Stan's Safari *32*

STAN IS PLANNING HIS 'DEFINITIVE SPEAKER SYSTEM', AND EXPLAINS WHY HE'S GOING FOR HIGH EFFICIENCY

"horns have something of a mixed reputation for some commentators, yet their use has a long and enviable record in the high quality reproduction of sound"

In my last column I talked about the low-frequency section of my new 'work in progress' loudspeaker system. The dominant criteria is that it must produce an emotionally satisfying sound, rather than the usual 'specification performance' of low-colouration; flat frequency response, and so on. And a major part of my thinking led me to an unusual preference for high efficiency loudspeakers.

By high efficiency I mean loudspeakers that generate a sound pressure level at one metre of over 100dB when fed with a signal level of 2.83V (equivalent to 1W into an 8ohm load). How have I arrived at this preference? Let's consider a few obvious advantages. The amplitude of the coil and diaphragm movement is reduced by as much as ten times, so the effects of magnetic field and suspension nonlinearities are similarly reduced, as is the likelihood of large physical movements causing diaphragm break up and consequently quite audible coloration.

The signal level is much lower with high efficiency, so far less heat is generated by the voice-coil and thermal compression is avoided. (In simple terms this means that as the voice coil gets hotter so its resistance rises so the current in the coil reduces so reducing the cone movement and thus the sound level plateaus.) Having this higher efficiency also eases the problems of playing music with a full dynamic range.

My planned loudspeaker will generate 105dB from 1W one watt so peaks of 115dB it will only require a 10W input. By comparison a typical modern loudspeaker with a sensitivity of, say, 85dB would require a humungous 1,000W output amplifier to hit 115dB, and the bulk of that power would be wasted as heat. And as an aside, let me state that it's a darn sight easier to design an excellent 10W amplifier than a 1000W amplifier of comparable quality.

But for me the defining advantage of high efficiency loudspeakers is a characteristic of sound quality that I've never been able to define or measure, but which is apparent with remarkable consistency. When I listen to various horn loudspeakers (or indeed early JBL and Wharfedale models), I'm always struck by the free-flowing nature of the sound. It's not something tangible like transient response, but more a sense of the sound somehow having less inertia.

I'm not alone in this because the annual Munich High End show invariably includes a demonstration using various very old horn loudspeakers. And although they exhibit a restricted frequency response and quite audible colourations, they also bring a free-flowing nature to the music which keeps me and many others in the room. So in many ways my quest is to design a pair of loudspeakers which embody this unique characteristic, while still achieving the conventional desirability of a wide and level frequency range with low distortion and coloration.

At this point you would be entitled to ask if high efficiency is such a good idea, how come most loudspeakers today are very inefficient, with sensitivity figures in the upper 80s? In part, from a designer's viewpoint it is a whole lot easier to design such types. In earlier times valve amplifiers had outputs in the 3-25W region, so efficient loudspeakers were essential to get realistic sound levels.

This was particularly true in the early days of cinema when a 20W amplifier coupled to efficient horn-loaded loudspeakers could fill the auditorium with sound. But the modern transistor amplifier is relatively inexpensive to make and in reality a 400W amplifier doesn't cost that much more to build than a 100W amplifier. And with the availability of high-temperature voice coils and glues, loudspeaker drive units are able to absorb huge levels of power before they burn out.

The designer can now see a number of benefits. These include cheaper magnet assemblies with smaller magnets, and wider gaps between the pole pieces, instead of the difficult to make narrow gaps essential for the high levels of 'shove' required by sensitive drivers. Instead of lightweight cone assemblies, much heavier diaphragms incorporating enhanced stiffness and damping could now be fitted.

The modern drive unit is inexpensive to produce; is very reliable; can handle high power inputs; offers good linearity and can generate very low levels of coloration. Compare them with the James B Lansing drivers used cinemas in the 1940s -1960s. The latter have big, heavy, expensive Alnico magnets. Voice-coils are wound with flat wire inserted into very narrow magnet gaps, requiring real skill and precision during assembly or repair; lightweight cones are formulated to generate low levels of coloration but in truth have inadequate stiffness; and quite low power handling by today's standards. Along with the above factors comes the clincher

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that if such drivers were made today, very few could afford them.

But having found or designed suitable drive units, there are two ways of increasing the efficiency of the system. One is to use multiple drive units: for example, two drivers side-by-side will produce four times the intensity on the principle axis at lower frequencies. However as the frequency rises, phase anomalies occur as the distance between the cones and the listener is not identical for multiple drivers. This is the result of wave interference through the adding and subtracting of various frequencies at various angles, which causes comb filtering and hence an irregular response.

Because such interference is not present at low frequencies, I've used this design approach for the bass section using two (or possibly even four) JBL drive units, not dissimilar to the ones just described. But for higher frequencies we need a single source of sound so the preference is to use the other route to high efficiency; the horn.

Unfortunately (and possibly because of their common use in many squarky public address systems) horns have something of a mixed reputation for some commentators, yet their use has a long and enviable record in the high quality reproduction of sound. A quick trawl of patent office records and papers on acoustic matters reveals a huge wealth of literature. Indeed it often seems that every generation of audio designers has had a go at improving the genre since Lord Rayleigh's 1828 *Theory of Sound* treatise, and a large part of that literature describes numerous improvements or variations of the flare shape of the horn.

The simplest shape is the conical horn and we are all familiar with the exponential horn beloved of public address system designers. But there are many, many more, and since my first hands on experience at Martin Audio I have experimented with at least nine types of horn shapes, each of which has its own limitations of efficiency, response, resonances due to driver loadings, and so on.

In recent times many other flare shapes have been made possible through computer simulations. Many of these go by term 'waveguide', which somehow sounds more technical and modern than the traditional horn. Indeed the more you live with horns the more complex these essentially simple devices become. Design, materials and construction determine their efficiency; power handling; frequency response flatness; directional characteristics; and various different resonances and colorations. Regrettably most designs have failed in one or more of these parameters, so my next column will explore how horns work and how you can grow to love them. And how the input of friendly Soviet scientists working on acoustic weapons of war opened my ears to how the very air we breathe can distort the sounds we listen to.

But before you review the list of all the ways that a badly designed horn can screw up the sound, let me balance things out by saying that it often, surprisingly, just doesn't seem to matter. Despite what the measurements say, horns are often working at such a small fraction of their operating dynamic range (what we designers call infinitesimal amplitude) that many of those limitations just aren't heard.

This is where I go all philosophical, and maybe throw in a bit of 1960s Zen totality. It has always struck me that if the music is coherent (as when, for example, it is performed in a concert hall), then the music is heard through whatever coloration and clutter is created by the hall acoustics and the audience. In contrast, when we're at home we try to listen in almost sterile conditions: low ambient noise; ideal stereo seating position; low-coloration sound system; even maybe optimised lighting. Yet at least part of me reckons that if the music is reproduced in that important coherent form, then it can be listened through a degree of minor imperfections without any loss of emotional pleasure. Some of my most cherished musical experiences have been in clubs were the music passes through a cacophony of conversation and clattering cutlery. Just saying.

JBL K2 S9900