

Crossovers de-mystified



BSS Audio

Sub-title

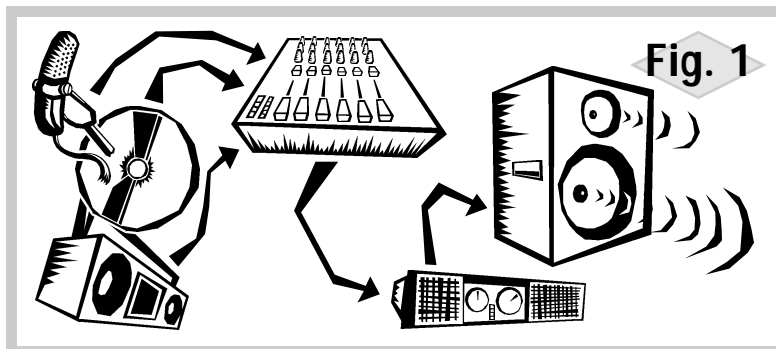
Cross, v; an individual produced as a result of this process.

Cross-, *combining form*; indicating action from one individual, group, etc.

Loudspeakers

Loudspeaker systems come in many shapes and sizes, from the Hi-Fi in your home to a concert system at Wembley Stadium.

Despite the difference in size between these systems they all do the same job; converting electrical signals into sound that we can hear (see figure 1).



How do they do that?

When a loudspeaker cone moves, the air in front is compressed and then expanded, creating a sound wave. It is these compressions and expansions that transmit the sound from the loudspeaker to the listeners' ears. In most loudspeakers this conversion from electrical energy to sound energy is achieved by using a moving coil design. The electrical signal is passed through a tightly wound coil of wire which is placed in a magnetic field, as shown in figure 2.

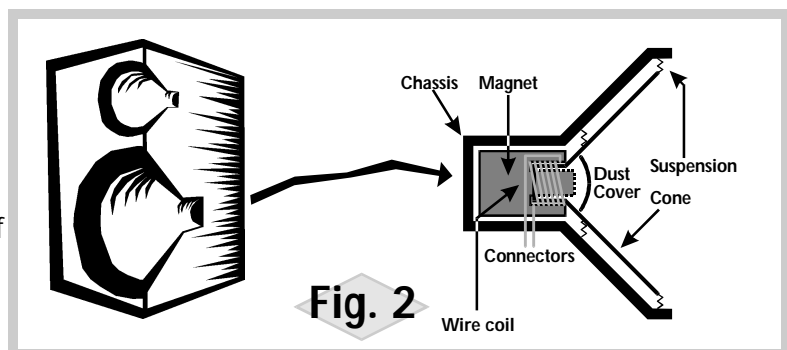
The alternating current passing through a magnetic field causes the coil to move, displacing the cone (or diaphragm) backwards and forwards, generating sound.

Why are there different sizes of drivers?

If you remove the grille from a typical Hi-Fi speaker, you will see two different sized drivers inside. We accept that the larger of these is for bass frequencies, and the smaller for treble, but why is it necessary to have these two drivers?

To reproduce bass frequencies, a large volume of air must be moved. This is done by allowing a large amount of movement in the driver, and also by increasing its size. As the size is increased, the effectiveness of reproducing high frequencies is reduced as it becomes more difficult for the cone to vibrate at the speed required.

High frequencies generally require less energy to be transferred to the air, and so much smaller drivers can be used. These smaller, lighter drivers can vibrate very



quickly reproducing treble, but are obviously unsuitable for low frequencies. If bass frequencies are applied to these delicate high frequency drivers it will often result in their destruction; a costly mistake.

The graphs in figure 3 show the frequency response of the components found in a typical Hi-Fi speaker.

