

L o u d s p e a k e r T e c h n o l o g y L i m i t e d



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Billy Woodman
Managing Director and founder of
ATC Loudspeaker Technology Limited



Company profile

ATC (Acoustic Transducer Company) was formed in 1974 to manufacture custom drive units for the professional sound industry – the first product being the 12" PA75-314 PA Drive unit. This was followed in 1976 by the ground breaking 3" Soft Dome mid driver which was to become, and still is, the ATC

signature drive unit. Built to handle 300Hz to 3kHz it was able to produce very low distortion and very high sound pressure levels. Hand-made using a massive magnet structure and tight tolerances it presented a quantum leap forward in loudspeaker design. With constant improvement over the years it is still unique in performance and innovation; flatteringly it has been widely copied without success.

By 1978, the company was producing its first speaker systems. The bass-reflex S50 and the

infinite baffle S85 establishing the naming tradition of the model number representing the internal volume of the enclosure, in this case 50 and 80 litres respectively. Another key to the ATC, studio-derived philosophy was the crossover arrangement: the S50 and S85 allowing for true active triamping. Through a rotary switch the user could access either the internal passive crossover or an external electronic device.

By 1985, the company had established itself in a big way in the professional sector, the growth forcing a move to the company's current premises in Gloucestershire. The BBC,

Abbey Road Studios, Pink Floyd, Nimbus, Peter Gabriel, Sony Music Studios, Sydney Opera House, Covent Garden Royal Opera House, Paramount, Warner Bros – these are but a few of the clients ATC has attracted over the years.

But one in particular inspired one of those watershed moments which was to forever alter a company's destiny. In 1985, Danish Radio needed a rugged, robust and transportable active speaker system. Tim Isaac had already developed for ATC an electronic crossover system. It was then a matter of matching this to a three-way amplifier, fitting it internally and producing what would evolve into one of ATC's greatest hits, the SCM50A. This in turn would establish ATC as one of the few companies able to produce an active system which lived up to the concept's promise.

Because of resistance to active systems in some hi-fi quarters, ATC released its major models in both active and passive versions, eventually creating smaller, more domestically suitable passive models such as the SCM10 and SCM20, and eventually producing its own range of stand-alone electronics, including pre, power and integrated amplifiers.

ATC has entered the multi-channel market with an advantage over all of the other manufacturers forced to convert from two-channel to 5.1. The secret weapon is ATC's direct involvement

with leading players in the industry, including Sony Music Studios in New York City, which uses ATC loudspeakers for the six channel DVD monitoring systems, TODD-AO in California, and others at the cutting edge.

Evolution therefore has equipped ATC with a current range not so different from the classic models of the pre-surround sound era, but eminently suited for whatever multi-channel may bring. The major advantage in its new multi-channel concept is the company's insistence that all products have the same sound signature or timbre. This allows seamless mixing of any product to form a 5.1 system. The current range offers six variants from C1 to C6 all with dedicated centre channels and subs.

The C7 was described by Stereophile Guide To The Home Theatre in the US as "The best loudspeaker system in the world". It received both the best loudspeaker and best component award for 2000 by the same magazine. The SCM70 which formed L.R and surrounds has now been superseded by the both Anniversary SCM50 and 100 Tower which on its launch won the 34th Japan Stereo Compo Grand Prix.

In the studio and mastering field ATC continues to fulfil the needs of the most discerning organisations, its user list reading like a who's who of the recording industry.

The Black Box Company has completed a studio complex in India using ATC which is reputed to have the best sound in Mumbai. Its fame has led to film premiers and celebrity galas being held there. The film industry in Mumbai is the largest in the world and has been known for inferior sound equipment up until now. It is fully expected that other studios will follow this new high quality route to the advantage of ATC in the near future.

The latest and one of the biggest projects in the world has been the exclusive equipping of the British Grove complex in London. Using no fewer than ten SCM300's and 8 subs it represents the pinnacle of studio sound quality. The SCM300 has become the monitor of choice for discerning engineers based in the USA, whilst the full professional range finds favour where quality is the criteria, throughout the world.

Other areas of excellence include the equipping of large concert halls with studio-type equipment. One of the most important of these is the Disney Concert Hall in Los Angeles, home of the L.A. Philharmonic.

A completely custom system was designed and built in-house to address this technically demanding project. The ATC approach is unique for this type of venue and offers a solution not available from other manufacturers.

ATC has recognised that there is a niche market for high quality, active sound reinforcement in theatres, wine bars and jazz clubs. To fill this need the new PA65 fully active cabinet has been designed. It now has a foothold in some of the most prestigious night clubs in London: China White, Aura, Umbaba, and Movida. Combined with custom-built sub bass units this combination offers the ideal solution for venues requiring high SPL's with Studio quality sound.



