# *Pi Speaker Alignment Theory*

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For audio reproduction to be completely accurate, all signal components must be transferred perfectly from electrical to acoustic energy. The listening environment must be completely free of echoes and reflections. In this manner, the original recording ambience can be perfectly reproduced without interference from the listening environment. If the audio transducers faithfully transform the electrical signal into an audio one, then the listener will enjoy the original recording just as it was created.

Obviously, loudspeakers and the listening environment in which they occupy are not perfect. Even if the audio equipment is very good, listening rooms are usually filled with reflective surfaces and resonances, and these color the reproduced sounds. These problems aside, a loudspeaker design should present an audio signal that matches the electrical signal it is sent, as closely as possible.

This document describes electrical and mechanical properties that must be considered in loudspeaker design, which is most important, and where compromises must be made – it shows which properties are the most important. Further, formulas to quantify each parameter are offered, so that loudspeakers can be designed with a procedural method.

Audio equipment is rated by specific measures of quality. These measures include bandwidth – commonly called frequency response, distortion and power handling, and they are an indication of how accurately equipment can reproduce sound. Each of these specification parameters is important, but some are more so than others. Whenever a compromise must be made, it should be made to affect the specification that matters least.

Bandwidth or frequency response is a measure of the highest and lowest frequencies that can be reproduced. A frequency response curve is a graph showing the acoustic energy produced by the speaker over its entire bandwidth. The response curve can show how linear amplitude is through the bandwidth, and is a more important indicator of quality than simply knowing the upper and lower cutoff frequencies.

A frequency response curve should be as close to linear as possible. If a speaker has large peaks and dips in its response curve, then certain frequencies presented to the loudspeaker will be louder than others, even when presented with equal amplitudes. So a broad and linear response curve is unquestionably the most important aspect of loudspeaker design.

Low distortion is nearly as important as bandwidth linearity. Distortion is a measurement of inaccuracy exhibited when a signal of single frequency is presented. Usually measured as a percentage, it gives and indication of how much of a signal heard is the result of speaker imperfections.

Several things can cause distortion, and most can be avoided through careful speaker design. Most speaker elements are made using an electromagnetic or electrostatic linear motor assembly connected to a cone. In this arrangement, the motor should be carefully designed to ensure that it moves in a controlled fashion, and that it moves exactly with the signal presented to it. The cone attached to the motor may not be rigid enough and may begin to vibrate at harmonic frequencies when presented with certain signals. Further, the mounting of the cone requires a suspension, which may prevent the cone from free movement and may cause the cone to twist or limit its motion more in one direction than the other. So even though the motor assembly is fairly simple, it is important to engineer and build the motors carefully, and to choose only those that are of high quality.

Dynamic range is a combination of two parameters – efficiency and power handling. A broad dynamic range is important because it describes the difference between the lowest and highest levels of acoustic amplitude that the loudspeaker can reproduce. Having a speaker that can handle an enormous amount of power is unimpressive if its efficiency is so low that high power levels cause it to provide a relatively low volume of sound. While dynamic range, efficiency and power handling aren't measures of accuracy – they are very important indicators of the overall quality of a loudspeaker.

Efficiency is rated in decibels – over our threshold of hearing – with one watt of electrical energy. Specifically, the amount of energy required for us to even hear a sound is 0.00002 newtons/sq. meter, and is defined as 0db. Further, the accepted way to measure efficiency is relative to this reference value, when providing a signal of 1 watt to a speaker and measuring its acoustic output at 1 meter's distance.

A difference of 3db is equivalent to a difference of two in power. Decibels are a logarithmic scale – *10logX/Y* - which means that every time you multiply power figures, you can describe it by adding decibel figures. As an example, a four-fold increase of power will yield a 6 decibel gain. To gain 12 decibels, you need 16 times as much power, and a 100-fold increase yields a 20 decibel gain. Therefore, whatever a loudspeaker is rated at 1watt/1meter, 20db is added to calculate its output at 100 watts.

Power handling is a measure of how much power can be presented to a speaker without causing it to be damaged. Often times, cone travel is impaired by the loudspeaker suspension at power levels much lower than its rated maximum, so that fact must be taken into consideration. It is generally safest to assume that distortion rises dramatically between 50% and 70% of a speaker's maximum rated power handling capacity.

Power handling is measured in watts, and because of the transient nature of audio signals an averaged measurement of rating known as R.M.S. is commonly used. R.M.S. stands for Root-Mean-Squared, and when comparing specifications it is safest to use this figure.

