Power Handling Capacity of speakers

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What is power

- It is measured as energy produced over a defined period of time.
- We measure power from two aspects
- 1) output of amplifiers[overheat may cause problems]
- 2) Power handling capacity of loud speakers [calculated based on different frequencies]

Why loud speakers will fail

- ***Thermal failure: Loud speaker voice coil has a limit to the amount of signal it can handle at any point of time.***
- ***Excursion failure: This is physical type of failure. Cone or diaphragm is made to vibrate beyond its capacity.***

Specification of power

- RMS : Root Mean Square-average power of signal- heating capacity of the signal
- Peak: the maximum level that a signal attains. The excursion capability of loud speaker can be assessed by the peak power measure.
- Crest factor: pink noise signal ratio of the peak power to RMS. Crest factor of 6 db result peak voltage ie =2* RMS

Specification of power continue

- Total system power: total power the system consumes
- Peak music power output(PMPO) :different manufactures use different definitions. it is considered to be derived from some peak power of each amplifier in a system.

AES power handling specification

- ANSI has processed AES2-1984 aloud speakers power handling capacity
- <u>High frequency loudspeaker mounting</u>: A device which reasonably simulates the acoustical loading of a horn is used to mount the driver.
- <u>Low frequency loud speaker mounting: </u>the driver is mounted in free air such that the motion is in horizontal plane.

AES power handling continue

- Test signal : The test is done starting at the low frequency limit of the device using pink noise with crest factor of 6 db restricted to one octave
- Power calculation: the loud speaker measured on the RMS voltage which is then used to calculate the power and minimum impedance of the driver.

•AES power handling

- Duration of test : several tests are conducted using successively higher power levels. In b/w test loudspeaker is allowed to stabilize
- Rated power: the power it can endure for two hours with less than 10 % change in the acoustical, mechanical or electrical characteristics

Columns and Enclosures for speakers

- Speakers need to be kept in some sort of box or cabinet, referred to as column or enclosures
- The enclosure also contains electronic hardware such as crossover circuits and amplifiers.

The purpose of Enclosures

1) Avoid interaction of sound waves .

- Sound generated by the rearward facing surface of the diaphragm and at the front of driver
- 2) Avoid introducing echo and reverberation effects that are not part of the original sound
- 3)Restrict any vibration caused by the driver, any moving air mass and heat generated by driver voice coils and amplifiers.

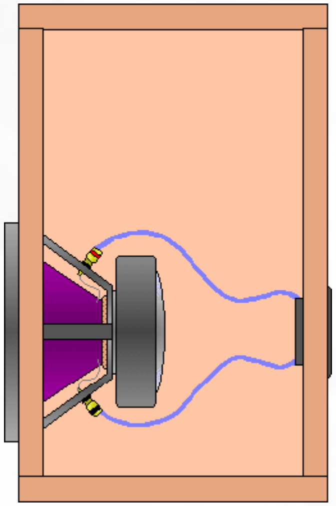
- 4 house the feet fixed to the base

Types of Enclosures

- Designing an enclosure is a fine balancing act between low bass extension, linear frequency response, efficiency, distortion, loudness, enclosure size

Sealed(closed) enclosure

- A resonant frequency of a driver is determined by its moving mass and compliance
- Damping properties of the system affect the low frequency response of sealed box systems
- Fiber glass ,long fiber wool



•Infinite baffle

- "**Infinite baffle**" or simply "IB" is also used as a generic term for sealed enclosures of any size, the name being used because of the ability of a sealed enclosure to prevent any interaction between the forward and rear radiation of a driver at low frequencies   image
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Acoustic suspension

- This variation of the closed-box enclosure has box size that can utilise the resulting linear air spring.
- An exceptionally complaint woofer suspension along with the air inside the enclosure restores the cone to neutral position.

Isobaric loading

- Two identical woofers operating simultaneously in a common box along one side of each diaphragm.
- They contribute to improve low-end frequency without increasing cabinet size.
- Cost and weight go up

•Bass reflex

- Also called vented system.
- It transmits low-frequency energy from rear of the speaker to listener with the use of cabinet openings
- The interior may be empty line filled or stuffed with damping materials

•Passive radiator

- Similar to ported enclosure in that it produces low frequency extension or efficiency increase
- The passive driver moves in response to changing enclosure pressures and is not connected to an amplifier

•Compound and bandpass

- It contains vented box that simulates 4^{th} order electrical band pass filter where the rear face of the driver cone pressure is trapped in a sealed box
- Enclosure has two chambers with only one that is ported and the driver is mounted on the dividing wall

