DSP DESIGN FOR DIGITAL MONITORING LOUDSPEAKERS

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0 INTRODUCTION

There is a move afoot, in this world of ours, to 'improve' all aspects of our lives. Unless good design principles are used, the price we pay is often increased complexity and confusion.

This paper will survey the progress which Tannoy has made with loudspeaker equalisation, its implementation into product, and the approach which has been used to apply this technique in a usable fashion to studio monitoring.

1 OVERVIEW

This paper is primarily a discussion of the factors which effect the application of DSP equalisation (both Phase and Amplitude) to monitor loudspeakers, in particular those based on point source technology. Factors which apply to the broader application of this technique, such as in domestic and PA technology, are also relevant.

An ideal loudspeaker has a unity transfer function. If we ignore, at this time, the inherent non-linearity of its motor system etc., the ideal configuration is Point Source, Phase Linear and Amplitude Linear. The reasoning is thus:

Point Source: Allows the generation of full range audio from a single spatial point, without the interference effects normally associated with discrete (multiple driver) designs. These effects occur mainly in the crossover region, between high and low frequency drivers when they are located close to one another. In this region both drivers attempt to reproduce the same tone with similar amplitudes, creating audible interference fringes. Thus, conventional loudspeakers are oriented along a vertical axis: generally people move around more in the horizontal plane than the vertical, so perception of the interference is minimised. In a studio environment this is less acceptable, where there might be people standing behind a seated engineer, both hearing slightly different sounds. These fringes do not occur with a Point Source driver where both the HF and LF driver acoustic centres are made coincident by tailoring the geometry of the combined driver .

Phase Linearity: This can also be thought of as applying a constant time delay to all frequencies, and is determined by a constant rate of change of phase with frequency.

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Consider the spectral content of music. A considerable amount of what makes individual instruments both musical and individually identifiable is determined by the presence of harmonic structures. Further, consider that most instruments have fundamental tones which occur below the crossover frequency, while some, or all, of their harmonics occur above the crossover frequency. In order perform a credible reconstruction of the musical instrument, it can be seen that the LF and HF sections of the loudspeaker must be time aligned. As a corollary, if you experimentally separate the harmonics from the fundamental (in time) then there is increasing confusion in the perception of the instrument, until a point is reached where you hear two distinct instruments: neither with the qualities of the original. It is proposed that Phase Linearity within a monitoring system is of significant importance in the psycho-acoustic reconstruction of the musical stage.

Amplitude Linearity: Maintaining a constant, controlled amplitude response is the first step in increasing the performance of monitoring systems. The advantage may not derive from keeping a constant amplitude with frequency, but in being able to tailor the response to follow a particular characteristic while maintaining Phase Linearity.

System Requirements: Given the three factors outlined above, it is then possible to define system requirements. To obtain the degree of equalisation required to give a worthwhile performance advantage, conventional analogue electronics may be discarded in favour of the complex mathematical processes which can be performed within a Digital Signal Processor (DSP). Similarly, the high power components used in a conventional crossover may, and should, be discarded. These components, no matter how good, have some non-linear characteristics, which give a degree of variability over time, temperature, power and physical stress. By removing them as a variable it is possible to further refine the performance of the digital equalisation process. The DSP incorporates the crossover function, so making the loudspeaker fully Active.

In studio monitoring, minimal system delay is mandatory, that is the time delay between the signal entering the system and its reproduction at the loudspeaker. This is not a system used only for recording playback, where time delay is not a significant issue; but a system which may be used in live monitoring applications where audio feedback delay must be minimised. Analogue systems do not suffer this problem, but it has caused no end of head-scratching amongst digital desk manufacturers. The development system has been limited to less then 40ms overall delay as a realisable and realistic target.

Finally, in order perform as a viable commercial product, the system must be as generic and modular as possible, and is implemented in a separate enclosure. At first sight this appears to contradict the desire most studios express: to have completely integrated monitors. The reasoning behind this decision is of some importance and will be discussed further.

