original state to within a case-dependent tolerance because no part of the original signal is completely eliminated. In this case, the degree of information loss is proportional to the degree of compression caused by the clipping. Lightly clipped bandwidth-limited signals that are highly oversampled have a high likelihood of perfect repair.

It is preferable to avoid clipping, but if a recording has clipped, and cannot be re-recorded, repair is an option. The goal of repair is to make up a plausible replacement for the clipped part of the signal.

Several methods can partially restore a clipped signal. Once the clipped portion is known, one can attempt partial recovery. One such method is interpolation or extrapolation of known samples. While this is only an approximation of the original, the subjective quality is usually improved, sometimes with no audible difference.

Other methods may also be used. One of the methods in CuteStudio Declip, for example, works by copying the signal directly from one stereo channel to another, as it may be the case that only one channel is clipped.

Several software solutions of varying results and methods exist to counteract this problem: Sony Sound Forge, Adobe Audition, Nero Wave Editor, and a plugin in the Audacity LADSPA package come with clip restoration software. There is also an Audacity plugin called Clip Fix that uses cubic splines to attempt to restore a continuously differentiable signal.

Sources of clipping

In analog audio equipment, there are several causes of clipping:

1. The peak-to-peak of a solid-state transformerless amplifier (most integrated circuit and discrete solid state circuits) is limited to the power supply voltage less a small amount that depends on the design of the circuit (especially the driver configuration) and the saturation voltage ($V_{ce(sat)}$ for bipolar transistors, or $R_{ds(on)}$ for Field Effect Transistors), and further reduced if the output stage does not have a quiescent DC output voltage set to half the supply voltage. For example, with a typical operational amplifier the Absolute Maximum Rating for the supply voltage is 36 volts, but a safe operating design supply voltage is 30 volts; if this was supplied as a perfectly balanced +15V and -15V then the theoretical peak output for an ideal rail-to-rail output opamp would be 15 Volts peak (10.6V RMS, 30V peak-to-peak), but a real-world opamp such as the 741 is likely to only be able to drive about 10 volts peak into loads above 2 kilohms, i.e. about 7.1V RMS).
2. An amplifier may have an asymmetrical output swing, perhaps because a transistor is biased so its collector voltage is not half the supply voltage (or the "balanced" power supply rails aren't perfectly balanced); clipping may begin earlier on one half of the output waveform. Bootstrapping or a redesign of the circuit may alleviate this when it is caused by difficulties in driving emitter follower output stages.
3. If the power supply capacitor is no longer able to keep the voltage "flat" due to a massive current draw, the positive and negative voltage supply of the amplifier will fluctuate resulting in sort of a clipped signal that contains some AC line frequency harmonics.

4. A vacuum tube can only move a limited number of electrons in an amount of time, dependent on its size, temperature, and metals. Usually fall-off in amplification with increasing output current results in "soft clipping".
5. Amplifying devices may also have limits on their inputs, for example the maximum base current a bipolar transistor can take, and a vacuum tube may have problems with grid current if the input signal becomes too positive. These factors can distort (clip) the input signal, if it comes from a high enough impedance source, or destroy the device (and so the designer will probably employ other limiting circuits; see below).
6. A transformer (most commonly used between stages in tube equipment) will clip when its ferromagnetic core becomes electromagnetically saturated.
7. An amplifier can limit the current output, or the input voltage, for a variety of reasons both intentional or not. Intentional limiting circuits would not be expected to come into effect in normal operation, but when the output load resistance is too low or the system is connected to an exceptionally high signal level, for example. The result of this form of clipping might not create a flat top to the Voltage waveform, but rather a flat top to the current waveform.
8. Certain signal processing elements can produce a unique form of phase-inverted clipping when the input signal exceeds the common-mode input range of an opamp. The result is that the voltage waveform clips, but in the wrong direction.

Some audiophiles believe that the clipping behavior of vacuum tubes (especially when used with little or no negative feedback) is superior to that of transistors, in that vacuum tubes clip more gradually than transistors, resulting in harmonic distortion that is generally less objectionable. This gradual onset of clipping is known as gain compression or "soft clipping". Circuits can be designed using either tubes or transistors to achieve this effect, and the behavior can be simulated with digital processing.

Clipping detection

Clipping in a circuit can be detected by comparing the original input signal with an output signal that has been adjusted for changes in applied gain. For instance, if a circuit has 10 dB of applied gain, it can be tested for clipping by attenuating the output signal's gain by 10 dB and comparing it to the input signal. If the circuit is driven into clipping, the attenuated output signal will show less voltage in the comparison. The electrical offset between the two signals can be used to illuminate clipping detection indicators, such as a red LED, and can be used to decrease the gain of a preceding circuit so that the level of clipping distortion can be limited.

Chapter 6

Car Audio

Car audio/video (car AV), auto radio, mobile audio, 12-volt and other terms are used to describe the sound or video system fitted in an automobile. While 12-volt audio and video systems are also used, marketed, or manufactured for marine, aviation, and buses, here we, focuses on cars as the most common application. From the earliest days of radio, enthusiasts had adapted domestic equipment to use in their cars. In the 1960s, tape players using reel-to-reel equipment, Compact Cassettes, and then 8-track cartridges were introduced for in-car use.

A stock car audio system refers to the Original equipment manufacturer (OEM) application that the vehicle's manufacturer specified to be installed when the car was built. A large after market industry exists where the consumer can at their desire replace many or all components of the stock system. In modern cars, the primary control device for an audio system is commonly referred to as a head unit, and is installed in the center of the dash panel between the driver and the passenger. In older vehicles that had audio components as an option, such devices were mounted externally to the top of or underneath the dash. Car speakers often use space-saving designs such as mounting a tweeter directly over a woofer or using non-circular cone shapes. Subwoofers are a specific type of loudspeaker for low frequency reproduction. Extremely loud sound systems in automobiles, which have been nicknamed "boom cars", may violate the noise ordinance of some municipalities.

Motorcycles have been utilized with similar equipment since they also have the so-called "car audio" experience. Even pedal bicycles, as well as homemade boomboxes have utilized sealed lead-acid batteries (or 12V power supplies) for applications outside of motor vehicle use, likewise the store displays which mount in demo models prior to aftermarket purchases for installation.

History

Early 1970s tractor with a radio/8-track system

1930s

From the earliest days of radio, enthusiasts had adapted domestic equipment to use in their cars. The commercial introduction of the fitted car radio came in the 1930s from the Galvin Manufacturing Corporation. Galvin Manufacturing was owned and operated by Paul V. Galvin and his brother Joseph E. Galvin. The Galvin brothers purchased a battery eliminator business in 1928 and the corporation's first product was a battery eliminator that allowed vacuum tube battery-powered radios to run on standard household electric current. In 1930, the Galvin Corporation introduced one of the first commercial car radios, the Motorola model 5T71, which sold for between $110 and $130 (*2009: $1,700*) and could be installed in most popular automobiles. Founders Paul Galvin and Joe Galvin came up with the name 'Motorola' when his company started manufacturing car radios. A number of early companies making phonographs, radios, and other audio equipment in the early 20th century used the suffix "-ola," the most famous being Victrola; RCA made a "radiola"; there was also a company that made jukeboxes called Rock-Ola, and a film-editing device called a Moviola. The Motorola prefix "motor-" was chosen because the company's initial focus was in automotive electronics.

In Germany Blaupunkt fitted their first radio to a Studebaker in 1932 and in the United Kingdom Crossley offered a factory fitted wireless in their 10 hp models from 1933. The early car radio receivers used the battery voltage (6.3 volts at the time) to run the vacuum tube filaments, and generated the required high voltage for the plate supply using a vibrator to drive a step-up transformer. The receivers required more stages than the typical home receiver in order to ensure that enough gain was available to allow the AGC to mask signal fading as the car was driven. When cars switched to 12-volt batteries, the same arrangement was used, with tubes with 12-volt heaters. In 1952 Blaupunkt became the first maker to offer FM receivers.

1950s

A common feature of modern car radios is the "seek" function which allows tuning from one station to the next at the push of a button. This was a popular option on some Ford products in the 1950s. It was known as the "Town & Country" radio since it used a pair of switches marked "Town" and "Country." Pressing the Town button actuated a motor to rotate the tuning mechanism while the receiver sensitivity was reduced so that only local (stronger) signals would be received. When a station was tuned, the motor stopped. Pressing the Country button had the same effect except that full sensitivity was enabled so that the very next available station would be selected. In addition, for repeated seeking operations, pressing a foot switch on the driver's floor up to the left where the "dead pedal" is located on modern cars would reactivate the Seek at whatever sensitivity was last selected.

1960s-1970s

The introduction of semiconductors (transistors) allowed the output stage to change to a transistor, which soon lead to the elimination of the vibrator, and the use of "space charge" tubes that only required 12 volts on their plates without a high voltage plate power supply (typical example was the 6GM8/ECC86). Advances in electronics allowed additions to the basic radio and Motorola offered 16 2/3 rpm disc players fitted to some Chryslers known as "Hiway HI FI" from as early as 1956 and ran through 1958. Records were produced under license by Comumbia "Special Products division and sold exclusively through Chrysler dealers. The 45 rpm record player was introduced in 1959 and ran through the early 60"s under the RCA and ARC brand. Earl "Mad Man Muntz" introduced the "4 track" tape player in the early 60's using a continuous loop cartidge and was the first commercially available "car stereo. Tape players using reel to reel equipment followed, but their bulk ensured limited popularity. This changed in 1964 when Philips launched the Compact Cassette. During the '60s Lear invented and introduced the 8track cartridge in competition with the cassette system. Other early manufacturers and enthusiasts began building extra audio amplifiers to run on 12 volts (the standard voltage in automotive electrical systems). Jim Fosgate, later to become the founder of Rockford Fosgate, was one such pioneer. The company *a/d/s* also brought an amplifier to market in 1978.

1980s-1990s

In 1983, Zed Audio became the first company to build a 200 watt per channel car amplifier, which was invented by company founder Steven Mantz. At first, speakers from the home audio and professional markets were simply installed into vehicles. However, they were not well suited to the extremes of temperature and vibration which are a normal part of the environment of an automobile. Different manufacturing techniques, and different component materials were used in construction to adapt to these conditions.

Car audio competitions started in the early 1980s

The first known occurred in 1981 in Bakersfield, CA and evolved into an annual event. It was called The Summertime Car Show and Sound Off Competition, which at its height drew upwards of 300 contestants and continued into the 1990s. The Summertime Car Show and Sound Off Competition began as a promotional event for Cars on Camera, a magazine founded by owners Steve Silver and Scott Burud. Since the magazine derived a large part of its advertising revenue from local car stereo shops (TransLex, AutoSounds and others) it made sense to hold a sound off competition in order to create higher demand for magazine ad space. The original event took place in the parking lot of the local Zody's chain store on Ming Avenue, in Bakersfield, CA. However, the following year it was moved to the Kern County fairgrounds in order to accommodate the thousands of participants. By the second year, the event added a men's great legs contest and a bikini contest that attracted contestants from all over California. Cars on Camera changed its name to Camera Ads, which was then sold to Buck Owens Productions.

The most important of these were CAN (formed by Alpine) and NACA (supported by shop owners and amp manufacturers). Both organizations sanctioned countrywide regional events and hosted National Championship events in the late 1980s. They merged to form IASCA in 1990. Despite the move to "quality" based judging, volume was still a significant portion of most early 1990s competitions. Since then, the two styles—SPL vs. sound quality—have become almost mutually exclusive. The loudness competitions have become known as dB drag racing. Currently, MEASQ conducts Sound Quality Competitions nationally in Australia. This back to basics competition format was developed by Marc Rushton, the founder of one of the largest enthusiast organizations known as Mobile Electronics Australia.

Common components and terms

Stock unit

A *stock* car audio system refers to the OEM application that the vehicle's manufacturer specified to be installed when the car was built and nowadays at least includes a CD-radio, with MP3 player and an aux-in. A large after market industry exists where the consumer can at their desire replace or complement many or all components of the stock system (i.e. kits to include a USB port and A2DP bluetooth to the stock radio-CD). Nowadays, the most valued port (40% of the users) is the USB.

Head unit

A Panasonic single DIN head unit, combining radio, CD and MP3

In modern cars, the primary control device for an audio system is commonly referred to as a head unit, and is installed in the center of the dash panel between the driver and the passenger. In older vehicles that had audio components as an option, such devices were mounted externally to the top of or underneath the dash.

The headunit itself is usually a multi-purpose device that houses multiple types of components in its housing. The most common components are a radio receiver/tuner usually with AM and FM bands, and a small amplifier for driving an audio signal to speakers. Other possible components include various media devices, such as (in older vehicles) a tape player (either 8-track or cassette), CD player, DVD player, USB flash memory, and even a portable hard disk drive typically used in notebook computing. Many head units also feature a DSP component, and equalization component (such as bass and treble controls), or a control interface for another feature on the car (such as a back-up/parking camera, navigation system, trip odometer, etc.).