Aperiodic enclosures

- Design wise it comes somewhere between suspension and bass reflex enclosures
- It improves low frequency reproduction

Damping in the port is done by blocking the port precisely with sufficiently tightly packed fiber filling

•Horn enclosure

- Here we use horn similar to driver cone that improves the coupling between the speaker driver and air
- The driver appears to have higher efficiency and helps control dispersion at higher frequencies

•Multiple entry horn

- It has aliases like coentrant horn, unity horn and synergy horn.
- At stepped distances from the horn’s apex several different drivers are mounted

•Tapped horn

- It has two sides of long-excursion high power driver ported with different path lengths which combine in phase at the horn’s mouth restricted to the required frequency range.

•Transmission line

- It come with infinitely long line and are stuffed with absorbent material to completely absorb the rear radiation of the driver up to the lowest frequencies.
- Hear combination of materials are used to overcome the insufficient stuffing

•Quarter wave enclosure

- A well tuned transmission line that , at a frequency somewhat below the drivers frequency forms a standing quarter wave is a quarter wave resonator.

•Tapered quarter –wave pipe

- It is a combination of transmission line and horn effects forms a tapered quarter –wave pipe
- It works on the premise that the tampering tube will progressively reflect and absorb the sound emitted from the rear of the loudspeaker

•Crossover networks in columns

- In multi driver speaker systems each driver works on a different frequency range.
- A single input signal needs to be separated according too these different frequency range
- This is the work of crossover
- This ensures minimum distrotion in the drivers and eliminates interference.

•Cross over network continue

- The following are not possible
- ✓ absolute block at the edges of the passband
- ✓ No amplitude variation within the passband
- ✓ No phase changes across the frequency band boundaries the crossover establishes

•Cross over network continue

- If there is requirement to mix the bands then the crossover splits the incoming audio signal,.
- The separate bands must not overlap

•Different types of crossover

- 1)classification based on the number of filter sections
- 2)classification based on components
- 3)classification based on filter order or slope
- 4) classification based on circuit topology

1. Classification based on the number of filter sections

- N-way speaker
- N denotes number of filter sections
- Three types of filters

1 Low Band Filters(LBF)

2 High-Pass filters(HPF)

3) band-pass filter(BPF)

Classification based on the number of filter sections continue

- BPF is a combination of HPF and LPF sections
- 2 way speaker contain an LBF and HBF
- 3 way speaker have a combination of LBF, BPF and HBF
- For protection an N way may have an extra HPF section which will safe guard the lowest frequency driver from frequencies lower than it can safely handle

2.Classification based on components

- Passive
- Active
- Digital
- Mechanical

2.1Passive

- It use resistors, capacitors and inductors
- it is expensive
- To protect from accidental overpowering it uses fuses, PTC devices bulbs or circuit breakers.

The draw back of passive network

- They are bulky
- Cause power loss
- They are frequency specific
- They are also impedance specific

•2.2Active

- The name come from the active components used it its filters
- Power amplifier input levels are the basis of active crossover operations
- Each output band requires the use of power amplifiers
- 2 amplifier
- One for woofer and one for tweeter

The advantages of active crossovers

- The frequency response depend/affect by dynamic changes in driver electrical characteristics.
- Each frequency band can be fine tuned to the specific drivers used easily
- Reduction in intermodulation distortion and overriding

•Advantages of active continue

- Amplifier damping control of the speaker maximized because the speaker drivers are directly connected to the power amplifiers
- There is no loss of energy as result power requirement is low
- Reduction in costs and increase in quality

•2.3Digital

- A DSP chip or other microprocessors can be used to implement it
- IIR filters: These are digital approximations to traditional analog circuits
- Similar to analog filters
- Do not require CPU power
- They are recursive

•Digital continue

- Finite impulse response (FIR) filters : it require more recourses for similar characteristics
- Longer delay time is incurred to achieve linear phase response

.2.4Mechanical

- Filtering is achieved from the properties of the materials in a driver diaphragm
- This type of filter uses the dust cap as a high frequency radiator as it radiates low frequencies
- The low mass and reduced damping increases radiation energy at higher frequencies

3. Classification based on filter order or slope

- Crossover follow different orders based on the filter slope.
- The final acoustic slope established by

  Electrical filter: each driver must have a flat response with signal strength of at least 10 db down from the passband

Classification based on filter order or slope continue

- A combination of the electrical filter's slope and the natural characteristics of the driver: normally a steeper final acoustic slope is achieved compared to that of the electrical filters used

- First order
- Second order
- Third order
- Fourth order
- Higher order
- Mixed order

First to fourth order electrical filters are most commonly used in audio crossovers.