Above left: The S50 bass reflex, the first model to be launched by ATC

Right: Doug Sax Studio, Ojai, California

Far right: Custom installation for the Disney Concert Hall, Los Angeles



The design and development of high performance loudspeakers

The aims of the forefathers of the industry seem to have been completely forgotten and many loudspeakers of today's manufacturers are described as being musically involving, having pace, rhythm and slam or as just being a musical experience, words which might have a definite subjective meaning to the originator, cause confusion and suspicion in the mind of the public, and provide the less scrupulous with a cover for rather cynical products poorly engineered.

To be able to describe a loudspeaker as being of high performance it must comply with a range of related yet quite complicated criteria. These, when detailed, may appear obvious, however, the significance of the following simple ideas and design criteria, and their relative importance to each other in the design of a high performance loudspeaker, are all too often not properly understood.

The performance of a loudspeaker can be defined by its linear and non-linear behavior. Linear performance is defined by the impulse response and non-linear performance by harmonic distortion measurements.

The most important elements to consider in a practical design, which are encompassed by the characteristics of linear and non-linear behavior, are detailed under the following headings:

1. Magnitude Response

2. Phase Response
3. Time Domain Anomalies
4. Dispersion and Directivity
5. Harmonic Distortion
6. Amplitude Intermodulation Distortion
7. Hysteresis Distortion
8. Dynamic Range
9. Motional Impedance

Linear Distortion

1. Magnitude Response

The magnitude response of a loudspeaker, measured using analogue techniques, has been the mainstay of most loudspeaker assessment for decades.

By definition a "Linear Magnitude" refers to a magnitude response that has a constant level with frequency and only then will it not cause any linear distortion. We all know that this is practically not achievable and that the impulse response of a loudspeaker is largely dominated by the low and high frequency roll-off characteristics and by any resonant peaks in the amplitude response.

It is possible, however, to produce loudspeaker systems that maintain a variation in magnitude response within $\pm 1.5\text{dB}$ consistently between 100 Hz and 10 kHz and that have an excellent overall balance between bands. We believe that the balance between drive unit frequency bands is critical, particularly between bass and midrange in three way systems, and should always be better than 1dB.

Time is well spent on drive unit development in order to meet this magnitude response limit. It is much more elegant to use properly developed drive units which will then enable the use of simple crossover filters than to use complex equalization in an effort to correct drive unit magnitude response anomalies.

2. Phase Response

As with the magnitude response, the phase response of a system is usually measured on a single reference axis, midway between the bass/mid and high frequency drive units in a two way system and on the axis of the midrange drive unit for a three way system.

A system will be defined as being "Linear Phase" if the phase response is a straight line, when the frequency response has a linear scale and passes through the origin. The effect is then of a true time delay and will therefore not cause any linear distortion.

In practical loudspeaker systems however, the aim is to



design for a minimum phase response free from any abrupt changes that are usually indicative of high Q resonances. Even order frequency dividing networks using Butterworth filters offer the special characteristic that the phase of complimentary low and high pass filters are the same. The result is a greatly improved polar response and therefore improved coherence of the audio signal.

It is also relevant to include here that the delay between drive units due to acoustic centre misalignment is not audible, we believe, for delays below 2 ms. Therefore, providing the overall delay is within 2 ms. and there are no sharp phase response irregularities, then the system should be free from any subjective phase effects. ATC has incorporated analogue phase correction, operating through the crossover regions, in its active loud-

speakers since 1982. This is achieved by the addition of an all-pass filter (i.e. one with a magnitude response of unity for all frequencies, but a varying phase response) enabling correction for the delays due to the extra sound path length from the various drive units in a multi drive unit system. Such correction



serves to steer the main radiation lobe at the crossover frequency toward the listener. The result of such active filtering is to give much better control over the filter shapes with greater phase coherence and therefore a more uniform group delay characteristic. The subjective result, when compared with the same loudspeaker system but with a passive crossover, is of a broader and more stable stereo sound field with much more coherent drive unit integration and improved openness and timbre of reproduced sounds.

Digital signal processing promises several advances in phase response manipulation in the future:

A. Linear Phase Crossovers

Delays enables crossover filters to be constructed with a constant group delay, i.e. no changes in phase in the audio band. ►

The design and development of high performance loudspeakers

B. Excess Phase Equalization

The phase response of a loudspeaker through its low frequency roll-off can be equalized using digital delays producing subjectively a deeper and tighter bass response.

C. Magnitude Response Equalization

Digital signal processing can also be used to equalize a drive units magnitude response. However, in most cases, response anomalies will be polar dependent and therefore not equalizable with a single dimension equalizer. Magnitude equalization should therefore be applied with great caution.

