Audio Today

Developments in audio reviewed by Wally Parsons

SEVERAL MONTHS AGO, in the letters section of these pages, I printed a letter from a reader in California on the subject of bi-amping a set of automotive speakers. This reader turns out to be an old friend with whom I had lost contact, but who has been active over the years in professional audio. It is not a subject on which he really needed my advice, and was written with tongue firmly planted in cheek. So, why did I play it straight and run it, complete with answer (aside, of course, from filling space and proving that ETI really does get down into Canada's 11th province)?

The truth is, RWS touched directly on the fact that the whole subject of bi-amping, the techniques and appropriate applications, is not too well understood by many audiophiles, even many technically sophisticated ones, so this month seems to be as good a time as any to discuss it.

Bi-amplification, or, for that matter, tri- or multi-amplification involves powering each driver in a multi-way speaker system with its own amplifier, and splitting the audio spectrum up before the amplifiers, using either passive filters, similar in design to those used in passive cross-overs and single amplifiers, or active filters, using either discrete or integrated filters. There are several advantages to this, in addition to swelling the coffers of the amplifier manufacturers and the finance companies.

AMPLIFIER STABILITY

Although it is possible to design a passive cross-over network which will present a constant resistance load, in practice it is not so easy to build such a network. The large value capacitors required and the physically large inductors involved are not easily held to close tolerances, so that calculated performance and real performance may be quite different. The calculated response of such networks can only be obtained when each leg is terminated in a resistive load, a condition which does not hold with real drivers. At the low resonance a driver presents a reactive load whose value may be considerably different from its rated value, and at the upper part of its range it will show a rising impedance characteristic due to voice coil inductance. Clearly, the actual power delivered to an individual driver may not correspond to the calculated cross-over characteristic, and the impedance appearing at the cross-over input terminals is not only not constant at all frequencies but contains considerable reactance over several portions of the audio spectrum. Some amplifiers get very unhappy with this state of affairs and show their displeasure in their performance, a phenomenon apparently little understood, judging by the mystery which seems to surround the subject of speaker-amplifier matching, and which is undoubtedly a major factor contributing to the large differences frequently observed in the sonic character of many amplifiers of similar specifications.

DAMPING

In a well-designed speaker system, each driver's main resonance will be kept well below its operating pass band. If an odd order network is used the driver will be fed from a source impedance which increases as frequency becomes remote from the pass band, so it doesn't benefit from amplifier damping, even if fed from an attenuator pad.

OPTIMUM CONDITIONS

Undoubtedly, the ideal mode of operation would be one in which each driver operates substantially in the pass band in which its impedance is relatively constant and essentially resistive, with the low resonance, although outside the major pass band, well damped by an amplifier's low output resistance, that same amplifier presenting a constant resistive load to its section of the cross-over, which can now be designed to tailor such characteristics as phase response and frequency slope, in order to optimize total system performance. While we're at it, we can sometimes get an output greater than the sum of the amplifiers' powers, in other words, something for nothing.

Whoa! Back up, there. What happened to the "no free lunch" principle? Actually, it still holds, but we are now optimizing our available power. Press on, and we'll see how.

POWER, VOLTAGE, CURRENT, AND ALL THAT JAZZ

We are accustomed to thinking of power amplifiers in terms of power output, but often forget that they are really constant voltage devices, and that signal levels within the amplifier are voltages, so we are now going to look at the output **voltage** of an amplifier when handling signals of different frequency and amplitude.

Fig. 1 shows the composite waveform which results from combining two pure tones remote in frequency and in which the higher frequency is of lower amplitude than the lower. The frequencies involved are 100 Hz and

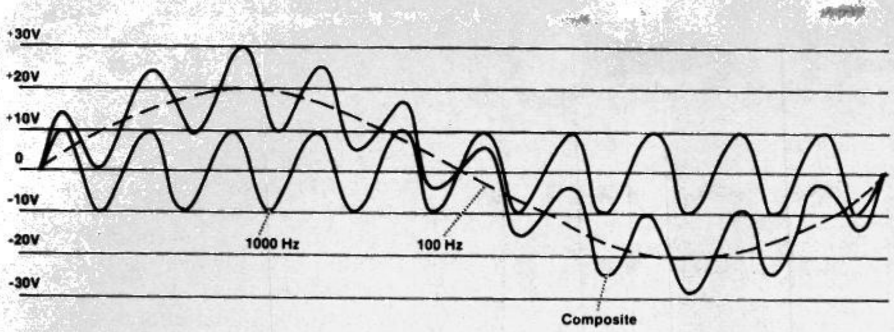

Fig. 1. Two signals of differing frequencies demand a surprising amount of power.