Passive crossovers for loudspeakers normally do not implement higher orders except in elctronic equipment.

- First order [best for cross overs]: These work on a 20 db/decade slope
- They pass amplitude and phase unchanged across the required range
- It has lesser parts giving lowest insertion loss in passive cases.
- Designing first order is challenge because of the large overlapping bandwidth requirement

•Second order

- Here the slope is 40 db/decade
- The design choices and the components used determine the filter characteristics.
- Complexity ,response and higher frequency driver make it ideal passive crossovers.
- A symmetrical polar response as in even order crossovers can be achieved when time aligned physical placement is included.

•Third order

- It have an 60 db/decade slope.
- similar to first order crossover the level sum is flat
- It has an asymmetric polar response
- First or second order filter circuits are used too build third order acoustic crossovers

•Fourth order

- It have an 80 db/decade slope
- The components interact with each other making the design of the passive form complex.
- The issue with this is that they are parts value deviations or tolerances are not registered

•Higher order

- Due to cost and complexity these type are very uncommon
- It has 96 db slope

•Mixed order

- Lie combining a second order lowpass with a third order highpass
- When a computer program optimization is done to derive component values
- To compensate the time offset, caused by non-aligned acoustic centers between the woofer and tweeter.

•4 classification based on circuit topology

- Parallel
- Series
- Derived

•1. Parallel

- It is commonly used
- Filters are electrically parallel
- It is easy to design two-way
- It ensures isolation of component tolerance variations
- It also allows biwiring of the speaker drivers

•Series

- It is done with individual filters
- But driver combination follows a parallel connection with each filter
- The drivers and filters that present low impedance allow the signal to pass

•Derived

- Using a differential amplifier with active crossovers, one crossovers response is derived from the other
- Advantage

 the high pass and low pass sections do not show a phase difference at any frequency

Disadvantages of Derived

- Levels of attenuation in their stop bands are different for high pass and low pass sections producing an asymmetrical slope
- One or both sections response peaks near the crossover frequency
- Higher volume levels are required

•Introduction to Equalisation

- It is the process of balancing frequency components with an electronic signal
- An equalizer is the equipment that facilitates equalisation by strengthening or weakening the energy of specific frequency bands
- Linear filters are commonly used to alter the frequency response
- Bass and treble adjustments are done with simple filters

•Introduction to Equalisation continue

- It can be seen in recording studios, radio studios production control rooms etc..
- Other use of equalisers are

1helps to eliminate the unwanted sound

2 Can make certain instrument or voices more prominent

Other use of equalisers are continue

- Helps to enhance particular aspects of an instrument ‘s tone
- Can be used in public address system to combat feedback or howling
- Used to adjust the timbre of individual instruments in music production

Filter types

- The theory of linear filters governs the function of equalisation
- The relative gains b/w frequencies are altered with the help of first order filter functions referred to as shelving controls
- Second order filter functions are implemented by the one or more sections in parametric equalizers

•Filter types

- IN semi parametric equalizers the bandwidth is preset by the designer with no control or limited to switch b/w two presets
- Equalization is used in sound recording as well as in sound playback.
- In Live events parametric equalizer is best

Low pass filter

- It will pass all signals having a frequency that is lower than a certain cutoff frequency
- Frequencies higher than the cutoff frequency are attenuated
- The filter design determines the amount of attenuation
- Another filter the band pass filter combines a low pass high pass filter

•Low pass filter continue

- Different forms of low pass filters exist
- Electronic circuit like a hiss filter used in audio
- Anti-aliasing filters to condition signals before analog to digital conversion
- Digital filters to smooth sets of data, acoustic barriers, blurring of images

Low pass filter continue

- Smoother form of a signal can be attained using a low pass filter
- An optical filter is actually a low pass filter
- A true low pass filter will totally eliminate all frequencies that fall above the cutoff frequency
- Signal below cutoff are passed without any change

•Low pass filter continue

- <u>Acoustic example</u>
- A good example of a low pass filter is a physical barrier like a wall.
- When music played in one room , the low notes can be clearly heard in another room but the high note are attenuated

•High pass filter

- It will allow signals that have a higher frequency than a specified cutoff frequency
- Attenuate all signals lower than cutoff frequency
- The filter design defines the amount of attenuation to be done for each frequency

