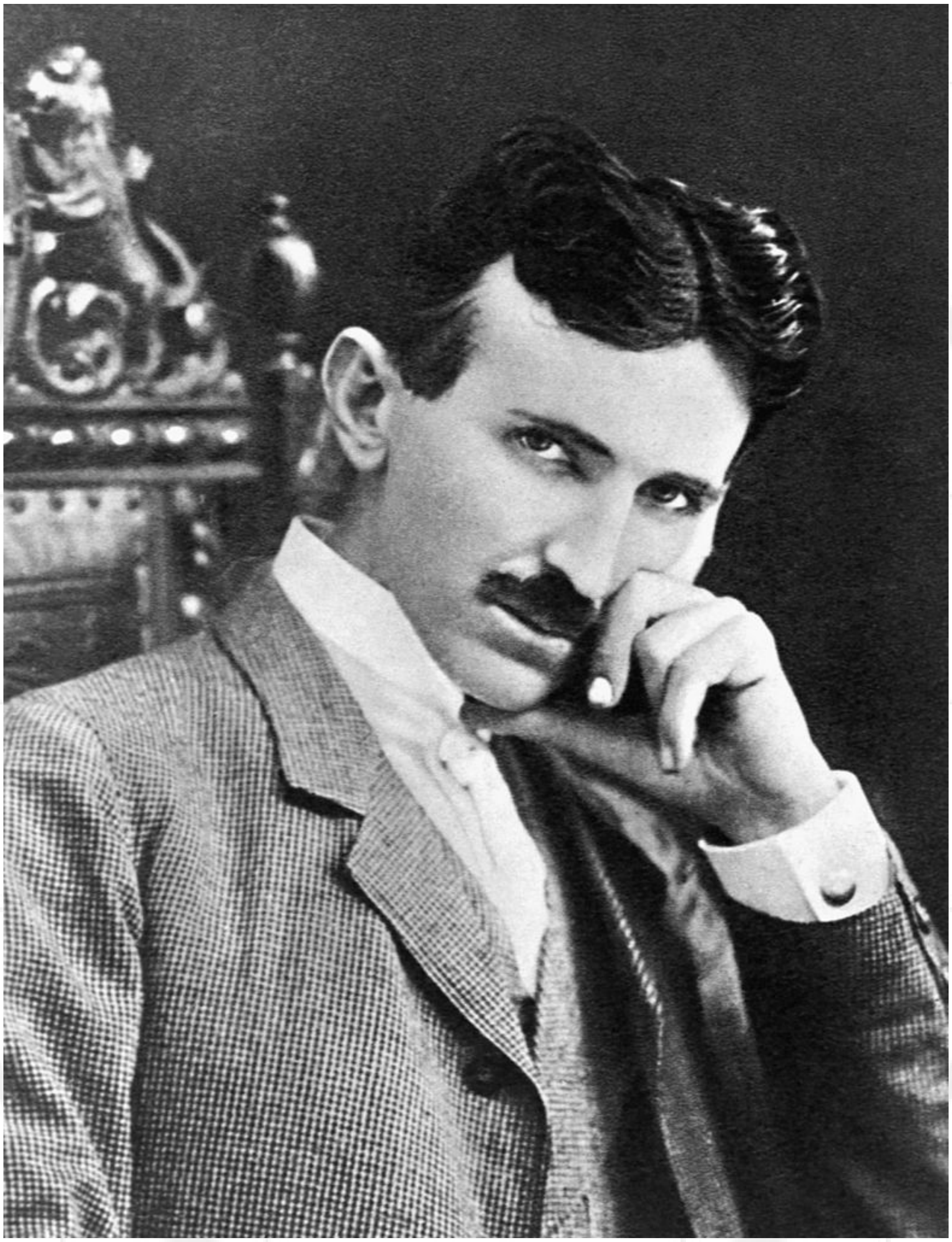
The Media Streaming Journal

October 2016



Covering Audio and Video Internet Broadcasting

Brought To You By

RADIOSOLUTION

www.radiosolution.info

publicdomainvectors.org/en/free-clipart/Vintage-microphone-vector-graphics/6111.html

