Music Technology

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Introduction

Music technology is a term that refers to all forms of technology involved with the musical arts, particularly the use of electronic devices and computer software to facilitate playback, recording, composition, storage, and performance. This subject is taught in many educational levels including k-12 through college and university. Furthermore, music technology encompasses the technical and scientific aspects of the music such as acoustic science, programming, music psychology & sociology, and music industry business practices. In many contemporary experimental musical instruments new music technology inventions are being used to create new sound possibilities.

The concept of music technology is intimately connected to both musical and technological creativity. People are constantly striving to devise new forms of expression through music, and physically creating new devices to enable them to do so. Because of this, our definition of what music technology encompasses must continually expand. Although the term is now most commonly used in reference to modern electronic devices, such as a monome, the piano and guitar are also examples of music technology. In the computer age the ontological range of music technology has greatly increased. It may now be mechanical, electronic, software or indeed even purely conceptual.

Sequencer software is perhaps the most widely-used form of software music technology. Such programs allow the user to record audio or MIDI musical sequences, which then may be organized along a time line. Musical segments can be copied and duplicated ad infinitum, edited and processed using a variety of audio effects. Music Technology includes many forms of music reproduction. Music technology and sound technology both refer to the use of sound engineering in a commercial or leisurely manner. The two may sometimes be classed as the same but actually refer to different fields of work, the names of which are self explanatory but where sound engineering may refer primarily to the use of sound technology for media logical purposes.

Chapter 1

Multitrack Recording

The TASCAM 85 16B analog tape recorder can record 16 tracks of audio on 1-inch (2.54cm) tape. Professional analog units of 24 tracks on 2-inch tape were common, with specialty tape heads providing 16 or even 8 tracks on the same tape width, for greater fidelity.

The Digidesign 192 i/o. A digital audio interface for the Pro Tools computer-based hard disk recording system. Digital audio quality is measured in data resolution per channel.

Multitrack recording (also known as **multitracking** or just **tracking** for short) is a method of sound recording that allows for the separate recording of multiple sound sources to create a cohesive whole. Multitracking became possible with the idea of simultaneously recording different audio channels to separate discrete "tracks" on the same tape—a "track" was simply a different channel recorded to its own discrete area on tape whereby their relative sequence of recorded events would be preserved, and playback would be simultaneous or *synchronized.*

In the 1980s and 1990s, computers provided means by which both sound recording and reproduction could be digitized, revolutionizing audio distribution. In the 2000s, multitracking hardware and software for computers was of sufficient quality to be widely used for high-end audio recording. Though tape has not been universally replaced as a recording medium, the advantages of non-linear editing and recording have resulted in digital systems largely superseding tape.

Process

Multitracking can be achieved with analogue, tape-based equipment (from simple, cassette-based four or eight trackers to 2" reel-to-reel 24-track machines), digital equipment that relies on tape storage of recorded digital data (such as ADAT eight-track machines) and hard disk-based systems often employing a computer and audio recording software. Multitrack recording devices vary in their specifications, such as the number of simultaneous tracks available for recording at any one time; in the case of tape-based systems this is limited by, among other factors, the physical size of the tape employed. Some of the biggest professional analog recording studios used a computer to synchronize multiple 24-track machines, effectively multiplying the number of available tracks into the hundreds. The rock group Toto recorded their fourth album on four computer-synced 24-track machines, for example.

For computer-based systems the trend is towards unlimited numbers of record/playback tracks, although issues such as memory and CPU available will in fact limit this from machine to machine. Moreover, on computer-based systems, the number of simultaneously available recording tracks is limited by the sound card discrete analog or digital inputs.

When recording, audio engineers can select which track (or tracks) on the device will be used for each instrument, voice, or other input.

At any given point on the tape, any of the tracks on the recording device can be recording or playing back using sel-sync or Selective Synchronous recording. This allows an artist to be able to record onto track 2 and, simultaneously, listen to track 1, 3 and 7, allowing them to sing or to play an accompaniment to the performance already recorded on these tracks. They might then record an alternate version on track 4 while listening to the other tracks. All the tracks can then be played back in perfect synchrony, as if they had originally been played and recorded together. This can be repeated until all of the available tracks have been used, or in some cases, reused. During mix down a separate set of playback heads with higher fidelity are used.

At any given point in the recording process, any number of existing tracks can be "bounced" into one or two tracks and the original tracks erased, making more room for more tracks to be reused for fresh recording. Beatles producer George Martin used this technique extensively to achieve multiple track results, while still being limited to using only multiple four-track machines, until an eight-track machine became available during the recording of the Beatles' White Album. The Beach Boys' *Pet Sounds* also made innovative use of multitracking with 8-track machines of the day (1965–66).

The TEAC 2340, a popular early (1973) home multitrack recorder, four tracks on ¼ inch tape (1/16 inch/track headroom).

Korg D888 eight-track digital recorder

Multitrack recording also allows any recording artist to record multiple "takes" of any given section of their performance, allowing them to refine their performance to virtual perfection. A recording engineer can record only the section being worked on, without erasing any other section of that track. This process of turning the recording mechanism on and off is called "punching in" and "punching out".

When recording is completed, the many tracks are "mixed down" through a mixing console to a two-track stereo recorder in a format which can then be duplicated and distributed. (Movie and DVD soundtracks can be mixed down to four or more tracks, as needed, the most common being five tracks, with an additional subwoofer track, hence the "5.1" surround sound most commonly available on DVDs.)

Most of the records, CDs and cassettes commercially available in a music store are recordings that were originally recorded on multiple tracks, and then mixed down to stereo.

In some rare cases, as when an older song is technically "updated", these stereo (or mono) mixes can in turn be recorded (as if it were a "submix") onto two (or one) tracks of a multitrack recorder, allowing additional sound (tracks) to be layered on the remaining tracks.

Flexibility

During multitracking, multiple musical instruments (and vocals) can be recorded, either one at a time or simultaneously, onto individual tracks, so that the sounds thus recorded can be accessed, processed and manipulated individually to produce the desired results. For example, after recording some parts of a song, an artist might listen to *only* the guitar part, by 'muting' all the tracks except the one on which the guitar was recorded. If one then wanted to listen to the vocals in isolation, one would do so by muting all the tracks apart from the vocals track. If one wanted to listen to the entire song, one could do so by un-muting all the tracks. If one did not like the guitar part, or found a mistake in it, and wanted to replace it, one could do so by re-recording *only* the guitar part (i.e., re-recording only the track on which the guitar was recorded), rather than re-recording the entire song.

If all the voices and instruments in a recording are individually recorded on distinct tracks, then the artist is able to retain complete control over the final sculpting of the song, during the mix-down (re-recording to two stereo tracks for mass distribution) phase.

For example, if an artist wanted to apply one effect to a synthesizer part, a different effect to a guitar part, a 'chorused reverb' effect to the lead vocals, and different effects to all the drums and percussion instruments, they could not do so if they had all been originally recorded together onto the same track. However, if they had been recorded onto separate tracks, then the artist could blend and alter all of the instrument's sounds with complete freedom.

Multitracking a song also leaves open the possibilities of remixes by the same or future artists, such as DJs. If the song was not available in a multitrack format recording, the job of the remixing artist could be very difficult, or impossible, because once the tracks have been re-recorded together during the mixdown phase, they are inseparable. Theoretically, one could use frequency selective filters for this, but in reality this has not been done with any great degree of success because of the multi-harmonic (having many frequencies) nature of many musical instruments and voices.

History

The process was conceived and developed by Ross Snyder at Ampex in 1955 resulting in the first 8-track machine which used 1-inch tape. This 8-track recorder was sold to Les Paul for $10,000. It became known as the "Octopus". Les Paul, Mary Ford and Patti Page used the technology in the late 1950s to enhance vocals and instruments. From these beginnings, it evolved in subsequent decades into a mainstream recording technique.

With computer

In the 2000s, many performers have recorded albums using only a personal computer as a tracking machine. To use a personal computer as a multitracking device, the computer must have an analog to digital interface, and multitrack recording software must be installed (software is available at all price ranges or even free, in the case of free software or open source). As well, a microphone is needed to record the vocals of a singer or any other sources of sound or a line-level input to accept analog signals from other equipment.

This is all that is needed to use a computer as a digital multitrack. Alternately, a standard personal computer sound card can be used to capture sounds, albeit with less fidelity. This is done simply by attaching either a microphone to the microphone input jack if a vocal track is to be recorded, or a stereo cable from the electronic device (such as a synthesizer or a guitar amplifier) to the line input of the sound card. Computers with appropriate software and hardware can record multiple audio tracks at once. This audio interface hardware sends audio signals to the computer and may interface with the computer via a PCI card, USB or FireWire connections. There are a range of audio interface options available. Popular brands include Apogee Electronics, Digidesign, Echo Digital Audio, MOTU, RME, M-Audio and Presonus. The instruments and singers' voices are recorded as individual files on the computer's hard drive, and function as tracks as per traditional multitracking. Effects such as reverb, chorus, and delays can be applied by the computer software. When the musicians are happy with the sound, the multiple tracks are mixed down onto two clean tracks, again within the multitracking software. Finally, the final stereo recording can be burned to a CD, which can be copied and distributed.

Multitracking software for a personal computer includes: Adobe Audition, Pro Tools from AVID, Pyramix from Merging Technologies, Digital Performer from Mark of the Unicorn, SONAR from Cakewalk, Samplitude from Magix, Nuendo, Cubase from Steinberg, FL Studio from Image-Line, and Logic from Apple. Mixcraft from Acoustica, Inc., REAPER from Cockos and n-Track Studio from FASoft are affordable alternatives to high end multi-track software. Audacity and Ardour are popular open source programs for multi-track recording. Jokosher (open source as well) is quite new, but seems to be gaining popularity among Linux users.

2007 Song Galaxy has released an Audio Multi-Track format that is delivered in a single file, which loaded into the player software gives the user the ability to mute or adjust the volume level of individual instruments. Tracks can be exported as individual WAV files which can then be loaded into other Multitracking software for further editing.

Order of recording

In most modern popular songs, drums and percussion instruments are the first instruments to be recorded. There are various reasons for this. The drums are usually the rhythm leaders; it is much easier for musicians recording later tracks to keep to the common beat of the drums, also due to the precise attack of drum sounds. A drummer might find it very difficult to play along with a backing track recorded without percussion, due to the likely variations in the musicians' tempo. Furthermore, in order to accurately keep to a pre-established rhythm, a drummer would need the sound of the other instruments to be very loud to compete with their drum kit; apart from the possibility of the drum microphones picking up the sound of the other instruments from the drummer's headphones, prolonged exposure to such volume would damage their hearing. Also, it allows the drums to be recorded for a few seconds, then looped. Click (metronome) tracks are also often used as the first sound to be recorded, especially when the drummer isn't available for the initial recording, and/or the final mix will be synchronized with motion picture and/or video images. Another practical reason refers to the song key. While having the basic rhythmic track laid down, musicians can experiment with the song key (e.g. C or D). This turns useful at the writing period or when songs are meant to be performed by a not yet defined singer (*e.g.* music producer searching for a jingle singer).

Also, though the drums might eventually be mixed down to a couple of tracks, each individual drum and percussion instrument might be initially recorded to its own individual track. The drums and percussion combined can occupy the largest number of tracks utilized in a recording. This is done so that each percussion instrument can be processed individually for maximum effect. Equalization (or EQ) is often used on individual drums, to bring out each one's characteristic sound.

The last tracks to be recorded are usually the vocals (though a temporary vocal track might be recorded early on either as a reference or to guide subsequent musicians; this is sometimes called a "Guide Vocal", "Ghost Vocal" or "Scratch vocal"). One reason for this is that singers will often temper their vocal expression in accordance with the accompaniment.

Concert music

For classical and jazz recordings (particularly instrumentals) where multitracking is chosen as the recording method (as opposed to direct to stereo, for example), a different arrangement is used; all tracks are recorded simultaneously. Sound barriers are often placed between different groups within the orchestra, e.g. pianists, violinists, percussionists, etc. When barriers are used, these groups listen to each other via headphones.

Chapter 2

Guitar Amplifier

Mesa Boogie Mark IV, a guitar combo amplifier

A **guitar amplifier** (or **guitar amp**) is an electronic amplifier designed to make the signal of an electric or acoustic guitar louder so that it will produce sound through a loudspeaker. Guitar amplifiers also modify the instrument's tone by emphasizing or de-emphasizing certain frequencies and adding electronic effects.

Amplifiers consist of one or more circuit stages which have unique responsibilities in the modification of the input signal. The power amplifier or output stage produces a high current signal to drive a speaker to produce sound. One or more preamplifier stages precede the power amplifier stage. The preamplifier is a voltage amplifier that amplifies the guitar signal to a level that can drive the power stage. There may be one or more tone stages which affect the character of the guitar signal: before the preamp stage (as in the case of guitar pedals), in between the preamp and power stages (as in the cases of effects loop or many dedicated amplifier tone circuits), in between multiple stacked preamp stages, or in feedback loops from a post-preamp signal to an earlier pre-preamp signal (as in the case of presence modifier circuits). The tone stages may also have electronic effects such as equalization, compression, distortion, chorus, or reverb. Amplifiers may use vacuum tubes (in Britain they are called valves), or solid state (transistor) devices, or both.

There are two configurations of guitar amplifiers: combination ("combo") amplifiers, which include an amplifier and one, two, or four speakers in a wooden cabinet; and the standalone amplifier (often called a "head" or "amp head"), which does not include a speaker, but rather passes the signal to a speaker cabinet or "cab". Guitar amplifiers range in price and quality from small, low-powered practice amplifiers, designed for students, which sell for less than US$50, to expensive "boutique" amplifiers which are custom-made for professional musicians and can cost thousands of dollars.

History

The first electric instrument amplifiers were not designed for use with electric guitars. The earliest examples appeared in the early 1930s when the introduction of electrolytic capacitors and rectifier tubes allowed the production of economical built-in power supplies that could be plugged into wall sockets, instead of heavy multiple battery packs, since rechargeable batteries wouldn't be lightweight until later on. While guitar amplifiers from the beginning were used to amplify acoustic guitar, electronic amplification of guitar was first widely popularized by the 1930s and 1940s craze for Hawaiian music, which extensively employed the amplified lap steel Hawaiian guitar.

Tone controls on early guitar amplifiers were very simple and provided a great deal of treble boost, but the limited controls, the loudspeakers used, and the low power of the amplifiers (typically 15 watts or less prior to the mid-1950s) gave poor high treble and bass output. Some models also provided effects such as spring reverb and/or an electronic tremolo unit. Early Fender amps labeled tremolo as "vibrato" and labeled the vibrato arm of the Stratocaster guitar as a "tremolo bar".

In the 1960s, guitarists experimented with distortion produced by deliberately overdriving their amplifiers. The Kinks guitarist Dave Davies produced early distortion effects by connecting the already distorted output of one amplifier into the input of another. Later, most guitar amps were provided with preamplifier distortion controls, and "fuzz boxes" and other effects units were engineered to safely and reliably produce these sounds. In the 2000s overdrive and distortion has become an integral part of many styles of electric guitar playing, ranging from blues rock to heavy metal and hardcore punk.

Guitar amplifiers were at first used with bass guitars and electronic keyboards, but other instruments produce a wider frequency range and need a suitable amplifier and full-range speaker system. Much more amplifier power is required to reproduce low-frequency sound, especially at high volume. Reproducing low frequencies also requires a suitable woofer or subwoofer speaker and enclosure. Woofer enclosures need to be larger and more sturdily built than cabinets for mid-range or high-frequency (tweeter) speakers.

Types

Two combo amplifiers

Guitar amplifiers are manufactured in two main forms. The "combination" (or "combo") amplifier contains the amplifier head and guitar speakers in a single unit which is typically housed in a rectangular wooden box. The amplifier head or "amp head" contains the electronic circuitry constituting the preamp, built-in effects processing, and the power amplifier. Combo amps have at least one 1/4" input jack where the patch cord from the electric guitar can be plugged in. Other jacks may also be provided, such as an additional input jack, "send" and "return" jacks to create an effects loop (for connecting electronic effects such as compression, reverb, etc.), an extension speaker jack (for connecting an additional speaker cabinet). Some smaller practice amps have stereo RCA jacks for connecting a CD player, iPod or other sound source and a 1/4" headphone jack so that the player can practice without disturbing neighbours or family members.

Kustom 200 bass amp - amp head and speakers, 100 watts RMS, two channels, two 15" speakers, 1971

Some amplifiers have a line out jack for connecting the amplifier's signal to a PA system or recording console or to connect the amplifier to another guitar amp. In but most styles of rock and blues guitar, the line out is not used to connect the guitar amp to a PA system or recording console, because the tonal coloration and overdrive from the amplifier and speaker is considered an important part of the amplifier's sound. However, players do use the line out to connect one guitar amplifier to another amplifier, in order to create different tone colors or sound effects.

In the "amp head" form, the amplifier head is separate from the speakers, and joined to them by speaker cables. The separate amplifier is called an amplifier head, and is commonly placed on top of one or more loudspeaker enclosures. A separate amplifier head placed atop a guitar speaker enclosure or guitar speaker cabinet forms an amplifier "stack" or "amp stack". Amp heads may also have the different types of input and output jacks listed above in the combo section. In addition to a 1/4" input jack, acoustic guitar amplifiers typically have an additional input jack for a microphone, which is easily identified because it will use a three-pin XLR connector. Phantom power is not often provided on general-use amps, restricting the choice of microphones for use with these inputs. However, for high-end acoustic amplifiers, phantom power is often provided, so that musicians can use condenser microphones.

Amplifiers used with electric guitars may be solid state, which are lighter in weight and less expensive than tube amplifiers. Most guitarists, particularly in the genres of blues and rock, prefer the sound of vacuum tube amplifiers despite their higher cost, heavier weight, the need to periodically replace tubes and need to re-bias the output tubes (every year or two with moderate use). Some companies design amplifiers that require no biasing as long as properly rated tubes are used. Some modern amplifiers use a mixture of tube and solid-state technologies.

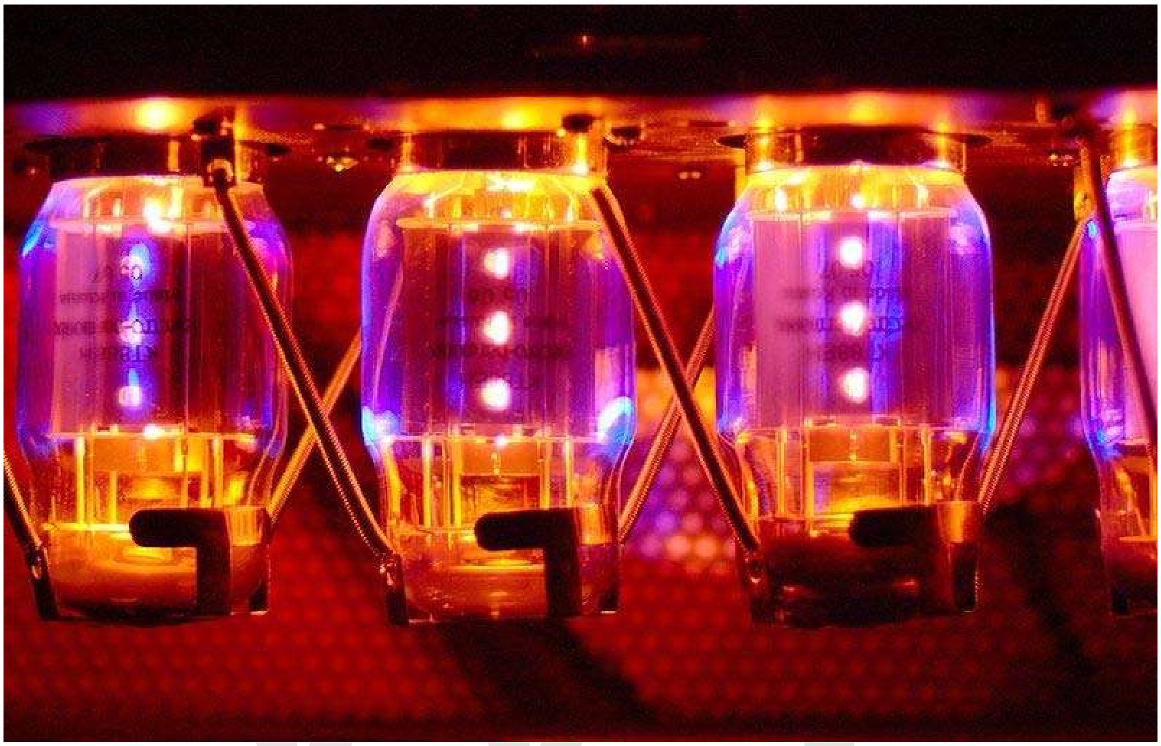
Since the advent of microprocessors and digital signal processing, "modeling amps" have been developed in the late 1990s, these can simulate the sounds of a variety of well-known tube amplifiers without needing to use vacuum tubes. Amplifiers with processors and software emulate the sound of a classic amp well, but from the player's point of view the response of these amplifiers may not feel the same as the digital modeling does not accurately model all aspects of a tube amplifier.

A wide range of instrument amplifiers is available, some for general purposes and others designed for specific instruments or particular sounds. These include:

- **"Traditional" guitar amplifiers**, with a clean, warm sound, a sharp treble roll-off at 5 kHz or less and bass roll-off at 60–100 Hz, and often built-in reverb and tremolo ("vibrato") units. These amplifiers, such as the Fender "Tweed"-style amps, are often used by traditional rock, blues, and country musicians.Traditional amps have more recently become popular with musicians in indie and alternative bands
- **Hard rock-style guitar amplifiers**, which often include preamplification controls, tone filters, and distortion effects that provide the amplifier's characteristic tone. Users of these amplifiers use the amplifier's tone to add "drive", intensity, and "edge" to their guitar sound. Amplifiers of this type, such as Marshall amplifiers, are used in a range of genres, including hard rock, metal, and punk.
- **Bass amplifiers**, with extended bass response and tone controls optimized for bass guitars (or more rarely, for upright bass). Higher-end bass amplifiers sometimes include compressor or limiter features, which help to keep the amplifier from distorting at high volume levels, and an XLR DI output for patching the bass signal directly into a mixing board. Bass amplifiers are often provided with external metal heat sinks or fans to help keep the amplifier cool.

- ***Acoustic* amplifiers**, similar in many ways to keyboard amplifiers but designed specifically to produce a "clean," transparent, "acoustic" sound when used with acoustic instruments with built-in transducer pickups and/or microphones.

Vacuum tube amplifiers



The glow from four "Electro Harmonix KT88" brand power tubes lights up the inside of a Traynor YBA-200 guitar amplifier

Vacuum tubes (valves) were by far the dominant active electronic components in most instrument amplifier applications until the 1970s, when semiconductors (transistors) started taking over for performance and economic reasons, including heat and weight reduction, and improved reliability. High-end tube instrument amplifiers have survived as one of few exceptions, because of the sound quality. Typically, one or more dual triodes are used in the preamplifier section in order to provide sufficient voltage gain to offset losses by tone controls and to drive the power amplifier section.

Rear view of a tube (valve) combo guitar amplifier. Visible are two glass output tubes, six smaller preamp tubes in their metal tube retainers, and both the power transformer and the output transformer.

The output tubes are often arranged in a class AB push-pull connection to improve efficiency; this requires another triode or dual triode to split the phase of the signal. The tubes of the power amplifier stage are almost always of the pentode or beam tetrode type (also known as "kinkless tetrodes", hence the KTxx nomenclature). Some high power models use paralleled pairs of output tubes (four or more in total) in push-pull. Except for the light negative feedback from the secondary end of the output transformer to the driver stage, most amplifying stages work in "raw" open-loop mode. Some designs employ current feedback via unbypassed cathode resistors.

Since most tubes show "soft clipping" gain non-linearity, applying an input signal high enough to overdrive any stage tends to produce favorably natural distortion. Today, most vacuum tube amplifiers are based on the ECC83/12AX7/7025 (dual triode) tubes for the preamplifier and driver sections and the EL84/6BQ5 or EL34/6CA7/KT77 or 6L6/KT66 or 6V6 tubes for the power output section. Some use the KT88/6550 beam power tubes in the output stage. The differing codes for equivalent tubes generally reflect those used by the original European or U.S.A. based manufacturers. These tubes are now mainly manufactured in Russia, China and Eastern European countries. Some amplifiers, such as the Marshall Silver Jubilee, use solid state components in the preamp, most commonly diodes, to create distortion, a design feature known as diode clipping.

Tube instrument amplifiers are often equipped with lower-grade transformers and simpler power regulation circuits than those of hi-fi amplifiers. They are usually not only for cost-saving reasons, but also are considered as a factor in sound creation. For example, a simple power regulation circuit's output tends to sag when there is a heavy load (that is, high output power) and vacuum tubes usually lose gain factors with lower power voltages. This results in a somewhat compressed sound which could be criticized as a "poor dynamic range" in case of hi-fi amplifiers, but could be desirable as "long sustain" of sounds on a guitar amplifier. Some tube guitar amplifiers use a rectifier tube instead of solid-state diodes specifically for this reason.

Unfortunately, most amplifiers offer a fixed amount of sag, and this fixed amount can only be attained at full volumes. A small minority of amplifiers offer sag control via either multiple rectifiers or the Sag Circuit (a non-traditional power supply design patented by Maven Peal® Instruments). Amplifiers with multiple rectifiers can offer up to two sag settings (amounts), while the Sag Circuit provides a Sag control knob, which allows range of sag control at all volumes (by interacting with a wattage control knob).

Some models have a "spring reverb" unit that simulates the reverberation of an echoic ambient. A reverb unit usually consists of one or more coil springs driven by the preamplifier section using a transducer driver similar to a loudspeaker at one end and an electro-magnetic pickup and preamplifier stage at the other end that picks up the long sustaining spring vibration, which is then mixed with the original signal. Some guitar amplifiers have a tremolo control. An internal oscillator generates a low frequency continuous signal which can modulate the input signal's amplitude or the output tubes' bias, thereby producing a tremolo effect.

Tube amps have the following technical disadvantages in comparison to solid-state amps. They are bulky and heavy, primarily due to the iron in power and output transformers. Solid-state amplifiers still require power transformers, but are usually direct-coupled and don't need output transformers. Glass tubes are fragile, and require more care and consideration when equipment is moved repeatedly. Tube performance can deteriorate slightly over time before eventual catastrophic failure.

When tube vacuum is maintained at a high level, though, excellent performance and life is possible. They are prone to pick up mechanical noises (microphonic noise), although such electro-mechanical feedback from the loudspeaker to the tubes in combo amplifiers may contribute to sound creation. Tubes benefit from a heater warm-up period before the application of high tension anode voltages; this allows the tube cathodes to operate without damage and so prolongs tube life. This is of particular importance for amplifiers with solid state rectifiers.

Tube amps have the following technical advantages over solid-state amps. Compared to semiconductors, tubes have a very low "drift" (of specs) over a wide range of operating conditions, specifically high heat/high power. Semiconductors are very heat-sensitive by comparison and this fact usually leads to compromises in solid-state amplifier designs. When a tube fails, it is replaceable. While solid state devices are also replaceable, it is usually a much more involved process (i.e., having the amplifier tested by a professional, removing the faulty component, and replacing it).

For working musicians this is usually a huge problem by comparison to looking in the back of a tube amp at the tubes and simply replacing the faulty tube. In addition, tubes can easily be removed and tested, while transistors cannot. Tube amplifiers respond differently from transistor amplifiers when signal levels approach and reach the point of clipping. In a tube-powered amplifier, the transition from linear amplification to limiting is less abrupt than in a solid state unit, resulting in a less grating form of distortion at the onset of clipping. For this reason, some guitarists prefer the sound of an all-tube amplifier; the aesthetic properties of tube versus solid state amps, though, are a topic of debate in the guitarist community.

Solid-state amplifiers

Most inexpensive guitar amplifiers currently produced are based on semiconductor (solid state) circuits, and some designs incorporate tubes in the preamp stage for their subjectively warmer overdrive sound. Tubes create warm overdrive sounds because instead of cutting the peaked signal off, they more or less pull the peaked audio information back (like natural compression) which creates a fuzzy overdrive sound. While this is a desirable attribute in many cases, the tube's characteristic will "color" all the sounds at any volume, unlike solid state. However, solid state in general have the quickest response time, perhaps even more so than modeling amps.

High-end solid state amplifiers are less common, since many professional guitarists tend to favor vacuum tubes. Some jazz guitarists, however, tend to favor the "colder" sound of solid-state amplifiers, preferring not to color the sound of their guitar with the tube distortion and compression so popular with rock, blues, and metal musicians.. Solid-state amplifiers vary in output power, functionality, size, price, and sound quality in a wide range, from practice amplifiers to professional models. Some inexpensive amplifiers have only a single volume control and a one or two tone controls.

Hybrid amplifiers

A tube power amp may be fed by a solid-state pre-amp circuit, as in the Fender Super Champ XD and the Roland Bolt amplifier, which is thereby classed as a 'hybrid' amp. Randall Amplifier's current flagship models, the V2 and T2, use hybrid amp technology. Alternatively, a tube pre-amp can feed a solid state output stage, as in models from Kustom and Vox. This approach dispenses with the need for an output transformer and allow modern power levels to be easily achieved.

The Roland Micro Cube, left, a small and portable digital modeling amplifier.

Modeling amplifiers

Modeling amplifiers use amplifier modeling to simulate the sound of well-known guitar amps, cabinets, and effects, as well as simulating the way traditional speaker cabinets sound when mixed with different types of microphones. They may also be an original creation not meant to simulate any particular real world guitar amp at all, instead allowing the user to create their own unique sound. Such as the original creations of companies like AcmeBarGig or Peavey. This is usually achieved through digital processing, although there are analog modeling amps as well, such as the Tech 21 Trademark. Modeling technology offers several advantages over traditional amplification. A modeling amp typically is capable of a wide range of tones and effects, and offers cabinet simulation, so it can be recorded without a microphone. Most modeling amps digitize the input signal and use a DSP, a dedicated microprocessor, to process the signal with digital computation. Some modeling amps incorporate vacuum tubes, digital processing, and some form of power attenuation.

Acoustic guitar amplifiers

These amplifiers are designed to be used with acoustic guitars, especially for the way these instruments are used in relatively quiet genres such as folk and bluegrass. They are similar in many ways to keyboard amplifiers, in that they have a relatively flat frequency response, and they are usually designed so that neither the power amplifier nor the speakers will introduce additional coloration.

To produce this relatively "clean" sound, these amplifiers often have very powerful amplifiers (providing up to 800 watts RMS), to provide additional "headroom" and prevent unwanted distortion. Since an 800 watt amplifier built with standard Class AB technology would be very heavy, some acoustic amplifier manufacturers use lightweight Class D amplifiers, which are also called "switching amplifiers."

Acoustic amplifiers are designed to produce a "clean", transparent, "acoustic" sound when used with acoustic instruments with built-in transducer pickups and/or microphones. The amplifiers often come with a simple mixer, so that the signals from a pickup and microphone can be blended. Since the early 2000s, it has become increasingly common for acoustic amplifiers to be provided with a range of digital effects, such as reverb and compression. As well, these amplifiers often contain feedback-suppressing devices, such as notch filters or parametric equalizers.

Amplifier configuration

A 3 x 6 stack of Marshall guitar cabinets for Jeff Hanneman of Slayer

In the case of electric guitars, an amplifier stack consisting of a head atop one cabinet is commonly called a *half stack*, while a head atop two cabinets is referred to as a *full stack*. The cabinet which the head sits on often has an angled top in front, while the lower cabinet of a full stack has a straight front. The first version of the *Marshall stack* was an amp head on an 8x12 cabinet, meaning a single speaker cabinet containing eight 12" guitar speakers. After six of these cabinets were made, the cabinet arrangement was changed to an amp head on two 4x12 cabinets, meaning four 12" speakers, to enable transporting the amp rig.

In heavy metal bands, the term "double stack" or "full stack" is sometimes used to refer to two stacks, with a second amplifier head serving as a slave to the first and four speaker cabinets in total. Another name for the "Head & Cab" that comes from the 1960s and 1970s is "Piggyback". Vox amp stacks could be put on a tiltable frame with casters. Fender heads could be attached to the cab and had "Tilt-Back" legs, like those used on larger Fender combo amps. Typically, a guitar amp's preamplifier section provides sufficient gain so that an instrument can be connected directly to its input, and sufficient power to connect loudspeakers directly to its output, both without requiring extra amplification.

Bands such as Immortal have used stage sets which appear to be stacks of guitar amplifiers.

