Electro-Voice
CROSSOVER COOKBOOK

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Setting-up equalization and crossover networks for active multi-way systems has always been a very time-consuming and tedious task with analog crossover units because of the limited flexibility of those units. The Electro-Voice digital loudspeaker processor Dx38 on the contrary offers extreme flexibility and an audio quality level, which outperforms even the most sophisticated analog units by an order of magnitude. But setting-up loudspeaker systems with such an advanced unit has become even more complex due to the enormous number of different parameters that can be accessed.

In a former paper [1] it has been shown that in most cases it is necessary to know the acoustic transfer function of a speaker system, otherwise equalization and crossover functions cannot be adjusted properly. Only a limited number of customers have access to an anechoic chamber and the expensive measuring equipment. Speaker system measurements in real rooms often suffer from severe reflection problems especially at low frequencies, from ambient noise, from a limited measurement resolution and from the cost of appropriate measuring equipment.

But with the Electro-Voice RACE (Real-time Acoustic Cluster Editor) software package, written by Michael Aumer, it is now possible to display the on-axis anechoic frequency response of a loudspeaker system combined with the crossover, equalizers and the power amplifiers on a PC in real-time. In addition, the software shows the position-dependent on-axis frequency response and the steady-state vertical plane low-frequency SPL distribution of a loudspeaker cluster including all equalization parameters, crossovers and power amplifiers.

Parameter adjustments on the Dx38 controller can be made, displayed on a PC and heard on-the-fly in real-time. The design of crossover and equalizer functions can be done in a straightforward manner with the RACE software, which is a significant advantage, compared with trial-and-error set-up practices.

Many customers only infrequently are doing set-ups of active multi-way systems. This paper has been written in order to give some condensed "cookbook" information on setting-up loudspeaker systems without any mathematics involved. Some basic information on equalization, crossover filters and the graphical representation of phase response has been included in order to support customers who are not dealing on a daily basis with signal theory and linear circuits.

The RACE software comes with every Dx38 unit and can be downloaded from the Electro-Voice website for free.
Loudspeaker files and amplifier files are available on request for interested customers.

1 A Thirty Minutes Cookery Class on using RACE for the Design of Your Personal Crossover Presets

1.1 Appetizer – Active 2-Way System with Sx300 and Sb121

Our appetizer will be an active 2-way system with Sb121 and Sx300 and we want to have a linear frequency response 50Hz-20kHz of the system first. Some eq spice can be added later.

First we have to bring our ingredients to the kitchen.

We start the RACE software. The program comes up with the Stereo 2-Way Mode. We load the Sb121 file (Sb121_S00) and the Sx300 file (Sx300_S00) into the speaker menu.. We will use the built-in default power amplifiers but we also could easily load P-Series and Q-Series amps. But this is only necessary if we would use different models e.g. Q66 for the Sb121 and Q44 for the Sx300.

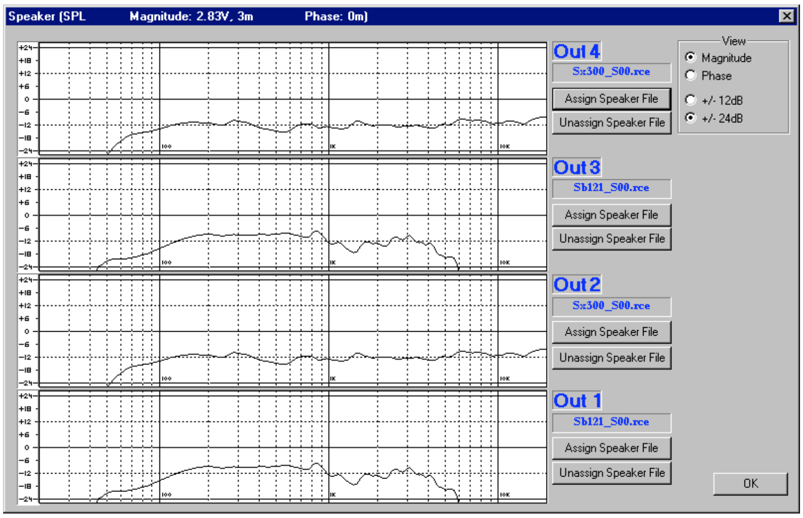


Fig. 1 Frequency response of Sb121 and Sx300

A visual inspection of our components shows, that both cabinets can deliver sufficient SPL around our planned 100Hz crossover point so we do not have to expect severe problems.
From [2] we know that the phase response of the system components should be matched in the vicinity of the crossover point, so we now have to look at the phase response of our components.

We click the "View Phase" and Fig. 2 shows up on the screen. WOW…never seen before.

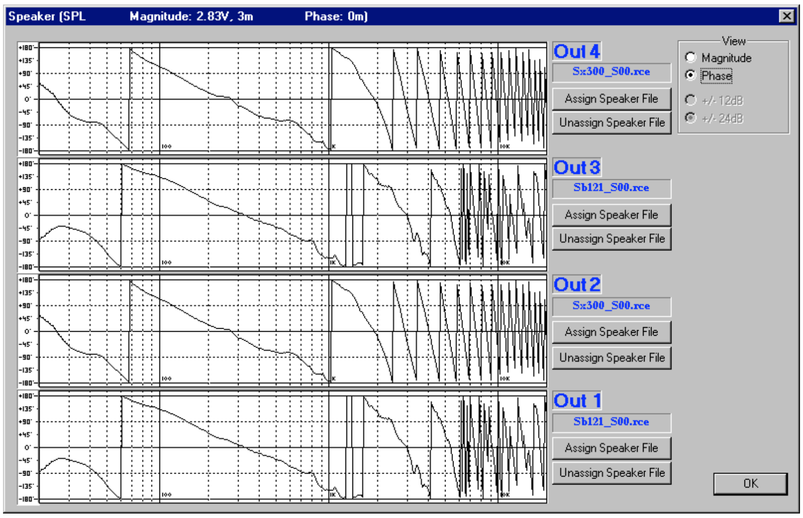


Fig. 2 Phase response of Sb121 (1,3) and Sx300 (2,4) at grille position

The phase responses around 100Hz are not very different; hence aligning both cabinets into equal acoustical plane will be fairly easy.

We now open the x-over menu where the popular Linkwitz-Riley 24dB/Oct. crossover function is applied by default. We now mute channels 3 and 4 in order to prevent any confusion when adding outputs acoustically. The “Sum” always sums up all channels but we only want to see one side of the stereo setup.

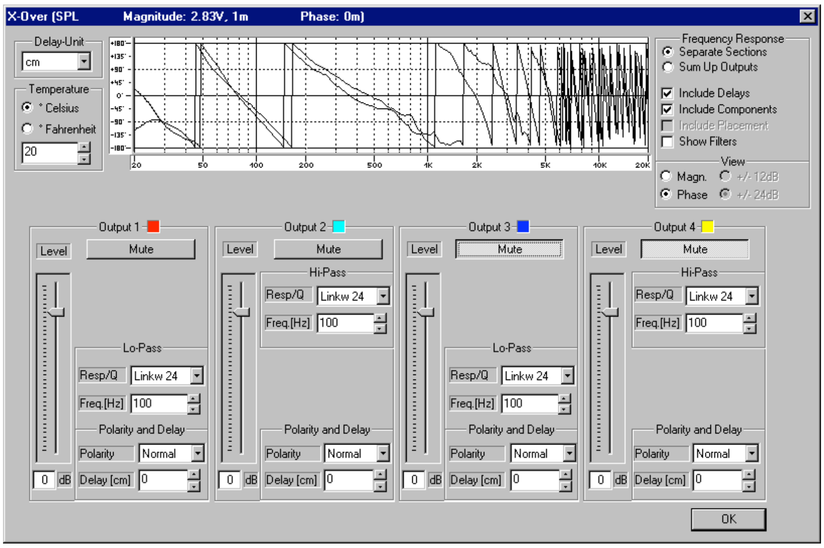


Fig. 3 Phase response of Sb121 and Sx300 driven by a Linkwitz-Riley 24dB/Oct. crossover

In Fig. 3 we can now see the phase response of the Sb121 and of the Sx300 combined with the respective Linkwitz-Riley crossover functions.

Clicking “View Magn.” brings us back to the frequency response of our components now including the active crossover.

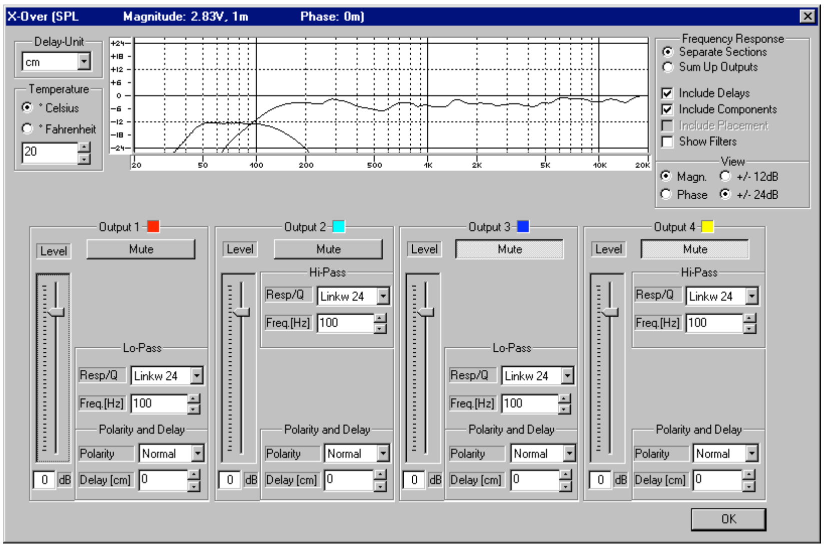


Fig. 4 Frequency response of Sb121 and Sx300 with Linkwitz-Riley 24dB/Oct. crossover

The acoustic output from 50Hz – 100Hz of the bass cabinet has to be raised a little bit in order to achieve a linear frequency response of the system. We want to use an overall equalization to prevent phase problems so we will click to the “Master Eq” menu.

At the first glance we see nothing, we have to mark the Out 1 and Out 2 boxes in order to see our outputs.
We now equalize our roll-offs with a 12dB/Oct. low shelving eq at 100Hz and set the gain to 6dB. For speaker protection against unwanted rumble and low-frequency overload we add a high-pass filter with 12dB/Oct. slope, 50Hz pole frequency and Q=1.

The resulting frequency response of our system can now be seen in Fig. 5, the upper curve shows just the electrical response of our eq filters so we can see how much additional dBs we are driving out of our power amplifiers into the cabinets.

The lower curves now show the frequency response of our cabinets combined with the crossovers, the 100Hz/12dB/Oct. shelving eq and the 50Hz/12dB/Oct. Q=1 high-pass (Lo-Cut) filter.

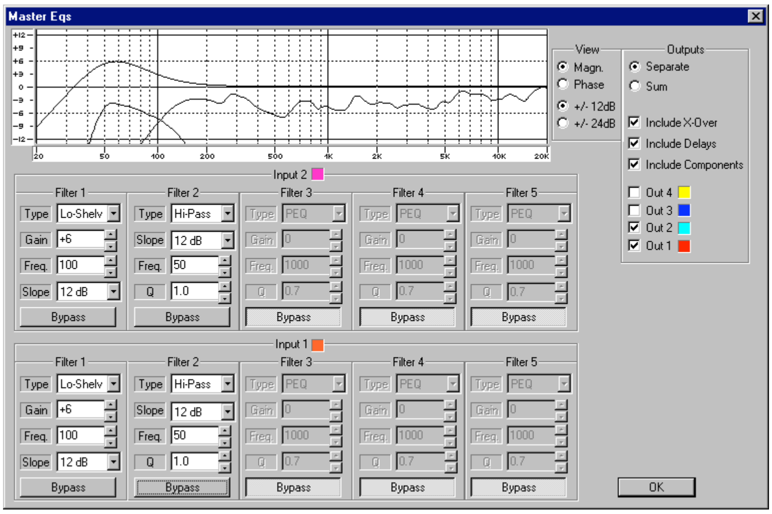


Fig. 5 Eq and Lo-Cut frequency response (upper curve), acoustic response (lower curves)

We now go back to the X-Over menu and click the "Sum Up Outputs" button because at our ears we receive the sum of the acoustic output of the Sb121 and Sx300. The level for the Sb121 has been raised by 3dB for an improved matching of the respective acoustic outputs. The result is quite good for a cook apprentice and is shown in Fig. 6.