The Media Streaming Journal Staff

Derek Bullard
Publication Director
info@radiosolution.info

David Childers
Editor In Chief
editor@radiosolution.info

Advertising
advertising@radiosolution.info

www.radiosolution.info

Welcome to The Media Streaming Journal

Greetings,

The weather is changing, and so are the colors of the seasons. With the upcoming elections in the United States, perhaps a voice of sanity can be brought forth and changes made to foster the online broadcast industry. With more people turning to the Internet for their daily musical fill, it is important to level the playing field and realize that Internet broadcasters are not the enemy that the recording associations portray them as. As more “artists” go public, competition for an audience becomes paramount. Without a medium to sustain public exposure of individual artists, how do these same artists expect to be recognized or even heard?

The ability to control the distribution medium provides the recording associations to control the artists, and yet these individual artists lack the foresight to see this. Thus the question arises, who is controlling whom.

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

www.linkedin.com/pub/david-childers/4/736/72a

The Grand Master of Digital Disaster
(Editor In Chief)

The Media Streaming Journal

What is in this edition of the Media Streaming Journal

Magazine cover graphic:

https://en.wikipedia.org/wiki/File:N.Tesla.JPG

RADIOSOLUTION

www.radiosolution.info

Our Mission

Let our friendly, knowledgeable staff assist you to build your project, such as an online radio station using our high end reliable video and audio streaming technologies. We want to become your partner for all your hosting needs, as well as your one stop shop for radio products such as custom DJ drops and radio ID's.

Start An Internet Radio Station

Whatever you need to start Internet radio station, we will deliver! We provide high quality Internet Radio services to make your music radio project a success. We can provide Wowza, Icecast, SHOUTcast hosting and internet radio services to hobbyists, deejays, amateurs and established professionals. No radio station client is too big or too small for Radiosolution.

Choose between complete hassle-free service packages or new features to add to start internet radio station. Benefit from customized services and the latest in internet radio technology. You will receive professional, personalized and better Internet Radio Station services than you have received up till now. If you already have an Icecast or SHOUTcast hosting provider, we can still help you transfer your radio server over to us with no hassle and at no charge.

Internet Radio Station Services

Launch your internet, digital, satellite or AM/FM radio station anywhere in the world with all of the right tools. A broadcasting specialist is on standby to help you get started with an SHOUTcast or Icecast hosting package. We have servers ready for reliable streaming in North America and Europe. Our hosting packages have all the features you need to make your radio station project a success.

If you stream live or with an Auto DJ, we can provide you with the latest in web-based Cloud technology. You will love the simple to use control panel. Discover how easy it is to manage live deejays, upload fresh music and create custom scheduled programming. You will be able to track your listeners by getting real time statistics.

Starting your own Internet radio has never been easier. Get in touch with us anytime to start your Internet radio station.

Radiosolution is a SHOUTcast hosting provider located in Quebec Canada. We also offer Icecast, Wowza and Web Hosting services. Contact us to discuss the best option available as you start internet radio station. Radiosolution can provide personalized service in English, Dutch, and French. Starting an internet radio station can be intimidating, many people want to start one, but have no idea where to start. Radiosolution will be there for you every step of the way. Everyday people are searching the internet for free SHOUTcast servers. With Radiosolution SHOUTcast hosting we will allow you to try our services for FREE. By trying our services, you can be confident that you have chosen the best radio server hosting provider. You have nothing to loose because we offer a 30 day satisfaction guarantee. What are you waiting for? Contact us now! Radiosolution offers everything you need to start internet radio station. You will not need to go anywhere else. We can create your website, market your station and help you submit your station to online directories. We also feature the voice of Derek Bullard aka Dibblebee He can create affordable commercials, DJ intros, sweepers, jingles, ids and so much more.

