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TOWARDS A DIGITAL SPEAKER MONITORING STANDARD

Alex Garner

Tannoy Ltd, Rosehall Industrial Estate, Coatbridge, Scotland, ML5 4TF.

1. INTRODUCTION

Recording studios, Broadcast studios and associated quality monitoring services increasingly have a substantial digital signal content. Given that digital signals when processed properly are of inherently high quality, it is essential that the quality monitoring speakers (albeit essentially analogue in fundamental energy conversion) have specifications at least equal to if not better than the digital signal itself.

Currently, popular speaker monitoring systems are equalised in the analogue domain, either within the internal passive crossover, or by electronic filter design in an external active crossover system. This is because the drive units themselves, making up the speaker system, have non ideal characteristics. Also the reaction of drive units with both surroundings and cabinet design requires careful consideration and must be taken into account in the primary equalisation of the speaker system when used as a true monitor.

Digital signal processing can provide an extra degree of freedom to the monitor speaker designer, allowing equalisation responses and characteristics which cannot be realised with analogue active or passive crossover circuitry. In particular the ability to correct amplitude response independently of phase response becomes available using FIR filters, assuming that a small overall delay can be tolerated between the signal input to the monitoring system and the acoustic output.

However in setting a digital standard for broadcast and studio quality monitoring speakers it is essential to take account of those parameters which can successfully be corrected by DSP and those which cannot. DSP is not a substitute for poor drive unit design, nor is it a panacea for all ills. Good fundamental acoustic transducer design is required with careful addition of the extra degree of freedom available with DSP techniques.

2. DIFFICULTIES AND COMPROMISES - A BRIEF OVERVIEW

Figure 1 shows some of the basic difficulties facing the speaker designer. The bandwidth of human hearing, speech and music sound propagation is very wide. At low frequencies large volumes of air have to be moved with high velocities to create realistic sound pressure levels. At high frequencies air volumes are less but velocities are much higher. Using two or more drive units to reduce the driver design compromises leads to further complications in joining the drive unit acoustic radiation together at the crossover frequencies. To find out how DSP can be helpful in solving these difficulties we have to go back over some fundamental concepts in speaker design.

3. CABINET FUNDAMENTALS

To prevent the front and backside radiation from a moving piston (speaker cone) cancelling out at low frequencies we need either an infinite baffle (impractical) or a closed box around the unit. Apart from supporting the driver, this is the only reason for having a cabinet. Immediately, 50% of the radiation from the driver is trapped inside the cabinet is not usefully employed and must be absorbed. Forming a mechanical cabinet structure around the driver leads to the possibility of resonances occurring, excited by the acoustic energy trapped inside. The cabinet around the driver traps air inside which acts like a spring on the moving piston. The strength of the spring is proportional to the volume of trapped air and the surface area of the piston. The resultant air spring compliance reacts with the piston mass to produce a fundamental resonance. The piston radiates acoustic energy freely above this resonance frequency but reduces sharply below it, becoming increasingly inefficient. The range, quantity and quality of low frequencies is determined by the frequency of this resonance and the amount of air volume velocity capability in the piston. Large amounts of low frequency energy require large pistons. To keep the fundamental resonance in the cabinet low, so that low frequencies can be reproduced, the compliance of the air in the cabinet must be high, necessitating a large cabinet. Smaller pistons can be used but they must be capable of sweeping large volumes of air and therefore have high linear excursion.

The piston and cabinet can be analysed as an electrical analogue yielding a band pass system. This has been well documented in the literature over the last 15 years. A simplified view of the amplitude response of the bandpass system is shown in Figure 2 for reference. Electronic external processing can be helpful in overcoming the inherent compromise of cabinet/drive unit limitations by boosting signals below fundamental resonance and controlling or modifying the Q value of the resonance. However, the linear excursion limitations of the drive unit are fundamental to high quality low frequency reproduction and should not be exceeded. The electronics must therefore have sharp high pass filtering at frequencies preceding the boost to ensure minimum drive unit generated distortion or even mechanical damage. Electronic feedback systems can be helpful in extending responses but the fundamental ability of the drive unit to produce air volume velocity is paramount in a successful design. DSP can be helpful in calculating the amount of boost that can be given without causing the drive unit to over excurt. DSP is also capable of creating very sharp cut off slopes, although there are fundamental difficulties in creating sharp low frequency slopes in the region required. DSP cannot contribute to the fundamental problem of generating low frequency energy from a direct radiating piston without the use of a device to prevent front and rear radiations cancelling.