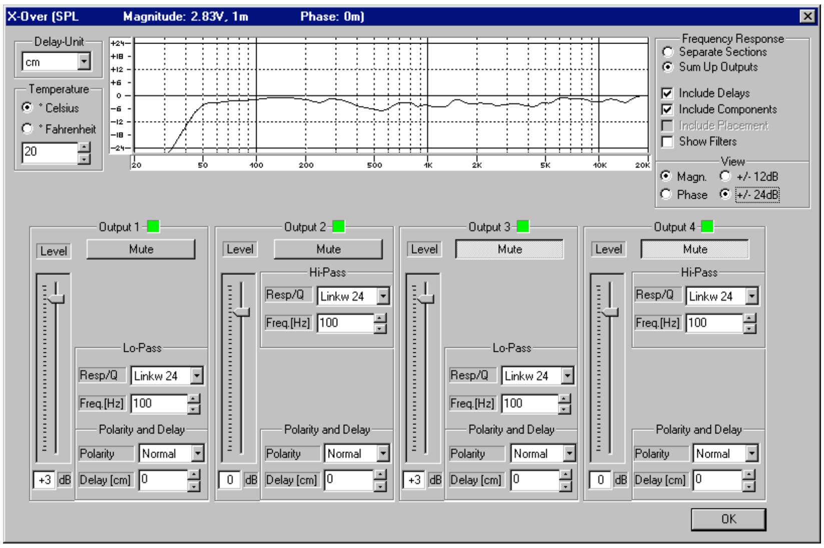


Fig. 6 Output 1 (Sb121) and Output 2 (Sx300) summed together

We now can fine tune our alignment for positioning the transducers in "equal acoustical plane" and click the "View Phase" and the "Separate Sections" buttons. Fig. 7 shows the curves without the adjustment for matched phase.

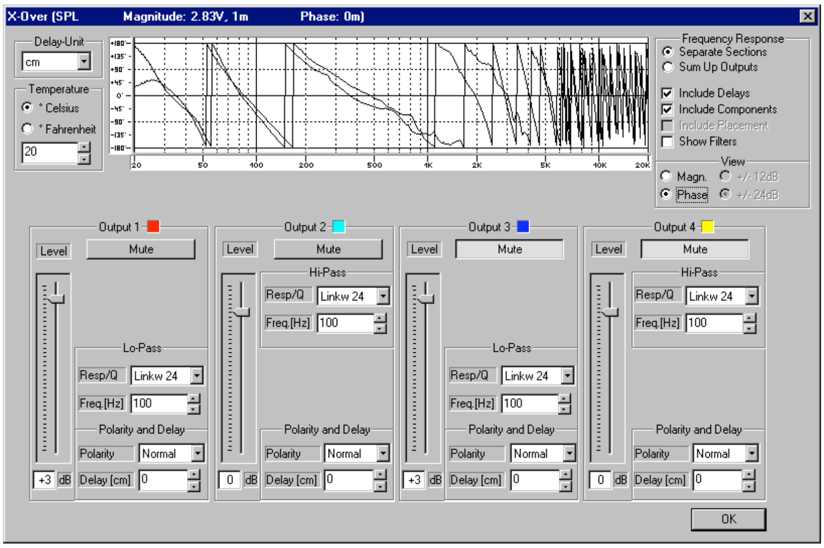


Fig. 7 Phase response without fine-tuning for matching near the crossover point

Adding now an alignment delay of 500us or 17cm to the woofer brings the phase into perfect match in the vicinity of the 100Hz crossover point as can be seen in Fig. 8.

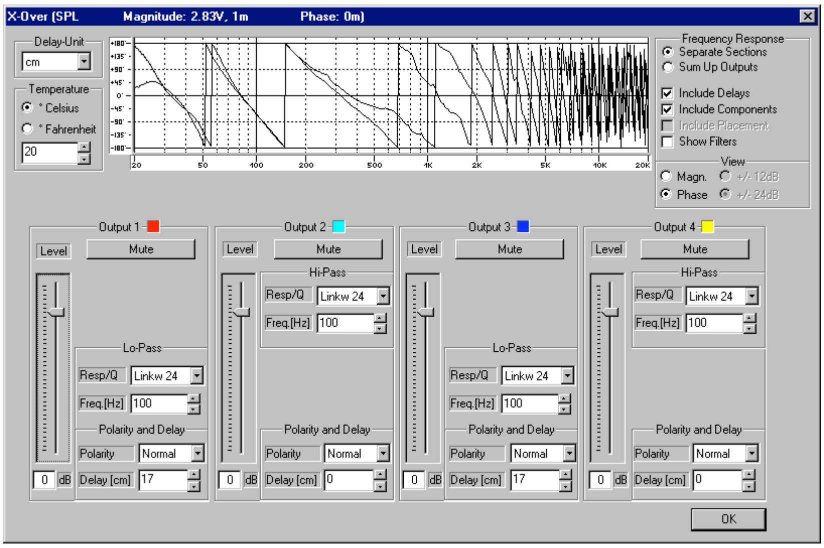


Fig. 8 17cm alignment delay added to the woofer

In [2] a simple method has been published how to check whether the aligning was done correctly. One has to look at the summed level response and invert the polarity of one channel. If a deep notch occurs the alignment has been done properly.

One has to be a little bit cautious because a notch occurs again if one has added accidentally “one or more wavelengths” during the aligning process. In our example here if we set the alignment delay to 3.44m we again would see a notch in the summed response when inverting one channel.

But clicking back to the “View Phase” and “Separate Sections” immediately uncovers that something must be wrong. We still have equal “relative” phase at the crossover point but in the vicinity the phase curve of the woofer has a much steeper slope and we are approximately 3.27m distant from the correct position. Fig. 9 shows this misalignment.

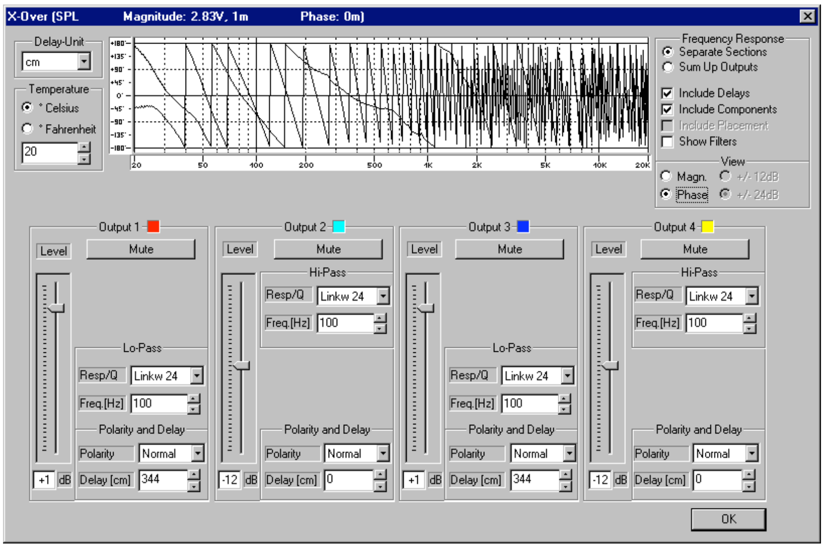


Fig. 9 Misaligned crossover with still perfect match at the 100Hz crossover point

Our appetizer is now nearly ready for serving but we can do some additional fine-tuning in order to bring maximum satisfaction to our customers. We will do this in the “Master Eq” section. Any additional modifications will then not change our phase relations between Sb121 and Sx300.

When fine-tuning is done in the Channel Eqs, one has always to re-check the delay alignments because the phase relations will change with every modification of a channel eq. This can be time consuming in more complex systems.

We have 3 additional eqs available in the master eq section and can use 2 of them to flatten out the slight peaks at 144Hz and 300Hz as can be seen in Fig. 10. The remaining eq in the master section can now be used to make an overall adjustment to the personal taste of a client.

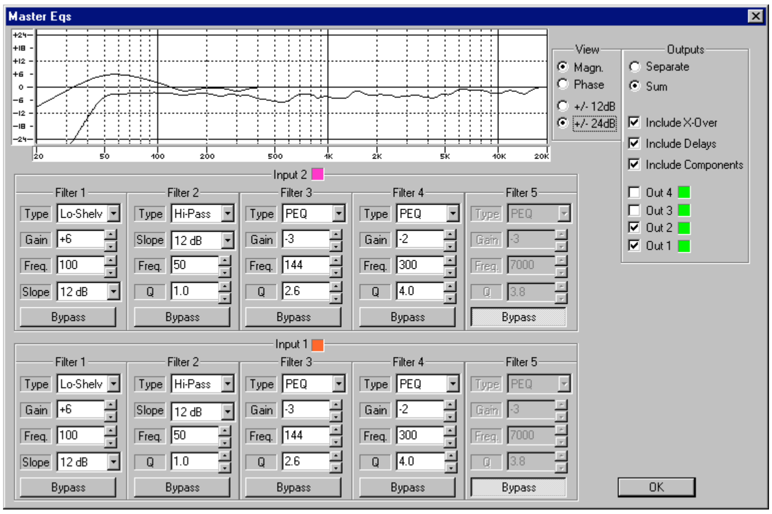


Fig. 10 Slight peaks at 144Hz and 300Hz flattened

Filter 5 in the Master Eqs is now set to a center frequency of 1kHz, and a Q=0.7. The filter acts very similar as a regular linked bass-treble control. Fig. 11 shows a 1kHz boost of 6dB which would make the system sound more aggressive but with a very good intelligibility for speech. Normally such an extreme boost should be avoided and this is shown here only for demonstration purposes.

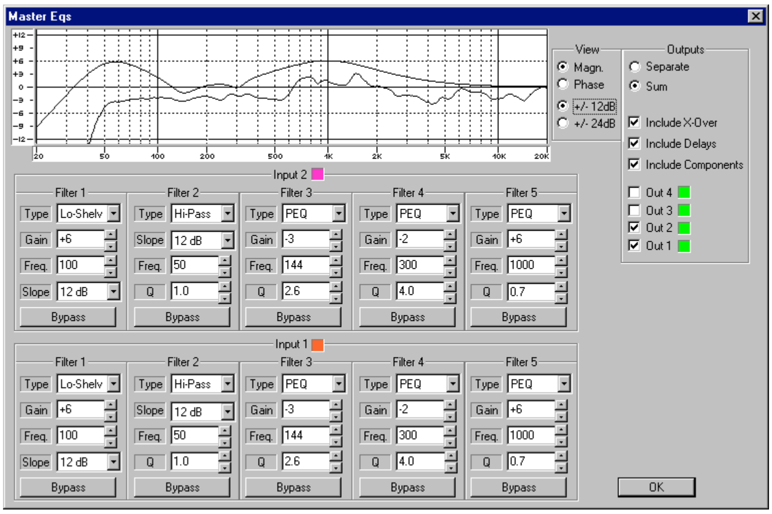


Fig. 11 Extreme boost at 1kHz, high intelligibility, slightly aggressive

We now turn the gain control of filter 5 into the opposite direction and lower the 1kHz level by –6dB. The system now sounds overly friendly and the vocals “move back” significantly. This is another extreme just shown for demonstration purposes and should normally not be used. Fig. 12 shows the eq settings and the resulting frequency response.

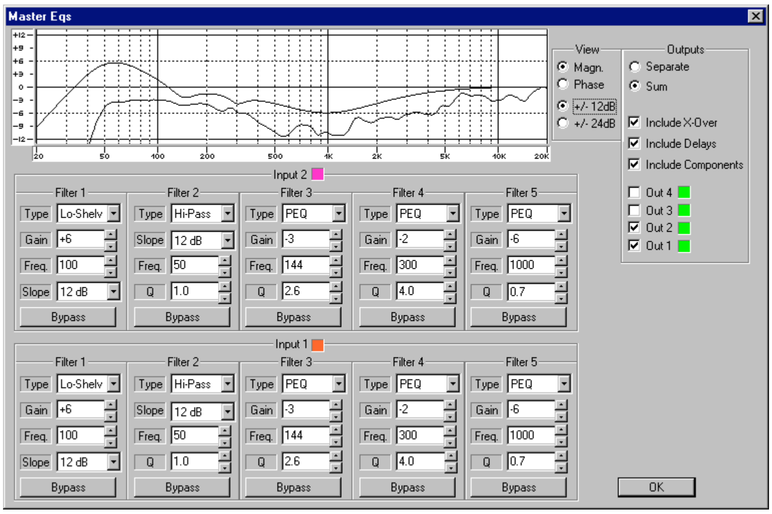


Fig. 12 Extreme attenuation at 1kHz for people who do not like the lyrics

We have now completed our appetizer and are ready to enjoy our client.

One week later our client calls and after some nice words tells us that he is quite satisfied but “…. that his stomach asks for more punch….”. We look into our nice preset and still think that everything is o.k. Nevertheless we jump in a car and rush to the client´s club. The installer has placed the Sb121s in front of the stage and the Sx300s at the most convenient mounting positions some 1.5m behind in 2.5m height. As fast as we can we power-up our laptop, load our preset including cabinets, click the placement menu button and punch in the coordinates of the cabinets. Fig. 13 shows the situation. We have a pronounced “hole” around 100Hz at the customer’s ear and stomach.