Social Media For Business
David Childers

Social Media is an Internet platform that allows friends to share information and stay in touch. This platform can also be used in a business environment to attract new customers while retaining the existing customer base. The key to business social media followers is that they can be guided into doing specific things or given a particular mindset regarding the individual business or the product/ services it delivers.

Most Social Media platforms allow a mix of content to be incorporated which enables the account users to use different approaches for presenting information. This ability is important as variety is the spice of life that will help to increase and retain a customer base.

When using Social Media for business you should consider several key concepts:

Identification of the target audience

It is important to define your target audience and participant communities because, very often, a social media engagement that has been designed for everyone ends up being for no one in particular. Successful films, television programs, newspapers or posters are never made for "everyone." On the other hand, a well-made social media engagement that targets a specific audience can very easily end up being liked by many different groups of people.

State the overall goal of the social media message

Every social media action should have a particular function and incorporate a call to action. The goal can be modified to conform to changes in the target audience, marketing requirements or promotional needs.

Crafting the message

Presenting a clear and concise message is a critical step in creating your strategy for connecting with consumers. What information do you want to convey and how will you present that information is important. Your message is what will pull people to you.

Remember that an effective message should:

- Present information clearly.
- Tell people something new or useful.
- Be engaging and informative.
- Inform the audience what action needs to taken and why they should do the action.

Measure the impact

It takes a lot of effort to ensure people will remember your message and take action on it. Measuring the impact of social media engagements is important so that you know what works and what does not.

Planning and Conducting Multimedia Influence Campaigns
David Childers
January 2015

There are several points also that should be used in the planning and execution of social media engagements.

These include:

Educate consumers about a business

This represents businesses branding.

Branding establishes a market identity for the business.

- What is the purpose of the business?
- What products does the business sell?
- What services does the business offer?
- Why is this business different?

Inform consumers about a business

This represents marketing.

Marketing enables the following:

- Attraction of new customers.
- Retention of satisfied customers.

Make announcements of business operation:

Sales of products.
Discuss services.
Discuss products.
Discuss hours of operation

Discuss business hours.
Discuss which days the business is open.
New business staff.

Call To Action

This actively encourages consumers to perform an immediate task.

State clearly what action you want people to take.

Use the KISS system, Keep It Simple Stupid.

You can also combine several call to actions, but remember to make the wording catchy and short. You can provide the consumer with more detailed information when they have performed the action.

- Click here to register.
- Order your coffee cup today.
- Watch the latest music video.
- Subscribe to the bands news letter now.

Entertain consumers

Give consumers a smile and brighten their day, they will remember that.

Post pictures, video, or audio from:

- Business events.
- Business staff.
- Industry news.
- General entertainment.

Consumer communication

Allow consumers to give feedback to the business.

- Suggestions for improvement.
- Problems encountered.
- General thoughts.

Consumer content share

Allow content to go viral.

- Encourage consumers to share posted items.

We need creative people working with broadcasters, making smart content to inspire people to be geniuses.

will.i.am

Audio Filters

These are a frequency dependent amplifier circuits that work in the audio frequency range, 0 Hz to beyond 20 kHz. Audio filters can amplify ("boost"), pass or attenuate ("cut") some frequency ranges. Many types of filters exist for different audio applications including hi-fi stereo systems, musical synthesizers, sound effects, sound reinforcement systems, instrument amplifiers and virtual reality systems.

Classification of filters

Active filters

These are implemented using a combination of passive and active (amplifying) components and require an outside power source. Operational amplifiers are frequently used in active filter designs. These can have high Q factor and can achieve resonance without the use of inductors. However, their upper frequency limit is limited by the bandwidth of the amplifiers.

Passive filters

These are passive implementations of linear filters which use combinations of resistors (R), inductors (L) and capacitors (C). These types of filters do not depend on an external power supply or they do not contain active components such as transistors.