4. DIFFRACTION AND REFLECTION FUNDAMENTALS

Diffraction in speakers occurs around corners and edges of cabinets and drive unit mechanical mounting details. Even if the cabinet did not exist there would still be diffraction and reflection around the piston edges and supporting chassis. Smooth contours are required to reduce these effects to a minimum. Figure 3 shows for reference the work done by Olson in quantifying the effects of diffraction and reflection around cabinet shapes. There is very little that external DSP routines can do to correct such real physical problems. "Pre-echo" can be generated and phased to minimise a major troublesome reflection caused by a large architectural problem in the studio space but the myriad of small reflections and diffractions around a speaker cabinet cannot be controlled by DSP. Diffractions and reflections interfere with the main driver radiations especially in the near field and it is therefore essential to take the physical design of the speaker cabinet and driver mounting into account in monitor speakers.

5. DISPERSION FUNDAMENTALS

With a given piston diameter the acoustic radiation narrows from omnidirectional dispersion as the reproduced frequency increases. The relationship between dispersion and reproduced frequency is demonstrated in figure 4. As long as the reproduced wavelength is larger than the piston circumference the radiation will be substantially omnidirectional. For radiations with wavelengths increasingly less than the piston circumference the dispersion becomes increasingly narrow. Control over dispersion depends on the formation of acoustic wavefronts, spherical wavefronts having wide dispersion. Wavefront control, either by selection of size for direct radiating pistons, or horn mouth shape for horn loaded systems, is fundamental in achieving dispersion patterns which can be beneficial to the final use of a speaker system. For Studio and Broadcast monitoring systems it is important to have dispersion which does not change suddenly through the spectrum nor has widely differing characteristics in the horizontal or vertical plane.

Although control rooms are usually acoustically absorptive environments and therefore most of the energy from a monitor speaker is received directly, the presence of the console and other devices can create unwanted reflections. It is important therefore for the speaker to have relatively constant directivity and symmetrical dispersion. In this way there will not be any surprises when moving about the console and for near field use the speaker may be mounted vertically or horizontally to meet sight lines etc. If off axis energy is smooth and symmetrical then the reverberant field will be consistent and without distracting unnatural acoustic colourations.

Unfortunately DSP cannot help to control dispersion in single speaker applications and therefore the design of the drive unit complement in the monitor system is all important.

6. AMPLITUDE FUNDAMENTALS

There can be little argument that the amplitude response of a monitor speaker should be smooth, that is, free from excessive undulations in the amplitude response. Whether the response should be 'flat', that is, parallel to the frequency axis in a conventional dB/Hz plot is more a function of the dispersion or overall directivity of the system. It is important that the sound power into the room is smooth and even. In some speakers the changes in response from a 'flat' (horizontal) amplitude response reflect the necessity to boost radiation in some areas to maintain sound power. In some instances it is beneficial to shape the monitor response by say 0.15 to 0.2 dB/octave over the full bandwidth to create a constant sound power radiation into the room over the dispersion envelope.

There is no doubt that DSP can bring enormous advantages in amplitude response control to monitor speakers. Small changes in overall slope (0.1 - 0.3 dB/octave) can be achieved to compensate for directivity if necessary. The overall response undulations can be smoothed further resulting in an amplitude response which is more characteristic of a piece of electronic equipment rather than a speaker.

Figures 5 and 6 show the typical amplitude response of the woofer and tweeter sections of the Tannoy System 8 NFM II near field monitor measured under anechoic conditions. Figures 7 and 8 show the amplitude correction and crossover filtering using DSP techniques. Figure 9 shows the combined response of the System 8 NFM II with a full DSP crossover and amplitude equalisation implementation.

7. PHASE FUNDAMENTALS

Well designed speaker drive units behave as maximally flat band pass systems over a substantial portion of their designed response. This means that improvements to the amplitude response benefit the phase response and vice versa. Outside the pass band, in the 'break up' areas the phase response becomes meaningless as many areas of the piston surface are radiating simultaneously and often in antiphase. Nevertheless when restricted to their intended ranges the phase responses of well designed drive units are reasonable. This shows that the drive units are capable of reproducing sound without introducing widely different time delays in different portions of their bandwidth.

The concept of acoustic phase is bound up with inherent time delay differences as the sound energy propagates through air. The concept of flat phase response in monitor speakers can be a little misleading and the portrayal of measured phase responses in the literature can also be misleading as logarithmic or linear frequency scales are used. It is more important that the speaker can be shown to have a constant time delay associated with it over a substantial portion of the passband. That is, all frequencies within the pass band are delayed equally by a fixed amount - constant group delay. It follows that the rate of change of phase with frequency must therefore be a constant. Phase response measurements showing responses inclined at an angle to the horizontal axis are perfectly valid (vs linear frequency) as long as they are straight lines. On a logarithmic frequency axis a phase response parallel to the horizontal axis shows frequency invariant time delay characteristics - constant time delay.