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2 THE SYSTEM

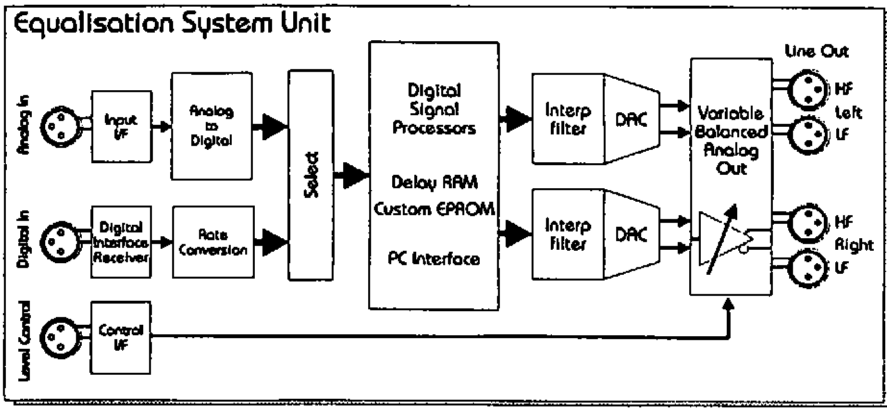

Fig 1

The bulk of the electronics used in this project are contained in a single rack mounted enclosure. It follows the basic pattern for this kind of system, having a stereo ADC or digital receiver, followed by a signal processing core, followed by a DAC stage which develops a two way stereo output. These outputs have level control applied and then feed into the amplifiers. The system is inherently modular, so it can be configured as an N channel by M way system of a cost/complexity defined by the national deficit of the country of your choice. Any system can be configured in the factory or on site by running customising software on a PC, and once installed is 'invisible' except for the level control input. The software within the DSP system performs the functions of Amplitude and Phase Equalisation, in addition to an active crossover.

Historically, much of the textbook content related to DSP filtering concerned itself with the operation of recursive/non-recursive (IIR/FIR) techniques and the implementation of classical filter structures. In recent years, due in no small part to the work of members of the IOA, AES etc., we find that methodologies designed with the DSP in mind are becoming better understood and more widely used. For example: adaptive filtering, noise shaping and inverse function filtering (where the filter is the inverse of the measured system response). More importantly, the limitations of such systems are becoming better understood.

The application of the DSP system from fig 1. is shown in fig 2, where the loudspeaker analysis mode is described.

