# The Media Streaming Journal

February 2017

Covering Audio and Video Internet Broadcasting

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**Welcome to The Media Streaming Journal**

Greetings,

Among the many people that have given me a kind word here or a swift kick in the posterior to get me motivated, I would like to take the opportunity to thank a fellow journalism student from the dark ages of time. I had the great opportunity of learning under the tutelage of Mr. Walt Bishop Esquire Extraordinaire. Not only was Walt a many of many words, but he was keen on the deadline and a steady hand on the electric typewriter. He planted a seed that would take a long time to take root, but it finally came to fruition with surprising results.

Thank you, Walt for taking the time and instilling a love for writing. It was a pleasure to work with you and learn the finer points of the art of writing for the mass communication medium.

It’s none of their business that you have to learn to write. Let them think you were born that way.

Ernest Hemingway

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

Namaste

David Childers

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The Grand Master of Digital Disaster
(Editor In Chief)

# The Media Streaming Journal

## What is in this edition of the Media Streaming Journal

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Edited By
David Childers

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Edited By
David Childers

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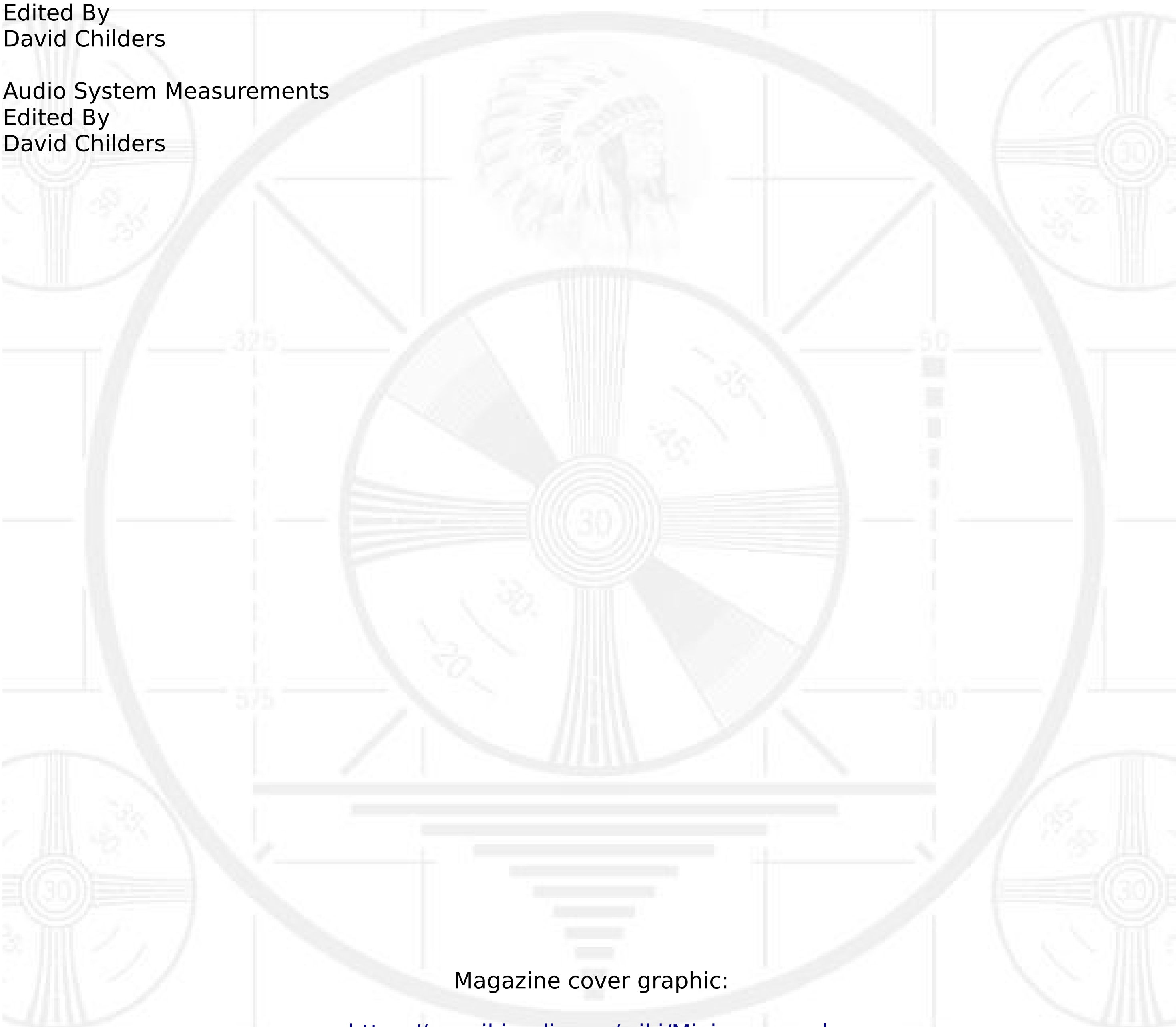
Edited By
David Childers



Magazine cover graphic:

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

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**Multitrack Recording**
**Edited By**
**David Childers**

Multitrack recording (MTR) also known as multitracking, double tracking, or tracking is a method of sound recording that allows for the separate recording of multiple sound sources or of sound sources recorded at different times to create a cohesive whole.