3. Time Domain Anomalies

A high performance loudspeaker should have no high Q or delayed resonance's and must also minimize multiple arrivals of the same signal caused by reflections and diffraction effects as these add a hard and claustrophobic coloration to the sound, masking ambient detail and confusing the stereo image. Time domain anomalies are without doubt the most intrusive and tiring to the listener of all distortions. Careful drive unit and crossover design can ensure a flat and even magnitude response free from any low Q broad band resonances or response irregularities. High Q and delayed resonances however, which are common in hard undamped diaphragms and poorly designed crossover filters, are not so easily ameliorated. In fact the only successful solution is to design heavily damped flexible diaphragm structures having high internal resistance and great structural integrity even under high drive levels.

Best results have been achieved using quite steep curvilinear and domed diaphragms formed from polycotton and acrylic fabrics which are impregnated with plasticized PVA and other

viscous damping mediums to control resonant break-up modes which occur at high frequencies.

It is also equally important for the fundamental system resonance to be well damped, that is, have a Q between 0.3 and 0.6.

Loudspeakers with an under damped system resonance produce ill defined bass which sounds uncontrolled and excessive and masks midrange detail.

In fact what is really being said here is that for a high performance loudspeaker all resonant systems should be at least critically damped whether they are due to diaphragm break-up or the fundamental system resonance.

4. Dispersion and Directivity

The relationship between direct and reverberant sound is very important in high performance loudspeakers. It is clear that not only must the on-axis magnitude response be accurate and linear but also that the behavior off-axis must be both broad and even with frequency exhibiting no abrupt dips in amplitude. The aim should be to achieve a horizontal dispersion of +/-80 deg. With a -6dB @ 10KHz and a vertical dispersion of at least +/-10 deg. To ensure that in a well behaved room with a good RT vs frequency characteristic, The reverberant sound will be consistent with the direct sound in the listening area.

To achieve this criteria the highest performance loudspeakers will be either small two way systems with bass/mid drive units up to 160mm diameter or preferably three way designs where the midrange is no more than 75mm diameter and crosses over from the bass drive unit at around 300 Hz. The tweeter in this system should not be greater in diameter than 34mm and should be

crossed over from the midrange at around 3 KHz.

In a well behaved room when listening to a stereo pair of loudspeakers with a good relationship between direct and reverberant responses you will first hear the direct sound and then the reverberant field. It is generally agreed and probably true that the reverberant field masks periodic signals, however, it is also apparent that the ear has a precedence effect which means that for impulsive sounds the ear can hear phase dependent effects. Therefore, I believe that any critical judgment of reproduced sound is made principally on the first arrival or direct sound which gives most of the phase related cues and also the low level detail which is quickly lost in the reverberant field.

However, the way we perceive magnitude band balance and the full energy of percussive or impulsive sounds, is dependent upon the power response of the loudspeaker or how evenly it excites the reverberant field with frequency.

Clearly it is impossible to exclude from such a relationship the effect of room acoustics, however, for the purpose of discussing loudspeaker performance we will assume that the listening room has been properly treated and has no serious intrusive acoustic problems.

It should also be stated here that the use of DSP to equalize loudspeaker room interface problems is not an acceptable solution to that problem in critical listening environments if it involves modifying the direct sound from the loudspeaker. A dramatic effect of poor midrange dispersion, common in many two way loudspeaker systems, is demonstrated by recording engineers making incorrect magnitude band judgments and applying equalization, usually to the upper midrange, in an attempt to compensate for the apparent lack of

energy in that region. Many examples of pop recordings are available which demonstrate this characteristic. That is, a hard strident upper midrange which masks high frequencies, and makes vocals sound recessed while accentuating the bass.

Non Linear Distortion

5. Harmonic Distortion

Non linear distortion is the product of non-linearity in a system's transfer function. There are three principal sources of non-linear distortion in loudspeakers and they are all related to the drive system.

The first relates to the voice coil and magnet gap geometry and the non-uniformity of the distribution of magnetic lines along the length of the magnet gap. A short coil in a long gap renders the best solution regarding geometry, although not the most commonly used, and the distribution of magnetic lines will be improved by the use of an undercut centre pole. Further advantages of this geometry are the improved heat dissipation and therefore reduced operating temperature of the voice coil as well as a reduction in the variation of voice coil induction in relation to its instantaneous position in the magnet gap.

The second principal source of non-linear distortion is generated in the suspension system of the diaphragm assembly and is mainly contributed to by the spider. The spider presents a number of difficult design compromises when longer excursions are required in high power drive units. It must exhibit high axial compliance while also being progressively resistive towards the extension extremes so as to avoid mechanical damage and at the same time be very stable normal to the axis so as to ensure good voice coil centering in small magnet gap clearances. ►

The design and development of high performance loudspeakers

The third source of distortion is due principally to the inherently non-linear magnetic performance of steel. The alternating magnetic field created by the voice coil induces eddy currents into both the pole and front plate, adjacent to the coil, of the permanent magnet assembly. These eddy currents flow in such a way as to oppose the magnetic field producing them, (i.e. from the voice coil), and cancel out much of the self inductance.