Due to auto manufacturing differences over the years, aftermarket headunit products are manufactured in multiple form factors. The primarily used size is mostly referred to by its legacy name of DIN, which refers to ISO 7736. DIN headunits come as single DIN or double DIN. A third less common standard is used mostly by Chrysler group and for a time Mitsubishi in their OEM devices.

Speakers

A set of speaker drivers removed from a passenger vehicle.

Car **speakers** are largely functionally identical to any other loudspeaker design with key components specialized for use in mobile environments, and generally serve an identical purpose. One major key design difference is mult-axial mounting of different types of loudspeakers in the same footprint, such as a tweeter directly mounted over a woofer. Another key difference is non-circular cone shapes, such as square, oval, or even triangular. Both of these features reflect a significant reduction in space and size that a speaker may occupy in a vehicle cabin.

Material construction may also include more exotic and hearty components more suitable to mobile use. Marine speakers may have plating for corrosion resistance. Cones may be coated with a substance to resist expansion and contraction under high vehicle cabin temperatures, known to reach 140 °F (60 °C) in the sun. Subwoofers may also be found in mobile audio applications where a cabin speaker may lack the desired low frequency response on its own.

Before stereo radio was introduced, the most common speaker location was in the middle of the dashboard pointing through perforations towards the front windshield. In most modern applications, speakers are mounted certain common locations including the front deck (or dash), the rear deck (or parcel shelf), the kick panel (located in the footwell below the A-pillar,) or the doors. In the case of subwoofers, mountings are usually under the seat or in the trunk. Each position has certain strengths and limitations from both a quality of sound, and a vehicle manufacturing perspective.

5.1 and even 7.1 channel surround sound systems, as well as THX II Certified, are now being integrated into some cars by both aftermarket enthusiasts and car manufacturers themselves. These systems include the full complement of front left, right and center speakers along with rear right and left surround speakers.

Amplifiers

A car audio amplifier.

Blaupunkt Class T amplifier

Basically a mobile audio amplifier, a car 'amp' is a term used to refer to a dedicated electronic amplifier separated from the other components of the system. Though most head units have an amplifier, some do not, or lack the desired power or additional features (e.g., equalization controls or crossover systems). External amplification is available and most often used when existing amplification is insufficient. External amplifiers can be mounted in a different part of the car than the "head unit"; in many cases, an additional amp is mounted in the trunk. This is usually the case when powering a subwoofer, where desired wattage may be several multiples more compared to other cabin speakers.

Though less common, OEM external amplification can be found in 'premium' audio packages, or in luxury cars. More common is aftermarket amplification installed later to satisfy the expansion of an existing system in some way. During operation, it is common for a vehicle's charging system to fluctuate, so a regulated amplifier will maintain its power output regardless of voltage fluctuation. Amplifiers rated at 100 watts at 14.4 volts can not be regarded equal as to an amplifier that can maintain 100 watts at 12 volts. Outside of certain standards, it is not uncommon for manufacturers to list a 14.4 rating and not post a 12 volt value.

Subwoofers

Subwoofers are a specific type of loudspeaker for low frequency reproduction. Mobile 'subs' are not very different from any other application of sub in terms of construction. However it is more common in aftermarket that visual aesthetics take on a more significant role in design than other types of sub drivers, including high contrast paint schemes, grill covers, translucent or refractive materials. Typical subwoofer drivers range in size from an 8" diameter to 10", 12" or 15"; more rarely, some car systems may have 18", 21", 22", 24" or even 32" subwoofers.

A subwoofer is used when existing low frequency production is unsatisfactory, either in frequency range or in volume. Design goals have led to subwoofer, both driver's alone and whole packages, with some extreme difference from one another. Space conscious design has reduced some driver depth to 2" or less, or enclosure depth to 3". Pure loudness through increasing sound pressure has led to some drivers with excursions as great as 4" and vented components to cool the "motor" of the speaker. Quality and clarity has led to driver enclosures being tuned by construction to resonate or neutralize certain frequencies.

Capacitors

A powerful after-market audio system installation in a Toyota

Capacitors are used to store energy for the amplifier to draw on demand. They come in many different sizes ranging from 0.5 farad to well over 100 farads and their intended function is to temporarily cover the short-burst electrical demands of a car audio system that have exceeded the general electrical capabilities of the vehicle. There is little evidence to suggest they impart any benefit to the system, however, due to their low energy storage (compared with the battery) and exponential nature of capacitor voltage decay.

Damping

Sound deadening material is often used in the door cavities and boot/trunk area to damp excess vibration of the panels in the car in response to loud subwoofer bass tones, especially the boot/trunk. The most common type of deadening is either butyl or rubberized asphalt, a product which has an adhesive quality and can be applied by simply pressing it into place with a roller and using a heat gun (or hair dryer). Other types of deadening can be sprayed on, but they are less common because of the additional installation difficulties.

Other components

Uniden BCT-15 radio scanner installed with aftermarket head unit

Other components that make up high-end car audio installations may include:

- Multiple-CD Changer
- amplifiers
- audio processors
- cables
- crossovers
- equalizers
- mobile video (VCRs, television, DVD and navigation)
- Controls, including on steering wheel interface, as well as remote controls

- Car computer, fully functional computer (i.e. Internet, Music, games) that is operable from the interface.
- Gaming consoles – passenger entertainment

Legal problems

Extremely loud sound systems in automobiles may violate the noise ordinance of some municipalities. Some cities have even outlawed so called "boom cars", vehicles containing loud stereo systems that emit low frequency sound, usually with an intense amount of bass. A number of organizations and websites are dedicated to lobbying for tougher restrictions on boom cars, citing that they disturb the peace and cause documented health problems. Noise Free America, a 501(c)(3) non-profit group, cites boom cars as one of the most problematic sources of noise pollution. In 2007, the U.S. Department of Justice issued a guide to police officers on how to deal with problems associated with boom cars.

Chapter 7

Sound Recording and Reproduction

Sound recording and reproduction is an electrical or mechanical inscription and re-creation of sound waves, such as spoken voice, singing, instrumental music, or sound effects. The two main classes of sound recording technology are analog recording and digital recording. Acoustic analog recording is achieved by a small microphone diaphragm that can detect changes in atmospheric pressure (acoustic sound waves) and record them as a graphic representation of the sound waves on a medium such as a phonograph (in which a stylus senses grooves on a record). In magnetic tape recording, the sound waves vibrate the microphone diaphragm and are converted into a varying electric current, which is then converted to a varying magnetic field by an electromagnet, which makes a representation of the sound as magnetized areas on a plastic tape with a magnetic coating on it. Analog sound reproduction is the reverse process, with a bigger loudspeaker diaphragm causing changes to atmospheric pressure to form acoustic sound waves. Electronically generated sound waves may also be recorded directly from devices such as an electric guitar pickup or a synthesizer, without the use of acoustics in the recording process other than the need for musicians to hear how well they are playing during recording sessions.

Digital recording and reproduction converts the analog sound signal picked up by the microphone to a digital form by a process of digitization, allowing it to be stored and transmitted by a wider variety of media. Digital recording stores audio as a series of binary numbers representing samples of the amplitude of the audio signal at equal time intervals, at a sample rate so fast that the human ear perceives the result as continuous sound. Digital recordings are considered higher quality than analog recordings not necessarily because they have higher fidelity (wider frequency response or dynamic range), but because the digital format can prevent much loss of quality found in analog recording due to noise and electromagnetic interference in playback, and mechanical deterioration or damage to the storage medium. A digital audio signal must be reconverted to analog form during playback before it is applied to a loudspeaker or earphones.

History

Origins

The automatic reproduction of music can be traced back as far as the 9th century, when the Banū Mūsā brothers invented "the earliest known mechanical musical instrument", in this case a hydropowered organ which played interchangeable cylinders automatically. According to Charles B. Fowler, this "cylinder with raised pins on the surface remained the basic device to produce and reproduce music mechanically until the second half of the nineteenth century." The Banu Musa also invented an automatic flute player which appears to have been the first programmable machine.

In the 14th century, Flanders introduced a mechanical bell-ringer controlled by a rotating cylinder. Similar designs appeared in barrel organs (15th century), musical clocks (1598), barrel pianos (1805), and musical boxes (1815).

All of these machines could play stored music, but they could not play arbitrary sounds, could not record a live performance, and were limited by the physical size of the medium. The first device that could record sound mechanically (but could not play it back) was the phonautograph, developed in 1857 by Parisian inventor Édouard-Léon Scott de Martinville. The earliest known recordings of the human voice were phonautograms also made in 1857. These earliest known recordings include a dramatic reading in French of Shakespeare's *Othello* and music played on a guitar and trumpet. The recordings consist of groups of wavy lines scratched by a stylus onto fragile paper that was blackened by the soot from an oil lamp. One of his phonautograms of *Au Clair de la Lune*, a French folk song, was digitally converted to sound in 2008. While this is an interesting playback that sounds like a girl singing, the creator of this recording, Patrick Feaster of Indiana University in Bloomington, reports that phonautograms his team had previously transcribed, using a laser as a virtual stylus, had been played back at twice the actual speed. What sounded like a girl singing the French folksong was actually Léon Scott singing, Feaster concluded in May, 2009. Since the above recording was recovered, the same team have since recovered a recording of a 435-Hz tuning fork (at that time the French standard concert pitch for A' — now 440 Hz). The tuning fork is barely audible.

The player piano, first demonstrated in 1876, used a punched paper scroll that could store an arbitrarily long piece of music. This piano roll moved over a device known as the 'tracker bar', which first had 58 holes, was expanded to 65 and then was upgraded to 88 holes (generally, one for each piano key). When a perforation passed over the hole, the note sounded. Piano rolls were the first stored music medium that could be mass-produced, although the hardware to play them was much too expensive for personal use. Technology to record a live performance onto a piano roll was not developed until 1904. Piano rolls have been in continuous mass production since around 1898. A 1908 U.S. Supreme Court copyright case noted that, in 1902 alone, there were between 70,000 and 75,000 player pianos manufactured, and between 1,000,000 and 1,500,000 piano rolls produced. The use of piano rolls began to decline in the 1920s although one type is still being made today. The fairground organ, developed in 1892, used a similar system of accordion-folded punched cardboard books.

Phonograph

Phonograph cylinder

Frances Densmore recording Blackfoot chief Mountain Chief on a cylinder phonograph for the Bureau of American Ethnology (1916)

The first practical sound recording and reproduction device was the mechanical phonograph cylinder, invented by Thomas Edison in 1877 and patented in 1878. The invention soon spread across the globe and over the next two decades the commercial recording, distribution and sale of sound recordings became a growing new international industry, with the most popular titles selling millions of units by the early 1900s. The development of mass-production techniques enabled cylinder recordings to become a major new consumer item in industrial countries and the cylinder was the main consumer format from the late 1880s until around 1910.

Disc phonograph

The next major technical development was the invention of the gramophone disc, generally credited to Emile Berliner and commercially introduced in the United States in 1889. Discs were easier to manufacture, transport and store, and they had the additional benefit of being louder (marginally) than cylinders, which by necessity, were single-sided. Sales of the Gramophone record overtook the cylinder ca. 1910, and by the end of World War I the disc had become the dominant commercial recording format. Edison, who was the main producer of cylinders, created the Edison Disc Record in an attempt to regain his market. In various permutations, the audio disc format became the primary medium for consumer sound recordings until the end of the 20th century, and the double-sided 78 rpm shellac disc was the standard consumer music format from the early 1910s to the late 1950s.

Although there was no universally accepted speed, and various companies offered discs that played at several different speeds, the major recording companies eventually settled on a *de facto* industry standard of nominally 78 revolutions per minute, though the actual speed differed between America and the rest of the world. The specified speed was 78.26 rpm in America and 77.92 rpm throughout the rest of the world, the difference in speeds a result of the difference in cycle frequencies of the AC power driving the synchronous motor) and available gearing ratios. The nominal speed of the disc format gave rise to its common nickname, the "seventy-eight" (though not until other speeds had become available). Discs were made of shellac or similar brittle plastic-like materials, played with needles made from a variety of materials including mild steel, thorn and even sapphire. Discs had a distinctly limited playing life which was heavily dependent on how they were reproduced.

The earlier, purely acoustic methods of recording had limited sensitivity and frequency range. Mid-frequency range notes could be recorded but very low and very high frequencies could not. Instruments such as the violin transferred poorly to disc; however this was partially solved by retrofitting a conical horn to the sound box of the violin. The horn was no longer required once electrical recording was developed.

The Vinyl microgroove was invented by a Hungarian engineer Peter Carl Goldmark. The vinyl microgroove record was introduced in the late 1940s, and the two main vinyl formats — the 7-inch single turning at 45 rpm and the 12-inch LP (long-playing) record turning at 33 1/3 rpm — had totally replaced the 78 rpm shellac (sometimes vinyl) disc by the end of the 1950s. Vinyl offered improved performance, both in stamping and in playback, and came to be generally played with polished diamond styli, and when played properly (precise tracking weight, etc.) offered longer life. Vinyl records were, overoptimistically, advertised as "unbreakable". They were not, but were much less brittle and breakable than shellac. Nearly all were tinted black, but some were colored, as red, swirled, translucent, etc.

