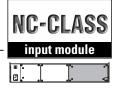
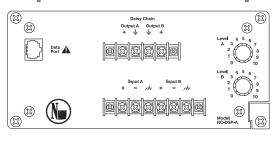
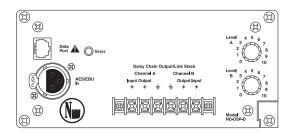
application note



dsp modules in amplifier systems







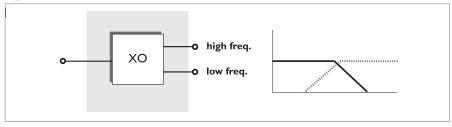
crossovers and speaker controllers

traditional crossovers

Most active crossover systems are implemented as low-level signal processors in rack mounted enclosures. They are intended to split a full spectrum signal into various frequency bands which drive external amplifiers connected to horns and woofers in biamp or tri-amp systems. An example of a two-way crossover and it's band-split spectrum is shown in figure 1.

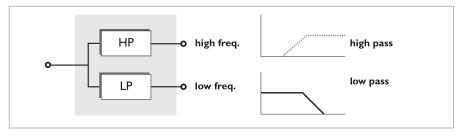
Octopus STO wiring employs a three-pin pheonix connector intended **only** for use with CC-STL modules.

figure 1



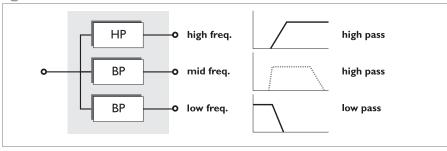
Crossover circuit functions usually include separate highpass HP and lowpass LP filters as shown in figure 2 for the bi-amp example. This allows independent control of each band's parameters: corner frequency, slope, and filter type, as required for a wide range of applications.

figure 2



Multi-way crossovers employ additional bandpass BP filters for the extra mid-band frequency splits as in figure 3 for a three-way system. Again, the separate HP, BP, and LP filters are to provide independent parametric control of each of these bands as required for the individual transducers.

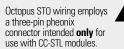




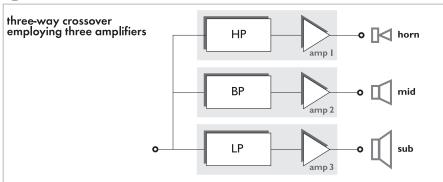
These filter blocks are the heart of all crossovers and, with the inclusion of equalizers, time delays, and limiters, are the basis for speaker controllers, as well. With the advent of digital signal processors DSP in amplifiers, these crossover and controller functions are implemented without the need for external devices. The allocation of the processing blocks is distributed among the DSP's in each amplifier as discussed below.

slicing crossover functions into amplifiers

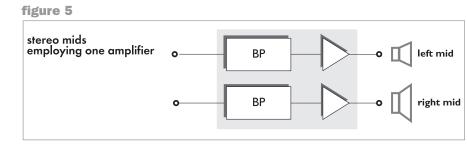
Each amplifier in a multi-way system has a particular band's transducer connected to it highs, mids, lows. The DSP in each amplifier provides a slice of the appropriate crossover function LP, BP, HP for the transducer driven by it. figure 4 shows the three-way system implemented with three amplifiers containing DSP's, which provide the crossover filters required only for that frequency band. As we will see later, the DSP's can also provide other transducer processing functions as well.







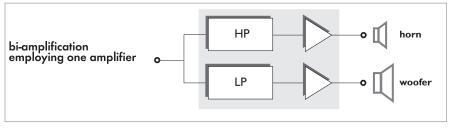
In a stereo, or a multi-channel system, one two-channel amplifier with DSP can provide a particular band's processing requirements for both stereo channels. The same slice of the crossover function is implemented for both signal inputs, and, since DSP based filters can be exactly duplicated without drift or misalignment, the stereo image is well preserved. figure 5 shows the BP function of a three-way stereo system implemented in one amplifier. In this case, each amplifier can be chosen for the appropriate power level for each transducer band.



crossovers and speaker controllers

In other cases, a system designer may want to use one amplifier with DSP to power a bi-amp configuration. The HP and LP slices of a crossover or controller are implemented in the one DSP. Fig 6 shows such an application.

figure 6

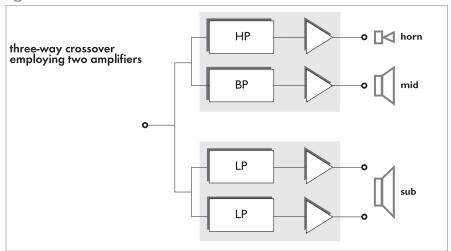


Octopus STO wiring employs a three-pin pheonix connector intended **only** for use with CC-STL modules.