Inductors block high-frequency signals and conduct low-frequency signals, while capacitors do the reverse. A filter in which the signal passes through an inductor, or in which a capacitor provides a path to ground, presents less attenuation to low-frequency signals than high-frequency signals and is therefore a low-pass filter. If the signal passes through a capacitor or has a path to ground through an inductor, then the filter presents less attenuation to high-frequency signals than low-frequency signals and therefore is a high-pass filter. Resistors on their own have no frequency-selective properties but are added to inductors and capacitors to determine the time-constants of the circuit, and therefore the frequencies to which it responds.

http://en.wikipedia.org/wiki/Electronic_filter

Types of audio filters

Low-pass

Common types include low-pass filters, which pass through frequencies below their cutoff frequencies and progressively attenuates frequencies above the cutoff frequency. Low-pass filters are used in audio crossovers to remove high-frequency content from signals being sent to a low-frequency subwoofer system.

High-pass

A high-pass filter does the opposite, passing high frequencies above the cutoff frequency, and progressively attenuating frequencies below the cutoff frequency. A high-pass filter can be used in an audio crossover to remove low-frequency content from a signal being sent to a tweeter.

Bandpass

A bandpass filter passes frequencies between its two cutoff frequencies while attenuating those outside the range. A band-reject filter attenuates frequencies between its two cutoff frequencies while passing those outside the 'reject' range.
All-pass

An all-pass filter passes all frequencies, but affects the phase of any given sinusoidal component, according to its frequency.

Applications

In some applications, such as in the design of graphic equalizers or CD players, the filters are designed according to a set of objective criteria. Examples include pass band, pass band attenuation, stop band, and stop band attenuation, where the pass bands are the frequency ranges for which audio is attenuated less than a specified maximum, and the stop bands are the frequency ranges for which the audio must be attenuated by a specified minimum. In more complex cases, an audio filter can provide a feedback loop, which introduces resonance (ringing) alongside attenuation. Audio filters can also be designed to provide gain (boost) as well as attenuation. In other applications, such as with synthesizers or sound effects, the aesthetic of the filter must be evaluated subjectively.

Audio filters can be implemented in analog circuitry as analog filters or DSP code or computer software as digital filters. Generically, the term 'audio filter' can be applied to mean anything which changes the timbre, or harmonic content of an audio signal.

http://en.wikipedia.org/wiki/Audio_filter

Other types of audio filters include:

Analogue filters

These are a basic building block of signal processing much used in electronics. Amongst their many applications are the separation of an audio signal before application to bass, mid-range and tweeter loudspeakers; the combining and later separation of multiple telephone conversations onto a single channel; the selection of a chosen radio station in a radio receiver and rejection of others.

Passive linear electronic analog filters are those filters which can be described with linear differential equations (linear); they are composed of capacitors, inductors and, sometimes, resistors (passive) and are designed to operate on continuously varying (analog) signals. There are many linear filters which are not analog in implementation (digital filter), and there are many electronic filters which may not have a passive topology – both of which may have the same transfer function of the filters described in this article. Analogue filters are most often used in wave filtering applications, that is, where it is required to pass particular frequency components and to reject others from analog (continuous-time) signals.

http://en.wikipedia.org/wiki/Analogue_filter

Digital filter

This is an electronic system that performs mathematical operations on a sampled, discrete-time signal to reduce or enhance certain aspects of that signal. This is in contrast to the other major type of electronic filter, the analog filter, which is an electronic circuit operating on continuous-time analog signals.