Other measurements are sometimes used, but they can be very misleading. For example, Peak power is a figure that is 41% greater than R.M.S. and Peak-to-Peak is 182% greater. Thus, a speaker with a 50 watt R.M.S. rating has the same power capacity as one rated at 70 watts Peak or one rated at 140 watts Peak-to-Peak.

Root-Mean-Squared, Peak and Peak-to-Peak are all valid technical terms. They can all be used to quantify signal amplitude, and they can all be used to describe a maximum value. But to connect a loudspeaker rated at 150 watts Peak-to-Peak to an amplifier rated at 100 watts will probably result in damage to the loudspeaker with the amplifier at half its rated output.

Phase shifts are differences in time between two components of an audio signal. For example, when a snare drum is played, two distinct sound components are created at the same time – a low frequency sound that is the result of the resonation of the drum and a higher frequency sound that is caused by the impact of the drum head and the vibrations of the snares. If the two signals are separated and sent to different speaker elements, and if one of the elements is much closer to the listener than the other, then one component of the sound will reach the listener sooner than the other.

Phase shifts are also introduced to a small degree by crossover components and by the electrical characteristic of the driver motors themselves. But these shifts are much less than one cycle and are negligible. The only problem that can arise from these small phase shifts is the possibility of two drivers receiving a portion of a signal, and then being shifted near 180 degrees so that the sound output of the individual drivers partially cancels or modifies the other.

A greater annoyance is when there is a large multi-cycle shift. In this sense, it is more of a time-delay than a phase shift. This has the same effect as is heard in large meeting halls or outdoor events where several loudspeakers are placed at great distances and the impression of echoes is heard because of the difference in time that the listener hears the closest loudspeaker from those that are further away. When several loudspeakers are used together to present the same signal, or to present components of the same signal, they should all be placed as close together as possible. Every attempt should be made to ensure that each speaker driver is placed at the same distance from the audience.

Reflections are caused by interference from objects away from the source of the projected sound. These objects can be outside the loudspeaker enclosure – in the listening environment - or they may be a result of the loudspeaker itself. Except for horn loaded enclosures, reflections caused by the speaker enclosure are generally limited to those from the edge of the speaker and are negligible. Occasionally, manufacturers will design a certain amount of reflection into a speaker system. But in reality, reflections play a very small role in loudspeaker performance, and in any case are unavoidable in a listening room of any complexity.

Impedance is a measure of how much difficulty electrical energy has when passing through a conductor. It is important that the impedance of a loudspeaker be matched with its amplifier. Eight ohms is the most common value, but four and sixteen ohms are also popular in certain applications. Too high an impedance will not allow the amplifier to transfer energy into them, and will often damage vacuum tube amplifiers. Too low an impedance will draw too much power from an amplifier without properly transferring the electrical signal, which will cause the amplifier to prematurely clip and create harmonic distortion that is annoying and damaging.

For the loudspeaker designer, it is important to choose components of the same impedance except in a few special cases. Typically, the system will contain more than one driver element, but by using crossover components, only one will present the majority of the load to the amplifier. So the designer will usually choose drivers of the same impedance and connect them with crossover components.

The most common exceptions are when more than one driver is connected in series or in parallel. Drivers connected in parallel increase efficiency but not power handling and drivers in series increase power handling but not efficiency. A loudspeaker can be designed with two midrange drivers connected in series to double power handling capacity of the midrange system. Perhaps two low frequency drivers are connected in parallel to increase bass efficiency by 3db. In these examples, series connected drivers are usually chosen that are of half the impedance as the other drivers in the system and parallel connected drivers should have double the intended loudspeaker system impedance.

Careful loudspeaker design is important for accurate signal reproduction. The best equipment in the world cannot help the quality of sound that a poorly designed speaker makes. Frequency linearity and distortion characteristics of the speakers chosen color the rest of the sound system, and the speaker's dynamic range sets the upper volume limit of the system. So designing a loudspeaker, which has favorable specifications, is an exacting task, and arguably plays the most important role in a sound system's overall quality.

Loudspeaker design should start with the low end of the audio spectrum. First, a woofer should be chosen which exhibits a flat response curve and a low cutoff frequency. The factors for consideration should be the required low frequency cutoff point and power handling capacity. No consideration should be made for woofers with non-linear response curves or high total harmonic distortion levels – only the components tested and shown to have the most superior performance levels in frequency linearity and distortion should be used.