A three-way speaker system can be realized using two-amplifiers with DSP, as shown in figure 7. Here, one amplifier provides the HP and BP crossover slices for the horn and mid-range drivers. The other amplifier provides the LP slice for the subs, which are driven in bridge mode. The DSP implements identical functions on each channel for the out-of-phase operation of the amplifier as required for bridge mode.

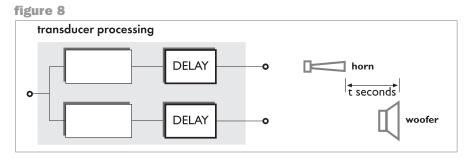
figure 7



transducer processing

Multi-way speaker systems typically have drivers whose acoustic centers are not in the same plane. A DSP can implement a delay line to compensate for the acoustic center offset of the furthest transducer and align all centers into the same plane. This eliminates combing in the frequency response at the crossover band edges.

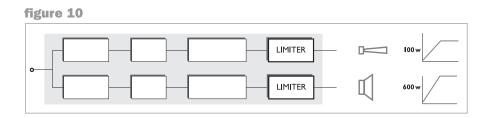
figure 8 shows a two-way system with a horn's acoustic center behind that of the woofer. This physical offset corresponds to a delay of T seconds and delaying the signal to the woofer by T seconds will align them.



The addition of an equalizer to the signal path can be used to compensate for variations in the frequency response of each driver. A four-band parametric EQ can equalize the overall frequency response anomalies of a given transducer in it's useable crossover band. Figure 9 shows the need for EQ for the drivers in a two-way system.



Protection limiters give safety margins for excursion limits and thermal overload of transducers. High slopes 20:1 provide for no limiting effect up to the power-handling capabilities of the driver as shown in Figure 10. The attack and release times are set to allow headroom for short-duration transients to exceed the threshold of limiting.



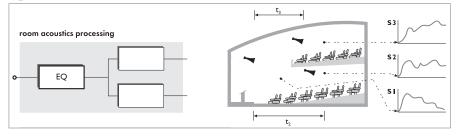
crossovers and speaker controllers

acoustics processing

With speaker systems positioned in various locations throughout a venue, the acoustic environment they serve will be different and require different equalization. A parametric EQ in the signal path ahead of the crossover, as shown in Figure I I, provides this equalization for each zone.

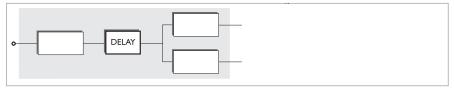
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figure 11

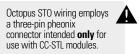


A long delay ahead of the crossover, as in Figure 12, Is used to coordinate the arrival time of the source sound with the local speaker. This eliminates the slap echo that occurs without such a delay.

figure 12



dsp modules in amplifier systems



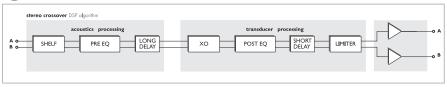
NC-DSP amplifier algorithm sets

The NC-DSP modules implement a set of algorithms that satisfy the vast majority of signal processing required in a signal path downstream of the power amplifier, specifically, for the transducer and the acoustic zone. Four major algorithm-sets provide the necessary Function Blocks in the signal flow to implement stereo, multi-channel, or mono crossover systems as well as full range mono systems. A Function Block is an algorithm segment that implements one of the seven unique signal processing functions required by the transducer and the zone (Equalizer, Shelf, Crossover, Delay, Limiter, Gain, and Mix). The appropriate Algorithm Set is chosen based on the speaker/amplifier configuration.

stereo crossover algorithm

The signal flow in this Algorithm consists of two independent channel paths with identical processing on each one. It is used in stereo multi-way crossover systems, where each amp channel services the same type of driver and frequency band for each stereo channel. It requires identical signal flow and identical function settings, which promotes better stereo separation and imaging.

figure 13



The first three Function Blocks SHELF, EQ, and DELAY are used for acoustic zone processing. The settings for each Function Block will typically be copied to each frequency band slice DSP in other amps servicing the same zone. The SHELF and EQ Function Blocks are available to equalize the acoustic environment and the long DELAY provides alignment for distributed systems

