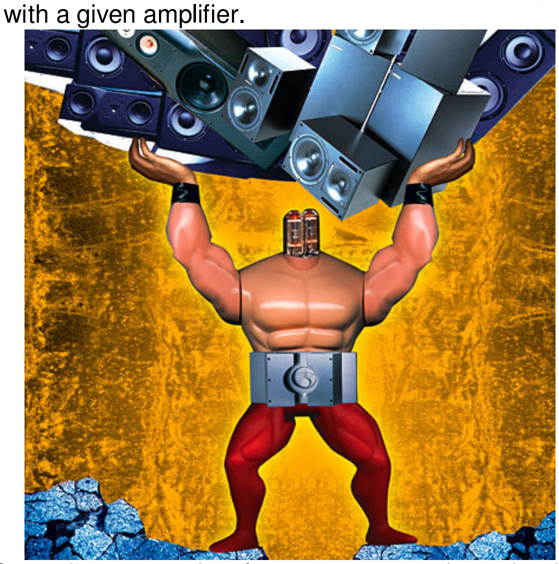
Heavy Load: How Loudspeakers Torture Amplifiers

By Keith Howard • Posted: Jul 29, 2007
http://www.stereophile.com/reference/707heavy/index.html#ywTIKYGqIfszYWSG.99

Why, in loudspeaker reviews, is impedance measured (assuming that the magazine in question bothers to measure anything)? Generally, for one principal reason only: to establish whether the speaker presents an "easy" or a "difficult" load to its partnering amplifier. In the design context, much more information can be extracted from a graph of speaker impedance vs frequency—such as details of the bass alignment, and indications of internal or structural resonances that can be difficult to identify by acoustical measurements. But for a magazine audience, the principal interest in a loudspeaker's load impedance lies in gaining some indication of its compatibility with a given amplifier.



So you may be surprised to learn that conventional measurements of speaker impedance don't do a very good job of it. This is not—as Matti Otala suggested in the 1980s (footnotes 1, 2)—because loudspeakers replaying music signals sometimes present impedance loads significantly less than indicated by conventional steady-state measurements. Although Otala and colleagues did demonstrate that you can contrive signals that will cause loudspeakers to behave this way, they never offered any evidence that such waveforms occur with significant regularity within music recordings.

That was left to others to check, and their results suggest that [*sigh of relief*] this is *not* a phenomenon with any practical relevance. That is what Dolby Labs' Eric Benjamin found when he investigated the issue in 1994 (footnote 3). It's what I found, too, when I unwittingly reprised some of Benjamin's work in 2005 (footnote 4), albeit using a software-analysis approach rather than an oscilloscope. While in this context you can't prove a negative—there is always the possibility that some pieces of music will contain just the waveform necessary for a particular speaker to demonstrate the Otala effect—the available evidence suggests that this probably occurs extremely rarely, if at all.

No, the problem with conventional impedance measurements lies not in the measurement method itself but the way in which its results are presented. To understand why this is the case, we need first to look at the amplifier output stage—here I consider only class-B transistor amplifiers—and what constraints the output devices have to work within if they are not to suffer catastrophic failure.

Out of bounds

Fig.1 shows the safe operating area (SOA) graph for a notional output device that might be used in an audio amplifier. Current (in amps) is plotted on the vertical axis and voltage (in volts) on the horizontal axis, the SOA being bounded along each by the device's absolute maximum current and voltage ratings, which must not be exceeded under any circumstances. For our device, these are 25A and 120V, respectively. The other boundary to the SOA, which accounts for most of its perimeter, is the device's maximum power dissipation; *ie*, the maximum product of current through and voltage across the device, which here is 125W. Note that this is very much less than we get by multiplying the absolute maximum ratings (25A *x* 120V = 3000W), which is why the SOA is shaped like a rectangle with a very large, hyperbola-shaped bite taken out of its top right corner—that is, provided the current and voltage scales are linear. In most transistor specification sheets the two axes are in fact logarithmic, in which case the same SOA looks like fig.2 (which has the advantage of making the SOA appear larger . . .).