Electrical recording

RCA-44, a classic ribbon microphone

Sound recording began as a mechanical process and remained so until the early 1920s (with the exception of the 1899 Telegraphone) when a string of groundbreaking inventions in the field of electronics revolutionised sound recording and the young recording industry. These included sound transducers such as microphones and loudspeakers, and various electronic devices such as the mixing desk, designed for the amplification and modification of electrical sound signals.

After the Edison phonograph itself, arguably the most significant advances in sound recording, were the electronic systems invented by two American scientists between 1900 and 1924. In 1906 Lee De Forest invented the "Audion" triode vacuum-tube, electronic valve, which could greatly amplify weak electrical signals, (one early use was to amplify long distance telephone in 1915) which became the basis of all subsequent electrical sound systems until the invention of the transistor. The valve was quickly followed by the invention of the Regenerative circuit, Super-Regenerative circuit and the Superheterodyne receiver circuit, all of which were invented and patented by the young electronics genius Edwin Armstrong between 1914 and 1922. Armstrong's inventions made higher fidelity electrical sound recording and reproduction a practical reality, facilitating the development of the electronic amplifier and many other devices; after 1925 these systems had become standard in the recording and radio industry.

While Armstrong published studies about the fundamental operation of the triode vacuum tube before World War I, inventors like Orlando R. Marsh and his Marsh Laboratories, as well as scientists at Bell Telephone Laboratories, achieved their own understanding about the triode and were utilizing the Audion as a repeater in weak telephone circuits. By 1925 it was possible to place a long distance telephone call with these repeaters between New York and San Francisco in 20 minutes, both parties being clearly heard. With this technical prowess, Joseph P. Maxfield and Henry C. Harrison from Bell Telephone Laboratories were skilled in using mechanical analogs of electrical circuits and applied these principles to sound recording and reproduction. They were ready to demonstrate their results by 1924 using the Wente condenser microphone and the vacuum tube amplifier to drive the "rubber line" wax recorder to cut a master audio disc.

Meanwhile, radio continued to develop. Armstrong's groundbreaking inventions (including FM radio) also made possible the broadcasting of long-range, high-quality radio transmissions of voice and music. The importance of Armstrong's Superheterodyne circuit cannot be over-estimated — it is the central component of almost all analog amplification and both analog and digital radio-frequency transmitter and receiver devices to this day.

Beginning during World War One, experiments were undertaken in the United States and Great Britain to reproduce among other things, the sound of a Submarine (u-boat) for training purposes. The acoustical recordings of that time proved entirely unable to reproduce the sounds, and other methods were actively sought. Radio had developed independently to this point, and now Bell Laboritories sought a marriage of the two disparate technologies, greater than the two separately. The first experiments were not very promising, but by 1920 greater sound fidelity was achieved using the electrical system than had ever been realized acoustically. One early recording made without fanfare or announcement was the dedication of the Tomb of the Unknown Soldier at Arlington Cemetery.

By early 1924 such dramatic progress had been made, that Bell Labs arranged a demonstration for the leading recording companies, the Victor Talking Machine Company, and the Columbia Phonograph Co. (Edison was left out due to their decreasing market share and a stubborn Thomas Edison). Columbia, always in financial straits, could not afford it, and Victor, essentially leaderless since the mental collapse of founder Eldridge Johnson, left the demonstration without comment. English Columbia, by then a separate company, got hold of a test pressing made by Pathé from these sessions, and realized the immediate and urgent need to have the new system. Bell was only offering its method to United States companies, and to circumvent this, Managing Director Louis Sterling of English Columbia, bought his once parent company, and signed up for electrical recording. Although they were contemplating a deal, Victor Talking Machine was apprised of the new Columbia deal, so they too quickly signed. Columbia made its first released electrical recordings on February 25, 1925, with Victor following a few weeks later. The two then agreed privately to "be quiet" until November 1925, by which time enough electrical repertory would be available.

Other recording formats

In the 1920s, the early talkies featured the new sound-on-film technology which used photoelectric cells to record and reproduce sound signals that were optically recorded directly onto the movie film. The introduction of talking movies, spearheaded by *The Jazz Singer* in 1927 (though it used a sound on disk technique, not a photoelectric one), saw the rapid demise of live cinema musicians and orchestras. They were replaced with pre-recorded soundtracks, causing the loss of many jobs. The American Federation of Musicians took out ads in newspapers, protesting the replacement of real musicians with mechanical playing devices, especially in theatres.

This period also saw several other historic developments including the introduction of the first practical magnetic sound recording system, the magnetic wire recorder, which was based on the work of Danish inventor Valdemar Poulsen. Magnetic wire recorders were effective, but the sound quality was poor, so between the wars they were primarily used for voice recording and marketed as business dictating machines. In the 1930s radio pioneer Guglielmo Marconi developed a system of magnetic sound recording using steel tape. This was the same material used to make razor blades, and not surprisingly the fearsome Marconi-Stille recorders were considered so dangerous that technicians had to operate them from another room for safety. Because of the high recording speeds required, they used enormous reels about one metre in diameter, and the thin tape frequently broke, sending jagged lengths of razor steel flying around the studio.

The K1 Magnetophon was the first practical tape recorder, developed by AEG in Germany in 1935. The other major invention in sound recording in this period was the optical sound-on-film system, also generally credited to Lee De Forest. Although famous early "Talkies" like *The Jazz Singer* used a sound-on-disc system, the film industry rapidly adopted the optical sound-on-film system and it revolutionised the movie industry in the 1930s, ushering in the era of 'talking pictures'. Optical sound-on-film, based on the photoelectric cell, became the standard film audio system throughout the world and remains so for theatrical release prints despite attempts in the 1950s to substitute magnetic recording methods. Currently all release prints on 35mm film include an analogue optical soundtrack (usually stereo with Dolby SR noise reduction). In addition an optically recorded digital soundtrack in Dolby Digital and/or Sony SDDS form is likely to be present. Optically recorded timecode is also commonly found in order to synchronise CDROMs containing a DTS soundtrack.

Magnetic tape

Other important inventions of this period were magnetic tape and the tape recorder (Telegraphone). Paper-based tape was first used but was soon superseded by polyester and acetate backing due to dust drop and hiss. Acetate was more brittle than polyester and snapped easily. This technology, the basis for almost all commercial recording from the 1950s to the 1980s, was invented by German audio engineers in the 1930s, who also discovered the technique of AC biasing, which dramatically improved the frequency response of tape recordings. Tape recording was perfected just after the war by American audio engineer John T. Mullin with the help of Crosby Enterprises (Bing Crosby), whose pioneering recorders were based on captured German recorders, and the Ampex company produced the first commercially available tape recorders in the late 1940s.

A typical Compact Cassette

Magnetic tape brought about sweeping changes in both radio and the recording industry. Sound could be recorded, erased and re-recorded on the same tape many times, sounds could be duplicated from tape to tape with only minor loss of quality, and recordings could now be very precisely edited by physically cutting the tape and rejoining it. Within a few years of the introduction of the first commercial tape recorder -- the Ampex 200 model, launched in 1948 -- American musician-inventor Les Paul had invented the first multitrack tape recorder, ushering in another technical revolution in the recording industry. Tape made possible the first sound recordings totally created by electronic means, opening the way for the bold sonic experiments of the Musique Concrète school and avant garde composers like Karlheinz Stockhausen, which in turn led to the innovative pop music recordings of artists such as Frank Zappa, The Beatles and The Beach Boys.

Magnetic tape allowed the radio industry for the first time to pre-record many sections of program content such as advertising, which formerly had to be presented live, and it also enabled the creation and duplication of complex, high-fidelity, long-duration recordings of entire programs. Also, for the first time, broadcasters, regulators and other interested parties were able to undertake comprehensive logging of radio broadcasts. Innovations like multitracking and tape echo enabled radio programs and advertisements to be pre-produced to a level of complexity and sophistication that was previously unattainable and the combined impact of these new techniques led to significant changes to the pacing and production style of program content, thanks to the innovations like the endless-loop broadcast cartridge.

Stereo and hi-fi

In 1881, it was noted during experiments in transmitting sound from the Paris Opera that it was possible to follow the movement of singers on the stage if earpieces connected to different microphones were held to the two ears. However, this observation was not followed up or investigated further at the time.

In 1931 Alan Blumlein, a British electronics engineer working for EMI, designed a way to make the sound of an actor in a film follow their movement across the screen. In December 1931 he submitted a patent including the idea, and in 1933 this became UK patent number 394,325. Over the next two years, Blumlein developed stereo microphones and a stereo disc-cutting head, and recorded a number of short films with stereo soundtracks.

Magnetic tape enabled the development of the first practical commercial sound systems that could record and reproduce high-fidelity stereophonic sound. The experiments with stereo during the 1930s and 1940s were hampered by problems with synchronization. A major breakthrough in practical stereo sound was made by Bell Laboratories, who in 1937 demonstrated a practical system of two-channel stereo, using dual optical sound tracks on film. Major movie studios quickly developed three-track and four-track sound systems, and the first stereo sound recording for a commercial film was made by Judy Garland for the MGM movie *Listen, Darling* in 1938.

The first movie commercially released with a stereo soundtrack was Walt Disney's *Fantasia*, released in 1940. The original 1941 release of this production used the "Fantasound" sound system. This system employed a separate film for the sound, which ran in synchronism with the film carrying the picture. On this sound film were four double-width optical soundtracks, three of which carried left, center and right audio whilst the fourth was a "control" track on which were recorded three tones which controlled the playback volume of the three audio channels. Because of the complex equipment required to present it, it was shown as a roadshow, but only in the United States. Regular releases of the film were on standard mono optical 35 mm stock until 1956 when the film was released with a stereo soundtrack using the "Cinemascope" four-track magnetic sound system.

German audio engineers working on magnetic tape are reported to have developed stereo recording by 1943, but it was not until the introduction of the first commercial two-track tape recorders by Ampex in the late 1940s that stereo tape recording became commercially feasible. However, despite the availability of multitrack tape, stereo did not become the standard system for commercial music recording for some years and it remained a specialist market during the 1950s. This changed after the late 1957 introduction of the "Westrex stereo phonograph disc", which used the groove format developed earlier by Blumlein. Decca Records in England came out with FFRR (Full Frequency Range Recording) in the 1940s which became internationally accepted and a worldwide standard for higher quality recordings on vinyl records. The Ernest Ansermet recording of Igor Stravinsky's Petrushka was key in the development of full frequency range records and alerting the listening public to high fidelity in 1946.

Most pop singles were mixed into monophonic sound until the mid 1960s, and it was common for major pop releases to be issued in both mono and stereo until the early 1970s. Many Sixties pop albums now available only in stereo were originally intended to be released only in mono, and the so-called "stereo" version of these albums were created by simply separating the two tracks of the master tape. In the mid Sixties, as stereo became more popular, many mono recordings (such as The Beach Boys' *Pet Sounds*) were remastered using the so-called "fake stereo" method, which spread the sound across the stereo field by directing higher-frequency sound into one channel and lower-frequency sounds into the other.

1950s and beyond

Magnetic tape transformed the recording industry, and by the late-1950s the vast majority of commercial recordings were being mastered on tape. The electronics revolution that followed the invention of the transistor brought other radical changes, the most important of which was the introduction of the world's first "personal music device", the miniaturized transistor radio, which became a major consumer luxury item in the 1960s, transforming radio broadcasting from a static group experience into a mobile, personal listening activity. An early multitrack recording made using magnetic tape was "How High the Moon" by Les Paul, on which Paul played eight overdubbed guitar tracks. In the 1960s Brian Wilson of The Beach Boys, Frank Zappa and The Beatles (with producer George Martin) were among the first popular artists to explore the possibilities of multitrack techniques and effects on their landmark albums *Pet Sounds*, *Freak Out!* and *Sgt. Pepper's Lonely Hearts Club Band*.

The next important innovation was small cartridge based tape systems of which the compact cassette, introduced by the Philips electronics company in 1964 is the best known. It eventually entirely replaced the competing formats, the larger 8-track tape (used primarily in cars) and the fairly similar 'Deutsche Cassette' developed by the German company Grundig. This latter system was not particularly common in Europe and practically unheard of in America. The compact cassette became a major consumer audio format and advances in microelectronics eventually allowed the development of the Sony Walkman, introduced in the 1970s, which was the first personal music player and gave a major boost to the mass distribution of music recordings. Cassettes became the first successful consumer recording/re-recording medium. The gramophone record was a pre-recorded playback only medium, and reel-to-reel tape was too difficult for most consumers and far less portable.