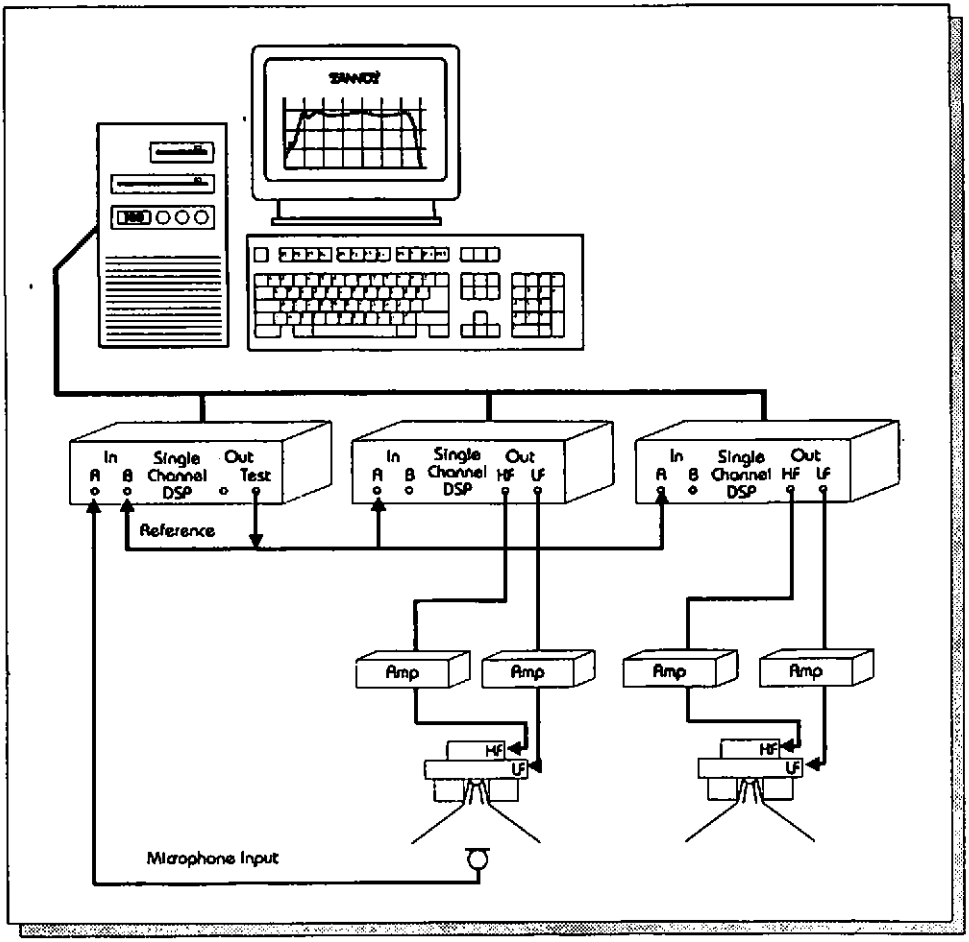

Fig 2.

A test signal (noise source) drives the unit under test, either the HF or LF drive unit. A microphone feeds back the signal to the Analysis DSP. The test and return signals are then compared in order to derive the amplitude and phase responses across the audio frequency band (15Hz to 24kHz). The PC, from which the system is controlled during the analysis phase, calculates the inverse filter function which, when applied to the driver under test, will linearize the response of both phase and amplitude. Given the response of the microphone, which can be removed from the equation, the programme linearizes the complete system, from analogue/digital input to analogue output, amplifier and speaker drive units.

The PC programme exhibits manual control over a number of features, such as the start and stop frequencies of the equalisation band. In addition to the crossover specification, there is control over the specific amplitude response at arbitrary frequencies, while still maintaining the linear phase target.

For the LF driver, the upper frequency of the band over which equalisation is performed is determined by the crossover point, or rather a point a little above the cut-off at ~1.6kHz. Here the desired amplitude response will be 80-100dB down and there is little point in using processor power to equalise a driver beyond this point. In the development system the crossover occurs around 1.6kHz, and so the increasingly common technique of decimation was used to bring the sampling frequency down (to about 6kHz) which enables a single DSP to perform Equalisation down to 15Hz. As a corollary to this, equalisation is performed on bands of 15Hz width.