Three parameters describe the operating characteristics of a woofer, excluding dynamic range. They are the Equivalent Volume (*Vad*) - which is a measure of the volume of air that has the same stiffness as the driver, Resonant Frequency (*Frd*) – is the frequency of resonance in free air, and the Ratio of Resonance to Bandwidth (*Qd*) – a measure of the bandwidth of the peak caused by resonance. Two characteristics describe the dynamic range of a woofer, being (*Eff*) – the Efficiency in decibels measured at 1 watt and 1 meter, and Power Handing Capacity (*Pwr*), which is measured in watts.

After choosing a woofer that provides the desired low frequency cutoff and power handling capacity, determine the volume of the enclosure required by the woofer. At this time, you will also determine the optimal resonant frequency for the enclosure.

To determine the optimum enclosure volume, resonance and bandwidth, use the following formula:

$$Ve = {Vad}/{Qd}$$

$$Fre = {3FrdQd}/{8}$$

$$Qe = {Fre}/{2\left|Fre - Frd\right|}$$

where $Ve$ is the optimal enclosure volume (in cubic feet)
$Fre$ is the optimal enclosure resonant frequency, and
$Qe$ is the optimal enclosure damping bandwidth

The next step is to find a port that will tune the enclosure to the optimum frequency and damping bandwidth:

$$Lc = Lp + {8Dp}/{3\pi}$$

$$Fre = \frac{l}{2\pi}\sqrt{{Ap}/{VeLc}}$$

$$Qe = {Lp^2}/{Ap}$$

where $l$ is the speed of sound in air, 13548 inches per second
$Lp$ is the length of the port (in inches)
$Lc$ is the corrected port length (offset due to area)
$Dp$ is the port diameter (in inches)
$Ap$ is the area of the port (square inches)
$Fre$ is the actual resonant frequency of the enclosure
$Qe$ is the actual damping bandwidth of the enclosure

Notice that the first set of equations derive optimum enclosure values when the woofer driver specifications are given. The second set of equations calculate actual enclosure resonance when port dimensions and enclosure volume are given.

Different values of *Ap* and *Lp* can be chosen which will tune the enclosure to the same resonant frequency. For example, you may find that a port with 3 inch diameter and 3 inches length will tune the enclosure to the optimum frequency, but so will one that is 8 inches long and 4 inches in diameter. The port should be chosen which is the closest match to optimum, for both *Fre* and *Qe*. This is not a trivial task, and often several iterations are required. But bass reproduction in the critical resonance region is largely affected by these parameters.

Now that the woofer enclosure has been designed, the high frequency section should be tailored to match.  If the woofer is small and can reproduce the midrange section, then a tweeter alone may suffice for high frequency response.  Larger woofers do not reach into the midrange band and systems incorporating these woofers typically break the audio spectrum into three or even four parts for proper frequency response.  Do not split the audio spectrum unless the choice of drivers makes it imperative.  Every crossover point is accompanied by a corresponding phase shift and should be avoided unless necessary.

Midrange and midbass enclosures can be tuned the same way as woofer enclosures; however, it is important to note that most likely the driver will not receive frequencies at or near its resonant.  Usually, midband enclosures are designed to receive only frequencies much higher than the resonant.  Only if a midbass enclosure is expected to reproduce frequencies at resonance should the enclosure be tuned.

In the usual situation – where the midrange band is significantly higher than resonant – the enclosure simply needs to be large enough to resonate below the lowest frequency presented to it.  These midrange enclosures are usually placed within the woofer enclosure.  When doing so, volume within the woofer enclosure is displaced so it is made that much larger to compensate.

All sections of the response curve must be equally represented and of equal efficiency. High efficiency drivers must be attenuated or low efficiency ones need to be supplemented.  A pair of identical drivers connected in parallel will act as a single driver would, but will provide a 3db gain in efficiency at the cost of an impedance drop of one half.  Drivers can be attenuated 6db by connecting a series resistor equal to the driver's impedance.  Whatever methods are chosen, efficiency of all driver sections must be made equal or frequency linearity of the system will be poor.

Usually a driver should be allowed to receive frequencies up to, and sometimes beyond its upper cutoff point. In most cases, drivers are more prone to distort when presented signals below their intended range than they are from signals above.  Most speakers have reduced output of signals above their intended frequency range, but do not distort or become nonlinear.  So the best option usually is to cross a driver at the frequency where the one below it begins to have reduced output.