Some touring bands have used the appearance of a large array of guitar amplifiers for aesthetic reasons. Some of these arrangements include one or more actual guitar cabinets, while others do not. Gizmodo writer Rosa Golijan investigated the phenomenon and found it "not too uncommon".

Another arrangement, often used for public address amplifier systems, is to provide two stages of amplification in separate units. First a *preamplifier* or *mixer* is used to boost the instrument output, normally to line level, and perhaps to mix signals from several instruments. The output from this preamplifier is then connected to the input of a power amplifier, which powers the loudspeakers.

Performing musicians that use the "two-stage" approach (as opposed to an amplifier with an integrated preamplifier and power amplifier) often want to custom-design a combination of equipment that best suits their musical or technical needs, and gives them more tonal and technical options. Some musicians require preamps that include specific features. Acoustic performers sometimes require preamps with "notch" filters (to prevent feedback), reverb, an XLR DI output, or parametric equalization. Hard rock, metal, or punk performers may desire a preamplifier with a range of distortion effects. As well, some musicians have specific power amplifier requirements, such as low-noise design, very high wattage, the inclusion of limiter features to prevent distortion and speaker damage, or biamp-capable operation.

With the "two-stage" approach, the preamplifier and power amplifier are often mounted together in a rack case. This case may be either free-standing or placed on top of a loudspeaker cabinet. If many rack-mounted effects are used, the rack may be a large unit on wheels. Some touring players need several racks of effects units to reproduce on stage the sounds they have produced in the studio. At the other extreme, if a small rack case containing both preamp and power amp is placed on top of a guitar speaker cabinet, the distinction between a rack and a traditional amp head begins to blur. Another variation is to combine the power amplifier into the speaker cabinet, an arrangement called a *powered speaker*, and use these with a separate preamp, sometimes combined into an effects pedal board or floor preamp/processor.

Preamplifiers are also used to connect very low-output or high-impedance instruments to instrument amplifiers. When piezoelectric transducers are used on upright bass or other acoustic instruments, the signal coming directly from the transducer is often too weak and it does not have the correct impedance for direct connection to an instrument amplifier. Small, battery-powered preamps are often used with acoustic instruments to resolve these problems.

Distortion, power, and volume

Power output

For electric guitar amplifiers, there is often a distinction between "practice" or "recording studio" guitar amps, which tend to have output power ratings of 20 watts down to a small fraction of a watt, and "performance" amps, which are generally 50 watts or higher. Traditionally, these have been fixed-power amplifiers, with a few models having a half-power switch to slightly reduce the listening volume while preserving power-tube distortion. The relationship between perceived volume and power output is not immediately obvious. A 5-watt amplifier is perceived to be half as loud as a 50-watt amplifier (a tenfold increase in power), and a half-watt amplifier is a quarter as loud as a 50-watt amp. Doubling the power of an amplifier results in a "just noticeable" increase in volume, so a 100-watt amplifier is held to be only just noticeably louder than a 50-watt amplifier. Such generalizations are also subject to the human ear's tendency to behave as a natural compressor at high volumes.

Power attenuation can be used with either low-power or high-power amplifiers, resulting in variable-power amplifiers. A high-power amplifier with power attenuation can produce power-tube distortion through a wide range of listening volumes. Speaker efficiency is also a major factor affecting a tube amplifier's maximum volume. For bass instruments, higher-power amplifiers are needed to reproduce low-frequency sounds. While an electric guitarist would be able to play at a small club with a 50-watt amplifier, a bass player performing in the same venue would probably need an amplifier with 200 or more watts.

Distortion and volume

Distortion is a feature available on many guitar amplifiers that is not typically found on keyboard or bass guitar amplifiers. Tube guitar amplifiers can produce distortion through pre-distortion equalization, preamp tube distortion, post-distortion EQ, power-tube distortion, tube rectifier compression, output transformer distortion, guitar speaker distortion, and guitar speaker and cabinet frequency response. Distortion sound or "texture" from guitar amplifiers is further shaped or processed through the frequency response and distortion factors in the microphones (their response, placement, and multi-microphone comb filtering effects), microphone preamps, mixer channel equalization, and compression. Additionally, the basic sound produced by the guitar amplifier can be changed and shaped by adding distortion and/or equalization effect pedals before the amp's input jack, in the effects loop just before the tube power amp, or after the power tubes.

Power-tube distortion

Power-tube distortion is required for amp sounds in some genres. In a standard master-volume guitar amp, as the amp's final or master volume is increased beyond the full power of the amplifier, power tube distortion is produced. The "power soak" approach places the attenuation between the power tubes and the guitar speaker. In the re-amped or "dummy load" approach, the tube power amp drives a mostly resistive dummy load while an additional low power amp drives the guitar speaker. In the isolation box approach, the guitar amplifier is used with a guitar speaker in a separate cabinet. A soundproofed isolation cabinet, isolation box, isolation booth, or isolation room can be used.

Volume controls

A variety of labels are used for level attenuation potentiometers in a guitar amplifier and other guitar equipment. Electric guitars and basses have a volume control to attenuate whichever pickup is selected. There may be two volume controls in parallel to mix the signal levels from the neck and bridge pickups. Rolling back the guitar's volume control also changes the pickup's equalization or frequency response, which can provide pre-distortion equalization.

The simplest guitar amplifiers have only a volume control. Most have at least a gain control and a master volume control. The gain control is equivalent to the distortion control on a distortion pedal, and similarly may have a side-effect of changing the proportion of bass and treble sent to the next stage.

A simple amplifier's tone controls typically include passive bass and treble controls. In some cases, a midrange control is provided. The amplifier's master volume control restricts the amount of signal permitted through to the driver stage and the power amplifier. When using a power attenuator with a tube amplifier, the master volume no longer acts as the master volume control. Instead, the power attenuator's attenuation control controls the power delivered to the speaker, and the amplifier's master volume control determines the amount of power-tube distortion. Power-supply based power reduction is controlled by a knob on the tube power amp, variously labeled "Wattage", "Power", "Scale", "Power Scale", or "Power Dampening".

Use with other instruments

Musicians often run sound-sources other than guitars through guitar amps. For live performances, synthesizers and drum machines or keyboards are often put through guitar amps to create a richer sound than can be obtained by patching the direct-outs right into the PA system. Guitar amplifiers can add tonal coloration, roll off unwanted high frequencies, and add overdrive or distortion. Deep Purple's Jon Lord played his Hammond Organ through a distorted Marshall amp to create a sound more suitable for heavy rock. String instruments and vocals are also put through guitar amps to add distortion effects. Some blues harp players also use guitar amps to create a warmer overdrive sound for their harmonica playing; 1950s-style "tweed" amps are often used for this purpose, such as Fender Bassman combo amps.

Recording engineers occasionally run pre-recorded parts through miked guitar amps, a process called re-amping. When a guitar part is recorded "dry" (e.g., without effects or distortion), straight into the mixing board for a recording, this gives the producer and mixing engineer much more flexibility to create new re-mixes or new tones from the recording. If a guitar player records an electric guitar part that is run through a chorus pedal and a distortion pedal, there is little that can be done at the "mix-down" stage to change the sound of this recording, beyond "tweaking" the equalization and modifying the level. Since re-mixing or mixdown can take place weeks, months, or even years after the original recording session, it may be impossible to have the guitarist come in to re-record a new part.

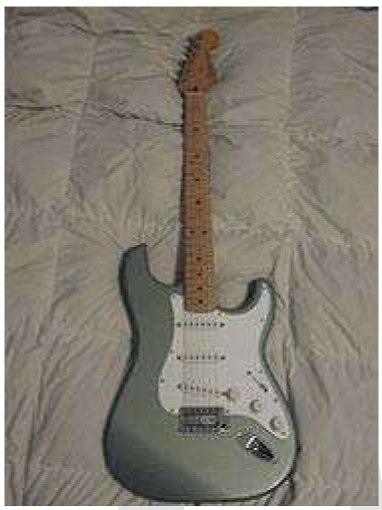
If the dry guitar sound is recorded, though, the mixing engineers can add any effects they want to the signal and then re-play it through a miked guitar amplifier which is being recorded. The effects, amplifiers, cabinets, and miking processes can be changed to any combination. When a dry guitar has been recorded, it can be a useful tool for "updating" an older recording. For example, if a band wants to re-release a 1980s-era album on which the guitar has a very dated 1980s sound, with heavy flanging and artificial-sounding electronic distortion, the band can update the guitar sound by re-amping the dry signal and using 2000s-era effects.

Mixing guitar amp signals with other signals is also done by some musicians. Chris Squire of Yes produced his bass guitar sound by playing through a guitar amplifier with its bass turned down, treble turned up, and volume turned up well into distortion; the miked guitar speaker signal was then mixed with a direct-input (DI) signal, a technique that has been used for processing synth keyboards as well. A bass guitar can also be played through a bass amp in parallel with a distorted guitar amp by using a DI box; the bass amp provides the low frequencies, while the guitar amp – which is not capable of reproducing the lowest frequencies of the bass guitar– emphasizes the upper harmonics of the instrument's tone.

Chapter 3

Electric Guitar

Electric guitar



Fender Stratocaster.

String instrument	
Classification	String instrument (plucked, either by fingerpicking, or with a pick.)
Hornbostel-Sachs classification	321.322 (Composite chordophone)
Playing range	

(a standard tuned guitar)

An **electric guitar** is a guitar that uses the principle of electromagnetic induction to convert vibrations of its metal strings into electric signals. Since the generated signal is too weak to drive a loudspeaker, it is amplified before sending it to a loudspeaker. Since the output of an electric guitar is an electric signal, the signal may easily be altered using electronic circuits to add color to the sound. Often the signal is modified using effects such as reverb and distortion. Conceived in 1931, the electric guitar became a necessity as jazz musicians sought to amplify their sound. Since then, it has evolved into a stringed musical instrument capable of a multitude of sounds and styles. It served as a major component in the development of rock and roll and countless other genres of music.

History

Various experiments at electrically amplifying the vibrations of a string instrument date back to the early part of the twentieth century. Patents from the 1910s show telephone transmitters adapted and placed inside violins and banjos to amplify the sound. Hobbyists in the 1920s used carbon button microphones attached to the bridge, however these detected vibration from the bridge on top of the instrument, resulting in a weak signal. With numerous people experimenting with electrical instruments in the 1920s and early 1930s, there are many claimants to have been the first to invent an electric guitar.

Electric guitars were originally designed by guitar makers and instrument manufacturers. Guitar innovator Les Paul experimented with microphones attached to guitars. Some of the earliest electric guitars adapted hollow bodied acoustic instruments and used tungsten pickups. An electrically amplified guitar was developed by George Beauchamp in 1931. Commercial production began in late summer of 1932 by Electro-Patent-Instrument Company Los Angeles, a partnership of Adolph Rickenbacker, Paul Barth and George Beauchamp, the inventor. The wooden body of the prototype was built by Harry Watson, a craftsman who had worked for the National Resophonic Guitar Company (where the men met). By 1934 the company was renamed Rickenbacker Electro Stringed Instrument Company.

The need for the amplified guitar became apparent during the big band era as orchestras increased in size, particularly when guitars had to compete with large brass sections. The first electric guitars used in jazz were hollow archtop acoustic guitar bodies with electromagnetic transducers. By 1932 an electrically amplified guitar was commercially available. Early electric guitar manufacturers include: Rickenbacker (first called Ro-Pat-In) in 1932, Dobro in 1933, National, AudioVox and Volu-tone in 1934,Vega, Epiphone (Electrophone and Electar), and Gibson in 1935 and many others by 1936.

The solid body electric guitar is made of solid wood, without functionally resonating air spaces. Rickenbacher, offered a cast aluminum electric steel guitar, nicknamed "The Frying Pan" or "The Pancake Guitar", developed in 1931 with production beginning in the summer of 1932. This guitar sounds quite modern and aggressive.

The first solid body "Spanish" standard guitar was offered by Vivi-Tone no later than 1934. An example of this model, featuring a guitar-shaped body of a single sheet of plywood affixed to a wood frame. Another early, substantially solid Spanish electric guitar, called Electro Spanish, was marketed by the "Rickenbacker" guitar company in

1935 and made of Bakelite. By 1936, the Slingerland company introduced a wooden solid body electric model.

The earliest documented performance with an electrically amplified guitar was in 1932, by Gage Brewer. The Wichita, Kansas-based musician had an Electric Hawaiian A-25 (frypan, lap-steel) and a standard Electric Spanish from George Beauchamp of Los Angeles, California. Brewer publicized his new instruments in an article in the *Wichita Beacon* of October 2, 1932 and through performances that month.

The first recordings using the electric guitar were by Hawaiian style players,in 1933. Bob Dunn of Milton Brown's Musical Brownies introduced the electric Hawaiian guitar to Western Swing with his January 1935 Decca recordings, departing almost entirely from Hawaiian musical influence and heading towards Jazz and Blues. Alvino Rey was an artist who took this instrument to a wide audience in a large orchestral setting and later developed the pedal steel guitar for Gibson. An early proponent of the electric Spanish guitar was jazz guitarist George Barnes who used the instrument in two songs recorded in Chicago on March 1, 1938, "Sweetheart Land" and "It's a Low-Down Dirty Shame". Some incorrectly attribute the first recording to Eddie Durham, but his recording with the Kansas City Five was 15 days later. Durham introduced the instrument to a young Charlie Christian, who made the instrument famous in his brief life and would be a major influence on jazz guitarists for decades thereafter.

Gibson's first production electric guitar, marketed in 1936, was the ES-150 model ("ES" for "Electric Spanish"; and "150" reflecting the $150 price of the instrument, along with a matching amplifier). The ES-150 guitar featured a single-coil, hexagonally shaped "bar" pickup, which was designed by Walt Fuller. It became known as the "Charlie Christian" pickup (named for the great jazz guitarist who was among the first to perform with the ES-150 guitar). The ES-150 achieved some popularity, but was suffered from unequal loudness across the six strings.

At an Engineering Fair in 1940, first prize went to NC State University physics professor Sidney Wilson for his invention of the world's first fully electric guitar. Wilson's guitar was also the first to have single-string pick-up, which addressed the unequal loudness problem of the ES-150's single coil. Professor Wilson also disposed of the acoustical body, reasoning that it was not necessary for a fully electric instrument. He developed the guitar shown here and entered it in the annual engineering fair. Patents from academia were quite unusual in the 1940s, so Professor Wilson did not patent his invention. In 1949 Gibson incorporated both the individual string pick-up and the cut-away body in its model ES-175. The design was attributed to Ted McCarthy of Gibson Corporation, but the features were first conceived and implemented by NC State physicists.

Early proponents of the electric guitar on record include: Jack Miller (Orville Knapp Orchestra), Alvino Rey (Phil Spitalney Orchestra), Les Paul (Fred Waring Orchestra), Danny Stewart (Andy Iona Orchestra), George Barnes (under many aliases), Floyd Smith, Big Bill Broonzy, T-Bone Walker, George Van Eps, Charlie Christian (Benny Goodman Orchestra) Tampa Red, Memphis Minnie, and Arthur Crudup.

A functionally solid body electric guitar was designed and built by Les Paul from an Epiphone acoustic archtop. His "log guitar" (so called because it consisted of a simple 4x4 wood post with a neck attached to it and homemade pickups and hardware, with two detachable Swedish hollow body halves attached to the sides for appearance only) shares nothing in design or hardware with the solid body "Les Paul" model sold by Gibson. However, the feedback problem associated with hollow-bodied electric guitars was understood long before Paul's "log" was created in 1940; Gage Brewer's Ro-Pat-In of 1932 had a top so heavily reinforced that it essentially functioned as a solid-body instrument.

In 1945, Richard D. Bourgerie made an electric guitar pickup and amplifier for professional guitar player George Barnes. Bourgerie worked through World War II at Howard Radio Company making electronic equipment for the American military. Barnes showed the result to Les Paul, who then arranged for Bourgerie to have one made for him.

Construction

Legend:
1. Headstock:
1.1 machine heads
1.2 truss rod cover
1.3 string guide
1.4 nut
2. Neck:
2.1 fretboard
2.2 inlay fret markers
2.3 frets
2.4 neck joint
g 3. Body

3.1 "neck" pickup
3.2 "bridge" pickup
3.3 saddles
3.4 bridge
3.5 fine tuners
3.6 tremolo arm (whammy bar)
3.7 pickup selector switch
3.8 volume and tone control knobs
3.9 output connector (input jack)(TRS)
3.10 strap buttons
4. Strings:
4.1 bass strings
4.2 treble strings

While guitar construction has many variations, in terms of the materials used for the body, the shape of the body, and the configuration of the neck, bridge, and pickups, there are features which are found in almost every guitar. The photo below shows the different parts of an electric guitar. The headstock (1) contains the metal machine heads, which are used for tuning; the nut (1.4), a thin fret-like strip of metal, plastic, graphite or bone which the strings pass over as they first go onto the fingerboard; the machine heads (1.1), which are worm gears which the player turns to change the string tension and thus adjust the tuning; the frets (2.3), which are thin metal strips which stop the string at the correct pitch when a string is pressed down against the fingerboard; the truss rod (1.2), a metal cylinder used for adjusting the tension on the neck (not found on all instruments); decorative inlay (2.2), a feature not found on lower-cost instruments.

The neck and the fretboard (2.1) extend from the body; at the neck joint (2.4), the neck is either glued or bolted to the body; the body (3) of this instrument is made of wood which is painted and lacquered, but some guitar bodies are also made of polycarbonate or other materials; pickups (3.1, 3.2), which are usually magnetic pickups, but which may also be piezoelectric transducer pickups; the control knobs (3.8) for the volume and tone potentiometers; a fixed bridge (3.4) -on some guitars, a spring-loaded hinged bridge called a "tremolo system" is used instead, which allows players to "bend" notes or chords down in pitch or perform a vibrato embellishment; and a plastic pickguard, a feature not found on all guitars, which is used to protect the body from scratches or cover the control cavity which holds most of the electric guitar's wiring.

The wood that the body (3) is made of is a very disputed subject considered by some to largely determine the sonic qualities of the guitar, while others believe that the sonic difference in a solid body guitar is very subtle between woods. In acoustic and archtop guitars there is a more pronounced sonic definition caused by the type of wood used. Typical woods include alder (brighter, but well rounded), swamp ash (similar to alder, but with more pronounced highs and lows), mahogany (dark, bassy, warm), poplar (similar to alder) and basswood (very neutral). Maple, a very bright tonewood, is also a popular body wood, but is very heavy. For this reason it is often placed as a 'cap' on a guitar made of primarily of another wood. Cheaper guitars are often made of cheaper woods, such as plywood, pine or agathis, not true hardwoods, which can affect the durability and tone of the guitar.Although most guitars are made from wood, any material may be used in the construction of a guitar. Materials such as plastic or cardboard are examples of unusual but possible materials that affect the overall sound of the guitar.

The input jack normally uses a TRS connector.

Pickups

Compared to an acoustic guitar, which has a hollow body, electric guitars make comparatively little audible sound simply by having their strings plucked, and so electric guitars are normally plugged into a guitar amplifier, which makes the sound louder. When an electric guitar is strummed, the movement of the strings generates (i.e., "induces") a very small electric current in the magnetic pickups, which are magnets wrapped with coils of very fine wire. That current is then sent through a cable to a guitar amplifier. The current induced is proportional to such factors as the density of the string or the amount of movement over these pickups. That vibration is, in turn, affected by several factors, such as the composition and shape of the body.

A close-up of the pickups on a Fender Squier "Fat Strat" guitar; on the left is a "humbucker" pickup and on the right are two single-coil pickups.

Some "hybrid" electric-acoustic guitars are equipped with additional microphones or piezoelectric pickups (transducers) that sense mechanical vibration from the body.

Because in some cases it is desirable to isolate the pickups from the vibrations of the strings, a guitar's magnetic pickups will sometimes be embedded or "potted" in epoxy or wax to prevent the pickup from having a microphonic effect.

Because of their natural inductive qualities, all magnetic pickups tend to pick up ambient and usually unwanted electromagnetic noises. The resulting noise, the so-called "hum", is particularly strong with single-coil pickups, and aggravated by the fact that very few guitars are correctly shielded against electromagnetic interference. The most frequent cause is the strong 50 or 60 Hz component that is inherent in the generation of electricity in the local power transmission system. As nearly all amplifiers and audio equipment associated with electrical guitars rely on this power, there is in theory little chance of completely eliminating the introduction of unwanted hum.

Double-coil or "humbucker" pickups were invented as a way to reduce or counter the unwanted ambient hum sounds (known as 60 cycle hum). Humbuckers have two coils of opposite magnetic and electric polarity. This means that electromagnetic noise hitting both coils should cancel itself out. The two coils are wired in phase, so the signal picked up by each coil is added together. This high combined inductance of the two coils leads to the richer, "fatter" tone associated with humbucking pickups. Optical pickups are a type of pickup which sense string and body vibrations using infrared LED light.

Vibrato arms

Some electric guitars have a tremolo arm (sometimes called a "whammy bar" or "vibrato arm" and occasionally abbreviated as *trem.* Technically, "vibrato arm" is correct, since moving the arm creates vibrato, not tremolo.), a lever attached to the bridge which can slacken or tighten the strings temporarily, changing the pitch, thereby creating a vibrato or a portamento effect. The name "tremolo bar" is somewhat misleading. It would be more accurate and appropriate to call it a vibrato bar. Tremolo is a fluctuation of volume. Vibrato is a fluctuation of pitch, which is what the whammy bar produces. Early vibrato systems, such as the Bigsby vibrato tailpiece, tended to be unreliable and cause the guitar to go out of tune quite easily, and also had a limited range. Later Fender designs were better, but Fender held the patent on these, so other companies used Bigsby-style vibrato for many years.

Detail of a Squier-made Fender Stratocaster. Note the vibrato arm, the 3 single-coil pickups, the volume and tone knobs.

With the expiration of the Fender patent on the Stratocaster-style vibrato, various improvements on this type of internal, multi-spring vibrato system are now available. Floyd Rose introduced one of the first improvements on the vibrato system in many years when in the late 1970s he began to experiment with "locking" nuts and bridges which work to prevent the guitar from losing tuning even under the most heavy whammy bar acrobatics.

Guitar necks

Electric guitar necks can vary according to composition as well as shape. The primary metric used to describe a guitar neck is the *scale*, which is the overall length of the strings from the nut to the bridge. A typical Fender guitar uses a 25.5 inch scale, while Gibson uses a 24.75 inch scale in their *Les Paul*. While Gibson's scale has often claimed to be 24.75", it has varied through the years by as much as a half inch. The frets are placed proportionally according to the scale length; thus, the smaller the scale, the tighter the spacing of the frets.

Necks are described as *bolt-on*, *set-in*, or *neck-through* depending on how they are attached to the body. Set-in necks are glued to the body in the factory, and are said to have a warmer tone and greater sustain; this is the most traditional type of joint. Bolt-on necks were pioneered by Leo Fender to facilitate easy adjustment and replacement of the guitar neck. Neck-through instruments extend the neck itself to form the center of the guitar body, and are known for long sustain and for being particularly sturdy. While a set neck can be carefully unglued by a skilled luthier, and a bolt-on neck can simply be unscrewed, a neck-through design is difficult or even impossible to repair, depending on the damage. Historically, the bolt-on style has been more popular for ease of installation and adjustment; since bolt-on necks can be easily removed, there is an after-market in replacement bolt-on necks from companies such as Warmoth and Mighty Mite. Some instruments, notably most Gibson models, have continued to use set/glued necks. Neck-through bodies are somewhat more common in bass guitars.

The materials used in the manufacture of the neck have great influence over the tone of the instrument. Hardwoods are very much preferred, with maple, ash, and mahogany topping the list. The neck and fingerboard can be made from different materials, such as a maple neck with a rosewood fingerboard. In the 1980s, exotic man-made materials such as graphite began to be used, but are pricey and never have replaced wood in production instruments. Such necks can be retrofitted to existing bolt-on instruments.

There are several different neck shapes used on guitars, including shapes known as C necks, U necks, and V necks. These refer to the cross-sectional shape of the neck (especially near the nut). There are also several sizes of fret wire available, with traditional players often preferring thin frets, and metal shredders liking thick frets. Thin frets are considered better for playing chords, while thick frets allow lead guitarists to bend notes with less effort. An electric guitar with a neck which folds back called the "Foldaxe" was designed and built for Chet Atkins by Roger C. Field (featured in Atkins' book "Me and My Guitars."). Steinberger guitars developed a line of exotic instruments without headstocks, with tuning done on the bridge instead.

Sound and effects

While an acoustic guitar's sound is largely dependent on the vibration of the guitar's body and the air within it, the sound of an electric guitar is largely dependent on a magnetically induced electrical signal, generated by the vibration of metal strings near sensitive pickups. The signal is then "shaped" on its path to the amplifier by using a range of effect devices or circuits that modify the tone and characteristics of the signal. The amplifiers and speakers used also add (intentional) coloration to the final sound.

Built-in sound shaping

Electric guitars usually have up to three magnetic pickups. Identical pickups will have different tones depending on how near they are to the neck or bridge, with bridge pickups having a bright or trebly timbre, and neck pickups being more warm or bassy. The type of pickup also affects tone, with dual-coil pickups sounding warmer, thicker, perhaps even muddy, and single coil pickups sounding clear, bright, perhaps even biting. Guitars do not have to be fitted with a uniform type of pickup: a common mixture is the "fat strat" arrangement of one dual-coil at the bridge position, with single coils in the middle and neck positions.

Where there is more than one pickup, selector switching is fitted. These often allow the outputs of two or more pickups to be combined, so that two-pickup guitars have three-way switches, and three-pickup guitars have five-way switches. Further circuitry is sometimes provided to combine the pickups in different ways. For instance, phase switching places one pickup out of phase with the other(s), leading to a "honky", "nasal", or "funky" sound. Individual pickups can also have their timbre altered by switches, typically coil tap switch, which effectively short-circuits some of a dual-coil pickup's windings, giving a tone like a single coil pickup.

The final stages of on-board sound-shaping circuitry are the volume control (potentiometer) and tone control (which "rolls off" the treble frequencies). Where there are individual volume controls for different pickups, and where pickup signals can be combined, they would affect the timbre of the final sound by adjusting the balance between pickups from a straight 50:50.

The strings fitted to the guitar also have an influence on tone. Rock musicians often prefer the lightest gauge of roundwound string, which are easier to bend, while jazz musicians go for heavier, flatwound strings with a rich, dark sound.

Classic amplifier sounds

In the 1960s, some guitarists began exploring a wider range of tonal effects by distorting the sound of the instrument. To do this, they used overdrive — increasing the gain, of the preamplifier beyond the level at which the signal could be faithfully reproduced, resulting in a "fuzzy" sound. This effect is called "clipping" by sound engineers, because when viewed with an oscilloscope, the wave forms of a distorted signal appear to have had their peaks "clipped off", approximating a square wave. This was not actually a new development in the instrument, but rather a shift of aesthetics, the sound having not been recognized as desirable previously.

Distortion achieved by overdrive necessarily involves high volumes and is therefore often combined with audio feedback.

After distortion became popular, amplifier manufacturers included various provisions for it, making amps easier to overdrive, and providing separate "dirty" and "clean" channels so that distortion could easily be switched in and out. The distortion characteristics of vacuum tube amplifiers are particularly sought-after, and various attempts have been made to emulate them without the disadvantages (fragility, low power, expense) of actual tubes.

Guitar amplifiers have long included at least a few effects, often tone controls and a spring reverb unit. The use of offboard effects is assisted by the provision of effect loops, an arrangement that allows effects to be taken out of circuit when not required.

Effects units

A Boss distortion pedal in use.

In the 1960s, the tonal palette of the electric guitar was further modified by introducing an Effects unit in its signal path. modifiers, wave-shaping circuits, voltage-controlled oscillators, or digital delays. Effects units come in several formats, the most common of which are the stomp-box and the rack-mount unit. A "stomp box" (or "pedal") is a small metal or plastic box containing the circuitry which is placed on the floor in front of the musician and connected in line with the patch cord connected to the instrument. The box is typically controlled by one or more foot-pedal on-off switches and it typically contains only one or two effects. "Guitar pedalboards" are used by musicians who use multiple stomp-boxes; these may be a DIY project made with plywood or a commercial pedalboard.

A rack-mount effects unit may contain the identical electronic circuit, but is mounted in a standard 19" equipment rack. Usually, however, rack-mount effects units contain several different types of effects. They are typically controlled by knobs or switches on the front panel, and often by a MIDI digital control interface.

Typical effects include:-

- Effects such as stereo chorus, phasers and flangers which shift the pitch of the signal by a small and varying amount, creating swirling, shimmering and whooshing noises.
- Effects such as octavers, which displace pitch by an exact musical interval.
- Distortion, such as transistor-style fuzz, or effects incorporating or emulating vacuum tube distortion.
- Filters such as wah-wah
- Envelope shapers, such as compression/sustain or volume/swell.
- Time-shift effects such as delay and reverb.

Modern amplifier techniques

In the 1970s, as effects pedals proliferated, their sounds were combined with tube amp distortion at lower, more controlled volumes by using power attenuators such as Tom Scholz' Power Soak as well as re-amplified dummy loads such as Eddie Van Halen's use of a variac, power resistor, post-power-tube effects, and a final solid-state amp driving the guitar speakers. A variac is one approach to power-supply based power attenuation, to make the sound of power-tube distortion more practically available.

Recent amplifiers may include digital technology similar to modern effects pedals, including the ability to model or emulate a variety of classic amps.

Digital and software-based effects

The Zoom 505 multi-effect pedal.

A **multi-effects device** (also called a "multi-FX" device) is a single electronics effects pedal or rackmount device that contains many different electronic effects. In the late 1990s and throughout the 2000s, multi-FX manufacturers such as Zoom and Korg produced devices that were increasingly feature-laden. Multi-FX devices allow several of the effects to be used together, and most devices allow users to set "preset" combinations of different effects including distortion, chorus, reverb, compression, and so on. This allows musicians to have quick on-stage access to different effects combinations. Some multi-FX pedals contain modelled versions of well-known effects pedals or amplifiers.

The Boss GT-8 is a higher-end multi-effect processing pedal; note the preset switches and patch bank footswitches and built-in expression pedal.

Multi-effects devices have garnered a large share of the effects device market because they offer the user such a large variety of effects in a single package. A low-priced multi-effects pedal may provide 20 or more effects for the price of a regular single-effect pedal. More expensive multi-effect pedals may include 40 or more effects, amplifier modelling, and the ability to combine effects and/or modelled amp sounds in different combinations, as if the user was using multiple guitar amps. More expensive multi-effects pedals may also include more input and output jacks (e.g., an auxiliary input or a "dry" output), MIDI inputs and outputs, and an expression pedal, which can control volume or modify effect parameters (e.g., the rate of the simulated rotary speaker effect).

By the 1980s and 1990s, software effects became capable of replicating the analog effects used in the past. These new digital effects attempted to model the sound produced by analog effects and tube amps, to varying degrees of quality. There are many free guitar effects computer programs for computers that can be downloaded via the Internet. Now, computers with sound cards can be used as digital guitar effects processors. Although digital and software effects offer many advantages, many guitarists still use analog effects.

Synthesizer and digital guitars

In 2002, Gibson announced the first digital guitar, which performs analog-to-digital conversion internally. The resulting digital signal is delivered over a standard Ethernet cable, eliminating cable-induced line noise. The guitar also provides independent signal processing for each individual string. Also, in 2003 amp maker Line 6 released the Variax guitar. It differs in some fundamental ways from conventional solid-body electrics. For example it uses piezoelectric pickups instead of the conventional electromagnetic ones, and has an on-board computer capable of modifying the sound of the guitar to model the sound of many instruments.

Playing techniques

A prepared guitar

The sound of a guitar is not only adapted by electronic sound effects, but also heavily by all kinds of new techniques developed or becoming possible in combination with the electric amplification. This is called extended technique.

Extended techniques include:-

- String bending (or radial finger vibrato.). This is not quite unique to the electric instrument, but is greatly facilitated by the light strings typically used on solid body guitars.

- Neck bending, by holding the upper arm on the guitar body and bending the neck either to the front or pulling it back. This is used as a substitute for a tremolo bar, although not as effective and too powerful of force use could snap the guitar neck.
- The use of the whammy bar or "tremolo" arm, including the extreme technique of dive bombing.
- Tapping, in which both hands are applied to the fretboard. This is only feasible with the assistance of amplification.
- Pinch harmonics, sometimes called "squealies".
- Volume swells, in which the volume knob is repeatedly rolled to create a violin-like sound. Note that the same result can also be accomplished through the use of an external swell pedal, although the knob technique can enhance showmanship and conveniently eliminate the need for another pedal.
- Use of audio feedback to enhance sustain and change timbre.
- Substitution of another device for the plectrum, for instance the cello bow (as famously used by Jimmy Page) and the e-bow, (a device using electromagnetic feedback to vibrate strings without direct contact). Like feedback, these techniques increase sustain, bring out harmonics and change the acoustic envelope.
- Sustainers built into the guitar itself.
- Use of slide or bottlenecks.
- Sometimes guitars are even adapted with extra modifications to alter the sound, such as Prepared guitar and 3rd bridge.