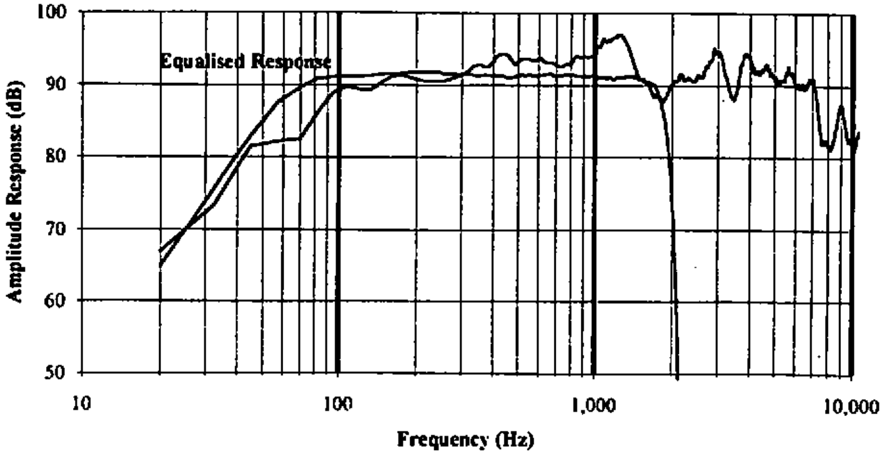

fig 3a. LF Driver Amplitude Response

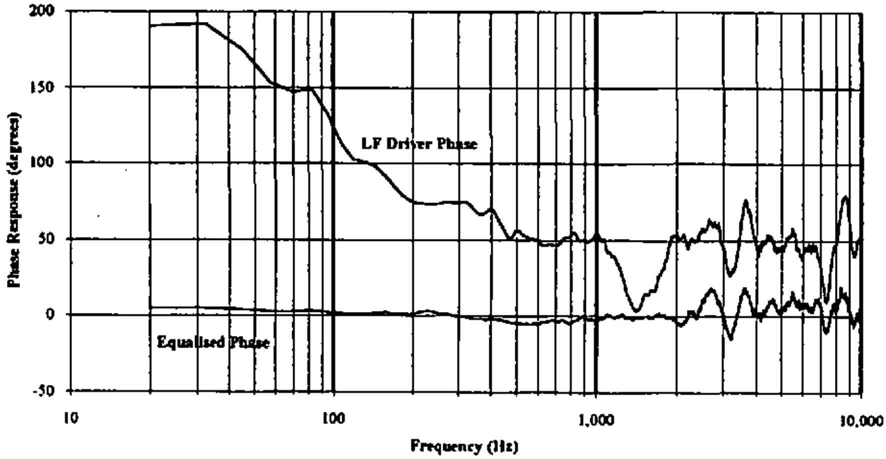

fig 3b. LF Driver Phase Response

Note that the Equalised phase response in figures 3b, and later in 4b, have been compensated for time delay and therefore appear to have approximately zero gradient. This flat line representation better illustrates the phase linearity of the system. Note also that the compensation above 2kHz is turned off, illustrated by the increased variation in equalised phase in fig 3b.

Equalisation of the HF driver is less demanding of processor resource than the LF driver, and is implemented at the system sample rate of 48kHz. When considering the range covered by the amplitude characteristic of the Dual Concentric unit used, special attention must be given to the scaling of the filter response. This is made necessary by the complexity of the filter process. If the characteristic were to be equalised to the peak then we should be applying 12dB of boost at certain frequencies. This is undesirable because of the potential limiting which can occur within the 24 bit fixed point DSP. In practice we apply a little boost to certain frequencies, but mostly we cut the response around the 2.5kHz peak of the HF unit. There is a careful balancing act to optimise performance, taking into account the factors of Noise, Stability and adequate Headroom. This is one of the few areas where the user exercises direct, iterative, control over the analysis programme parameters.

Dither, as always, is an important issue. Within the instructions which remain available in the 20us time frame of the DSP, it is possible to apply dither with a triangular probability distribution function (TPDF).

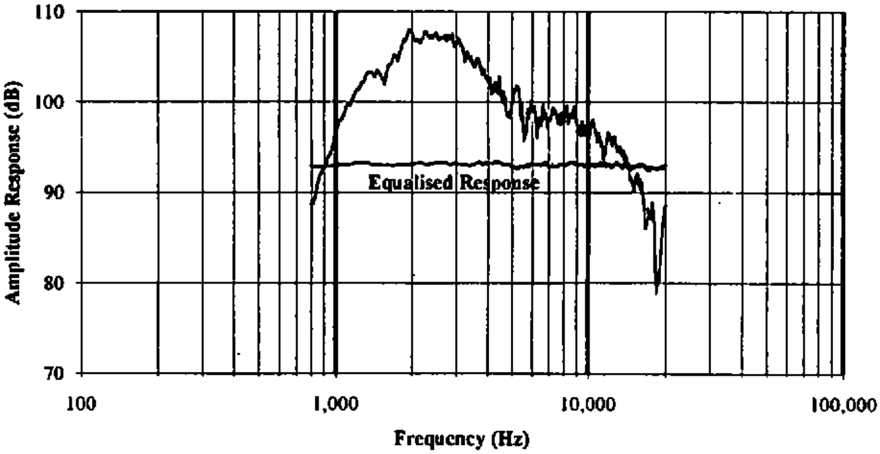

fig 4a. HF Driver Amplitude Response (800Hz to 20kHz only)

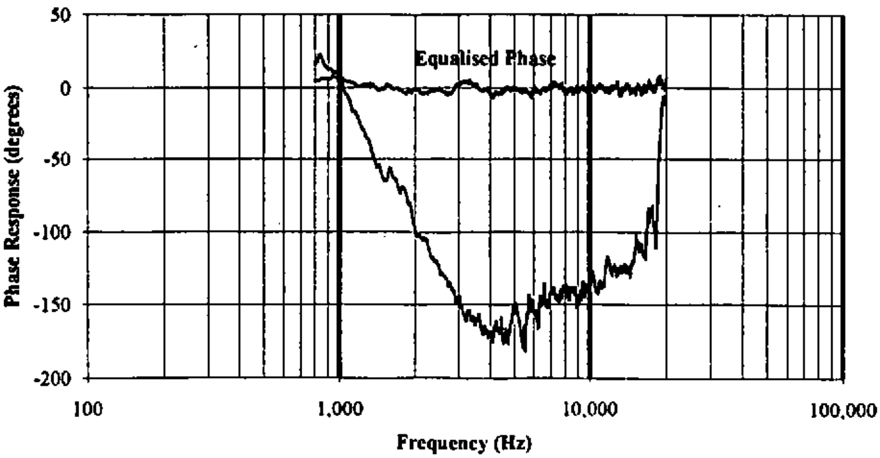

fig 4b. HF Driver Phase Response (800Hz to 20kHz only)

3 HARDWARE

Consider now, some of the requirements which effect the system design and the effect of design parameters on system characteristics.