Exceptions to this include drivers that become nonlinear or distorted at higher frequencies.  Occasionally, drivers have a good response curve up to their upper limit, but after they begin to roll-off, they have one or two annoying peaks above their intended range.  When using drivers of this type, the crossover should attenuate their upper cutoff.

Each crossover point should be made without overlap.  If any driver has a response curve that overlaps another, then the system response curve will show a peak in the overlapping region.  Each driver should be crossed over at the point where the lower frequency driver is cutoff.  No overlap should be allowed, nor should any gap.

Crossover component values are calculated by knowing the driver's impedance and the desired crossover frequency. Using the proper series inductance or capacitance value provides an attenuation of 6db per octave from the frequency chosen.

Inductors allow low frequencies to pass and attenuate high frequencies. Capacitors allow the passage of high frequencies but block low frequencies. A capacitor and an inductor connected in series act as a band pass filter, blocking both low and high frequencies and allowing only midrange frequencies to pass.

As an example, a midrange that should cross at 700hz and 8000hz should have a capacitor and an inductor connected in series with the midrange. The calculations for both the capacitor and inductor should use *X* equal to the impedance of the midrange, but for the capacitor *F*=700 and for the inductor *F*=8000.

To determine the proper 6db/octave crossover components:

$$C = \frac{1}{2\pi Fx}$$

$$L = \frac{x}{2\pi F}$$

where *F* is the desired frequency
*X* is the impedance of the driver
*C* is a crossover capacitance value
*L* is a crossover inductance value

Many loudspeaker designs are optimal with this type of crossover circuitry. It provides a smooth rolloff and matches well with the gradual rise in acoustic output of the higher frequency drivers in the system. But occasionally drivers are chosen that do not work well with 6db/octave crossovers. The reason is that crossover networks do not completely eliminate unwanted frequencies.

For example, you would expect a woofer that is crossed over at 1000hz would be –6db at 2000hz. But if the woofer has a sudden peak of 6db at 2000hz, then the driver will be just as loud at 2000hz as it was at 500hz. Further, the midrange driver will also be online at 2000hz, so there will be a noticeable increase in volume for sounds near this frequency. Clearly, a 6db/octave crossover is not suitable for designs of this type.

Another important consideration is that a driver's impedance rises at its resonant. Caution should be taken to ensure that excessive power isn't coupled into high frequency drivers at their resonant, even if the intended crossover frequency is significantly higher. A 6db/octave crossover is usually insufficient if the target crossover frequency is less than an octave from the driver's resonant frequency.

Sometimes, a crossover rolloff is desirable for smoothing a gradual 6db/octave rise in a driver's response curve near the cutoff region.  In this case, the driver that is being crossed to will be crossed later.  When a driver has a gradual rise, it is appropriate to attenuate the 6db peak and then to bring the higher frequency driver online much later.

For example, if the woofer has a 6db rise at 2000hz, then it is appropriate to install components to "crossover" at 1000hz – but then do not crossover the midrange until above 2000hz, where the combined response of the woofer and the crossover coil is 3db below average.  This would produce an entirely flat response curve.

Sometimes a design using 6db/octave crossover components simply isn't appropriate.  A 12db/octave rolloff can be provided by using a component connected in parallel with the driver in addition to the series component.  Calculation of this type of network is similar to the 6db/octave crossover, except that instead of choosing a single low pass or high pass component – one is chosen to pass in series and another is to shunt in parallel.

For example, to cross a woofer at 1000hz with 12db/octave attenuation, the inductor should be connected in series with the woofer and the capacitor should be placed in parallel.  The calculations for both the capacitor and inductor should use $F$=1000 and $X$ equal to the impedance of the woofer.

When using a 12db/octave crossover design, it is usually desirable to connect drivers in a manner that would usually be considered "out of phase."  This is because the phase shift of the crossover itself will more properly match to an adjacent driver's corresponding phase shift – if the two are connected in reverse.

The reasons for this are significant, and caution should be taken in this regard.  The phase relationships for reactive circuits are described with the following formula:

$$\theta = ArcTan\frac{|Xl - Xc|}{R}$$

where $\theta$ is the phase angle
$Xl$ is the impedance of an inductor
$Xc$ is the impedance of a capacitor, and
$R$ is DC resistance

Since the 6db/octave crossover is a simple series circuit, phase shift is limited to 45 degrees.  But the 12db/octave crossover has a series component and a parallel component, and can therefore shift phase up to 90 degrees.  Both 6db and 12db crossovers have a phase shift in opposite directions for high-pass and low-pass filters.