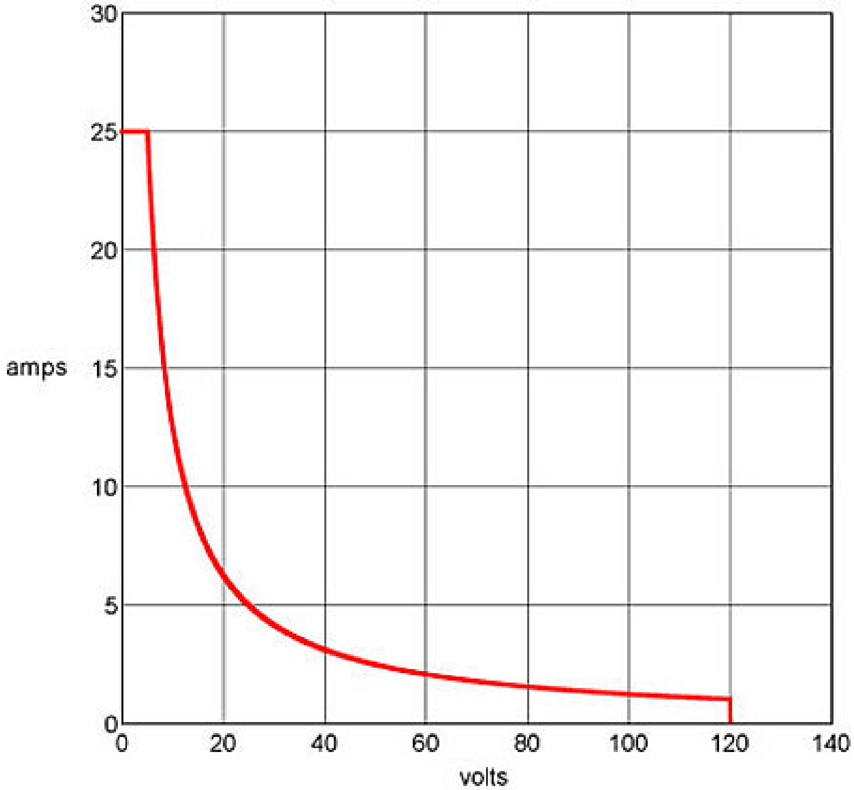


Fig.1 Safe operating area of a notional output transistor.

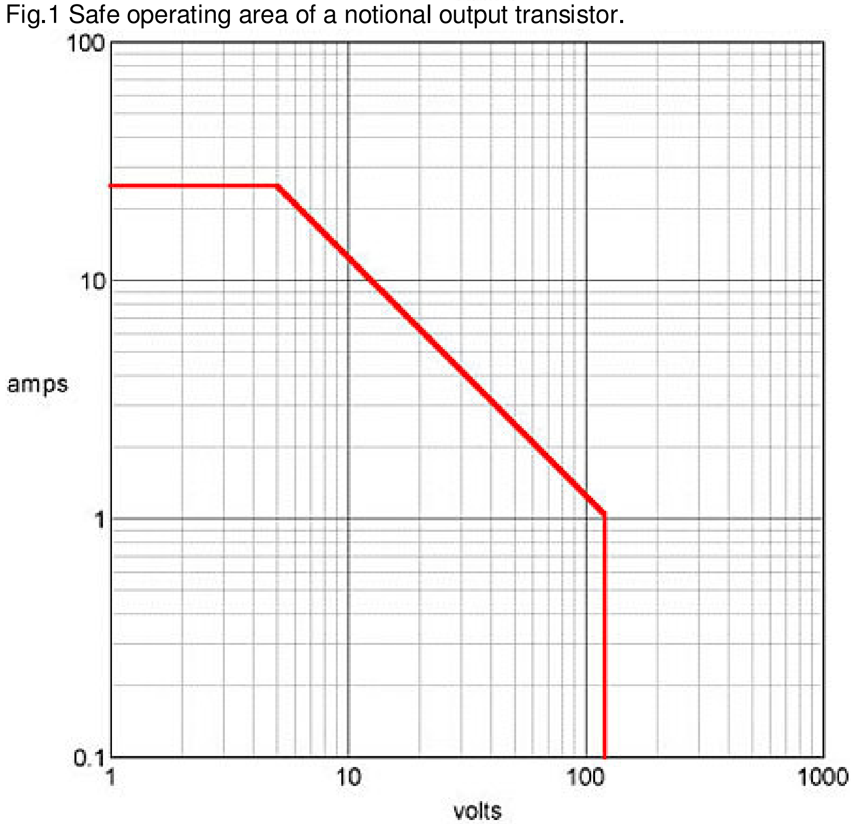


Fig.2 As fig.1 but with logarithmic axes, as typically used in transistor data sheets.

For various reasons, this is an oversimplification of actual SOAs. We haven't considered another limiting factor, *secondary breakdown*, which takes another bite out of the SOA; neither have we considered the effect of time. Many published SOAs show a number of superimposed graphs: one for DC conditions, as we've just considered, and others for shorter signal intervals, for which the SOA is somewhat larger. But no matter: our simplified SOA is quite good enough to illustrate the point that needs making.

Let's consider what happens when our notional output transistor is used, as one half of a class-B push-pull output stage, to drive a load. In the first instance we'll consider a resistive dummy load, as is usually employed for amplifier power measurements. If the dummy load has a resistance of 8 ohms, then at the transistor's 120V maximum the current will be (120/8 =) 15A. By joining these two points on the axes of the SOA graph we construct for this resistance a so-called *load line* (blue line in fig.3) that represents the combinations of voltage across and current through the device when driving this load. Immediately we see that this load line exceeds the SOA, and that the voltage across the transistor (and therefore the peak voltage deliverable to the load) must be reduced. In fact, it has to be cut to 63V in order for the load line (green) to just graze the limiting curve imposed by the device's power rating. If we now plot the equivalent for a 4 ohm resistor, the result is the pink load line, and the amplifier's rail voltage has decreased still further, to 44V.