Other techniques such as axial finger vibrato, pull-offs, hammer-ons, palm muting, harmonics and altered tunings are also used on the classical and acoustic guitar. Shred guitar is a genre involving a number of extended techniques.

Types

Paul Reed Smith Standard 22

Solid body

Solid body electric guitars have no hollow internal cavity to accommodate vibration and no sound holes such as those used to amplify string vibrations in acoustic guitars. Solid body instruments are generally made up of hardwood with a lacquer coating. The wood is dried for 3 to 6 months in heated storage before being cut to shape. The sound that is audible in music featuring electric guitars is produced by pickups on the guitar that convert the string vibrations into an electrical signal. The signal is then fed to an amplifier (or amp) and speaker.

One of the first solid body guitars was invented by Les Paul. Gibson did not present their 'Les Paul' guitar prototypes to the public, as they did not believe it would catch on. The first mass-produced solid-body guitar was Fender's Broadcaster (later to become the 'Telecaster') first made in 1948, five years after Les Paul made his prototype. The Gibson Les Paul appeared soon after to compete with the Broadcaster.

String-through body

When discussing electric guitar construction, the term **string-through body** is used to describe a type of solid body electric guitar body in which the strings are threaded through holes drilled into the bottom of the guitar body. The strings are typically held in place using metal ferrules screwed or glued into the holes.

The advantages of a string-through body mostly relate to improvements in a guitar's sustain and timbre. It is also by nature impossible to install a tremolo arm ***and*** have the string ends anchored through the body. Tremolo systems change string tension by changing the physical length of the string. This requires the end of the string to be anchored to the (tremolo) bridge unit directly, instead of to the wood of the body. [*A talented guitar luthier or repair person is certainly capable of installing a tremolo bridge on this type of guitar, however, this is rarely done as it eliminates whatever tonal benefits are provided by the string-through body system and permanently degrades the value and collectability of the original instrument.*]

Examples of string-through bodies on guitars include the Fender Telecaster Thinline and Telecaster Deluxe.

Semi-acoustic

An Epiphone brand semi-acoustic hollow-body guitar.

These guitars have a hollow body and electronic pickups mounted on its body. They work in a similar way to solid body electric guitars except that because the hollow body also vibrates, the pickups convert a combination of string and body vibration into an electrical signal. A variant form, the semi-hollow body guitar, strikes a balance between the characteristics of solid-body and hollow-body guitars. Advocates of semi-hollow-body guitars argue that they have greater resonance and sustain than true solid-body guitars, as a solid wood body. Several metal bodies were made in the 1950s by violin and cello makers. In the 1970s, John Veleno made a polished aluminum guitar. Liquid Metal Guitars makes a metal body guitar made out of a solid block of aluminum and then chrome or gold-plates the instrument.

Chambered

Many guitars otherwise sold as solid-bodied instruments, such as the Gibson Les Paul or the PRS Singlecut, are built with "weight relief" holes bored into the body which affect the sound of the instrument. The Les Paul Supreme edition is currently described by the manufacturer as a "chambered" instrument, with a weight relief system designed to positively affect the sound.

Electric acoustic

Some steel-string acoustic guitars are fitted with pickups purely as an alternative to using a separate microphone. They may also be fitted with a piezoelectric pickup under the bridge, attached to the bridge mounting plate, or with a low mass microphone (usually a condenser mic) inside the body of the guitar that will convert the vibrations in the body into electronic signals, or even combinations of these types of pickups, with an integral mixer/preamp/graphic equalizer. These are called electric acoustic guitars, and are regarded as acoustic guitars rather than electric guitars because the pickups do not produce a signal directly from the vibration of the strings, but rather from the vibration of the guitar top or body.

These should not be confused with semi-acoustic guitars, which have pickups of the type found on solid body electric guitars, or solid-bodied hybrid guitars with piezoelectric pickups.

String, bridge, and neck variants

Although rare, the one-string guitar is sometimes heard, particularly in Delta blues, where improvised folk instruments were popular in the 1930s and 1940s. Eddie "One String" Jones had some regional success with a Mississippi blues musician Lonnie Pitchford played a similar, homemade instrument. In a more contemporary style, Little Willie Joe, the inventor of the Unitar, had a rhythm and blues instrumental hit in the 1950s with "Twitchy", recorded with the Rene Hall Orchestra. The best-known exponent of the four-string guitar, often called the tenor guitar was Tiny Grimes, who played on 52nd Street with the beboppers and played a major role in the Prestige Blues Swingers. Grimes' guitar omitted the bottom two strings. Deron Miller of CKY only uses four strings, but plays a six string guitar with the two highest strings removed. Many banjo players use this tuning: DGBE, mostly in Dixieland. Guitar players find this an easier transition than learning plectrum or tenor tuning.

Seven-string

Most Seven-string guitars add a low "B" string below the low "E". Both electric and classical guitars exist designed for this tuning. A high "A" string above the high "E" instead of the low "B" is sometimes used. Another less common seven-string arrangement is a second G string situated beside the standard G string and tuned an octave higher, in the same manner as a twelve-stringed guitar. Jazz guitarists using a seven-string include veteran jazz guitarists George Van Eps, Lenny Breau, Bucky Pizzarelli and his son John Pizzarelli.

Seven-string electric guitars were popularized among rock players in the 1980s by Steve Vai. Along with the Japanese guitar company Ibanez, Vai created the Universe series seven string guitars in the 1980s, with a double locking tremolo system for a seven string guitar. These models were based on Vai's six string signature series, the Ibanez Jem. Seven-string guitars experienced a resurgence in popularity in the 2000s, championed by Limp Bizkit, Slayer, KoRn, Fear Factory, Strapping Young Lad, Nevermore, Muse and other hard rock/metal bands. Metal musicians often prefer the seven-string guitar for its extended lower range. The seven-string guitar has also played an essential role in progressive metal rock, and is commonly used in bands such as Dream Theater, Pain of Salvation and by experimental guitarists such as Ben Levin.

Eight and nine-string

Eight-string electric guitars are rare, but not unused. One is played by Charlie Hunter (manufactured by Novax Guitars). The largest manufacturer of 8- to 14-strings is Warr Guitars. Their models are used by Trey Gunn (ex King Crimson) who has his own *signature* line from the company. Also, Mårten Hagström and Fredrik Thordendal of Meshuggah used 8-string guitars made by Nevborn Guitars and now guitars by Ibanez.Alan Aldo of the band Mariachi Terror uses acoustic and electric 8 string guitars to produce his sound. Munky of nu metal band KoRn is also known to use seven-string Ibanez guitars and it is rumored that he is planning to release a K8 eight-string guitar similar to his K7 seven-string guitar. Another Ibanez is Tosin Abasi, lead guitar player of progressive metal band Animals as Leaders, who uses an Ibanez RG2228 to mix bright chords with very heavy low riffs on the 7 and 8th strings. Stephen Carpenter of Deftones also switched from 7 to 8 string in 2008 and released his signature STEF B-8 with ESP Guitars. In 2008 Ibanez released the Ibanez RG2228-GK which is the first mass produced eight-string guitar. Jethro Tull's first album uses a nine-string guitar on one track. Minarik Guitars manufactures the "Inferno V" 9 stringed guitar that has the top three strings doubled up with strings that are an octave higher, like 12 stringed guitars. Bill Kelliher, guitarist for the heavy metal group Mastodon, worked with First Act on a custom mass-produced nine-string guitar.

Ten-string

B.C.Rich manufacture a ten-string six-course electric guitar known as the *Bich*, whose radical shape was specifically designed to allow the machine heads for the four secondary strings to be positioned on the body, avoiding the head-heaviness of many electric twelve-string guitars. However many players bought it for the body shape or electrics and simply removed the extra strings. The company recognized this and released six-string models of the Bich, but ten-string models also remain in production.

In October 2008, a ten-string electric jazz guitar by Mike Shishkov was demonstrated at the 3rd International Ten String Guitar Festival. This instrument was based on the ten-string extended-range classical guitar.

Twelve-string

Twelve string electric guitars feature six pairs of strings, usually with each pair tuned to the same note. The extra E, A, D, and G strings add a note one octave above, and the extra B and E strings are in unison. The pairs of strings are played together as one, so the technique and tuning are the same as a conventional guitar, although creating a much fuller tone. They are used almost solely to play harmony and rhythm. They are relatively common in folk rock music. Lead Belly is the folk artist most identified with the twelve-string guitar, usually acoustic with a pickup.

George Harrison of The Beatles and Roger McGuinn of The Byrds brought the electric twelve-string to notability in rock and roll. During the Beatles' first trip to the US, in February 1964, Harrison received a new "360/12" model guitar from the Rickenbacker company, a 12-string electric made to look onstage like a 6-string. He began using the 360 in the studio on Lennon's "You Can't Do That" and other songs. Roger McGuinn began using electric 12-string guitars to create the jangly sound of The Byrds. Another notable guitarist to utilize electric 12-string guitars is Jimmy Page, the guitarist with hard rock-heavy metal and rock group Led Zeppelin.

3rd bridge

The 3rd bridge guitar is an electric prepared guitar with an additional 3rd bridge. This can be a normal guitar with for instance a screwdriver placed under the strings, but can also be a custom made instrument. Lee Ranaldo of Sonic Youth plays with a 3rd bridge.

Double neck guitar

A white Gibson EDS-1275

Double neck (or, less commonly, "twin-neck") guitars enable guitarists to play guitar and bass guitar or, more commonly, a six-string and twelve-string. An early user was John McLaughlin, but the double-neck guitar was popularized by Jimmy Page, who used a custom-made Gibson EDS-1275 to perform the "Stairway to Heaven" and "The Song Remains the Same", although "Stairway to Heaven" was actually recorded using a Fender Telecaster. Don Felder of the Eagles also used the Gibson EDS-1275 during the Hotel California tour. Muse guitarist and vocalist Matthew Bellamy uses a silver Manson Double Neck on his bands' The Resistance Tour.

Uses

Popular music

Popular music and rock groups often use the electric guitar in two roles: as a rhythm guitar which provides the chord sequence or "progression" and sets out the "beat" (as part of a rhythm section), and a lead guitar, which is used to perform melody lines, melodic instrumental fill passages, and guitar solos. In some rock or metal bands with two guitarists, the two performers may perform as a guitar tandem, and trade off the lead guitar and rhythm guitar roles. In bands with a single guitarist, the guitarist may switch between these two roles, playing chords to accompany the singer's lyrics, and then playing a guitar solo in the middle of the song.

In the most commercially available and consumed pop and rock genres, electric guitars tend to dominate their acoustic cousins in both the recording studio and the live venue, especially in the "harder" genres such as heavy metal and hard rock. However the acoustic guitar remains a popular choice in country, western and especially bluegrass music, and it is widely used in folk music.

Jazz and jazz fusion

Jazz guitar playing styles include rhythm guitar-style "comping" (accompanying) with jazz chord voicings (and in some cases, walking basslines) and "blowing" (improvising solos) over jazz chord progressions with jazz-style phrasing and ornaments. The accompanying style for electric guitar in most jazz styles differs from the way chordal instruments accompany in many popular styles of music. In rock and pop, the rhythm guitarist usually performs the chords in rhythmic fashion which sets out the beat of a tune. In contrast, in many modern jazz styles, the guitarist plays much more sparsely, intermingling periodic chords and delicate voicings into pauses in the melody or solo. Jazz chord voicings are usually rootless and emphasize the 3rd and 7th notes of the chord.

When jazz guitar players improvise, they use the scales, modes, and arpeggios associated with the chords in a tune's chord progression. Jazz guitarists have to learn how to use scales (whole tone scale, chromatic scale, etc.) to solo over chord progressions. Jazz guitar improvising is not merely the recitation of jazz scales and rapid arpeggios. Jazz guitarists often try to imbue their melodic phrasing with the sense of natural breathing and legato phrasing used by horn players such as saxophone players. As well, a jazz guitarists' solo improvisations have to have a rhythmic drive and "time feel" that creates a sense of "swing" and "groove".

Most jazz guitarists play hollow body instruments, but solid body guitars are also used. Hollow body instruments were the first guitars used in jazz in the 1930s and 1940s. During the 1970s jazz fusion era, many jazz guitarists switched to the solid body guitars that dominated the rock world.

Contemporary classical music

Until the 1950s, the acoustic, nylon-stringed classical guitar was the only type of guitar favored by classical, or art music composers. In the 1950s a few contemporary classical composers began to use the electric guitar in their compositions. Examples of such works include Karlheinz Stockhausen's *Gruppen* (1955–57); Donald Erb's *String Trio* (1966), Morton Feldman's *The Possibility of a New Work for Electric Guitar* (1966); George Crumb's *Songs, Drones, and Refrains of Death* (1968); Hans Werner Henze's *Versuch über Schweine* (1968); Francis Thorne's *Sonar Plexus* (1968) and *Liebesrock* (1968–69), Michael Tippett's *The Knot Garden* (1965–70); Leonard Bernstein's *MASS* (1971) and *Slava!* (1977); Louis Andriessen's *De Staat* (1972–76); Helmut Lachenmann's *Fassade, für grosses Orchester* (1973, rev. 1987), Steve Reich's *Electric Counterpoint* (1987), Arvo Pärt's *Miserere* (1989/92), György Kurtág's *Grabstein für Stephan* (1989), and countless works composed for the quintet of Ástor Piazzolla. Alfred Schnittke also used electric guitar in several works, like the "Requiem", "Concerto Grosso N°2" and "Symphony N°1".

In the 1980s and 1990s, a growing number of composers (many of them composer-performers who had grown up playing the instrument in rock bands) began writing contemporary classical music for the electric guitar. These include Shawn Lane, Steven Mackey, Nick Didkovsky, Scott Johnson, Lois V Vierk, Tim Brady, Tristan Murail, John Rogers, and Randall Woolf.

Yngwie Malmsteen released his Concerto Suite for Electric Guitar and Orchestra in 1998, and Steve Vai released a double-live CD entitled *Sound Theories*, of his work with the Netherlands Metropole Orchestra in June 2007. The American composers Rhys Chatham and Glenn Branca have written "symphonic" works for large ensembles of electric guitars, in some cases numbering up to 100 players, and the instrument is a core member of the Bang on a Can All-Stars (played by Mark Stewart). Still, like many electric and electronic instruments, the electric guitar remains primarily associated with rock and jazz music, rather than with classical compositions and performances. R. Prasanna plays a style of Indian classical music (Carnatic music) on the electric guitar.

In the 21st century, European avant garde composers like Richard Barrett, Fausto Romitelli and Karlheinz Essl used the electric guitar (together with extended playing techniques) in solo pieces or ensemble works. Probably the most ambitious and perhaps significant work to date is *Ingwe* (2003–2009) by Georges Lentz (written for Australian guitarist Zane Banks), a 60-minute work for solo electric guitar, exploring that composer's existential struggles and taking the instrument into realms previously unknown in a concert music setting.

Chapter 4

Electric Piano

Rhodes Mark II 73 Stage Piano

Neo-Bechstein (1929)

Vierling-Förster piano (1937)

An **electric piano** is an electric musical instrument.

Electric pianos produce sounds mechanically and the sounds are turned into electronic signals by pickups. Unlike a synthesizer, the electric piano is not an electronic instrument, but electro-mechanical. The earliest electric pianos were invented in the late 1920s; the 1929 *Neo-Bechstein* electric grand piano was among the first. Probably the earliest stringless model was Lloyd Loar's *Vivi-Tone Clavier*.

The popularity of the electric piano began to grow in the late 1950s, reaching its height during the 1970s, after which they were eventually replaced by synthesizers capable of piano-like sounds without the disadvantages of moving mechanical parts. Many models were designed for home or school use, or to replace a heavy and un-amplified piano on stage, while others were conceived for use in school or college piano labs for the simultaneous tuition of several students using headphones.

Due to their size and weight, digital stage pianos have replaced many of the original electromechanical instruments in contemporary usage. However, In 2009, Rhodes Music Corporation started producing a new line of electro-mechanical pianos, known as the Rhodes Mark 7.

Tone Production

Yamaha CP-70M

Strings and hammers of Yamaha CP-70

The actual method of tone production varies from one model to another:

Struck strings

Yamaha, Baldwin, Helpinstill and Kawai's electric pianos are actual grand or upright pianos with strings and hammers. The Helpinstill models have a traditional soundboard; the others have none, and are more akin to a solid-body electric guitar. On Yamaha, Baldwin and Kawai's pianos, the vibration of the strings is converted to an electrical signal by piezoelectric pickups under the bridge. Helpinstill's instruments use a set of electromagnetic pickups attached to the instrument's frame. All these instruments have a tonal character similar to that of an acoustic piano.

Struck reeds

Wurlitzer 200A

Wurlitzer electric pianos use flat steel reeds struck by felt hammers. The reeds fit within a comb-like metal plate, and the reeds and plate together form an electrostatic or capacitive pickup system, using a DC voltage of 170v. This system produces a very distinctive tone – sweet and vibraphone-like when played gently, and developing a hollow resonance as the keys are played harder. The reeds are tuned by adding or removing mass from a lump of solder at the free end of the reed. Replacement reeds are furnished with a slight excess of solder, and thus tuned "flat"; the user is required – by repeated trial and error – to gradually file off the excess solder until the correct tuning is achieved. The "Columbia Elepian," also branded as "Maestro" uses a reed system similar to the Wurlitzer.

Struck tuning-forks

Rhodes Mark II Stage 73

Tuning forks of Fender Rhodes Mark I

The tuning-fork here refers to the struck element having two vibrating parts – physically it bears little resemblance to a traditional type. In Fender Rhodes instruments, the struck portion of the "fork" is a tine of stiff steel wire. The other part of the fork, parallel and adjacent to the tine, is the tonebar, a sturdy steel bar which acts as a resonator and adds sustain to the sound. The tine is fitted with a spring which can be moved along its length to allow the pitch to be varied for fine-tuning. The tine is struck by the small neoprene (originally felt) tip of a hammer activated by a greatly simplified piano action (each key has only three moving parts including the damper). Each tine has an electromagnetic pickup placed just beyond its tip. The Rhodes piano has a distinctive bell-like tone, fuller than the Wurlitzer, with longer sustain and with a "growl" when played hard. Hohner's "Electra-Piano" uses a similar system, with a metal reed replacing the Rhodes' tine. Its sound is correspondingly somewhere between the Rhodes and Wurlitzer.

Plucked reeds

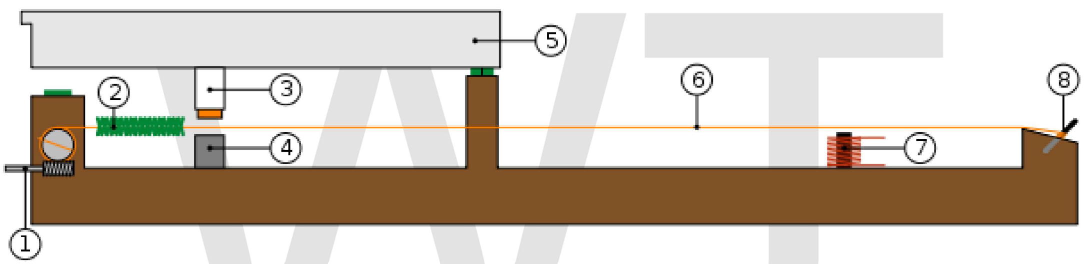
Hohner Pianet (below)

Weltmeister Claviset (Selmer Pianotron)

Hohner's original Pianet uses adhesive pads made from foam rubber and leather impregnated with a viscous silicone oil to pluck metal reeds. When the key is released, the pad acts as a damper. An electrostatic pickup system similar to Wurlitzer's is used. The tone produced resembles that of the Wurlitzer but brighter and with less sustain. The same firm's "Cembalet" uses rubber plectra and separate dampers but is otherwise almost identical. Hohner's later "Pianet T" uses silicone rubber suction pads rather than adhesive pads and replaces the electrostatic system with passive electromagnetic pickups similar to those of the Rhodes, the reeds themselves however being magnetized. The Pianet T has a far mellower sound not unlike that of the Rhodes instruments. None of the above instruments have the facility for a sustain pedal.

A close copy of the Cembalet is the "Weltmeister Claviset," also marketed as the "Selmer Pianotron." This has electromagnetic pickups with a battery-powered preamplifier, and later models have multiple tone filters and a sustain pedal.

Other electric keyboard instruments



Tangent action of a Clavinet : 1.Tuning / 2.Damper / 3.Tangent / 4.Anvil / 5.Key / 6.String / 7.Pickup / 8. Tailpiece

Hohner Clavinet D6

Although not technically pianos, the following are electric harpsichords and clavichords.

Baldwin's "Solid-Body Electric Harpsichord" or "Combo Harpsichord" is an aluminum-framed instrument of fairly traditional form, with no soundboard and with two sets of electromagnetic pickups, one near the plectra and the other at the strings' mid-point. The instrument's sound has something of the character of an electric guitar, and has occasionally been used to stand in for one in modern chamber music. Roger Penney of Bermuda Triangle Band worked on the design and development of the original instrument for the Cannon Guild Company, a premier harpsichord maker located in Cambridge Massachusetts. This instrument had an aluminum bar frame, a spruce wood soundboard, bar magnetic pickups, and a Plexiglas (clear plastic) openable lid. The prototypes and design were sold to Baldwin who made some modifications, and then manufactured the instrument under their own name.

Hohner's "Clavinet" is essentially an electric clavichord. A rubber pad under each key presses the string onto a metal anvil, causing the "fretted" portion of the string to vibrate. When the key is released, the whole string is theoretically free to vibrate but is immediately damped by yarn woven across the far end. Two electromagnetic single-coil pickups under the strings detect the vibrations which are then preamplified and filtered.

Playing technique and styles

As with electric vs. acoustic guitars, the sound of most electric pianos differs considerably from that of an acoustic instrument, and the electric piano has thus acquired a musical identity of its own, far beyond that of simply being a portable, amplified piano. In particular, the Rhodes piano lends itself to long, sustained "floating" chords in a way which would be impossible on an acoustic instrument, while the Hohner Clavinet has an instantly recognizable vocabulary of percussive riffs and figures which owe less to conventional keyboard styles than to funk rhythm guitar and slap bass. Early Wurlitzer models had vacuum tube amplifiers, which could be over-driven to create a distinctive distortion. Later transistorized models, while sharing a similar mechanical approach to sound generation, didn't replicate the "fat" sound of the tube-based models, but instead sported a soulful and useful tremolo.

- Examples:
    - Rhodes piano
    - Hohner Cembalet, Clavinet, Pianet, Electra Piano
    - Wurlitzer EP-200A
    - Yamaha CP-70 Electric Grand Piano

Rhodes Mark 7 (2009) on stage

- Popular pieces with electric pianos:
  - Fender Rhodes:
    - *The Beatles*: "Get Back", "Don't Let Me Down" (both played by Billy Preston)
    - *The Doors*: "L.A. Woman", "Riders on the Storm" (played by Ray Manzarek)
    - *Chick Corea*: "Spain", "La Fiesta"
    - *Herbie Hancock*: "Chameleon"
    - *Billy Joel*: "Just the Way You Are"
    - *Stevie Wonder*: "You Are the Sunshine of My Life", "Isn't She Lovely", "I Believe (When I Fall In Love It Will Be Forever)"
    - *Pink Floyd*: "Dogs", "Hey You"; "Sheep"
    - *Elton John*: "Daniel"; "Sorry Seems To Be The Hardest Word"; "Little Jeannie"
    - *Peter Frampton*: "Baby, I Love Your Way"
    - *One Day as a Lion*: "Wild International"
  - Hohner Cembalet:
    - *Manfred Mann*: "Do Wah Diddy Diddy"
    - *Elvis Costello*: "Veronica"
    - *The Stranglers*: "No More Heroes"

- Hohner Clavinet:

The Clavinet C, used on Stevie Wonder's Superstition.

  - *The Band*: "Up On Cripple Creek"
  - *Stevie Wonder*: "Superstition"
  - *Led Zeppelin*: "Custard Pie", "Trampled Underfoot"
  - *Steely Dan*: "Kid Charlemagne"
  - *Pink Floyd*: "Pigs (Three Different Ones)", "Shine On You Crazy Diamond (Parts 6-9)"
  - *Gentle Giant: "Cogs In Cogs, "Experience", "So Sincere"*
  - *Van der Graaf Generator: "The Undercover Man", "Scorched Earth", "Arrow"*
- Hohner Electra-Piano:
  - *Led Zeppelin*: "Stairway To Heaven", "Misty Mountain Hop", "No Quarter", "Down By The Seaside" (studio recordings only, when played live these songs were played on a Fender Rhodes)
- Hohner Pianet:
  - *The Association*: "Never My Love"
  - *The Beatles*: "The Night Before", "I Am the Walrus", "Tell Me What You See", "You Like Me Too Much"
  - *The Guess Who*: "These Eyes"
  - *The Zombies*: "She's Not There"
  - *The Kingsmen*: "Louie Louie"
  - *The Lovin' Spoonful*: "Summer In The City"
  - *Soft Machine: "Slightly All The Time", "Out-Bloody-Rageous"*

        - *Van Der Graaf Generator: "Plague of lighthouse"*
    - Wurlitzer Electric Piano:

Wurlitzer 200A used by Supertramp

        - *Ray Charles*: "What'd I Say"
        - *Cannonball Adderley Quintet*: "Mercy, Mercy, Mercy" Only first studio recording, all subsequent live verisions are Fender-Rhodes.
        - *The Buckinghams*: "Hey, Baby (They're Playing Our Song)"
        - *Steely Dan*: "Do It Again"
        - *Pink Floyd*: "Time", "Money", "Have A Cigar", "Shine On You Crazy Diamond (Parts 6-9)"
        - *Queen*: "You're My Best Friend"
        - *King Harvest*: "Dancing in the Moonlight"
        - *Supertramp*: "Dreamer", "Lady (supertramp song)", "Bloody Well Right", "The Logical Song"; "Goodbye Stranger"
    - Baldwin Combo Harpsichord:
        - *The Association*: "Along Comes Mary"
        - *The Beatles*: "Because"

Chapter 5

Electronic Musical Instrument

Ondes Martenot created by Maurice Martenot, 1928

An **electronic musical instrument** is a musical instrument that produces its sounds using electronics. Such an instrument sounds by outputting an electrical audio signal that ultimately drives a loudspeaker.

An electronic instrument may include a user interface for controlling its sound, often by adjusting the pitch, frequency, or duration of each note. However, it is increasingly common for the user interface and sound-generating functions to be separated into a music controller (input device) and a music synthesizer, respectively, with the two devices communicating through a musical performance description language such as MIDI or Open Sound Control.

All electronic musical instruments can be viewed as a subset of audio signal processing applications. Simple electronic musical instruments are sometimes called sound effects; the border between sound effects and actual musical instruments is often hazy.

French composer and engineer Edgard Varèse created a variety of compositions using electronic horns, whistles, and tape. Most notably, he wrote Poème Électronique for the Phillips pavilion at the Brussels World Fair in 1958.

Electronic musical instruments are now widely used in most styles of music. The development of new electronic musical instruments, controllers, and synthesizers continues to be a highly active and interdisciplinary field of research. Specialized conferences, notably the International Conference on New interfaces for musical expression, have organized to report cutting edge work, as well as to provide a showcase for artists who perform or create music with new electronic music instruments, controllers, and synthesizers.

Early electronic musical instruments

Emergence of electronic sound technology

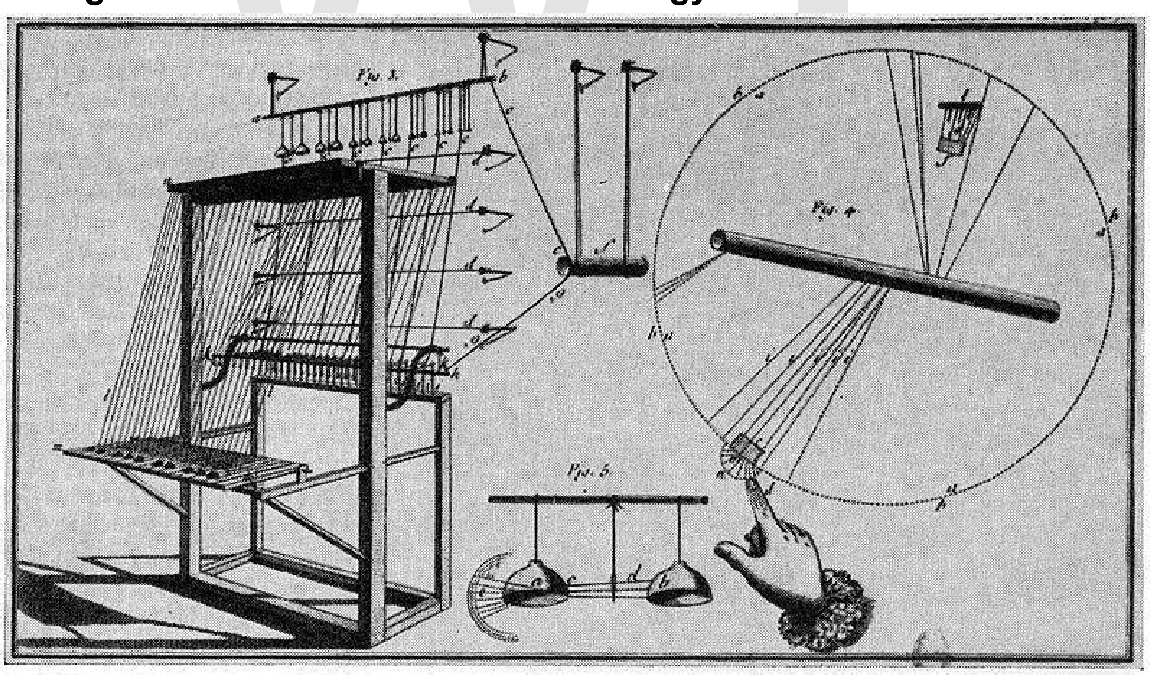


Diagram of the clavecin électrique

In the 18th-century, musicians and composers adapted a number of acoustic instruments to exploit the novelty of electricity. Thus, in the broadest sense, the first electrified musical instrument was the Denis d'or, dating from 1753, followed shortly by the Clavecin électrique by the Frenchman Jean-Baptiste de Laborde in 1761. The former instrument consisted of a keyboard instrument of over 700 strings, electrified temporarily to enhance sonic qualities. The later was a keyboard instrument with plectra (picks) activated electrically. However, neither instrument used electricity as a sound-source.

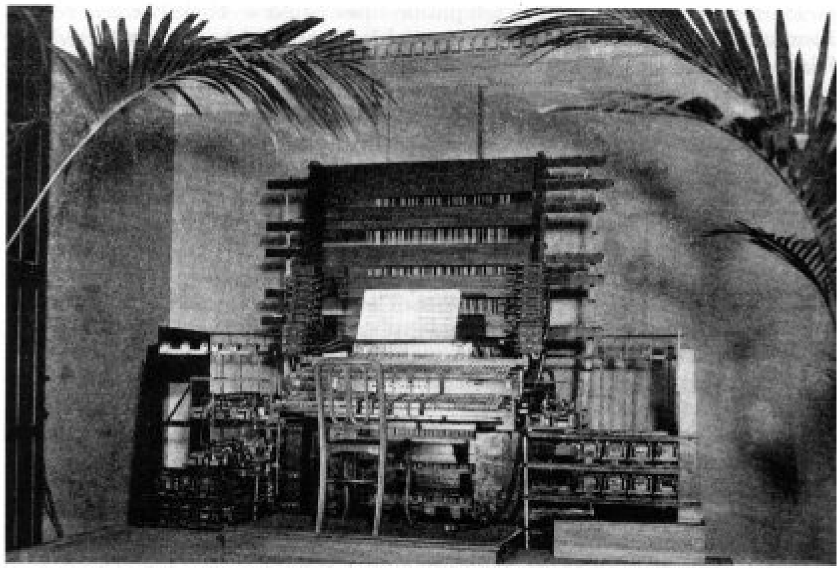
The first electric synthesizer was invented in 1876 by Elisha Gray. The "Musical Telegraph" was a chance by-product of his telephone technology when Gray accidentally discovered that he could control sound from a self-vibrating electromagnetic circuit and so invented a basic oscillator. The Musical Telegraph used steel reeds oscillated by electromagnets and transmitted over a telephone line. Gray also built a simple loudspeaker device into later models, which consisted of a diaphragm vibrating in a magnetic field.