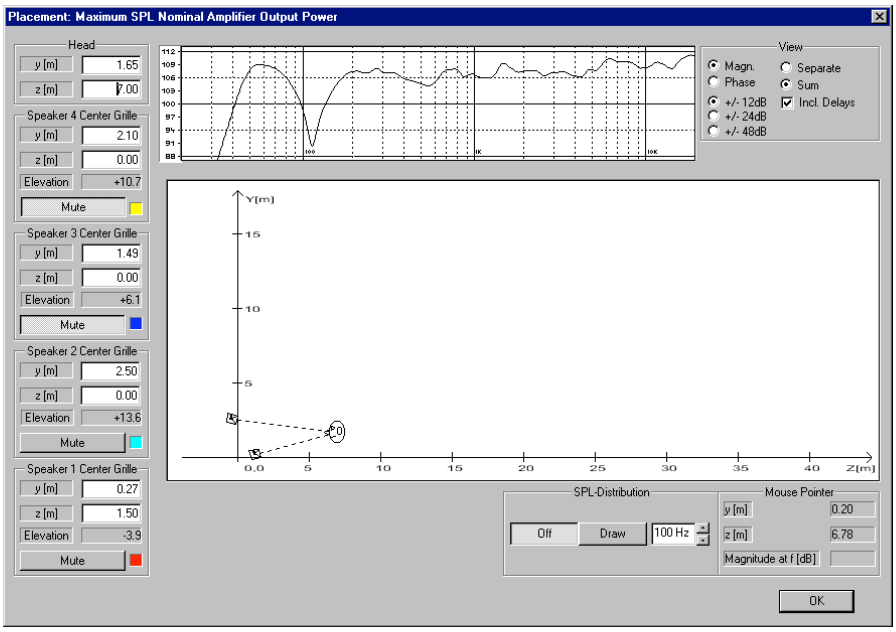


Fig. 13 A hole in the frequency response due to distance difference induced interference

The travel distance to the ear and stomach of the customer is approximately half a wavelength at 100Hz due to the chosen mounting positions and hence the signals are “out of phase” at the aiming point. Pushing now the “Draw” button shows the steady state SPL distribution at 100Hz in the vertical plane (without any reflections). Fig. 14 shows, that the main lobe points toward the ceiling and not to the stomach of the client.

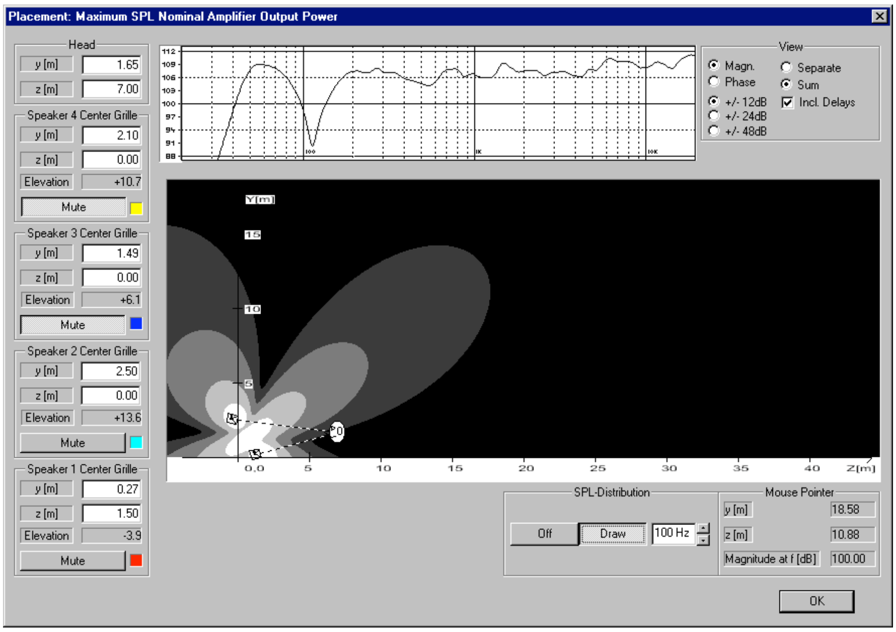


Fig. 14 Upwards rotated main lobe at 100Hz due to horizontal displacement of the cabinets

We now have a lengthy discussion with the client and the installer on changing the positions of the cabinets to the optimum place but could not convince any of them. But thanks to the Dx38 and RACE some relief is at hand to improve the situation. We ”shift” the Sb121 back 1.5m increasing the alignment delay to 1.67m and have the significantly improved situation shown in Fig. 15. The hole at 100Hz disappeared and the main lobe now massages the stomach of our client.

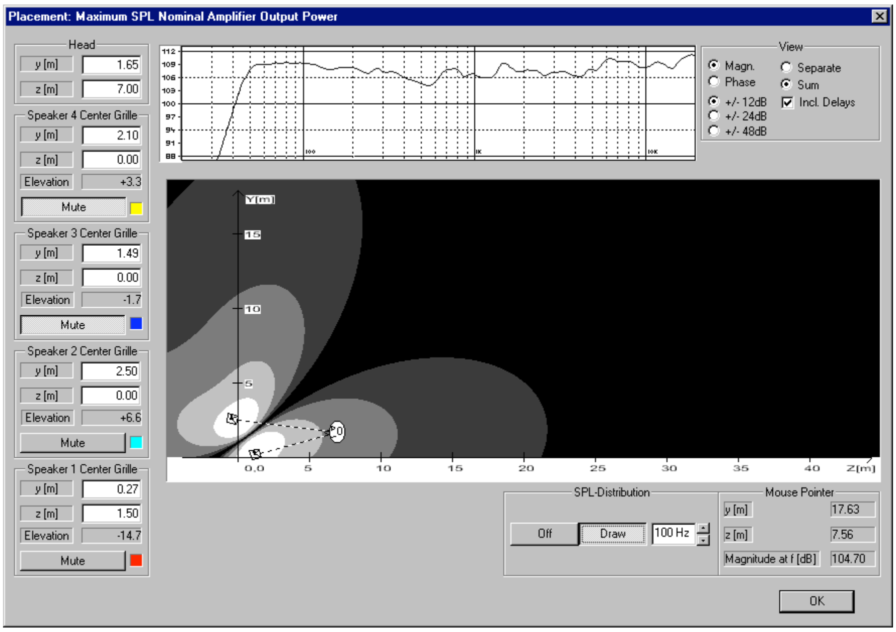


Fig. 15 Main lobe at 100Hz rotated downwards using an additional 1.5m alignment delay for the Sb121

The optimal physical positioning of the cabinets in the same vertical plane still would be a better as can be seen in Fig. 16 because adding some additional delay to a cabinet does not change the point of origin of the sound wave for the respective cabinet. Nevertheless, the improvement in Fig. 15 fully satisfied our client.

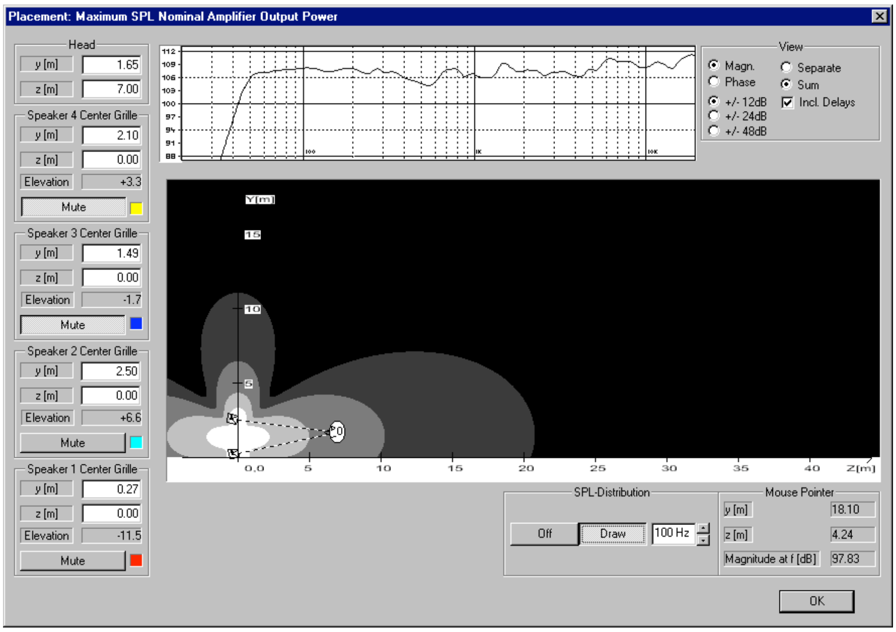


Fig. 16 Optimal positioning without additional delay

Whether such irregularities are audible or not heavily depends on the acoustics of the environment. Open air and in large, well-damped rooms such irregularities can be very irritating. In small or very reverberant rooms the early reflections and the reverberant sound often are dominant and even ridicuously large distances between the cabinets subjectively do not make much difference.

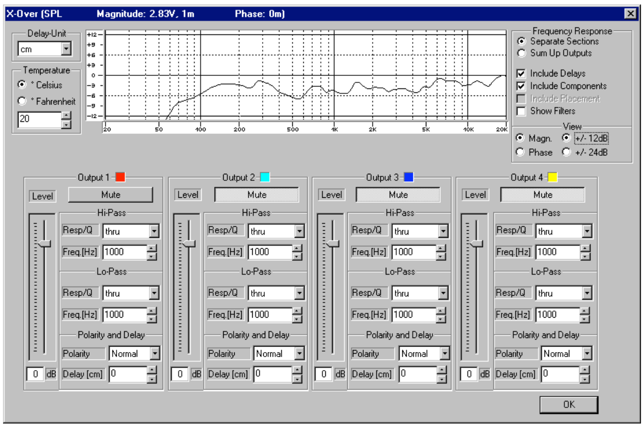
1.2 Desert – Using the Dx38 as a Powerful Equalizer

The Dx38 with all the filter functions, the built-in delay, the limiters and compressors together with the RACE software forms an extremely powerful digital signal processing system for general purpose full-range applications. As a fully programmable 2-In-4 unit with free routing of the outputs one can feed up to 4 different speaker zones with completely independent parameter settings and different presets can easily be recalled directly from the unit or via the contact closure option.

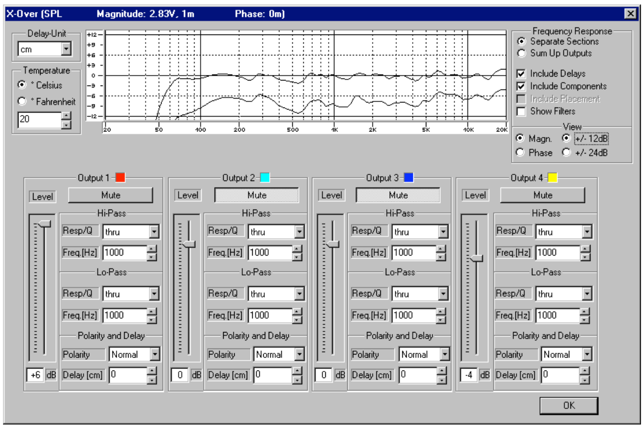
A typical example would be a small club where the Dx38 could drive a stereo system for a performer, the ceiling speakers in the rear part and small utility speakers in the bathrooms. Every different zone could be given a different equalization, a delay for the ceiling speakers, a level limitation for ceiling speakers and the bathroom and one could make different presets for different events.

Doing the same with conventional Equalizers, Limiters, Delay units and Compressors would cost a fortune, would eat up a lot of rack space, would be difficult to set-up and operate and would not even come close to the immaculate audio performance of the Dx38.

As an example we show here the unequalized frequency response of an Sx300, which is already excellent. For some applications additional equalization can bring some benefits with regard to extended frequency response and improved suppression of acoustic feedback. We open the RACE program and switch to full-edit 2-In-4 mode and load our Sx300_S00 file.



We add some low-frequency equalization, a high-pass filter for removing unwanted rumble and flatten out the response with different shelving and parametric eqs. The resulting frequency response is extremely smooth and the low-frequency cut-off has been shifted down approximately ½ octave. For reasons of clarity the equalized curve (upper) and the non-equalized (lower) have been spaced 10dB apart.



2 Design Targets

The never ending debate on the perceived quality of different sound systems and different equalization philosophies seems to suggest that there is no common method to set up a system. But as a matter of fact, good sounding systems always show similar measurement results e.g. a fairly linear frequency response and measurable and perceived differences often are attributable to the personal taste of the operator during fine tuning. Every good sounding system has to fulfill first some basic requirements on frequency response, distortion, maximum achievable SPL and coverage of the audience and can then always be fine tuned to the operator´s taste. The opposite is not necessarily true in all cases.