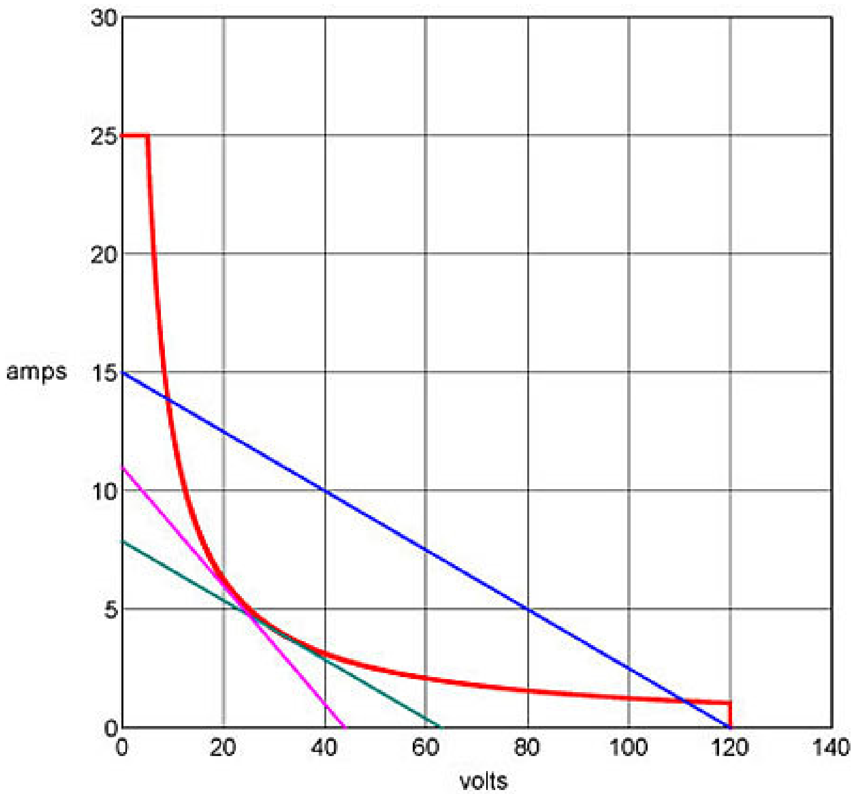


Fig.3 SOA of fig.1 with resistive load lines added (green line, 8 ohms; pink line, 4 ohms).

But loudspeaker loads are not, in general, resistive. They comprise complex impedances with both resistive and reactive (capacitive and inductive) elements, as a result of which the voltage and current waveforms are, at most frequencies, out of phase with each other. This is why, for a complete characterization of a speaker's load impedance, both modulus *vs* frequency and phase *vs* frequency have to be determined. Worst-case phase angles of 60° or more are not unusual in loudspeakers, so let's see what happens when we plot the load line for an 8 ohm load having a 60° phase angle, using the 63V rail voltage we determined for the resistive case. As shown in fig.4—where the current axis has been zoomed for clarity—what was previously a straight load line (green, as before) is now half an ellipse (orange trace), and this comprehensively busts the SOA budget. To prevent this, we have to reduce the rail voltage to 33V (purple trace), at which the load line again just grazes the SOA's limiting power boundary. Note that this is a more severe voltage reduction than is imposed by a 4 ohm resistive load. As far as our amplifier is concerned, then, an 8 ohm load with 60° phase angle is considerably more challenging that a 4 ohm load with 0° phase angle.