Process

Multi-tracking can be achieved with analog and digital recording equipment. Multi-track 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.

When recording, audio engineers can select which track (or tracks) on the device will be used for each instrument, voice, or other input and can even blend one track with two instruments to vary the music and sound options available. At any given point on the tape, any of the tracks on the recording device can be recorded or played back using sel-sync or Selective Synchronous recording. This ability 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 of track four 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 ability can be repeated until all of the available tracks have been used, or in some cases, reused. During mixdown a separate set of playback heads with higher fidelity are used.

Before all tracks are filled, 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. In 1963, The Beatles were using twin-track for Please Please Me. The 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 (circa 1965). Motown also began recording with 8-track machines in 1965 before moving to 16-track machines in mid-1969.

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 by making additional "takes" of songs or instrumental tracks. 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." (See "Punch in / 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 Low-Frequency Effects 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, such as when an older song is technically updated, this stereo (or mono) mixes can, in turn, be recorded (as if it were a "sub-mix") onto two (or one) tracks of a multitrack recorder. This ability allows 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. In the 2010s, many rock and pop bands record each part of the song one after the other. First, the bass and drums are often recorded, followed by the chordal rhythm section instruments. Then the lead vocals and guitar solos are added. As the last step, the harmony vocals are added. On the other hand, orchestras are always recorded with all 70 to 100 instrumentalists playing their parts simultaneously. If each group of instruments has their own microphone, and each instrument with a solo melody has its own microphone, the different microphones can record on multiple tracks simultaneously. After recording the orchestra, the record producer and conductor can adjust the balance and tone of the different instrument sections and solo instruments, because each section and solo instrument was recorded to its own track.

With the rock or pop band 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 lead vocals in isolation, one would do so by muting all the tracks apart from the lead 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 can retain complete control over the final sculpting of the song, during the mix-down (re-recording to two stereo tracks for mass distribution) phase.

Here is an example. An artist wanted to apply one effects unit to a synthesizer part, a different effect to a guitar part, a 'chorused reverb' effect to the lead vocals, and different effects on all the drums and percussion instruments.  This process 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 and vocal 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.

Since the early 2000s, many performers have recorded music using only a PC as a tracking machine. The computer must have a sound card or another type of digital audio interface with one or more Analog-to-digital converters. Multitrack recording software must be installed on the computer. Microphones are needed to record the sounds of vocalists or acoustic instruments. Depending on the capabilities of the system, some instruments, such as a synthesizer or electric guitar, can also be sent to an interface directly using Line level or other inputs. Direct inputs eliminate the need for microphones and can provide another range of sound control options.

There are tremendous differences in computer audio interfaces. Such units vary widely in price, sound quality, and flexibility. The most basic interfaces use audio circuitry that is built into the computer motherboard. The most sophisticated audio interfaces are external units of professional studio quality which can cost thousands of dollars. Professional interfaces usually use one or more IEEE 1394 (commonly known as FireWire) connections. Other types of interfaces may use internal PCI cards or external USB connections. Popular manufacturers of high-quality interfaces include Apogee Electronics, Avid Audio (formerly Digidesign), Echo Digital Audio, Focusrite, MOTU, RME Audio, M-Audio, and PreSonus.

Microphones are often designed for highly specific applications and have a major effect on recording quality. A single studio quality microphone can cost $5,000 or more, while consumer quality recording microphones can be bought for less than $50 each. Microphones also need a microphone preamplifier to prepare the signal for use by other equipment. These preamplifiers can also have a major effect on the sound and come in different price ranges, physical configurations, and capability levels. Microphone preamplifiers may be external units or a built-in feature of other audio equipment.

Software

Multitrack recording software can record multiple tracks at once. The software traditionally uses the graphic notation for an interface and offers numerous views of the music. Most multitrackers also provide audio playback capability. Some multitrack software also provides MIDI playback functions not just for audio; during playback, the MIDI data is sent to a softsynth or virtual instrument (e.g., VSTi) which converts the data to audio sound. Multitrack software may also provide other features that qualify it being called a digital audio workstation (DAW). These features may include various displays including showing the score of the music, as well as editing capability. There is often overlap between many of the categories of musical software. In this case, scorewriters and full featured multitrackers such as DAWs have similar features for playback but may have less similarity for editing and recording.

Multitrack recording software varies widely in price and capability. Popular multitrack recording software programs include the following: Propellerhead Reason, Ableton Live, FL Studio, Adobe Audition, Pro Tools, Digital Performer, Cakewalk Sonar, Samplitude, Nuendo, Cubase, and Logic. Mixcraft, REAPER and n-Track Studio are popular low-cost alternatives to more expensive and more versatile software options. Open Source and Free software programs are also available for multitrack recording. These range from very basic programs such as Audacity and Jokosher to Ardour, which is capable of performing many functions of the most sophisticated programs.