A significant invention, which later had a profound effect on electronic music, was Lee DeForest's triode audion. This was the first thermionic valve, or vacuum tube, invented in 1906, which led to the generation and amplification of electrical signals, radio broadcasting, and electronic computation, amongst other things.

Other early synthesizers included Thaddeus Cahill's Dynamophone or Telharmononium (before 1907), Jorg Mager's Spherophone (1924) and Partiturophone, the Theremin (1927), which was marketed by RCA, Taubmann's similar Electronde (1933), the Ondes Martenot (1928), Trautwein's Trautonium (1930). The **Mellertion** (1933) used a non-standard scale, Bertrand's Dynaphone could produce octaves and perfect fifths, while the **Emicon** was an American, keyboard-controlled instrument constructed in 1930 and the German **Hellertion** combined four instruments to produce chords. Three Russian instruments also appeared, Oubouhof's Croix Sonore (1934), Ivor Darreg's microtonal 'Electronic Keyboard Oboe' (1937) and the ANS synthesizer, constructed by the Russian scientist Evgeny Murzin from 1937 to 1958. Only two models of this latter were built and the only surviving example is currently stored at the Lomonosov University in Moscow. It has been used in many Russian movies - like *Solaris* - to produce unusual, "cosmic" sounds.

Hugh Le Caine, John Hanert, Raymond Scott, composer Percy Grainger (with Burnett Cross), and others built a variety of automated electronic-music controllers during the late 1940s and 1950s. In 1959 Daphne Oram produced a novel method of synthesis, her "Oramics" technique, driven by drawings on a 35 mm film strip; it was used for a number of years at the BBC Radiophonic Workshop. This workshop was also responsible for the theme to the TV series Doctor Who, a piece, largely created by Delia Derbyshire, that more than any other ensured the popularity of electronic music in the UK.

Telharmonium



Telharmonium console by Thaddeus Cahill 1897

In 1897 Thaddeus Cahill patented an instrument called the Telharmonium (or Teleharmonium, also known as the Dynamaphone). Using tonewheels to generate musical sounds as electrical signals by additive synthesis, it was capable of producing any combination of notes and overtones, at any dynamic level. This technology was later used to design the Hammond organ. Between 1901 and 1910 Cahill had three progressively larger and more complex versions made, the first weighing seven tons, the last in excess of 200 tons. Portability was managed only by rail and with the use of thirty boxcars. By 1912, public interest had waned, and Cahill's enterprise was bankrupt.

Theremin

Theremin (1924)

Fingerboard Theremin

Another development, which aroused the interest of many composers, occurred in 1919-1920. In Leningrad, Leon Theremin (actually Lev Termen) built and demonstrated his Etherophone, which was later renamed the Theremin. This led to the first compositions for electronic instruments, as opposed to noisemakers and re-purposed machines.

Composers who ultimately utilized the Theremin included Varèse—in his piece *Ecuatorial* (1934)—while conductor Leopold Stokowski experimented with its use in arrangements from the classical repertory. In 1929, Joseph Schillinger composed *First Airphonic Suite for Theremin and Orchestra*, premièred with the Cleveland Orchestra with Leon Theremin as soloist. The next year Henry Cowell commissioned Theremin to create the first electronic rhythm machine, called the Rhythmicon. Cowell wrote some compositions for it, and he and Schillinger premiered it in 1932.

Ondes Martenot

Ondes Martenot (ca.1974, 7th generation model)

The 1920s have been called the apex of the Mechanical Age and the dawning of the Electrical Age. In 1922, in Paris, Darius Milhaud began experiments with "vocal transformation by phonograph speed change." These continued until 1927.

This decade brought a wealth of early electronic instruments—along with the Theremin, there is the presentation of the Ondes Martenot, which was designed to reproduce the microtonal sounds found in Hindu music, and the Trautonium. Maurice Martenot invented the Ondes Martenot in 1928, and soon demonstrated it in Paris. Composers using the instrument ultimately include Boulez, Honneger, Jolivet, Koechlin, Messiaen, Milhaud, Tremblay, and Varèse. Radiohead guitarist and multi-instrumentalist Jonny Greenwood also uses it in his compositions and a plethora of Radiohead songs. In 1937, Messiaen wrote *Fête des belles eaux* for 6 ondes Martenot, and wrote solo parts for it in *Trois petites Liturgies de la Présence Divine* (1943–44) and the *Turangalîla-Symphonie* (1946–48/90).

Trautonium

Volks Trautonium (1933, Telefunken Ela T 42)

The Trautonium was invented in 1928. It was based on the subharmonic scale, and the resulting sounds were often used to emulate bell or gong sounds, as in the 1950s Bayreuth productions of *Parsifal*. In 1942, Richard Strauss used it for the bell- and gong-part in the Dresden première of his *Japanese Festival Music*. This new class of instruments, microtonal by nature, was only adopted slowly by composers at first, but by the early 1930s there was a burst of new works incorporating these and other electronic instruments.

Hammond organ and Novachord

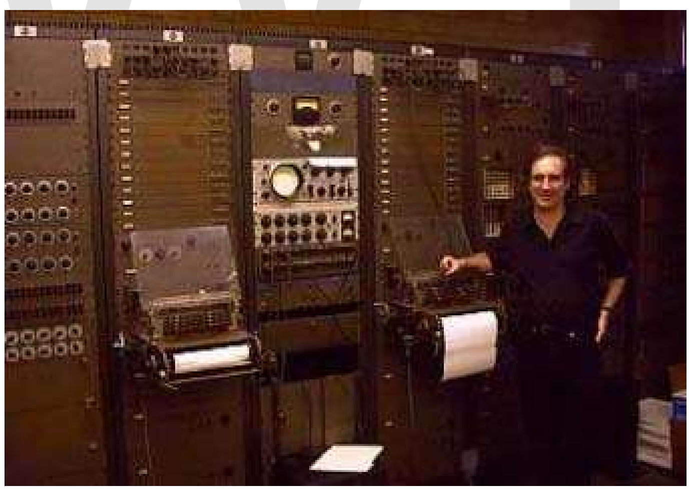
Hammond Novachord (1939)

In 1929 Laurens Hammond established his company for the manufacture of electronic instruments. He went on to produce the Hammond organ, which was based on the principles of the Telharmonium, along with other developments including early reverberation units.

The first commercially-manufactured synthesizer was the Novachord, built by the Hammond Organ Company from 1938 to 1942, which offered 72-note polyphony using 12 oscillators driving monostable-based divide-down circuits, basic envelope control and resonant low-pass filters. The instrument featured 163 vacuum tubes and weighed 500 pounds. The instrument's use of envelope control is significant, since this is perhaps the most significant distinction between the modern synthesizer and other electronic instruments.

Analogue synthesis 1950-80

Siemens Synthesizer at Siemens Studio For Electronic Music (ca.1956)



The RCA Mark II (ca.1957)

The most commonly used electronic instruments are synthesizers, so-called because they artificially generate sound using a variety of techniques. All early circuit-based synthesis involved the use of analogue circuitry, particularly voltage controlled amplifiers, oscillators and filters. An important technological development was the invention of the Clavivox synthesizer in 1956 by Raymond Scott with subassembly by Robert Moog.

Modular synthesizers

RCA produced experimental devices to synthesize voice and music in the 1950s. The Mark II Music Synthesizer, housed at the Columbia-Princeton Electronic Music Center in New York City. Designed by Herbert Belar and Harry Olson at RCA, with contributions from Vladimir Ussachevsky and Peter Mauzey, it was installed at Columbia University in 1957. Consisting of a room-sized array of interconnected sound synthesis components, it was only capable of producing music by programming, using a paper tape sequencer punched with holes to control pitch sources and filters, similar to a mechanical player piano but capable of generating a wide variety of sounds. The vacuum tube system had to be patched to create timbres.

Robert Moog

In the 1960s synthesizers were still usually confined to studios due to their size. They were usually modular in design, their stand-alone signal sources and processors connected with patch cords or by other means and controlled by a common controlling device. Harald Bode, Don Buchla, Hugh Le Caine, Raymond Scott and Paul Ketoff were among the first to build such instruments, in the late 1950s and early 1960s. Buchla later produced a commercial modular synthesizer, the Buchla Music Easel. Robert Moog, who had been a student of Peter Mauzey and one of the RCA Mark II engineers, created a synthesizer that could reasonably be used by musicians, designing the circuits while he was at Columbia-Princeton. The Moog synthesizer was first displayed at the Audio Engineering Society convention in 1964. It required experience to set up sounds but was smaller and more intuitive than what had come before, less like a machine and more like a musical instrument. Moog established standards for control interfacing, using a logarithmic 1-volt-per-octave for pitch control and a separate triggering signal. This standardization allowed synthesizers from different manufacturers to operate simultaneously. Pitch control was usually performed either with an organ-style keyboard or a music sequencer producing a timed series of control voltages. During the late 1960s hundreds of popular recordings used Moog synthesizers. Other early commercial synthesizer manufacturers included ARP, who also started with modular synthesizers before producing all-in-one instruments, and British firm EMS.

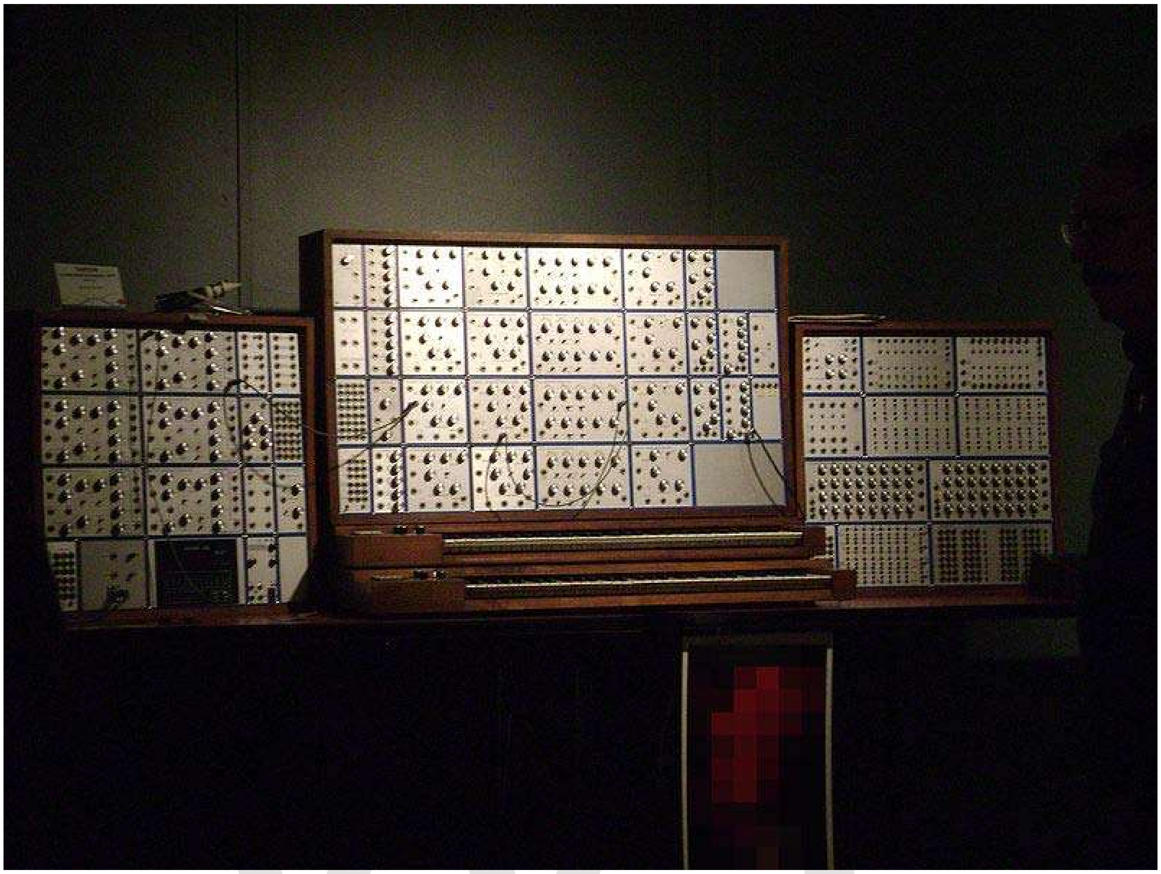
Minimoog (1970, R.A.Moog)

Integrated synthesizers

In 1970, Moog designed the Minimoog, a non-modular synthesizer with a built-in keyboard. The analogue circuits were interconnected with switches in a simplified arrangement called "normalization." Though less flexible than a modular design, normalization made the instrument more portable and easier to use. The Minimoog sold 12,000 units. further standardized the design of subsequent synthesizers with its integrated keyboard, pitch and modulation wheels and VCO->VCF->VCA signal flow. It has become celebrated for its "fat" sound - and its tuning problems. Miniaturized solid-state components allowed synthesizers to become self-contained, portable instruments that soon began to be used in live performance and quickly became the most widely used synthesizer in both popular and electronic art music.

During the late 1970s and early 1980s, DIY (Do it yourself) designs were published in hobby electronics magazines (notably the Formant modular synth, a DIY clone of the Moog system, published by Elektor) and kits were supplied by companies such as Paia in the US, and Maplin Electronics in the UK.

Yamaha GX-1 (ca.1973)



E-mu Modular System (ca.1973)

Sequential Circuits Prophet-5 (1977)

Polyphony

Many early analog synthesizers were monophonic, producing only one tone at a time. Popular monophonic synthesizers include the Moog Minimoog, and Roland SH-101. A few, such as the Moog Sonic Six, ARP Odyssey and EML 101, could produce two different pitches at a time when two keys were pressed. Polyphony (multiple simultaneous tones, which enables chords) was only obtainable with electronic organ designs at first. Popular electronic keyboards combining organ circuits with synthesizer processing included the ARP Omni and Moog's Polymoog and Opus 3.

By 1976 affordable polyphonic synthesizers began to appear, notably the Yamaha CS-50, CS-60 and CS-80, the Sequential Circuits Prophet-5 and the Oberheim Four-Voice. These remained complex, heavy and relatively costly. The recording of settings in digital memory allowed storage and recall of sounds. The first practical polyphonic synth, and the first to use a microprocessor as a controller, was the Sequential Circuits Prophet-5 introduced in late 1977. For the first time, musicians had a practical polyphonic synthesizer that allowed all knob settings to be saved in computer memory and recalled by pushing a button. The Prophet-5's design paradigm became a new standard, slowly pushing out more complex and recondite modular designs.

Tape recording

Phonogene (1953) for musique concrète

Mellotron M4000

In 1935, another significant development was made in Germany. Allgemeine Elektrizitäts Gesellschaft (AEG) demonstrated the first commercially produced magnetic tape recorder, called the "Magnetophon". Audio tape, which had the advantage of being fairly light as well as having good audio fidelity, ultimately replaced the bulkier wire recorders.

The term "electronic music" (which first came into use during the 1930s) came to include the tape recorder as an essential element: "electronically produced sounds recorded on tape and arranged by the composer to form a musical composition") It was also indispensable to Musique concrète.

Tape also gave rise to the first, analogue, sample-playback keyboards, the Chamberlin and its more famous successor the Mellotron, an electro-mechanical, polyphonic keyboard originally developed and built in Birmingham, England in the early 1960s.

Sound sequencer

One of the earliest digital sequencer, EMS Synthi Sequencer 256 (1971)

In 1951 former jazz composer Raymond Scott invented the first sequencer, which consisted of hundreds of switches controlling stepping relays, timing solenoids, tone circuits and 16 individual oscillators.

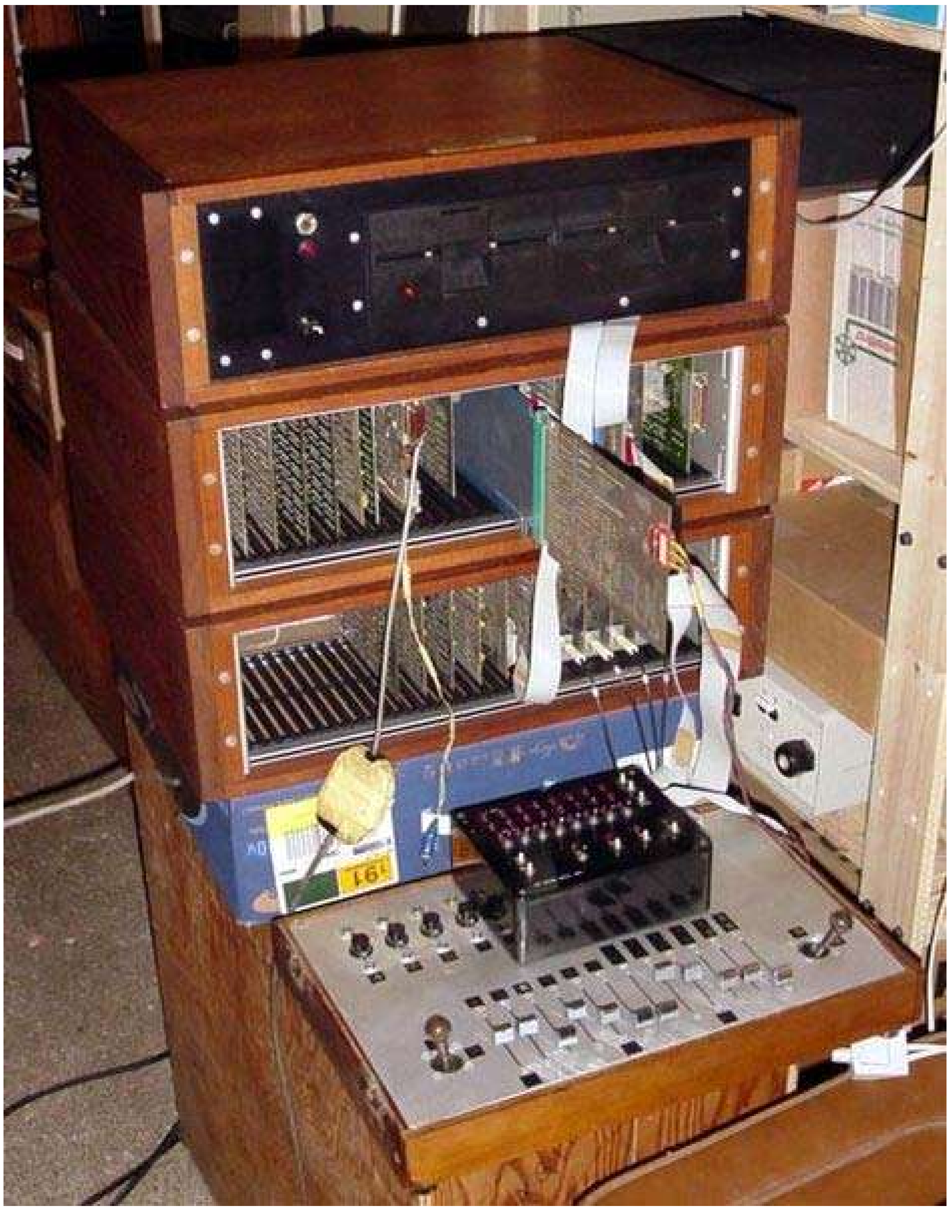
Hardware hacking

It was within this period (1966–67) that Reed Ghazala discovered and began to teach "circuit bending"—the application of the creative short circuit, a process of chance short-circuiting, creating experimental electronic instruments, exploring sonic elements mainly of timbre and with less regard to pitch or rhythm, and influenced by John Cage’s aleatoric music concept.

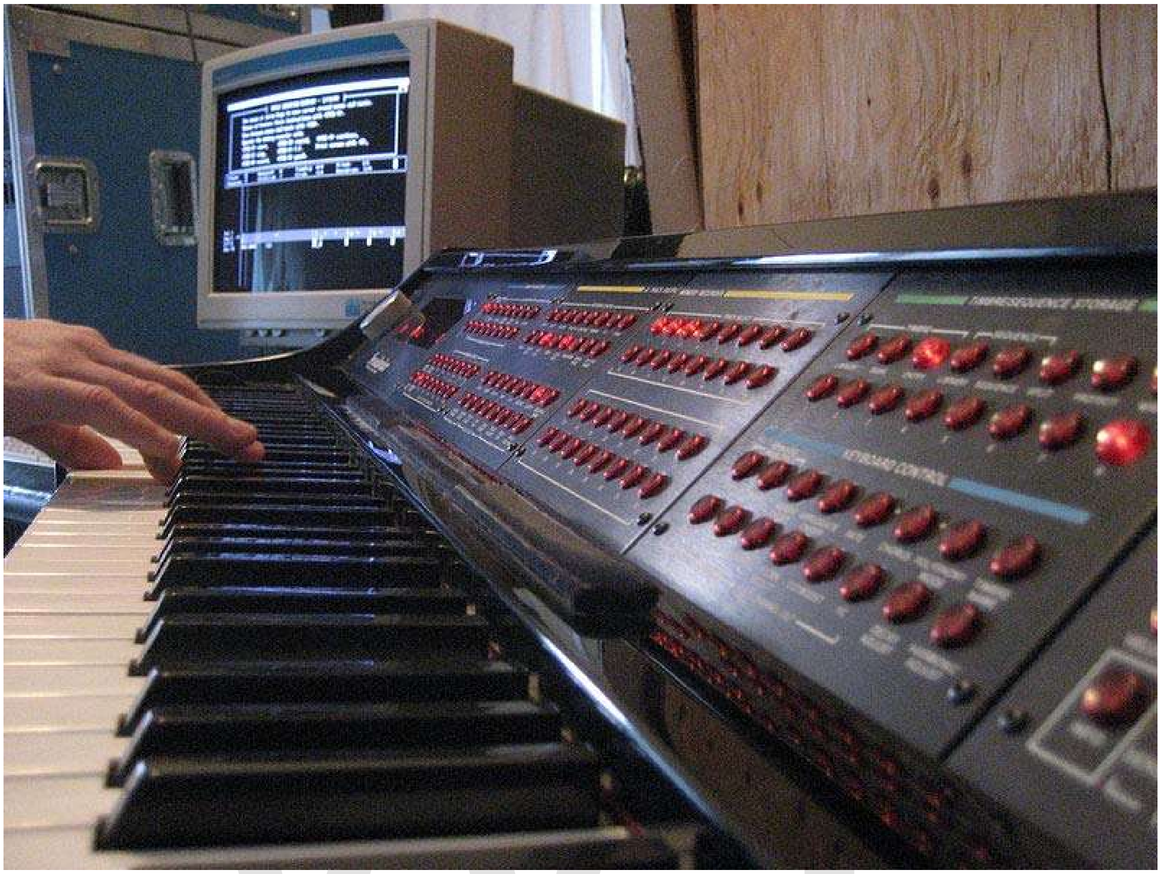
Much of this manipulation of circuits directly, especially to the point of destruction, was pioneered by Louis and Bebe Barron in the early 1950s, such as their work with John Cage on the Williams Mix and especially in the soundtrack to Forbidden Planet.

The digital era 1980-2000

Digital synthesis



Synclavier I (1977)



Synclavier PSMT (1984)

Yamaha GS-1 (1980)

Yamaha DX7 (1983)

The first digital synthesizers were academic experiments in sound synthesis using digital computers. FM synthesis was developed for this purpose, as a way of generating complex sounds digitally with the smallest number of computational operations per sound sample. In 1983 Yamaha introduced the first stand-alone digital synthesizer, the DX-7. It used frequency modulation synthesis (FM synthesis), first developed by John Chowning at Stanford University during the late sixties. Chowning exclusively licensed his FM synthesis patent to Yamaha in 1975. Yamaha subsequently released their first FM synthesizers, the GS-1 and GS-2, which were costly and heavy. There followed a pair of smaller, preset versions, the CE20 and CE25 Combo Ensembles, targeted primarily at the home organ market and featuring four-octave keyboards. Yamaha's third generation of digital synthesizers was a commercial success; it consisted of the DX7 and DX9 (1983). Both models were compact, reasonably priced, and dependent on custom digital integrated circuits to produce FM tonalities. The DX7 was the first mass market all-digital synthesizer. It became indispensable to many music artists of the 1980s, and demand soon exceeded supply. The DX7 sold over 200,000 units within three years.

The DX series was not easy to program but offered a detailed, percussive sound that led to the demise of the electro-mechanical Rhodes piano. Following the success of FM synthesis Yamaha signed a contract with Stanford University in 1989 to develop digital waveguide synthesis, leading to the first commercial physical modeling synthesizer, Yamaha's VL-1, in 1994.

Sampling

A Fairlight CMI keyboard (1979)

Kurzweil K250 (1984)

The Fairlight CMI (Computer Musical Instrument), the first polyphonic digital sampler, was the harbinger of sample-based synthesizers. Designed in 1978 by Peter Vogel and Kim Ryrie and based on a dual microprocessor computer designed by Tony Furse in Sydney, Australia, the Fairlight CMI gave musicians the ability to modify volume, attack, decay, and use special effects like vibrato. Sample waveforms could be displayed on-screen and modified using a light pen. The Synclavier from New England Digital was a similar system. Jon Appleton (with Jones and Alonso) invented the Dartmouth Digital Synthesizer, later to become the New England Digital Corp's Synclavier. The Kurzweil K250, first produced in 1983, was also a successful polyphonic digital music synthesizer, noted for its ability to reproduce several instruments synchronously and having a velocity-sensitive keyboard.

Computer music

Max Mathews (1970s) playing realtime software instrument.

ISPW, a succesor of 4X, was a DSP platform based on i860 and NeXT, by IRCAM.

An important new development was the advent of computers for the purpose of composing music, as opposed to manipulating or creating sounds. Iannis Xenakis began what is called "musique stochastique," or "stochastic music", which is a method of composing that employs mathematical probability systems. Different probability algorithms were used to create a piece under a set of parameters. Xenakis used graph paper and a ruler to aid in calculating the velocity trajectories of glissandi for his orchestral composition *Metastasis* (1953–54), but later turned to the use of computers to compose pieces like *ST/4* for string quartet and *ST/48* for orchestra (both 1962).

The impact of computers continued in 1956. Lejaren Hiller and Leonard Isaacson composed *Iliac Suite* for string quartet, the first complete work of computer-assisted composition using algorithmic composition.

In 1957, Max Mathews at Bell Lab wrote MUSIC-N series, a first computer program family for generating digital audio waveforms through direct synthesis. Then Barry

Vercoe wrote MUSIC 11 based on MUSIC IV-BF, a next-generation music synthesis program (later evolving into csound, which is still widely used).

In mid 80s, Miller Puckette at IRCAM developed graphic signal-processing software for 4X called Max (after Max Mathews), and later ported it to Macintosh (with Dave Zicarelli extending it for Opcode) for real-time MIDI control, bringing algorithmic composition availability to most composers with modest computer programming background.

MIDI

MIDI, a LAN for music, enables connections between digital musical instruments

In 1980, a group of musicians and music merchants met to standardize an interface by which new instruments could communicate control instructions with other instruments and the prevalent microcomputer. This standard was dubbed MIDI (Musical Instrument Digital Interface). A paper was authored by Dave Smith of Sequential Circuits and proposed to the Audio Engineering Society in 1981. Then, in August 1983, the MIDI Specification 1.0 was finalized.

The advent of MIDI technology allows a single keystroke, control wheel motion, pedal movement, or command from a microcomputer to activate every device in the studio remotely and in synchrony, with each device responding according to conditions predetermined by the composer.

MIDI instruments and software made powerful control of sophisticated instruments easily affordable by many studios and individuals. Acoustic sounds became reintegrated into studios via sampling and sampled-ROM-based instruments.

Modern electronic musical instruments

Wind synthesizer

SynthAxe

The increasing power and decreasing cost of sound-generating electronics (and especially of the personal computer), combined with the standardization of the MIDI and Open Sound Control musical performance description languages, has facilitated the separation of musical instruments into music controllers and music synthesizers.

By far the most common musical controller is the musical keyboard. Other controllers include the radiodrum, Akai's EWI, the guitar-like SynthAxe, the BodySynth, the Buchla Thunder, the Continuum Fingerboard, the Roland Octapad, various isomorphic keyboards including the Thummer, and Kaossilator Pro, and kits like I-CubeX.

The reactable

Reactable

The reactable is a round translucent table, used in a darkened room, and appears as a backlit display. By placing blocks called *tangibles* on the table, and interfacing with the visual display via the tangibles or fingertips, a virtual modular synthesizer is operated, creating music or sound effects.

Percussa AudioCubes

AudioCubes are autonomous wireless cubes powered by an internal computer system and rechargeable battery. They have internal RGB lighting, and are capable of detecting each other's location, orientation and distance. The cubes can also detect distances to the user's hands and fingers. Through interaction with the cubes, a variety of music and sound software can be operated. AudioCubes have applications in sound design, music production, DJing and live performance.

Kaossilator

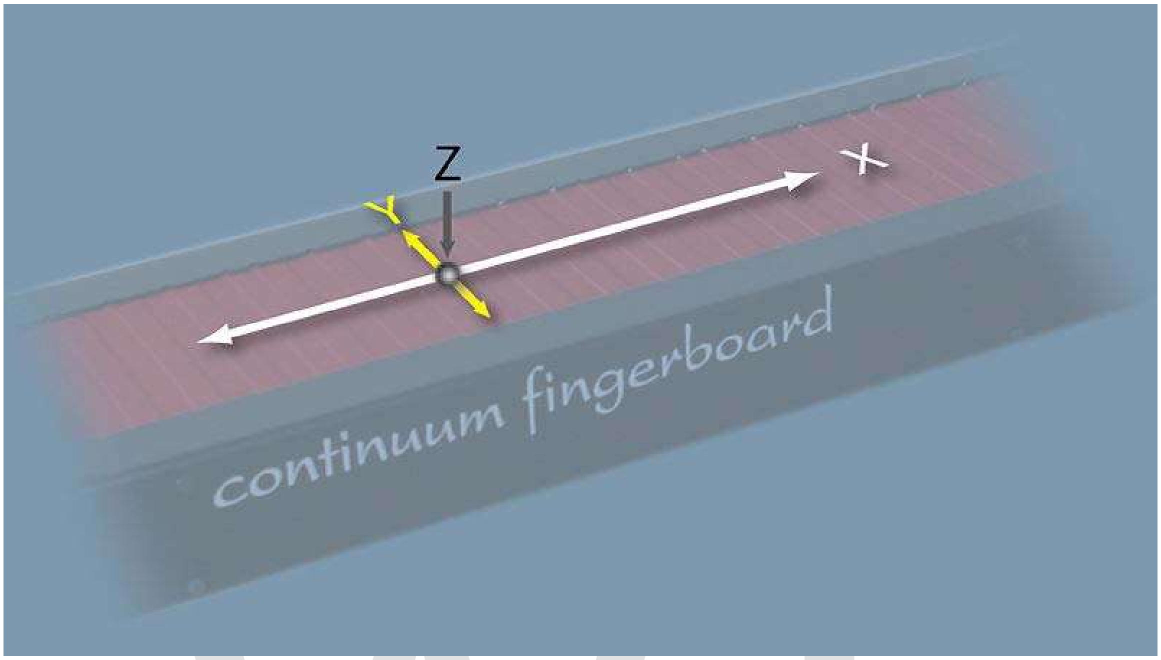
Korg Kaossilator

The Kaossilator and Kaossilator Pro are compact instruments where the position of a finger on the touch pad controls two note-characteristics; usually the pitch is changed with a left-right motion and the tonal property, filter or other parameter changes with an up-down motion. The touch pad can be set to different musical scales and keys. The instrument can record a repeating loop of adjustable length, set to any tempo, and new loops of sound can be layered on top of existing ones. This lends itself to electronic dance-music but is more limited for controlled sequences of notes, as the pad on a regular Kaossilator is featureless.