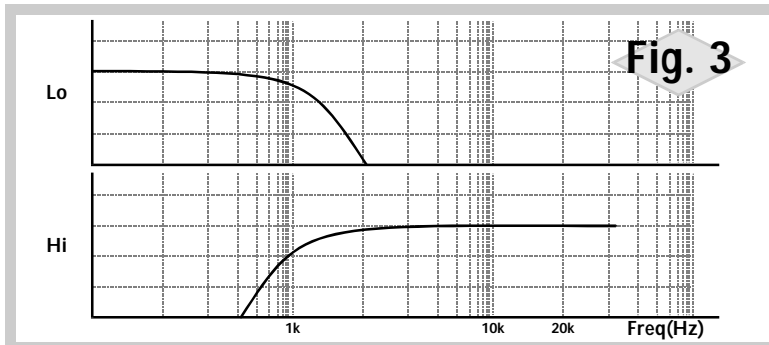


Fig. 3

The ability to “chop up” our frequency range into different bands allows the designer of a loudspeaker system to choose specialised drivers for each range of frequencies. By carefully choosing these drivers and the frequency bands used, it is possible to achieve lower distortion and a better reproduction of the source material.

This division is achieved by a piece of electronic circuitry known as a *crossover* (or occasionally as a *frequency dividing network*).

Crossover filter types

Every frequency division system uses the same basic filter types enabling us to split up the frequency spectrum as required.


Using drivers together

For the highest quality of reproduction we must use a number of loudspeaker drivers together, each one reproducing part of the whole frequency range. Our typical Hi-Fi speaker mentioned earlier would have a loudspeaker driver with a broad frequency range, and a high frequency driver to compensate for the main drivers' frequency roll off.

For any system with more than one driver to work correctly, we must ensure that each driver only receives the part of the signal that it is intended to reproduce. In other words, the audio frequency range is divided into a number of *bands*, one band for each driver. For example, our “2-way” Hi-Fi speaker might have a low band covering frequencies below 2kHz, and a high band covering frequencies above 2kHz. A more sophisticated sound reinforcement speaker might be “3-way”, and have a low band covering frequencies up to 1.5kHz, a mid-range speaker covering frequencies from 1.5kHz to 7kHz, and a tweeter covering frequencies above 7kHz.

 **High-pass filters** - Only allow higher frequencies to pass

 **Low-pass filters** - Only allow lower frequencies to pass

 **Band-pass filters** - A combination of high and low pass filters only allowing a range of frequencies to pass.

Our simplest example of a low cost Hi-Fi speaker could use just a high-pass filter on the tweeter, which starts to pass signal just at the point in the audio spectrum where the low frequency driver ceases to be effective. This filter is often nothing more than a capacitor which stops any low frequencies from passing through, shown below in figure 4.

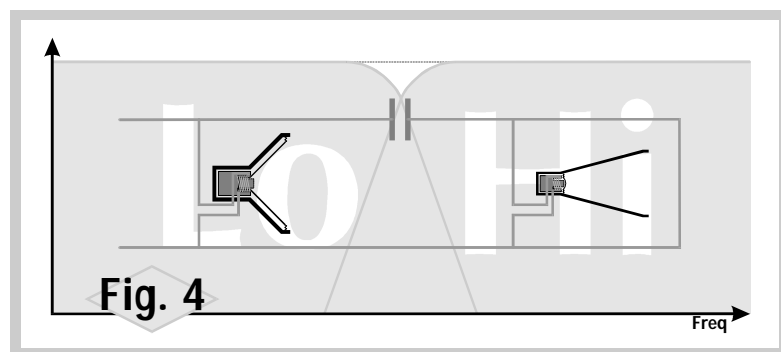
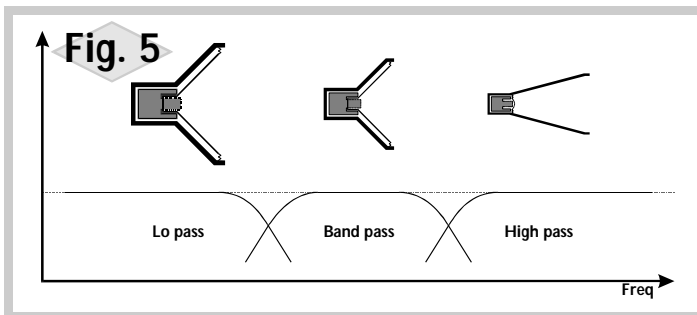