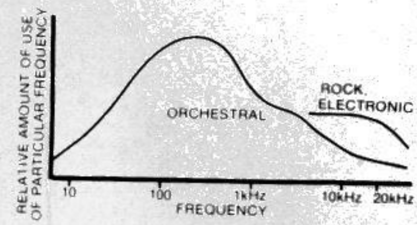

Fig. 2. Energy distribution of (classical) music.

1000 Hz, respectively, and the voltages are 20 V and 10 V, peak voltages. These voltages will result in power levels of 50 W and 12.5 W respectively, across an 8 ohm load, provided that the load is resistive. It can be seen that the sum of the two voltages will result in a condition in which the total voltage is 30 V, for a total maximum instantaneous power of 112.5 W. In other words, if an amplifier is required to deliver high power at some low frequency, and is called upon to simultaneously handle another frequency at only half the level of the lower frequency, it still must deliver **over twice as much power** as it would if it only had to handle the lower frequency. Conversely, if it is called upon to deliver power at some middle or high frequency, and then we add a lower frequency at only twice the level, the amplifier must be capable of supplying almost **ten times as much power** as it would otherwise have to deliver.

Wow! No wonder so many of the super amps sound so clean. It doesn't take much to drive even a 300 W behemoth to clipping. If our aforementioned 100 Hz signal were driving the amplifier to full output at 50 peak watts, a state of affairs easily encountered on transient peaks (and remember, bass drums, tympani, and the like, have their fundamentals in this region) there is *no headroom to handle anything else.* And since real music contains more than single frequencies, this headroom really is needed.

Then too, this assumes a resistive load. If the load is reactive at any of the frequencies in question, then we may have to derate our amplifier, reducing the headroom even further. In fact, numerous tests have shown that even the biggest super amps are operating under clipping conditions a very high percentage of the time, especially on uncompressed signal sources.

The Fletcher-Munsen effect is characterized by a rise in the ear's response threshold as frequency is lowered. Consequently, at relatively low listening levels, including low performance volume, a higher proportion of the energy in a complex signal representing real music condition is concentrated in the lower frequency range, so a fairly large drop in level is required to produce a significant improvement in headroom.

ENERGY DISTRIBUTION

Just how serious is this problem in dealing with real programme, as distinguished from laboratory test tones? We all know that energy is not equally distributed throughout the spectrum in most music. Fig. 2 shows a generalized energy distribution curve for symphonic music, generally the most demanding type to reproduce (and if the Rock fans don't believe it, try one of the Brahms symphonies some time at concert hall level — if you can). Both Rock and Electronic music will contain higher amplitudes below 100 Hz and above about 3000 Hz, but the general distribution remains about the same. (See also ETI, March 1978 "About Equalization").

The very high levels encountered below 500 Hz, particularly between 500 Hz and about 150 Hz are obviously likely to produce the conditions described in Fig. 1. A very large part of the musical fundamental information appears in this region, which explains why it dominates the distribution curve, and, in conjunction with the bass region, is easily capable of driving even fairly powerful amplifiers into overload. In this area we also find large reactive components in the load, due either to driver or cross-over characteristics. Small wonder, then, that higher frequency fundamentals, as well as harmonics suffer so much mutilation. Obviously, we can solve this problem by simply acquiring sufficient electrical horsepower, even if it means bringing in a separate 100 Amp service from the power company to handle it.

But, cost and efficiency aside, will that really do the job?

SLEWED-UP TRANSIENTS

Some time in the future I shall probably do a piece on amplifier slewing and transient performance, including so-call Transient Intermodulation Distortion (T.I.D.), but for now a brief description of this characteristic will have to suffice. Basically, slewing refers to the rate at which an amplifier's output voltage will change in response to an input voltage step. Since a transient signal generally is very similar to the leading edge of a voltage step (including a square wave) slewing performance provides a good indication of transient response. (Fig. 3).