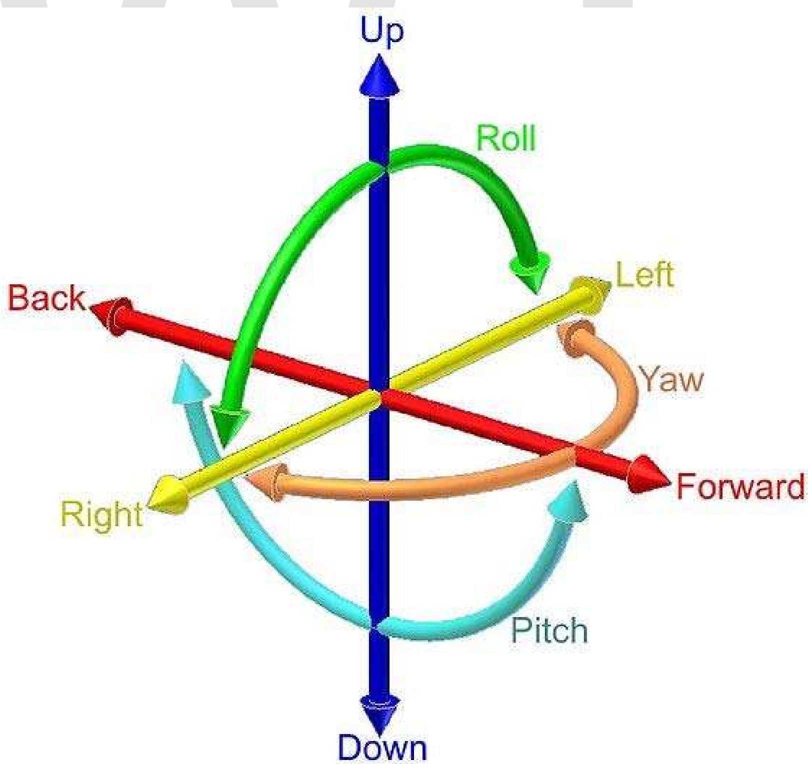
Eigenharp

The Eigenharp is a large instrument resembling a bassoon, which can be interacted with through touch-sensitive buttons, a drum sequencer and a mouthpiece. The sound processing is done on a separate computer.

Future electronic musical instruments



three degree of freedom on Continuum



Six degrees of freedom

According to a forum post in December 2010, Sixense Entertainment is working on musical control with the Sixense TrueMotion motion controller.

Immersive virtual musical instruments, or immersive virtual instruments for music and sound aim to represent musical events and sound parameters in a virtual reality so that they can be perceived not only through auditory feedback but also visually in 3D and possibly through tactile as well as haptic feedback, allowing the development of novel interaction metaphors beyond manipulation such as prehension.

Chapter 6

Amplifier

Generally, an **amplifier** or simply **amp**, is a device for increasing the power of a signal.

In popular use, the term usually describes an electronic amplifier, in which the input "signal" is usually a voltage or a current. In audio applications, amplifiers drive the loudspeakers used in PA systems to make the human voice louder or play recorded music. Amplifiers may be classified according to the input (source) they are designed to amplify (such as a guitar amplifier, to perform with an electric guitar), the device they are intended to drive (such as a headphone amplifier), the frequency range of the signals (Audio, IF, RF, and VHF amplifiers, for example), whether they invert the signal (inverting amplifiers and non-inverting amplifiers), or the type of device used in the amplification (valve or tube amplifiers, FET amplifiers, etc.).

A related device that emphasizes conversion of signals of one type to another (for example, a light signal in photons to a DC signal in amperes) is a transducer, a transformer, or a sensor. However, none of these amplify power.

Figures of merit

The quality of an amplifier can be characterized by a number of specifications, listed below.

Gain

The gain of an amplifier is the ratio of output to input power or amplitude, and is usually measured in decibels. (When measured in decibels it is logarithmically related to the power ratio: $G(\mathrm{dB})=10\log(P_{out}/(P_{in}))$. RF amplifiers are often specified in terms of the maximum **power gain** obtainable, while the voltage gain of audio amplifiers and instrumentation amplifiers will be more often specified (since the amplifier's input impedance will often be much higher than the source impedance, and the load impedance higher than the amplifier's output impedance).

- Example: an audio amplifier with a gain given as 20 dB will have a *voltage gain* of ten (but a power gain of 100 would only occur in the unlikely event the input and output impedances were identical).

If two equivalent amplifiers are being compared, the amplifier with higher gain settings would be more sensitive as it would take less input signal to produce a given amount of power.

Bandwidth

The bandwidth of an amplifier is the range of frequencies for which the amplifier gives "satisfactory performance". The definition of "satisfactory performance" may be different for different applications. However, a common and well-accepted metric is the half power points (i.e. frequency where the power goes down by half its peak value) on the output vs. frequency curve. Therefore bandwidth can be defined as the difference between the lower and upper half power points. This is therefore also known as the −3 dB bandwidth. Bandwidths (otherwise called "frequency responses") for other response tolerances are sometimes quoted (−1 dB, −6 dB etc.) or "plus or minus 1dB" (roughly the sound level difference people usually can detect).

The gain of a good quality full-range audio amplifier will be essentially flat between 20 Hz to about 20 kHz (the range of normal human hearing). In ultra high fidelity amplifier design, the amp's frequency response should extend considerably beyond this (one or more octaves either side) and might have −3 dB points < 10 and > 65 kHz. Professional touring amplifiers often have input and/or output filtering to sharply limit frequency response beyond 20 Hz-20 kHz; too much of the amplifier's potential output power would otherwise be wasted on infrasonic and ultrasonic frequencies, and the danger of AM radio interference would increase. Modern switching amplifiers need steep low pass filtering at the output to get rid of high frequency switching noise and harmonics.

Efficiency

Efficiency is a measure of how much of the power source is usefully applied to the amplifier's output. Class A amplifiers are very inefficient, in the range of 10–20% with a max efficiency of 25% for direct coupling of the output. Inductive coupling of the output can raise their efficiency to a maximum of 50%.

Class B amplifiers have a very high efficiency but are impractical for audio work because of high levels of distortion (See: Crossover distortion). In practical design, the result of a tradeoff is the class AB design. Modern Class AB amplifiers commonly have peak efficienies between 30–55% in audio systems and 50-70% in radio frequency systems with a theoretical maximum of 78.5%.

Commercially available Class D switching amplifiers have reported efficiencies as high as 90%. Amplifiers of Class C-F are usually known to be very high efficiency amplifiers.

RCA manufactured an AM broadcast transmitter employing a single class-C low mu triode with an RF efficiency in the 90% range.

More efficient amplifiers run cooler, and often do not need any cooling fans even in multi-kilowatt designs. The reason for this is that the loss of efficiency produces heat as a by-product of the energy lost during the conversion of power. In more efficient amplifiers there is less loss of energy so in turn less heat.

In RF linear Power Amplifiers, such as cellular base stations and broadcast transmitters, special design techniques can be used to improve efficiency. Doherty designs, which use a second output stage as a "peak" amplifier, can lift efficiency from the typical 15% up to 30-35% in a narrow bandwidth. Envelope Tracking designs are able to achieve efficiencies of up to 60%, by modulating the supply voltage to the amplifier in line with the envelope of the signal.

Linearity

An ideal amplifier would be a totally linear device, but real amplifiers are only linear within limits.

When the signal drive to the amplifier is increased, the output also increases until a point is reached where some part of the amplifier becomes saturated and cannot produce any more output; this is called clipping, and results in distortion.

In most amplifiers a reduction in gain takes place before hard clipping occurs; the result is a *compression* effect, which (if the amplifier is an audio amplifier) sounds much less unpleasant to the ear. For these amplifiers, the 1 dB compression point is defined as the input power (or output power) where the gain is 1 dB less than the small signal gain. Sometimes this nonlinearity is deliberately designed in to reduce the audible unpleasantness of hard clipping under overload.

Ill effects of nonlinearity can be reduced with negative feedback.

Linearization is an emergent field, and there are many techniques, such as feedforward, predistortion, postdistortion, in order to avoid the undesired effects of the non-linearities.

Noise

This is a measure of how much noise is introduced in the amplification process. Noise is an undesirable but inevitable product of the electronic devices and components; also, much noise results from intentional economies of manufacture and design time. The metric for noise performance of a circuit is noise figure or noise factor. Noise figure is a comparison between the output signal to noise ratio and the thermal noise of the input signal.

Output dynamic range

Output dynamic range is the range, usually given in dB, between the smallest and largest useful output levels. The lowest useful level is limited by output noise, while the largest is limited most often by distortion. The ratio of these two is quoted as the amplifier dynamic range. More precisely, if S = maximal allowed signal power and N = noise power, the dynamic range DR is $DR = (S + N) / N$.

In many switched mode amplifiers, dynamic range is limited by the minimum output step size.

Slew rate

Slew rate is the maximum rate of change of the output, usually quoted in volts per second (or microsecond). Many amplifiers are ultimately slew rate limited (typically by the impedance of a drive current having to overcome capacitive effects at some point in the circuit), which sometimes limits the full power bandwidth to frequencies well below the amplifier's small-signal frequency response.

Rise time

The rise time, t_r, of an amplifier is the time taken for the output to change from 10% to 90% of its final level when driven by a step input. For a Gaussian response system (or a simple RC roll off), the rise time is approximated by:

$\mathbf{t_r * BW = 0.35}$, where t_r is rise time in seconds and BW is bandwidth in Hz.

Settling time and ringing

The time taken for the output to settle to within a certain percentage of the final value (for instance 0.1%) is called the settling time, and is usually specified for oscilloscope vertical amplifiers and high accuracy measurement systems. Ringing refers to an output variation that cycles above and below an amplifier's final value and leads to a delay in reaching a stable output. Ringing is the result of overshoot caused by an underdamped circuit.

Overshoot

In response to a step input, the overshoot is the amount the output exceeds its final, steady-state value.

Stability

Stability is an issue in all amplifiers with feedback, whether that feedback is added intentionally or results unintentionally. It is especially an issue when applied over multiple amplifying stages.

Stability is a major concern in RF and microwave amplifiers. The degree of an amplifier's stability can be quantified by a so-called stability factor. There are several different stability factors, such as the Stern stability factor and the Linvil stability factor, which specify a condition that must be met for the absolute stability of an amplifier in terms of its two-port parameters.

Electronic amplifiers

There are many types of electronic amplifiers, commonly used in radio and television transmitters and receivers, high-fidelity ("hi-fi") stereo equipment, microcomputers and other electronic digital equipment, and guitar and other instrument amplifiers. Critical components include active devices, such as vacuum tubes or transistors.

Other amplifier types

Carbon microphone

One of the first devices used to amplify signals was the carbon microphone (effectively a sound-controlled variable resistor). By channeling a large electric current through the compressed carbon granules in the microphone, a small sound signal could produce a much larger electric signal. The carbon microphone was extremely important in early telecommunications; analog telephones in fact work without the use of any other amplifier. Before the invention of electronic amplifiers, mechanically coupled carbon microphones were also used as amplifiers in telephone repeaters for long distance service.

Magnetic amplifier

A **magnetic amplifier** is a transformer-like device that makes use of the saturation of magnetic materials to produce amplification. It is a non-electronic electrical amplifier with no moving parts. The bandwidth of magnetic amplifiers extends to the hundreds of kilohertz.

Rotating electrical machinery amplifier

A Ward Leonard control is a rotating machine like an electrical generator that provides amplification of electrical signals by the conversion of mechanical energy to electrical energy. Changes in generator field current result in larger changes in the output current of the generator, providing gain. This class of device was used for smooth control of large motors, primarily for elevators and naval guns.

Field modulation of a very high speed AC generator was also used for some early AM radio transmissions.

Johnsen-Rahbek effect amplifier

The earliest form of audio power amplifier was Edison's "electromotograph" loud-speaking telephone, which used a wetted rotating chalk cylinder in contact with a stationary contact. The friction between cylinder and contact varied with the current, providing gain. Edison discovered this effect in 1874, but the theory behind the Johnsen-Rahbek effect was not understood until the semiconductor era.

Mechanical amplifiers

Mechanical amplifiers were used in the pre-electronic era in specialized applications.

Early **autopilot** units designed by Elmer Ambrose Sperry incorporated a mechanical amplifier using belts wrapped around rotating drums; a slight increase in the tension of the belt caused the drum to move the belt. A paired, opposing set of such drives made up a single amplifier. This amplified small gyro errors into signals large enough to move aircraft control surfaces. A similar mechanism was used in the Vannevar Bush differential analyzer.

The **Electrostatic drum amplifier** used a band wrapped partway around a rotating drum, and fixed at its anchored end to a spring. The other end connected to a speaker cone. The input signal was transformed up to high voltage, and added to a high voltage dc supply line. This voltage was connected between drum and belt. Thus the input signal varied the electric field between belt and drum, and thus the friction between them, and thus the amount of lateral movement of the belt and thus speaker cone.

Other variations on the theme also existed at one time.

Optical amplifiers

Optical amplifiers amplify light through the process of stimulated emission.

Miscellaneous types

- There are also mechanical amplifiers, such as the automotive servo used in braking.
- Relays can be included under the above definition of amplifiers, although their transfer function is not linear (that is, they are either open or closed).
- Also purely mechanical manifestations of such digital amplifiers can be built (for theoretical, didactical purposes, or for entertainment).
- Another type of amplifier is the fluidic amplifier, based on the fluidic triode.

Chapter 7

Reel-to-Reel Audio Tape Recording

A reel-to-reel tape recorder (Sony TC-630), typical of those which were once common audiophile objects. Note the distinctive Scotch tape spool at left.

Reel-to-reel, **open reel** tape recording is the form of magnetic tape audio recording in which the recording medium is held on a reel, rather than being securely contained within a cassette.

In use, the *supply reel* or *feed reel* containing the tape is mounted on a spindle; the end of the tape is manually pulled out of the reel, threaded through mechanical guides and a tape head assembly, and attached by friction to the hub of a second, initially empty *takeup reel*. The arrangement is similar to that used for motion picture film.

History

Magnetophon from a German radio station in World War II.

The reel-to-reel format was used in the very earliest tape recorders, including the pioneering German Magnetophon machines of the 1930s. Originally, this format had no name, since all forms of magnetic tape recorders used it. The name arose only with the need to distinguish it from the several kinds of tape cartridges or cassettes which were introduced in the early 1960s.

Reel-to-reel tape was also used in early tape drives for data storage on mainframe computers, video tape machines, and later for high quality analog audio recorders as early as the late 1940s, up until modern day studios where it is still in use. Studer, Stellavox, Nagra, Denon and Otari currently manufacture analog reel-to-reel recorders.

The earliest machines produced distortion during the recording process which German engineers significantly reduced during the Nazi era by applying a high-frequency bias current to the recording head along with the desired signal. American audio engineer Jack Mullin was a member of the U.S. Army Signal Corps during World War II. His unit was assigned to investigate German radio and electronics activities, and in the course of his duties, he acquired two Magnetophon recorders and 50 reels of I.G. Farben recording tape from a German radio station at Bad Nauheim (near Frankfurt). He had these shipped home. Over the next two years, he worked to develop the machines for commercial use, hoping to interest the Hollywood film studios in using magnetic tape for movie soundtrack recording.

Unitra ZK-147, a vintage Polish-made reel-to-reel tape recorder

Mullin gave a demonstration of his recorders at MGM Studios in Hollywood in 1947, which led to a meeting with Bing Crosby, who immediately saw the potential of Mullin's recorders to pre-record his radio shows. Crosby invested $50,000 in a local electronics company, Ampex, to enable Mullin to develop a commercial production model of the tape recorder. Using Mullin's tape recorders and with Mullin as his chief engineer, Crosby became the first American performer to master commercial recordings on tape and the first to regularly pre-record his radio programs on the medium. Ampex and Mullin subsequently developed commercial stereo and multitrack audio recorders, based on the system invented by Ross Snyder of Ampex Corp. Les Paul had been given one of the first Ampex Model 200 tape decks by Crosby in 1948 and went on to use Ampex eight track "Sel Sync" machines for multitracking. Ampex went on to develop the first practical videotape recorders in the early 1950s to pre-record Crosby's TV shows.

7-inch reel of ¼-inch-wide (6.4 mm) recording tape, typical of non-professional use in the 1950s–70s. Studios generally used 10½ inch reels on PET film backings.

Inexpensive reel-to-reel tape recorders were widely used for voice recording in the home and in schools before the Philips compact cassette, introduced in 1963, gradually took over. Cassettes eventually displaced reel-to-reel recorders for consumer use. However, the narrow tracks and slow recording speeds used in cassettes compromised fidelity.

Following the example set by Bing Crosby, high-speed reel-to-reel tape recorders rapidly became the main recording format used by audiophiles and professional recording studios until the late 1980s when digital audio recording techniques began to allow the use of other types of media (such as Digital Audio Tape (DAT) cassettes and hard disks).

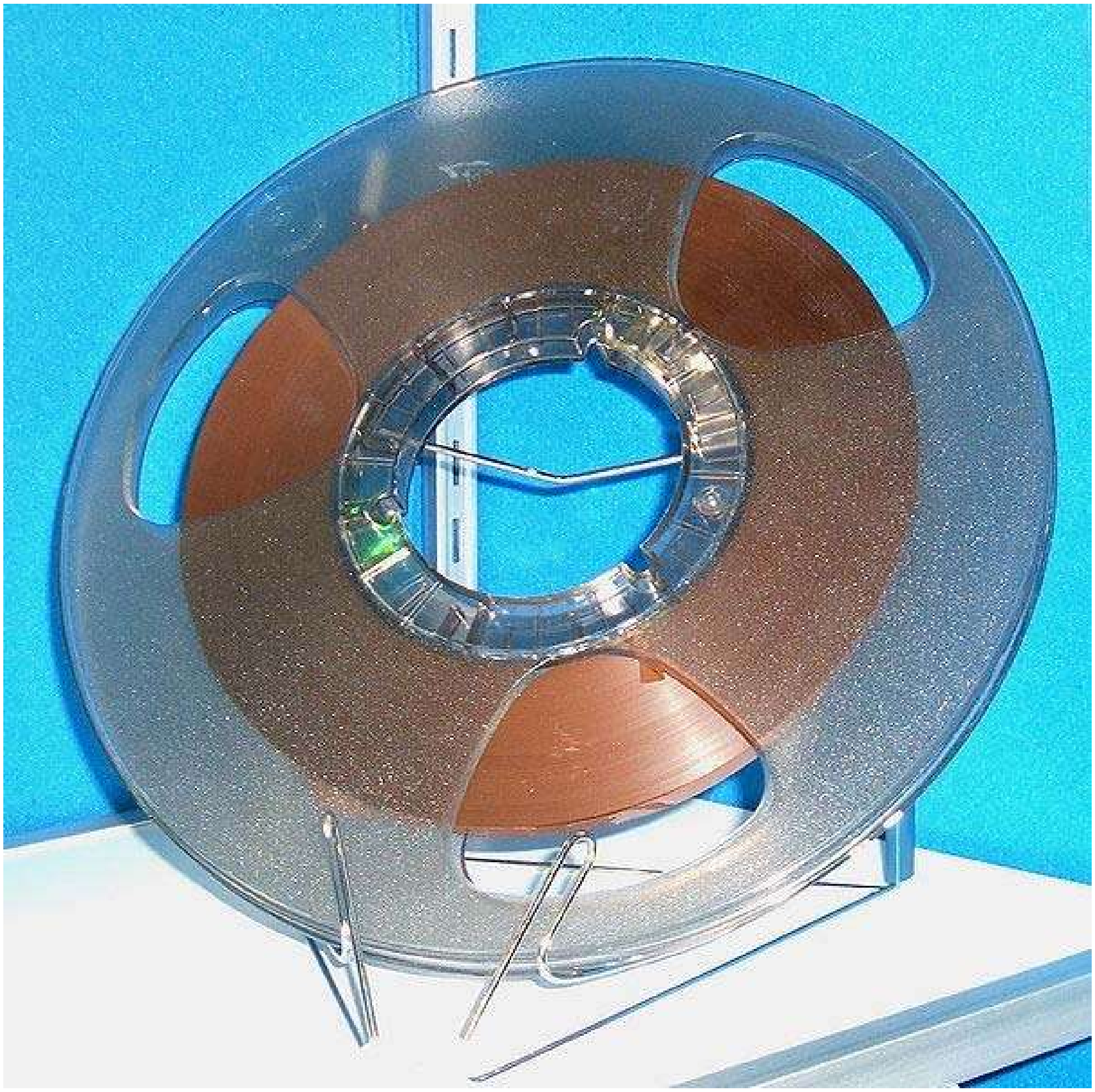
Even today, some artists of all genres prefer analog tape's "musical", "natural" and especially "warm" sound. Due to harmonic distortion, bass can thicken up, creating the illusion of a fuller-sounding mix. In addition, high end can be slightly compressed, which is more natural to the human ear. It is common for artists to record to digital and re-record the tracks to analog reels for this effect of "natural" sound. In addition to all of these attributes of tape, tape saturation is a unique form of distortion that many rock and blues artists find very pleasing.

Euphonic distortion and noise levels aside, high-quality analog tape currently outstrips the transparency of all but the best digital recording/playback systems: digital systems can suffer from (among other problems) clock jitter, inferior analog circuitry, inferior digital filter design, improper wordlength conversion, and/or lack of correct dithering. Dramatic improvements in the average quality of digital hardware design are narrowing the gap, though, and might soon eliminate the quality distinction altogether.

Description

The earliest reel-to-reel systems used metal wire as a medium, which is robust, but suffered from a number of problems – poor fidelity, required a strong current to imprint the signal onto the wire, editing inconvenience (needing physical cuts to effect an edit), and potential kinking or even tangling of the recording wire. The invention of cellulose acetate plastic tape coated with iron oxide solved these problems, opening up the use of tape recorders in studios.

The great advantage of tape for studios was twofold – it allowed a performance to be recorded without the 30 minute time limitation of a phonograph disc, and it permitted a recorded performance to be edited. For the first time, audio could be manipulated as a physical entity. Tape editing is performed simply by cutting the tape at the required point, and rejoining it to another section of tape using adhesive tape, or sometimes glue. This is called a *splice*. The splicing tape has to be very thin to avoid impeding the tape's motion, and the adhesive is carefully formulated to avoid leaving a sticky residue on the tape or deck. Usually, the cut is made at an angle across the tape so that any "click" or other noise introduced by the cut is spread across a few milliseconds of the recording. The use of reels to supply and collect the tape also made it very easy for editors to manually move the tape back and forth across the heads to find the exact point they wished to edit. Tape to be spliced was clamped in a special *splicing block* attached to the deck near the heads to hold the tape accurately while the edit was made. A skilled editor could make these edits very rapidly and accurately. A side effect of cutting the tape at an angle is that on stereo tapes the edit occurs on one channel a split-second before the other.



Professional-style tape reel designed to fit large NARTB hub.

The performance of tape recording is greatly affected by the width of the tracks used to record a signal, and the speed of the tape. The wider and faster the better, but of course this uses more tape. These factors lead directly to improved frequency response, signal-to-noise ratio, and high-frequency distortion figures. Tape can accommodate multiple parallel tracks, allowing not just stereo recordings, but multi-track recordings too. This gives the producer of the final edit much greater flexibility, allowing a performance to be remixed long after the performance was originally recorded. This innovation was a great driving force behind the explosion of popular music in the late 1950s and 1960s. The first multi-tracking recorders had four tracks, then eight, then sixteen, twenty-four, and so on. It was also discovered that new effects were possible using multi-tracking recorders, such as phasing and flanging, delays and echo, so these innovations appeared on pop recordings shortly after multi-tracking recorders were introduced.

A typical home reel to reel tape recorder, this one made by Sonora. It could play stereo quarter track tapes, but record only in one quarter track mono. Home equipment with missing features were fairly common in the 1950s and 1960s.

For home use, simpler reel-to-reel recorders were available, and a number of track formats and tape speeds were standardised to permit interoperability and prerecorded music.

Reel-to-reel tape editing also gained cult-status when many used this technique on hit-singles in the 1980s.

Pre-recorded Reel Tapes

The first prerecorded reel-to-reel tapes were introduced by RCA Victor in 1954. The heyday of prerecorded reel tapes was the mid-1960s, but after the introduction of less complicated cassette tapes and 8 track tapes, the number of albums released on prerecorded reel tape dropped dramatically despite their superior sound quality. By the latter 1960s, their retail prices were considerably higher than competing formats, and musical genres were limited—classical, soundtracks, original cast albums, major pop stars—to those most likely to appeal to audiophiles willing to contend with the cumbersome threading of open-reel tape. The introduction of the Dolby noise-reduction system narrowed the performance gap between cassettes and open-reel, and by 1973 the prerecorded open-reel offerings had almost completely disappeared, even from record stores and audio equipment shops. Columbia House continued to offer a select number of new releases in the format for its club members until 1981.

Current manufacturers

Reel-to-reel recorders

Otari makes the MX5050 1/4" recorder.

Denon makes the broadcast-oriented DN-3602RG 1/4" recorder for Asian markets.

Nagra makes the 4.2 portable 1/4" recorder available in several different versions for film and radio use.

Stellavox makes the modular TD-9 1/4" recorder and the portable SD-9 1/4" recorder.

Tape

When Ampex broke apart in the 1990s, Quantegy Inc. was formed, later becoming Quantegy Recording Solutions in 2004. Quantegy (and formerly Ampex) led the field in reel-to-reel technology, and Quantegy was the only company left making reel-to-reel tape in the world for a period of two years. In 2007, Reel Deal Pro Audio purchased the majority of Quantegy's reel to reel audio tape and accessories and began to sell it on their web site.

In 2006, Recorded Media Group International (RMGI) in the Netherlands began manufacturing EMTEC specification tape in Oosterhout and is now the largest open reel tape manufacturer in the world.

ATR Magnetics of York, PA, longtime service and modification shop for multitrack and master recorders, began manufacturing analog multitrack tape, and in November 2006 began beta testing a new formula.

Jai Electronic Industries in India are currently making audio tape in 6.35 mm(1/4") and 12.7 mm(1/2") width, and perforated 16 mm and 35 mm audio tape for the film industry.

Daniel Technology in the USA are making 3.81 mm tape for the Nagra SN-series tape recorders.

Pyral in France are making perforated 16 mm, 17.5 mm and 35 mm audio tape.

Tape speeds

In general, the faster the speed the better the sound quality. In addition to faithfully recording higher frequencies and increasing the magnetic signal strength and therefore the signal-to-noise ratio (S/N), higher tape speeds spread the signal longitudinally over more tape area, reducing the effects of defects in or damage to the medium. Slower speeds conserve tape and are useful in applications where sound quality is not critical.

- 15/16ths of an inch per second (in/s) or 2.38 cm/s — used for very long-duration recordings (e.g. recording a radio station's entire output in case of complaints, aka "logging")
- 1⅞ in/s or 4.76 cm/s — usually the slowest domestic speed, best for long duration speech recordings
- 3¾ in/s or 9.52 cm/s — common domestic speed, used on most single-speed domestic machines, reasonable quality for speech and off-air radio recordings
- 7½ in/s or 19.05 cm/s — highest domestic speed, also slowest professional; used by most radio stations for "dubs", copies of commercial announcements; Through the early-mid '90s many stations could not handle 15 IPS.
- 15 in/s or 38.1 cm/s — professional music recording and radio programming
- 30 in/s or 76.2 cm/s — used where the best possible treble response is demanded, e.g., many classical music recordings

Speed units of inches per second or in/s are also abbreviated IPS. 3¾ in/s and 7½ in/s are the speeds that were used for (the vast majority of) consumer market releases of commercial recordings on reel-to-reel tape. 3¾ in/s is also the speed used in 8-track cartridges. 1⅞ in/s is also the speed used in Compact cassettes.)

In some early prototype linear video tape recording systems developed in the early 1950s from companies such as Bing Crosby Enterprises, RCA, and the BBC's VERA, the reel speed was extremely high, over 200 in/s, to adequately capture the large amount of image information. The need for a high linear tape speed was made unnecessary with the introduction of the now-obsolete professional Quadruplex system from 1956, which segmented the fields of a television image by recording (and reproducing) several tracks at a high-speed across the width of the tape per field of video by way of a spinning headwheel with 4 separate video heads mounted on its edge (a technique called *transverse scanning*), allowing for the linear tape speed to be much slower. Transverse scanning was superseded by the later technology of helical scanning, which could record one whole field of video per helically-recorded track, recorded at an angle across the width of the tape.

Quality aspects

Even though a recording on tape may have been made at studio quality, tape speed was the limiting factor, much like sample rate is today. Decreasing the speed of analog audio tape causes a uniform decrease in high-frequency presence, increased background noise (hiss), more noticeable dropouts where there are flaws in the magnetic tape, and shifting of the (Gaussian) background noise spectrum toward lower frequencies (where it sounds more "granular",) *regardless* of the audio content. An MP3 of a noisy rock band at a low bit rate will have many more artifacts than a simple flute solo at the same bitrate, whereas either on low-speed tape will have the same uniform background noise profile and the same limited frequency spectrum (rolled-off high end) but no dynamic distortion patterns.

A recording on magnetic audio tape is linear; unlike today's digital audio, not only was jumping from spot to spot to edit time consuming, editing was destructive—unless the recording was duplicated before edit, normally taking the same amount of time to copy, in order to preserve 75-90 percent of the quality of the original. Editing was done either with a razor blade—by physically cutting and splicing the tape, in a manner similar to motion picture film editing—or electronically by dubbing segments onto an edit tape. The former method preserved the full quality of the recording but not the intact original; the latter incurred the same quality loss involved in dubbing a complete copy of the source tape, but preserved the original.

Tape speed is not the only factor affecting the quality of the recording. Other factors affecting quality include track width, tape formulation, and backing material and thickness. The design and quality of the recorder are also important factors, in many ways that are not applicable to digital recording systems (of any kind.) The machine's speed stability (wow-and-flutter), head gap size, head quality, and general head design and technology, and the machine's alignment (mostly a maintenance issue, but also a matter of design—how well and precisely it can be aligned) electro-mechanically affect the quality of the recording. The regulation of tape tension affects contact between the tape and the heads and has a very significant impact on the recording and reproduction of high frequencies. The track width of the machine, which is a question of format rather than individual machine design, is one of two major machine factors controlling signal-to-noise ratio (assuming the electronics have high enough S/N not to be a factor), the other being tape speed. S/N ratio varies directly with track width, due to the Gaussian nature of tape noise; doubling the track width doubles the S/N ratio (hence, with good electronics and comparable heads, 8-track cartridges should have half the signal-to-noise of quarter-track 1/4" tape at the same speed, 3-3/4 IPS.) Tape formulation affects the retention of the magnetic signal, especially high frequencies, the frequency linearity of the tape, the S/N ratio, print-through, optimum AC bias level (which must be set by a technician aligning the machine to match the tape type used, or more crudely set with a switch to approximate the optimum setting.) Tape formulation varies between different tape types (ferric oxide [FeO], chromium dioxide [CrO_2], etc.) and also in the precise composition of a specific brand and batch of tape. (Studios therefore generally align their machines for one brand and model number of tape and use only that brand and model.) Backing material type and thickness affect the tensile strength and elasticity of the tape, which affect wow-and-flutter and tape stretch; stretched tape will have a pitch error, possibly fluctuating. Backing thickness also effects print-through, the phenomenon of adjacent layers of tape wound on a reel picking up weak copies of the magnetic signal from each other. Print-through on analog tape causes unintended pre- and post-echoes on playback, and is generally not fully reversible once it has occurred.

Noise reduction

Electronic noise reduction techniques were also developed to increase the signal-to-noise ratio and dynamic range of analog sound recordings. Dolby noise reduction includes a suite of standards (designated A, B, C, S and SR) for both professional and consumer recording. The Dolby systems use frequency dependant compression/expansion (companding) during the recording/playback, respectively. DBX is another noise reduction system that uses a more aggressive companding technique to improve both dynamic range and noise level. However, DBX recordings do not sound acceptable when played on non-DBX equipment.

Dolby B eventually became the most popular system for Compact Cassette noise reduction. Today Dolby SR is in widespread use for professional analog tape recording and is only surpassed in quality (although difference is almost negligible) by digital audio technologies.

Multi-track recorders

As studio audio production progressed and became more and more advanced, it became desirable to record the separate instruments and human voices separately and mix them down to one, two, or more speaker channels later, rather than in real time in the studio before recording. In addition to allowing recording engineers and producers to experiment with different mixing arrangements, effects, etc. on the same performance and to produce multiple versions of a recording (without having multiple duplicates of all the studio control room equipment used for mixing), multi-tracking enables the use of non-real-time effects or effects that cannot be produced in the same studio where the musicians perform. Reel-to-reel recorders with eight, sixteen, twenty four, and even thirty two tracks were eventually built, with as many heads recording synchronized parallel linear tracks. Some of these machines were larger than a laundry washing machine and used tape as wide as 2 inches. A single new reel of 1" or wider tape, blank, could easily cost over $100, to $200. Still, in professional studios, most tapes were recorded only once, and all recording was on new tape, to ensure the maximum quality, as studio time and the time of skilled musicians was much higher than the cost of tape, making it not worth the risk of a recording being lost or degraded due to using media that had been previously recorded upon.