2.1 Frequency Response, Maximum SPL, Distortion

Live music sound reinforcement systems should create a SPL of approximately 100 dB (peak) at the listener´s ear and should have a useable frequency response of approximately 40Hz to 15kHz in order to sound fully satisfactory and realistic. This does not mean that the –3dB corners have to be at 40 Hz and 15kHz for a sound system in order to sound good, but the –10dB points should be somewhere between 40Hz and 80Hz and 10kHz and 15kHz, respectively. The level variations in the frequency response should be comparably low, as a target midband +- 3dB or better, a requirement not so easy to achieve for high SPL components. Linearity is extremely critical in the vocal range, let us say between 500Hz and 2kHz where level differences of just 1dB can easily be heard and severe misadjustments can lead to a loss of “music” and a bad intelligibility.

There are sometimes special requirements for a significant lower cutoff frequency of a system for some disco and cinema applications which can only be satisfied with special low-efficiency subwoofers. On stage the reproduction of extremely low frequencies can create a lot of problems with rumble, mechanical overload of speakers and weird low-frequency feedback. Frequencies well above 12kHz -15kHz cannot be projected sufficiently loud over any significant distance due to the absorption in air and diffraction effects created by temperature differences of the environment.

100dB maximum SPL at a listener position, 30m distant from the sound system, means that the system has to have a (calculated) maximum of approximately 130dB SPL. Distortion figures should be sufficiently low from a purely subjective point of view, especially in the vocal range in to order to achieve real customer satisfaction. Even the best speaker components with highest sensitivity figures create THD figures of 10% and more at such high levels. If the maximum achievable SPL is not sufficient in a system one simply has to use significantly more cabinets or some sort of a distributed system e.g. delayed loudspeaker stacks. Increasing the output power of amplifiers can help a little bit in some cases but the mechanical and physical limitations of transducers cannot be overcome by any brute force approach.

2.2 Polars

The sound system should cover the whole audience as evenly as possible. For vertically arrayed speaker components even with non-coincident drivers it is not difficult to have acceptable horizontal dispersion. But due to the always existing vertical interference patterns [2] there can be regions inside the audience where the sound system can sound consistently bad, unmusical or uninspired in the midrange walking from left to right. Coaxial systems inherently are much less sensitive to such effects [1], [2] in the critical vocal region compared with stacked component systems for large-scale sound reinforcement.

The physical size of typical coaxial systems can often not be used in medium to small applications and one has to use vertically stacked components with non-coincident drivers e.g. a typical high-power 2-way cabinet. In order to prevent vertical interference pattern problems in the mid-range with such cabinets it is often best to choose the acoustic crossover point between the components as high as possible let us say somewhere between 2kHz and 3kHz. The resulting multi-finger high-mid vertical lobing is subjectively not nearly as destructive as projecting a pronounced vertical-off-axis 1kHz notch to some parts of the audience.

In addition, distortion figures of compression drivers rise dramatically in the region of horn and driver cutoff frequencies, so shifting the crossover point as high as possible lowers the perceived distortion of the system and increases the reliability of compression drivers significantly.

For low-level applications e.g. cinema 2-way systems the acoustic crossover point in the mid-range can be chosen well below 1kHz. Vertical polar pattern problems then will not exist due to the increased wavelengths at the crossover point compared to the physical distance of the transducers [2].

3 Visual Inspection of the Frequency Response of Components

Before we can start designing a crossover we have to have a look at the raw frequency response of the loudspeaker components in order to know what region of the speaker can be used for the required application. As a matter of fact, there is not much freedom in choosing crossover points between different components due to the mechanical nature of transducers and basic laws of acoustics.

The following gives a very short overview on the behaviour of typical professional components and can be used as a guideline.

3.1 Large Format Bass Cabinets

Fig. 17 shows the frequency response of a typical high quality large-format bass cabinet equipped with two 18” transducers. The SPL with 2.83V drive level is approximately 98dB and the –3B corner frequency is about 50Hz. Below 50Hz the cabinet´s frequency response rolls of with approximately 24dB/Oct. Above 300Hz the frequency response becomes very irregular which is typical for large high power subs. The frequency response inside the useable range is excellent and needs only a very moderate amount of low-frequency equalization. The cabinet can be safely used up to 200Hz even though a lower crossover point around 80Hz-150Hz is often preferrable between subs and lo-mid cabinets. The steep 24dB/Oct. roll-off below 50 Hz cannot be equalized because enormous amounts of amplifier power would drive the transducer into mechanical and thermal problems without having any significant additional acoustic output

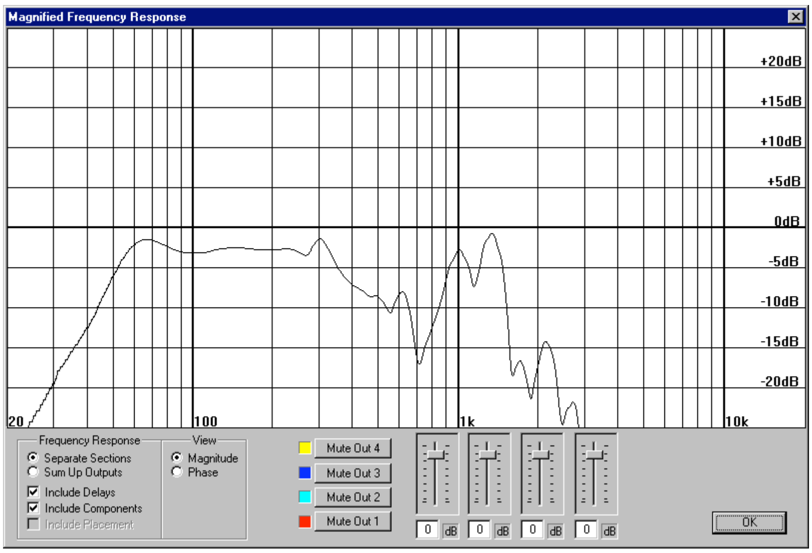


Fig. 17 Frequency response of a dual 18” high-quality bass cabinet

RECIPE 1
LARGE-FORMAT BASS CABINETS NEED ONLY A MODERATE AMOUNT OF LOW FREQUENCY EQUALIZATION.

RECIPE 2
CROSSOVER FREQUENCIES TO LO-MID CABINETS SHOULD BE BETWEEN 80HZ AND 150HZ.

3.2 Small Format Bass Cabinets and Full-Range Cabinets

Fig. 18 shows the frequency response of a typical small-format bass cabinet equipped with a 12" woofer. The cabinet´s frequency rolls off gradually below 200Hz with approximately 8dB/Oct. followed by a steep roll-off below 50Hz. The cabinet's frequency reponse is excellent up to 3kHz and rolls off rapidly above. The cabinet can be used as sub because the gradual roll-off to low frequencies can be equalized easily.

The cabinet could also be used as a high-quality low-mid component in a small active 2-way system due to its excellent linearity in the mid-range. SPL around 500Hz is approximately 100dB/2.83V/1m

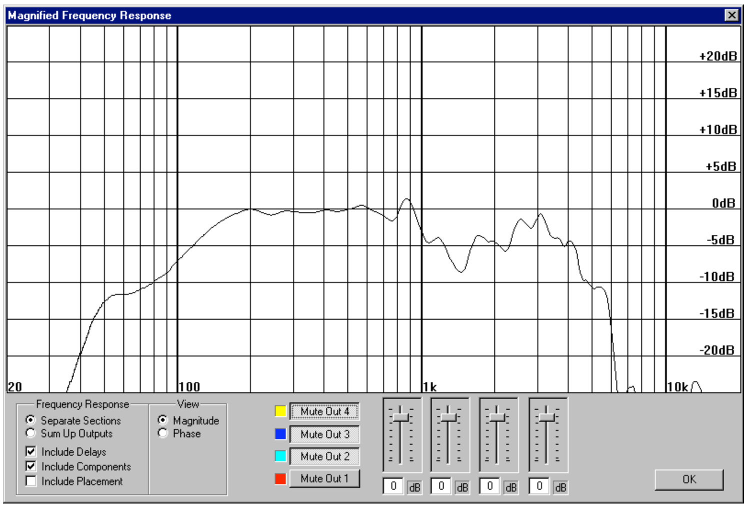


Fig. 18 Frequency response of a 12" small-format bass cabinet

RECIPE 3
SMALL FORMAT BASS CABINETS WITH A GRADUAL ROLL-OFF CAN BE EQUALIZED FOR AN EXTENDED BASS RANGE.

In Fig. 19 the frequency response of a very popular 12" 2-way cabinet is shown. The low frequency roll-off starts with a slope of approximately 12dB/Oct. and rolls off more rapidly below 30Hz. High-frequency SPL is excellent, the mid-range linearity is quite good.
The cabinet can be used without any significant additional equalization as mid-hi component in a typical small active 2-way system with an appropriate sub. One would choose a crossover point about 100Hz for such a system. SPL is approximately 96dB/2.83V/1m. Such components therefore are excellent for small venues but not for medium and large-scale sound reinforcement applications.

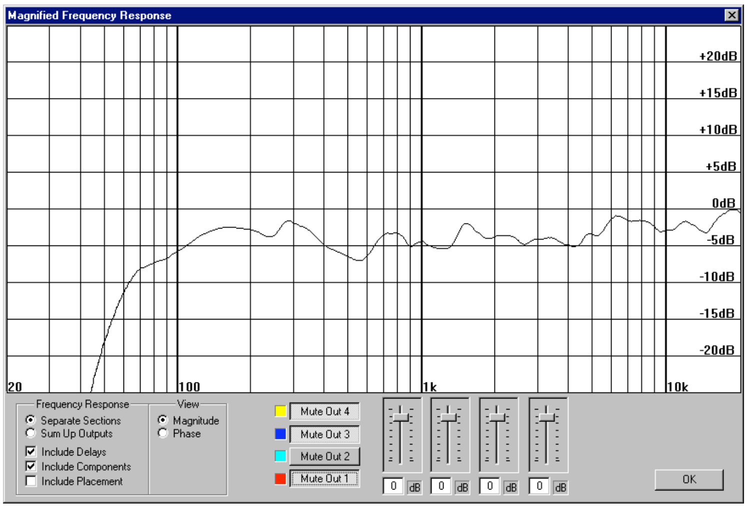


Fig. 19 Frequency response of a medium sensitivity 12" 2-way full-range cabinet

RECIPE 4
CROSSOVER FREQUENCIES BETWEEN NORMAL FULL-RANGE CABINETS AND SUBS SHOULD BE CHOSEN BETWEEN 80HZ AND 120HZ.

3.3 Horn-Loaded Lo-Mid Cabinets

Good direct radiator cabinets for sound reinforcement can reach a mid-range sensitivity of approximately 100dB/2.83V/1m. Even when driven with large power amplifiers the maximum SPL will not exceed 120dB/2.83V/1m significantly, due to thermally induced power compression and mechanical limitations of the transducers.

For medium and large-scale concert sound applications therefore the sensitivity has to be an order of magnitude higher otherwise a satisfactory SPL over longer projection distances cannot be achieved. Horn-loading a cone transducer is one of the best ways to increase sensitivity in the mid-range and for higher frequencies. Professional horn-loaded components reach mid-range sensitivities of 110dB/2.83V/1m and more with well-defined directional characteristics.

Fig. 20 shows the frequency response of a typical medium-sized horn-loaded lo-mid cabinet. The mid-range SPL is approximately 108dB/2.83V/1m, more than 10dB higher than the SPL of a regular direct radiator 2-way cabinet.

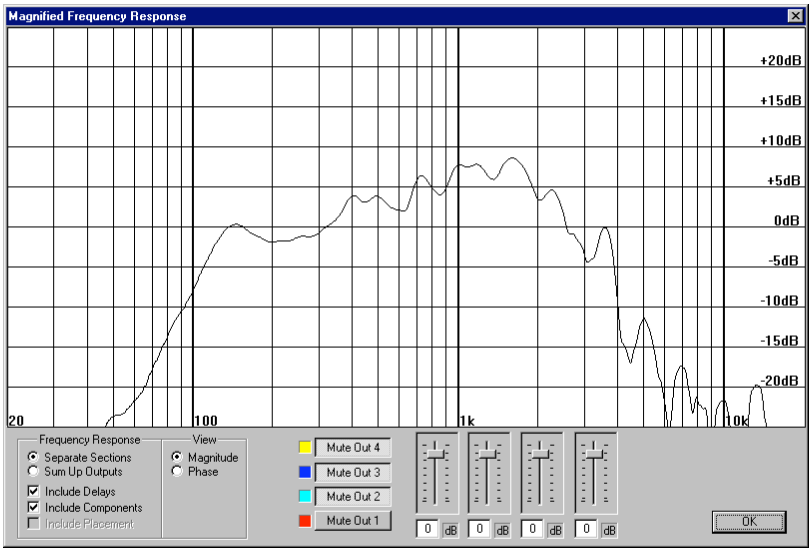


Fig. 20 Medium-sized horn-loaded lo-mid cabinet

The steep roll-offs below 150Hz and above 3kHz are typical for such high efficiency speakers and one has to be careful not to try to milk-out SPL by enormous amounts of equalization outside the gradual roll-off region. The component can be equalized between 150Hz and 2kHz. The region between 1kHz and 2kHz is exceptionally good for this unit and a crossover point between 1.5kHz and 2kHz will give excellent results. The peak at 150Hz can be equalized with a parametric Eq.