Instruments and voices are usually recorded as individual files on a computer hard drive. These function as tracks which can be added, removed or processed in many ways. Effects such as reverb, chorus, and delays can be applied by electronic devices or by computer software. Such effects are used to shape the sound as desired by the producer. When the producer is happy with the recorded sound finished tracks can be mixed into a new stereo pair of tracks within the multitrack recording software. Finally, the final stereo recording can be written to a CD, which can be copied and distributed.

## Order Of Recording

In modern popular songs, drums, percussion instruments, and electric bass are often among the first instruments to be recorded. These are the core instruments of the rhythm section. Musicians recording later tracks use the precise attack of the drum sounds as a rhythmic guide. In some styles, the drums may be recorded for a few bars and then looped. Click (metronome) tracks are also often used as the first sound to be recorded, especially when the drummer is not available for the initial recording, and/or the final mix will be synchronized with motion picture and/or video images. One reason that a band may start with just the drums is that this process allows the band to pick the song's key later on. The producer and the musicians can experiment with the song's key and arrangement against the basic rhythm track. 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 a large number of tracks utilized in a recording. This process 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 recorded are often the vocals (though a temporary vocal track may 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 by the accompaniment. Producers and songwriters can also use the guide/scratch vocal when they have not quite ironed out all the lyrics or for flexibility based on who sings the lead vocal (as The Alan Parsons Project's Eric Woolfson often did).

## Concert Music

For classical and jazz recordings, especially 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.

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

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**Audio Mixing**
**Edited By**
**David Childers**

Audio mixing is the process of combining multitrack recordings into a single track, and these tracks that are blended together are done so by using various processes such as EQ, Compression, and Reverb. The track may be mixed in mono, stereo, or surround sound. The are numerous approaches, methods, and techniques involved in Audio mixing; some of these practices include levels setting, equalization, stereo panning, and effects. Audio mixing techniques and approaches can vary widely, and these can greatly affect the qualities of the sound recording.

Audio mixing techniques largely depend on music genres and the quality of sound recordings involved. The process is generally carried out by a mixing engineer, though sometimes the musical producer or music artist may assist. After mixing, a mastering engineer prepares the final product for production.

Audio mixing may be used in conjunction with a mixing console or digital audio workstation.

**Equipment**

<u>Mixing consoles</u>

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

<u>Mixers offer three main functionalities</u>:

*Mixing* – summing signals together, which is normally done by a dedicated summing amplifier or in the case of digital by a simple algorithm.

<u>Routing</u> – allows the routing of source signals to internal buses or external processing units and effects.

<u>Processing</u> – many mixers also offer on-board processors, like equalizers and compressors.

Mixing consoles used for dubbing can often be seen as large and intimidating, due to the exceptional amount of controls. However, because many of these controls are duplicated, much of the console can be learned by studying one part of it. The controls on a mixing console will typically fall into one of two categories: processing and configuration. Processors are the controls used to manipulate the sound. These can vary in complexity, from simple internal level controls to sophisticated outboard reverberation units. Configuration controls deal with the signal routing from the input to the output of the console through the various processes.

Digital audio workstations (DAW) have many mixing features which potentially have more processes available than that of a major console. The distinction between a large console and a DAW equipped with a control surface is that a digital console will typically consist of dedicated digital signal processors for each channel. It is thus designed not to "overload" under the burden of signal processing, which may crash or lose signals. DAWs can dynamically assign resources like digital audio signal processing power but may run out if too many signal processes are in simultaneous use. This overload can be solved fairly easily by simply plugging more hardware into the DAW, although the cost of such an endeavor may begin to approach that of a major console.

<u>Outboard gear and plugins</u>

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

<u>Processors</u> – these devices are normally connected in series to the signal path, so the input signal is replaced with the processed signal (e.g. equalizers).

Effects – these can be considered as any unit that has an effect on the signal, the term is mostly used to describe units that are connected in parallel to the signal path, and therefore they add to the existing sounds but do not replace them. Examples would include reverb and delay.

**Multiple level controls in signal path**

A single signal can pass through a large number of level controls – such as an individual channel fader, subgroup master fader, master fader and monitor volume control. According to audio engineer Tomlinson Holman, problems are created due to the multiplicity of the controls. Each and every console has their own dynamic range, and it is important to utilize this correctly to avoid excessive noise or distortions. Attacking this problem – of the correct setting for the variety of controls - can be accomplished relatively quickly. Holman refers to the scale of the control as a clue for the solution of this problem. With 0 dB being the nominal setting of the controls, many have a "gain in hand," which goes above 0 dB. This means that one can turn it up from the nominal setting to have something that sounds clear. Other controls, such as sub masters and master level controls, are used for slight trims to the overall section-by-section balance or for the main fade-ins and fade-outs of the overall mix.

<u>Processes that affect levels</u>

<u>Faders</u> – used to attenuate or boost the level of signals.