Fig. 4

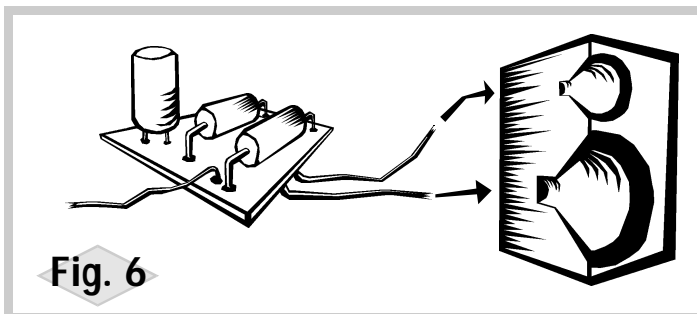
A more sophisticated three way crossover may need one low-pass filter, one band-pass filter and one high-pass filter to split the signal up. The low pass filter will be used to stop midrange or high frequencies from being sent to the bass driver, the band-pass will stop bass or treble from going to the midrange driver, and the high pass will prevent bass or midrange from reaching the high frequency driver. This is shown in figure 5 below.



Types of crossover unit

Crossovers come in two types:

A **passive** crossover consists of filter components (mainly capacitors and inductors) placed inside the loudspeaker cabinet. A single power amplifier channel is used for the whole frequency range. Figure 6 shows a simple two way loudspeaker using a passive crossover network.



With an **active** crossover, each frequency band is amplified separately using a different power amplifier channel; the crossover is a box of electronics placed in the signal path between the signal source (usually a mixer) and the power amplifiers. A loudspeaker system which requires an active crossover is often known as *bi-*

amped (for a 2-way cabinet) or *tri-amped* (for a 3-way cabinet) - see figure 7.

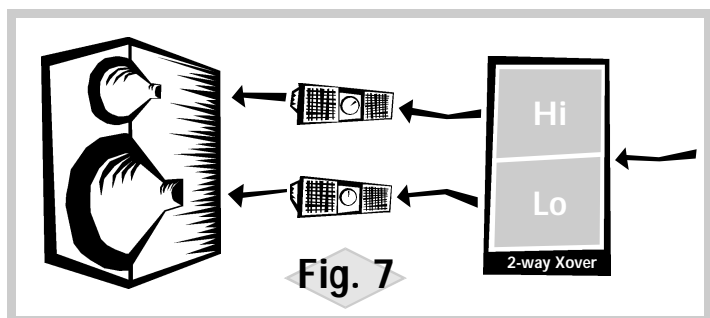
For smaller systems, passive crossovers provide a very cost effective solution, with only one power amplifier and one cable required for each speaker cabinet. However, as the power increases, the cost of the passive components required becomes prohibitive, and significant energy is wasted within the crossover filter.

For higher power systems, active crossovers provide major advantages. Because the crossover operates on line level signals, as opposed to speaker levels, the cost of the circuitry needed to produce accurate filters does not increase as the power goes up. Much more sophisticated filter circuitry can be provided, the user can be given

much greater control over signal processing, and additional functions such as band equalisation can be included.

For example, the BSS FDS-310 and FDS-318 crossovers provide front panel controls for the setting of crossover points and band output gain, the user manipulating the filter positions and band gains to achieve the best response. The BSS FDS-360 uses plug in

cards to determine the crossover frequencies; these cards can also be set to provide adjust overlaps between bands as well as various different filter slopes. In this way, it is possible to tailor any setup to suit whatever loudspeaker components are being used, ensuring the best possible results.



Digital Crossovers

The advent of digital crossovers has taken the power of active crossovers one step further. Many processes that would traditionally be achieved by using many units can now be incorporated into one crossover system providing an integrated solution.

A digital unit such as the BSS FDS-388 Omnidrive is not simply a crossover; it is a complete Loudspeaker Management System. As well as the crossover, functions are provided for limiting, time delay and equalisation. Because the unit is completely controlled by a microprocessor, each unit can store many programs internally. The same Omnidrive can be used for many different speaker systems owned by a rental company, and set up for a particular system or particular gig by simply recalling a program from the Omnidrives' memory.

Perhaps the most important benefit to come from digital crossovers is the provision of driver alignment, the nature and importance of which are described later. Whilst it is possible to use delay lines connected to analogue crossovers to achieve a perfectly driver aligned system, this is an expensive and often complicated solution. The integration of delay into a digital crossover enables the benefits of driver alignment to be easily achieved without great expense.