Again DSP is an excellent tool for linearising time delay or group delay. There are time delays inherent in the DSP amplitude response equalisation which can be further manipulated to arrive at a flat phase, constant time delay solution.

Figures 5 and 6 show the typical phase responses of woofer and tweeter used in the monitor described in section 6 above and figures 7 and 8 the DSP phase equalised versions. The combined amplitude and phase response of the whole DSP system is shown in figure 10 using a linear frequency scale. Figure 9 shows the amplitude response of the DSP system on a logarithmic frequency scale.

8. CROSSOVER FUNDAMENTALS

The acoustic output from individual drive units must add up in amplitude and phase throughout the crossover transition frequency band. The physical displacement of drive units on the front face of the cabinet poses some fundamental problems which cannot all be solved by DSP techniques. Low, mid and high frequency drive units have widely differing acoustic virtual sources. Adjusting time delays between units can align acoustic sources to a plane but not to a single point for all positions in the listening environment. DSP is of limited value therefore with spaced drive units. It is of much greater value where the drive unit has co- axial sources. DSP techniques can align a coaxial or dual concentric drive unit construction to a single point source where flat amplitude and constant time delay are valid for all listening positions within the constant directivity envelope.

9. THE DIGITAL MONITOR CONCEPT

In summarising the areas where DSP can be useful in monitoring applications the fundamental design of the drive units and overall system must play a leading role. Only by optimising both the transducer and the DSP technique will we approach a standard.
A professional monitor system is a commercial tool, used by organisations in making gainful profits. Not only must it have optimum qualities in traditional measured parameters such as bandwidth, distortion, amplitude, phase, dispersion, diffraction; but it must be reliable, robust, easy to service and affordable by most studios, not just the elite. It must also be capable of building a reputation for producing successful commercial productions and be highly analytical in portraying fault conditions.

There are two practicalities at present in defining a digital monitoring standard:

> Setting a monitoring standard for studios and broadcast who have fully digital signal flow and storage, operating wholly in the digital domain right up to the monitor power amplifier, by providing monitoring with a near perfect amplitude and phase response over a given symmetrical dispersion angle.

> Setting a monitoring standard for all studios whether digital or analogue by providing monitoring with near perfect amplitude and phase response over a given symmetrical dispersion angle.

The first defines a system where the console has a digital signal monitoring output. The signal therefore only has one stage of conversion before the speaker. Digital console monitor output is fed directly into the monitor DSP equalisation system and only converted by DACs immediately before the power amplifiers. In this way the signal receives minimum conversions and therefore monitoring is closer to the production. Control of monitor volume must be done at the console in the digital domain and therefore the volume coding within the data stream must be standardised so that the monitor system will work with all digital consoles.

The second defines a system for organisations where the production is predominantly analogue and the console has conventional variable analogue monitor output. Clearly a stage of ADC is needed before the DSP. It is important that the resolution of the ADC and DAC in the monitor speaker are equal to or exceed the studio or broadcast production system. Control of monitor volume becomes a fundamental issue here. Monitor volume controlled from the console in the conventional way will reduce the resolution in the ADC stages before the DSP. To maintain resolution the best solution is to take monitor line level from the console and control volume after the DACs following the DSP using a high quality VCA controlled from the console control surface..

The basic target specification required of DSP monitor speaker systems and power amplifiers might be summarised as follows:

Sensitivity:	92-103dB for 1 watt at 1 metre
Power capacity:	100 - 400 watts to IEC spec
Power amplifier:	100 - 500 watts into 8 ohms
Acoustic Source	Coincident or Coaxial
Dispersion	Constant over a 90 degree angle over a given bandwidth.

Good power response (no compression)
Low colouration and distortion
Good diffraction cabinet performance

Note that DSP techniques cannot correct for any of the above.

Amplitude response:	$\pm$0.2 dB over quoted bandwidth
Phase response:	$\pm$1 degree over quoted bandwidth

Note that DSP techniques are inherent in the last two targets.

10. CONCLUSIONS

Fundamental monitor speaker design parameters have been discussed and those found suitable for improvement by DSP techniques highlighted. In producing a DSP monitor the basic design philosophy of drive unit and system are all important for the final DSP equalisation to be successful. The issue of whether to have an all digital system without input ADC or whether to incorporate high resolution ADC conversion for those organisations using consoles with analogue monitor outputs only, (the majority?), will probably be resolved by demand rather than philosophy.