<u>Pan pots</u> – A fundamental part of the configuration in recording console is panning. Pan pots are devices that place sound among the channels: L, C, R, LS, and RS. They are also used to pan signals to the left or right and in surround, and to the back or front.

<u>Compressors</u> – A device which automatically varies the volume range of tracks being mixed, so that one track is not obscured by another when a low volume level on the primary track coincides with a high volume level on a secondary track. Compressors are equipped with various controls to vary the volume range over which the action of compression occurs, the amount of compression, and how quickly or slowly the compressor acts.

<u>Expansion</u> – The Expansion device does exactly the opposite of what the compressor does. It increases the volume range of a source and may do so across a wide dynamic range or may be restricted to a narrower region by control functions. Restricting expansion to only low-level sounds helps to minimize noise. This function is often referred to as downward expansion, noise gating, or keying and reduces the level below a threshold set by a specific control. Noise gates have numerous audible problems. (e.g.: In a dialog recording with air conditioning noise in the background, the threshold of the noise gate may remove the air conditioner sound between lines of dialog which can create an exaggerated difference that could be much more noticeable than if the audio had been left unprocessed.)

<u>Limiters</u> – A limiter acts on signals above a certain threshold. Above a specific threshold, the level is controlled so that for each dB of increase in the input, the gain is reduced by the same amount. Therefore, the output level above the threshold will stay precisely the same, regardless of any increases in the input level. Limiters can be used to catch occasional events that might not otherwise be controlled, to bring them into a range in which the recording medium can handle the signal linearly.

These items discussed thus far affect the level of an audio signal. The most commonly used process is level control, which is used even on the simplest of mixers.

<u>Processes that affect frequency response</u>

Processes that primarily affect the frequency response of the signal are commonly seen as second in importance to level control. These processes clean the audio signal, enhance interchangeability between other signals, adjust for the loudness effect, and typically create a much more pleasant or deliberately worse sound. There are two principle frequency response processes – equalization and filtering.

<u>Equalizers</u> – This is the process of altering the frequency response in a manner similar to what tone controls do on a stereo system. Professional EQs dissect the audio spectrum into three or four parts which may be called the low-bass, mid-bass, mid-treble, and high-frequency controls.

Filters – Filters are used to eliminate certain frequencies from the output. Filters cab strip off any part of the audio spectrum. There are various types of filters. A high-pass filter (low-cut) is used to remove excessive room noise at low frequencies. A low-pass filter (high-cut) is used to help isolate a low-frequency instrument playing in a studio along with others. And a band-pass filter is a combination of high- and low-pass filters, also known as a telephone filter (because a sound lacking in high and low frequencies resembles the quality of sound transmitted and received by telephone).

Processes that affect time

Reverbs – Reverbs are used to simulate boundary reflections created in a real room, adding a sense of space and depth to otherwise 'dry' recordings. Another use is to distinguish among auditory objects. All sound having one reverberant character will be categorized together by human hearing in a process called auditory streaming. This aspect is an important feature in layering sound, in depth, from in front of the speaker to behind it.

Before the advent of electronic reverb and echo processing, physical means were used to generate the effects. An echo chamber, a large reverberant room, could be equipped with a speaker and at least two spaced microphones. Signals were then sent to the speaker and the reverberation generated in the room was picked up by the two microphones, constituting a "stereo return."

**Downmixing**

Downmixing is the process of converting a program with a multiple-channel configuration into a program with fewer channels. Common examples include downmixing from 5.1 surround sound to stereo, and stereo to mono. In the former case, the left and right surround channels are blended with the left and right front channels. The center channel is blended equally with the left and right channels. The LFE channel is either mixed with the front signals or not used. Because these are common scenarios, it is common practice to verify the sound of such downmixes during the production process to ensure stereo and mono compatibility.

The alternative channel configuration can be explicitly authored during the production process with multiple channel configurations provided for distribution. For example, a stereo mix can be put on DVDAudio discs or Super Audio CDs along with the surround mix. Alternatively, the program can be automatically downmixed by the end consumer's audio system. For example, a DVD player or sound card may downmix a surround sound program to stereophonic sound (two channels) for playback through two speakers.

**Mixing in surround sound**

Any device having several bus consoles (typically having eight or more buses) can be used to create a 5.1 surround sound mix, but this may be frustrating if the device is not designed to facilitate signal routing, panning, and processing in a surround sound environment. Whether working in an analog hardware, digital hardware, or DAW "in-the-box" mixing environment, the ability to pan mono or stereo sources and place effects in the 5.1 soundscape and monitor multiple output formats without difficulty can make the difference between a successful or compromised mix. Mixing in surround is very similar to mixing in stereo except that there are more speakers, placed to "surround" the listener. In addition to the horizontal panoramic options available in stereo, mixing in surround lets the mix engineer pan sources within a much wider and more enveloping environment. In a surround mix, sounds can appear to originate from much more or almost any direction depending on the number of speakers used, their placement and how audio is processed.

