Technical Notes Volume 1, Number 16

Power Ratings of JBL Loudspeakers and JBL/UREI Amplifiers

Introduction and Scope of this Technical Note:

JBL and UREI manufacture both loudspeakers and amplifiers, and it is the company's obligation to give the user guidelines for proper matching of these system elements. The task is not as easy as it would seem, because there are many modes of both loudspeaker and amplifier operation, depending on the specific application.

In this Technical Note we will examine both amplifier and loudspeaker rating methods in detail. Both tutorial and real world cases will be examined, and the reader should, after digesting this Technical Note, be able to choose the correct loudspeaker and amplifier power class for a given job. First, let us look at traditional amplifier and loudspeaker ratings.

Amplifier Power Specifications:

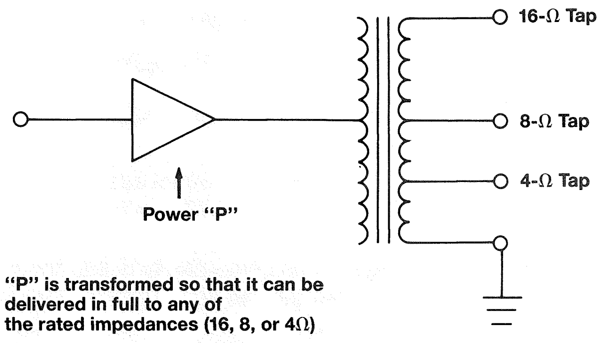
1. Basic Rating Methods:

A. Power Delivered to Nominal Load Impedances: In the days of tube amplifiers, the tradition developed of rating amplifiers by measuring the continuous average sine wave power the amplifier could deliver into a nominal load impedance, consistent with some maximum allowable level of distortion. As a general rule, loudspeaker impedances were assumed to be integral multiples or sub-multiples of 8 ohms. Life was made relatively simple through the use of output transformers, and this meant that the amplifier's power output capability was constant over the entire range of impedances (4, 8, and 16 ohms) – provided the correct taps were used on the output transformer.

Loudspeakers rarely present the ideal 4, 8, and 16-ohm loads which their specifications imply, but these errors were usually overlooked, inasmuch as the amplifiers were relatively tolerant of load variations.

It was an orderly world; the user would simply connect an 8-ohm loudspeaker rated at 50 watts to the 8-ohm output tap of a 50-watt amplifier, and that was all there was to it. The loudspeaker's rating of 50 watts implied that it could, over its operating bandwidth, safely handle the output power of a 50-watt amplifier, and few, if any, problems ensued. Figure 1 shows the rating method in detail.

Figure 1. **Maximum power into any design impedance.**



B. Maximum Voltage Swing into Minimum Load Impedance: With the advent of transistorized amplifiers, alternate rating methods have arisen. These amplifiers are normally operated without output transformers, so they cannot deliver full power to a wide range of loads. Such amplifiers usually have a maximum output voltage swing which they can deliver, and this value is related to the positive and negative voltages delivered to the output transistors by the power supply. This maximum output voltage swing can be delivered to a load as long as the maximum current rating of the output devices is not exceeded. Even momentary excess output current can cause instant failure of the output devices, and it is customary in transistor power amplifiers to incorporate some kind of internal current limiting as a failsafe feature.

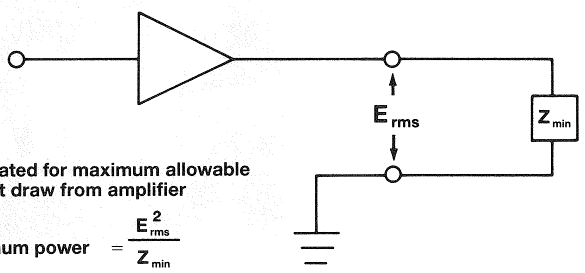
Thus, we can rate a transistorized amplifier's output capability by stating its maximum output voltage swing (in volts rms) and the minimum impedance across which this voltage can safely be applied. For a resistive load, the minimum resistance is given by:

$$R = (E_{rms})/(I_{rms})$$

where E_{rms} is the maximum voltage swing the amplifier can deliver, and I_{rms} is the limiting output current capability.

Taking into account complex loads such as loudspeakers, the amplifier's limiting current may be somewhat different than in the case of a purely resistive load. In the interest of completeness, this method of output rating would include both minimum resistive and minimum complex loads for the amplifier. Figure 2 shows the basis of this output rating method.