Finally, for DSP to be fully effective over the whole studio monitoring area it must be used in conjunction with a coaxial or dual concentric drive unit design. If conventional spaced multi unit systems are used the full benefit of near perfect phase or frequency invariant time delay over the constant directivity envelope cannot be realised. For all areas of a control room to hear a true linearised DSP standard the basic monitor system before equalisation must be a point source speaker.

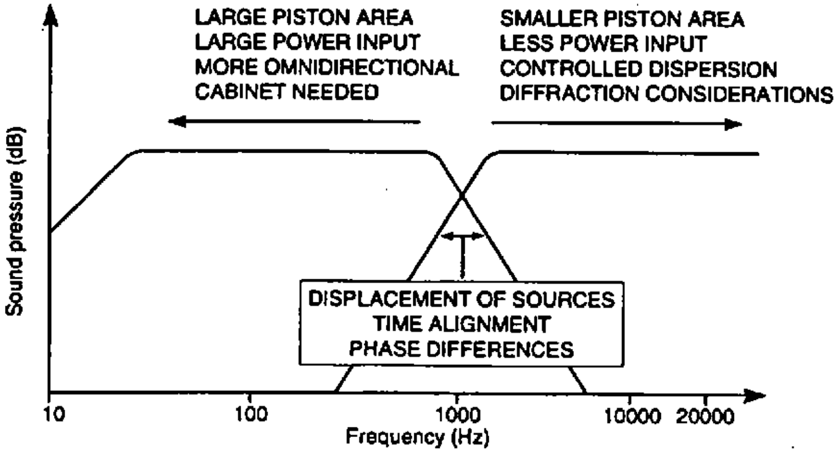


Figure 1: Basic difficulties facing the speaker designer.

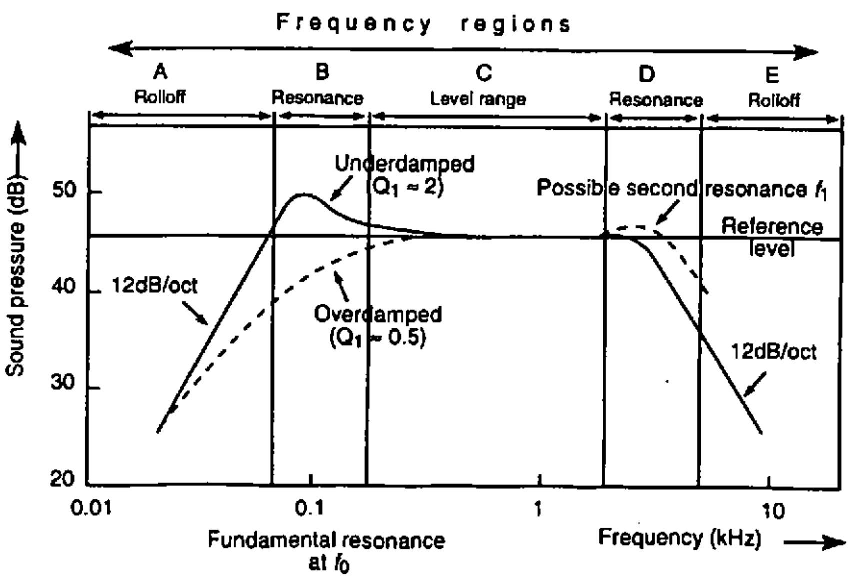


Figure 2: Developing the electrical analogue of a speaker leads to a bandpass filter.

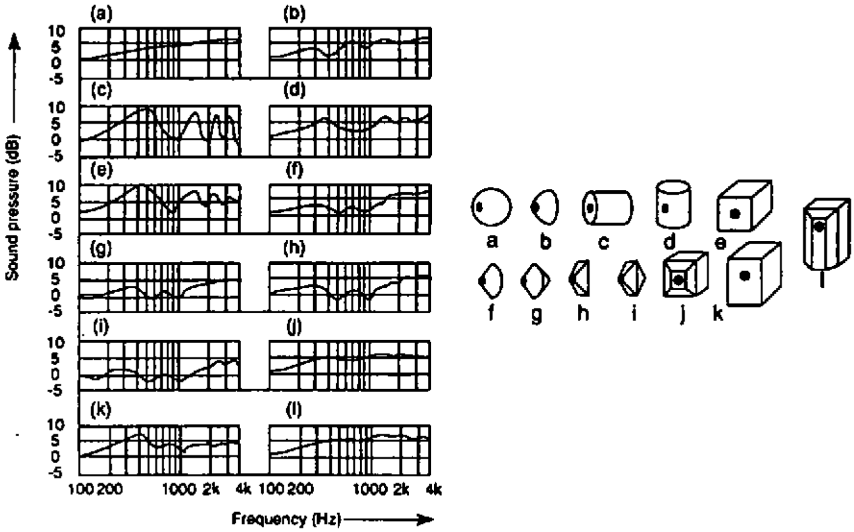


Figure 3: From Harry F Olson's book 'Acoustical Engineering' showing the effects of diffraction and reflection around various cabinet shapes on a known good drive unit.