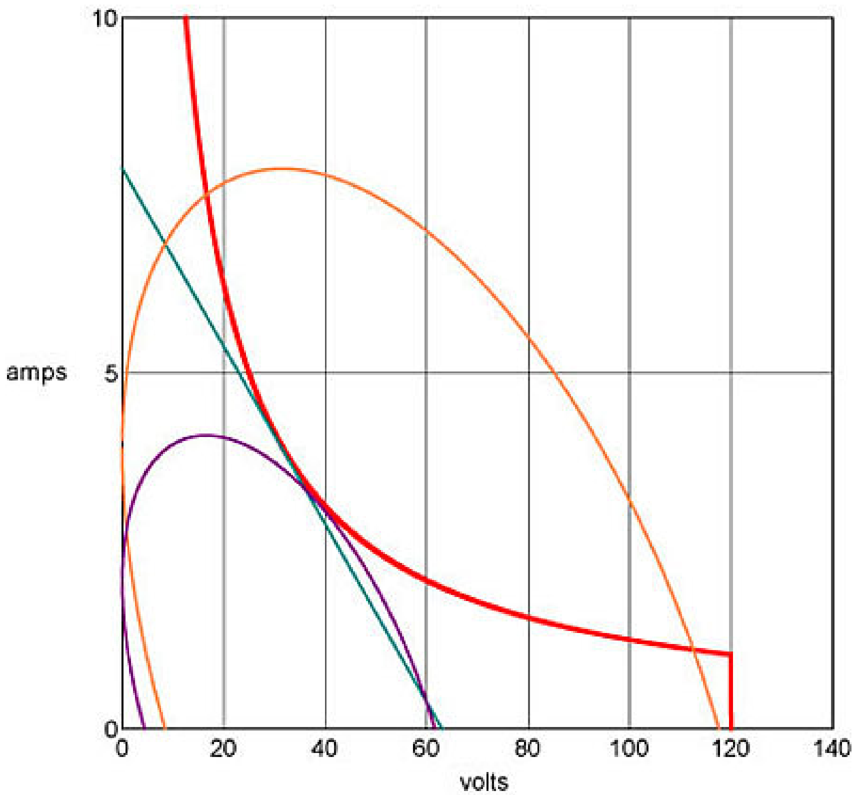


Fig.4 Same SOA as in fig.3, this time with zoomed current axis, showing elliptical load lines for an 8 ohm impedance with 60° phase angle.

Fig.4 makes the point that impedance phase angle is a critical factor when assessing, from the amplifier's point of view, how severe a load a loudspeaker is. But even if modulus *vs* frequency and phase *vs* frequency graphs are both plotted, we still don't get a clear picture of this, because what matters is the interplay of modulus and phase angle. Generally speaking, as we will see, it is not the frequency where the minimum modulus occurs at which the loudspeaker represents the severest amplifier load, nor will the amplifier be stressed equivalently to the speaker if we test it using a resistive load equal to the speaker's minimum modulus.

Footnote 1 M. Otala *et al*, "Input Current Requirements of High-Quality Loudspeaker Systems," AES 73rd Convention, March 1983 (available from www.aes.org).
Footnote 2 M. Otala *et al*, "Peak Current Requirement of Commercial Loudspeaker Systems," AES 79th Convention, October 1985 (available from www.aes.org).
Footnote 3 E. Benjamin, "Audio Power Amplifiers for Loudspeaker Loads," *JAES*, Vol.42 No.9, September 1994 (available from www.aes.org).
Footnote 4 K. Howard, "Current Affairs," *Hi-Fi News*, February 2006.

Part 2

All this was clear from Eric Benjamin's aforementioned AES article, but the message seems not to have filtered through in the 13 years since (footnote 5). Loudspeaker impedance continues to be assessed by considering modulus and phase separately when, as Benjamin showed, there is a much better way to reflect the load's severity from the amplifier's viewpoint. What he did—as Douglas Self did later (footnote 6), though by the less elegant means of SPICE circuit simulation rather than mathematical analysis—was to plot peak output-stage power dissipation *vs* frequency with respect to a stated resistive load. Benjamin chose 4 ohms as his reference, while Self preferred 8 ohms. Such curves can be calculated analytically from conventional modulus and phase data—no additional measurements are necessary.

Using this same technique to assess three loudspeakers I've recently measured for UK magazine *Hi-Fi News* gives the results shown in figs. 5–7. Note that each of these graphs has a different vertical scale. In each case, 1 on the vertical axis represents the output-device power dissipation for an 8 ohm resistive load, assuming a perfect class-B amplifier—that is, one able to deliver its full rail voltage to the load. Although no class-B amplifier is actually able to do this, some come close, and making this assumption allows for a worst-case calculation.

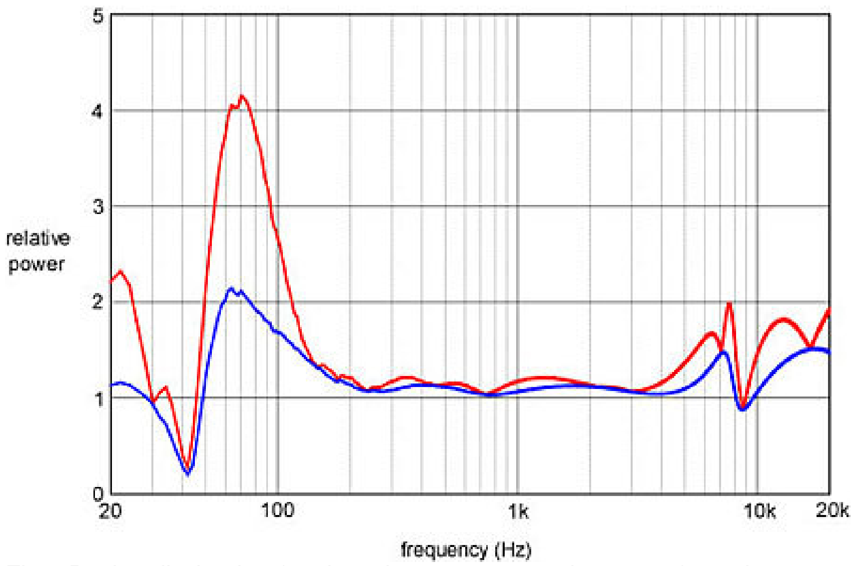


Fig.5 Device dissipation (peak, red trace; average, blue trace) relative to an 8 ohm resistive load for the JBL 1400 Array.