Figure 2. **Maximum voltage swing across mininum load impedance.**



Zmin rated for maximum allowable current draw from amplifier

Maximum power $= \dfrac{E_{rms}^2}{Z_{min}}$

Maximum current $= \dfrac{E_{rms}}{Z_{min}}$

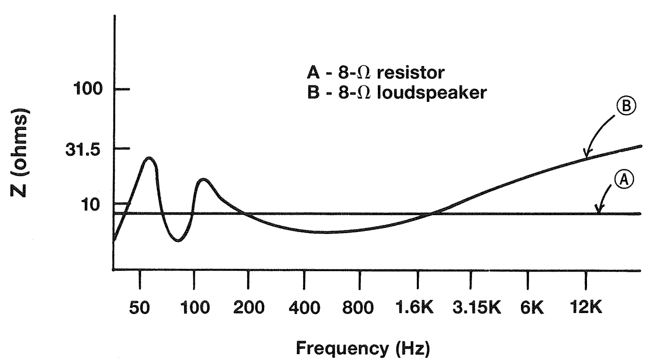
JBL and UREI have retained the traditional rating method for amplifiers, inasmuch as there is a wide universe of transducer hardware whose ratings are consistent with the older method. As an example, we will give the power output ratings of the JBL/UREI model 6290 stereo amplifier:

Rated Power (20 Hz-20 kHz)	
8-ohm (per channel)	300 watts
4-ohm (per channel)	600 watts
16-ohm bridged	600 watts
8-ohm bridged	1200 watts

What these ratings tell us immediately is that the same voltage swing at the output can be accommodated with either 8 or 4 ohms. The 4-ohm load will draw twice the current, and therefore produce twice the output power. What about operating the amplifier into a 2-ohm load? Some professional amplifiers carry such a rating — or, rather, a "derating." In our opinion this is risky, since the load will fall below 2 ohms at some frequencies. We would rather see designers maintain nominal loads no less than 4 ohms. Similarly, in the bridged mode the nominal load must be no less than 8 ohms.

2. Real versus theoretical loads: An 8-ohm resistor provides a constant load over the frequency range, while a loudspeaker does not. There are no simple loudspeaker loads; they are all reactive to some extent, and a typical example is shown in Figure 3. This curve shows the steady-state magnitude of impedance over the frequency range of interest.

Figure 3. **Resitive and loudspeaker loads.**



When we label Curve B as "8 ohms," we are implying that the average load is somewhere around 8 ohms. At resonance peaks the value is considerably higher, while at some points in the mid-band it is a little lower. The general implication however, is that on music or speech program the average load seen by the amplifier will be about 8 ohms.

But the real situation can be more complex. Recent studies (1) have shown that complex loudspeaker loads can, under specific transient drive conditions, produce dynamic loads which can be as low as one-half the steady-state minimum value! Thus, a nominal 8-ohm low-frequency loudspeaker with an impedance minimum of, say, 6.2 ohms, may, under the right drive conditions, present a momentary dynamic resistive load to the amplifier of 3.1 ohms.

Such factors as these are rarely considered *per se,* and they are often responsible for triggering current limiting in amplifiers in systems where conventional design wisdom has indicated that no problems exist.

Since it is virtually impossible to specify an amplifier's input signal, the only way to guard against the adverse effects of dynamic load variation is to design into the system the capability of coping with one-half the steady-state minimum impedance.

Thus, if an 8-ohm load presents a steady-state minimum value of 6.2 ohms, it should be powered by an amplifier capable of handling a load as low as 3.1 ohms, with current capability corresponding to the maximum rated output voltage of the amplifier. This is an extreme requirement, and not many systems have been designed to satisfy it. Many amplifiers are audibly unstable under such operating conditions, while others may handle such transient signals in stride. The reader should be aware that high-efficiency loudspeaker systems, such as those generally used in sound reinforcement work, present the greatest load variation to an amplifier, due to the relatively high back EMF these loudspeakers produce.

The situation becomes even more critical when stereo amplifiers are bridged for mono application. In this mode of operation, the two amplifier sections are operating in series, and the output voltage is effectively doubled, making it possible for the amplifier to deliver large currents. Specifications for bridged operation will clearly indicate the minimum load impedance and the maximum safe power which can be delivered to that impedance. It is important that such precautions be carefully followed.