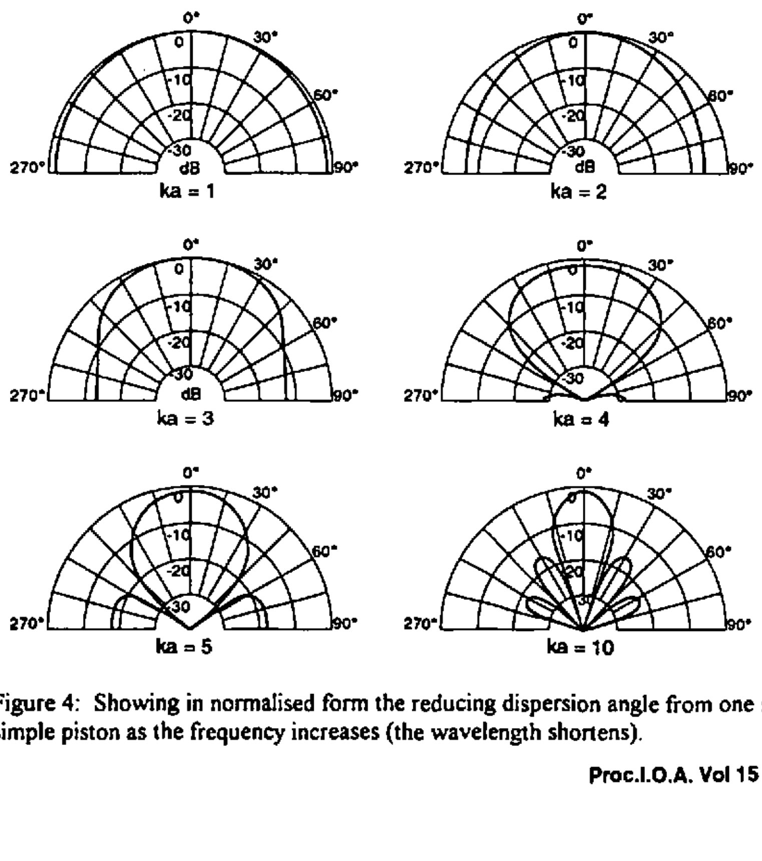


Figure 4: Showing in normalised form the reducing dispersion angle from one side of a simple piston as the frequency increases (the wavelength shortens).