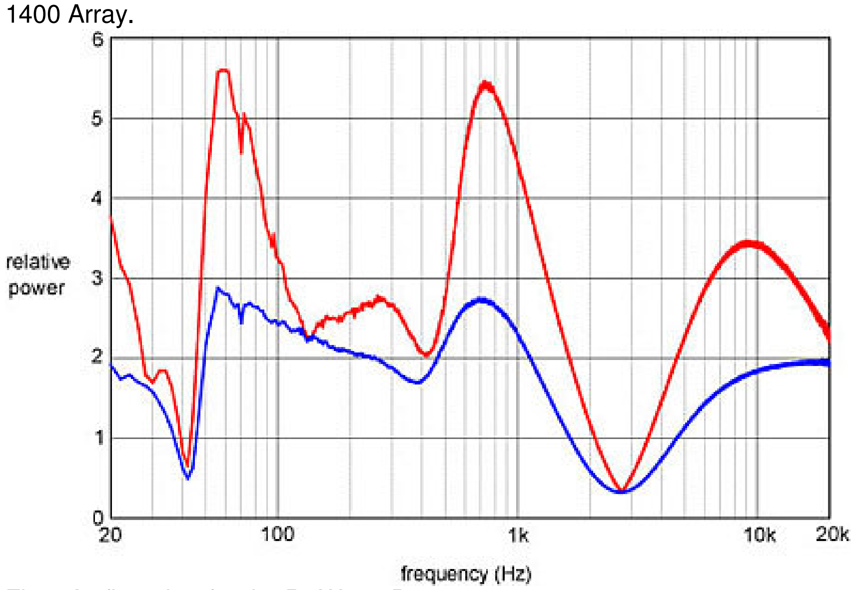


Fig.6 As fig.5, but for the B&W 802D.

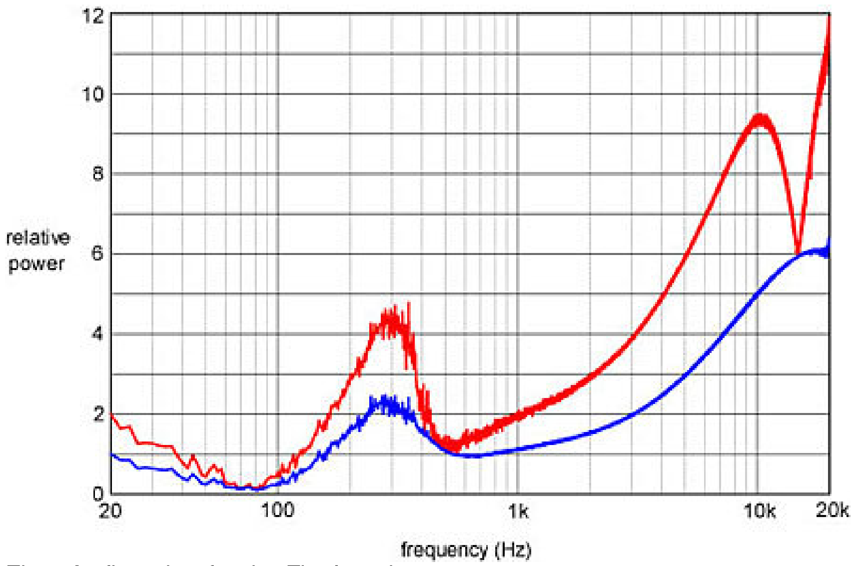


Fig.7 As fig.5, but for the Final 600i.

Fig.5 is for the JBL 1400 Array, fig.6 for the B&W 802D, and fig.7 for the Final 600i electrostatic—a dipole speaker of scarily low impedance at high frequencies. In each case, the red trace represents the peak instantaneous device dissipation, while the blue trace represents the average dissipation. It is the former, peak result that we are most interested in here as a measure of load severity; average device dissipation relates to heatsink provision and so is of interest primarily to the amplifier designer (although, since we consumers pay dearly for heatsink capability, it is not something for us to ignore).

Although both Benjamin and Self chose this means to display the device dissipation imposed by a loudspeaker, it suffers two problems. First, the scaling of the vertical axis depends on the chosen reference resistance; second, we are used to identifying the *minima* in modulus *vs* impedance graphs as potential problem areas, whereas here it is the *maxima* which are of concern. Both difficulties can be solved by plotting what I term equivalent peak dissipation resistance (EPDR) *vs* frequency, which inverts the graph and removes any need of a reference. EPDR is simply the resistive load that would give rise to the same peak device dissipation as the speaker itself. Adopting the EPDR view, the red traces of figs.5–7 become those of figs.8–10.