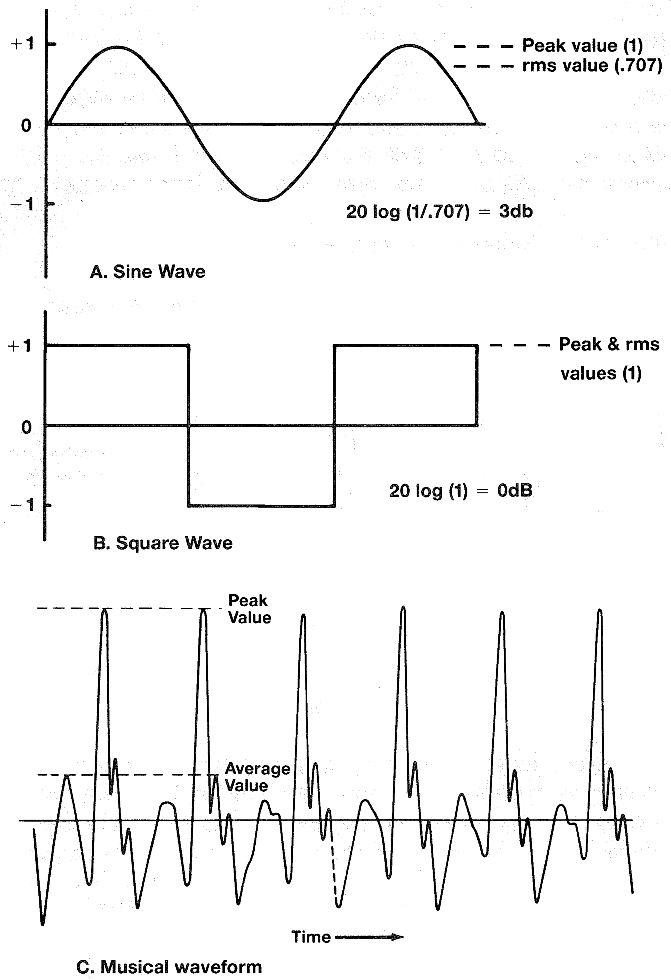
The Onset of Distortion in Amplifiers:

In many professional applications it is difficult to ensure that an amplifier will never be driven into distortion. It is thus essential that we know how a given amplifier will recover from momentary overloads and what the audible effects of these may be. Certain kinds of speech and music signals may be subjected to clipping of the waveforms with little or no audible effect. The key here is the crest factor of the signal.

1. Crest factor:

Crest factor is the ratio of the rms value of an audio waveform and its peak value. As shown in Figure 4 the crest factor of a continuous sine wave is 1.4, or 3 dB. The crest factor of a square wave is unity, or 0 dB. Many audio signals

***Figure 4.* Crest factors in audio signals.**



may have crest factors in excess of 10 dB, as shown in Figure 4C. Full band audio signals usually have fairly high crest factors, while bi- or tri-amplified signals, due to frequency division and consequent narrower bandwidth signals, generally have lower crest factors.

The designer can easily specify a sound system which will never be driven into amplifier distortion, but the cost to the user may be prohibitive. For example, let us assume that a 50-watt amplifier will handle the anticipated average power requirements for speech program in a reinforcement system. Now, in order to accommodate crest factors in speech of 10 dB, an amplifier capable of 500 watt peaks will have to be specified. To accommodate a crest factor of 12 dB, an amplifier capable of 800 watt peaks will be required.