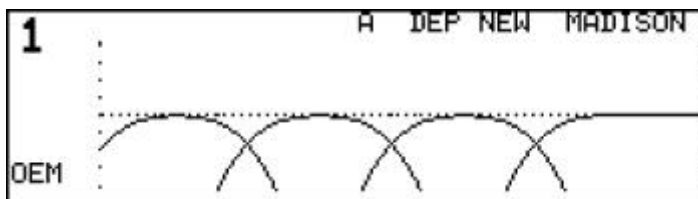
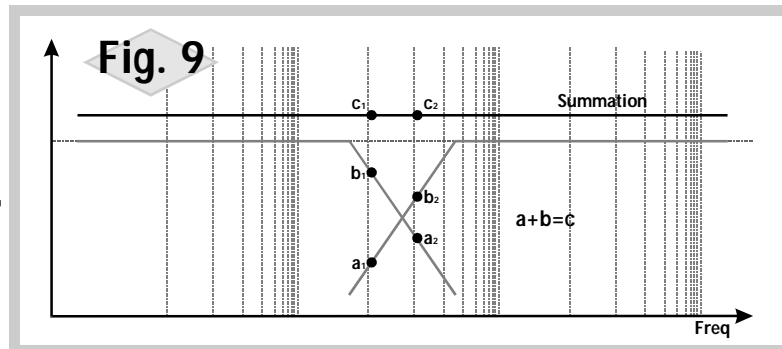


Fig. 8

Adjusting crossover settings

Figure 8 shows a typical representation of the crossover network, as displayed on the screen of a BSS Omnidrive, clearly showing where one band stops and the next begins. It is undesirable to abruptly stop each band and so each high or low-pass filter actually rolls off the edge of its band at a certain rate. This rate is called the slope of the filter, and will typically range from 12 to 48dB/Octave.

The aim in using such a crossover is to maintain a flat frequency response across our crossover points. The slopes of adjacent bands achieve this; one band fades out as the next band fades in, giving a constant level across the transition.



If we look at the actual response of the crossover, we can see that the crucial point is where one output is rolling off and the other is rolling in. The ideal setup will give a flat response across the transition from one band to the next, demonstrated in figure 9.

If we consider point b_1 , where the LF band has started to roll off, the addition of the signal from the HF band (a_1) coming in should maintain a flat response - c_1 . The point at which the low and high pass filters cross is called the crossover point, and the rate at which the filters roll off is the slope. This slope is measured in dB/Octave and should be chosen dependant on the drivers used, although in most cases a 24dB/Octave slope is acceptable.

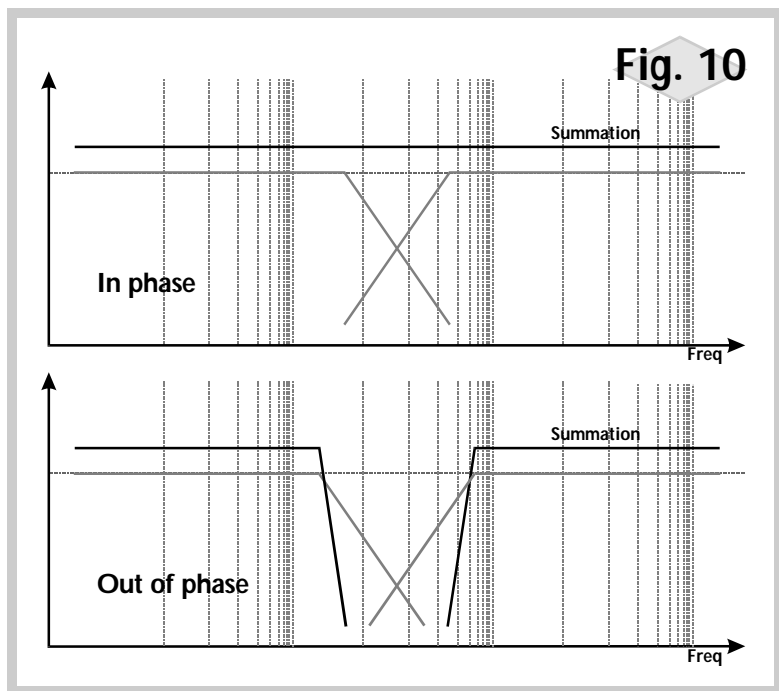
As discussed earlier, the ideal response of a crossover is flat across the crossover point. This relies on the perfect summation of the two bands being achieved, and leads us to consider the phase of the signals. The introduction of any filter will introduce some shift in a signals' phase, as will the characteristics of the speakers. A phase shift will affect the summation of the signals around the crossover transition, shown in figure 10.