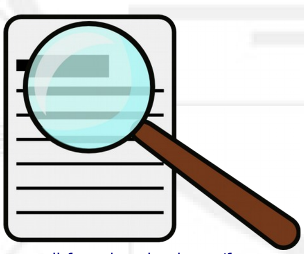
A digital filter system usually consists of an analog-to-digital converter to sample the input signal, followed by a microprocessor and some peripheral components such as memory to store data and filter coefficients, etc. Finally, a digital-to-analog converter to complete the output stage. Program Instructions (software) running on the microprocessor implement the digital filter by performing the necessary mathematical operations on the numbers received from the ADC. In some high-performance applications, an FPGA or ASIC is used instead of a general purpose microprocessor, or a specialized DSP with specific paralleled architecture for expediting operations such as filtering.

http://en.wikipedia.org/wiki/Digital_filter

Scenic Television is an Internet television station that presents the sights and sounds of nature 24 hours a day. Let us soothe and relax you wherever you are. Savor the tropical beaches of Puerto Rico or relax at a rain forest in Costa Rica. Meditate at the Danube River in Germany, or relish the view of Lake Zurich in Switzerland. We have scenic videos from locations all over the world.

Scenic Television originates from the Gulf coast of South Alabama and broadcasts to a global audience. The television broadcast is accessible on any device with an Internet connection. Such electronic devices include desktop computers, laptops, tablets, smartphones, game platforms, and Internet-connected televisions.

http://television.scenicradio.com



all-free-download.com/free-vector/download/magnifying_glass_clip_art_23181.html

We Are Your Information Resource

Are you looking for specialized data?

Are you swamped with information overload?

Do you need help finding the right information?

**We Can Help You
Find The Information
That You Need**

Our experienced data research analysts can wade through the vast information wasteland and find the information that you need.

We can save you both time and money.

We can streamline data requirement planning.

We can provide business critical information acquisition.

Contact us today

info@radiosolution.info

Power Conditioners

These are devices that are intended to improve the quality of the power that is delivered to electrical load equipment. While there is no official definition of a power conditioner, the term most often refers to a device that acts in one or more ways to deliver a voltage of the proper level and characteristics to enable load equipment to function properly. In some uses, power conditioner refers to a voltage regulator with at least one other function to improve power quality (e.g. power factor correction, noise suppression, transient impulse protection, etc.)

The terms "power conditioning" and "power conditioner" can be misleading, as the word "power" here refers to the electricity rather than the more technical electric power.

Conditioners specifically work to smooth the sinusoidal A.C. waveform and maintain a constant voltage over varying loads.

Types

An AC power conditioner is the typical power conditioner that provides "clean" AC power to sensitive electrical equipment. This is typically used for home or office applications and has up to 10 or more receptacles or outlets and commonly provides surge protection as well as noise filtering.

Power line conditioners take in power and modify it based on the requirements of the machinery to which they are connected. Attributes to be conditioned are measured with various devices, such as Phasor measurement units. Voltage spikes are most common during electrical storms or malfunctions in the main power lines. The surge protector stops the flow of electricity from reaching a machine by shutting off the power source.

The term "Power Conditioning" has been difficult to define historically. However, with the advances in power technology and recognition by IEEE, NEMA, and other standards organizations, a new actual engineering definition has now been developed and accepted to provide an accurate depiction of this definition.

"Power Conditioning" is the ability to filter the AC line signal provided by the power company. "Power Regulation" is the ability to take a signal from the local power company and turn it into a DC signal that will run an oscillator. This will generate a single frequency sine wave, determined by the local area needs, is fed to the input stage of a power amplifier, and is then output as specified as the ideal voltage present at any standard wall outlet.
Design

A good quality power conditioner is designed with internal filter banks to isolate the individual power outlets or receptacles on the power conditioner. This eliminates interference or "cross-talk" between components. If the application is a home theater system, the noise suppression rating listed in the technical specifications of the power conditioner will be very important. This rating is expressed in decibels (dB). The higher the dB rating, the better the noise suppression.

The power conditioner will also have a "joule" rating. A joule is a measurement of energy or heat required to sustain one watt for one second, known as a watt second. Since electrical surges are momentary spikes, the joule rating indicates how much electrical energy the suppressor can absorb at once before becoming damaged itself. The higher the joule rating, the greater the protection.
Uses

Power conditioners vary in function and size, generally according to their use. Some power conditioners provide minimal voltage regulation while others protect against six or more power quality problems. Units may be small enough to mount on a printed circuit board or large enough to protect an entire factory.