The last five Function Blocks XO, EQ, DELAY, LIMITER, GAIN are used for transducer processing. The settings for each Function Block are unique for each driver and frequency band in the system. The settings will be the same wherever a particular drive is used in a system and may be copied into the DSP Algorithm serving it. The XO crossover Function Block implements the band splitting function and the EQ Block is available to equalize the transducer response. The DELAY Function allows for acoustic center alignment of the transducers while the LIMITER provides protection for overdrive.

dual mono crossover algorithm

This Algorithm has the same flow as the Stereo Crossover, but, here, the settings of each channel's Function Block are independently adjustable. This is useful in multi-way clusters and array systems where each channel services a different driver and/or different acoustic zone but requires the same power level in the amp.

figure 14



The Function Blocks are the same as in the Stereo Crossover Algorithm, where the first three are for acoustic zone processing and the last five are for transducer processing.

An application of this Algorithm is an amplifier driving a long and a short throw horn, each serving a different zone. The same power is required in each amp channel, but the transducer parameters and the acoustic zone parameters differ and require independent settings.

mono crossover algorithm

Contraction of the

This algorithm has a signal flow to process a mono signal (or mix of two channels) and provide one set of acoustic zone Function Blocks and two sets of transducer Function Blocks. It is used in mono two-way or three-way systems where two frequency bands are covered by one amplifier.

rigure 15				
	mono crossover DSP algorithm			
А о В о	acoustics processing	transducer processing XO POST EQ SHORT DELAY XO POST EQ SHORT DELAY LIMITER DELAY		рА

A two-way monitor would require a High Pass Crossover on one channel and a Low Pass Crossover on the other. A three-way system might have one amplifier serving both a horn and a mid-range driver, with the HP and BP Crossovers implemented in it's DSP. A second amplifier, in bridge mode, drives the subs, with the LP Crossover is implemented on both channels as required for bridge operation.

mono full range EQ

This signal flow has two independent signal paths without a Crossover Function Block. It is intended for use in full range applications that are typical of distributed systems or multi-way cabinets with passive crossovers.

figure 16				
d	lual mono eq DSP algorithm			
A0				
Во	SHELF EQ DELAY LIMITER B			

The SHELF can be used to bandwidth limit the program material or enhance the low end and high end of the transducers. The five-band EQ is available to equalize the acoustic zone. A long DELAY is provided to align distributed speakers while the LIM-ITER protects them.

general set-up procedure

The overall approach to setting the DSP program's Processing Block parameters is, first, choose the appropriate DSP Algorithm for your amplifier/speaker configuration, then, setup the transducer processing Function Block parameters XO, EQ, DELAY, LIMITER, and, finally, setup the acoustic processing Function Block parameters SHELF, EQ, DELAY.

The sequence is as follows:

- Set Crossover according to speaker manufacturer's recommendations or mea sured driver frequency response
- 2 EQ each transducer band's spectrum individually to correct the driver's frequen cy response measurement
- Set Short Delay to align the transducer's acoustic centers per measurements or manufacturer's specs
- Set protection Limiters for operation below transducer's thermal and excursion limits as per specs
- 6 EQ each acoustic coverage zone individually using spectrum analyzer
- 6 Set Long Delay for distributed system wavefront alignment

measurements

Many acoustic measurement systems are applicable for setting up the DSP Function Block parameters: Dual Channel FFT, MLS, SIM and TEF among others. Each analyzer should be capable of measuring frequency response, or transfer function, and time delay, impulse response or coherence.

To setup Equalizers, Crossovers and Shelves use pink or white noise sources and measure frequency response and coherence. Alternately, use wide band program material and measure transfer function. The Delays can be setup using impulse response measurements, delay locator, or ETC measurements.

In all cases iterate through cycles of measure, adjust, and listen as you proceed through the setup sequence.

setting the crossover function block

Choose the filter shape (Low Pass, Band Pass, High Pass) according to which crossover slice frequency band the DSP and amp are serving. The filter type (Butterworth, Linkwitz-Riley, Bessel) is chosen for desired transition band performance. The Butterworth has maximally flat amplitude and is –3dB at the crossover frequency. The Linkwitz-Riley is similar to the Butterworth and is –6dB at crossover, giving smoother combining response in some cases. The Bessel has maximally flat time delay over frequency, but does not roll off as steeply as the others. Choose the Slope of the filter roll off based on the desired frequency overlap and combining between

bands. This is typically low for two way systems and high for three- and four-way systems. Set the Crossover Frequency per manufacture's recommendation or measured frequency power response.

setting the parametric EQ function block

These filters are used for acoustic zone or transducer equalization and have independent control of boost/cut Gain, center Frequency and Bandwidth. It is preferable to use cut rather than boost in equalization of systems as this reduces the possibility of ringing and/or overload.