This mechanism is minimized in ATC bass and bass/mid drive units by the use of a new material, which has the unique properties of high magnetic permeability and saturation as well as low electrical conductivity. We call it a super linear magnet material (SLMM). With this material fitted to the pole and front plate adjacent to the voice coil the eddy currents are suppressed and the impedance (self inductance) increases. The result is that third harmonic distortion is reduced by between 12–15dB.

It is evident from experiment, that distortion caused by eddy currents in the magnet assembly, is worse in long gap than short gap magnets.

In practice it will be careful drive unit magnet system and suspension design that will most effectively minimize non-linear distortion.

Having said all of that, since the main use of loudspeakers is to listen to music and speech, both of which have complex structures dominated by harmonically related tones, the presence of low order harmonic distortion is generally considered to be less audible and more tolerable than other forms of distortion.

6. Amplitude Intermodulation Distortion

Amplitude intermodulation distortion, however, is much more intrusive than harmonic distortion due to the products not being harmonically related to the original sound.

A recent review of active and passive loudspeakers at ATC confirmed that active loudspeakers, due to each drive unit amplifier operating only over a restricted frequency band, will have much lower amplifier borne amplitude intermodulation distortion than the same loudspeaker operated passively driven over the full audio frequency range, in fact, a full 20dB difference.

7. Hysteresis Distortion

The presence of hysteresis distortion implies that the system transfer characteristic is not always singlevalued for a given instantaneous input and will vary with both the change of direction and the level of the input and that it will therefore produce distortion that has a different phase to that produced by harmonic distortion.

Hysteresis distortion, as much as it exists in loudspeaker suspension systems and heavily damped soft diaphragm assemblies, does not manifest itself as an intrusive distortion. It is certainly not particularly evident in other measurements, for example, transient response, magnitude response or in harmonic distortion measurements. In fact, if care is taken over the choice of both diaphragm and suspension materials then they will largely have the characteristics of a simple damped spring and exhibit negligible hysteresis.

8. Dynamic Range

The issue of dynamic range is a complex one and although it is primarily controlled by voice coil operating temperature and magnet total flux it must be considered along with the mechanical integrity and freedom from break-up of the diaphragm and suspension structure. There can be no doubt that system dynamic range significantly effects the clarity of reproduced sound. Even quite simple combinations of instruments, for example a string quartet, will produce a maximum SPL well in excess of 100dB at 2m when starting from just audible pianissimo passages.

A loudspeaker that has significant power compression will tend to sound dull and boomy and the high voice coil temperature and consequent resistance rise will effect the loading of the passive crossover and therefore also modify the magnitude response of the system.

The dynamic range of direct radiating loudspeakers is in fact almost entirely determined by cost. Designers do strive to produce more sensitive small systems through the use of very light diaphragm structures but the scope for maneuver is limited if a correct balance between bass and midrange magnitude response is to be achieved for a given diameter of drive unit. Furthermore, light diaphragm structures almost always have low internal damping and therefore a tendency to exhibit high Q resonances.

To qualify in all respects as a high performance loudspeaker the requirements of dynamic range will for most designs be the largest compromise. A choice, which is made much more difficult as a consequence of the rapid developments in digital electronics



ATC SCM150A active monitors, Blackbird Studios, Nashville

during the past decade. Digital recording mediums offer a huge dynamic range with a peak to average of typically 12–16dB which means that even the most modest loudspeaker wearing the tag "high performance" must be capable of continuous output of at least 94dB at 1m while being driven from an amplifier of 100 watts or more.

9. Motional Impedance

The complex motional impedance of a typical two or three way passive loudspeaker system must have a modulus of impedance which varies within defined limits, never falling below the voice coil resistance. A minimum impedance modulus which does fall below the voice coil resistance indicates a ringing filter in the passive crossover which will cause time domain distortion as well as presenting a difficult load for the driving amplifier.

A Review of the Limitations of DSP as Applied to Room Acoustics

Recording studio control rooms are a particular case in which loudspeakers must interface with the acoustic environment of a room to produce a neutral fidelity so that critical judgements can be made of live and recorded/reproduced sound quality.

Listeners appear to judge sound source quality and character largely despite the room response characteristics. This is clearly demonstrated by the easy recognition of a familiar voice in many different acoustic environments. Thus, it seems likely that the direct sound from the loudspeaker will play the most significant part in any judgement of sound quality.

Loudspeaker anechoic performance can therefore be considered, in any listening room, to be more important than the room response. That is not to deny, however, that the room does have an effect, but not to the extent that the measurement might suggest. Nor does it put into context the relative importance of the direct and reverberant sound.