There are two common ways to approach mixing in surround:

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

<u>Complete Surround/All speakers are treated equally</u> – Instead of following the traditional ways of mixing in stereo, this much more liberal approach lets the mix engineer do anything he or she wants. Instruments can appear to originate from anywhere, or even spin around the listener. When done appropriately and with taste, interesting sonic experiences can be achieved, as was the case with James Guthrie's 5.1 mix of Pink Floyd's The Dark Side of the Moon, albeit with input from the band. This sound is a much different mix from the 1970s quadrophonic mix.

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

<u>MSS – Multi Stereo Surround</u> – This approach treats the speakers in a surround sound system as a multitude of stereo pairs. For example, a stereo recording of a piano, created using two microphones in an ORTF configuration, might have its left channel sent to the left rear speaker and its right channel sent to the center speaker. The piano might also be sent to a reverb having its left and right outputs sent to the left front speaker and right rear speaker, respectively. Additional elements of the song, such as an acoustic guitar recorded in stereo, might have its left and right channels sent to a different stereo pair such as the left front speaker and the right rear speaker with its reverb returning to yet another stereo pair, the left rear speaker, and the center speaker. Thus, multiple clean stereo recordings surround the listener without the smearing comb-filtering effects that often occur when the same or similar sources are sent to multiple speakers.

https://en.wikipedia.org/wiki/Audio_mixing_%28recorded_music%29

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www.pulsradio.com/trance/

**Audio System Measurements**
**Edited By**
**David Childers**

Audio system measurements are made for several purposes. Designers take measurements so that they can specify the performance of a piece of equipment. Maintenance engineers make them to ensure equipment is still working to specification, or to ensure that the cumulative defects of an audio path are within limits considered acceptable. Some aspects of measurement and specification relate only to intended usage. Audio system measurements often accommodate psychoacoustic principles to measure the system in a way that relates to human hearing.

## Subjectivity and frequency weighting

Subjectively valid methods came to prominence in consumer audio in the UK and Europe in the 1970s, when the introduction of compact cassette tape, DBX, and Dolby noise reduction techniques revealed the unsatisfactory nature of many basic engineering measurements. The specification of weighted CCIR-468 quasi-peak noise and weighted quasi-peak wow and flutter became particularly widely used, and attempts were made to find more valid methods for distortion measurement.

Measurements based on psychoacoustics, such as the measurement of noise, often use a weighting filter. It is well established that human hearing is more sensitive to some frequencies than others, as demonstrated by equal-loudness contours, but it is not well appreciated that these contours vary depending on the type of sound. The measured curves for pure tones, for instance, are different from those for random noise. The ear also responds less well to short bursts, below 100 to 200 ms, than to continuous sounds such that a quasi-peak detector has been found to give the most representative results when noise contains click or bursts, as is often the case for noise in digital systems.For these reasons, a set of subjectively valid measurement techniques have been devised and incorporated into BS, IEC, EBU and ITU standards. These methods of audio quality measurement are used by broadcast engineers throughout most of the world, as well as by some audio professionals, though the older A-weighting standard for continuous tones is still commonly used by others.

No single measurement can assess audio quality. Instead, engineers use a series of measurements to analyze various types of degradation that can reduce fidelity. Thus, when testing an analog tape machine, it is necessary to test for wow and flutter and tape speed variations over longer periods, as well as for distortion and noise. When testing a digital system, testing for speed variations is normally considered unnecessary because of the accuracy of clocks in digital circuitry, but testing for aliasing and timing jitter is often desirable, as these have caused audible degradation in many systems.

Once subjectively valid methods have been shown to correlate well with listening tests over a wide range of conditions; then such methods are typically adopted as preferred. Standard engineering methods are not always sufficient when comparing like with like. One CD player, for example, might have higher measured noise than another CD player when measured with an RMS method, or even an A-weighted RMS method, yet sound quieter and measure lower when 468-weighting is used. This method could be because it has more noise at high frequencies, or even at frequencies beyond 20 kHz, both of which are less important since human ears are less sensitive to them. This effect is how Dolby B works and why it was introduced. Cassette noise, which was predominately high frequency and unavoidable given the small size and speed of the recorded track could be made subjectively much less important. The noise sounded 10 dB quieter but failed to measure much better unless 468-weighting was used rather than A-weighting.

## Measurable performance

### Analog electrical

#### Frequency response (FR)

This measurement tells you over what frequency range output level for an audio component will remain reasonably constant (either within a specified decibel range, or no more than a certain number of dB from the amplitude at 1kHz). Some audio components such as tone controls are designed to adjust the loudness of signal content at particular frequencies, e.g., a bass control allows the attenuation or accentuation of low frequency signal content, in which case the specification may specify the frequency response is taken with tone controls "flat" or disabled. Preamplifiers may also contain equalizers, filters for example to play LPs requiring RIAA frequency response correction, in which case the specification may describe how closely the response matches the standard. By comparison, the frequency range is a term sometimes used of loudspeakers and other transducers to indicate the frequencies that are usable, without normally specifying a decibel range. Power bandwidth is also related to frequency response - indicating the range of frequencies usable at high power (since frequency response measurements are normally taken at low signal levels, where slew rate limitations or transformer saturation would not be a problem.