The BSS FDS-360 and Omnidrive units offer the facility to trim the phase alignment between bands, which allows the system to be set up so that the target of a maximally summation (flat response) at the crossover point can be achieved. Obviously when the bands are in phase the output at the crossover frequency will be at its greatest, and when 180 degrees out of phase it will be at its least.

There are many ways to set up the phase trim, but here is a very simple method. All that is required is some form of signal generator, or even a test CD with spot tones on it. If we select a tone equal to the crossover point frequency and play it through the system, we will hear its level change as the phase is adjusted.

To use this effect to set up a system we should first set band gains such that they have roughly the same amount of output from both drivers. The phase can then be adjusted until the signal is at its loudest, signifying correct phase alignment. The loudest point can be measured using an SPL meter or RTA, but simply using your ears will allow a surprisingly close approximation.

Using the same principle, we can make it a little easier by reversing the phase of one of the outputs. This can be achieved by



swapping the amplifier output connections, or by using the phase reverse button if the crossover has one. Now listen for the point of maximum cancellation; the point at which the output is quietest. If we then correct our amplifier connections (swap them around again), or switch the phase reverse button out, the loudest possible signal will return, and with it the correct phase alignment. This method is useful as it is often easier to hear the quietest point than the loudest.

Limiting

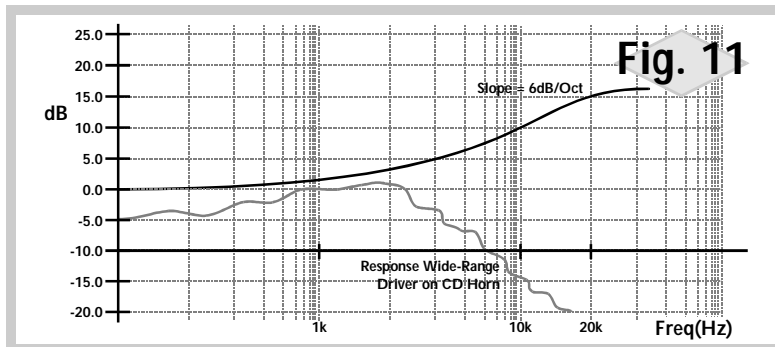
Depending on how a speaker system is used, it is often necessary to provide signal limiting for protection. The BSS FDS-360 crossover and the Omnidrive loudspeaker management systems contain protection limiters for this purpose. This subject is itself fairly complex, and the BSS application note on "Loudspeaker protection" deals with it in detail.

Equalisation

Many crossovers, both analogue and digital, offer the ability to add some equalisation to the outputs. This is often necessary to correct for the frequency response of some component or cabinet, and can be achieved within the crossover unit.

Constant Directivity (CD) horn equalisation

CD horns are designed to give a constant dispersion across their specified frequency range and this creates the need for some corrective equalisation. Typically the CD horn will cause the output to roll off with increasing frequency, and so a CD horn EQ is normally a 6dB/Octave slope, as shown in figure 11.



Component tuning

In addition to CD horn EQ, it is frequently necessary to add some further EQ to compensate for the frequency response of other components; perhaps a boost in the bass region, or a cut in the midrange. This type of EQ is often carried out on the main equaliser with the same setting being used every day. If it were possible to “program” this EQ into the crossover, it would enable you to concentrate on room tuning with your main equaliser, rather than compensating for any problems in the speaker system.