RECIPE 5
HIGH EFFICIENCY HORN-LOADED LOW-MID CABINETS NORMALLY CAN BE USED BETWEEN 120HZ - 200HZ AND 800HZ – 2KHZ, THE FREQUENCY EXTREMES DEPENDING ON THE SIZE OF THE CABINET AND THE QUALITY OF ITS COMPONENTS.

3.4 2” and 1” Exit Compression Drivers with Horns

High-power 10”, 12” and 15” cone transducers cannot reproduce frequencies at reasonable SPLs above 2kHz - 3kHz due to physical reasons. The mass of the voice coil and cone assembly is too high and the dimensions of the cone in principle lead to a narrowing of the polar behaviour. For the transmission of frequencies above 1kHz-3kHz one has to use compression drivers loaded with horns. Diaphragm and voice coil assemblies of compression drivers are orders of magnitude lighter than the equivalent cone transducer parts and the use of phasing plugs overcomes the high-frequency beam narrowing of the diaphragm itself.

But compared with cone transducers compression drivers are delicate and fragile units and can easily be driven into full (and expensive) destruction with low-frequencies even at low power levels. Hence crossover points and settings of equalizers and limiters have to be chosen with extreme care.

Typical high-power compression drivers have a voice-coil diameter of 4” and an exit diameter to the horn of 1.3” to 2”. Fig. 21 shows the raw frequency response of such a unit.

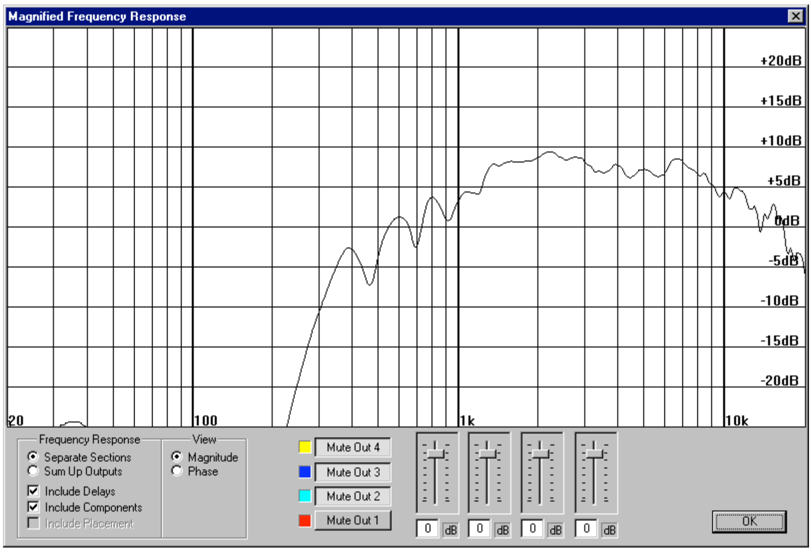


Fig. 21 Frequency response of a 2” driver coupled to a constant directivity horn coaxially mounted in a low-mid horn

Below 1.2kHz the frequency response has a roll-off due to the size of the horn and the resonance frequency of the compression driver. The ripple which can be seen in the response stems from reflections inside of the low-mid horn and cannot be equalized. This is not a big problem because the horn-driver combination can never be used safely at such low frequencies except for low-level systems.

The unit reaches a maximum SPL of 109dB/2.83V/1m at approximately 2kHz, has a gentle roll-off up to 10kHz and a more steep roll-off beyond. This behaviour is typical for any horn-loaded 2” compression driver and can be equalized without too much problems.

RECIPE 6
CROSSOVER POINTS TO HORN-LOADED 2” COMPRESSION DRIVERS SHOULD NEVER BE CHOSEN BELOW 1KHZ. PREFERABLY THE CROSSOVER POINT SHOULD BE CHOSEN AROUND 1.5KHZ-2KHZ IF THE LOW –MID CABINET IS ABLE TO REPRODUCE FREQUENCIES UP TO 1.5KHZ-2KHZ.

1” compression drivers in principle behave as their bigger brothers but often have a significantly better performance in the 10kHz to 15kHz region. But due to their smaller voice coil assembly they cannot take as much power as 2” drivers and should normally only be used as supertweeters or in systems which are not driven at extreme levels.

RECIPE 7:
CROSSOVER POINTS TO HORN-LOADED 1” COMPRESSION DRIVERS SHOULD NEVER BE CHOSEN BELOW 2KHZ. PREFERABLY THE CROSSOVER POINT SHOULD BE AT ABOUT 2.5KHZ IF THE LOW–MID CABINET IS ABLE TO REPRODUCE FREQUENCIES UP TO 2.5KHZ. 1” DRIVERS ARE FINE FOR SUPERTWEETER APPLICATIONS OR IN SMALL TO MEDIUM SIZED FULL-RANGE CABINETS.

4 Equalization of Components

True equalization of a linear system is linearization of the amplitude AND phase response or, speaking in circuit theory language, removing poles and zeroes from the transfer function or shifting poles and zeroes out of the interesting range.

For loudspeaker systems one normally does not know in detail the nature of any response irregularity and perfect equalization is very difficult to achieve. Nevertheless, some rules of thumb exist how to equalize pronounced irregularities in a straightforward way.

4.1 Large Format Bass Cabinets

Fig. 22 shows the raw frequency response of a large-format bass cabinet which has a moderate peak at 66Hz. An isolated peak in the frequency response is always a sign of the existence of some sort of resonance and can be equalized with a parametric eq.

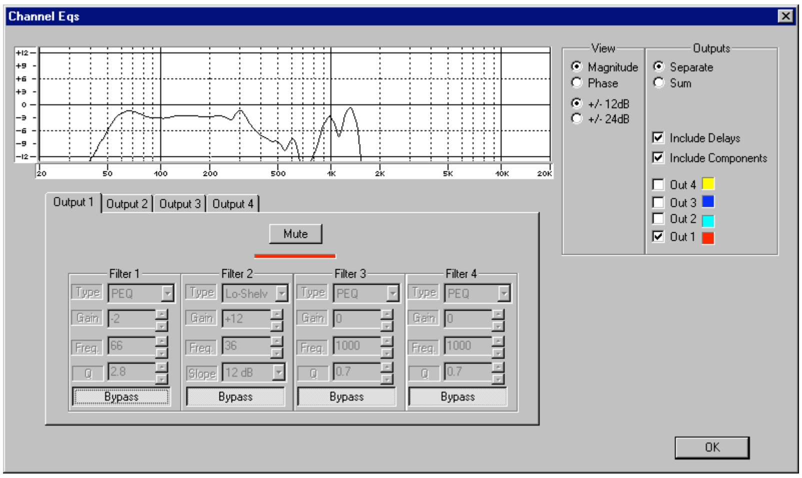


Fig. 22 Raw frequency response of a large-format bass cabinet. Slight peaking at 66Hz

One of the big advantages of the RACE software is, that the frequency response of the speaker component and the frequency response of the eqs can be displayed simultaneously in the graphs, so it is easy to see what influence the adjustment has. The software can control the Dx38 in real-time, hence one has an immediate information on the acoustic result of an adjustment. Fig. 23 shows how the peak can be equalized in our sample cabinet. The upper curve is the frequency response of the PEQ, the lower shows the resulting equalized frequency response of the cabinet where the peak at 66Hz now has been removed. We will not do any equalization for higher frequencies because the region above 200Hz should not be used with large-format bass cabinets and will be cut-off with the crossover low-pass section.

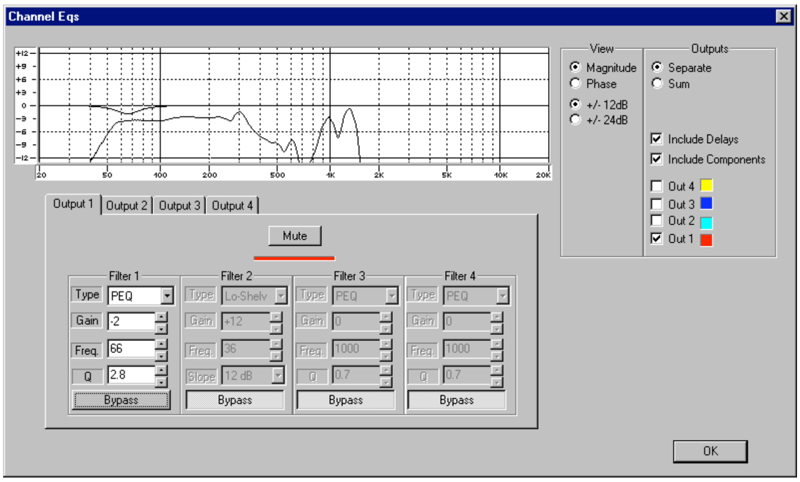


Fig. 23 Equalized response of the sample cabinet. The upper curve shows the eq only

RECIPE 8
PRONOUNCED ISOLATED PEAKS IN THE FREQUENCY RESPONSE CAN BE REMOVED WITH A PARAMETRIC DIP EQ.

4.2 Small Format Bass Cabinets and Full-Range Cabinets

High-efficiency small-format bass and full-range cabinets can often benefit from extending the low frequency response with a moderate amount of equalization. But this makes only sense if the cabinets show a gradual roll-off to low frequencies. Cabinets with an immediate 24dB/octave or locally steeper roll-off like large woofers, some bass horns and many bandpass cabinets should not be boosted at low frequencies because the roll-off in most cases is to rapid and additional equalization only creates hot voice coils, warm air, heavy breathing out of the vents and cones dancing like hell, but no significant SPL.

The low-frequency eq should always be combined with a suitable high-pass filter in order to prevent low-frequency overdrive of the transducer. A 12dB/Oct. shelving Eq combined with a 12dB/Oct. high-pass filter works pretty good in many cases. The shelving eq should be set around 100Hz with a gain of 6dB – 12dB. The high-pass filter with a 12dB/Oct. slope should be set somewhere between 40Hz-60Hz and to a pole Q ranging from 0.7 – 1.

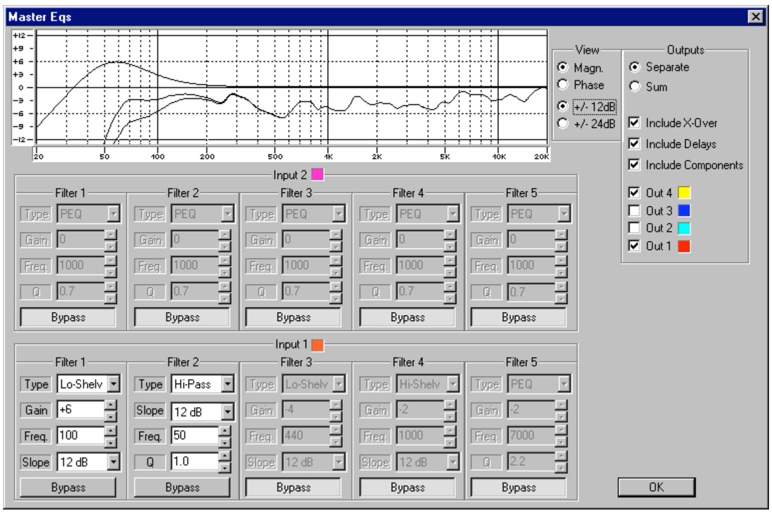


Fig. 24 Low frequency equalization with 12dB/Oct. shelving eq and 12dB/Oct. high-pass filter

Fig. 24 shows such a filter combination applied to a typical high-quality 2-way cabinet. The lower curve is the unequalized response, the upper curve is the shelving eq combined with the high-pass filter. The curve inbetween shows the equalized response. Only a moderate amount of boost is applied to the cabinet and protection against unwanted low-frequency signals has been greatly enhanced.

4.3 Horn-Loaded Lo-Mid Cabinets

The low-mid horn shown in Fig. 25 shows a gradual roll-off from 1.5kHz down to 200Hz, followed by a moderate peak around 150Hz. Above 2kHz there is a steep roll-off. Equalization of this roll-off does not make too much sense because the polar pattern rapidly narrows and not many useful SPL could be milked-out from the component above 2kHz.