The Direct Sound

The characteristics of the direct sound are defined by the loudspeaker performance. The magnitude part of the frequency response, the mainstay of most loudspeaker assessment for decades, is largely defined by the impulse response which is dominated by the low and high frequency roll off characteristics and by any resonant peaks and dips in the magnitude response. A high performance loudspeaker should have a magnitude response which is free from any peaks and dips and maintains a magnitude within +/- 2dB between 100Hz and 10KHz whilst having -6dB points at 60Hz or below and 15KHz or above. The roll-off characteristics at both ends of the magnitude response should be

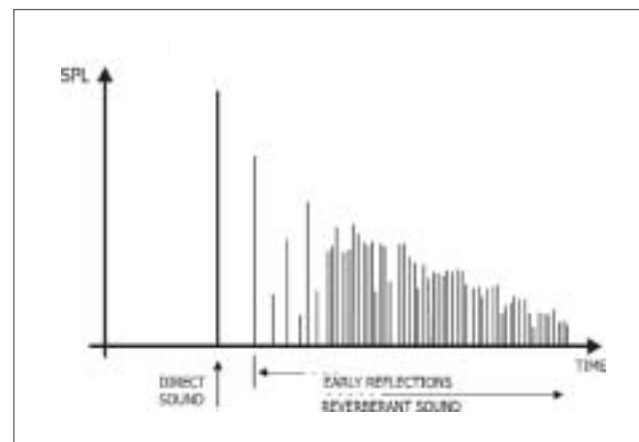


Fig. 1: Arrival times and levels of a sound

smooth and of a slow rate.

Fig. 1 shows the arrival of the direct sound, followed by the early reflections and the reverberant field.

The phase part of the frequency response is equally as important and should be of minimum phase and free from abrupt changes usually indicative of high Q resonances or uncompensated crossover filters. For a waveform to be reproduced accurately, all of the frequency components should be reproduced not only with the correct relative amplitudes but also with the correct relative phase. Significant changes in the relative phase of the components of a waveform will cause changes in timbre and the clarity of pitch. Furthermore, for transient or short duration sounds (common in music) we discriminate only between sounds in the relative phase of their frequency components and not in their relative magnitude. This is

true for sounds with durations as small as 2-3ms, i.e. accruing before the arrival at the listener's ear of early reflections, usually occurring between 5-80ms after the initial sound. Therefore, the room response has a minimal effect upon the subjective judgement of impulsive and short duration sounds.

The direct sound from the loudspeaker should also be free from high Q or delayed resonances and multiple arrivals of the same sound caused by reflections and/or diffraction (time domain distortion) as these effects add a hard claustrophobic colouration to the sound, masking ambient and low level detail.

Finally, the dispersion (polar response) of a loudspeaker, particularly in the mid and high frequency region, should be as constant as is possible with frequency and of the order of +/- 80 degrees. This will ensure that, if the listening room is well designed, the reverberant field will be evenly excited with frequency and the room response, therefore, will be as uniform as possible.

Fig. 2 shows the relationship between driver diameter, frequency and directivity. In order for a loudspeaker to have broad and even dispersion, the drive unit utilized for each frequency band should only operate up to the frequency limits shown by the curved line $ka=2$.

All ATC loudspeakers are designed to achieve this.

The Room Response

In all listening rooms reflected sound is always present and will, to some extent, influence the perception of sound quality. The listener, however, will first hear the direct sound followed quickly by early reflections (typically 5-80ms later) and then by

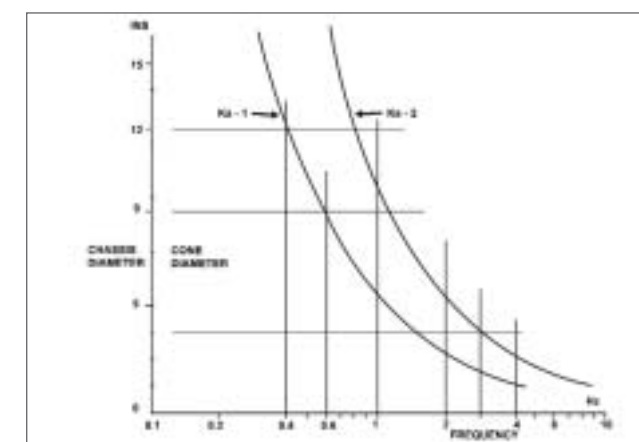


Fig. 2: Curves of Drive Unit Diameter vs Frequency

the diffuse and modal regions of the room response (see Fig. 1). By definition, the room response is the sum of its natural resonant frequencies. At mid and high frequencies all modes overlap producing high modal density and consequently the response is diffuse and uniform. At low frequencies however, the modal density is low, but due to the magnitude and slow decay of the predominately axial room modes, frequency response fluctuations, particularly suck-outs, can exceed 40dB. Discrete standing waves can occur in typical listening rooms up to 400Hz and increasing acoustic absorption is the only successful means of providing better measured results. In fact, although low frequency absorption is very difficult to implement the aim should be for acoustic absorption to be uniform with frequency and for the Q of any discrete room resonances to be less than 0.6. Individual room resonances will then be undetectable. The

A Review of the Limitations of DSP as Applied to Room Acoustics

RT should then be uniform over the broadest possible frequency band and a source of direct sound, as described above, will ensure the best possible room response.