The Parametric EQs provide features of both "constant Q" and "constant BW" equalizers by way of the symmetry control. When symmetry is enabled, the shape of the filter frequency response is symmetric about 0 dB for equal values of boost or cut. This is useful for general program oriented equalization or broadband attenuation for room effects since the attenuate shape is broader for a give bandwidth setting. If symmetry is not enabled, the attenuate shape of the filter is narrower and more notch like. This is useful for narrow-band EQ, as is required to notch out resonances in room or transducer frequency responses.

To adjust a particular filter, set BW to the narrowest, and Gain to full boost or cut to accentuate the spectral problem of interest. Sweep the Frequency control until the filter is aligned with the problem frequency, then adjust Gain to equalize the level to desired one. Finally, adjust BW to cover the full range of the problem. Several iterations of Gain, Frequency and Bandwidth adjustment will usually be necessary.

setting the shelf EQ function block

There are both an LF and HF shelving filter in this function block. The LF Shelf can be used to roll off low end response for speakers without low frequency handling capability. It can also be used to boost the low end response on subwoofers to extend their response. The HF Shelf can be used to boost high end to compensate for driver deficiencies or to produce the Constant Directivity EQ required by some horns. Alternately it can be used to cut high-end for bandwidth limiting of program.

setting the time delay function block

There are two time delay controls; one for short delay to align acoustic offset and another to provide long delay for distributed systems. To align the acoustic centers, adjust the Short Delay to add delay to all transducers but the one with the longest offset. This has the effect of moving all other drivers "back" to align with the furthest acoustic center and you will be adding delay equal to the difference from the furthest one. Use an analyzer with microphone near the speaker system to measure these offsets or compute the differences from spec sheets or, for a rough estimate, measure voice coil locations (not recommended).

Setting Long Delay for distributed systems is accomplished in a similar manner using the analyzer with microphone in the far zone. Measure the delay from each source separately and add the difference to the closest source. You can also measure the distance and enter the delay in feet and get results. In either case, you may want to add another 5 mS to allow for Haas effect (the human ear localizes on the first arriving sound source even if it is not the loudest). Thus, the perceived source location will be from the stage, or central cluster, not the distributed reinforcement speaker.

setting the limiter function block

The Protection Limiters are set to prevent overdriving the transducers, since amplifier power levels are typically greater than the long-term power rating of the drivers. This provides headroom before clipping for good transient performance. The ratio of compression is 20:1

Set the Threshold for desired power handling limits of all drivers connected to the amp channel. A Threshold of 0 dB gives no limiting, while a threshold of -6dB places the limiter onset just at amplifier clip level (the DSP system clips at +16dBu input while the amplifier clips at +10dBu, giving 6 dB of headroom). To protect the drivers, set the Threshold X dB below -6dB, where

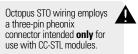
 $X = 10\log(P \text{ limit } / P \text{ amplifier}).$

Here, P limit is the desired protection power for the drivers and P amplifier is the power rating of the amplifier channel. As an example for a 100W amp and a protection power of 50W (-3dB), set the Threshold at -9dB (- 6dB headroom, - 3dB power ratio).

The Attack time should be set long enough to allow short duration transients to pass. This is, typically, twice the period of the lowest frequency in the crossover band. The Release time should also be long enough to prevent audible pumping and level modulation after a signal passes below the Threshold. This is usually 20 times the period of the lowest frequency in band. Thus :

T attack = 2 / F crossover and T release = 20 / F crossover

dsp modules in amplifier systems



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figure 5

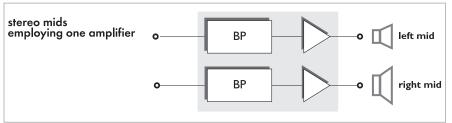
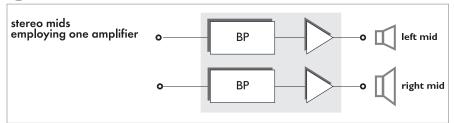


figure 5



dsp modules in amplifier systems

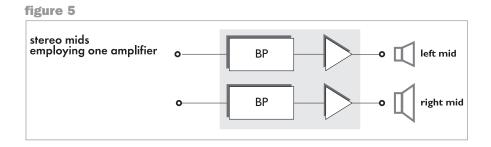


figure 5