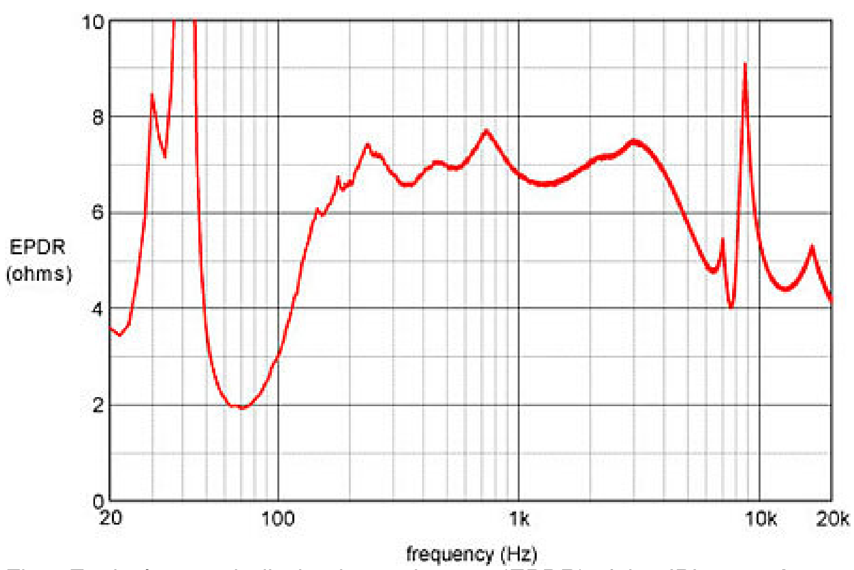


Fig.8 Equivalent peak dissipation resistance (EPDR) of the JBL 1400 Array.

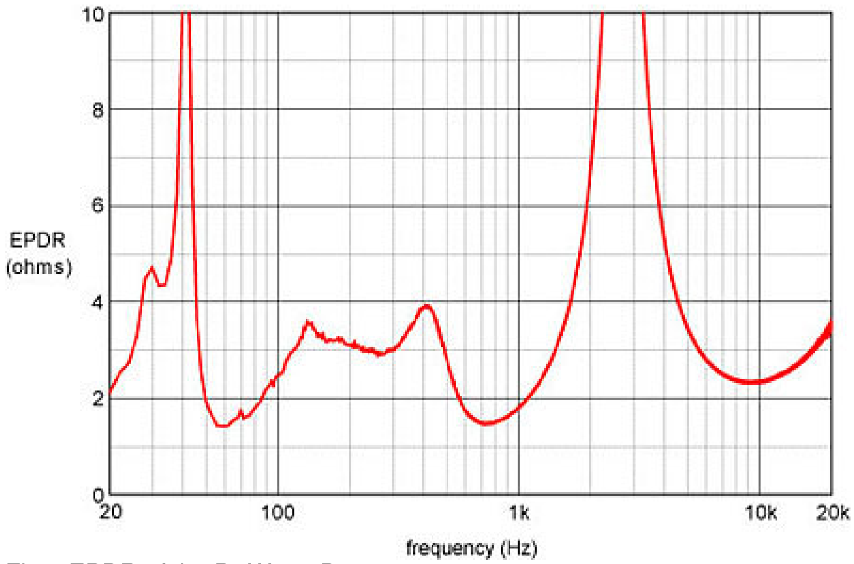


Fig.9 EPDR of the B&W 802D.

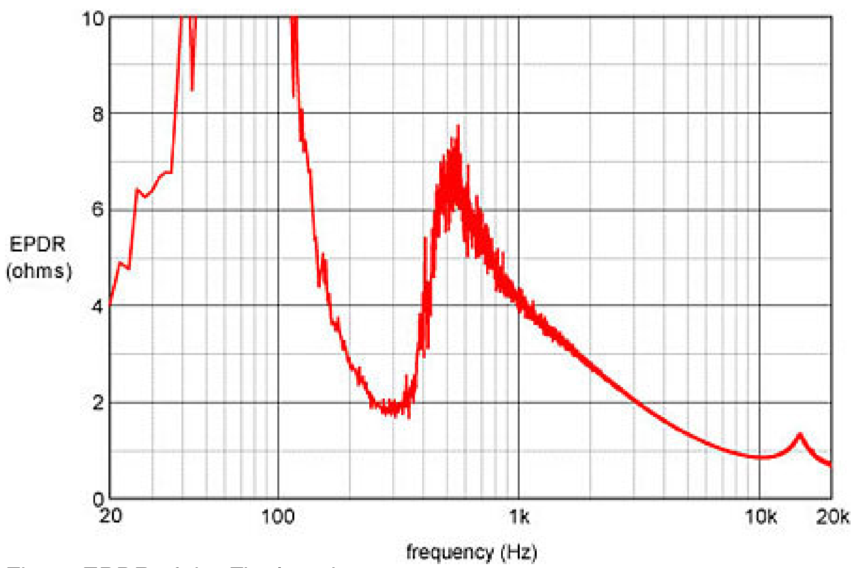


Fig.10 EPDR of the Final 600i.