The Application of Digital Signal Processing to Equalization of Room Response

There will always be significant limitations in the combined performance of room and loudspeaker if DSP compensation is applied to the room response which involves modification of the direct sound from the loudspeaker. Any system of room response compensation, regardless of its complexity, by the method of pre-convolution of one or more loudspeaker signals (i.e. modifying the direct sound) with some pre-defined inverse response, cannot guarantee a region of equalization, within the listening area, greater than half a wavelength from the measuring point. This suggests that the use of DSP room equalization, if used at all, should be limited in application to the low frequency room mode region where the equalized area will be functionally large enough to be practical for the listener.

A more effective solution can be implemented using conventional passive room treatment or even active low frequency absorbers within the listening room and, if correctly placed, will improve the low frequency performance, particularly with regard to suck-outs.

Equalization of suck-outs in the low frequency region of the room response will then only be limited by the performance of the passive absorption, the room dimensions and the dynamic range of the electronics and loudspeaker that make up the active

absorber. However, just equalizing the low frequency region of the room response using an active absorber does leave a decay lacking in low frequency spectral content, which may be as undesirable from a psycho-acoustic point of view, as the fluctuations in room response before equalization.

In summary, as discussed earlier, since most critical judgements are made from the direct sound, no room response equalization that involves modification of the anechoic response of the loudspeaker can ever be acceptable in a critical listening environment. At best, active absorbers could be used in conjunction with low frequency room treatment.

Therefore, the application of DSP to room response equalization will always be inferior to a room correctly designed in the first place or one later modified using passive or active absorbers. Even given this, the single most important factor in this complex equation is the use of correctly designed and manufactured loudspeakers displaying the characteristics discussed earlier.

In this respect ATC have no equal.

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Sony Music Studios, New York



The benefits of Active over Passive loudspeaker systems

The performance benefits of active over passive loudspeakers is substantial. Even a system, which incorporates the best available stand-alone power amplifier, will never achieve the performance of a similar active system. There are very good engineering reasons why this is true and the following brief will introduce some of the issues.



ATC's 350W Tri-Amplifier

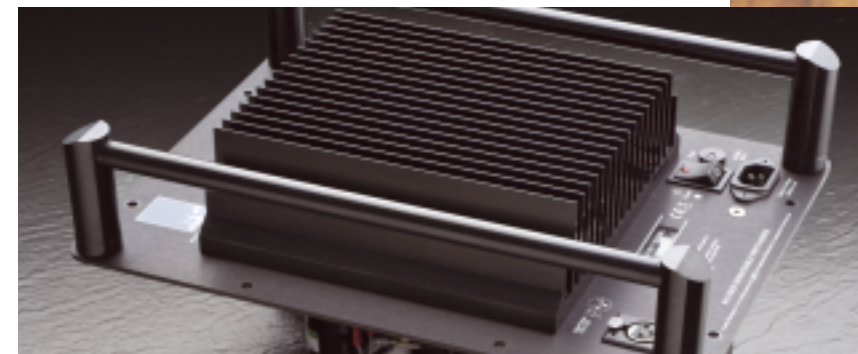
Far right (main picture): ATC fully discrete Tri-Amplifier



1. The magnitude of the frequency response of both active and passive loudspeakers can be controlled, with good design, to be within 1dB of one another. However, the phase component of the frequency response will always be better in an active system. The active filters produce better filter roll-off characteristics at crossover. Combine this with the inclusion of a variable all-pass filter at each crossover point to correct the phase response of the drive units through the crossover regions and the result is a loudspeaker with much better group delay characteristics. The benefit to the listener will be improved polar response and therefore radiated power response. Such an active loudspeaker will, therefore, have a large stable sound field with stable imaging and source location not possible with a passive loudspeaker.

2. A passive crossover will only operate correctly into the load impedance of a particular loudspeaker drive unit. However, the impedance of a loudspeaker drive unit will change with the amount of power input. This is because loudspeakers are very inefficient and most of the input power is dissipated as heat in the voice coil. As a result the temperature of the voice coil will rise and, because copper has a positive temperature coefficient of resistance, the impedance of the loudspeaker drive unit will rise. The result will be frequency response errors as the filters move from their designed response with increased input power. This effect does not occur in active loudspeakers where the filter response is maintained independent of input power to the loudspeaker.

3. Because the amplifiers in an active loudspeaker system are only required to operate over reduced frequency bands the intermodulation distortion products present in a passive system will be dramatically reduced, by typically 20dB, in an active system.



4. In an active system the absence of a passive crossover and long cable runs together with a known amplifier damping factor prevents the modification of the loudspeaker drive unit "Q" ensuring better controlled low frequency performance.