Digital reel-to-reel

As professional audio evolved from analog magnetic tape to digital media, engineers adapted magnetic tape technology to digital recording, producing digital reel-to-reel magnetic tape machines. Before large hard disks became economical enough to make hard disk recorders viable, studio digital recording meant recording on digital tape. Mitsubishi's ProDigi and Sony's Digital Audio Stationary Head (DASH) were the primary digital reel-to-reel formats in use in recording studios from the early 1980s through the mid 1990s. Nagra introduced digital reel-to-reel tape recorders for use in film sound recording. Digital reel-to-reel tape eliminated all the traditional quality limitations of analog tape, including background noise (hiss), high frequency roll-off, wow and flutter, pitch error, nonlinearity, print-through, and degeneration with copying, but the tape media was even more expensive than professional analog open reel tape, and the linear nature of tape still placed restrictions on access, and winding time to find a particular spot was still a significant drawback. Also, while the quality of digital tape did not progressively degrade with use of the tape, the physical sliding of the tape over the heads and guides meant that the tape still did wear, and eventually that wear would lead to digital errors and permanent loss of quality if the tape was not copied before reaching that point. Still, digital reel-to-reel tape represented a significant advance in audio recording technology, and most who could afford to record using digital tape generally did.

As a musical instrument

Early reel-to-reel users realized that segments of tape could be spliced together and otherwise manipulated by adjusting playback speed or direction of a given recording. In the same way as modern keyboards allow sampling and playback at different speeds, a reel-to-reel could accomplish similar feats in the hands of a talented user. Consider:

- The title track of Jimi Hendrix's album *Are You Experienced*, on which the guitar solo and much of the drum track was recorded, then played backwards on a reel-to-reel.
- The Beatles recorded many songs using reel to reel tape as a part of the creative process. Examples include "Being for the Benefit of Mr. Kite" and "Yellow Submarine" which used a technique where stock recordings were cut up and then randomly reassembled and overdubbed on to the songs (On "Mr. Kite", recordings of calliope organs and on "Yellow Submarine" recordings of Marching Bands). On "Tomorrow Never Knows" multiple tape machines were used all interconnected patching tape loops that had been prepared by the band. The loops were played in a variety of ways such as backwards, sped up and slowed down. To record the song the machines, which were located in separate studio rooms, were all manned by individual technicians and played at once to record on the fly. "Strawberry Fields Forever" combined two different taped versions of the song. The versions were independently altered in speed to end up together miraculously both on pitch and tempo. "I am the Walrus" used a radio tuner patched into the sound console to layer random live broadcast over an existing taped track. "Revolution 9" also had many effects produced using a reel-to-reel and tape editing techniques.
- Delia Derbyshire, who performed the original Doctor Who theme by recording various sounds including oscillators and then manually cutting together each individual note on a group of reel-to-reels.
- Aaron Dilloway, founding member of Wolf Eyes, often utilizes a reel to reel tape machine in his solo performances.
- Yamantaka Eye of the band Boredoms uses a reel-to-reel tape as an instrument in live performances and in post-production (a good example would be in the track 'Super You' from the album Super æ).

- The Gasman who produced much of his early work on Planet Mu Records splicing old reel-to-reel classical music into loops.
- Mission of Burma, whose fourth member Martin Swope "played" a reel-to-reel tape recorder live, either playing previously recorded samples at certain times or recording part of the band's performance and playing it back either in reverse or at different speeds. When the band re-formed in 2002, audio engineer Bob Weston took over Swope's role at the tape deck.
- Musique concrète in general.
- Pink Floyd's cash register introduction to their track "Money" was made using a loop of "splices" which was continually run through the reel-to-reel mechanism.
- Steve Tibbetts is a recording artist that includes tape editing as a significant portion of the creative process.
- Frank Zappa's *Lumpy Gravy* and *We're Only In It For the Money*, both of which featured edits too numerous to mention, in addition to multiple instances of speed alteration and intricately layered samples upon samples.

In addition, multiple reel-to-reel machines used in tandem can also be used to create echo and delay effects. The Frippertronics configuration used by Brian Eno and Robert Fripp on numerous of their 1970s and '80s recordings illustrates these possibilities.

Chapter 8

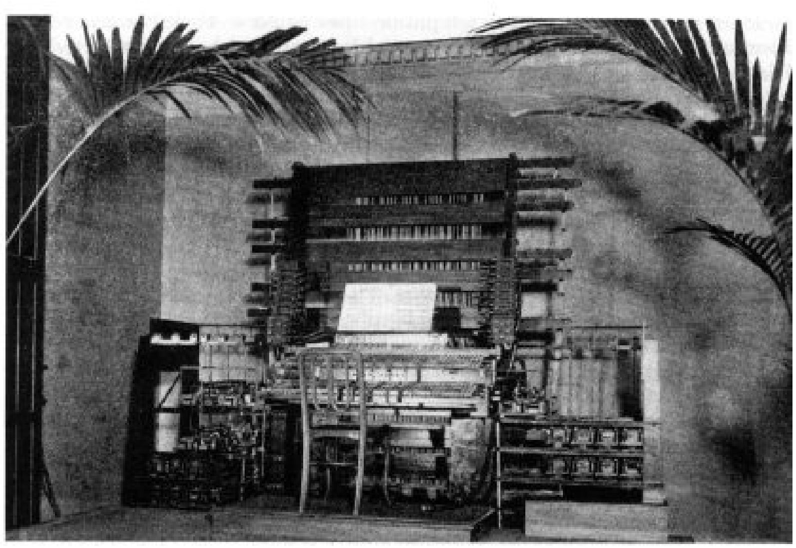
Electronic Music

Electronic music is music that employs electronic musical instruments and electronic music technology in its production. In general a distinction can be made between sound produced using electromechanical means and that produced using electronic technology. Examples of electromechanical sound producing devices include the telharmonium, Hammond organ, and the electric guitar. Purely electronic sound production can be achieved using devices such as the Theremin, sound synthesizer, and computer.

Electronic music was once associated almost exclusively with Western art music but from the late 1960s on the availability of affordable music technology meant that music produced using electronic means became increasingly common in the popular domain. Today electronic music includes many varieties and ranges from experimental art music to popular forms such as electronic dance music.

History

Late 19th century to early 20th century



Telharmonium, Thaddeus Cahill, 1897

The ability to record sounds is often connected to the production of electronic music, but not absolutely necessary for it. The earliest known sound recording device was the phonautograph, patented in 1857 by Édouard-Léon Scott de Martinville. It could record sounds visually, but was not meant to play them back.

In 1878, Thomas A. Edison patented the phonograph, which used cylinders similar to Scott's device. Although cylinders continued in use for some time, Emile Berliner developed the disc phonograph in 1887. A significant invention, which was later to have a profound effect on electronic music, was Lee DeForest's triode audion. This was the first thermionic valve, or vacuum tube, invented in 1906, which led to the generation and amplification of electrical signals, radio broadcasting, and electronic computation, amongst other things.

Before electronic music, there was a growing desire for composers to use emerging technologies for musical purposes. Several instruments were created that employed electromechanical designs and they paved the way for the later emergence of electronic instruments. An electromechanical instrument called the Telharmonium (sometimes Teleharmonium or Dynamophone) was developed by Thaddeus Cahill in the years 1898-1912. However, simple inconvenience hindered the adoption of the Telharmonium, due to its immense size. The first electronic instrument is often viewed to be the Theremin, invented by Professor Léon Theremin circa 1919–1920. Other early electronic instruments include the *Croix Sonore*, invented in 1926 by Nikolai Obukhov, and the Ondes Martenot, which was most famously used in the *Turangalîla-Symphonie* by Olivier Messiaen as well as other works by him. The Ondes Martenot was also used by other, primarily French, composers such as Andre Jolivet.

Sketch of a New Esthetic of Music

In 1907, just a year later after the invention of the triode audion, Ferruccio Busoni published *Sketch of a New Esthetic of Music*, which discussed the use of electrical and other new sound sources in future music. He wrote of the future of microtonal scales in music, made possible by Cahill's Dynamophone: "Only a long and careful series of experiments, and a continued training of the ear, can render this unfamiliar material approachable and plastic for the coming generation, and for Art."

Also in the *Sketch of a New Esthetic of Music*, Busoni states:

Music as an art, our so-called occidental music, is hardly four hundred years old; its state is one of development, perhaps the very first stage of a development beyond present conception, and we—we talk of "classics" and "hallowed traditions"! And we have talked of them for a long time! We have formulated rules, stated principles, laid down laws;—we apply laws made for maturity to a child that knows nothing of responsibility! Young as it is, this child, we already recognize that it possesses one radiant attribute which signalizes it beyond all its elder sisters. And the lawgivers will not see this marvelous attribute, lest their laws should be thrown to the winds. This child—it *floats on air*! It touches not the earth with its feet. It knows no law of gravitation. It is well nigh incorporeal. Its material is transparent. It is sonorous air. It is almost Nature herself. It is—free! But freedom is something that mankind have never wholly comprehended, never realized to the full. They can neither recognize or acknowledge it. They disavow the mission of this child; they hang weights upon it. This buoyant creature must walk decently, like anybody else. It may scarcely be allowed to leap—when it were its joy to follow the line of the rainbow, and to break sunbeams with the clouds.

Through this writing, as well as personal contact, Busoni had a profound effect on many musicians and composers, perhaps most notably on his pupil, Edgard Varèse, who said:

Together we used to discuss what direction the music of the future would, or rather, should take and could not take as long as the straitjacket of the tempered system. He deplored that his own keyboard instrument had conditioned our ears to accept only an infinitesimal part of the infinite gradations of sounds in nature. He was very much interested in the electrical instruments we began to hear about, and I remember particularly one he had read of called the Dynamophone. All through his writings one finds over and over again predictions about the music of the future which have since come true. In fact, there is hardly a development that he did not foresee, as for instance in this extraordinary prophecy: 'I almost think that in the new great music, machines will also be necessary and will be assigned a share in it. Perhaps industry, too, will bring forth her share in the artistic ascent.

Futurists

In Italy, the Futurists approached the changing musical aesthetic from a different angle. A major thrust of the Futurist philosophy was to value "noise," and to place artistic and expressive value on sounds that had previously not been considered even remotely musical. Balilla Pratella's "Technical Manifesto of Futurist Music" (1911) states that their credo is: "To present the musical soul of the masses, of the great factories, of the railways, of the transatlantic liners, of the battleships, of the automobiles and airplanes. To add to the great central themes of the musical poem the domain of the machine and the victorious kingdom of Electricity."

On 11 March 1913, futurist Luigi Russolo published his manifesto "The Art of Noises". In 1914, he held the first "art-of-noises" concert in Milan on April 21. This used his Intonarumori, described by Russolo as "acoustical noise-instruments, whose sounds (howls, roars, shuffles, gurgles, etc.) were hand-activated and projected by horns and megaphones." In June, similar concerts were held in Paris.

The 1920–1930s

This decade brought a wealth of early electronic instruments and the first compositions for electronic instruments. The first instrument, the Etherophone, was created by Léon Theremin (born Lev Termen) between 1919 and 1920 in Leningrad, though it was eventually renamed the Theremin. This led to the first compositions for electronic instruments, as opposed to noisemakers and re-purposed machines. In 1929, Joseph

Schillinger composed *First Airphonic Suite for Theremin and Orchestra*, premièred with the Cleveland Orchestra with Leon Theremin as soloist.

In addition to the Theremin, the Ondes Martenot was invented in 1928 by Maurice Martenot, who debuted it in Paris.

The following year, Antheil first composed for mechanical devices, electrical noisemakers, motors and amplifiers in his unfinished opera, *Mr. Bloom*.

Recording of sounds made a leap in 1927, when American inventor J. A. O'Neill developed a recording device that used magnetically coated ribbon. However, this was a commercial failure. Two years later, Laurens Hammond established his company for the manufacture of electronic instruments. He went on to produce the Hammond organ, which was based on the principles of the Telharmonium, along with other developments including early reverberation units. Hammond (along with John Hanert and C. N. Williams) would also go onto invent another electronic instrument, the Novachord, which Hammond's company manufactured from 1939–1942.

The method of photo-optic sound recording used in cinematography made it possible to obtain a visible image of a sound wave, as well as to realize the opposite goal—synthesizing a sound from an artificially drawn sound wave.

In this same period, experiments began with sound art, early practitioners of which include Tristan Tzara, Kurt Schwitters, Filippo Tommaso Marinetti, and others.

Developments from 1945 to 1960

Musique concrète

Low-fidelity magnetic wire recorders had been in use since around 1900 and in the early 1930s the movie industry began to convert to the new optical sound-on-film recording systems based on the photoelectric cell. It was around this time that the German electronics company AEG developed the first practical audio tape recorder, the "Magnetophon" K-1, which was unveiled at the Berlin Radio Show in August 1935.

During World War II, Walter Weber rediscovered and applied the AC biasing technique, which dramatically improved the fidelity of magnetic recording by adding an inaudible high-frequency tone. It extended the 1941 'K4' Magnetophone frequency curve to 10 kHz and improved the signal-to-noise ratio up to 60 dB, surpassing all known recording systems at that time.

As early as 1942 AEG was making test recordings in stereo. However these devices and techniques remained a secret outside Germany until the end of WWII, when captured Magnetophon recorders and reels of Farben ferric-oxide recording tape were brought back to the United States by Jack Mullin and others. These captured recorders and tapes were the basis for the development of America's first commercially-made professional tape recorder, the Model 200, manufactured by the American Ampex company with support from entertainer Bing Crosby, who became one of the first performers to record radio broadcasts and studio master recordings on tape.

Magnetic audio tape opened up a vast new range of sonic possibilities to musicians, composers, producers and engineers. Audio tape was relatively cheap and very reliable, and its fidelity of reproduction was better than any audio medium to date. Most importantly, unlike discs, it offered the same plasticity of use as film. Tape can be slowed down, sped up or even run backwards during recording or playback, with often startling effect. It can be physically edited in much the same way as film, allowing for unwanted sections of a recording to be seamlessly removed or replaced; likewise, segments of tape from other sources can be edited in. Tape can also be joined to form endless loops that continually play repeated patterns of pre-recorded material. Audio amplification and mixing equipment further expanded tape's capabilities as a production medium, allowing multiple pre-taped recordings (and/or live sounds, speech or music) to be mixed together and simultaneously recorded onto another tape with relatively little loss of fidelity. Another unforeseen windfall was that tape recorders can be relatively easily modified to become echo machines that produce complex, controllable, high-quality echo and reverberation effects (most of which would be practically impossible to achieve by mechanical means).

It wasn't long before composers began using the tape recorder to develop a new technique for composition called Musique concrète. This technique involved editing together recorded fragments of natural and industrial sounds. The first pieces of *musique concrète* were assembled by Pierre Schaeffer, who went on to collaborate with Pierre Henry.

On 5 October 1948, Radiodiffusion Française (RDF) broadcast composer Pierre Schaeffer's *Etude aux chemins de fer*. This was the first "movement" of *Cinq études de bruits*, and marked the beginning of studio realizations and musique concrète (or acousmatic art). Schaeffer employed a disk-cutting lathe, four turntables, a four-channel mixer, filters, an echo chamber, and a mobile recording unit.

Not long after this, Henry began collaborating with Schaeffer, a partnership that would have profound and lasting effects on the direction of electronic music. Another associate of Schaeffer, Edgard Varèse, began work on *Déserts*, a work for chamber orchestra and tape. The tape parts were created at Pierre Schaeffer's studio, and were later revised at Columbia University.

In 1950, Schaeffer gave the first public (non-broadcast) concert of musique concrète at the Ecole Normale de Musique de Paris. "Schaeffer used a PA system, several turntables, and mixers. The performance did not go well, as creating live montages with turntables had never been done before." Later that same year, Pierre Henry collaborated with Schaeffer on *Symphonie pour un homme seul* (1950) the first major work of musique concrete. In Paris in 1951, in what was to become an important worldwide trend, RTF established the first studio for the production of electronic music. Also in 1951, Schaeffer and Henry produced an opera, *Orpheus*, for concrete sounds and voices.

Elektronische Musik

Karlheinz Stockhausen in the Electronic Music Studio of WDR, Cologne, in 1991

Karlheinz Stockhausen worked briefly in Schaeffer's studio in 1952, and afterward for many years at the WDR Cologne's Studio for Electronic Music.

In Cologne, what would become the most famous electronic music studio in the world was officially opened at the radio studios of the NWDR in 1953, though it had been in the planning stages as early as 1950 and early compositions were made and broadcast in 1951. The brain child of Werner Meyer-Eppler, Robert Beyer, and Herbert Eimert (who became its first director), the studio was soon joined by Karlheinz Stockhausen and Gottfried Michael Koenig. In his 1949 thesis *Elektronische Klangerzeugung: Elektronische Musik und Synthetische Sprache*, Meyer-Eppler conceived the idea to synthesize music entirely from electronically produced signals; in this way, *elektronische Musik* was sharply differentiated from French *musique concrète*, which used sounds recorded from acoustical sources.

With Stockhausen and Mauricio Kagel in residence, it became a year-round hive of charismatic avante-gardism [*sic*]" on two occasions combining electronically generated sounds with relatively conventional orchestras—in *Mixtur* (1964) and *Hymnen, dritte Region mit Orchester* (1967). Stockhausen stated that his listeners had told him his electronic music gave them an experience of "outer space," sensations of flying, or being in a "fantastic dream world" More recently, Stockhausen turned to producing electronic music in his own studio in Kürten, his last work in the genre being *Cosmic Pulses* (2007).

American electronic music

In the United States, sounds were being created electronically and used in composition, as exemplified in a piece by Morton Feldman called *Marginal Intersection*. This piece is scored for winds, brass, percussion, strings, 2 oscillators, and sound effects of riveting, and the score uses Feldman's graph notation.

The Music for Magnetic Tape Project was formed by members of the New York School (John Cage, Earle Brown, Christian Wolff, David Tudor, and Morton Feldman), and lasted three years until 1954. Cage wrote of this collaboration: "In this social darkness, therefore, the work of Earle Brown, Morton Feldman, and Christian Wolff continues to present a brilliant light, for the reason that at the several points of notation, performance, and audition, action is provocative.

Cage completed *Williams Mix* in 1953 while working with the Music for Magnetic Tape Project. The group had no permanent facility, and had to rely on borrowed time in commercial sound studios, including the studio of Louis and Bebe Barron.

Columbia-Princeton

In the same year Columbia University purchased its first tape recorder—a professional Ampex machine—for the purpose of recording concerts.

Vladimir Ussachevsky, who was on the music faculty of Columbia University, was placed in charge of the device, and almost immediately began experimenting with it.

Herbert Russcol writes: "Soon he was intrigued with the new sonorities he could achieve by recording musical instruments and then superimposing them on one another."

Ussachevsky said later: "I suddenly realized that the tape recorder could be treated as an instrument of sound transformation."

On Thursday, May 8, 1952, Ussachevsky presented several demonstrations of tape music/effects that he created at his Composers Forum, in the McMillin Theatre at Columbia University. These included *Transposition, Reverberation, Experiment, Composition*, and *Underwater Valse*. In an interview, he stated: "I presented a few examples of my discovery in a public concert in New York together with other compositions I had written for conventional instruments." Otto Luening, who had attended this concert, remarked: "The equipment at his disposal consisted of an Ampex tape recorder . . . and a simple box-like device designed by the brilliant young engineer, Peter Mauzey, to create feedback, a form of mechanical reverberation. Other equipment was borrowed or purchased with personal funds."

Just three months later, in August 1952, Ussachevsky traveled to Bennington, Vermont at Luening's invitation to present his experiments. There, the two collaborated on various pieces. Luening described the event: "Equipped with earphones and a flute, I began developing my first tape-recorder composition. Both of us were fluent improvisors and the medium fired our imaginations." They played some early pieces informally at a party, where "a number of composers almost solemnly congratulated us saying, 'This is it' ('it' meaning the music of the future)."

Word quickly reached New York City. Oliver Daniel telephoned and invited the pair to "produce a group of short compositions for the October concert sponsored by the American Composers Alliance and Broadcast Music, Inc., under the direction of Leopold Stokowski at the Museum of Modern Art in New York. After some hesitation, we agreed. . . . Henry Cowell placed his home and studio in Woodstock, New York, at our disposal. With the borrowed equipment in the back of Ussachevsky's car, we left Bennington for Woodstock and stayed two weeks. . . . In late September, 1952, the travelling laboratory reached Ussachevsky's living room in New York, where we eventually completed the compositions."

Two months later, on October 28, Vladimir Ussachevsky and Otto Luening presented the first Tape Music concert in the United States. The concert included Luening's *Fantasy in Space* (1952)—"an impressionistic virtuoso piece" using manipulated recordings of flute—and *Low Speed* (1952), an "exotic composition that took the flute far below its natural range." Both pieces were created at the home of Henry Cowell in Woodstock, NY. After several concerts caused a sensation in New York City, Ussachevsky and Luening were invited onto a live broadcast of NBC's Today Show to do an interview demonstration—the first televised electroacoustic performance. Luening described the event: "I improvised some [flute] sequences for the tape recorder. Ussachevsky then and there put them through electronic transformations."

1954 saw the advent of what would now be considered authentic electric plus acoustic compositions—acoustic instrumentation augmented/accompanied by recordings of manipulated and/or electronically generated sound. Three major works were premiered that year: Varèse's *Déserts*, for chamber ensemble and tape sounds, and two works by Luening and Ussachevsky: *Rhapsodic Variations for the Louisville Symphony* and *A Poem in Cycles and Bells*, both for orchestra and tape. Because he had been working at Schaeffer's studio, the tape part for Varèse's work contains much more concrete sounds than electronic. "A group made up of wind instruments, percussion and piano alternates with mutated sounds of factory noises and ship sirens and motors, coming from two loudspeakers."

Déserts was premiered in Paris in the first stereo broadcast on French Radio. At the German premiere in Hamburg, which was conducted by Bruno Maderna, the tape controls were operated by Karlheinz Stockhausen. The title *Déserts*, suggested to Varèse not only, "all physical deserts (of sand, sea, snow, of outer space, of empty streets), but also the deserts in the mind of man; not only those stripped aspects of nature that suggest bareness, aloofness, timelessness, but also that remote inner space no telescope can reach, where man is alone, a world of mystery and essential loneliness."

Stochastic music

An important new development was the advent of computers for the purpose of composing music, as opposed to manipulating or creating sounds. Iannis Xenakis began what is called "musique stochastique," or "stochastic music", which is a method of composing that employs mathematical probability systems. Different probability algorithms were used to create a piece under a set of parameters. Xenakis used graph paper and a ruler to aid in calculating the velocity trajectories of glissandi for his orchestral composition *Metastasis* (1953–54), but later turned to the use of computers to compose pieces like *ST/4* for string quartet and *ST/48* for orchestra (both 1962).

Mid to late 1950s

In 1954, Stockhausen composed his *Elektronische Studie II*—the first electronic piece to be published as a score.

In 1955, more experimental and electronic studios began to appear. Notable were the creation of the Studio de Fonologia (already mentioned), a studio at the NHK in Tokyo founded by Toshiro Mayuzumi, and the Phillips studio at Eindhoven, the Netherlands, which moved to the University of Utrecht as the Institute of Sonology in 1960.

The score for *Forbidden Planet*, by Louis and Bebe Barron, was entirely composed using custom built electronic circuits and tape recorders in 1956.

The world's first computer to play music was CSIRAC which was designed and built by Trevor Pearcey and Maston Beard. Mathematician Geoff Hill programmed the CSIRAC to play popular musical melodies from the very early 1950s. In 1951 it publicly played the Colonel Bogey March of which no known recordings exist. However, CSIRAC played standard repertoire and was not used to extend musical thinking or composition practice which is current computer music practice. CSIRAC was never recorded, but the music played was accurately reconstructed (reference 12). The oldest known recordings of computer generated music were played by the Ferranti Mark 1 computer, a commercial version of the Baby Machine from the University of Manchester in the autumn of 1951. The music program was written by Christopher Strachey.

The impact of computers continued in 1956. Lejaren Hiller and Leonard Isaacson composed *Iliac Suite* for string quartet, the first complete work of computer-assisted composition using algorithmic composition. "... Hiller postulated that a computer could be taught the rules of a particular style and then called on to compose accordingly." Later developments included the work of Max Mathews at Bell Laboratories, who developed the influential MUSIC I program. Vocoder technology was also a major development in this early era.

In 1956, Stockhausen composed *Gesang der Jünglinge*, the first major work of the Cologne studio, based on a text from the *Book of Daniel*. An important technological development of that year was the invention of the Clavivox synthesizer by Raymond Scott with subassembly by Robert Moog.

In 1957, MUSIC, one of the first computer programs to play electronic music, was created by Max Mathews at Bell Laboratories.

Also in 1957, Kid Baltan (Dick Raaymakers) and Tom Dissevelt released their debut album, Song Of The Second Moon, recorded at the Phillips studio.

Later, Milton Babbitt began composing electronic music:

From 1950 to 1960 the vocabulary of tape music shifted from the fairly pure experimental works which characterized the classic Paris and Cologne schools to more complex and expressive works which explored a wide range of compositional styles. More and more works began to appear by the mid-1950s which addressed the concept of combining taped sounds with live instruments and voices. There was also a tentative interest, and a few attempts, at incorporating taped electronic sounds into theatrical works.

The public remained interested in the new sounds being created around the world, as can be deduced by the inclusion of Varèse's *Poeme Electronique*, which was played over four hundred loudspeakers at the Phillips Pavilion of the 1958 Brussels World Fair. That same year, Mauricio Kagel, an Argentine composer, composed *Transición II*. The work was realized at the WDR studio in Cologne. Two musicians perform on a piano, one in the traditional manner, the other playing on the strings, frame, and case. Two other performers use tape to unite the presentation of live sounds with the future of prerecorded materials from later on and its past of recordings made earlier in the performance.

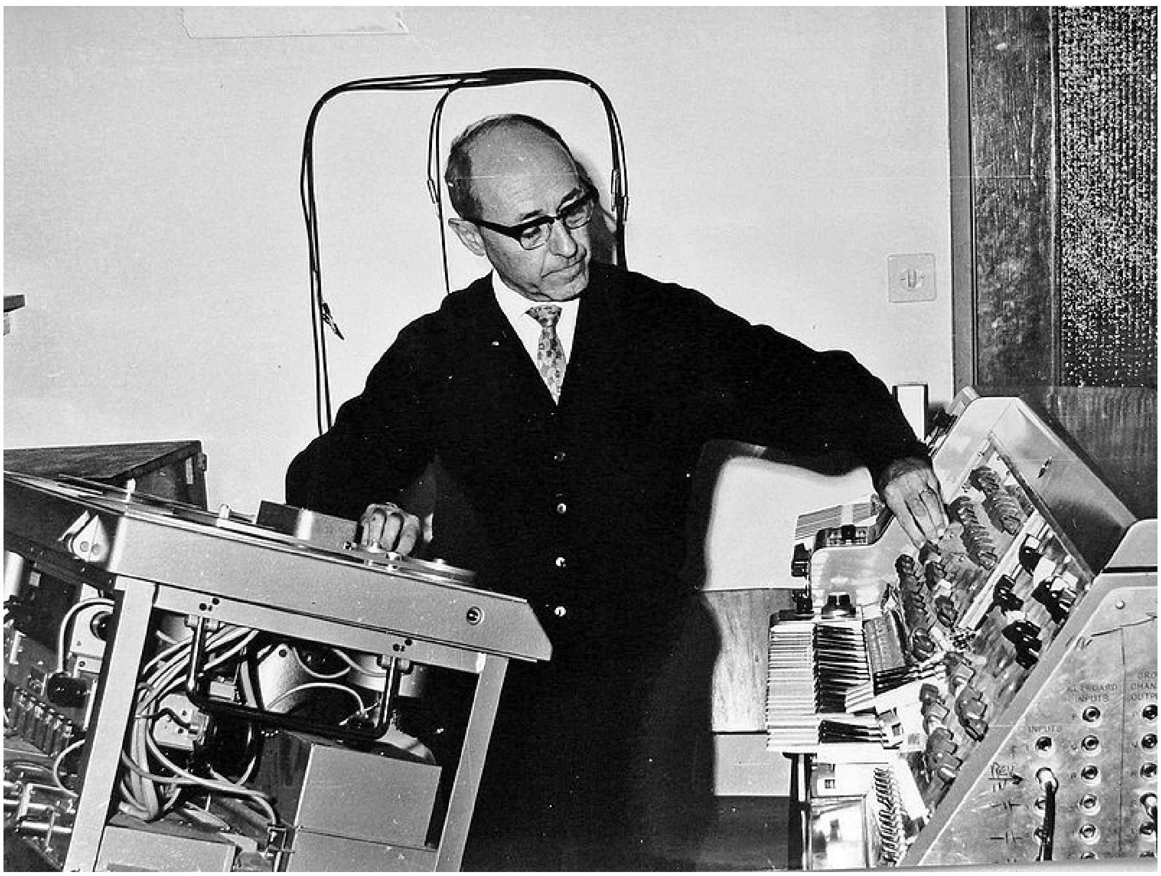
The 1960s

These were fertile years for electronic music—not just for academia, but for independent artists as synthesizer technology became more accessible. By this time, a strong community of composers and musicians working with new sounds and instruments was established and growing. 1960 witnessed the composition of Luening's *Gargoyles* for violin and tape as well as the premiere of Stockhausen's *Kontakte* for electronic sounds, piano, and percussion. This piece existed in two versions—one for 4-channel tape, and the other for tape with human performers. "In *Kontakte*, Stockhausen abandoned traditional musical form based on linear development and dramatic climax. This new approach, which he termed 'moment form,' resembles the 'cinematic splice' techniques in early twentieth century film."

The first of these synthesizers to appear was the Buchla. Appearing in 1963, it was the product of an effort spearheaded by *musique concrète* composer Morton Subotnick.

The theremin had been in use since the 1920s but it attained a degree of popular recognition through its use in science-fiction film soundtrack music in the 1950s (e.g., Bernard Herrmann's classic score for *The Day the Earth Stood Still*).

In the UK in this period, the BBC Radiophonic Workshop (established in 1958) emerged one of the most productive and widely known electronic music studios in the world, thanks in large measure to their work on the BBC science-fiction series *Doctor Who*. One of the most influential British electronic artists in this period was Workshop staffer Delia Derbyshire, who added a keen musical ear to her great technical prowess—she is famous for her landmark 1963 electronic realisation of the iconic *Doctor Who* theme, composed by Ron Grainer.



Israeli composer Josef Tal at the Electronic Music Studio in Jerusalem (~1965). On the right - Hugh Le Caine's sound synthesizer

In 1961 Josef Tal established the *Centre for Electronic Music in Israel* at The Hebrew University, and in 1962 Hugh Le Caine arrived in Jerusalem to install his *Creative Tape Recorder* in the centre. In the 1990s Tal conducted, together with Dr Shlomo Markel, in cooperation with the Technion – Israel Institute of Technology, and VolkswagenStiftung a research project (Talmark) aimed at the development of a novel musical notation system for electronic music.

Milton Babbitt composed his first electronic work using the synthesizer—his *Composition for Synthesizer*—which he created using the RCA synthesizer at CPEMC.

For Babbitt, the RCA synthesizer was a dream come true for three reasons. First, the ability to pinpoint and control every musical element precisely. Second, the time needed to realize his elaborate serial structures were brought within practical reach. Third, the question was no longer "What are the limits of the human performer?" but rather "What are the limits of human hearing?

The collaborations also occurred across oceans and continents. In 1961, Ussachevsky invited Varèse to the Columbia-Princeton Studio (CPEMC). Upon arrival, Varese embarked upon a revision of *Déserts*. He was assisted by Mario Davidovsky and Bülent Arel.

The intense activity occurring at CPEMC and elsewhere inspired the establishment of the San Francisco Tape Music Center in 1963 by Morton Subotnick, with additional members Pauline Oliveros, Ramon Sender, Terry Riley, and Anthony Martin. The center soon incorporated a voltage-controlled synthesizer based around automated sequencing by Don Buchla, and used in album-length Subotnick pieces such as *Silver Apples of the Moon* (1967) and *The Wild Bull* (1968).

Later, the Center moved to Mills College, directed by Pauline Oliveros, where it is today known as the Center for Contemporary Music.

Simultaneously in San Francisco, composer Stan Shaff and equipment designer Doug McEachern, presented the first "Audium" concert at San Francisco State College (1962), followed by a work at the San Francisco Museum of Modern Art (1963), conceived of as in time, controlled movement of sound in space. Twelve speakers surrounded the audience, four speakers were mounted on a rotating, mobile-like construction above. In an SFMOMA performance the following year (1964), *San Francisco Chronicle* music critic Alfred Frankenstein commented, "...the possibilities of the space-sound continuum have seldom been so extensively explored." In 1967, the first Audium, a "sound-space continuum" opened, holding weekly performances through 1970. In 1975, enabled by seed money from the National Endowment for the Arts, a new Audium opened, designed floor to ceiling for spatial sound composition and performance. "There are composers who manipulate sound space by locating multiple speakers at various locations in a performance space and then switching or panning the sound between the sources. In this approach, the composition of spatial manipulation is dependent on the location of the speakers and usually exploits the acoustical properties of the enclosure. Examples include Varese's Poem Electronique (tape music performed in the Phillips Pavilion of the 1958 World Fair, Brussels) and Stanley Schaff's Audium installation, currently active in San Francisco." Through weekly programs (over 4,500 in 40 years), Shaff "sculpts" sound, performing now-digitized spatial works live through 176 speakers.