High pass filter implementation

- Place an input voltage across the combination of a capacitor and resistor connected in series.
- Then taking the output from the voltage across the resistors
- Another first order high pass filter can be implimented by using an operational amplifier

•High pass filter Applications

- They can be used to audio crossovers to guide appropriate frequencies to tweeters and ensure that low(bass) frequencies do not damage speaker.
- Eliminate rumble or unwanted sounds
- To prevent the amplification of DC currents
- Mixing consoles at each channel strip
- It can be used in Directional microphones

•Parametric equalizer

- These equalizers allow a user to adjust all the parameters of tone shaping
- They offer continuous control over the many bands of frequencies of the audio signals frequency content
- The basic range from three to seven

•Parametric equalizer continue

- The following are the parameters that parametric equalizer provide control
- Frequency : The central frequency of the envelope in which the signal is boosted or cut
- Bandwidth or quality factor : The number of actaves over which the signal is affected by boosting or cutting
- Gain : The amount which the signal is boosted or cut

•parametric equalizer continue

- Semi-parametric equalizer are available with only frequency and gain control
- Fully parameterized have full continuous control on all three
- A very high level of equalization can be obtained with parametric equalizers.
- We can choose the frequency to be adjusted

•parametric equalizer continue

- The band width or range of each control can also be controlled by it
- These equalizer are more accurate
- It can seen in
- Car audios
- Home stereo systems
- Equalizer, Audio sytems etc

•Graphic equalizer

- It uses a slide controls for adjusting each frequency
- Slide is moved up and down to increase or decrease the amplitude of the frequencies
- Each filter only picks the signal that corresponds to its own frequency range or band
- The cost depends on the no frequency channel provided.

•Graphic equalizer continue

- Compared to a parametric equalizer a graphic equalizer has a set of fixed frequency and fixed Bandwidth
- For each frequency band there is only one boost or cut control.

Band stop or band reject filter

- It attenuates frequencies in a specified range to very low levels and passes all the other frequencies unaltered.
- A band stop filter with a narrow stop band meaning a high Q factor is notch filter
- Notch filters can be found in live sound reproduction.
- It help to reduce in some case prevent audio feed back

•Band stop or band reject filter

- Notch filters are also called band limit filter, T-notch filter , band elimination filter and band reject filter
- Normal width of a stop band is 1 to 2 decades
- Highest frequency that attenuates will be 10 to 100 times the lowest frequency that is attenuated

•Noise reduction Techniques

- Audio signals contains unwanted sound
- The process of removing this noise is called noise reduction.
- Hiss is the major form of noise seen in electronic recording devices.
- Another form of noise is clicks and cracklings that are short time impulsive disturbances.

•Four types of Noise reduction

- Single ended pre recording :at the time of recording ,this affects the recording medium
- Single ended hiss reduction: reduces the noise as it occurs before and after the recording process, and for live broad cast applications
- Single-ended surface noise reduction : This is applied to phonograph records at the time of playback

• Four types of Noise reduction

- Codec or dual ended systems: During recording it uses a pre-emphasis process and during playback it uses a de-emphasis process.

•Dolby and Dbx Noise Reduction systems

- In 1966, ray Dolby introduced this noise reduction technique
- The Dolby type A worked with encode/decode process
- This process worked on the amplitude of four bands by increasing the amplitude during encoding(recording)- & decreasing during placback

•Dolby and Dbx Noise Reduction systems continue

- Dolby B designed for consumer products and worked on a single band systems
- The output of Dolby B system was listenable without decoder on playback systems
- Another Noise Reduction system
- David E Blackmer The Dbx systems

•Dolby and Dbx Noise Reduction systems continue

- The encode/decode algorithm was a root-mean-squared .
- The entire audible bandwidth was covered by the Dbx achieving up to 30 db of noise reduction

•Dynamic Noise Limiter(DNL)& Dynamic Noise reduction DNR

- It is introduced by phillips in 1971 –unpatented audio noise reduction system
- Used in cassette decks
- Development in DNL resulted in Dynamic Noise Reduction
- This worked on long distance telephony,.
- Commercial use of DNR began in 1981.

•Dynamic Noise Limiter(DNL)& Dynamic Noise reduction DNR

- The DNL and DNR systems work only on the playback signal processing systems.
- They can be used on any audio signal –known as non complementary feature.
- They can be achieved as much as a 10 db reduction.

•Other approaches

- Dither systems- it adds noise to a signal to improve its quality.
- Software programs.

Infinite baffle picture