This means that the maximum phase difference for a 6db crossover is 90 degrees – an unwanted but relatively unimportant matter. The 12db crossover, however, has a maximum phase difference of 180 degrees – which causes signal cancellation – and would result in a 20db to 40db transient drop for frequencies where the phase difference is nearly 180 degrees.

Polarity reversal for 12db/octave circuits ensures that the overlapped region – where maximum phase difference occurs – will not be 180 degrees out of phase. In the audio spectrum where one driver is at midband, this reversal will not impact another driver because other drivers are offline.

Because of this peculiarity of the 12db/octave crossover, it is recommended that adjacent drivers using 12db/octave crossovers be always connected with opposite polarity, and avoided entirely when possible. When one driver uses a 12db/octave slope and an adjacent driver does not, the phase relationships of the signals during the overlap region should be calculated to know the best connection method.

A final word about crossover and compensation networks:

With a purely resistive load, a 6db/octave crossover cuts power by one-half at the chosen frequency and by about one-fifth an octave into the stop-band. At two octaves into the stop-band, power is down by a factor of twenty. As you can see, these numbers are equal to –3db at the chosen frequency, -7db at one octave into the stop-band and –13db at two octaves into the stop-band.

But since a speaker motor is not a purely resistive load, an engineer would be well advised to calculate power-transfer graphs for their crossover designs before putting them into production.

The reasons for this are two-fold. First, the 6db/octave crossover described can only produce a smooth rolloff with a purely resistive load. The 12db/octave network attenuates at a faster rate, but it too is dependent on its load. And the fact is that speaker motors are not purely resistive.

The second reason one should confirm their crossover design is that a driver’s impedance at resonance emulates that of a tuned circuit. After resonance, a driver can usually be considered to be primarily a resistive load, with only a small reactive component. But the impedance peak at resonance is often significant.

The impedance peak at resonance is usually more of a concern than the reactive component of the driver’s impedance during the rest of its usable bandwidth. Since the reactive component causes a slow linear change in impedance with respect to frequency, it is not usually troublesome. However, the peak at resonance causes impedance to change rapidly at one frequency, which is also usually very near the lower cutoff of the driver.

Since the resonant peak increases impedance, it will also affect the signal proportions distributed between components in a reactive network. If the resonant impedance isn't known, and the engineer assumes the driver to have linear impedance in this region, crossover frequency calculations will be inaccurate and will probably result in more power being dissipated by the tweeter at resonance than anticipated.

Actual impedance values of reactive circuits are described with the following formula:

$$Z = \sqrt{R^2 + |Xl - Xc|^2} \qquad \text{(for series circuits)}$$

$$Z = \frac{R|Xl - Xc|}{\sqrt{R^2|Xl - Xc|^2 + Xl^2 Xc^2}} \qquad \text{(for parallel circuits)}$$

where Z is the total impedance
*Xl* is the impedance of an inductor
*Xc* is the impedance of a capacitor, and
*R* is DC resistance

So drivers should be considered to be primarily resistive only after their resonant, and their actual impedance should be used in all crossover network calculations. Any impedance calculations should include the actual impedance of speaker drivers at the frequencies being calculated for, rather than to use the simple "advertised" impedance figure.

Power handling is an important and often misunderstood specification. All system components should be chosen that can handle the power required. System power handling capacity is only as great as its weakest link. There is a common misconception that power capacity of the midrange section need only be 50% of the bass, and that the power handling requirement of the high frequency tweeter need only be 30%. The reason for this does have some merit – it is an average power distribution of most audio content.

The problem with this reasoning is that sometimes signals will be presented to the loudspeakers that don't have this distribution. It is best to have the same, or similar, power handling capacity for all components in the system. And also do not forget to take into consideration the power capacity of the crossover components.

Adding series resistance to the driver can increase its power handling capacity. This is done at the cost of reduced efficiency, and it increases impedance. A better alternative is to connect two identical drivers in series, which doubles power handling capacity without decreasing efficiency. Here again, impedance is increased.

For inductors and resistors, current will be their limiting factor. Capacitors will be limited by their breakdown voltage. With all things, rated power capacity of a system should be listed as the component that can handle the least amount of power in the system.