A key advance in audio fidelity came with the Dolby A noise reduction system, invented by Ray Dolby and introduced in 1966. A competing system dbx, invented by David

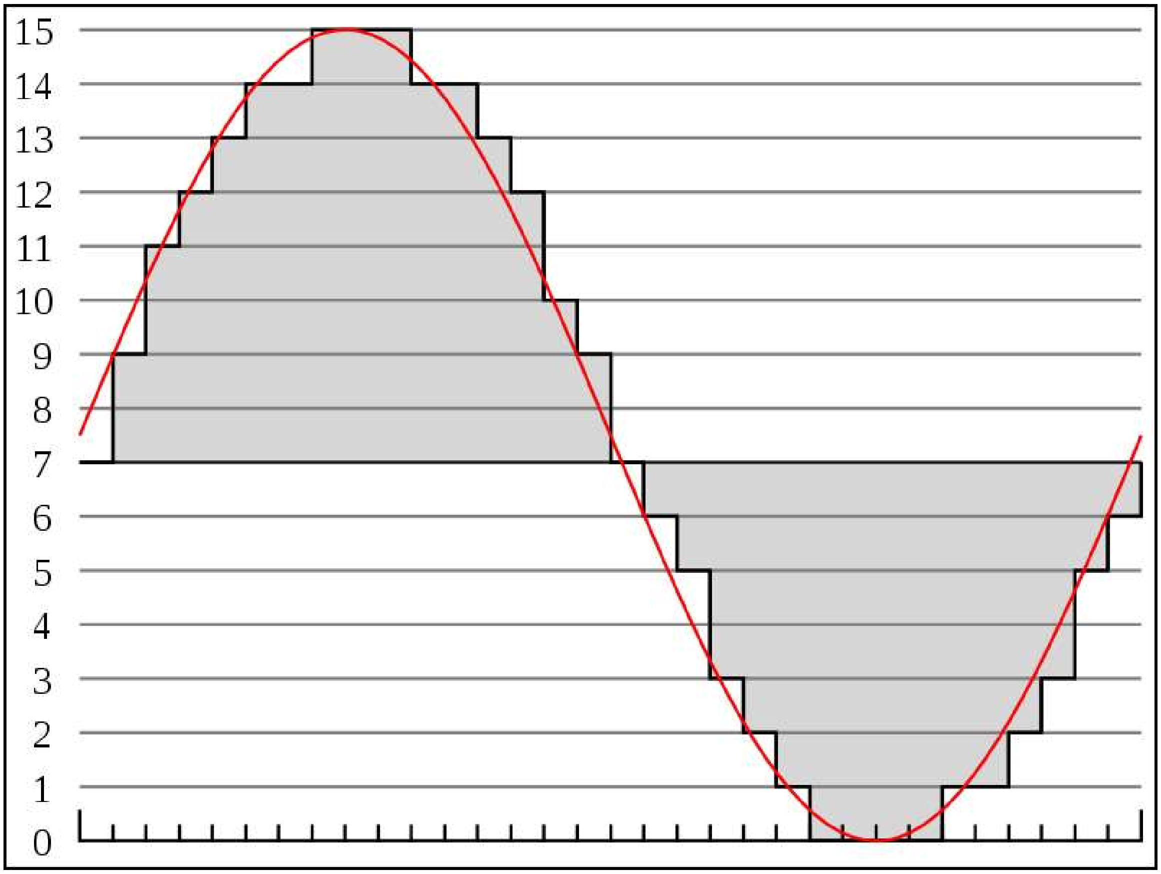
Blackmer, found most success in professional audio. A simpler variant of Dolby's noise reduction system, known as Dolby B greatly improved the sound of cassette tape recordings by reducing the practical effect of the recorded hiss inherent in the narrow tape used. It, and variants, also eventually found wide application in the recording and film industries. Dolby B was crucial to the popularisation and commercial success of the compact cassette as a domestic recording and playback medium, and became a part of the booming "hi-fi" market of the 1970s and beyond. The compact cassette also benefited enormously from developments in the tape material itself as materials with wider frequency responses and lower inherent noise were developed, often based on cobalt and/or chrome oxides as the magnetic material instead of the more usual iron oxide.

The multitrack audio cartridge had been in wide use in the radio industry, from the late 1950s to the 1980s, but in the 1960s the pre-recorded 8-track cartridge was launched as a consumer audio format by Bill Lear of the Lear Jet aircraft company (and although its correct name was the 'Lear Jet Cartridge', it was seldom referred to as such). Aimed particularly at the automotive market, they were the first practical, affordable car hi-fi systems, and could produce superior sound quality to the compact cassette. However the smaller size and greater durability — augmented by the ability to create home-recorded music "compilations" since 8-track recorders were rare — saw the cassette become the dominant consumer format for portable audio devices in the 1970s and 1980s.

There had been experiments with multi-channel sound for many years — usually for special musical or cultural events — but the first commercial application of the concept came in the early 1970s with the introduction of Quadraphonic sound. This spin-off development from multitrack recording used four tracks (instead of the two used in stereo) and four speakers to create a 360-degree audio field around the listener. Following the release of the first consumer 4-channel hi-fi systems, a number of popular albums were released in one of the competing four-channel formats; among the best known are Mike Oldfield's *Tubular Bells* and Pink Floyd's *The Dark Side of the Moon*. Quadraphonic sound was not a commercial success, partly because of competing and somewhat incompatible four-channel sound systems (e.g., CBS, JVC, Dynaco and others all had systems) and generally poor quality, even when played as intended on the correct equipment, of the released music. It eventually faded out in the late 1970s, although this early venture paved the way for the eventual introduction of domestic Surround Sound systems in home theatre use, which have gained enormous popularity since the introduction of the DVD. This widespread adoption has occurred despite the confusion introduced by the multitude of available surround sound standards.

The replacement of the thermionic valve (vacuum tube) by the smaller, cooler and less power-hungry transistor also accelerated the sale of consumer high-fidelity "hi-fi" sound systems from the 1960s onward. In the 1950s most record players were monophonic and had relatively low sound quality; few consumers could afford high-quality stereophonic sound systems. In the 1960s, American manufacturers introduced a new generation of "modular" hi-fi components — separate turntables, pre-amplifiers, amplifiers, both combined as integrated amplifiers, tape recorders, and other ancillary equipment (like the graphic equaliser), which could be connected together to create a complete home sound system. These developments were rapidly taken up by Japanese electronics companies, which soon flooded the world market with relatively cheap, high-quality components. By the 1980s, corporations like Sony had become world leaders in the music recording and playback industry.

Digital recording



Graphical representation of a sound wave in analog (red) and 4-bit digital (black).

The invention of digital sound recording and later the compact disc in 1982 brought significant improvements in the durability of consumer recordings. The CD initiated another massive wave of change in the consumer music industry, with vinyl records effectively relegated to a small niche market by the mid-1990s. However, the introduction of digital systems was initially fiercely resisted by the record industry which feared wholesale piracy on a medium which was able to produce perfect copies of original released recordings. However, the industry had to bow to the inevitable, but not without using various protection system (principally Serial Copy Management System, or SCMS).

A digital sound recorder from Sony

The most recent and revolutionary developments have been in digital recording, with the development of various uncompressed and compressed digital audio file formats, processors capable and fast enough to convert the digital data to sound in real time, and inexpensive mass storage. This generated a new type of portable digital audio player. The minidisc player, using ATRAC compression on small, cheap, re-writeable discs was introduced in the 1990s but became obsolescent as solid-state non-volatile flash memory dropped in price. As technologies which increase the amount of data that can be stored on a single medium, such as Super Audio CD, DVD-A, Blu-ray Disc and HD DVD become available, longer programs of higher quality fit onto a single disc. Sound files are readily downloaded from the Internet and other sources, and copied onto computers and digital audio players. Digital audio technology is used in all areas of audio, from casual use of music files of moderate quality to the most demanding professional applications. New applications such as internet radio and podcasting have appeared.

Technological developments in recording and editing have transformed the record, movie and television industries in recent decades. Audio editing became practicable with the invention of magnetic tape recording, but digital audio and cheap mass storage allows computers to edit audio files quickly, easily, and cheaply. Today, the process of making a recording is separated into tracking, mixing and mastering. Multitrack recording makes it possible to capture signals from several microphones, or from different 'takes' to tape or disc, with maximized headroom and quality, allowing previously unavailable flexibility in the mixing and mastering stages for editing, level balancing, compressing and limiting, adding effects such as reverberation, equalisation, flanging, and much more.

Digital recording and processing software

There are many different digital audio recording and processing programs running under several computer operating systems for all purposes, from professional through serious amateur to casual user.

A comprehensive list of digital recording applications is available on the digital audio workstation page.

Digital dictation software for recording and transcribing speech has different requirements; intelligibility and flexible playback facilities are priorities, while a wide frequency range and high audio quality are not.

Voice to note

Voice-to-note refers to the capability of personal computers to be able to recognize notes that are sung, hummed, or whistled into a microphone. The pitch and duration of the notes are then calculated and converted into MIDI music files.

Legal status

UK

Since 1934, sound recordings are treated differently from musical works under copyright law. *Copyright, Designs and Patents Act 1988* defines a sound recording to mean *(a) a recording of sounds, from which the sounds may be reproduced, or (b) a recording of the whole or any part of a literary, dramatic or musical work, from which sounds reproducing the work or part may be produced, regardless of the medium on which the recording is made or the method by which the sounds are reproduced or produced*. It thus covers vinyl records, tapes, compact discs, digital audiotapes, and MP3s which embody recordings.

Chapter 8

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.

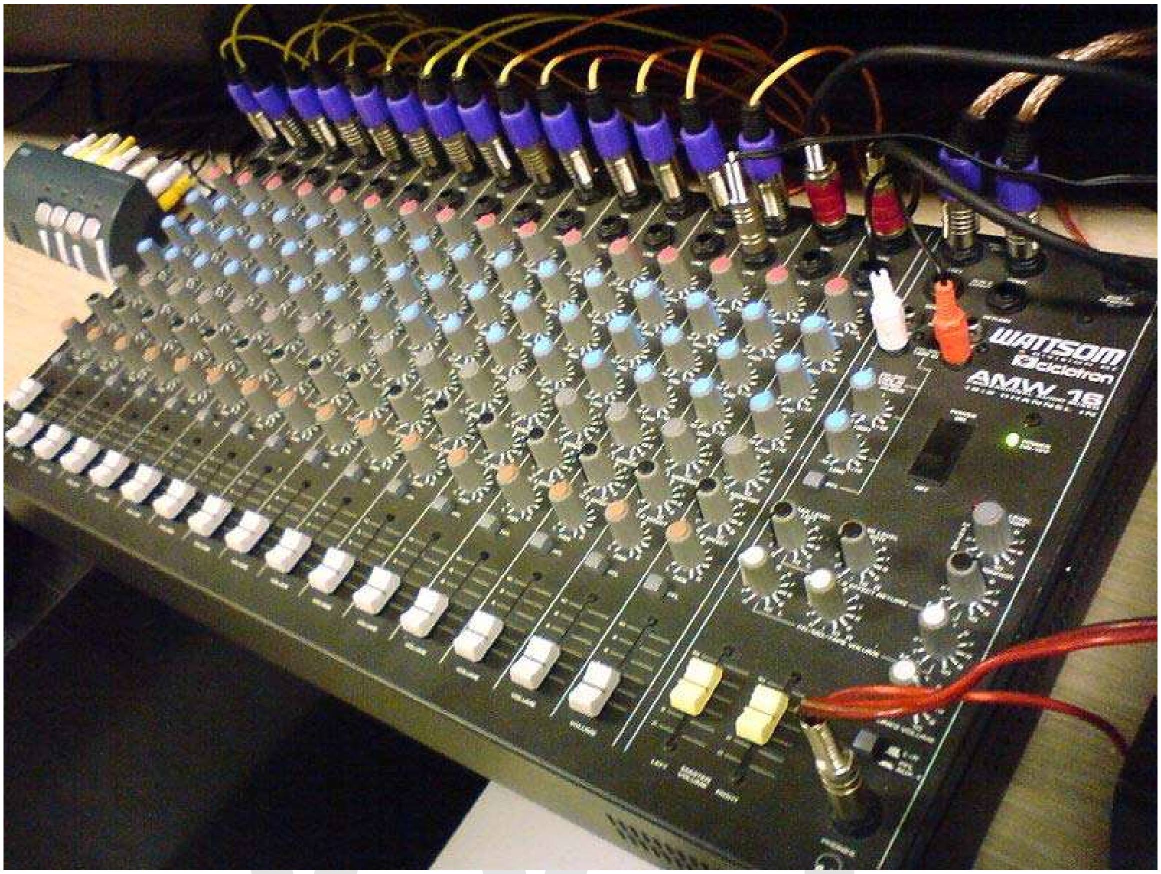
Propagation delay

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.