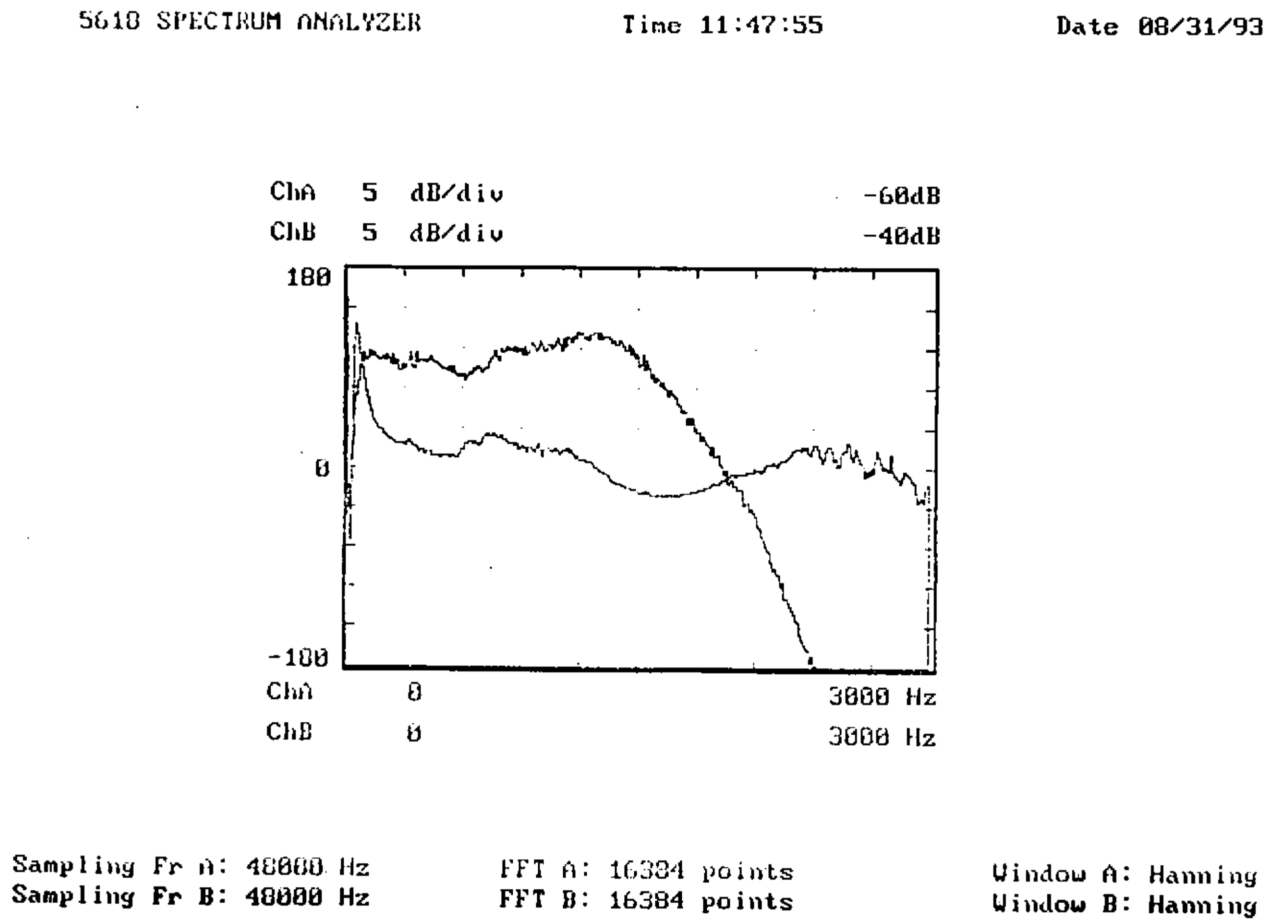
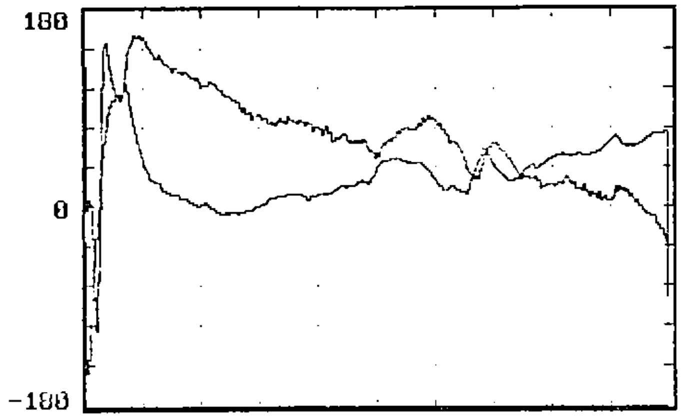


Figure 5 Woofer amplitude and phase response with no equalization

5618 SPECTRUM ANALYZER   Time 11:28:22   Date 08/31/93

ChA 5 dB/div -60dB
ChB 5 dB/div -40dB

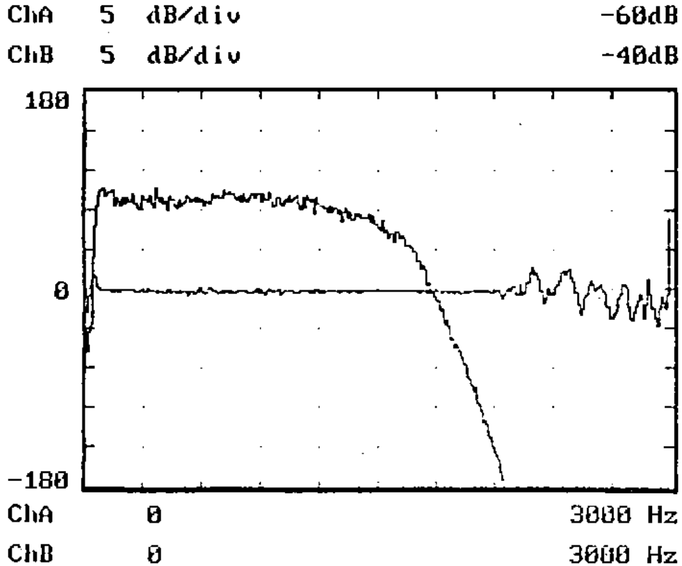


ChA 0 22000 Hz
ChB 0 22000 Hz

Sampling Fr A: 48000 Hz   FFT A: 16384 points   Window A: Hanning
Sampling Fr B: 48000 Hz   FFT B: 16384 points   Window B: Hanning