5. For a given amount of amplifier power an active loudspeaker can be expected to produce approximately 6dB more level than the equivalent passive system. Furthermore, powers may be more optimally specified in an active system. A tweeter, for example, requires much less power than a woofer to produce a balanced system performance.



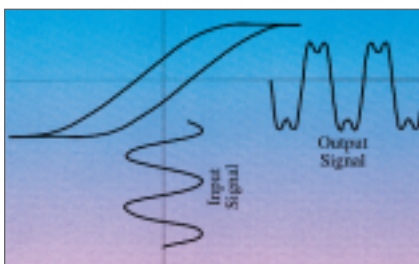
Super Linear Magnet Technology

Since the invention of the moving coil loudspeaker, designers have been looking for ways of improving the sonic performance of their systems. No-one has put more emphasis on this than ATC, and with the development of the Super Linear magnet system, one of the longest standing obstacles to audio engineering perfection has been removed.

The detrimental effects of magnetic hysteresis have been known for many years, but it has taken a combination of timing, with the right material coming to the market, and ATC's engineering abilities to bring the technology into loudspeakers.

Hysteresis

The magnetic performance of steel is inherently non-linear. From work first published back in the '30s (mainly concerning transformers and rotating machines) hysteresis has been known to be at the root of the problem, with the induction of eddy currents a compounding factor.



The effect of Hysteresis. The non-linearity of this magnetic mechanism results in a distortion of the input signal which is audible as colouration

Although much work has been published in the field of hysteresis distortion in loudspeakers, the work has never before seen a production transducer which obviates the distortion mechanism. ATC has finally cracked this nut with the help of a new material, which has the unique properties of high magnetic permeability and saturation level and low electrical conductivity. We call it Super Linear Magnetic Material (SLMM).

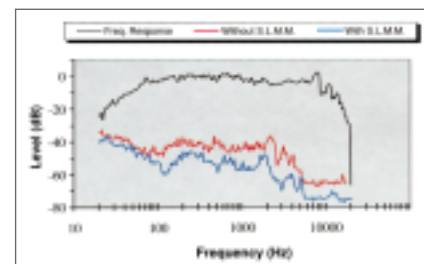
The effect of Hysteresis. The non-linearity of this magnetic mechanism results in a distortion of the input signal which is audible as colouration.

Super Linear

All ATC loudspeaker systems are now equipped with Super Linear technology. This is in the form of rings of SLMM which replace the steel regions concentric with the voice coil. The effect of the rings is to reduce the third harmonic distortion by 10-15dB between 100Hz and 3kHz. This makes for distortion comparable with most electronics.

Theory

Experiments were performed on a blocked voice coil with the magnet left un-energised. It could be thought of as a cored inductor. A current was passed through the coil and second and third harmonic distortion components were measured. Mathematical analysis, in conjunction with the experiments, has revealed some surprising answers to the question of why replacing the steel regions with SLMM has such a dramatic effect on the distortion.



Frequency response and third harmonic distortion in the acoustic output of the SB75-150 bass-mid driver

Firstly, the magnetic field in the regions concentric with the coil is reduced by a factor of around 10%. This is interesting because, within a non-conducting material, one would intuitively expect the magnetic field to be much lower, as the current density has to be lower. Not so in this case. The field is maintained by the steel pole and front plate.

Secondly, the presence of the SLMM increases the self-



SLMM rings shown here in green

inductance of the voice coil. When eddy currents are allowed to circulate in the system, they oppose the magnetic field producing them (ie. that from the coil) and 'cancel out' much of the self-inductance. With the SLMM in place eddy currents are suppressed and the self-inductance (ie. the impedance) goes up.

Thirdly, whilst the impedance, and therefore the fundamental voltage across a blocked coil goes up when the rings are fitted, the harmonic components, that are induced back into the voice coil, stay the same. This is because they are dependent only on magnetic field, which as we have seen, does not change very much. The net effect is a rise in the signal/distortion ratio.

The 'Bottom Line'

Two important issues result from this development in transducer technology.

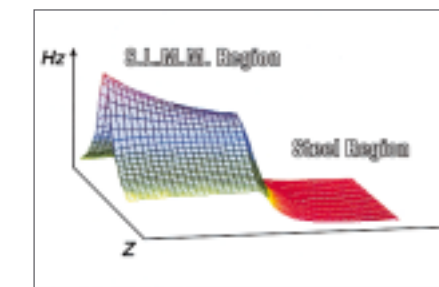
Aural benefits

Most importantly, we have achieved a significant improvement in sound quality. Reducing the level of distortion by such a dramatic amount has revealed another layer of information to the listener. Ambient sounds and low level effects that were previously masked are now clearly audible and provide an enhanced sense of realism. The articulation of male vocals is

markedly improved and piano reproduction is given a new lease of life.