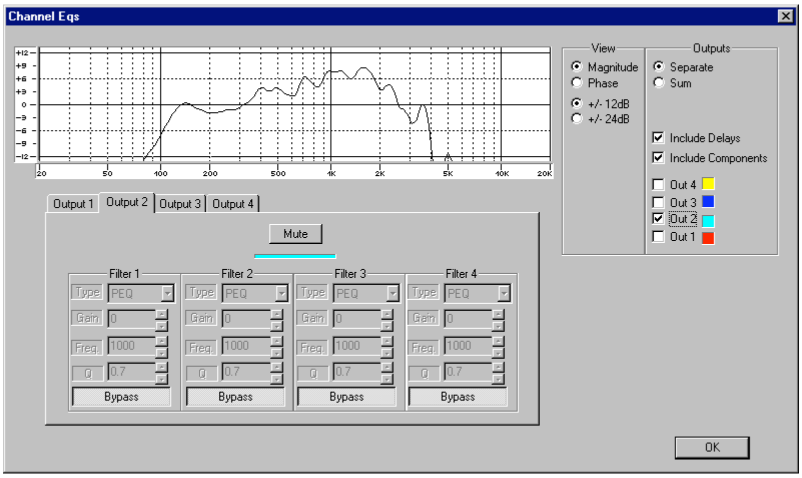


Fig. 25 Raw frequency response of the sample low-mid-horn

Equalizing gradual roll-offs can be done very efficiently with 12dB/Oct. shelving eqs. For the sample horn we use a 12dB/Oct. high-shelving filter and adjust the parameters until we have a satisfactory linearization result. This is shown in Fig. 26.

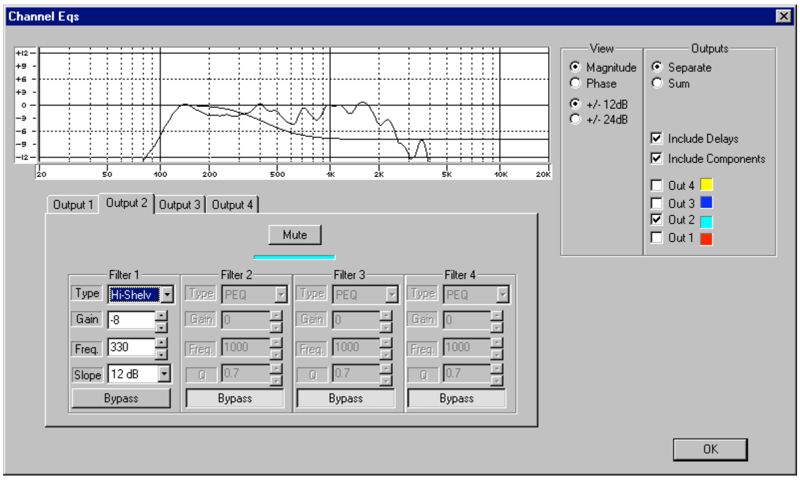


Fig. 26 Filter 1High-shelving eq, 8dB attenuation, 330Hz, 12dB/Oct. type

RECIPE 9
GRADUAL ROLL-OFFS CAN EFFICIENTLY BE EQUALIZED WITH 12DB/OCT. SHELVING EQS.

Now we remove the peak at 150Hz and the peak at 400Hz using parametric dip eqs. This is shown in Fig. 27.

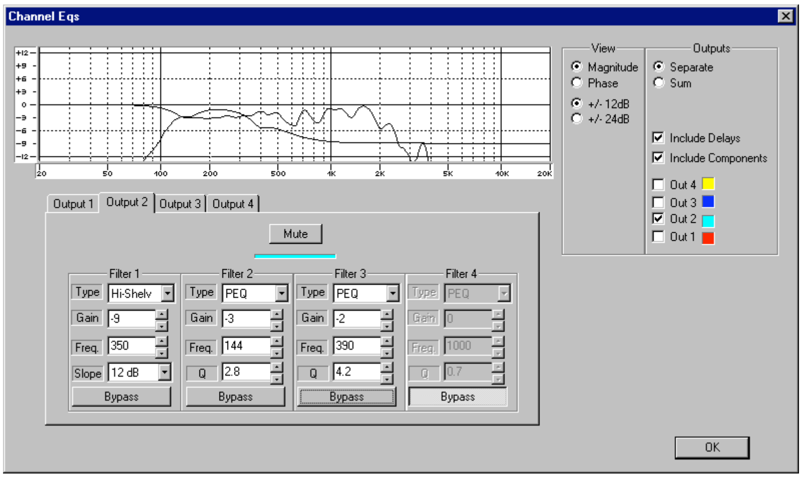


Fig. 27 Filter 2 and Filter 3 have been used to remove pronounced peaks

We still can see some moderate ripple above 500Hz but this looks more or less periodic and no pronounced peaking exists. Normally it is best not to touch such “periodicities” with Eqs because moderate “holes” in the frequency response curve are not very audible and such ripple characteristics in horn speakers often are induced from waves traveling back and forth along the horn creating comb filter effects or some mixture of different mechanisms.

RECIPE 10
LOCAL “HOLES” IN THE FREQUENCY RESPONSE SHOULD NOT BE EQUALIZED.

Fig. 28 shows the result of our equalization efforts. For comparison purposes the unequalized response is shown as the upper curve.

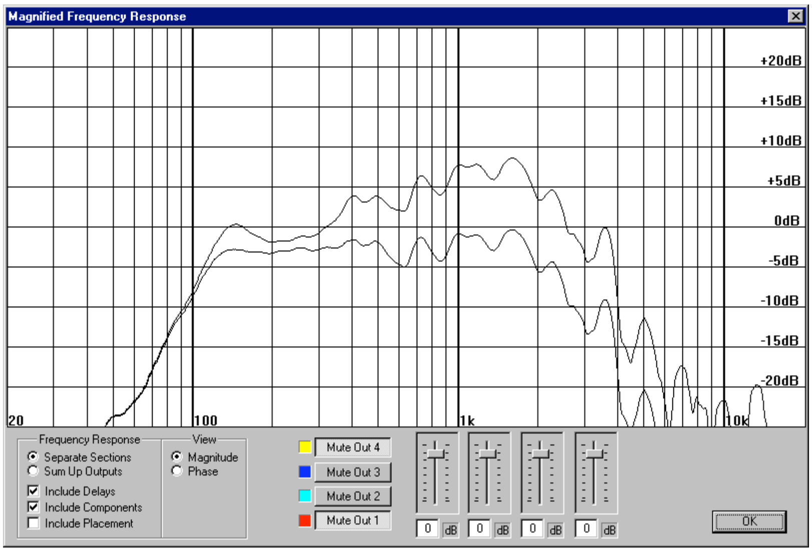


Fig. 28 Equalized and unequalized (upper) response of the sample low-mid horn

4.4 2” and 1” Exit Compression Drivers with Horns

The equalization of compression drivers loaded with horns is very similar to what we have done for the low-mid horn. Fig.29 shows the raw frequency response of a typical high-performance horn-driver combination. The combination has a gradual roll-off between 2kHz and 10kHz and a little bit steeper roll-off above. Below 1kHz we can see reflection-induced periodicities, more precise a comb-filter type response which cannot be equalized but this region of the driver will not be used due to the already mentioned distortion and overload problems.

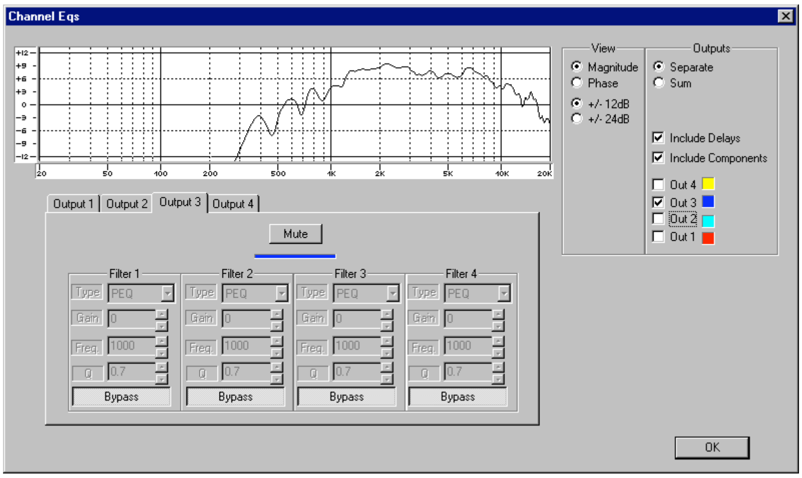


Fig. 29 Raw frequency response of a horn-loaded 2” driver

For this driver-horn combination a simple way for the basic linearization is to use a 12dB/Oct. low-shelving eq where the parameters have been set to 11600Hz and 7dB attenuation (-7dB). As Fig. 30 shows, the linearization is already remarkably good and we just need some additional fine-tuning on the remaining peaks.

The approach using a 12dB/Octave attenuating low-shelving eq works for nearly all compression driver-horn combinations. In cases where the high-frequency roll-off cannot be compensated with this simple method, one can add another shelving or parametric eq in order to correct the roll-off completely. All this sounds quite complicated but playing around a little bit with the RACE software gives an immediate idea on the progress compared with trial-and-error practices or on-site analyzer measurements and equalizer adjustments.

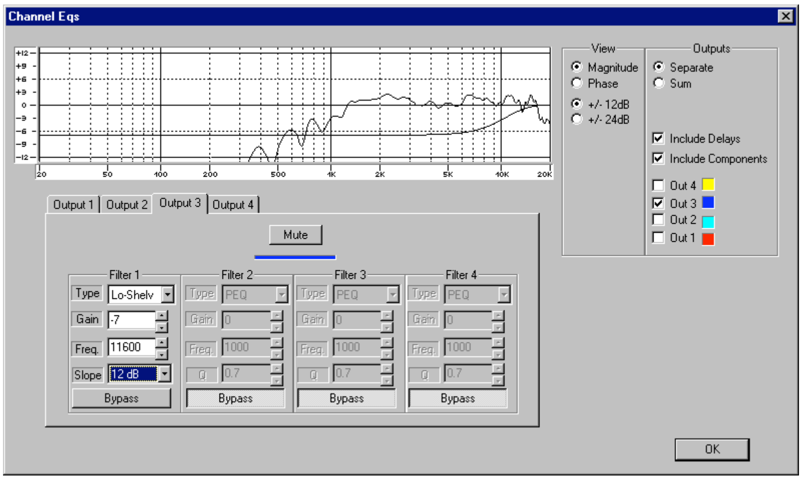


Fig. 30 Driver-Horn combination linearized with a 12dB/Oct. low-shelving eq

The remaining very moderate peaks at 2.2kHz and 7kHz can now be removed with parametric dip eqs. The result can be seen in Fig. 31.

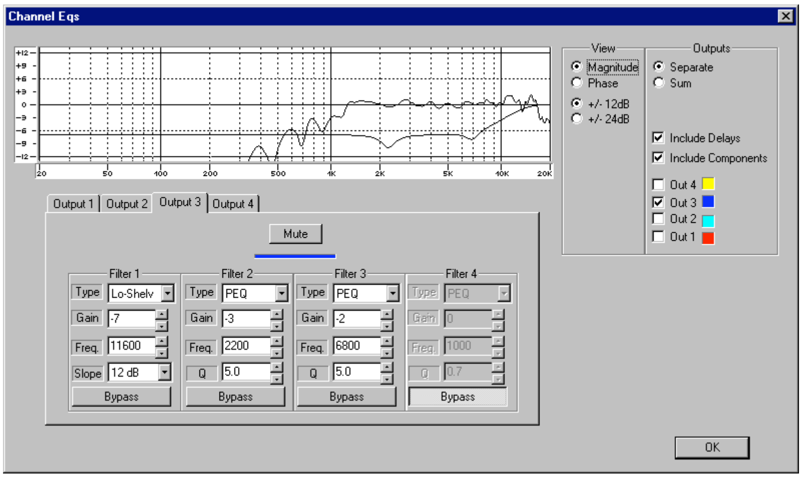


Fig. 31 Upper curve: Equalized driver-horn Lower curve: Eqs only

5 Crossover Functions

A large variety of network functions for realizing crossovers for loudspeakers have been published since the beginning of the century and there is a never ending discussion which crossover is best and why it should be chosen.

Siegfried H. Linkwitz in the early 70s published a paper [2] on the importance of a –6dB crossover point from the aspect of the linearity of the summed response. In addition he showed in his paper that vertical polars of non-coincident drivers depend heavily on the type of crossover function and that it is best to use only crossover functions which maintain an in-phase situation over frequency between the respective outputs. Very simplified this means that the air particles coming out of the respective transducers have to move in full synchronisation in the vicinity of the crossover point.