Analogue to Digital Input.
Since the vast majority of current studio systems are analogue based, even though many sources are digital, then a high quality Analogue to Digital Converter (ADC) is mandatory. Until recently commercial Analogue to Digital Converters have not had an entirely satisfactory range for this type of application. At least 6 to 12dB of headroom above normal maximum level (before clipping occurs) is required in order to cope with some of the wide dynamic range or transient sources found in a studio. As a result, a number of the available bits in the ADC are not routinely used. Bummer. The most advanced converters available must therefore be used in order to meet the system performance standards. Developments in ADC technology are occurring rapidly at the moment and the need to respond to these changes in a timely and cost effective manner illustrates the importance of having a modular and upgradable system.

Digital Input.
Digital studios will increase in numbers. As a result so will the need for a digital input on the monitoring system. There are, however, a number of problems to be considered:

The mathematical processes within the DSP are characterised by, and therefore tightly bound to, the sample frequency. This leaves the designer with two basic choices:

i) calculate different Equalisation functions for each pre-defined sample rate, or
ii) make the DSP system independent of the studio sample rate.

By choosing the second of the above options we can

i) simplify the software generation process
ii) avoid the need for arbitrary switching of filter coefficient sets
iii) allow the design of an ultra-low clock jitter system.

If everything within the DSP system is run from a low jitter system clock, instead of using a clock recovered from the mixing desk, then the system becomes independent of the unknown and uncontrollable state of the studio clock. The system then becomes dependent on a studio quality asynchronous (digital) rate converter of which there is a limited selection at the moment. The up side is that with increases in the number of digital sources and digital studios, there will be increased support from chip makers for this type of technology.

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DSP

In common with a large proportion of the industry, this system uses the Motorola 56K series DSP as its engine. The advantage of using a standard serial interface within a modular system is that the processor is, broadly speaking, irrelevant and can be replaced with cheaper and better components as they become available. The presence of a wide dynamic range signal and the complexity of its processing, indicates a need for a 32 bit fixed point architecture (more 'commercial' than 32 bit floating point) in the near future. The complexity of the mathematical process implemented in equalisation systems requires a great deal of attention to areas such as quantization and rounding errors. These errors can easily get out of hand and 24 bits of DSP hardware can easily be reduced to 16 bits effective resolution by poor algorithm selection.

By utilising the highest quality loudspeaker drive units as a starting point, the complexity of the signal processing is reduced, resulting in a greater potential resolution.

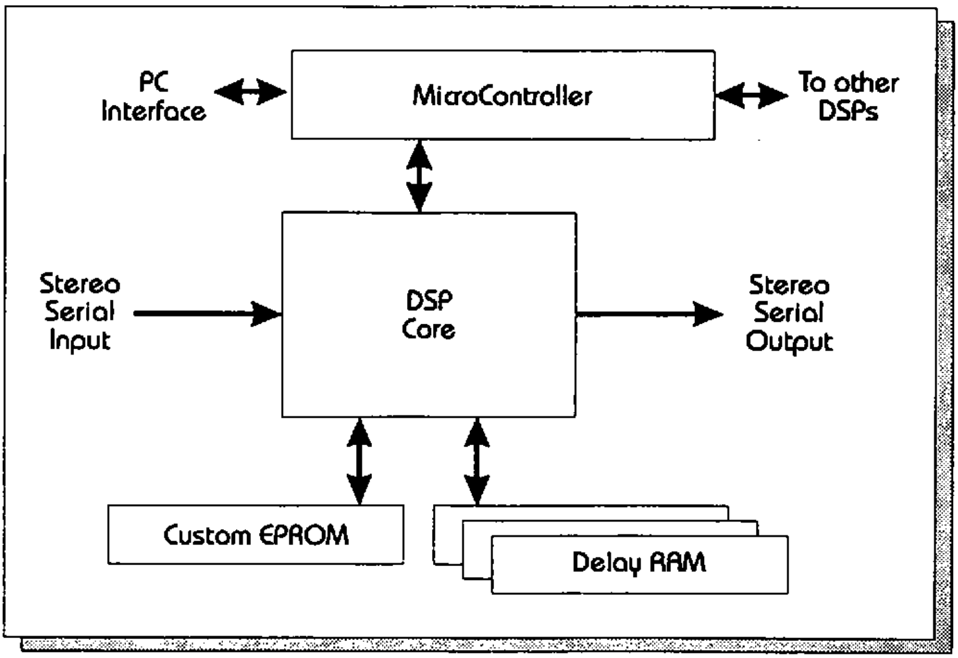

fig 5.