Small power conditioners are rated in volt-amperes (V·A) while larger units are rated in kilovolt-amperes (kV·A).

Ideally, electric power would be supplied as a sine wave with the amplitude and frequency given by national standards (in the case of mains) or system specifications (in the case of a power feed not directly attached to the mains) with an impedance of zero ohms at all frequencies.

No real life power feed will ever meet this ideal. Deviations may include:

- Variations in the peak or RMS voltage are both important to different types of equipment.

- When the RMS voltage exceeds the nominal voltage by 10 to 80% for 0.5 cycles to 1 minute, the event is called a "swell."

- A "dip" (in British English) or a "sag" (in American English – the two terms are equivalent) is the opposite situation: the RMS voltage is below the nominal voltage by 10 to 90% for 0.5 cycles to 1 minute.

- Random or repetitive variations in the RMS voltage between 90 and 110% of nominal can produce a phenomenon known as "flicker" in lighting equipment. Flicker is the impression of unsteadiness of visual sensation induced by a light stimulus on the human eye. A precise definition of such voltage fluctuations that produce flicker has been subject to ongoing debate in more than one scientific community for many years.

- Abrupt, very brief increases in voltage, called "spikes," "impulses," or "surges," generally caused by large inductive loads being turned off, or more severely by lightning.

- "Under-voltage" occurs when the nominal voltage drops below 90% for more than 1 minute. The term "brownout" in common usage has no formal definition but is commonly used to describe a reduction in system voltage by the utility or system operator to decrease demand or to increase system operating margins.

- "Over-voltage" occurs when the nominal voltage rises above 110% for more than 1 minute.

- Variations in the frequency.

- Variations in the wave shape – usually described as harmonics.

- Nonzero low-frequency impedance. (When a load draws more power, the voltage drops.)

- Nonzero high-frequency impedance. (When a load demands a large amount of current, then stops demanding it suddenly, there will be a dip or spike in the voltage due to the inductances in the power supply line.)

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

Audio System Measurements

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 take these measurements to ensure that the equipment is working at specification and 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, with the introduction of the 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 accurate method 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 measured lower when 468-weighting is used. This result could be due to 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. (See noise shaping.) 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

<u>Analog electrical</u>

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, 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. This frequency range 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. This 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 audio 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. See also Audio power.

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 extra current unless it is specified so. Power supply limitations may limit high current performance.

Intermodulation distortion (IMD)

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. These results are 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.

Noise

The level of unwanted noise generated by the system itself, 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 the separation between channels (e.g., in a stereo system). A crosstalk measurement yields a figure in dB relative to a nominal level of the 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 balanced audio systems, there are equal and opposite 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 generally only significant with long lines on 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 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 recommendation 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.

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 inadequate 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 inadequate 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 believed to be better. This measurement indicates 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.

<u>Mechanical</u>

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. These measurements are given as a percentage, and a lower reading is typically thought to be better.

Rumble

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

<u>Digital</u>

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 have their own 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 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.

Jitter

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

Sample rate

A specification of the rate at which measurements are obtained from the analog signal. This measurement is recorded in samples per second or hertz. A higher sampling rate allows a greater total bandwidth or pass-band 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.

Bit depth

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.

Sample accuracy/synchronization

Not as much a specification as an ability. Since independent digital audio devices are each run by their individual crystal oscillator, and no two crystals are exactly identical, 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

Linearity

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

Automated sequence testing

Sequence testing uses a specific sequence of test signals, for frequency response, noise, distortion, etc., generated and measured automatically to carry out a complete quality check on a piece of equipment or signal path. A single 32-second sequence was 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 a 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.

Unquantifiable

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 the 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 method 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 an average of a number of harmonics 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. Even order harmonics are said to be typically harder to hear than odd order. A number of 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