Back across the Atlantic, in Czechoslovakia, 1964, the First Seminar of Electronic Music was held at the Radio Broadcast Station in Plzen. Four government-sanctioned electroacoustic music studios were later established in the 1960s under the auspices of extant radio and television stations.

New instruments continued to develop. One of the most significant breakthroughs came in 1964, when Robert Moog introduced the Moog synthesizer, the first integrated modular voltage controlled analog synthesizer system. Moog Music later introduced a smaller synthesizer with a built-in keyboard and hardwired signal path called the Minimoog, which was introduced to many composers and universities and became widely used by popular musicians.

A well-known example of the use of Moog's full-sized Moog modular synthesizer is the *Switched-On Bach* album by Wendy Carlos, which triggered a craze for synthesizer music. In the late 1960s, Nonesuch Records commissioned and released to great acclaim a series of recordings of serious classical music composed on the Moog synthesizer including "Tragoedia" by Andrew Rudin, the first composer, according to Moog, to have written serious compositions for his synthesizer. Rudin's "Tragoedia" was re-issued in 2010 by the Warner Music Group's Rhino/Elektra Label.

Pietro Grossi was an Italian pioneer of computer composition and tape music, who first experimented with electronic techniques in the early sixties. Grossi was a cellist and composer, born in Venice in 1917. He founded the S 2F M (Studio de Fonologia Musicale di Firenze) in 1963 in order to experiment with electronic sound and composition.

Computer music

CSIRAC, the first computer to play music, did so publicly in August 1951 (reference 12). One of the first large-scale public demonstrations of computer music was a pre-recorded national radio broadcast on the NBC radio network program Monitor on February 10, 1962. In 1961, LaFarr Stuart programmed Iowa State University's CYCLONE computer (a derivative of the Illiac) to play simple, recognizable tunes through an amplified speaker that had been attached to the system originally for administrative and diagnostic purposes. An interview with Mr. Stuart accompanied his computer music.

The late 1950s, 1960s and 1970s also saw the development of large mainframe computer synthesis. Starting in 1957, Max Mathews of Bell Labs developed the MUSIC programs, culminating in MUSIC V, a direct digital synthesis language

Live electronics

In America, live electronics were pioneered in the early 1960s by members of Milton Cohen's Space Theater in Ann Arbor, Michigan, including Gordon Mumma and Robert Ashley, by individuals such as David Tudor around 1965, and The Sonic Arts Union, founded in 1966 by Gordon Mumma, Robert Ashley, Alvin Lucier, and David Behrman. ONCE Festivals, featuring multimedia theater music, were organized by Robert Ashley and Gordon Mumma in Ann Arbor between 1958 and 1969. In 1960, John Cage composed *Cartridge Music*, one of the earliest live-electronic works.

In Europe in 1964, Karlheinz Stockhausen composed *Mikrophonie I* for tam-tam, hand-held microphones, filters, and potentiometers, and *Mixtur* for orchestra, four sine-wave generators, and four ring modulators. In 1965 he composed *Mikrophonie II* for choir, Hammond organ, and ring modulators.

The Jazz composers and musicians Paul Bley and Annette Peacock performed some of the first live concerts in the late 1960s using Moog synthesizers. Peacock made regular use of a customised Moog synthesizer to process her voice on stage and in studio recordings.

In 1966–67, Reed Ghazala discovered and began to teach "circuit bending"—the application of the creative short circuit, a process of chance short-circuiting, creating experimental electronic instruments, exploring sonic elements mainly of timbre and with less regard to pitch or rhythm, and influenced by John Cage's aleatoric music concept.

1970s to mid-80s

In 1970, Charles Wuorinen composed *Time's Encomium*, the first Pulitzer Prize winner for an entirely electronic composition.

Synthesizers

Released in 1970 by Moog Music the Mini-Moog was among the first widely available, portable and relatively affordable synthesizers. It became the most widely used synthesizer in both popular and electronic art music. In 1974 the WDR studio in Cologne acquired an EMS Synthi 100 synthesizer which was used by a number of composers in the production of notable electronic works—amongst others, Rolf Gehlhaar's *Fünf deutsche Tänze* (1975), Karlheinz Stockhausen's *Sirius* (1975–76), and John McGuire's *Pulse Music III* (1978).

IRCAM

IRCAM in Paris became a major center for computer music research and realization and development of the Sogitec 4X computer system, featuring then revolutionary real-time digital signal processing. Pierre Boulez's *Répons* (1981) for 24 musicians and 6 soloists used the 4X to transform and route soloists to a loudspeaker system.

Rise of popular electronic music

Throughout the seventies bands such as The Residents and Can spearheaded an experimental music movement that incorporated electronic sounds. Other artists in the 1970s who composed primarily electronic instrumental music and managed to reach into the popular realm were Jean Michel Jarre, Tangerine Dream, Klaus Schulze, and

Vangelis. Also in the 1970s, rock bands from Genesis to The Cars began incorporating synthesizers into traditional rock arrangements.

In 1979, UK recording artist Gary Numan helped to bring electronic music into the wider marketplace of pop music with his hit "Cars" from the album *The Pleasure Principle*. Other successful hit electronic singles in the early 1980s included "Just Can't Get Enough" by Depeche Mode, "Don't You Want Me" by The Human League, "Whip It!" by Devo, and finally 1983's "Blue Monday" by New Order, which became the best-selling 12-inch single of all time. The Swiss duo Yello, Trevor Horn's Art of Noise, Naked Eyes, Prince, Kate Bush, Peter Gabriel, and Depeche Mode further incorporated early samplers like the Synclavier, Fairlight CMI, and E-mu Emulator into their hit records.

Birth of MIDI

In 1980, a group of musicians and music merchants met to standardize an interface by which new instruments could communicate control instructions with other instruments and the prevalent microcomputer. This standard was dubbed MIDI (Musical Instrument Digital Interface). A paper was authored by Dave Smith of Sequential Circuits and proposed to the Audio Engineering Society in 1981. Then, in August 1983, the MIDI Specification 1.0 was finalized.

The advent of MIDI technology allows a single keystroke, control wheel motion, pedal movement, or command from a microcomputer to activate every device in the studio remotely and in synchrony, with each device responding according to conditions predetermined by the composer.

MIDI instruments and software made powerful control of sophisticated instruments easily affordable by many studios and individuals. Acoustic sounds became reintegrated into studios via sampling and sampled-ROM-based instruments.

Miller Puckette developed graphic signal-processing software for 4X called Max (after Max Mathews) and later ported it to Macintosh (with Dave Zicarelli extending it for Opcode) for real-time MIDI control, bringing algorithmic composition availability to most composers with modest computer programming background.

Digital synthesis

In 1979 the Australian Fairlight company released the Fairlight CMI (Computer Musical Instrument) the first practical polyphonic digital synthesizer/sampler system. In 1983, Yamaha introduced the first stand-alone digital synthesizer, the DX-7. It used frequency modulation synthesis (FM synthesis), first experimented with by John Chowning at Stanford during the late sixties.

Barry Vercoe describes one of his experiences with early computer sounds:

At IRCAM in Paris in 1982, flutist Larry Beauregard had connected his flute to DiGiugno's 4X audio processor, enabling real-time pitch-following. On a Guggenheim at the time, I extended this concept to real-time score-following with automatic synchronized accompaniment, and over the next two years Larry and I gave numerous demonstrations of the computer as a chamber musician, playing Handel flute sonatas, Boulez's *Sonatine* for flute and piano and by 1984 my own *Synapse II* for flute and computer—the first piece ever composed expressly for such a setup. A major challenge was finding the right software constructs to support highly sensitive and responsive accompaniment. All of this was pre-MIDI, but the results were impressive even though heavy doses of tempo rubato would continually surprise my **Synthetic Performer**. In 1985 we solved the tempo rubato problem by incorporating *learning from rehearsals* (each time you played this way the machine would get better). We were also now tracking violin, since our brilliant, young flautist had contracted a fatal cancer. Moreover, this version used a new standard called MIDI, and here I was ably assisted by former student Miller Puckette, whose initial concepts for this task he later expanded into a program called **MAX**.

Late 1980s to 1990s

Rise of dance music

In the late 1980s, dance music records made using only electronic instruments became increasingly popular. The trend has continued to the present day with modern nightclubs worldwide regularly playing electronic dance music. Nowadays, electronic/dance music is so popular, that dedicated genre radio stations (e.g., RADIO 538, Q Radio) or TV Channels (e.g., NRJ Dance, MUSIC FORCE EUROPE) exist.

Advancements

In the 1990s, interactive computer-assisted performance started to become possible, with one example described as follows:

Automated Harmonization of Melody in Real Time: An interactive computer system, developed in collaboration with flutist/composer Pedro Eustache, for realtime melodic analysis and harmonic accompaniment. Based on a novel scheme of harmonization devised by Eustache, the software analyzes the tonal melodic function of incoming notes, and instantaneously performs an orchestrated harmonization of the melody. The software was originally designed for performance by Eustache on Yamaha WX7 wind controller, and was used in his composition *Tetelestai*, premiered in Irvine, California in March 1999.

Other recent developments included the Tod Machover (MIT and IRCAM) composition *Begin Again Again* for "hypercello", an interactive system of sensors measuring physical movements of the cellist. Max Mathews developed the "Conductor" program for real-time tempo, dynamic and timbre control of a pre-input electronic score. Morton Subotnick released a multimedia CD-ROM *All My Hummingbirds Have Alibis*.

2000s

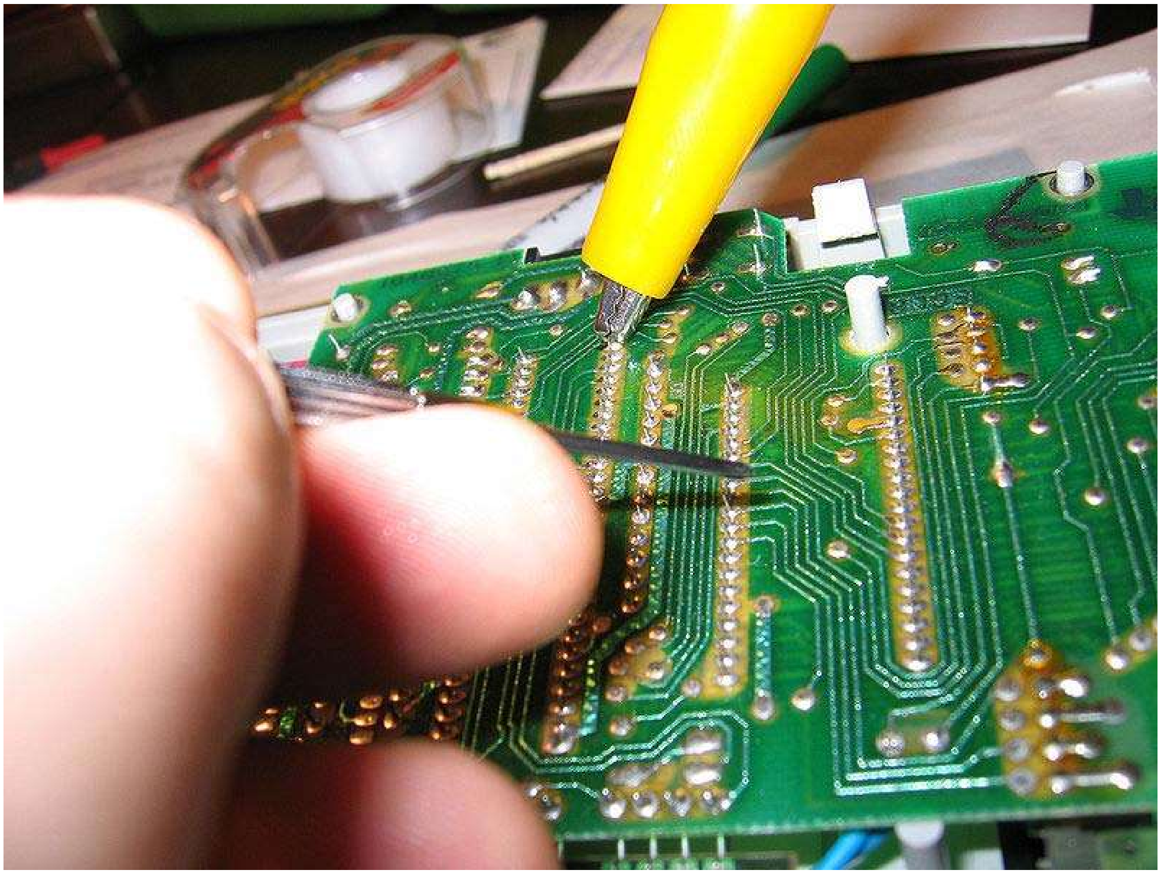
Qlimax, a large electronic music event that occurs each year in the Netherlands, celebrating the Hardstyle subgenre of electronic music

In recent years, as computer technology has become more accessible and music software has advanced, interacting with music production technology is now possible using means that bear no relationship to traditional musical performance practices: for instance, laptop performance (*laptronica*) and live coding. In general, the term Live PA refers to any live performance of electronic music, whether with laptops, synthesizers, or other devices.

In the last decade, a number of software-based virtual studio environments have emerged, with products such as Propellerhead's Reason and Ableton Live finding popular appeal. Such tools provide viable and cost-effective alternatives to typical hardware-based production studios, and thanks to advances in microprocessor technology, it is now possible to create high quality music using little more than a single laptop computer. Such advances have democratized music creation, leading to a massive increase in the amount of home-produced electronic music available to the general public via the internet.

Artists such as Fluker from Australia, can now also individuate their production practice by creating personalized software synthesizers, effects modules, and various composition environments. Devices that once existed exclusively in the hardware domain can easily have virtual counterparts. Some of the more popular software tools for achieving such ends are commercial releases such as Max/Msp and Reaktor and open source packages such as Pure Data, SuperCollider, and ChucK.

Chip Music



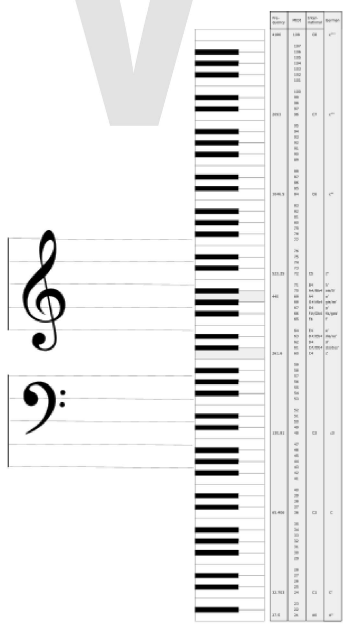
Probing for "bends" using a jeweler's screwdriver and alligator clips.

Chiptune, chipmusic, or chip music is music written in sound formats where many of the sound textures are synthesized or sequenced in real time by a computer or video game console sound chip, sometimes including sample-based synthesis and low bit sample playback. Many chip music devices featured synthesizers in tandem with low rate sample playback.

Chapter 9

Musical Instrument Digital Interface

MIDI is an industry-standard protocol that enables electronic musical instruments (synthesizers, drum machines), computers and other electronic equipment (MIDI controllers, sound cards, samplers) to communicate and synchronize with each other. Unlike analog devices, MIDI does not transmit an audio signal — it sends event messages about pitch and intensity, control signals for parameters such as volume, vibrato and panning, cues, and clock signals to set the tempo. As an electronic protocol, it is notable for its widespread adoption throughout the music industry. MIDI protocol was defined in 1982.



Note names and MIDI note numbers.

All MIDI-compatible controllers, musical instruments, and MIDI-compatible software follow the same MIDI 1.0 specification, and thus interpret any given MIDI message the same way, and so can communicate with and understand each other. MIDI composition and arrangement takes advantage of MIDI 1.0 and General MIDI (GM) technology to allow musical data files to be shared among many different devices due to some incompatibility with various electronic instruments by using a standard, portable set of commands and parameters. Because the music is stored as instructions rather than recorded audio waveforms, the data size of the files is quite small by comparison. Individual MIDI files can be traced through their own individual key code. This key code was established in early 1994 to combat piracy within the sharing of .mid files.

History

By the end of the 1970s, electronic musical devices were becoming increasingly common and affordable. However, devices from different manufacturers were generally not compatible with each other and could not be interconnected. Different interfacing models included analog control voltages at various standards (such as 1 volt per octave, or the logarithmic "hertz per volt"); analog clock, trigger and "gate" signals (both positive "V-trig" and negative "S-trig" varieties, between −15 V to +15 V); and proprietary digital interfaces such as Roland Corporation's DCB (digital control bus), the Oberheim system, and Yamaha's "keycode" system.

Following several months of discussion between US and Japanese manufacturers, in November 1981, audio engineer and synthesizer designer Dave Smith of Sequential Circuits, Inc. proposed a digital standard for musical instruments at the Audio Engineering Society show in New York. By the time of the January, 1983 Winter NAMM Show, Smith was able to demonstrate a MIDI connection between his Prophet 600 and a Roland JP-6. The MIDI Specification 1.0 was published in August 1983.

In the early 1980s, MIDI was a major factor in bringing an end to the "wall of synthesizers" phenomenon in progressive rock band concerts, when keyboard performers were often hidden behind huge banks of analog synthesizers and electric pianos. Following the advent of MIDI, many synthesizers were released in rack-mount versions, which meant that keyboardists could control many different instruments (e.g., synthesizers) from a single keyboard.

In the 1980s, MIDI facilitated the development of hardware and computer-based sequencers, which can be used to record, edit and play back performances. In the years immediately after the 1983 ratification of the MIDI specification, MIDI features were adapted to several early computer platforms including Apple II Plus and IIe, Apple Macintosh, Commodore 64, Commodore Amiga and the PC-DOS. This allowed the development of a market for powerful, inexpensive, and now-widespread computer-based MIDI sequencers. The standard Atari ST came equipped with MIDI ports and was commonly used in recording studios for this reason. Synchronization of MIDI sequences is made possible by the use of MIDI timecode, an implementation of the SMPTE time code standard using MIDI messages, and MIDI timecode has become the standard for digital music synchronization.

In 1991, the MIDI Show Control (MSC) protocol (in the Real Time System Exclusive subset) was ratified by the MIDI Manufacturers Association. The MSC protocol is an industry standard which allows all types of media control devices to talk with each other and with computers to perform show control functions in live and canned entertainment applications. Just like musical MIDI (above), MSC does not transmit the actual show media — it simply transmits digital data providing information such as the type, timing and numbering of technical cues called during a multimedia or live theatre performance.

Small file sizes made MIDI files a popular way of sharing music on the Internet in the early to mid 1990s, before broadband connections made it practical to share files in the MP3 format. Many gopher, and later web, sites hosted directories of MIDI files created by fans, thus avoiding the copyright issues that would later plague other forms of online music sharing.

MIDI initially made no provision for specifying timbre. In other words, each MIDI synthesizer had its own methods for producing the sound from MIDI instructions, with no standard sounds at all. For example, a producer might want a MIDI file played back through the Microsoft MIDI Synthesizer (included in any Windows operating system) to sound the same or similar on all machines. But because the quality of synthesis hardware might vary widely between machines — one might use a generic sound card, another might use professional-quality synthesis — there was no way to assure that what the listener heard was anything like what the producer intended.

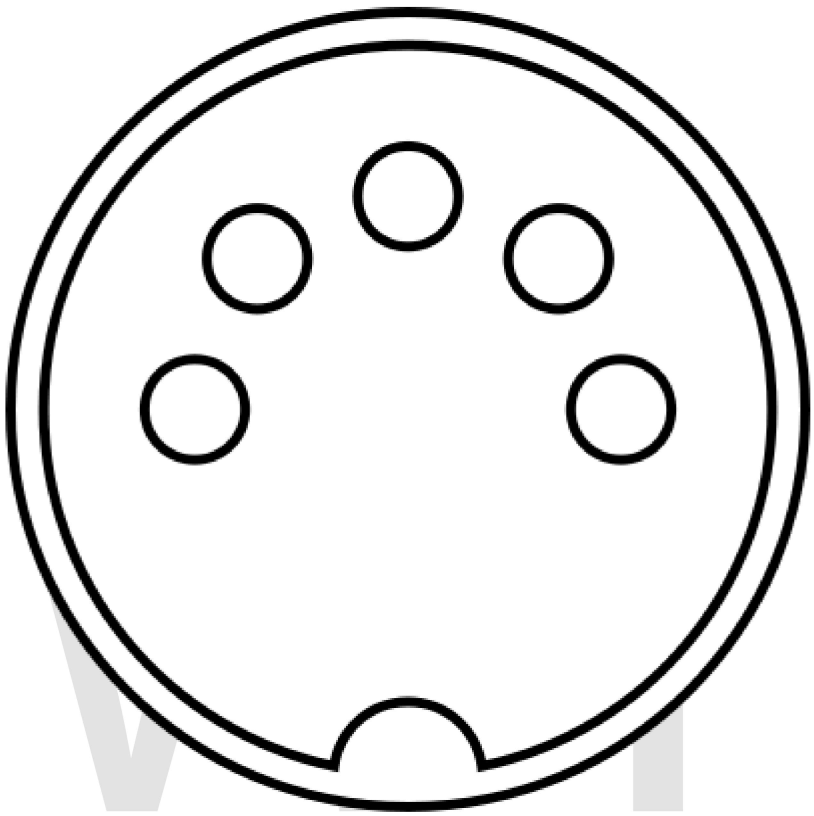
This situation was the impetus for the introduction of General MIDI in 1991. It created a standard set of 128 familiar sound *types* (piano, organ, guitar, strings). While manufacturers were still unable to decide what 'piano' sounded like, they at least had a standard to aim for and a location in which to place it.

In the early decades of MIDI, computer hardware was not able to play many samples or synthesize quality sounds. Quality hardware was too expensive; sound cards kept the price down, but many relied on unsophisticated synthesis methods to produce audio. As a result "the "MIDI sound" acquired a poor reputation with some critics.

Implications of MIDI

One of the implications of MIDI technology is the blurring of the roles for studios and instruments and the roles of musicians and producers. The adaptability of MIDI and the universal features has allowed for a rise not only in electronic music, but also 'bedroom producers'. The rise of MIDI has allowed musicians without formal engineering or production training to create their own music. Suddenly, a technology like MIDI allowed entire musical pieces to be recorded on a synthesizer or keyboard without the aid of producer. The keyboard becomes the studio and the musician the producer and vice versa. Over the past few decades there have always been cheap or affordable ways to record and produce music but never with such quality. MIDI allows the transfer of tracks between devices without degradation and allows an individual to create high quality professional tracks from their own home. MIDI has played an extremely large role in the ‘digitization’ of music.

Interfaces



MIDI connector diagram

MIDI connectors and a MIDI cable

The original physical MIDI connection uses DIN 5/180° connectors. Opto-isolating connections are used, to prevent ground loops occurring among connected MIDI devices.

The MIDI transceivers physically and logically separate the input and output lines, meaning that MIDI messages received by a device in the network not intended for that device must be re-transmitted on the output line (MIDI-OUT) by means of a "soft through". This can introduce a delay, one that is long enough to become musically significant on larger MIDI chains.

MIDI-THRU ports started to be added to MIDI-compatible equipment soon after the introduction of MIDI, in order to improve performance. The MIDI-THRU port avoids the aforementioned retransmission delay by linking the MIDI-THRU port to the MIDI-IN socket almost directly. The difference between the MIDI-OUT and MIDI-THRU ports is that data coming from the MIDI-OUT port has been generated on the device containing that port. Data that comes out of a device's MIDI-THRU port, however, is an exact duplicate of the data received at the MIDI-IN port.

Such chaining together of instruments via MIDI-THRU ports is unnecessary with the use of MIDI "patch bay," "mult" or "Thru" modules consisting of a MIDI-IN connector and multiple MIDI-OUT connectors to which multiple instruments are connected. *MIDI Thru Boxes* also clean up any skewing of MIDI data bits that might occur at the input stage.

Some equipment has the ability to merge MIDI messages into one stream; this is a specialized function and is not universal to all equipment. Such *MIDI Merge boxes* digitally merge all MIDI messages appearing at its inputs to its output, which allows a musician to plug in several MIDI controllers (e.g., two musical keyboards and a pedal keyboard) to a single synth voice device such as an EMU or Proteus.

All MIDI compatible instruments have a built-in MIDI. Some computers' sound cards have a built-in MIDI, whereas others require an external MIDI which is connected to the computer via the newer D-subminiature DA-15 game port, a USB connector or by FireWire, Ethernet or by MADI (RME standard). MIDI connectors are defined by the MIDI standard. In the 2000s, as computer equipment increasingly used USB connectors, companies began making USB-to-MIDI audio interfaces which can transfer MIDI channels to USB-equipped computers. As well, due to the increasing use of computers for music-making and composition, some MIDI keyboard controllers were equipped with USB jacks, so that they can be plugged into computers that are running "software synths" or other music software.

Controllers

In popular parlance, piano-style musical keyboards are called "keyboards", regardless of their functions or type. Amongst MIDI enthusiasts, however, keyboards and other devices used to trigger musical sounds are called "controllers", because with most MIDI set-ups, the keyboard or other device does not make any sounds by itself. MIDI controllers need to be connected to a voice bank or sound module in order to produce musical tones or sounds; the keyboard or other device is "controlling" the voice bank or sound module by acting as a trigger. The most common MIDI controller is the piano-style keyboard, either with weighted or semi-weighted keys, or with unweighted synth-style keys. Keyboard-style MIDI controllers are sold with as few as 25 keys (2 octaves), with larger models such as 49 keys, 61 keys, or even the full 88 keys being available.

MIDI controllers are also available in a range of other forms, such as electronic drum triggers; pedal keyboards that are played with the feet (e.g., with an organ); EWI wind controllers for performing saxophone-style music; and MIDI guitar synthesizer controllers. EWI, which stands for Electronic Wind Instrument, is designed for performers who want to play saxophone, clarinet, oboe, bassoon, and other wind instrument sounds with a synthesizer module. When wind instruments are played using a MIDI keyboard, it is hard to reproduce the expressive control found on wind instruments that can be generated with the wind pressure and embouchure. The EWI has an air-pressure level sensor and bite sensor in the mouthpiece, 13 touch sensors arrayed along the side of the controller, in a similar location to where sax keys are placed, and touch sensors for octaves and bends.

Pad controllers are used by musicians and DJs who make music through use of sampled sounds or short samples of music. Pad controllers often have banks of assignable pads and assignable faders and knobs for transmitting MIDI data or changes; the better-quality models are velocity-sensitive. More rarely, some performers use more specialized MIDI controllers, such as triggers that are affixed to their clothing or stage items (e.g., magicians Penn and Teller's stage show).

A MIDI foot-controller is a pedalboard-style device with rows of switches that control banks of presets, MIDI program change commands and send MIDI note numbers (some also do MIDI merges). Another specialized type of controller is the drawbar controller; it is designed for Hammond organ players who have MIDI-equipped organ voice modules. The drawbar controller provides the keyboard player with many of the controls which are found on a vintage 1940s or 1950s Hammond organ, including harmonic drawbars, a rotating speaker speed control switch, vibrato and chorus knobs, and percussion and overdrive controls. As with all controllers, the drawbar controller does not produce any sounds by itself; it only controls a voice module or software sound device.

While most controllers do not produce sounds, there are some exceptions. Some controller keyboards called "performance controllers" have MIDI-assignable keys, sliders, and knobs, which allow the controller to be used with a range of software synthesizers or voice modules; yet at the same time, the controller also has an internal voice module which supplies keyboard instrument sounds (piano, electric piano, clavichord), sampled or synthesized voices (strings, woodwinds), and Digital Signal Processing (distortion, compression, flanging, etc). These controller keyboards are designed to allow the performer to choose between the internal voices or external modules.

Messages

All MIDI compatible controllers, musical instruments, and MIDI-compatible software follow the same MIDI 1.0 specification, and thus interpret any given MIDI message the same way, and so can communicate with and understand each other. For example, if a note is played on a MIDI controller, it will sound at the right pitch on any MIDI instrument whose MIDI In connector is connected to the controller's MIDI Out connector.

When a musical performance is played on a MIDI instrument (or controller) it transmits *MIDI channel messages* from its MIDI Out connector. A typical MIDI channel message sequence corresponding to a key being struck and released on a keyboard is:

1. The user presses the middle C key with a specific *velocity* (which is usually translated into the volume of the note but can also be used by the synthesizer to set characteristics of the timbre as well). The instrument sends one *Note-On* message.
2. The user changes the pressure applied on the key while holding it down - a technique called *Aftertouch* (can be repeated, optional). The instrument sends one or more Aftertouch messages.
3. The user releases the middle C key, again with the possibility of velocity of release controlling some parameters. The instrument sends one *Note-Off* message.

Note-On, *Aftertouch*, and *Note-Off* are all channel messages. For the Note-On and Note-Off messages, the MIDI specification defines a number (from 0–127) for every possible note pitch (C, C♯, D etc.), and this number is included in the message.

Other performance parameters can be transmitted with channel messages, too. For example, if the user turns the pitch wheel on the instrument, that gesture is transmitted over MIDI using a series of *Pitch Bend* messages (also a channel message). The musical instrument generates the messages autonomously; all the musician has to do is play the notes (or make some other gesture that produces MIDI messages). This consistent, automated abstraction of the musical gesture could be considered the core of the MIDI standard.

Composition

MIDI composition and arrangement typically takes place using either MIDI sequencing/editing software on PC-type computers, or using specialized hardware music workstations. Some composers may take advantage of MIDI 1.0 and General MIDI (GM) technology to allow musical data files to be shared among various electronic instruments by using a standard, portable set of commands and parameters. On the other hand, composers of complex, detailed works to be distributed as produced audio typically use MIDI to control the performance of high-quality digital audio samples and/or external hardware or software synthesizers.

MIDI data files are much smaller than recorded audio waveforms. Many computer-sequencing programs allow manipulation of the musical data that composing for an entire orchestra of sounds is possible. This ability to manipulate musical data has also introduced the concept of surrogate orchestras, providing a combination of half sequenced MIDI recordings and half musicians to make up an entire orchestral arrangement; however, scholars believe surrogate orchestras have the possibility of effecting future live musical performances in which the use of live musicians in orchestral arrangements may terminate entirely because the composition of music via MIDI recordings proves to be more efficient and less expensive. Further, the data composed via the sequenced MIDI recordings can then be saved as a Standard MIDI File (SMF), digitally distributed, and reproduced by any computer or electronic instrument that also adheres to the same MIDI, GM, and SMF standards.

Another MIDI instrument that follows such standards of transferring musical compositions as previously mentioned is the MIDI harp. The MIDI harp possesses a piezo transducer that touches each string, allowing an electrical current to escape. The piezo pickup then outputs a current that is corresponding to the vibration when the string is plucked. Once this occurs, the MIDI harp's microprocessor that converts the analog signal to digital instruction devises a MIDI message that is sent by means of the MIDI-out port. In turn, the MIDI message can process the musician's "pluck" and decipher the volume of it and the duration of the "pluck." Once the harpist is satisfied with the music being created, the specific sounds are stored within the instrument's memory, similar to a computer file. The harpist may then proceed to transferring his or her composition by connecting the MIDI harp to a computer (preferably a PC). With sufficient software, the harpist can apply the use of the MIDI harp's "sustain petal" in which it will successfully transfer the harpist's composition measure-by-measure, and in its entirety.

Although a music distribution format, the Standard MIDI File was more attractive to computer users before broadband internet became widespread due to its much smaller file size. Also, the advent of high quality audio compression such as the MP3 format has decreased the relative size advantages of MIDI-encoded music to some degree, though MP3 is still much larger than SMF.

File formats

Standard MIDI (.mid or .smf)

MIDI messages (along with timing information) can be collected and stored in a computer file system, in what is commonly called a MIDI file, or more formally, a Standard MIDI File (SMF). The SMF specification was developed by, and is maintained by, the MIDI Manufacturers Association (MMA). MIDI files are typically created using computer-based sequencing software (or sometimes a hardware-based MIDI instrument or workstation) that organizes MIDI messages into one or more parallel "tracks" for independent recording and editing. In most sequencers, each track is assigned to a specific MIDI channel and/or a specific instrument *patch*; if the attached music synthesizer has a known instrument palette (for example because it conforms to the General MIDI standard), then the instrument for each track may be selected by name. Although most current MIDI sequencer software uses proprietary "session file" formats rather than SMF, almost all sequencers provide export or "Save As..." support for the SMF format.