5.1 Preferred Crossover Transfer Functions

For reasons published in [2], the preferred crossover function is 24dB/Oct. Linkwitz-Riley . In [1] it was shown that for some applications the 12dB/Oct. Butterworth can be an interesting alternative with regard to higher acoustic output around the crossover frequency but this comes at the expense of a peaked response and can lead to a muddy low-mid sound, due to the comparably slow roll-off of the 12dB/Oct. Butterworth filter. The transient response problems of 24dB/Oct. Butterworth filters are clearly audible at lower frequencies, so using 24dB/Oct. Butterworth filters or 48dB/Oct. Linkwitz-Riley filters is not recommended.

RECIPE 11
USE ONLY 24DB/OCT. LINKWITZ-RILEY CROSSOVER FUNCTIONS ESPECIALLY IN THE MID-RANGE.

RECIPE 12
12DB/OCT. BUTTERWORTH CROSSOVER FUNCTIONS CAN BE USED FOR LOW-FREQUENCY CROSSOVERS WHERE A CUSTOMER REQUIRES HIGHER ACOUSTIC OUTPUT IN THE CROSSOVER REGION. THE POLARITY OF THE BASS CHANNEL HAS TO BE INVERTED WITH 12DB/OCT. BUTTERWORTH CROSSOVERS. THE COMPARABLY SLOW ROLL-OFF OF THE 12DB/OCT. CROSSOVER CAN LEAD TO A MUDDY SOUND IN THE LOW-MID REGION.

5.2 "Equal Acoustical Plane" Adjustments

Fig. 32 shows a simplified sketch of a typical 2-way system. The low frequency and high-frequency transducers do not radiate out of the same plane and are non-coincident. The different distances to the aiming point create different arrival times for the high and low frequencies and shows up as a comb filter in the frequency response. The sketch shows a difference of approximately 3.5 periods at the sample frequency which is equivalent to a total phase shift of 3.5*360° = 1260°. At the aiming point we would have full cancellation for this frequency. A more detailed description of the problems involved can be found in [1] and [2].

The inherent phase shift of the transducers and the phase shifting of filters makes the situation even more complicated.

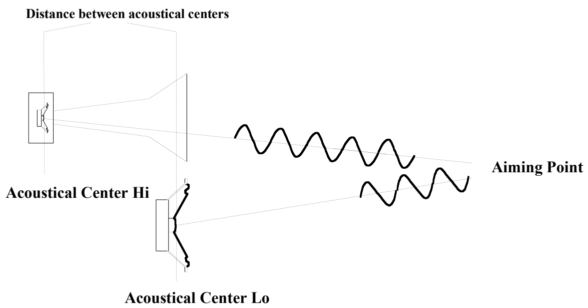


Fig. 32 Simplified sketch of a 2-Way System

But RACE can display the phase response of transducers, filters, crossover networks, power amplifiers and distance and we have an extremely powerful tool at hand for solving equal acoustical plane problems easily.

From [2] the following conclusions for equal acoustical plane adjustments and level settings can be drawn:

1. The acoustic output of the transducers should be In-Phase at the x-over frequency in order to prevent tilted vertical polar patterns.
2. The phase difference between Hi and Lo should be constant for all frequencies so that the symmetry of the radiation pattern is preserved above and below the x-over frequency.
3. For best steady-state linearity of the frequency response the acoustic output of the Hi and Lo cabinet should be 6dB down at the x-over frequency in order to avoid any peaking.

Without going into more details, only Linkwitz-Riley and even order Butterworth crossovers fulfill criterion 1 and 2.

Only Linkwitz-Riley crossovers fulfill criterion 1, 2 and 3.

5.2.1 Tutorial: Different Representations of the Phase Response of a Linear System

The phase response of linear systems can be displayed in different ways. Below 3 different graphs are shown which are fully equivalent and show the total phase shift of a delay filter with 333us delay time which is a travel distance in air of approximately 11.5cm.

In Fig. 33 the vertical axis is linearly marked in degrees from 0° to –1000°, the horizontal axis has linear frequency spacing from 20Hz to 20kHz. The phase response of our delay filter is displayed as a straight line. This picture shows where the term "Linear Phase", which means frequency independent delay comes from.

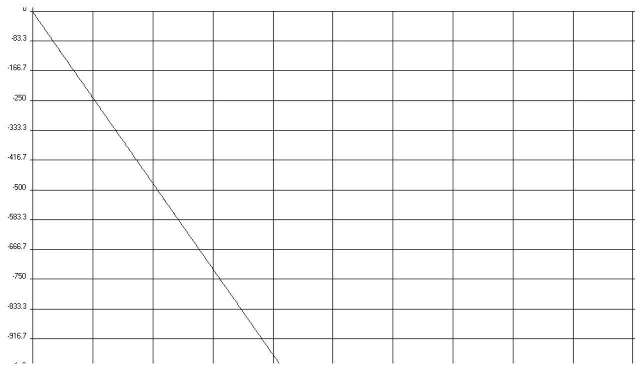


Fig. 33 Phase of a filter with 333us delay time on a linear-linear frequency response paper

Fig. 33 shows the phase from 0° to –1000°. If we want to expand the graph to -10.000° the y-axis of the graph would be 10 times longer and could not be printed on a sheet of paper with the same resolution.

In order to have a more manageable form one "wraps around" the phase curve after having reached –180° degrees and starts again from the top of the graph until at crossing the 0° one full period has been completed. Fig. 34 shows the "wrapped-around" phase display, the y-axis now marked +180° to –180°, the horizontal axis is still linear.

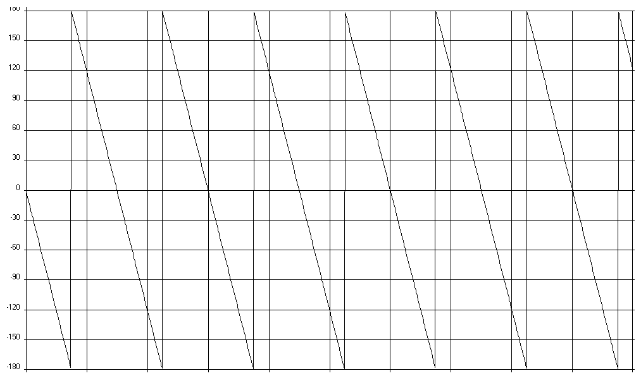


Fig. 34 Phase of a filter with 333us delay time "wrapped around"

Fig. 34 shows exactly the same as Fig. 33 but has the significant advantage that one can now display large amounts of phase shift without requiring ultra-long paper. Longer delays would show up here as a narrower spacing of the "saw-tooth", shorter delays as wider spacing of the saw tooth. The vertical lines of the saw-tooth do not represent "phase-jumps" but are the connection between the identical +180° and –180° points on a circle.

A linear frequency axis is not very comfortable for the representation of the frequency response of audio systems because the audio range is approximately 8 octaves wide and a logarithmic spacing on the frequency axis mimics to some extent the way our auditory system is working. In Fig. 35 the frequency axis has been rescaled to the commonly used 20Hz – 20kHz logarithmic scale where we now display the phase curve of the 333uS delay filter. The phase curve now looks "non-linear" but Fig. 35 shows exactly the same phase as Fig. 34 and Fig. 33, respectively. This is the representation of phase which is most commonly used but it is quite difficult to see whether we have pure delay or a minimum phase response like typical eq filters or both combined.

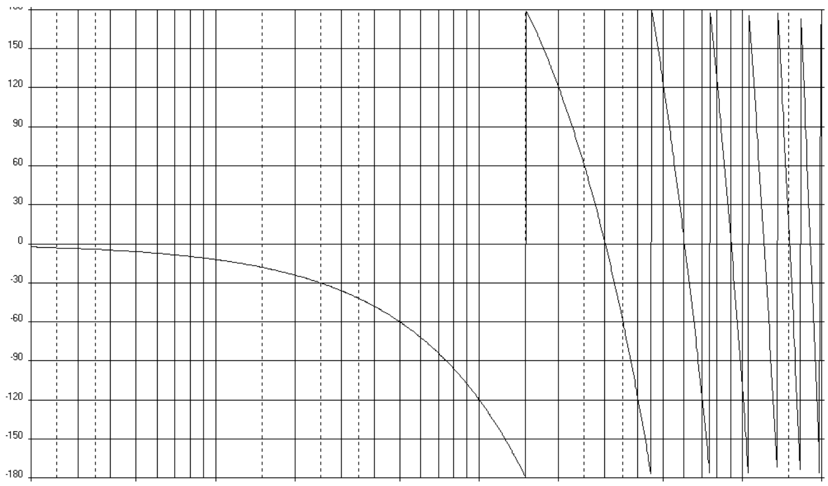


Fig. 35 Phase of a filter with 333us delay time on a “regular” frequency response paper

5.2.2 Adjustment for a matched Phase Response

Fig. 36 shows the frequency response of already equalized lo-mid and high-frequency horns. A 1600Hz 24dB/Oct. Linkwitz-Riley crossover has been applied.

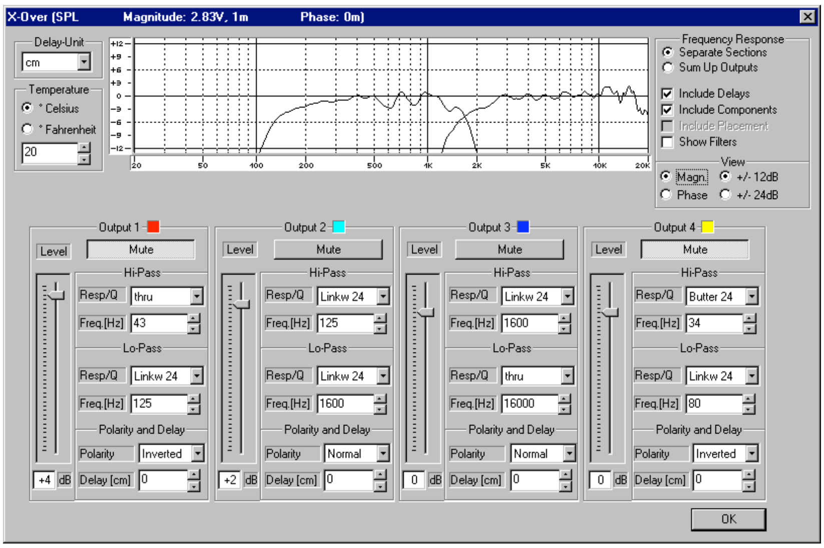


Fig. 36 Lo-mid and high-frequency horn with 1600Hz 24dB/Oct. Linkwitz-Riley crossover

The compression driver is coaxially mounted inside the lo-mid horn, hence the acoustical centers which are approximately located at the dome of the cone transducer and the diaphragm of the compression driver are not in the same vertical plane. We now click to the phase display and can see the different phase responses of both components in Fig. 37. Around 1600Hz the phase responses are not matched and after a while one can see that the slope of the low-mid horn´s phase is steeper what can be expected because the 12” cone transducer is located behind the compression driver.
The slope of the phase curves is steeper than we normally would expect, but RACE displays the phase at the grille position of cabinets and the drivers here are nearly 50cm behind the grille. It is important to note, that far outside of the transmission range of components the displayed phase becomes invalid because of a low signal to

noise ratio during measurement. But this does not play any significant role, because the SPL levels of components far outside their transmission range is of no relevance for our application.

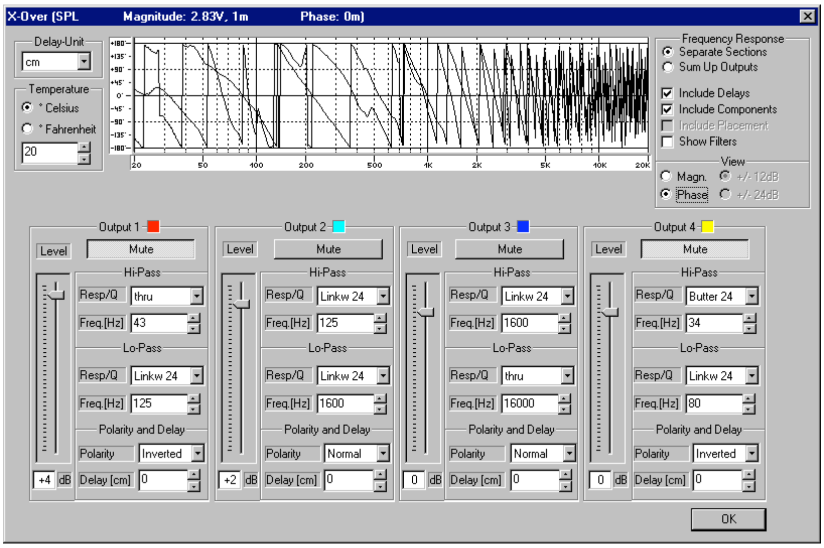


Fig. 37 Phase response of components

We now add 35cm delay to the compression driver and now the phase response of both units is perfectly matched in the vicinity of the 1600Hz crossover point. Playing around a little bit with the alignment delay shows that this is not as complicated as it looks like. Fig. 38 shows the matched phase response between 1kHz and 2kHz.