The first thing to note from these is that all three speakers have EPDRs that dip below 2 ohms. In the case of the Final, the minimum is well below 1 ohm, albeit at very high frequency (and clearly continues to decrease above the measurement limit of 20kHz, suggesting that this is not a loudspeaker to use in a system whose CD player does not have effective image filtering). Moreover, these minima are significantly lower than indicated by each speaker's minimum modulus, and they occur at different frequencies. In fact, in dynamic speakers the minimum EPDR generally occurs quite close to *peaks* in the modulus curve that relate to driver fundamental resonances. The modulus here is still well above minimum but the phase angle is large, hence the low EPDR. (Self believed he was the first to show this effect, but in fact it was clear from Benjamin's work.)

Let's look at specifics, beginning with the JBL 1400 Array. Its minimum modulus of 4.9 ohms occurs at 92Hz. Its minimum EPDR is 1.9 ohms and occurs at 70Hz, where the impedance modulus is 6.1 ohms—almost 25% above minimum—but the phase angle is a high –52°. The B&W 802D, by contrast, has two pronounced EPDR minima: the first of 1.4 ohms at 58Hz, and the second of 1.5 ohms at 722Hz. Its minimum modulus of 3.2 ohms is over twice the EPDR minima and occurs at 86Hz. Because it is an electrostatic dipole, the Final 600i behaves quite differently. Its EPDR minimum of around 0.7 ohm occurs at the measurement frequency limit of 20kHz and is due to the speaker presenting a severe capacitive load at high frequencies. Again, the impedance modulus minimum of 1.3 ohms is roughly twice the minimum EPDR and occurs at a different frequency (15.4kHz).

Footnote 5 It has been on my "To Do" list since I first read the Benjamin paper to calculate a Figure of Merit, such as Keith's EPDR, for the speakers reviewed in *Stereophile*, based on their measured impedance modulus and electrical phase angle. But like all things that may be important but are not urgent, this project has so-far remained unrealized.—**John Atkinson**

Footnote 6 D. Self, "Speaker Impedance Matters," *Electronics World*, November 1997.

Read more at http://www.stereophile.com/content/heavy-load-how-loudspeakers-torture-So is this the real Otala effect we are looking at here? Do some speakers present more difficult loads than their impedance measurements suggest, not because of a failure of conventional measurement but simply because the data are not interpreted properly? To be certain about this, we must do what Otala and his coworkers failed to do: put the idea to the test using representative music signals. Eric Benjamin performed this crucial reality check and concluded that the dissipation behavior depicted in figs.5–10 *is* relevant to music signals. Let's see if we achieve a similar result using a method different from Benjamin's that has certain advantages. Once again, we can't expect to just stumble on the program material that is most effective at promoting extreme behavior with each speaker, but we can hope to form a good idea of whether or not EPDR is a significant factor in practice.

Doing it in silicon

There are two obvious ways to go about determining device dissipation with music program. First, we could simultaneously measure voltage across and current through either the loudspeaker or the output device—at an output level that will not cause the amplifier protection to operate—and directly calculate instantaneous device dissipation. This is what Benjamin did in his work. Alternatively, we can use a computer-simulation approach, which has various advantages. We can obtain a result faster than in real time (provided we use FFT convolution), and we can assess any speaker for which we have impedance data, even though that speaker is no longer to hand. (The latter is an attractive benefit; the review samples of all three speakers whose EPDR I've plotted were returned many moons ago.)

How we go about this simulation is to design a digital filter that will convert the signal voltage across the speaker into the signal current through it, so that we generate the same data as we would by measurement. To design this filter, we first have to calculate the inverse of the speaker's complex impedance—*ie*, its admittance—because voltage times admittance gives us the current. This frequency-dependent, complex admittance can then be converted into an FIR digital filter by applying the inverse fast Fourier transform (IFFT) to take us from the frequency domain to the time domain. The resulting filter is then applied, by convolution, to the input voltage (represented by the sample values in a WAV file) to generate the current waveform. The output-device dissipation can then be calculated sample by sample, and the result analyzed to see whether it is as high on music signals as our analysis of the impedance data suggests.