An SMF consists of one header chunk and one or more track chunks. There exist three different SMF formats; the format of a given SMF is specified in its file header. A Format 0 file contains a single track and represents a single song performance. Format 1 may contain any number of tracks, enabling preservation of the sequencer track structure, and also represents a single song performance. Format 2 may have any number of tracks, each representing a separate song performance. Sequencers do not commonly support Format 2. Large collections of SMFs can be found on the web, most commonly with the extension `.mid` but occasionally with the `.smf`. These files are most frequently authored with the (rather dubious) assumption that they will only ever be played on General MIDI players.

A number of music file formats have been based on the MIDI bytestream. These formats are very compact; a file as small as 10 KiB can produce a full minute of music or more due to the fact that the file stores instructions on how to recreate the sound based on synthesis with a MIDI synthesizer rather than an exact waveform to be reproduced. A MIDI synthesizer could be built into an operating system, sound card, embedded device (e.g. hardware-based synthesizer) or a software-based synthesizer. The file format stores information on what note to play and when, or other important information such as possible pitch bend during the envelope of the note or the note's velocity. Small MIDI file sizes have also been advantageous for applications such as mobile phone ringtones, and some video games.

MIDI Karaoke (.kar)

MIDI-Karaoke (which uses the ".kar" file extension) files are an "unofficial" extension of MIDI files, used to add synchronized lyrics to standard MIDI files. SMF players play the music as they would a .mid file but do not display these lyrics unless they have specific support for .kar messages. These often display the lyrics synchronized with the music in "follow-the-bouncing-ball" or progressive highlighting of the lyric text fashion, essentially turning any PC into a karaoke machine. None of the MIDI-Karaoke file formats are maintained by any standardization body but they follow General MIDI standards.

XMF

The MMA has also defined (and AMEI has approved) a new family of file formats, XMF (Extensible Music File), some of which package SMF chunks with instrument data in DLS format (Downloadable Sounds, also an MMA/AMEI specification), to much the same effect as the MOD file format. The XMF container is a binary format (not XML-based, although the file extensions are similar).

RIFF-RMID

On Microsoft Windows, the system itself uses proprietary RIFF-based MIDI files with the `.rmi` extension. Note, Standard MIDI Files are not RIFF-compliant. A RIFF-RMID file, however, is simply a Standard MIDI File wrapped in a RIFF (Resource Interchange

File Format) chunk. For compatibility reasons many digital musicians overlook this format. One solution to this incompatibility is to extract the data part of the RIFF-RMID chunk, the result will be a regular Standard MIDI File. RIFF-RMID is not an official MMA/AMEI MIDI standard.

Extended RMID

In recommended practice RP-29 (), the MMA defined a method for bundling one Standard MIDI file (SMF) image with one Downloadable Sounds (DLS) image, using the RIFF container technology. However, this method was deprecated when the MMA introduced the Extensible Music Format (XMF) which, because of its many additional features, is generally preferred for MIDI-related resource-bundling purposes in the future.

Extended MIDI (.xmi)

The XMI format is a proprietary extension of the SMF format introduced by the Miles Sound System, a middleware driver library targeted at PC games. XMI is not an official MMA/AMEI MIDI standard.

Usage and applications

MMA and AMEI

MIDI technology was standardized and is maintained by the MIDI Manufacturers Association (MMA). All official MIDI standards are jointly developed and published by the MMA in Los Angeles, California, USA and for Japan, the MIDI Committee of the Association of Musical Electronics Industry (AMEI) in Tokyo**.**

Primary reference for MIDI is The Complete MIDI 1.0 Detailed Specification, document version 96.1, available only from MMA in English, or from AMEI in Japanese. Though the MMA site formerly offered free downloads of all MIDI specifications, links to the basic and general detailed specs have been removed. Printed documents can be purchased. However, considerable ancillary material is available at no cost on the website.

Extensions of the MIDI standard

Many extensions of the original official MIDI 1.0 spec have been standardized by MMA/JMSC.

General MIDI

The General MIDI Level 1 ("GM") specification defines the feature set important for MIDI content interoperability across multiple players. It addresses the indeterminacy of the basic MIDI 1.0 protocol standard regarding the meaning and behaviour of Program

Change and Control Change messages. Without GM, different synthesizers can, and actually do, sound completely different in response to the same MIDI messages.

The GM standard mandates:

- An assignment of specific instruments to each Program Number in Program Change messages (for example, Program Number 3 is "Electric Grand Piano")
- The mapping of several controller numbers to important effects
- Use of channel 10 for percussion only (a specific unpitched percussion sound in place of each note)
- Various minimum specifications such as number of simultaneous voices/notes and channels/parts

General MIDI 1 was introduced in 1991.

GM common misconceptions

Although the GM and GM2 specifications are dependent on the basic MIDI 1.0 protocol specification, they are separate standards from MIDI 1.0. As a result, MIDI products may legitimately implement MIDI 1.0 but not GM and/or GM2. Although GM is an important feature for MIDI content interoperability across multiple players, many important MIDI applications do not require such interoperability. For example, MIDI and the SMF format are used in professional music recording production where the MIDI file content will never be distributed, and custom or specialized synthesizers are used much more commonly than GM or GM2. As a direct consequence, not all SMF content is authored for GM or GM2 synthesizers. Because of the inherent risk of generating unintended and incorrect sounds as a result of playing any SMF- or MIDI-message stream on synthesizers other than originally intended, it can not be safely assumed that a given MIDI-message stream or MIDI file will be compatible, as a practical matter, with GM or GM2 synthesizers. In particular, it is a common misconception that all or nearly all SMF content anticipates being played using a GM- or GM2-compatible synthesizer. There is no such dependency in the actual MMA/AMEI specification; indeed, it is quite legitimate for SMF content to be written for non-GM synthesizers.

With the exception of RTP MIDI and the audio/sp-midi MIME type definition, there is currently no technical standard for indicating in advance what kind of synthesizer(s) a given SMF- or MIDI-message stream is intended to drive.

GS and XG

In order to improve upon the General MIDI standard, and to take advantage of the advancements in newer synthesizers, both Roland and Yamaha introduced new, proprietary, extended MIDI specifications – dubbed "GS" and "XG", respectively – along with numerous products based correspondingly upon them, designed with stricter requirements, new features, and backward compatibility with the GM specification. GS and XG are not mutually compatible, nor are they official MMA/AMEI MIDI standards. Adoption of each has been limited in general to its respective manufacturer; however, most popular MIDI/music software offerings now include them as built-in selectable options.

General MIDI level 2

Later, after the success of General MIDI was firmly established, member companies of Japan's AMEI developed the General MIDI Level 2 (GM2) specification:

- incorporating and harmonizing aspects of the Yamaha XG and Roland GS formats;
- further extending the instrument palette;
- specifying more message responses in detail;
- defining new messages for custom tuning scales and other new functionality, thereby * improving sound-editing features and quality.

In order to enable these new enhancements, new control messages needed to be incorporated into the MIDI specification; these include:

- controller directives,
- Registered Parameters (RPNs),
- MIDI tuning, and other
- MIDI Machine-Control (MMC) messages, including the notion of
- Universal Real-Time System-Exclusive (SysEx) messages.

The GM2 specs are maintained and published by the MMA and AMEI. General MIDI 2 was introduced in 1999 and is now implemented in many newer synthesizers.

SP-MIDI

Later still, GM2 became the basis of the instrument selection mechanism in Scalable Polyphony MIDI (SP-MIDI), a MIDI variant for mobile applications where different players may have different numbers of musical voices. SP-MIDI is a component of the 3GPP mobile phone terminal multimedia architecture, starting from release 5.

GM, GM2, and SP-MIDI are also the basis for selecting player-provided instruments in several of the MMA/AMEI XMF file formats (XMF Type 0, Type 1, and Mobile XMF), which allow extending the instrument palette with custom instruments in the Downloadable Sound (DLS) formats, addressing another major GM shortcoming.

Alternative tunings

By convention, most MIDI synthesizers generally default to the conventional Western 12-pitch-per-octave, equal temperament tuning system. This tuning system makes many types of music inaccessible, because they depend on different intonation systems. To address this issue in a standardized manner, in 1992 the MMA ratified the MIDI Tuning

Standard, or MTS. Instruments that support the MTS standard can be tuned to any desired tuning system by sending the MTS System Exclusive message (a Non-Real Time Sys Ex).

The MTS SysEx message uses a three-byte number format to specify a pitch in logarithmic form. This pitch number can be thought of as a three-digit number in base 128. To find the value of the pitch number *p* that encodes a given frequency *f*, use the following formula:

$$p = 69 + 12 \times \log_2 \left(\frac{f}{440\,\mathrm{Hz}} \right).$$

For a note in A440 equal temperament, this formula delivers the standard MIDI note number as used in the Note On and Note Off messages. Any other frequencies fill the space evenly. While support for MTS is at present not particularly widespread in commercial hardware instruments, it is nonetheless supported by some instruments and software, for example the free software programs TiMidity and Scala, as well as other microtuners.

MIDI Show Control

The MIDI Show Control (MSC) protocol (in the Real Time System Exclusive subset) is an industry standard ratified by the MIDI Manufacturers Association in 1991 which allows all types of media control devices to talk with each other and with computers to perform show control functions in live and canned entertainment applications. Just like musical MIDI (above), MSC does not transmit the actual show media — it simply transmits digital data providing information such as the type, timing and numbering of technical cues called during a multimedia or live theatre performance.

MSC can be seen with the creation of a Halloween haunted mansion designed by Brent Ross at his Mountain View, CA home in October 2007. The haunted mansion was solely run on MIDI in which Ross converted “real-time” recordings of MIDI musical notes and converted them into electrical signals to operate and turn pneumatic valves on and off. Using Cubase software for MIDI sequencing, Ross’s MIDI entertainment haunted mansion performance could be played, recorded and edited by manually pushing a specific control button. Ross could then access his “MIDI to switch” technology, further allowing him to send MIDI messages to turn a “note on or off” while controlling the activation of the various props, movements, theatrical lighting, and sounds for entertainment.

Console automation

Audio mixers can be controlled with MIDI during console automation.

Alternative hardware transports

In addition to the original 31.25 kbits/sec (baud is the signalling rate and is the reciprocal of the shortest signalling element; bits/sec is the data rate) current-loop transported on 5-pin DIN, other connectors have been used for the same electrical data, and transmission of MIDI streams in different forms over USB, IEEE 1394 a.k.a FireWire, and Ethernet is now common.

USB

A standard for MIDI over USB was developed in 1999 as a joint effort between IBM, Microsoft, Altec Lansing, Roland Corporation, and Philips. To transmit MIDI over USB a Cable Number and Cable Index are added to the message, and the result is encapsulated in a USB packet. The resulting USB message can be double the size of the native MIDI message. Since USB is over 15,000 times faster than MIDI (480,000 Kbits/sec vs 31.25 Kbits/sec,) USB has the potential to be much faster. However, due to the nature of USB there is more latency and jitter introduced that is usually in the range of 2 to 10 ms, or about 2 to 10 MIDI commands. Some comparisons done in the early part of the 2000s showed USB to slightly slower with higher latency, and this is still the case today. Despite the latency and jitter disadvantages, MIDI over USB is increasingly common on musical instruments.

XLR3

Some early MIDI implementations used XLR3 connectors in place of the 5-pin DIN. The use of XLR3 connectors allowed the use of standard low-impedance microphone cables as MIDI cables. As the 31.25 Kbits/sec current-loop requires only three conductors, there was no problem with the loss of two pins. An example of this use is the Octave-Plateau Voyetra-8 synthesizer.

Over a computer network

Compared to USB or FireWire, the computer network implementation of MIDI provides network routing capabilities, which are extremely useful in studio or stage environments (USB and FireWire are more restrictive in the connections between computers and devices). Ethernet is moreover capable of providing the high-bandwidth channel that earlier alternatives to MIDI (such as ZIPI) were intended to bring.

After the initial fight between different protocols (IEEE-P1639, MIDI-LAN, IETF RTP-MIDI), it appears that IETF's RTP MIDI specification for transport of MIDI streams over computer networks is now spreading faster and faster since more and more manufacturers are integrating RTP-MIDI in their products (Apple, CME, Kiss-Box, etc.). Mac OS X, Windows and Linux drivers are also available to make RTP MIDI devices appear as standard MIDI devices within these operating systems. Additionally, IEEE-P1639 is now a dead project. The other proprietary MIDI/IP protocols are slowly disappearing, since most of them require expensive licensing to be implemented (while RTP MIDI is completely open), or the MIDI implementation does not bring any real advantage (apart from speed) over original MIDI protocol.

RTP-MIDI transport protocol

The RTP-MIDI protocol has been officially released in public domain by IETF in December 2006 (IETF RFC4695). RTP-MIDI relies on the well-known RTP (Real Time Protocol) layer (most often running over UDP, but compatible with TCP also), widely used for real-time audio and video streaming over networks. The RTP layer is easy to implement and requires very little power from the microprocessor, while providing very useful information to the receiver (network latency, dropped packet detection, reordered packets, etc.). RTP-MIDI defines a specific payload type, that allows the receiver to identify MIDI streams.

RTP-MIDI does not alter the MIDI messages in any way (all messages defined in the MIDI norm are transported transparently over the network), but it adds additional features such as timestamping and sysex fragmentation. RTP-MIDI also adds a powerful 'journalling' mechanism that allows the receiver to detect and correct dropped MIDI messages.The first part of RTP-MIDI specification is mandatory for implementors and describes how MIDI messages are encapsulated within the RTP telegram. It also describes how the journalling system works. The journalling system is not mandatory (journalling is not very useful for LAN applications, but it is very important for WAN applications).

The second part of RTP-MIDI specification describes the session control mechanisms that allow multiple stations to synchronize across the network to exchange RTP-MIDI telegrams. This part is informational only, and it is not required.

RTP-MIDI is included in Apple's Mac OS X since 10.4 and iOS since 4.2, as standard MIDI ports (the RTP-MIDI ports appear in Macintosh applications as any other USB or FireWire port. Thus, any MIDI application running on Mac OS X is able to use the RTP-MIDI capabilities in a transparent way). However, Apple's developers considered the session control protocol described in IETF's specification to be too complex, and they created their own session control protocol. Since the session protocol uses a UDP port different from the main RTP-MIDI stream port, the two protocols do not interfere (so the RTP-MIDI implementation in Mac OS X fully complies to the IETF specification).

Apple's implementation has been used as reference by other MIDI manufacturers. A Windows XP RTP-MIDI driver for their own products only has been released by the Dutch company Kiss-Box , another Windows RTP-MIDI driver compatible to Windows XP up to Windows 7 (32bit and 64bit) has also been released and a Linux implementation is currently under development by the Grame association. So it seems probable that the Apple's implementation will become the "de-facto" standard (and could even become the MMA reference implementation).

Converting instruments to MIDI

Some older instruments, for example electronic organs built in the 1970s and 1980s, are becoming beyond repair, due to lack of spares and/or of technicians trained on such equipment. The best candidates for upgrade are what are referred to as "Console" sized, or have at least 2x keyboards of 61 notes, and at least a 25 note (preferably 32 note concave) pedal board. Smaller "Spinet" sized organs are probably not considered worthy of conversion. In some cases, they can be modified into MIDI instruments. Terms coined from *MIDI* + *modification* are often used, such as *midification* or *to midify*.

An old electronic organ could have almost all of its discrete component electronics replaced by modern circuitry which will cause the instrument to output MIDI signals. The instrument would then become a specialised MIDI keyboard. Its MIDI output would need to be fed to a MIDI engine of some sort.

In modern times new music keyboards have MIDI functions as standard and can be connected to the computers with a PC-to-MIDI circuit or simply via USB. Other forms of MIDI controllers include wind controllers, drums, guitars, accordion and many others.

Old synthesizers are not often modified to transmit MIDI but people sometimes modify them to receive it. The modification involves adding a circuit board that converts digital MIDI signals into analog control voltages, as well as a MIDI jack. The circuit boards are usually designed specially for one model of synthesizer and it takes some expertise to install them. This allows pre-MIDI analog synthesizers to be controlled by digital sequencers, whereas they formerly required the user to actually play them.

Other applications

MIDI 1.0 is also used as a control protocol in applications other than music, including:

- show control
- theatre lighting
- special effects
- sound design
- karaoke
- VJ-ing
- recording system synchronization
- audio processor control
- Digital DJing otherwise known as Controllerism
- computer networking, as solely demonstrated by the early first-person shooter game *MIDI Maze*, 1987
- animatronic figure control
- animation parameter control, as demonstrated by Apple Motion v2
- lighting control is accomplished through the MIDI Show Control protocol which was standardised in 1991.

Beyond MIDI 1.0

Although traditional MIDI connections work well for most purposes, a number of newer message protocols and hardware transports have been proposed over the years to try to take the idea to the next level. Some of the more notable efforts include:

OSC

The Open Sound Control (OSC) protocol was developed at CNMAT. OSC has been implemented in the well-known software synthesizer Reaktor, in other innovative projects including SuperCollider, Pure Data, Isadora, Max/MSP, Csound, vvvv, ChucK, Quartz Composer and LuaAV as well as in many general purpose programming languages such as C (liblo), Python (pyliblo), Haskell (hosc), Scheme (sosc) and Pure (pure-liblo). The Lemur Input Device, a customizable touch panel with MIDI controller-type functions, also uses OSC. OSC differs from MIDI 1.0 over traditional 5-pin DIN in that it can run at broadband speeds when sent over Ethernet connections. However, the differences are smaller compared to MIDI when run at broadband speeds over Ethernet connections. Few mainstream musical applications and no standalone instruments support the protocol so far, making whole-studio interoperability problematic. OSC is not owned by any private company; neither is it maintained by any standards organization. Since September 2007, there is a proposal for a common namespace within OSC for communication between controllers, synthesizers and hosts. This, too, would not be maintained by any standards organization.

mLAN

Yamaha has its mLAN protocol, which is based on the IEEE 1394 transport (also known as FireWire) and carries multiple MIDI 1.0 message channels and multiple audio channels. mLAN is not maintained by a standards organization as it is a proprietary protocol. mLAN is open for licensing, although covered by patents owned by Yamaha.

HD Protocol

Development of a version of MIDI for new products which is fully backward compatible is now under discussion in the MMA. First announced as "HD-MIDI" in 2005 and tentatively called "HD Protocol" since 2008, this new standard would support modern high-speed transports, provide greater range and/or resolution in data values, increase the number of Channels, and support the future introduction of entirely new kinds of messages. Representatives from all sizes and types of companies are involved, from the smallest speciality show control operations to the largest musical equipment manufacturers. No technical details or projected completion dates have been announced as of 2011, however the MMA indicates that a draft of the HD Protocol and an UDP-based transport is currently being reviewed by its members. Various transports have been proposed for use for the HD-Protocol physical layer, including a call for ACN to be used as the sole or primary transport in show control environments.

MIDI software

There is a wide range of MIDI software available such as auto accompaniment applications, notation programs, music teaching software, music producing, games, DJ/remix environments, etc.

Sample standard MIDI files

- Drum sample#1
- Drum sample#2
- Bass sample#1
- Bass sample#2
- A combination of the above four files, with piano, jazz guitar, a hi-hat and four extra measures added to complete the short song, in A minor.
- The above file being played on a MIDI-compatible synthesizer.

Chapter 10

Acousmatic Music and Effects Unit

Acousmatic music

Acousmatic music is a form of electroacoustic music that deals specifically with acousmatic sound as a compositional resource. The practice has a historical basis in musique concrète. It can be created using non-acoustic technology, exists only in a recorded format (as a fixed medium), and is composed for reception via loudspeakers. The compositional material is not restricted to the inclusion of sonorities derived from musical instruments or voices, nor to elements traditionally thought of as 'musical' (melody, harmony, rhythm, metre and so on), but rather admits any sound, acoustic or synthetic. With the aid of various technologies, such as tape recorders, digital signal processing tools and digital audio workstations, this material can then be combined, juxtaposed, and transformed, in any conceivable manner. In this context the compositional method can be seen as a process of *sound organisation*: a term first used by the French composer Edgard Varèse

Origins

The term *acousmatic* dates back to Pythagoras; the philosopher is believed to have tutored his students from behind a screen so as not to let his presence distract them from the content of his lectures. The term *acousmatique* was first used by the French composer Pierre Schaeffer. It is said to be derived from *akousmatikoi*, the outer circle of Pythagoras' disciples who only heard their teacher speaking from behind a veil. In a similar way, one hears acousmatic music from behind the 'veil' of loudspeakers, without seeing the source of the sound.

Developments

Within academia the term acousmatic music, or acousmatic art, has gained common usage, particularly when referring to contemporary musique concrète; however, there is some dispute as to whether acousmatic practice relates to a style of composition or a way of listening to sound. Scruton defines the experience of sound as inherently acousmatic, as Lydia Goehr (1999) paraphrases, "the sound world is not a space into which we can enter; it is a world we treat at a distance".

Style

Acousmatic music may contain sounds that have recognizably musical sources, but may equally present recognizable sources that are beyond the bounds of traditional vocal and instrumental technology. We are as likely to hear the sounds of a bird, or of a factory as we are the sounds of a violin. The technology involved transcends the mere reproduction of sounds. Techniques of synthesis and sound processing are employed which may present us with sounds that are unfamiliar and that may defy clear source attribution. Acousmatic compositions may present us with familiar musical events: chords, melodies and rhythms which are easily reconcilable with other forms of music, but may equally present us with events which cannot be classified within such a traditional taxonomy.

Performance practice

Acousmatic compositions are sometimes presented to audiences in concert settings that are often indistinguishable from acoustic recitals, albeit without performers. In an acousmatic concert the sound component is produced using pre-recorded media, or generated in real-time using a computer. The sound material will then be distributed spatially, via multiple loudspeakers, using a practice known as *diffusion*. The work is often *diffused* by the composer (if present) but the role of interpreter can also be assumed by another practitioner of the art. To provide a guideline for *spatialisation* of the work by an interpreter, many composers provide a *diffusion score*; in its simplest form this might be a graphic representation of the acousmatic work with indications for spatial manipulations, relative to a time-line.

Effects unit

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

Effects units are electronic devices that alter how a musical instrument or other audio source sounds. Some effects subtly "color" a sound, while others transform it dramatically. Effects can be used during live performances (typically with electric guitar, keyboard, or bass) or in the studio. While most frequently used with electric or electronic instruments, effects can also be used with acoustic instruments and drums. Examples of common effects units include wah-wah pedals, fuzzboxes, and reverb units.

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

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

Formats (form factor)

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

Stompboxes

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

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

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

Rackmounts

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

Built-in units

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

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

Multi-effects devices

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

Tabletop units

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

History

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

In 1948 DeArmond released the Trem-Trol, the first commercially available stand-alone effects unit. This device produced a tremolo by passing an instrument's electrical signal through a water-based electrolytic fluid. Most stand-alone effects of the 1950s and early 60s such as the Gibson GA-VI vibrato unit and the Fender reverb box, were expensive and impractical, requiring bulky transformers and high voltages. The original stand-alone units were not especially in-demand as many effects came built into amplifiers. The first popular stand-alone was the 1958 Watkins Copicat, a relatively portable tape echo effect made famous by the British band, The Shadows.

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

Distortion was not an effect originally intended by amplifier manufacturers, but could often easily be achieved by "overdriving" the power supply in early tube amplifiers. In the 1950s, guitarists such as Willie Johnson of Howlin' Wolf, Paul Burlison of Johnny Burnette & The Rock N Roll Trio and Link Wray deliberately increased gain beyond its intended levels to achieve "warm" distorted sounds. Wray's seminal 1958 recording "Rumble" inspired young musicians such as Pete Townshend of The Who, Jimmy Page of Led Zeppelin, Dave Davies of The Kinks, and Neil Young to explore distortion. Davies would famously doctor the speakers of his amp by slitting them with a razor blade to achieve an even grittier guitar sound on the 1964 song "You Really Got Me". In 1965, Marshall Amplification began selling the Marshall 1959, a guitar amplifier capable of producing the warm overtones and distorted "crunch" that rock musicians were starting to covet.

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

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

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

Types

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

Dynamics

A Guyatone VT2 Vintage Tremolo

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

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

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

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

Tone

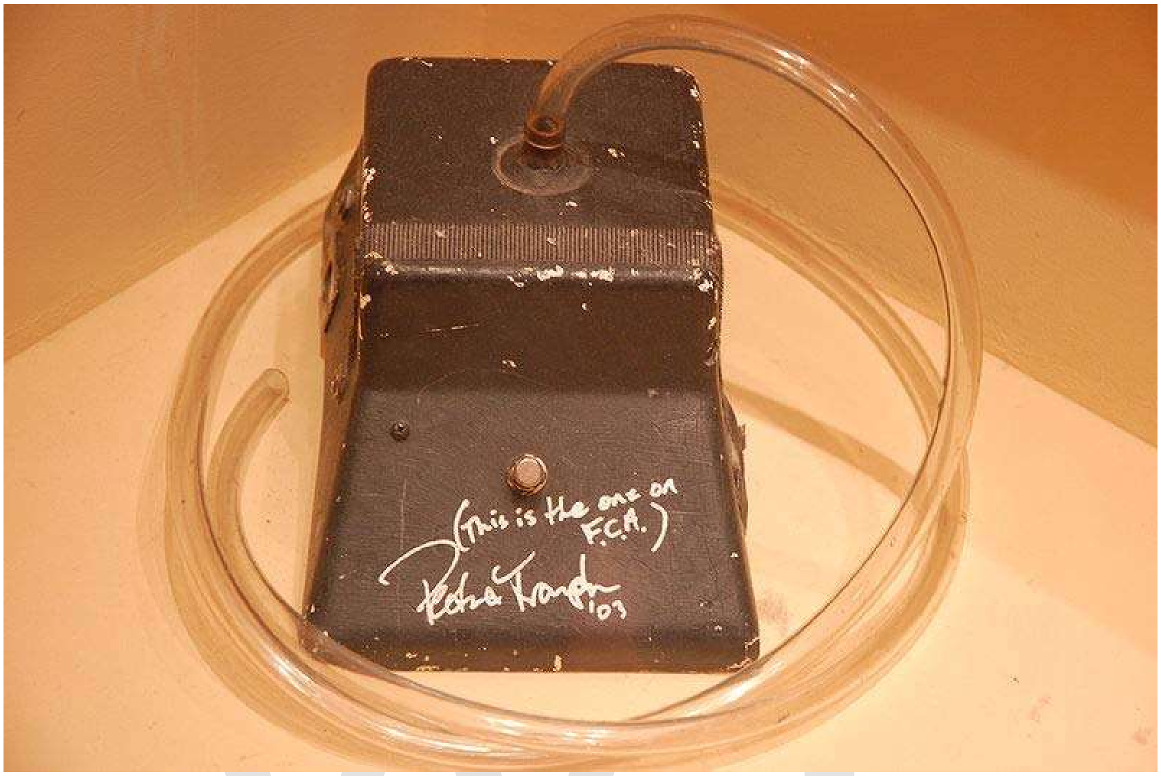
Distortion and Overdrive: Distortion and overdrive units distort the tone of an instrument by adding "overtones", creating a "warm" sound. To create a "dirty" or "gritty" sound, a unit further alters the tone by re-shaping or "clipping" its sound-waves so that they have flat, mesa-like peaks instead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

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

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

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

Filter



Peter Frampton's Talk box

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

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

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

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

Pitch/Frequency

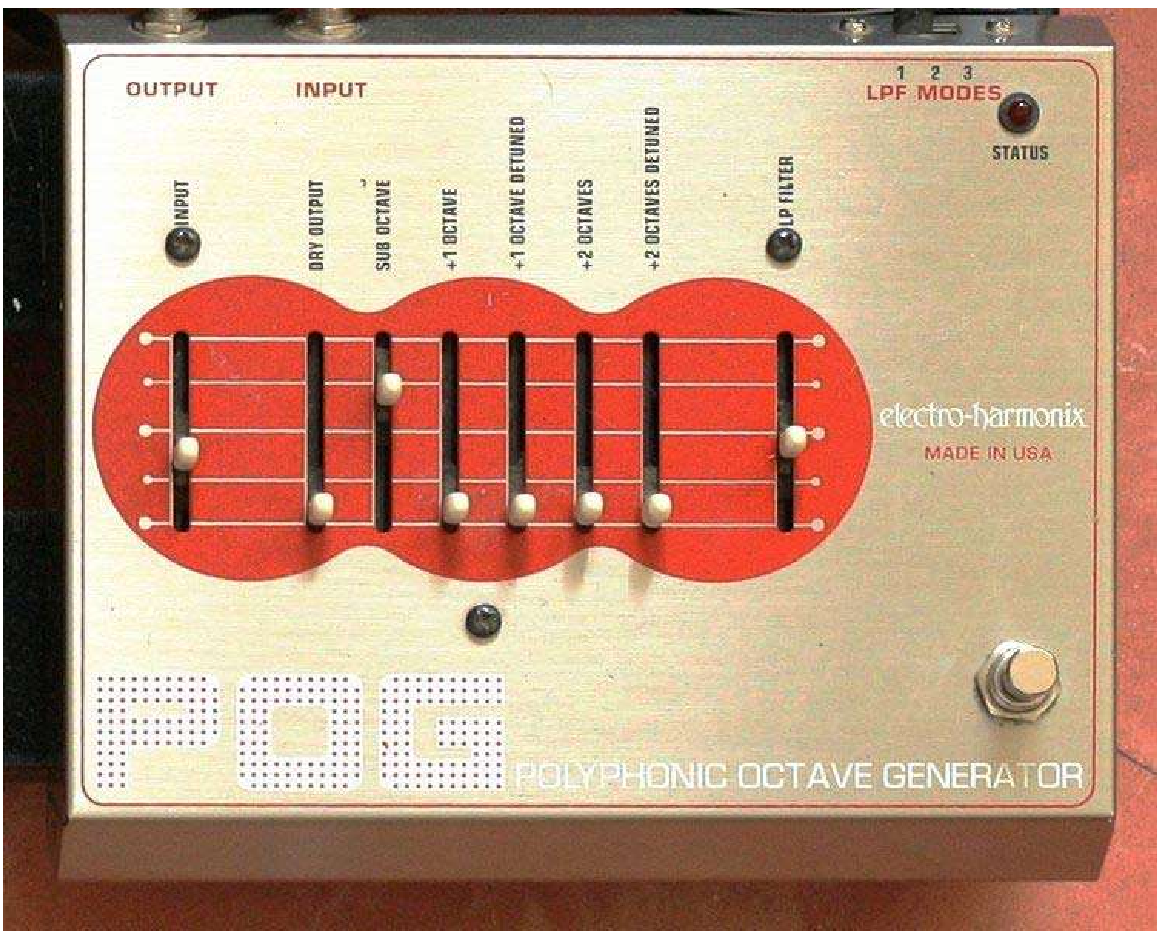
SmallClone chorus effect

Chorus: Chorus pedals mimics the effect produced naturally by choirs and string orchestras with very slight differences in timbre and pitch. A chorus effect splits the instrument-to-amplifier electrical signal, adding slight frequency variations or “vibrato” to part of the signal while leaving the rest unaltered. With extreme settings, a chorus effect can produce a "spacey" sound. A well-known usage of chorus is the lead guitar in “Come As You Are” by Nirvana. Chorus effects: Boss CE-1 Chorus Ensemble, Electro-

Harmonix Deluxe Memory Man, Electro-Harmonix Electric Mistress, Roger Mayer Voodoo Vibe, T.C. Electronic Stereo Chorus.

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

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



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

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

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

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

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

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

Time-based

Folded line reverberation device, which uses springs.

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

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

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

An EBow allows a guitar player to sustain a note.

Feedback/Sustain

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

The EBow, a handheld pickup/string driver, uses a small inductor coil to vibrate a guitar's strings, creating a bow-like sustained sound. Devices such as the Guitar Resonator, the Sustainiac Sustainer, and the Fernandes Sustainer create feedback by electrically vibrating (“driving”) the guitar strings while minimizing the highest-pitched overtones and providing true sustain.

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

Other effects

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

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

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

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

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

Boutique pedals

T-Rex brand "Mudhoney" overdrive pedal

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

Effects unit modification

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

Tributes by musicians

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

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

Fuzzbox and We're Gonna Use It were an all-female British band from the 1980s, and The Fuzztones were a 1980s garage rock revival band.

Other pedals and rackmount units

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