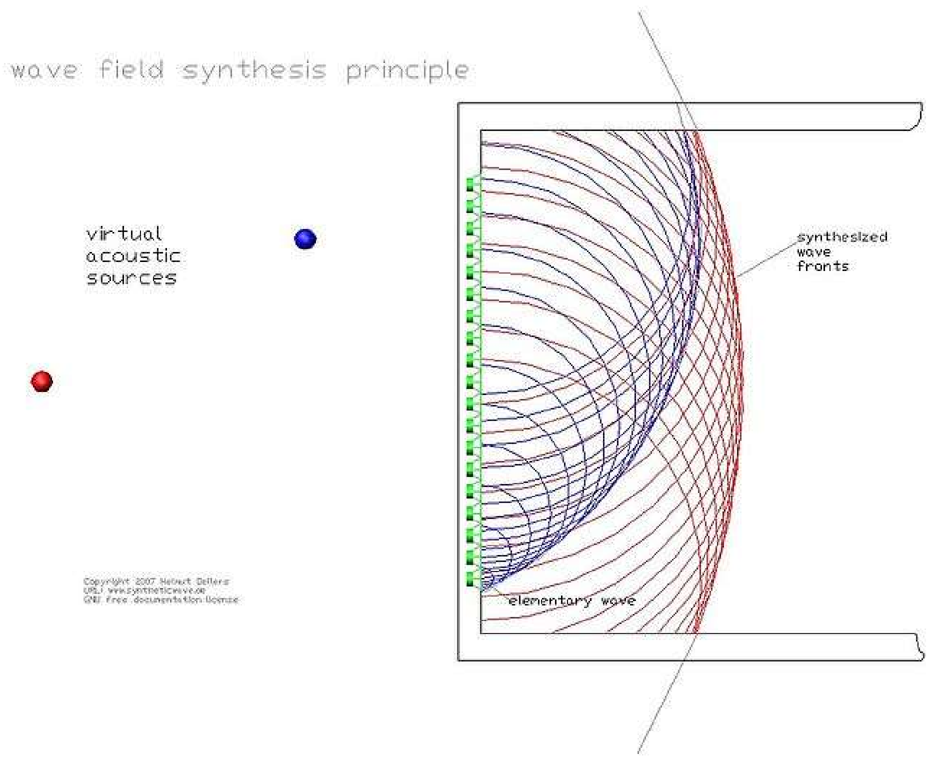
Mixing console manufacturers

- Alesis
- Allen & Heath
- Audient
- Automated Processes, Inc.
- AMS Neve
- Behringer
- Cadac Electronics
- Calrec Audio
- Carvin A&I
- Crest Audio
- D&R
- DiGiCo
- Digidesign
- DigiTech
- DHD
- EAW
- Euphonix
- Fairlight
- Daniel Flickinger
- Harris Corporation
- Harrison Audio Consoles
- Innovason
- Klotz Digital
- Langevin
- Mackie
- MCI
- Midas Consoles
- Peavey
- Phonic Corporation
- Pioneer Corporation
- Presonus
- Rane Corporation
- Roland Corporation
- Samick
- Samson Technologies
- Shure

- Solid State Logic
- Soundcraft
- SoundTracs
- Speck Electronics
- Studer
- Studiomaster
- Tapco
- Tascam
- Toft Audio Designs
- Ward-Beck Systems
- Yamaha Pro Audio

Chapter 9

Wave Field Synthesis



WFS Principle

Wave field synthesis (WFS) is a spatial audio rendering technique, characterized by creation of virtual acoustic environments. It produces "artificial" wave fronts synthesized by a large number of individually driven speakers. Such wave fronts seem to originate from a virtual starting point, the *virtual source* or *notional source*. Contrary to traditional spatialization techniques such as stereo, the localization of virtual sources in WFS does not depend on or change with the listener's position.

Physical fundamentals

WFS is based on Huygens' Principle, which states that any wave front can be regarded as a superposition of elementary spherical waves. Therefore, any wave front can be synthesized from such elementary waves. In practice, a computer controls a large array of individual loudspeakers and actuates each one at exactly the time when the desired virtual wave front would pass through it.

The basic procedure was developed in 1988 by Professor Berkhout at the Delft University of Technology. Its mathematical basis is the Kirchhoff-Helmholtz integral. It states that the sound pressure is completely determined within a volume free of sources, if sound pressure and velocity are determined in all points on its surface.

$$\boldsymbol{P}(w,z) = \iint_{dA} \left(G(w,z|z') \frac{\partial}{\partial n} P(w,z') - P(w,z') \frac{\partial}{\partial n} G(w,z|z') \right) dz'$$

Therefore any sound field can be reconstructed, if sound pressure and acoustic velocity are restored on all points of the surface of its volume. This approach is the underlying principle of **Holophony**.

For reproduction, the entire surface of the volume would have to be covered with closely spaced monopole and dipole loudspeakers, each individually driven with its own signal. Moreover, the listening area would have to be anechoic, in order to comply with the source-free volume assumption. In practice, this is hardly feasible.

According to Rayleigh II the sound pressure is determined in each point of a half-space, if the sound pressure in each point of its dividing plane is known. Because our acoustic perception is most exact in the horizontal plane, practical approaches generally reduce the problem to a horizontal loudspeaker line, circle or rectangle around the listener.

The origin of the synthesized wave front can be in any point on the horizontal plane of the loudspeakers. It represents the virtual acoustic source, which hardly differs from a material acoustic source at the same position. Unlike conventional (stereo) reproduction, the virtual sources do not move along if the listener moves in the room. For sources behind the loudspeakers, the array will produce convex wave fronts. Sources in front of the speakers can be rendered by concave wave fronts that focus in the virtual source and diverge again. Hence the reproduction inside the volume is incomplete - it breaks down if the listener sits between speakers and inner source.

Procedural advantages

By means of level and time information stored in the impulse response of the recording room or derived from a model-based mirror-source approach, a sound field with very stable position of the acoustic sources can be established by wave field synthesis. In principle, it would be possible to establish a virtual copy of a genuine sound field indistinguishable from the real sound. Changes of the listener position in the rendition area would produce the same impression as an appropriate change of location in the recording room. Listeners are no longer relegated to a "sweet spot" area within the room.

The Moving Picture Expert Group standardized the object-oriented transmission standard MPEG-4 which allows a separate transmission of content (dry recorded audio signal) and form (the impulse response or the acoustic model). Each virtual acoustic source needs its own (mono) audio channel. The spatial sound field in the recording room consists of the direct wave of the acoustic source and a spatially distributed pattern of mirror acoustic sources caused by the reflections by the recording room surfaces. Reducing that spatial mirror source distribution onto a few transmitting channels causes a significant loss of spatial information. Much more accurately this spatial distribution can be synthesized by the rendition side.

Concerning the conventional channel-orientated rendition procedures, WFS provides a clear advantage: "Virtual panning spots" called virtual acoustic sources guided by the signal content of the associated channels can be positioned far beyond the material rendition area. That reduces the influence of the listener position because the relative changes in angles and levels are clearly smaller as with closely fixed material loudspeaker boxes. This extends the sweet spot considerably; it can now nearly cover the entire rendition area. The procedure of the wave field synthesis thus is not only compatible, it clearly improves the reproduction for the conventional transmission methods.

Remaining problems

The most perceptible difference concerning the original sound field is the reduction of the sound field to two dimensions along the horizontal of the loudspeaker lines. This is particularly noticeable for reproduction of ambiance as acoustic damping is required in the rendition area for accurate synthesis. The damping, however, does not compliment natural acoustic sources.

Sensitivity to room acoustics

Since WFS attempts to simulate the acoustic characteristics of the recording space, the acoustics of the rendition area must be suppressed. One possible solution is to arrange the walls in an absorbing and non-reflective way. The second possibility is playback within the near field. For this to work effectively the loudspeakers must couple very closely at the hearing zone or the diaphragm surface must be very large.

High cost

A further problem is high cost. A large number of individual transducers must be very close together. Otherwise spatial Aliasing effects becomes audible. This is a result of having a finite number of transducers (and hence elementary waves).

Aliasing

There are undesirable spacial distortions caused by position-dependent narrow-band break-downs in the frequency response within the rendition range – in a word, aliasing. Their frequency depends on the angle of the virtual acoustic source and on the angle of the listener to the loudspeaker arrangement:

$$f_{\text{alias}} = \frac{c}{\Delta x \left|\sin \Theta^{\text{sec}} - \sin \Theta^{\text{v}}\right|}$$

For aliasing free rendition in the entire audio range thereafter a distance of the single emitters below 2 cm would be necessary. But fortunately our ear is not particularly sensitive to spacial aliasing. A 10-15 cm emitter distance is generally sufficient. On the other hand the size of the emitter field does limit the representation range; outside of its borders no virtual acoustic sources be produced.

Truncation effect

Another cause for disturbance of the spherical wavefront is the "Truncation Effect". Because the resulting wavefront is a composite of elementary waves, a sudden change of pressure can occur if no further speakers deliver elementary waves where the speaker row ends. This causes a 'shadow-wave' effect. For virtual acoustic sources placed in front of the loudspeaker arrangement this pressure change hurries ahead of the actual wave front whereby it becomes clearly audible.

In signal processing terms, this is spectral leakage in the spatial domain and is caused by application of a rectangular function as a window function on what would otherwise be an infinite array of speakers.

Research and market maturity

Early development of WFS was started in from 1988 by the Delft University. Further work was carried out in the context of the CARROUSO project, promoted by the European Union (January 2001 to June 2003). In Europe, ten institutes were included in this research. The WFS sound system IOSONO was developed by the Fraunhofer Institute for digital media technology (IDMT) by the technical University of Ilmenau.

Loudspeaker arrays implementing WFS have been installed in some cinemas and theatres and in public range with good success. The first live WFS transmission was on July 2008 from the Cologne cathedral lecture hall 104 by the Technical University of Berlin. The room contains the world’s largest speaker system with 2700 loudspeakers on 832 independent channels.

Development of home-audio application of WFS has only recently begun. In spite of the efforts, large acceptance problems remain.

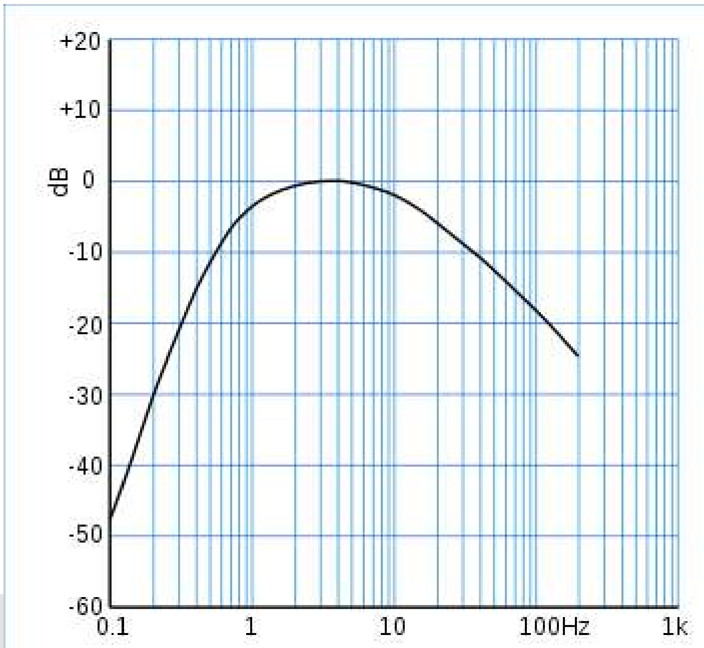
Chapter 10

Wow and Flutter Measurement

Wow and flutter measurement is carried out on audio tape machines, cassette recorders and players, and other analog recording and reproduction devices with rotary components (e.g. movie projectors, turntables (vinyl recording), etc.) This measurement quantifies the amount of 'frequency wobble' (caused by speed fluctuations) present in subjectively valid terms. Turntables tend to suffer mainly slow Wow. In digital systems, which are locked to crystal oscillators, wow and flutter are usually insignificant (variations in clock timing, referred to as jitter are a different issue, occurring over a much shorter timescale with different audible effect, and do not qualify as wow and flutter).

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

Measurement techniques



IEC 386 (Blk) Flutter weighting

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

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

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

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

Audible effects

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

Equipment performance

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

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

Absolute speed

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

Flutter correction

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

Scrape flutter

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

Chapter 11

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.