A component having a 'flat' frequency response will not change the weighting (i.e., intensity) of signal content across the specified frequency range. The frequency range often specified for audio components is between 20 Hz to 20 kHz, which broadly reflects the human hearing range. The highest audible frequency for most people is less than 20 kHz, with 16 kHz being more typical. Components with 'flat' frequency responses are often described as being linear. Most audio components are designed to be linear across their entire operating range. Well-designed solid-state amplifiers and CD players may have a frequency response that varies by only 0.2 dB between 20 Hz to 20 kHz. Loudspeakers tend to have a considerably less flat frequency responses than this.

Total harmonic distortion (THD)

Music material contains distinct tones, and some kinds of distortion involve spurious tones at double or triple the frequencies of those tones. Such harmonically related distortion is called harmonic distortion. For high fidelity, this is usually expected to be < 1% for electronic devices; mechanical elements such as loudspeakers usually have inescapable higher levels. Low distortion is relatively easy to achieve in electronics with the use of negative feedback, but the use of high levels of feedback in this manner has been the topic of much controversy among audiophiles. Essentially all loudspeakers produce more distortion than electronics, and 1–5% distortion is not unheard of at moderately loud listening levels. Human ears are less sensitive to distortion in the bass frequencies, and levels are usually expected to be under 10% at loud playback. The Distortion creates only even-order harmonics for a sine wave input is sometimes considered less bothersome than odd-order distortion.

Output power

Output power for amplifiers is ideally measured and quoted as maximum Root Mean Square (RMS) power output per channel, at a specified distortion level at a particular load. This measurement is considered the most meaningful measure of power available on music signals, though real, non-clipping music has a high peak-to-average ratio, and usually averages well below the maximum possible. The commonly given measurement of PMPO (peak music power out) is largely meaningless and often used in marketing literature; in the late 1960s there was much controversy over this point, and the US Government (FTA) required that RMS figures be quoted for all high fidelity equipment. Music power has been making a comeback in recent years.

Power specifications require the load impedance to be specified, and in some cases, two figures will be given (for instance, a power amplifier for loudspeakers will be typically measured at 4 and 8 ohms). Any amplifier will drive more current to a lower impedance load. For example, it will deliver more power into a 4-ohm load, as compared to 8-ohm, but it must not be assumed that it is capable of sustaining the additional current unless it is specified so. Power supply limitations may limit high current performance.

Intermodulation distortion (IMD)

Audio distortion that is not harmonically related to the signal being amplified is intermodulation distortion. It is a measure of the level of spurious signals resulting from an unwanted combination of different frequency input signals. This effect results from non-linearities in the system. Sufficiently high levels of negative feedback can reduce this effect in an amplifier. Many believe it is better to design electronics in a way to minimize feedback levels, though this is difficult to achieve while meeting other high accuracy requirements. Intermodulation in loudspeaker drivers is, as with harmonic distortion, almost always larger than in most electronics. IMD increases with cone excursion. Reducing a driver's bandwidth directly reduces IMD. This reduction is achieved by splitting the desired frequency range into separate bands and employing separate drivers for each band of frequencies and feeding them through a crossover filter network. Steep slope crossover filters are most effective at IMD reduction but may be too expensive to implement using high-current components and may introduce ringing distortion. Intermodulation distortion in multi-driver loudspeakers can be greatly reduced with the use of an active crossover, though it significantly increases system cost and complexity.

Noise

The level of unwanted noise generated by the system, or by interference from external sources added to the signal. Hum usually refers to noise only at power line frequencies (as opposed to broadband white noise), which is introduced through induction of power line signals into the inputs of gain stages. Or from inadequately regulated power supplies.

Crosstalk

The introduction of noise (from another signal channel) caused by ground currents, stray inductance or capacitance between components or lines. Crosstalk reduces, sometimes noticeably, the separation between channels (e.g., in a stereo system). A crosstalk measurement yields a figure in dB that is relative to a nominal level of a signal in the path receiving interference. Crosstalk is normally only a problem in equipment that processes multiple audio channels in the same chassis.

Common-mode rejection ratio (CMRR)

In a balanced audio system, there are equal, and opposing signals (difference-mode) in inputs and any interference imposed on both leads will be subtracted, canceling out that interference (i.e., the common-mode). CMRR is a measure of a system's ability to ignore such interference and especially hum at its input. It is only typically significant with long lines going to an input, or when some kinds of ground loop problems exist. Unbalanced inputs do not have common mode resistance; induced noise on their inputs appears directly as noise or hum.

Dynamic range and Signal-to-noise ratio (SNR)

The difference between the maximum level a component can accommodate and the noise level it produces. Input noise is not counted in this measurement. It is measured in dB.