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The DSP core processes data for a single channel 2 way system, driving a single Point Source loudspeaker. The DSP programme is down loaded from the system micro-controller, which also handles communications with the external computer (PC). The loudspeaker specific data is held in Electrically Erasable PROM which allows it to be updated on-line. The external memory is required to hold data from the intermediate stages of the DSP calculations.

Digital to Analogue Conversion
The same logic used to specify the ADC applies here. DAC resolution which is adequate for CD replay is not necessarily good enough to handle the dynamic range found in live monitoring applications. A good 18 bit converter is OK but a 20 bit resolution converter is preferred. In the best traditions of rampant commercialism, however, a modular system allows for the realisation of a number of different cost/performance options suited for different market areas.

Level Control
In many ways this is the most difficult part of the system to design. It has issues which relate not only to the performance of the system but to the way in which it is used, forming the bridge between a box which works on the lab bench, and the world of commercial application with which we are all inextricably linked.

An analogue input to the system should be at maximum line level in order to minimise quantization artefacts during the Analogue to Digital Conversion process. This means that the system cannot successfully use the monitor (variable) output from a mixing desk, since low level inputs will only use a few of the available converter bits and so will suffer disproportionately from signal quantization distortions. Quantization and associated effects can significantly reduce the Signal to Noise Ratio (SNR). In any case, the digital output from the desk is also at full scale and a large degree of compatibility between digital and analogue input systems would be preferred.

Scaling of the signal within the DSP is not a viable option, again because of the significant loss of resolution at low levels. This leads us to the conclusion that level control must be performed in the analogue domain, after the DAC stage. To remain useful, however, the control function must appear in a familiar form at the mixing desk. Level control is therefore performed using a relatively conventional electronic gain control, which derives its control signal from a pot on the desk. The engineer can choose to rewire the monitor level pot or to provide a new control on the desk, at their discretion. All the DSP level control requires is a control voltage.

The performance of the analogue control must match the rest of the system in terms of distortion and noise. Channel matching to within 0.3dB, or better, is desirable between the HF and LF channels of a particular loudspeaker. This must be maintained across the full operating range. Listening tests have shown that the enhanced performance of the monitors can be upset by small, asymmetric, changes in tonal balance (that is changes which only effect one loudspeaker enclosure). This degree of matching, required over 4 channels, is beyond commercially available potentiometer controls. Nor is remote control of such a pot available in a form which meets system requirements.

6 RELIABILITY

Reliability in any Pro' environment is of paramount importance. Any kind of system defect reflects badly on all concerned and will not only effect studio performance on the day (frustration has a well known psycho-acoustic effect on alcohol consumption), but future business for the studio and the supplier of the primary equipment. It therefore behoves the designer to get the system reliable in the first instance. Unfortunately we are up against the immutable laws of Physics and Murphy. Reliability and associated issues are subjects which we see aired far too little within these learned journals, particularly since they can bring the results of the most astute and elegant treatise crashing to the ground within seconds! Reliability is however a science into which component manufacturers put a great deal of time and effort.

For example: a solder joint; like every other component in a system, has a defined (statistical) lifetime, or Mean Time To Failure (MTTF). That solder joint might have an MTTF of 5,000,000 hours under constant conditions, but which might drop by a factor of 10 when subjected to the stresses of a normal environment: switched off at night, subject to a daily temperature swing of 50C, corrosion in a humid atmosphere, vibration (which is a good reason for keeping electronics outside the loudspeaker cabinet). Mathematically combine these effects and you find that most of the complex boxes within a typical studio will have an MTTF of 3 to 6 years of normal use (that is chronologically, from the date of purchase to the date it dies: operating for around 4400 hours per year). This applies to boxes with several hundred components and several thousand connections: which accounts for most everything, from an amplifier upwards. Include mechanical and laser optical components such as may be found in the media players and recorders in a studio and the MTTF drops to the point where system redundancy and a maintenance technician are required. As a conceptual guideline, reliability is inversely proportional to the number of separate components. Increased complexity usually means a shorter lifetime. A means of cheating this rule, slightly, is to rely on increased circuit integration into more complex chips, then let the chip manufacturer worry about increasing reliability. The reduced number of solder joints in the system also helps.