To calculate power requirements for crossover components, one must know how to derive voltage equivalents from power. Crossover components will not have to dissipate as much power as the drivers they are connected to, but will always be subjected to the full voltage of the system at certain frequencies. So it is important to know the power requirements of inductors and resistors to maintain full rated capacity and it is also important to calculate the voltage that is presented to the system so that capacitors will be chosen with a high enough breakdown voltage.

Inductors and resistors are rated in watts, but capacitors are rated in volts. That's why it is important to know the voltage presented to the system. Also, since the crossover components are dividing energies between components and themselves, calculations must be performed to know what the highest stresses are.

To calculate voltage, one must also know the current and impedance of the system. The given parameters are power and impedance of each component, so we must start by calculating system impedance using each individual component:

$$Zt = Z1 + Z2 + Z3\ldots \qquad \text{(for series)}$$

$$Zt = \frac{1}{1/Z1 + 1/Z2 + 1/Z3\cdots} \qquad \text{(for parallel)}$$

where *Zt* is total system impedance
*Z1* is impedance of series component 1
*Z2* is impedance of series component 2
*Z3* is impedance of series component 3

Most loudspeakers will have a combination of components, making a series-parallel network. Take for example, a simple system with one woofer using a crossover coil and one tweeter with a crossover capacitor. This configuration will have two series circuits connected in parallel with each other. Impedance can only be calculated at specific frequencies, and it is important to know impedance at two frequencies – each in the mid-band of a driver. A good pair of frequencies would be 100hz and 10Khz.

Series calculations will be taken for the woofer/inductor circuit and for the tweeter/capacitor circuit at 100hz. The impedance at 100hz will be very near the woofers impedance for the woofer circuit, but will be very high for the tweeter circuit. At 10Khz, the tweeter circuit will have *Zt* near to the tweeter's impedance but *Zt* will be very high for the woofer circuit.

Now parallel calculations will be done for the total system impedance at 100hz and 10Khz. Using the formula for parallel impedance and the series *Zt* values found above, find the system *Zt* at 100hz and at 10Khz. It will be near to the woofers impedance at 100hz and near the tweeter's at 10Khz.

Now we have total system impedance, the impedance of each series circuit at two frequencies, and the impedance of each component at two frequencies. Now we can plug these values into our power translation formulas:

$$P = I^2 Z \ , \quad P = E^2/Z \ \text{ and } \ P = EI$$
$$E = IZ \ , \quad E = \sqrt{PZ} \ \text{ and } \ E = P/I$$
$$I = E/Z \ , \quad I = \sqrt{P/Z} \ \text{ and } \ I = P/E$$
$$Z = E/I \ , \quad Z = P/I^2 \ \text{ and } \ Z = E^2/P$$

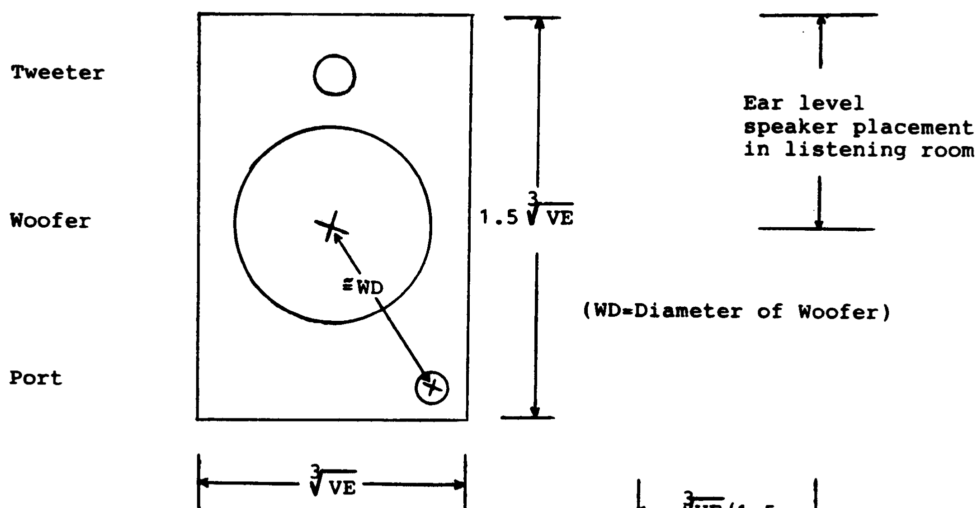
where *P* is power in watts
*E* is voltage in volts
*I* is current in amperes, and
*Z* is impedance in ohms

Knowing the total rated power and the system impedance, we can calculate voltage across the system, using the formula above. Then, taking this voltage, we can calculate the maximum current through each series circuit – the time when impedance is lowest. This current and the impedance for each component will yield the power dissipated by each component – which is the value required in order to have the system power capacity as described.