This process isn't quite so straightforward as just described because the sampling rate used to obtain the impedance data will generally not be the same as that of the WAV files we wish to process, so an adjustment must be made. In the case of the data used here—MLSSA impedance data measured using DRA Labs' supplied high-resolution setup script—the sampling rate is 65.57kHz and the output file contains 9995 data points from 2Hz to 20kHz. In order to use this to design an admittance filter, I extracted the first 8192 data points from the modulus and phase files, interpolated the 0Hz modulus, set the 0Hz phase to 0°, and then used these 8193 points to generate a 16,384-point filter. (We're not concerned with filter efficiency here, only accuracy.) To give the correct result, this filter was then applied to WAV files that had first been downsampled to (65.57kHz ˜ 2 =) 32,785Hz. As a result, the maximum signal frequency in the simulation is not 20kHz, as in the data, but 16.4kHz.

It's no accident that I've chosen as examples here three speakers whose EPDR behavior is significantly different. The JBL's EPDR dips to its minimum at low frequency; the B&W has prominent dips at both low and midrange frequencies, associated with bass and midrange system resonances, with a third, shallower dip associated with the tweeter resonance; while the Final's EPDR bottoms out at very high frequency, but also has a significant dip around 300Hz. Bearing in mind the typical spectrum of music, we might reasonably expect the B&W to present the most challenging amplifier load in practice, followed perhaps by the Final and then the JBL.

Four diverse music excerpts were chosen for the analysis, not entirely at random. I didn't have the luxury, as Eric Benjamin did, of assessing a large number of CDs and cherry-picking, so I selected source material with an eye to the EPDR results, hoping to choose recordings that would result in something like worst-case dissipation. Three were extracts from single-instrument recordings on the European Broadcasting Union's SQAM (sound quality assessment material) test disc, of flute, triangle, and soprano voice. The flute and soprano pieces have strong spectral content around 700Hz, which I hoped would probe the B&W 802D's midrange EPDR dip, while the triangle item has energetic HF to provoke the Final 600i. The fourth item—track 3 of Brian Bromberg's *Wood* (A440 Music 4001), a string-bass solo take on Lennon and McCartney's "Come Together"—was chosen for having strong midbass fundamentals where the 802D and JBL 1400 Array both have their lowest EPDRs.

Table 1 lists the highest device dissipations recorded for each channel of each track on each speaker, taking the worst-case dissipation into an 8 ohm resistance as the baseline. The second figure in brackets in each case is the equivalent EPDR. All the values fall within the ranges shown in figs. 5–7 and 8–10, and mostly they follow the expected pattern, the one surprise being the high values recorded for the Brian Bromberg track and the Final 600i. Taken together, these figures confirm that the orders of EPDR identified in figs. 8–10 are of real, practical significance when playing music signals: speakers really can make these high demands of amplifier output-device dissipation in normal use. If the amplifier's protection is invoked as a result, then its output will be clipped, even though the speaker's voltage and current demands may be within its capability.

Table One

Track	Channel	B&W 802D	JBL Array 1400	Final 600i
Flute	left	3.42 (2.34)	1.21 (6.61)	2.47 (3.24)
	right	4.03 (1.99)	1.21 (6.61)	2.59 (3.09)
Soprano	left	3.85 (2.08)	1.18 (6.78)	2.66 (3.00)
	right	4.71 (1.70)	1.17 (6.84)	2.37 (3.38)
Triangle	left	3.17 (2.52)	1.75 (4.57)	8.50 (0.94)
	right	2.71 (2.95)	1.54 (5.19)	8.38 (0.95)
"Come Together"	left	5.32 (1.50)	3.65 (2.19)	8.84 (0.90)
	right	4.78 (1.67)	2.88 (2.78)	8.70 (0.92)

How frequently do these extreme dissipation events occur? Some insight into this is given in Table 2, which shows the proportion of time that the dissipation factor fell within particular bounds for the B&W 802D on the left channel of the Bromberg track. For 5% of the time the speaker's EPDR is less than 4 ohms, and for 0.48% of the time below 2.7 ohms. So this is clearly a potentially significant effect with difficult source material. It's also obvious from these results why the B&W 802D has a reputation for being an amplifier ball-breaker.
image: http://cdn.stereophile.com/images/archivesart/707heavytab2.jpg

Table 2:

Peak Dissipation Ref. 8 ohm maximum	% of Track
≤ 1	77.02
>1, ≤ 2	17.48
>2, ≤ 3	5.00
>3, ≤ 4	0.48
>4	0.01

Heard before
All in all, there's little new here that Eric Benjamin's work didn't reveal 13 years ago. The EPDR concept is useful, I think. So is the simulation approach using digital filtering, since it allows results to be obtained more quickly, and with nothing more than conventional impedance modulus and phase measurements by way of input. But no excuse is necessary for reprising Benjamin's work, because its import seems not to have suffused audiophile consciousness. Speaker reviews don't address this issue, and neither do many speaker manufacturers, who are apparently happy to throw the output-device dissipation problem over the fence for amplifier designers to deal with. Jim Lesurf, who recently wrote about this issue for *Hi-Fi News* (May 2007, pp.100–102), jokingly postulated the existence of SCAMP—the Society for Cruelty to Amplifiers. If it existed, its membership would be thriving.

Part 3