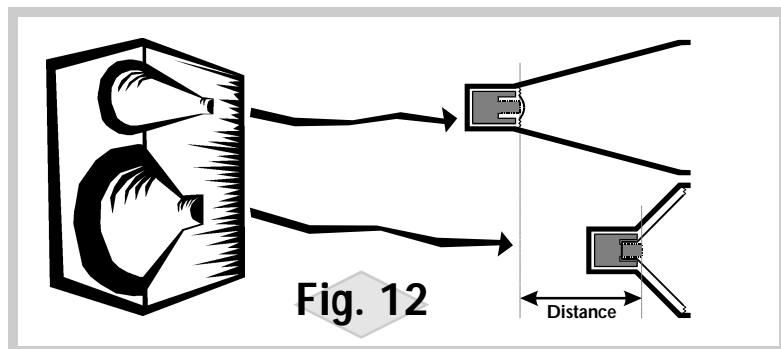
In some analogue crossovers, the FDS-360 for example, it is possible to add a circuit board onto which the desired EQ is pre-programmed. A better solution is offered by digital crossovers, where many bands of EQ are available for this component tuning. In the case of BSS Omnidrive products, many bands of EQ are available for each output.

Driver alignment

To understand driver alignment, we must first understand how sound travels. Any sound that we hear is as a result of compressions and rarefactions of the air. Because the transmission of sound relies on physical movement of the air, it takes some time to travel. The age old example of this is thunder and lightning: we see the lightning before hearing the thunder, even though they are

both the result of the same event. This is because light travels at about 300 million meters per second, whilst sound travels at about 330 meters per second. If the storm is 5 kilometres away we will see the light immediately, but we will not hear the thunder for 15 seconds.

This same time delay is true on a much smaller scale inside a loudspeaker. If we cut a typical loudspeaker in half (not a recommended practice!) you will see something resembling the diagram in figure 12.



In this example, you can see that the compression driver and cone driver are separated by a small horizontal distance. The sound from each driver will travel at the same speed, and as a result anyone listening to the speaker (before it has been cut in half) will hear the low frequencies slightly before the highs.

In an ideal situation the speaker designer will try to physically line up the centres of all drivers in a cabinet, but this is usually

difficult. Using time delay, it is possible to correct for this offset between drivers, delaying those closest to the front of the cabinet so that the sound from all drivers leaves the front of the box at the same time.

In the example from figure 12, the HF diaphragm and the cone driver may be separated by 0.50m, giving a delay of 1.5ms, as in figure 13a. To correct for this we can delay the cone driver by 1.5 milliseconds (see *The maths bit*). The listener will now hear the sound from both drivers at the same time, as shown in figure 13b.

How do I know how much delay I need?

We would not suggest that you do anything as drastic as actually cutting a cabinet in half to find out the displacement between drivers, but there are a few ways to get this information.

- Pick up the phone. This is the easiest method by far, most manufacturers will readily tell you the necessary delay times for their speaker systems.
- Use a delay time analyser. There are many systems available which you can use to give the necessary data.

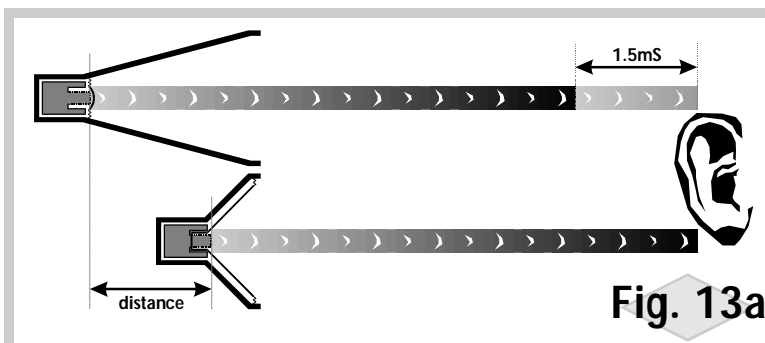


Fig. 13a

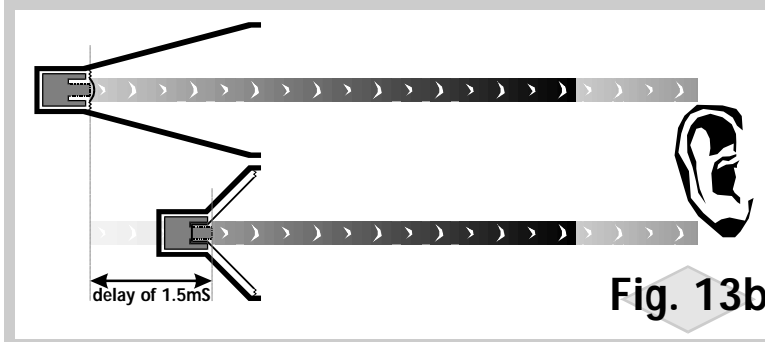


Fig. 13b

- Guess. Whilst the least elegant of solutions, you can actually achieve surprising results. If possible, take the grille off the speaker and measure the distance carefully with a ruler.