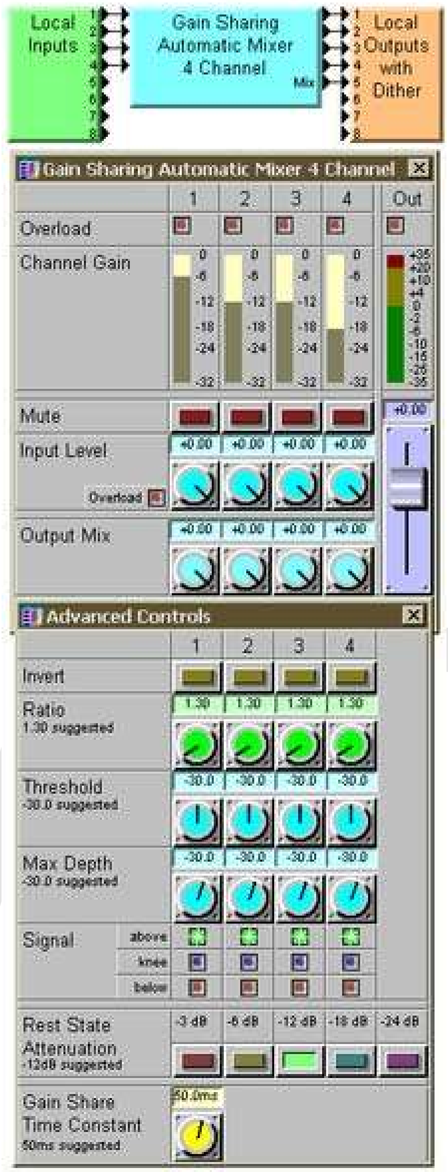
In 1971, Gregory Maston of Bell Labs filed for a patent involving a circuit that could switch between several audio sources based on their levels. The loudest one was latched into the mix. This system did not ramp switched signals smoothly in and out and did not maintain a constant ambience. It was intended 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 12

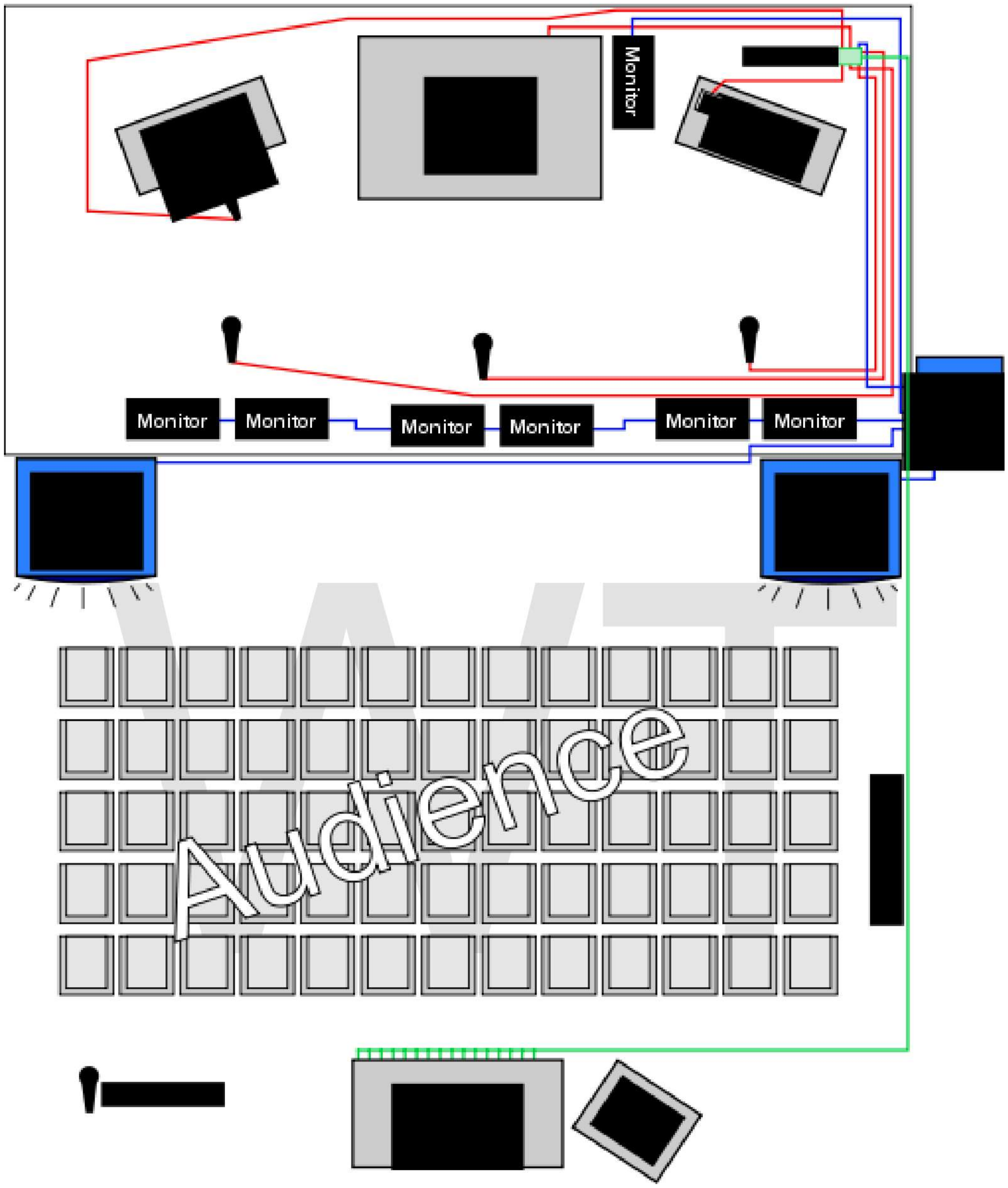
Sound Reinforcement System

Large outdoor concerts use complex and powerful sound reinforcement systems.

A **sound reinforcement system** is the combination of microphones, signal processors, amplifiers, and loudspeakers that makes live or pre-recorded sounds louder and may also distribute those sounds to a larger or more distant audience. In some situations, a sound reinforcement system is also used to enhance the sound of the sources on the stage, as opposed to simply amplifying the sources unaltered. A sound reinforcement system may be very complex, including hundreds of microphones, complex mixing and signal processing systems, tens of thousands of watts of amplification, and multiple loudspeaker arrays, all overseen by a team of audio engineers and technicians. On the other hand, a sound reinforcement system can be as simple as a small PA system in a coffeehouse, consisting of a single microphone connected to a self-powered 100-watt loudspeaker system. In both cases, these systems *reinforce* sound to make it louder or distribute it to a wider audience.

Some audio engineers and other sound industry professionals disagree over whether these audio systems should be called sound reinforcement (SR) systems or public address (PA) systems. Distinguishing between the two terms by technology and capability is common, while others distinguish by intended use (e.g., SR systems are for live music and PA systems are for reproduction of speech and recorded music in buildings and institutions). In some regions or markets, the distinction between the two terms is important, though the terms are considered interchangeable in many professional circles.

Basic concept



Basic Sound Reinforcement System

A typical sound reinforcement system consists of; **input transducers** (e.g., microphones), which convert sound energy into an electric signal, **signal processors** which alter the signal characteristics, **amplifiers,** which add power to the signal without otherwise changing its content, and **output transducers** (e.g., loudspeakers) , which convert the signal back into sound energy. These primary parts involve varying amounts of individual components to achieve the desired goal of reinforcing and clarifying the sound to the audience, performers, or other individuals.

Signal path

Sound reinforcement in a large format system typically involves a signal path that starts with an instrument pickup or a microphone (transducer) which is plugged into a multicore cable (often called a "snake"). The snake then routes the signals of all of the inputs to the Front of the House (FOH) and Monitor consoles. Once the signal is at a channel on the console, this signal can be equalized, compressed, or panned before being routed to an output bus. The signal may also be routed into an external effects processor, which outputs a *wet* (effected) version of the signal, which is typically mixed in varying amounts with the *dry* (ineffected) signal.

The signal is then routed to a bus, also known as a mix group, subgroup or simply 'group'. A group of signals may be routed through an additional bus before being sent to the main bus to allow the engineer to control the levels of several related signals at once. For example, all of the different microphones for a drum set might be sent to their own bus so that the volume of the entire drum set sound can be controlled with a single fader or a pair of faders. A bus can often be processed just like an individual input channel, allowing the engineer to process a whole group of signals at once. The signal is then typically routed with everything else to the stereo masters on a console. Mixing consoles also have additional sends, also referred to as *auxes*, on each input channel so that a different mix can be created and sent elsewhere.

The next step in the signal path generally depends on the size of the system in place. In smaller systems, the main outputs are often sent to an additional equalizer, or directly to a power amplifier, with one or more loudspeakers (typically two) then connected to that amplifier. In large-format systems, the signal is typically first routed through an equalizer then to a crossover. A crossover splits the signal into multiple frequency bands with each band being sent to separate amplifiers and speaker enclosures for low, middle, and high-frequency signals. Low-frequency sounds are sent to subwoofers, and middle and high-frequency sounds are typically sent to full-range speaker cabinets.

System components

Input transducers

Many types of input transducers can be found in a sound reinforcement system, with microphones being the most commonly used input device. They can be classified according to their method of transduction, pickup (or polar) pattern or their functional application. Most microphones used in sound reinforcement are either dynamic or condenser microphones.

Microphones used for sound reinforcement are positioned and mounted in many ways, including base-weighted upright stands, podium mounts, tie-clips, instrument mounts, and headset mounts. Headset mounted and tie-clip mounted microphones are often used with wireless transmission to allow performers or speakers to move freely. Early adopters of headset mounted microphones technology included country singer Garth Brooks, Kate Bush, and Madonna.

There are many other types of input transducers which may be used occasionally, including magnetic pickups used in electric guitars and electric basses, contact microphones used on stringed instruments, and piano and phonograph pickups (cartridges) used in record players.

In recent years, wireless technology has become increasingly popular in sound reinforcement, typically used for electric guitar, bass, and handheld microphones. This has granted performers the freedom to move about the stage during the show without the worry of tripping over or disconnecting the cable.

A Yamaha PM4000 and a Midas Heritage 3000 mixing console at the Front of House position at an outdoor concert.

Mixing consoles

Mixing consoles are the heart of a sound reinforcement system. This is where the operator can mix, equalize and add effects to sound sources. Multiple consoles can be used for different applications in a single sound reinforcement system. The Front of House (FOH) mixing console must be located where the operator can see the action on stage and hear the output of the loudspeaker system. Some venues with permanently installed systems such as religious facilities and theaters place the mixing console within an enclosed booth but this approach is more common for broadcast and recording applications. This is far less common in live sound reproduction since the engineer performs best when they can hear what the audience hears.

Large music productions often use a separate stage monitor mixing console which is dedicated to creating mixes for the performers' on-stage or in-ear monitors. These consoles are typically placed at the side of the stage so that the operator can communicate with the performers on stage. In cases where performers have to play at a venue that does not have a monitor engineer near the stage, the monitor mixing is done by the FOH engineer from the FOH console, which is located amongst the audience or at the back of the hall. This arrangement can be problematic because the performers end up having to request changes to the monitor mixes with "...hand signals and clever cryptic phrases". The engineer also cannot hear the changes that he is applying to the monitors on stage, often resulting in a reduction of the quality of the mix.

Signal processors

Digital signal processors

Small PA systems for venues such as bars and clubs are now available with features that were formerly only available on professional-level equipment, such as digital reverb effects, graphic equalizers, and, in some models, feedback prevention circuits which electronically sense and prevent feedback "howls" before they become a problem. Digital effects units may offer multiple pre-set and variable reverb, echo and related effects. Digital loudspeaker management systems offer sound engineers digital delay, limiting, crossover functions, EQ filters, compression and other functions in a single rack-mountable unit. In previous decades, sound engineers typically had to transport a substantial number of rack-mounted analog devices to accomplish these tasks.

Equalizers

Equalizers exist in sound reinforcement systems in two forms: graphic and parametric. Both of these are used in conjunction with high-pass and low-pass filters. Parametric equalizers are often built into each channel in mixing consoles and are also available as separate units. Parametric equalizers first became popular in the 1970s and have remained the program equalizer of choice for many engineers since then.

Graphic equalizers have faders which resemble a frequency response curve plotted on a graph. Sound reinforcement systems typically use graphic equalizers with one-third octave frequency centers. These are typically used to equalize output signals going to the main loudspeaker system or the monitors on stage.

High-pass (low-cut) and low-pass (high-cut) filters restrict a given channel's bandwidth extremes which can prevent subsonic disturbances and RF or lighting control disturbances from interfering with the audio system. Filter sections are often included with graphic and parametric equalizers to give full control of the frequency range. If their response is steep enough, high-pass filters and low-pass filters function as end-cut filters. A feedback suppressor is a specialized type of band-pass filter which automatically detects and suppresses feedback by greatly attenuating the frequency which is feeding back.

Compressors

Compressors are designed to manage the dynamic range of an audio signal. A compressor accomplishes this by reducing the gain of a signal that is above a defined level (threshold) by a defined amount (ratio). Without this gain reduction, a signal that gets, say 10% louder as an input, will be 10% louder at the output. With the gain reduced, a signal that gets 10% louder at the input will be perhaps 3% louder at the output. Most compressors available are designed to allow the operator to select a ratio within a range typically between 1:1 and 20:1, with some allowing settings of up to ∞:1. A compressor with an infinite ratio is typically referred to as a *limiter*. The speed that the compressor adjusts the gain of the signal (called the attack) is typically adjustable as is the final output of the device.

Compressor applications vary widely from objective system design criterion to subjective applications determined by variances in program material and engineer preference. Some system design criteria specify limiters for component protection and gain structure control. Artistic signal manipulation is a subjective technique widely utilized by mix engineers to improve clarity or to creatively alter the signal in relation to the program material. An example of artistic compression is the typical heavy compression used on the various components of a modern rock drum kit. The drums are processed to be perceived as sounding more punchy and full.