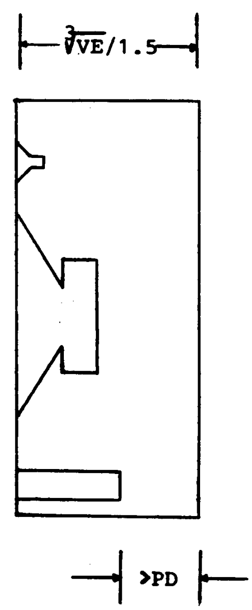
Similarly, we will want to calculate maximum voltage requirements. In each series circuit, the maximum voltage across capacitors will be when impedance is highest. It will be nearly the full voltage of the system, but to confirm it, calculate the minimum current through each series circuit – when impedance is highest. The current and impedance for each component will yield the voltage across each component – which, again, is the value required to allow full system power to be developed without damage.

Carefully designed loudspeakers are made from several components that are chosen using precise calculations. Any compromise that alters the results of these calculations from optimal will adversely affect the performance of the design. Exact tolerance and precise calculation will provide a design which has no compromise, and the quality of speakers built using this design will be as close to ideal as is possible. When combined with high quality drivers, speakers built in accordance with the preceding methods are as close to perfect as can be built.

Typical PI Speaker



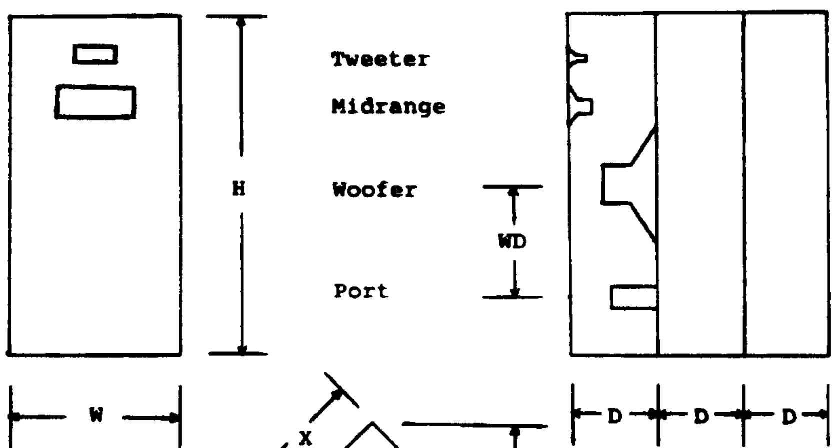

The crossover, matching components, and drivers should be securely mounted and free from vibrations. The enclosure should be sealed so that air can pass only through the port when the drivers are installed. Approximately one inch of fiberglass insulation should be attached to the bottom, the back, and the side nearest to the port on the inside of the enclosure. The enclosure should be constructed of particle board or plywood of sufficient thickness to prevent vibration of the wood.

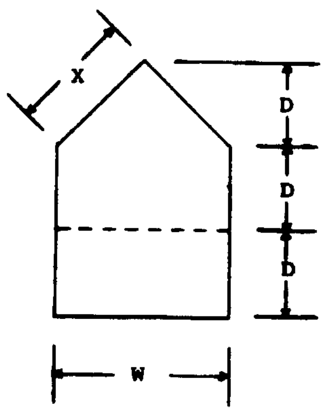



(PD=Diameter of Port)

## Typical Corner PI Speaker

**This enclosure raises the efficiency of the woofer 3dB when placed in the corner of the listening room. The walls of the room form a horn for the woofer.**




$W = 2\sqrt[3]{Ve/8}$

$H = 4\sqrt[3]{Ve/8}$

$D = \sqrt[3]{Ve/8}$

$X = W \sin 45$

**This portion of the enclosure faces the corner.**

**This portion is covered on top and bottom but open on the sides for bass to eminate.**

**This is the tuned speaker enclosure and is similar to enclosures without a horn except the woofer faces to the rear.**