The necessary maths to convert your values into milliseconds are given below, but users of BSS *Omnidrive* technology can simply change the delay units into milliseconds, metres or feet and dial in the necessary data.

The maths bit

At 20 degrees Celsius, the speed of sound in air is

330ms^{-1} (metres per second). By using a simple formula we can easily calculate the delay time required for a known distance.

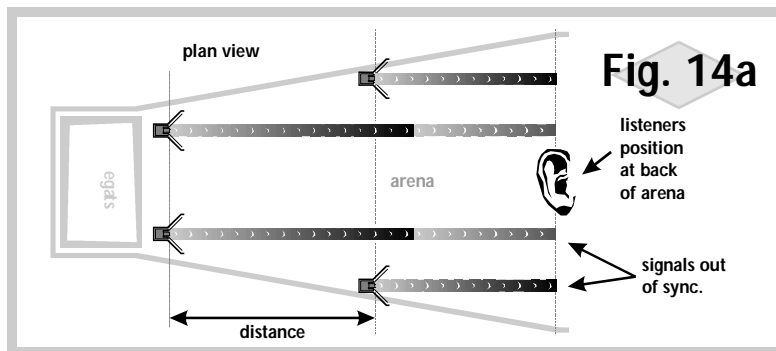
$$\text{Speed} = \frac{\text{Distance}}{\text{Time}}, \text{ so: } \text{Time} = \frac{\text{Distance}}{\text{Speed}}$$

Simply divide the distance by the necessary speed of sound in air and the answer is the time delay required. Remember that speed and distance should be given in the same units of measurement (eg: ms^{-1} & m).

Does 50 centimetres really matter? The simple answer to this is yes, size - or distance - does matter, and the difference is very noticeable. To illustrate how important these small distances are, think about your ears, which are separated by your head. This separation and its associated delay are used by the brain to determine where a sound comes from. OK, this is a simplification of the process, but it is a major factor.

Delay for 'delay speakers'

If you go to any large event, there are frequently speakers distributed around the venue. This is because it is difficult to maintain evenly distributed coverage from the main PA stacks, and so more speakers must be used. The example in figure 14a shows a simple setup with two main PA stacks, and another two stacks about halfway down the arena.



calculate how much delay must be added to the 'delay' speakers so that anyone behind them hears them at the same time as the main PA speakers. This maintains the listeners' focus on the stage and keeps the sound clear.

How do I calculate my delay times?

As explained in the driver alignment[®] section, you can use sophisticated measurement equipment, or a tape measure, to setup your delay time. This setting can be fine tuned by ear to get the best results. Often this involves playing an abrupt signal from a phase checker, or snare drum sample, through the system. The delay can then be fine tuned until only one sound is heard.

If all the speakers are fed with the same signal, the sound will leave each stack at the same time. This means that people at the back of the arena will hear the sound from the stack closest to them first, followed by the sound from the main stacks. An abrupt signal, for example a snare drum, will actually be heard twice.

Because of the way the brain interprets any sound heard, this delay between speakers can be quite uncomfortable. Our brain reinforces the sound that it hears first and then masks any sound that follows. This effect, known as the Haas effect, causes a dulling of the sound and a general loss of clarity. The listener also loses focus on the stage since it does not appear to their ears as if this is where the sound is coming from.

Delay can be used in this situation to ensure that those at the back of the arena hear the sound from both stacks at the same time (see figure 14b). If the distance between the stacks is measured, we can

Just when you thought it was safe to relax and enjoy the show

As mentioned, time delay is needed due to the time taken for sound to travel in air. This time is not constant, but depends greatly on temperature, so you'll need to alter your delays as things warm up. For example, if your delay tower is 100m from the main PA and the temperature is 20°C, a delay of 291.3ms is needed, but at a temperature of 25°C, the delay time should be 288.8ms. This delay adjustment can be made manually, although BSS delay lines and digital crossovers offer automatic adjustment of delays by way of a probe, or by just dialing the current temperature into the unit.