Figure 6 Tweeter amplitude and phase response with no equalization

5618 SPECTRUM ANALYZER　　Time 12:06:51　　Date 08/31/93

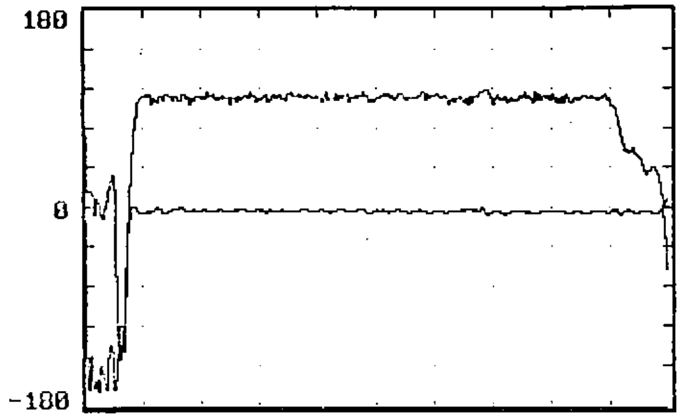


Sampling Fr A: 48000 Hz　　FFT A: 16384 points　　Window A: Hanning
Sampling Fr B: 48000 Hz　　FFT B: 16384 points　　Window B: Hanning

Figure 7　Woofer amplitude and phase response with equalization and FIR crossover.

5610 SPECTRUM ANALYZER Time 11:35:30 Date 08/31/93

ChA 5 dB/div -60dB
ChB 5 dB/div -40dB



ChA 0 22000 Hz
ChB 0 22000 Hz

Sampling Fr A: 48000 Hz FFT A: 16384 points Window A: Hanning
Sampling Fr B: 48000 Hz FFT B: 16384 points Window B: Hanning

Figure 8 Tweeter amplitude and phase response with equalization and FIR crossover.