A common methodology used by semiconductor manufacturers for establishing the reliability of chips (particularly for military products) is outlined below.

MTTF for a common exponential model is defined as the time after which 63% of the population of parts have failed, and is equal to $1/\lambda$ where λ is the failure rate in terms of FIT s

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$$\text{FIT s} = \frac{F \times m}{D \times T \times AF} \times 10^{9}$$

Where F = No of Failures
m = Factor to establish Confidence Interval
D = No of devices under test
T = Time (hours)
AF = Acceleration factor for failure mechanism at given temperature

The accelerated testing of semiconductor devices by elevating the temperature is a fairly well understood mechanism and is used extensively in determining reliability figures. Waiting for components to fail under test in real time is not an option.

Observing recent developments in integrated technology indicates that CMOS and similar digital processes can be operated into millions of hours. This is mainly the result of reduced gate topologies, lower power dissipation and improved process controls. Combine this fact with a highly integrated design, such as our DSP system, where most of the functionality appears in few chips, then system reliability increases dramatically.

Estimates show the MTTF of the DSP Equalisation system to be significantly longer than the more thermally stressed amplifiers, drive units, or the period within which somebody pours coffee into the system. Nevertheless, reliability is such an important issue that an active analogue backup system should be included for, if nothing else, peace of mind. It is worthwhile noting that long term reliability is increased by keeping the electronic systems isolated from the vibrations from the loudspeaker drive units.

7 AMPLIFICATION

As stated earlier, the channel matching between HF and LF channels of a given loudspeaker should ideally be kept to within 0.3dB. This will inevitably require a system to be calibrated for specific amplifiers, drivers and desk. The greatest variability and the likeliest prospect for being modified/replaced is the amplifier. A minimum performance specification for third party amplifiers is therefore required. The most important criteria will be for channel matching of gain characteristics over the normal variables of time and temperature.

Given the likelihood of an occasional modification to maintain peak performance, that wonder of modern technology: the multi-turn pot, is used. Calibration of the DSP system at the analogue input defines the maximum normal level fed to the ADC section. This is the only conventional analog control in the audio signal path. A trim function is applied to the output level control signal where it splits into the separate channel controllers. This allows independent channel matching to specific amplifiers without the need to insert another control into the signal path.

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8 SYSTEM ISSUES

As a result of conversations and tests at studios, potential customers usually desire a single speaker box to include all electronics and amplification. This eases system installation and movement around the studio. While it is certain that a 2-300W Near Field Monitor can be designed specifically to support this requirement, when the broader picture is considered this degree of system integration is unlikely to occur.

There are a number of considerations:

i) Performance and physical size issues would require the design of an amplifier for each speaker system. This raises a cost/performance issue for an amplifier which is only applied to a single speaker product.

ii) New loudspeaker and electronics enclosures would be required for each product, thus placing an additional cost burden on the customer.

iii) The increasingly stringent system requirements placed on designs by the European harmonised EMC (Electro-Magnetic Compatibility) standards, make it easier and more cost effective to design and test a single boxed system than to apply the standard separately to each system design. The cost benefit is passed on to the customer.

iv) A discrete system can be used to upgrade existing systems with relative ease.

v) The system becomes more flexible in terms of its generic application to custom installations.

9 CONCLUSION

One of the major concerns in audio product development today is the rate at which hardware and software techniques are changing, and the need to redesign product to maintain a competitive edge. Within the small to medium scale manufacturing in our industry the high ground will not go to companies designing highly integrated, fully customised products, but to those who can effectively modularise functions within a system design which can then be updated rapidly: creating flexible and highly responsive support to the customer base. Flexibility and speed of response will be the bywords of the coming decade and the attitudes which will engender these commercial qualities will, of necessity, begin with the people who define the objectives and methodologies used in audio systems. More importantly, those who buy and use the systems should be aware of the degree of consideration they are receiving from product manufacturers.