Processing racks at the FOH position at an outdoor concert.

Noise gates

A noise gate sets a threshold where if it is quieter it will not let the signal pass and if it is louder it opens the gate. A noise gate's function is in a sense the opposite to that of a compressor. Noise gates are useful for microphones which will pick up noise which is not relevant to the program, such as the hum of a miked electric guitar amplifier or the rustling of papers on a minister's podium.

Noise gates are also used to process the microphones placed near the drums of a drum kit in many hard rock and metal bands. Without a noise gate, the microphone for a specific instrument such as the floor tom will also pick up signal from nearby drums or cymbals. With a noise gate, the threshold of sensitivity for each microphone on the drum kit can be set so that only the direct strike and subsequent decay of the drum will be heard, not the nearby sounds.

Effects

Reverberation and delay effects are widely used in sound reinforcement systems to enhance the mix relative to the desired artistic impact of the program material. Modulation effects such as flanger, phaser, and chorus are also applied to some instruments. An exciter "livens up" the sound of audio signals by applying dynamic equalization, phase manipulation and harmonic synthesis of typically high frequency signals.

The appropriate type, variation, and level of effects is quite subjective and is often collectively determined by a production's engineer, artist, or musical director. Reverb, for example, can give the effect of signal being present in anything from a small room to a massive stadium, or even in a space that doesn't exist in the physical world. The use of reverb often goes unnoticed by the audience, as it often sounds more natural than if the signal was left dry. The use of effects in the reproduction of modern music is often in an attempt to mimic the sound of the studio version of the artist's music.

Power amplifiers

Power amplifiers boost a signal level and provide current to drive a loudspeaker. All output transducers require amplification of the signal by an amplifier, including loudspeakers, monitor speakers, and headphones. Most professional amplifiers provide protection from overdriven signals, short circuits across the output, and thermal overload. Limiters are often used to protect loudspeakers and amplifiers from signal overload.

Like most sound reinforcement equipment products, professional amplifiers are designed to be mounted within 19-inch racks. Many power amplifiers feature internal fans to draw air across their heat sinks. Since they can generate a significant amount of heat, thermal dissipation is an important factor for operators to consider when mounting amplifiers into equipment racks. Active loudspeakers feature internally mounted amplifiers that have been selected by the manufacturer to be the best amplifier for use with the given loudspeaker.

Power amplifiers have become lighter, smaller, more powerful and more efficient due to increasing use of switching power supplies and Class D amplifiers, which offer significant weight and space savings as well as increased efficiency. In the 1970s and 1980s, most PA amplifiers were heavy Class AB amplifiers. In the late 1990's these lightweight technologies spread into PA applications. Installations in railroad stations, stadia and airports, their high efficiency allow them to run with minimal additional cooling and with higher rack densities compared to older amplifiers. The use of class A amplifiers is typically limited to studio applications, where the highly detailed clarity of an amplifier is going to be much more noticeable and the power requirements are much less.

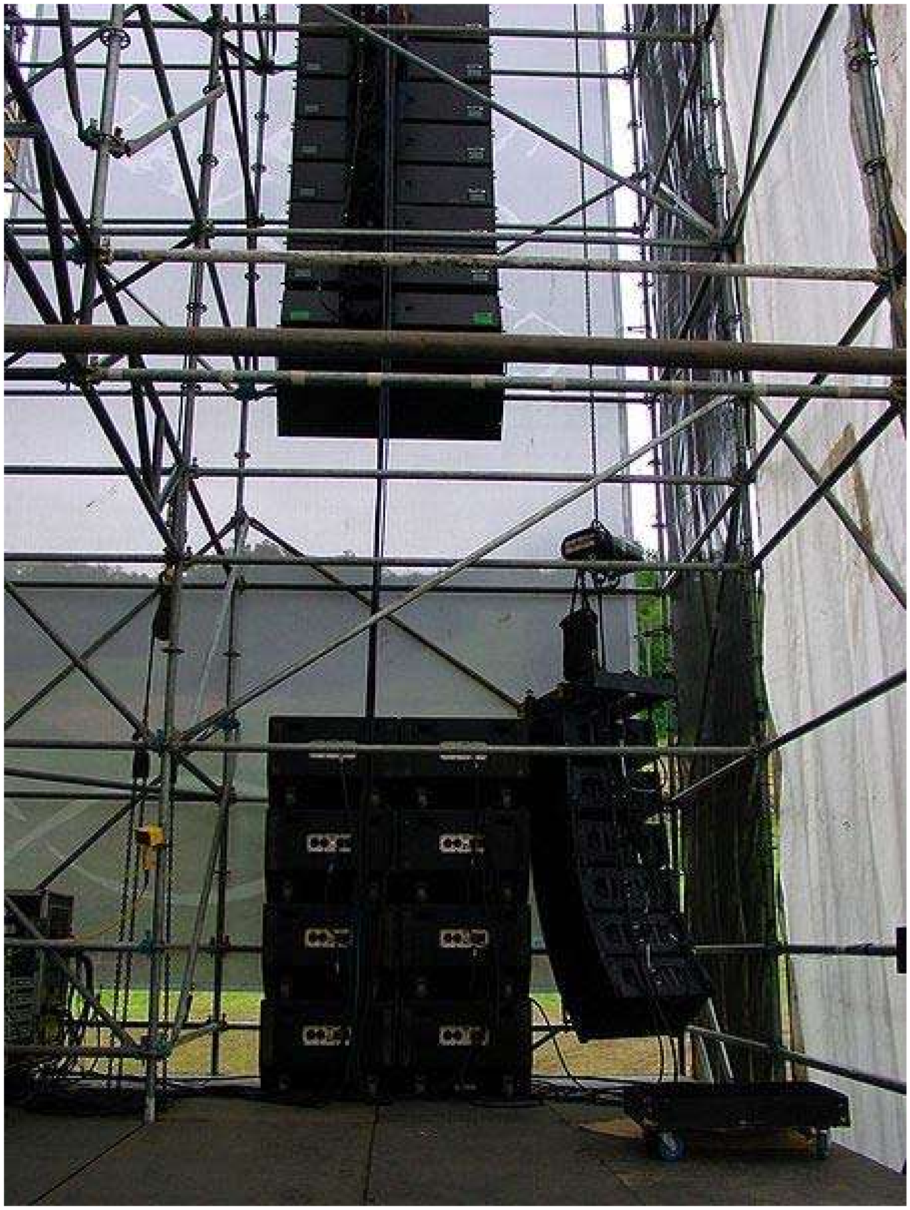
Digital loudspeaker management systems (DLMS) that combine digital crossover functions, compression, limiting, and other features in a single unit have become popular since their introduction. They are used to process the mix from the mixing console and route it to the various amplifiers in use. Systems may include several loudspeakers, each with its own output optimized for a specific range of frequencies (i.e. bass, midrange and treble). Bi, tri, or quad amplifying a sound reinforcement system with the aide of a DLMS results in a more efficient use of amplifier power by sending each amplifier only the frequencies appropriate for its respective loudspeaker. Most DLMS units that are designed for use by non-professionals have calibration and testing functions such as a pink noise generator coupled with a real-time analyzer to allow automated room equalization.

The amount of amplifier power used in a performance setting depends on a number of factors, such as the desired Sound Pressure Level of the performers, whether the venue is indoors or outdoors, and the presence of competing background noise. The following list gives a rough "rule of thumb" for the amount of amplifier power used in different settings:

- "Small Vocal" system - About 500 watts
- "Large Vocal" system - About 1,000 watts
- "Small Club" system - About 9,000 watts
- "Large Club" system - About 18,000 watts
- "Small Stadium" system - About 28,000 watts

Output transducers

Main loudspeakers



A large line array with separate subs and a smaller side fill line array.

A simple and inexpensive PA loudspeaker may have a single full-range loudspeaker driver, housed in a suitable enclosure. More elaborate, professional-caliber sound reinforcement loudspeakers may incorporate separate drivers to produce low, middle, and high frequency sounds. A crossover network routes the different frequencies to the appropriate drivers. In the 1960s, horn loaded theater loudspeakers and PA speakers were almost always "columns" of multiple drivers mounted in a vertical line within a tall enclosure. The 1970s to early 1980s was a period of innovation in loudspeaker design with many sound reinforcement companies designing their own speakers. The basic designs were based on commonly known designs and the speaker components were commercial speakers made by JBL, Altec Lansing, Electro-Voice, and others. The areas of innovation were in cabinet design, durability, ease of packing and transport, and ease of setup. This period also saw the introduction of the hanging or "flying" of main loudspeakers at large concerts. During the 1980s the large speaker manufactures started producing standard products using the innovations of the 1970s. These were mostly smaller two way systems with 12", 15" or double 15" woofers and a high frequency driver attached to a high frequency horn. The 1980s also saw the start of loudspeaker companies focused on the sound reinforcement market such as Meyer Sound and Eastern Acoustic Works. The 1990s saw the introduction of Line arrays where long vertical arrays of loudspeakers with a smaller cabinet are used to increase efficiency and provide even dispersion and frequency coverage. This period also saw the introduction of inexpensive molded plastic speaker enclosures mounted on tripod stands. Many feature built-in power amplifiers which made them practical for non-professionals to set up and operate successfully. The sound quality available from these simple 'powered speakers' varies widely depending on the implementation.

Many sound reinforcement loudspeaker systems incorporate protection circuitry, preventing damage from excessive power or operator error. Positive temperature coefficient resistors, specialized current-limiting light bulbs, and circuit-breakers were used alone or in combination to reduce driver failures. During the same period, the professional sound reinforcement industry made the Neutrik Speakon NL4 and NL8 connectors the standard input connectors, replacing 1/4" jacks, XLR connectors, and Cannon multipin connectors which are all limited to a maximum of 15 amps of current. XLR connectors are still the standard input connector on active loudspeaker cabinets.

The three different types of transducers are subwoofers, compression drivers, and tweeters. They all feature the combination of a voicecoil, magnet, cone or diaphragm, and a frame or structure. Loudspeakers have a power rating (in watts) which indicates their maximum power capacity, to help users avoid overpowering them. Thanks to the efforts of the Audio Engineering Society (AES) and the loudspeaker industry group ALMA, power-handling specifications became more trustworthy, although adoption of the EIA-426-B standard is far from universal. Around the mid 1990s trapezoidal-shaped enclosures became popular as this shape allowed many of them to be easily arrayed together.

A number of companies are now making lightweight, portable speaker systems for small venues that route the low-frequency parts of the music (electric bass, bass drum, etc.) to a powered subwoofer. Routing the low-frequency energy to a separate amplifier and subwoofer can substantially improve the bass-response of the system. Also, clarity may be enhanced, because low-frequency sounds take a great deal of power to amplify; with only a single amplifier for the entire sound spectrum, the power-hungry low-frequency sounds can take a disproportionate amount of the sound system's power.

Professional sound reinforcement speaker systems often include dedicated hardware for "flying" them above the stage area, to provide more even sound coverage and to maximize sight lines within performance venues.

The number of speaker enclosures used in a performance varies a great deal, but the following list gives a rough idea of how many cabinets are used in a typical venue:

- "Small Vocal" system - Two full range speakers mounted on tripod stands.
- "Large Vocal" system - Four full-range speakers for wide-area coverage.
- "Small Club" system - Two subwoofers and two mid/high speakers.
- "Large Club" system - Four subwoofers and four mid/high speakers.
- "Small Stadium" system - Four subwoofers, four mid-bass speakers, and four mid/high speakers.

Monitor loudspeakers

Monitor loudspeakers, also called 'foldback' loudspeakers, are speaker cabinets which are used onstage to help performers to hear their singing or playing. As such, monitor speakers are pointed towards a performer or a section of the stage. They are generally sent a different mix of vocals or instruments than the mix that is sent to the main loudspeaker system. Monitor loudspeaker cabinets are often a wedge shape, directing their output upwards towards the performer when set on the floor of the stage. Two-way, dual driver designs are common as monitor loudspeakers need to be smaller to save space on the stage. These loudspeakers typically require less power and volume than the main loudspeaker system, as they only need to provide sound for a few people who are in relatively close proximity to the loudspeaker. Some manufacturers have designed loudspeakers for use either as a component of a small PA system or as a monitor loudspeaker.

Using monitor speakers instead of in ear monitors typically results in an increase of stage volume, which can lead to more feedback issues and progressive hearing damage for the performers in front of them. The clarity of the mix for the performer on stage is also typically not as clear as they hear more extraneous noise from around them. The use of monitor loudspeakers, active or passive, requires more cabling and gear on stage, resulting in an even more cluttered stage. These factors, amongst others, have led to the increasing popularity of in- ear monitors.