Dynamic range refers to the ratio of maximum to minimum loudness in a given signal source (e.g., music or program material), and this measurement also quantifies the maximum dynamic range an audio system can carry. This measured range is the ratio (usually expressed in dB) between the noise floor of the device with no signal and the maximum signal (usually a sine wave) that can be output at a specified (low) distortion level.

Since the early 1990s, it has been recommended by several authorities including the Audio Engineering Society that measurements of dynamic range be made with an audio signal present. This measurement avoids questionable measurements based on the use of blank media or muting circuits.

Signal-to-noise ratio (SNR), however, is the ratio between the noise floor and an arbitrary reference level or alignment level. In "professional" recording equipment, this reference level is usually +4 dBu (IEC 60268-17), though sometimes 0 dBu (UK and Europe - EBU standard Alignment level). 'Test level', 'measurement level' and 'line-up level' mean different things, often leading to confusion. In "consumer" equipment, no standard exists, though −10 dBV and −6 dBu are common.

Different media characteristically exhibit different amounts of noise and headroom. Though the values vary widely between units, a typical analog cassette might give 60 dB, a CD almost 100 dB. Most modern quality amplifiers have >110 dB dynamic range, which approaches that of the human ear, usually taken as around 130 dB. See Program levels.

Phase distortion, Group delay, and Phase delay

A perfect audio component will maintain the phase coherency of a signal over the full range of frequencies. Phase distortion can be extremely difficult to reduce or eliminate. The human ear is largely insensitive to phase distortion, though it is exquisitely sensitive to relative phase relationships within heard sounds. The complex nature of our sensitivity to phase errors, coupled with the lack of a convenient test that delivers an easily understood quality rating, is the reason that it is not a part of conventional audio specifications. Multi-driver loudspeaker systems may have complex phase distortions, caused or corrected by crossovers, driver placement, and the phase behavior of the specific driver.

Transient response

A system may have low distortion for a steady-state signal, but not on sudden transients. In amplifiers, this problem can be traced to power supplies in some instances, to insufficient high frequency performance or excessive negative feedback. Related measurements are slew rate and rise time. Distortion in transient response can be hard to measure. Many otherwise good power amplifier designs have been found to have poor slew rates, by modern standards. In loudspeakers, transient response performance is affected by the mass and resonances of drivers and enclosures and by group delay and phase delay introduced by crossover filtering or marginal time alignment of the loudspeaker's drivers. Most loudspeakers generate significant amounts of transient distortion, though some designs are less prone to this (e.g. electrostatic loudspeakers, plasma arc tweeters, ribbon tweeters and horn enclosures with multiple entry points).

Damping factor

A higher number is typically believed to be better. This number is a measure of how well a power amplifier controls the undesired motion of a loudspeaker driver. An amplifier must be able to suppress resonances caused by mechanical motion (e.g., inertia) of a speaker cone, especially a low frequency driver with greater mass. For conventional loudspeaker drivers, this essentially involves ensuring that the output impedance of the amplifier is close to zero and that the speaker wires are sufficiently short and have a sufficiently large diameter. Damping factor is the ratio of the output impedance of an amplifier and connecting cables to the DC resistance of a voice coil, which means that long, high resistance speaker wires will reduce the damping factor. A damping factor of 20 or greater is considered adequate for live sound reinforcement systems, as the SPL of inertia-related driver movement is 26 dB less than signal level and won't be heard. Negative feedback in an amplifier lowers its effective output impedance and thus increases its damping factor.

Mechanical

Wow and flutter

These measurements are related to physical motion in a component, largely the drive mechanism of analog media, such as vinyl records and magnetic tape. "Wow" is slow speed (a few Hz) variation, caused by longer term drift of the drive motor speed, whereas "flutter" is faster speed (a few tens of Hz) variations, usually caused by mechanical defects such as out-of-roundness of the capstan of a tape transport mechanism. This measurement is given in as a percentage and a lower number is better.

Rumble

The measure of the low frequency (many tens of Hz) noise contributed by the turntable of an analog playback system. This noise is caused by imperfect bearings, uneven motor windings, vibrations in driving bands in some turntables, room vibrations (e.g., from traffic) that are transmitted by the turntable mounting and so to the phono cartridge. A lower frequency number is typically better.

Digital

Note that digital systems do not suffer from many of these effects at a signal level, though the same processes occur in the circuitry since the data being handled is symbolic. As long as the symbol survives the transfer between components, and can be perfectly regenerated (e.g., by pulse shaping techniques) the data itself is perfectly maintained. The data is typically buffered in a memory and is clocked out by a very precise crystal oscillator. The data usually does not degenerate as it passes through many stages because each stage regenerates new symbols for transmission.

Digital systems also have unique problems. Digitizing adds noise, which is measurable and depends on the audio bit depth of the system, regardless of other quality issues. Timing errors in sampling clocks (jitter) result in non-linear distortion (FM modulation) of the signal. One quality measurement for a digital system (Bit Error Rate) relates to the probability of an error in transmission or reception. Other metrics on the quality of the system are defined by the sample rate and bit depth. In general, digital systems are much less prone to error than analog systems; However, nearly all digital systems have analog inputs and/or outputs, and certainly all of those that interact with the analog world do so. These analog components of the digital system can suffer analog effects and potentially compromise the integrity of a well designed digital system.