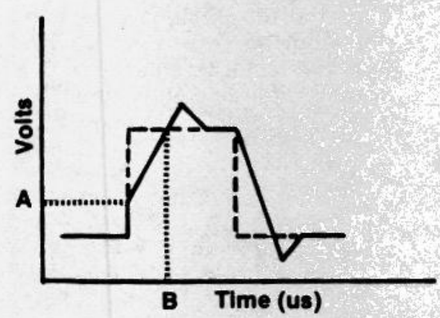

Fig. 3. Amplifier response to square pulse.

When a transient signal is applied to the input of the amplifier, output response is fairly fast up to some point determined by the amplifier's characteristics, beyond which a time delay occurs. Any attempt to pass a step at a faster rate will result in a slewing induced distortion, and it sounds pretty terrible. In other words, as long as either the transient signal's **amplitude** does not exceed "A" on the vertical, or amplitude co-ordinate of Fig. 3, or the **rise time** does not exceed "B" or the horizontal, or time co-ordinate, it will remain within the slewing capability of the amplifier. Now, you might think that 50 V/us is more than sufficient to handle the 0.8 V/us discovered as maximum slew requirements, by Peter Baxandall, but look what happens when the amplifier is already delivering signal at close to maximum output. It may already be delivering a voltage above that represented by point "A" of Fig. 3, and the introduction of a transient signal will then place demands on the amplifier which exceed its slewing capabilities. The kind of distortion which results sounds very much like hard clipping, and, in a sense, that's exactly what happens, even though the peak signal may not ordinarily result in a clipping condition.

The major disadvantage of using higher power amplifiers to overcome the problem is that in such equipment the output devices' characteristics make a major contribution to slew limiting, and the cure tends to be horrendously expensive when you get into the kilowatt class. Much of the amplifier equipment used with musical instruments in concert is in this power class and I don't think you have far to look to discover why they usually sound so horrible (aside from any musical judgements which might be made) in comparison with recordings.

SOMETHING FOR NOTHING

Let's go back to Fig. 1 and our two signals. If the 100 Hz signal is handled by one amplifier, the power output which would result from its 20 V level into 8 ohms would be 50 W, and if the 1000 Hz signal is handled by another amplifier, its 10V level would result in a power output into 8 ohms of 12.5 W, for a total of 62.5 W. Compare that with a power output of 112.5 W when the two are combined and we see some obvious benefits. By splitting the signal up this way there are no intermodulation products because the two signals do not appear in the same amplifier and so cannot produce IM. Even with today's devices, it is still easier and less expensive to design a really first rate low power amplifier than an equal quality high power unit. This is especially true with respect to slewing characteristics, which become more critically at higher frequencies, precisely the range to be handled by our lower powered unit.

Experiments have shown that fairly high levels of distortion at low frequencies can be tolerated in a complex signal if higher frequency components are reproduced cleanly; the higher frequency tends to mask the distortion components of the lower.

We are postulating amplifiers of different power ratings. Why? Well, if we examine Fig. 2 again we see that the output requirements in different pass bands are quite different, due to energy distribution. And there is an additional bonus. Drivers used for mid and high frequencies are usually more efficient than woofers, especially if the latter are designed as air suspension systems. So, if, in our hypothetical system the 1000 Hz signal is handled by a driver 10 times as efficient as the one handling 100 Hz, then our power requirement now drops to 1.25 W., all other conditions being equal!

IN PRACTICE

In actual practice, we are more likely to select each amplifier power on the basis of band-pass, driver efficiency, and driver power handling capacity, and it becomes a little more complicated than this brief run-down might suggest. But, with proper design, which really means designing the speaker, amplifiers, and cross-over as an integrated system, it is possible to achieve performance levels which would require power capabilities far exceeding the sum of the multi-amped system's amplifiers' outputs, and at considerably less cost.

Incidentally, one of the most common causes of tweeter burn-out is the very high level spikey wave-form with its high harmonic content which characterises the output of a single high power amplifier which is clipping. By eliminating this clipping, we are also protecting the tweeters.

Even the harmonics generated in the bass amplifier if driven to overload are generally not reproduced by the woofer, so even bass distortion is effectively reduced. Sub-woofing becomes a snap. The power levels required below, say, 50 Hz are more readily manageable when the sub-woofer has its own amplifier.

In fact, the same logic which dictates the use of multi-way speaker systems applies equally to the power amplifier portion of the chain.

But, more of this for a later column, with maybe some design examples.