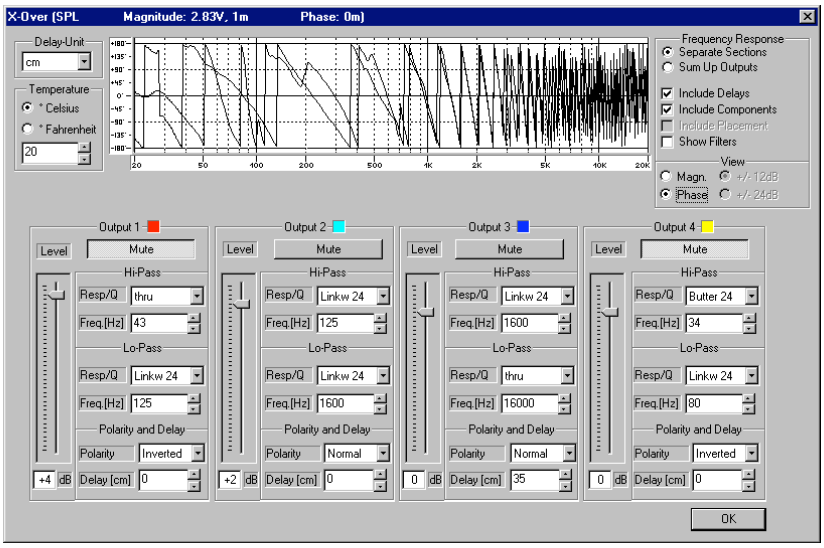


Fig. 38 Matched phase response around 1600Hz, 35cm alignment delay

5.2.3 Level Check for a Correct Alignment

In [2] a simple method has been published how to check whether the aligning was done correctly. One has to look at the summed level response and invert the polarity of one channel. If a deep notch occurs the alignment has been done properly. This level check with one channel inverted is shown in Fig. 39 and confirms that our alignment has been done properly.

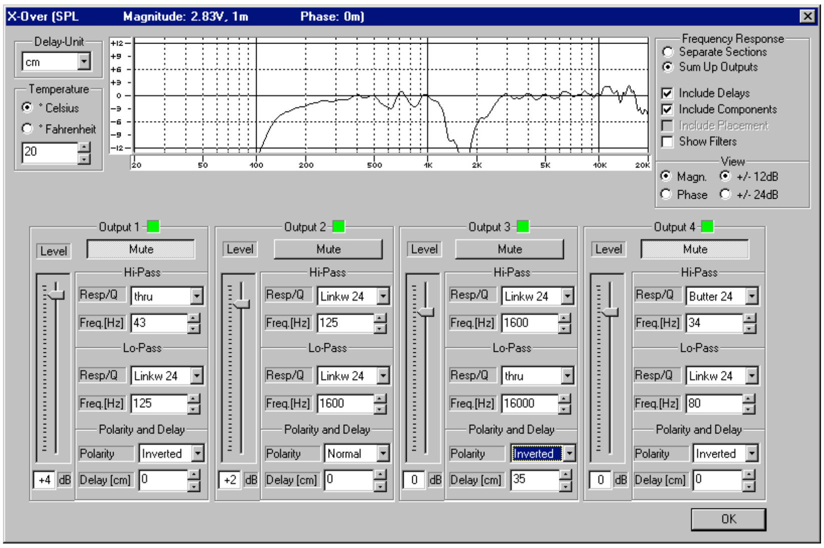


Fig. 39 Alignment check with inverted polarity of one channel

One has to be a little bit cautious because a notch occurs again if one has added accidentally “one or more wavelengths” during the aligning process. In our example here if we set the alignment delay to 55cm we again would see a notch in the summed response when inverting one channel.

But clicking back to the “View Phase” and “Separate Sections” immediately uncovers that something must be wrong. We still have equal “relative” phase at the crossover point but in the vicinity the phase curve of the compression driver now has a much steeper slope and we are approximately 22cm distant from the correct position.

Fig. 40 shows this misalignment.

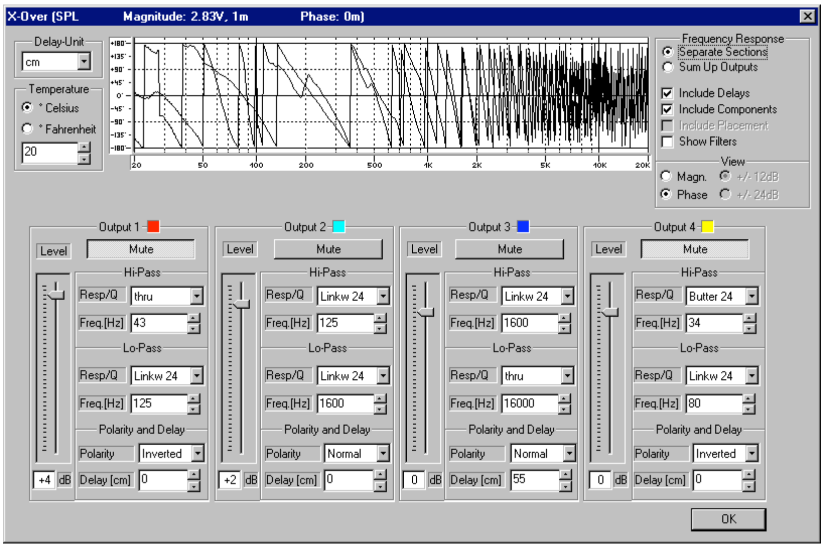


Fig. 40 Compression driver misaligned, but “In-Phase” at 1600Hz

6 Compressor and Limiter Settings

The Dx38 is equipped with a compressor and a limiter in every output channel. Limiters are primarily used for speaker protection purposes whereas compressors preferably are used more for tailoring the dynamic behaviour of the system long before reaching clip levels which helps to protect speakers too, but an overload situation still can occur during the attack time of the compressor.

6.1 Level settings

The Dx38 is a gain calibrated unit and the dB markings around the controls show the exact level change. Setting input and output levels to the “0” position means that the Dx38`s gain is “0dB”, hence 1V at the input will generate 1V at the output, presupposed that there is no internal filter or level control enabled.

The compressor and limiter thresholds refer to the internal digital transmission chain and are calibrated against the Output level in “0” position.

Example 1: We set the rotary output level control fully clockwise , to the “0” position and the limiter threshold to 1.55V. The maximum output voltage coming out of the unit will be 1.55V.

Example 2: We set the rotary output level control to the “-6” and the threshold to 1.55V again. Now the internal voltage is attenuated at the output by 6dB and the maximum voltage coming out of the unit will be 0.775V.

Example 3: We have a power amp with the manufacturer´s specification “Input sensitivity = 0dBu for rated output power”. We set the output level control fully clockwise and the threshold to “0dBu” and the power amp will never be overdriven because a maximum of 0dBu (=0.775V) comes out of the respective output of the Dx38.

Example 4: We have a power amp with the manufacturer´s specification “Input sensitivity 2.6V for rated output power”. We set the output level control fully clockwise and the threshold to “2.45V” and the power amp will always remain slightly below clipping because a maximum of 2.45V comes out of the respective output of the Dx38.

A warning should be raised here: One can choose the threshold level in Volts (what is easy to understand), in dBu (what is not so easy to understand) and in dB (what nobody understands because it means “dB distance to internal clipping” or dB distance to approximately 8.7V eff or whatever). So stay with the Volts or dBu but ignore the dB setting in the limiter and compressor menu.

6.2 Limiter Release Time

The default limiter time constant of 100ms is optimized for speaker protection and general-purpose applications. If the limiter should be used for limiting the systems output at lower levels the release time should be chosen approximately 500ms to 1s in order to prevent “pumping”. Care should be taken when microphones are involved because the slow level changes can lead to unexpected feedback situations. Turning up the volume control of the mic does not change the volume because of the limiter’s action but after a second the limiter release time is over, the gain goes up, the feedback starts again, we turn down the volume, not loud enough, turn up again and so on and so on.

6.3 Compressor Attack, Release Time and Compression Ratio

Compressor attack and release times have to be adjusted by ear or with a scope at hand. No general advice can be given because of the complex psychoacoustic interaction of all the parameters involved.

Ratio 1/1 means that there is no action of the compressor, 1/8 means that an 8dB level change (above threshold) will be reduced to a 1dB level change at the output, 1/2 and 1/4 are inbetween the extremes.

1/8 compression ratio is quite similar to the behaviour of a limiter but the compressor overshoots at the beginning of the signal which often sounds nice from a subjective point of view but contains the risk of short-term overload of the speaker components.

6.4 Combination of Compressors and Limiters

A combination of compressor and limiter can lead to problems with some signals because the limiter immediately catches the compressor overshoot and we have two coupled gain reduction devices involved. Nevertheless, in systems which are used primarily for speech transmission, combined compressors and limiters are a nice tool for tailoring the dynamic behaviour of the system.

7 List of Recipes

RECIPE 1
LARGE-FORMAT BASS CABINETS NEED ONLY A MODERATE AMOUNT OF LOW-FREQUENCY EQUALIZATION.

RECIPE 2
CROSSOVER FREQUENCIES TO LOW-MID CABINETS SHOULD BE BETWEEN 80HZ AND 150HZ.

RECIPE 3
SMALL FORMAT BASS CABINETS WITH A GRADUAL ROLL-OFF CAN BE EQUALIZED FOR AN EXTENDED BASS RANGE.

RECIPE 4
CROSSOVER FREQUENCIES BETWEEN NORMAL FULL-RANGE CABINETS AND SUBS SHOULD BE CHOSEN BETWEEN 80HZ AND 120HZ.

RECIPE 5
HIGH EFFICIENCY HORN-LOADED LOW-MID CABINETS NORMALLY CAN BE USED BETWEEN 120HZ - 200HZ AND 800HZ – 2KHZ, THE FREQUENCY EXTREMES DEPENDING ON THE SIZE OF THE CABINET AND THE QUALITY OF ITS COMPONENTS.

RECIPE 6
CROSSOVER POINTS TO HORN-LOADED 2" COMPRESSION DRIVERS SHOULD NEVER BE CHOSEN BELOW 1KHZ. PREFERABLY THE CROSSOVER POINT SHOULD BE CHOSEN AROUND 1.5KHZ-2KHZ IF THE LOW –MID CABINET IS ABLE TO REPRODUCE FREQUENCIES UP TO 1.5KHZ-2KHZ.

RECIPE 7:
CROSSOVER POINTS TO HORN-LOADED 1" COMPRESSION DRIVERS SHOULD NEVER BE CHOSEN BELOW 2KHZ. PREFERABLY THE CROSSOVER POINT SHOULD BE AT ABOUT 2.5KHZ IF THE LOW –MID CABINET IS ABLE TO REPRODUCE FREQUENCIES UP TO 2.5KHZ. 1" DRIVERS ARE FINE FOR SUPERTWEETER APPLICATIONS OR IN SMALL TO MEDIUM SIZED FULL-RANGE CABINETS.

RECIPE 8
PRONOUNCED ISOLATED PEAKS IN THE FREQUENCY RESPONSE CAN BE REMOVED WITH A PARAMETRIC DIP EQ.

RECIPE 9
GRADUAL ROLL-OFFS CAN EFFICIENTLY BE EQUALIZED WITH 12DB/OCT. SHELVING EQS.

RECIPE 10
LOCAL "HOLES" IN THE FREQUENCY RESPONSE SHOULD NOT BE EQUALIZED.

RECIPE 11
USE ONLY 24DB/OCT. LINKWITZ-RILEY CROSSOVER FUNCTIONS, ESPECIALLY IN THE MID-RANGE.

RECIPE 12
12DB/OCT. BUTTERWORTH CROSSOVER FUNCTIONS CAN BE USED FOR LOW-FREQUENCY CROSSOVERS WHERE A CUSTOMER REQUIRES HIGHER ACOUSTIC OUTPUT IN THE CROSSOVER REGION. THE POLARITY OF THE BASS CHANNEL HAS TO BE INVERTED WITH 12DB/OCT. BUTTERWORTH CROSSOVERS. THE COMPARABLY SLOW ROLL-OFF OF THE 12DB/OCT. CROSSOVER CAN LEAD TO A MUDDY SOUND IN THE LOW-MID REGION.

8 Appendix (Not yet existing)

8.1 Representation of Frequency and Phase Response of Speaker Components inside RACE

8.2 Representation of Frequency and Phase Response of Amplifiers inside RACE

9 Bibliography

[1] Krauss, Guenter J., "Design of Active Crossover Transfer Functions for Digital Sound System Processors", 21. Tonmeistertagung, 24.11. – 27.11.2000, Hannover, Germany

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