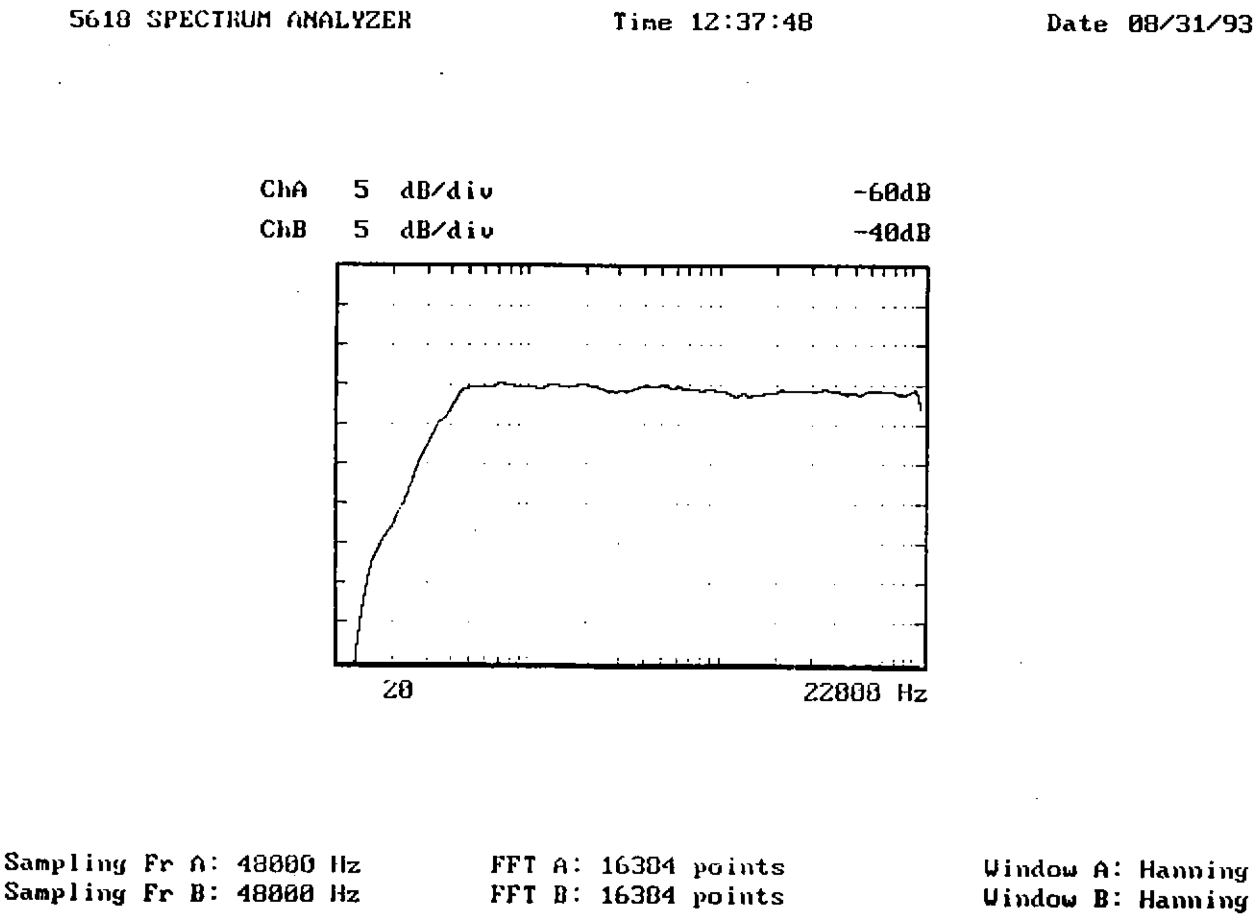


Figure 9 Combined amplitude response with equalization and FIR crossover.

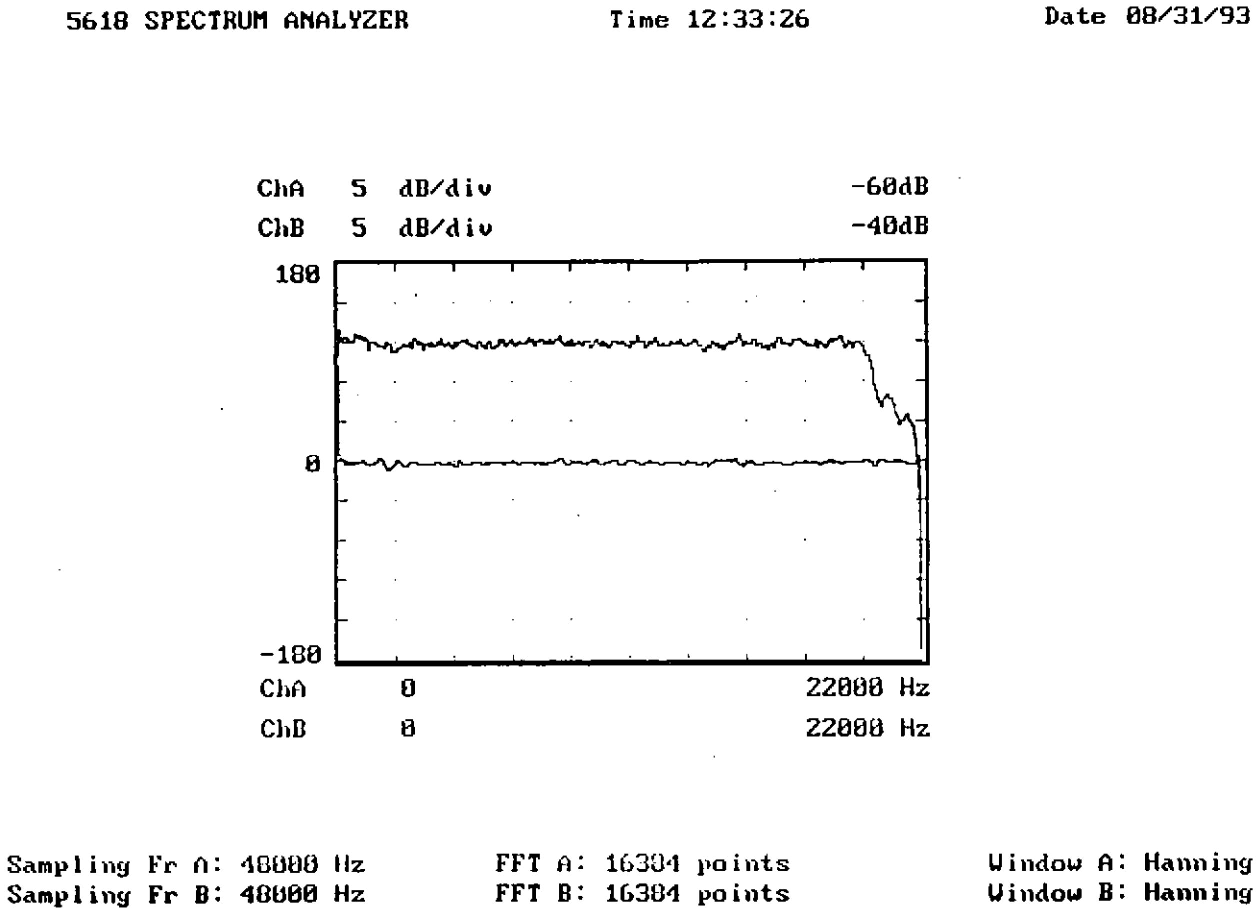


Figure 10 Combined amplitude and phase response with equalization and FIR crossover.