<u>Jitter</u>

A measurement of the variation in a specific period (periodic jitter) and absolute timing (random jitter) between measured clock timing versus an ideal clock. Less jitter is typically better for sampling systems.

<u>Sample rate</u>

A specification of the rate at which measurements are taken of the analog signal. This rate is measured in samples per second or hertz. A higher sampling rate allows a greater total bandwidth or passband frequency response and allows less-steep anti-aliasing/anti-imaging filters to be used in the stop-band, which can, in turn, improve overall phase linearity in the pass-band.

<u>Bit depth</u>

In Pulse-code modulation audio, the bit depth is the number of bits of information in each sample. Quantization, a process used in digital audio sampling, creates an error in the reconstructed signal. The Signal-to-quantization-noise ratio is a multiple of the bit depth.

Audio CDs use a bit depth of 16-bits, while DVD-Video and Blu-ray discs can use 24-bit audio. The maximum dynamic range of a 16-bit system is about 96dB, while for 24 bit it is about 144 dB.

Dither can be used in audio mastering to randomize the quantization error, and some dither systems use Noise shaping to spectral shape of the quantization noise floor. The use of shaped dither can increase the effective dynamic range of 16-bit audio to around 120 dB.

<u>Sample accuracy/synchronization</u>

Not as much a specification as an ability. Since independent digital audio devices are each run by their own crystal oscillator, and no two crystals are precisely the same, the sample rate will be slightly different. This variation will cause the devices to drift apart over time. The effects of this can vary. If one digital device is used to monitor another digital device, this will cause dropouts or distortion in the audio, as one device will be producing more or less data than the other per unit time. If two independent devices record at the same time, one will lag the other more and more over time. This effect can be circumvented with a wordclock synchronization. It can also be corrected in the digital domain using a drift correction algorithm. Such an algorithm compares the relative rates of two or more devices and drops or adds samples from the streams of any devices that drift too far from the master device. Sample rate will also vary slightly over time, as crystals change in temperature, etc. See also clock recovery

<u>Linearity</u>

Differential non-linearity and integral non-linearity are two measurements of the accuracy of an analog-to-digital converter. This measures how close the threshold levels for each bit are to the theoretical equally-spaced levels.

<u>Automated sequence testing</u>

Sequence testing uses a specific sequence of test signals, for frequency response, noise, distortion, etc., generated and measured automatically to carry out a complete quality check on a piece of equipment or signal path. A single 32-second sequence was standardized by the EBU in 1985, incorporating 13 tones (40 Hz–15 kHz at −12 dB) for frequency response measurement, two tones for distortion (1024 Hz/60 Hz at +9 dB) plus crosstalk and compander tests. This sequence, which began with an 110-baud FSK signal for synchronizing purposes, also became CCITT standard O.33 in 1985.

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

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

<u>Unquantifiable</u>

Many audio components are tested for performance using objective and quantifiable measurements, e.g., THD, dynamic range and frequency response. Some take the view that objective measurements are useful and often relate well to subjective performance, i.e., the sound quality as experienced by the listener. Floyd Toole has extensively evaluated loudspeakers in acoustical engineering research. In a peer reviewed scientific journal, Toole has presented findings that subjects have a range of abilities to distinguish good loudspeakers from bad, and that blind listening tests are more reliable than sighted tests. He found that subjects can more accurately perceive differences in speaker quality during monaural playback through a single loudspeaker, whereas subjective perception of stereophonic sound is more influenced by room effects. One of Toole's papers showed that objective measurements of loudspeaker performance match subjective evaluations in listening tests.

Some argue that because human hearing and perception are not fully understood, listener experience should be valued above everything else. This tactic is often encountered in the high-end home audio world, where it is used to sell amplifiers with poor specifications. The usefulness of blind listening tests and common objective performance measurements, e.g., THD, are questioned. For instance, crossover distortion at a given THD is much more audible than clipping distortion at the same THD, since the harmonics produced are at higher frequencies. This measurement does not imply that the defect is somehow unquantifiable or unmeasurable; just that a single THD number is inadequate to specify it and must be interpreted with care. Taking THD measurements at different output levels would expose whether the distortion is clipping (which increases with level) or crossover (which decreases with level).

Whichever the view, it should be noted that some measurements have been traditionally used, despite having no objective value. For example, THD is the average of several harmonics that are equally weighted, even though research performed decades ago identifies that lower order harmonics are harder to hear at the same level, compared with higher order ones. Also, even order harmonics are said to be typically harder to hear than odd order. Several formulas that attempt to correlate THD with actual audibility have been published. However, none have gained mainstream use.

The mass market consumer magazine Stereophile promotes the claim that home audio enthusiasts prefer sighted tests than blind tests.

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