Scientific benefits

The difficulties in solving non-linear field problems have constrained past research efforts to semi-empirical numerical approaches, and no-one has really been able to analytically pin-point the mechanism by which the distortion was entering the voice coil current. The complexity of the mechanism and the



A simulation of the magnetic field (Hz) within the pole of the SB75-150 Super Linear driver. In the cylindrical geometry used for the model, the z-direction is along the axis of the loudspeaker and the r-direction is along the radius of the pole/frontplate. The magnetic field does not change with the azimuth angle, so this direction is ignored. It is clear from the figure how the SLMM affects the magnetic field distribution, and this information allows for the calculation of the harmonic distortion induced in the voice coil, after

mathematically subjecting the magnetisation to a non-linear relationship (which represents the hysteresis within the pole and frontplate

diversity of contributing phenomena explain why it has taken the industry so long to solve this particular distortion.

Conclusion

The introduction of Super Linear technology has heralded probably the most important development in transducer design for the past fifteen years. It has been a practical success in that transducers incorporating the technology are in use across the whole range of ATC systems, and the improvements in sound quality are not subtle. Furthermore, ATC has analysed and succeeded in defining the complex non-linear electromagnetic mechanisms within the transducers. This work should pave the way for a new generation of transducer technology.

ATC users around the world

Doug Sax
Mark Knopfler
Sony Music Studios, New York
Warner Bros Burbank, CA
Polygram Wiseloord Studios, Holland
Walt Disney Concert Hall L.A.
Lenny Kravitz
Skin (Skunk Anansie)
K & A Productions (Naxos)
Pioneer Optical Disc (Barcelona)
Telarc
VTR TV Production Company
Paramount Pictures, Hollywood
BBC - UK
Lightning Seeds (Ian Broudie)
Pink Floyd's Studio
Dave Gilmour
Jarvis Recording Studios, NY
Dep International
John Kurlander
Ian Anderson
Manor Mobiles
Beacon Studio (Dublin)
Hans Zimmer
Karl Wallinger World Party
Albert's Music
Angel Studios, Islington
Chelsea Studio NY
London Post
Enya
Theierry Allard, Bruxelles
Matrix
Swedish Radio
Carlton Television (Nottingham & London)
York Street Studios
Boogie Park, Hamburg
Bruce Dunlop Assoc., London
Prince Sufri, Brunei
Tom Petty
Hokkaido Bunka Hoso
Broadcast Station (Sapporo)
Daniel Bedingfield

Sirensound Audio Archiving
Monster Music, Madrid
The Wigmore Hall
Crazy Sound, Guadeloupe
Oorong studio (Japan)
Yellow Shark (Cheltenham)
Expand Studio (London)
Aura London
Blacks Lab (Nashville)
Winds Over The Earth
Michael Winner
Abko NY
Sony-SACD NY
Chuck Ainley Backstage Studio Nashville
Fluke, London
Spectral Harmony, Bombay
Hamon Studio, Tel Aviv
Zaza Studios, Tel Aviv
Loco Studio, Wales
London College of Music
Polish Broadcast
Lakeside, Switzerland
Tape to Tape, London
Canadian Broadcasting Corporation
Blackbird Studios (Nashville)
Loud Recording (Nashville)
The Rolling Stones
DTS
US Airforce
Cambridge University
Cleveland Institute of Music
George Massenburg
Dhani Harrison
Cristalphonic
The Living Room NYC
Plus XXX Studios Paris
Air Studios
Matt Aitken
Beethoven Street
Hokkaido Television (Sapporo)
Asahi Television (Tokyo)
Fife University

Moles Studio
ARC Studio
Kash Productions, Madrid
Albert's Studio
Telegael, Eire (6 Studios)
Kate Bush
Dairy Studios
Sain Records
Mute Records (Depeche Mode, Erasure)
Todd AO, USA
Nick Whitaker
(Internationally Renowned Acoustician)
Ground Control, L.A
Angell Sound, London (5 Studios)
Royal Opera House, Covent Garden
Sydney Opera House
Royal College of Music
Birmingham University
Essex University
Bristol University
University of Surrey (Francis Rumsey)
Ronnie Scott's Jazz Club
Lou Reed
SBS Television, Australia
Lansdowne Recording Studio
Greg Walsh
(Producer Paul McCartney, Tina Turner, Elkie Brooks)
Peter Walsh
(Producer Stevie Wonder, Peter Gabriel, Simple Minds, Pulp)
James Guthrie (Pink Floyd, Toto, Chicago)
The Tate Gallery
Bob Ludwig Masterdisk
Bruce Leek
Vogler Audio Media
John Richards
Tony Wass
Ishikawa Television (Kanazawa)
Tokai Television (Nagoya)
Fuji Television (Tokyo)



Loudspeaker Technology Ltd Gypsy Lane, Aston Down, Stroud, Gloucestershire GL6 8HR United Kingdom

Telephone 01285 760561 **Fax** 01285 760683

Email: info@atc.gb.net **Websites:** www.atc.gb.net www.softdome.co.uk

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