In-Ear Monitors

A pair of universal fit in-ear monitors. This particular model is the Etymotic ER-4S

In-ear monitors are headphones that have been designed for use as monitors by a live performer. They are either of a "universal fit" or "custom fit" design. The universal fit in ear monitors feature rubber or foam tips that can be inserted into virtually anybody's ear. Custom fit in ear monitors are created from an impression of the users ear that has been made by an audiologist. In-ear monitors are almost always used in conjunction with a wireless transmitting system, allowing the performer to freely move about the stage whilst maintaining their monitor mix.

In-ear monitors offer considerable isolation for the performer using them, meaning that the monitor engineer can craft a much more accurate and clear mix for the performer. A downside of this isolation is that the performer cannot hear the crowd or other performers on stage that do not have microphones. This has been remedied by larger productions by setting up a pair of microphones on each side of the stage facing the audience that are mixed into the in-ear monitor sends.

Since their introduction in the mid-1980s, in-ear monitors have grown to be the most popular monitoring choice for large touring acts. The reduction or elimination of loudspeakers other than instrument amplifiers on stage has allowed for cleaner and less problematic mixing situations for both the front of house and monitor engineers.

Feedback is easier to manage and there is less sound reflecting off the back wall of the stage out into the audience, which affects the clarity of the mix the front of house engineer is attempting to create.

Applications

Sound reinforcement systems are used in a broad range of different settings, each of which poses different challenges.

Rental systems

Audio visual (AV) rental systems have to be able to withstand heavy use, and even abuse from renters. For this reason, rental companies tend to own speaker cabinets which are heavily braced and protected with steel corners, and electronic equipment such as power amplifiers or effects are often mounted into protective road cases. As well, rental companies tend to select gear which has electronic protection features, such as speaker-protection circuitry and amplifier limiters.

As well, rental systems for non-professionals need to be easy to use and set up, and they must be easy to repair and maintain for the renting company. From this perspective, speaker cabinets need to have easy-to-access horns, speakers, and crossover circuitry, so that repairs or replacements can be made. Some rental companies often rent powered amplifier-mixers, mixers with onboard effects, and powered subwoofers for use by non-professionals, which are easier to set up and use.

Many touring acts and large venue corporate events will rent large sound reinforcement Systems that typically include an audio engineer on staff with the renting company. In the case of rental systems for tours, there are typically several Engineers and Technicians from the Rental company that tour with the act to set up and calibrate the equipment for use by the band's production crew. The individual that actually mixes the act is often selected and provided by the band, as they are someone who has become familiar with the various aspects of the show and have worked with the act to establish a general idea of how they want the show to sound. The mixing engineer for an act sometimes also happens to be on staff with the rental company selected to provide the gear for the tour.

Live music clubs

A front-of-house sound engineer with a Digidesign D-Show Profile live digital mixer and a computer monitor.

Setting up sound reinforcement for live music clubs often poses unique challenges, because there is such a large variety of venues which are used as clubs, ranging from former warehouses or music theaters to small restaurants or basement pubs with concrete walls. In some cases, clubs are housed in multi-story venues with balconies or in "L"-shaped rooms, which makes it hard to get a consistent sound for all audience members. The solution is to use fill-in speakers to obtain good coverage, using a delay to ensure that the audience does not hear the same sound at different times.

Another problem with designing sound systems for live music clubs is that the sound system may need to be used for both prerecorded music played by DJs and live music. If the sound system is optimized for prerecorded DJ music, then it will not provide the appropriate sound qualities (or mixing and monitoring equipment) needed for live music, and vice versa. Lastly, live music clubs can be a hostile environment for sound gear, in that the air may be hot, humid, and smoky; in some clubs, keeping racks of power amplifiers cool may be a challenge. Often an air conditioned room just for the amplifiers is utilised.

Church sound

Designing systems in churches and similar religious facilities often poses a challenge, because the speakers may have to be unobtrusive to blend in with antique woodwork and stonework. In some cases, audio designers have designed custom-painted speaker cabinets so that the speakers will blend in with the church architecture. Some church facilities, such as sanctuaries or chapels are long rooms with low ceilings, which means that additional fill-in speakers are needed throughout the room to give good coverage. An additional challenge with church SR systems is that, once installed, they are often operated by amateur volunteers from the congregation, which means that they must be easy to run and troubleshoot.

Some mixing consoles designed for houses of worship have automatic mixers, which turn down unused channels to reduce noise, and automatic feedback elimination circuits which detect and notch out frequencies that are feeding back. These features may also be available in multi-function consoles used in convention facilities and multi-purpose venues.

Touring systems

Touring sound systems have to be powerful and versatile enough to cover many different rooms, often being of many different sizes and shapes. They also need to use "field-replaceable" components such as speakers, horns, and fuses, which are easily accessible for repairs during a tour. Tour sound systems are often designed with substantial redundancy features, so that in the event of equipment failure or amplifier overheating, the system will continue to function. Touring systems for acts performing for crowds of a few thousand people and up are typically set up and operated by a team of technicians and engineers that travel with the talent to every show.

It is not uncommon for mainstream acts that are going to perform in mid to large venues during their tour to schedule one to two weeks of tech rehearsal with the entire concert system and production staff at hand. This allows the audio and lighting engineers to become familiar with the show and establish presets on their digital equipment for each part of the show, if needed. Many modern musical groups work with their Front of House and Monitor Mixing Engineers during this time to establish what their general idea is of how the show should sound, both for themselves on stage and for the audience. This often involves programming different effects and signal processing for use on specific songs in an attempt to make the songs sound somewhat similar to the studio versions. To manage a show with a lot of these types of changes, the mixing engineers for the show often choose to use a digital mixing console so that they can recall these many settings in between each song. This time is also used by the system technicians to get familiar with the specific combination of gear that is going to be used on the tour and how it acoustically responds during the show. These technicians remain busy during the show, making sure the SR system is operating properly and that the system is tuned correctly, as the acoustic response of a room will respond differently throughout the day depending on the temperature, humidity, and number of people in the room.

Weekend band PA systems are a niche market for touring SR gear. Weekend bands need systems that are small enough to fit into a minivan or a car trunk, and yet powerful enough to give adequate and even sound dispersion and vocal intelligibility in a noisy club or bar. As well, the systems need to be easy and quick to set up. Sound reinforcement companies have responded to this demand by offering equipment that fulfills multiple roles, such as "amp-mixers" (a mixer with an integrated power amplifier and effects) and powered subwoofers (a subwoofer with an integrated power amplifier and crossover). These products minimize the amount of wiring connections that bands have to make to set up the system. Some subwoofers have speaker mounts built into the top, so that they can double as a base for the stand-mounted full-range PA speaker cabinets.

Live theater

Sound for live theater, operatic theater, and other dramatic applications may pose problems similar to those of churches, in cases where a theater is an old heritage building where speakers and wiring may have to blend in with woodwork. The need for clear sight lines in some theaters may make the use of regular speaker cabinets unacceptable; instead, slim, low-profile speakers are often used instead.

In live theater and drama, performers move around onstage, which means that wireless microphones may have to be used. Wireless microphones need to be set up and maintained properly, to avoid interference and reception problems.

Some of the higher budget theater shows and musicals are mixed in surround sound live, often with the show's sound operator triggering sound effects that are being mixed with music and dialogue by the show's mixing engineer. These systems are usually much more extensive to design, typically involving a separate sets of speakers for different zones in the theater.

Classical music and opera

A subtle type of sound reinforcement called acoustic enhancement is used in some concert halls where classical music such as symphonies and opera is performed. Acoustic enhancement systems help give a more even sound in the hall and prevent "dead spots" in the audience seating area by "...augment[ing] a hall's intrinsic acoustic characteristics." The systems use "...an array of microphones connected to a computer [which is] connected to an array of loudspeakers." However, as concertgoers have become aware of the use of these systems, debates have arisen, because "...purists maintain that the natural acoustic sound of [Classical] voices [or] instruments in a given hall should not be altered."

Kai Harada's article *Opera's Dirty Little Secret* states that opera houses have begun using electronic acoustic enhancement systems "...to compensate for flaws in a venue's acoustical architecture." Despite the uproar that has arisen amongst operagoers, Harada points out that none of the opera houses using acoustic enhancement systems "...use traditional, Broadway-style sound reinforcement, in which most if not all singers are equipped with radio microphones mixed to a series of unsightly loudspeakers scattered throughout the theatre." Instead, most opera houses use the sound reinforcement system for acoustic enhancement, and for subtle boosting of offstage voices, onstage dialogue, and sound effects (e.g., church bells in Tosca or thunder in Wagnerian operas).

Acoustic enhancement systems include LARES (Lexicon Acoustic Reinforcement and Enhancement System) and SIAP, the System for Improved Acoustic Performance. These systems use microphones, computer processing "with delay, phase, and frequency-response changes", and then send the signal "... to a large number of loudspeakers placed in extremities of the performance venue." Another acoustic enhancement system, VRAS (Variable Room Acoustics System) uses "...different algorithms based on microphones placed around the room." The Deutsche Staatsoper in Berlin and the Hummingbird Centre in Toronto use a LARES system. The Ahmanson Theatre in Los Angeles, the Royal National Theatre in London, and the Vivian Beaumont Theatre in New York City use the SIAP system.

Lecture halls and conference rooms

Lecture halls and conference rooms pose the challenge of reproducing speech clearly to a large hall, which may have reflective, echo-producing surfaces. In some conferences, sound engineers have to provide microphones for a large number of people, in the case of a panel conference or debate. In some cases, automatic mixers are used to control the levels of the microphones.

Sports sound systems

Systems for outdoor sports facilities and ice rinks often have to deal with substantial echo, which can make speech unintelligible. Sports and recreational sound systems often face environmental challenges as well, such as the need for weather-proof outdoor speakers in outdoor stadiums and humidity- and splash-resistant speakers in swimming pools.

Setting up and testing

Large-scale sound reinforcement systems are designed, installed, and operated by audio engineers and audio technicians. During the design phase of a newly constructed venue, audio engineers work with architects and contractors, to ensure that the proposed design will accommodate the speakers and provide an appropriate space for sound technicians and the racks of audio equipment. Sound engineers will also provide advice on which audio components would best suit the space and its intended use, and on the correct placement and installation of these components. During the installation phase, sound engineers ensure that high-power electrical components are safely installed and connected and that ceiling or wall mounted speakers are properly mounted (or "flown") onto rigging. When the sound reinforcement components are installed, the sound engineers test and calibrate the system so that its sound production will be even across the frequency spectrum.

System testing

A sound reinforcement system should be able to accurately reproduce a signal from its input, through any processing, to its output without any coloration or distortion. However, due to inconsistencies in venue sizes, shapes, building materials, and even crowd densities, this is not always possible without prior calibration of the system. This can be done in one of several ways.

The oldest method of system calibration involves a set of healthy ears, test program material (i.e. music or speech), a graphic equalizer, and last but certainly not least, a familiarity with the proper (or desired) frequency response. One must then listen to the program material through the system, take note of any noticeable frequency changes or resonances, and subtly correct them using the equalizer. Experienced engineers typically use a specific playlist of music every time they calibrate a system that they have become very familiar with. This process is still done by many engineers, even when analysis equipment is used, as a final check of how the system sounds with music or speech playing through the system.

Another method of manual calibration requires a pair of high-quality headphones patched into the input signal *before* any processing (such as the pre-fade-listen of the test program input channel of the mixing console, or the headphone output of the CD player or tape deck). One can then use this direct signal as a near-perfect reference with which to find any differences in frequency response. This method may not be perfect, but it can be very helpful with limited resources or time, such as using pre-show music to correct for the changes in response caused by the arrival of a crowd.

Because this is still a very subjective method of calibration, and because the human ear is so dynamic in its own response, the program material used for testing should be as similar as possible to that for which the system is being used.

Since the development of digital signal processing (DSP), there have been many pieces of equipment and computer software designed to shift the bulk of the work of system calibration from human auditory interpretation to software algorithms that run on microprocessors.

One tool for calibrating a sound system using either DSP or Analog Signal Processing is a Real Time Analyzer (RTA). This tool is usually used by piping pink noise into the system and measuring the result with a special calibrated microphone connected to the RTA. Using this information, the system can be adjusted to help achieve the desired response. The displayed response from the RTA mic cannot be taken as a perfect representation of the room as the analysis will be different, sometimes drastically, when the mic is placed in different position in front of the system.

More recently, sound engineers have seen the introduction of dual fft (fast-fourier transform) based audio analysis software which allows an engineer to view not only frequency vs. amplitude (pitch vs. volume) information that an rta provides, but also to see the same signals (sounds) in the time domain. This provides the engineer with much more meaningful data than an rta alone. Also, dual fft analysis allows one to compare the source signal with the output signal and view the difference. This is a very fast way to calibrate a system to sound as close as possible to the original source material. As with any such measurement tool, it must always be verified using actual human ears.

Some DSP system processing devices have been designed for use by non-professionals that automatically make adjustments in the system EQ based upon what is being read from the RTA mic. These are practically never used by professionals, as they almost never calibrate the system as well as a professional audio engineer can manually.

Chapter 13

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.