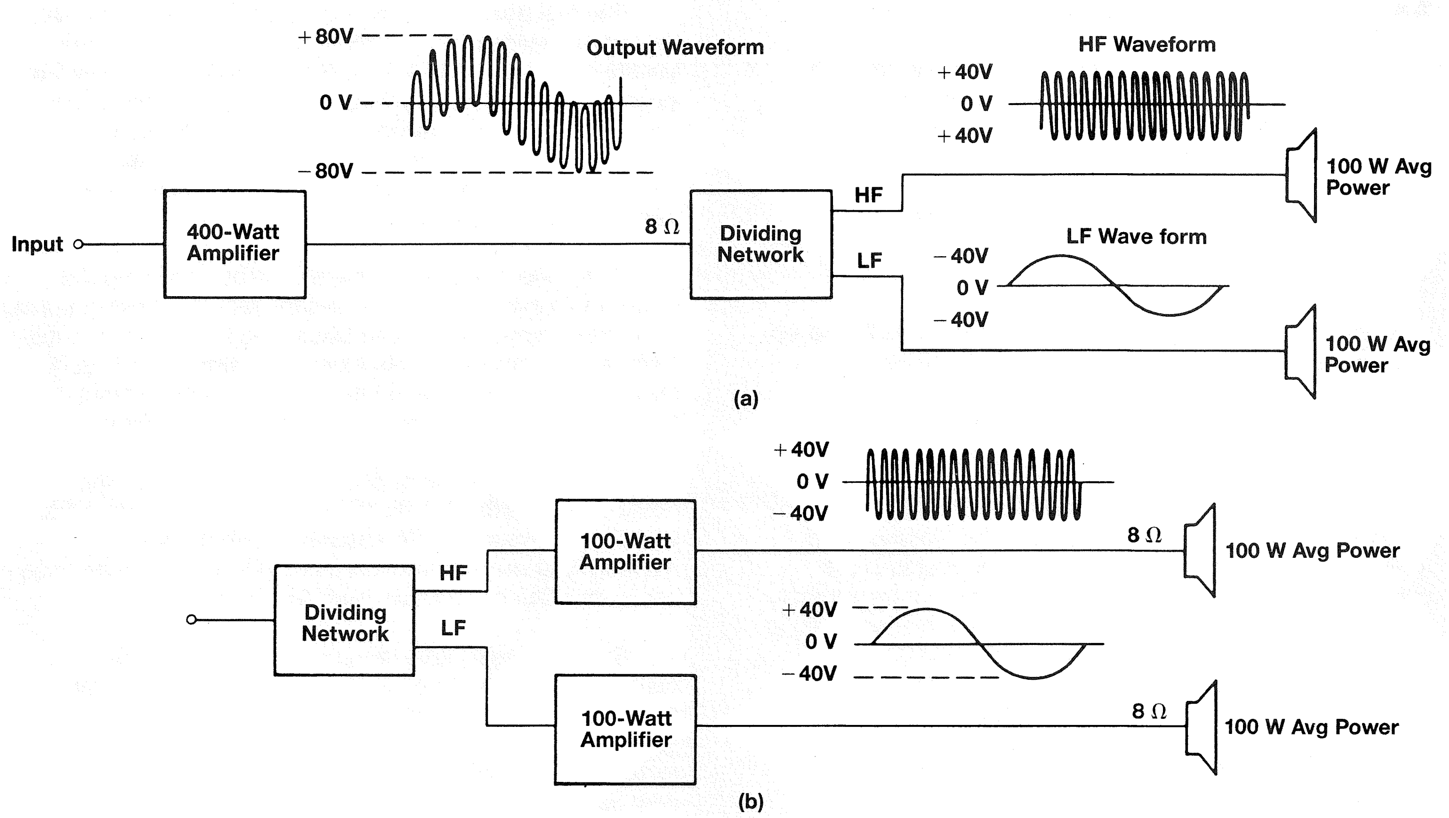
While such ampiifiers can be reasonably accommodated in modest size speech reinforcement systems, it is clear that many large music reinforcement systems will pose real economic problems. For example, assume that a music reinforcement system has 10,000 watts allocated to its high-frequency requirements and that the design crest factor is 3 dB. Simply raising the crest factor, or headroom, of the system to 6 dB would call for 10,000 additional watts.

Obviously, some kind of compromise is needed, and the following points are normally addressed:

A. Signal limiting. (Simple limiting can gauge the signal fed to the power amplifiers so that overload or clipping can be held to a fixed minimum.)

B. Bi- or tri-amplification. (Watt for watt, this approach is more effective than simply using larger full-range amplifiers in maintaining a given crest factor capability. See Figure 5 for added details.)

***Figure 5.* Principle of bi-amplification.**



C. Use of highly stable amplifiers. (Some amplifiers recover from internal voltage clipping and current limiting less audibly than others. All else being equal, these are preferred for professional audio applications.)

2. Amplifier failure modes:

The majority of power amplifier failures are due to burnout of the output transistors. Power transistors have both voltage and current limits they can safely handle, but the ratings are determined statistically for a given type of transistor. There may thus be variations from transistor to transistor, but these are usually sorted out and corrected during the "burn-in" testing phase of each power amplifier during manufacture.

In the amplifier design phase, the positive and negative operating voltages for a given type of output device are noted and not exceeded, and internal current limiting is carefully adjusted. Even then, random failures may be encountered in the field simply because of component variation, even though the amplifier has not been subject to particularly harsh operating conditions.

Highly reactive loads may be a problem, because they result in current flow at the amplifier's output, even when the output voltage is zero.

The heat generated by the power amplifier must be drawn away from it. Convection cooling is normally used in moderate size amplifiers, while forced air cooling is required on many large power amplifiers. Improper mounting of convection cooled amplifiers can hamper their cooling, and manufacturer's instructions should be carefully followed.

Transducer Power Ratings:

A transducer can fail basically in one of two ways; it can be heated to the point where it literally burns up; or, it can be mechanically stressed to the point where it is permanently rendered inoperative. The two failure modes are generally independent of each other; however, under certain operating conditions, one mode may aggravate the other.

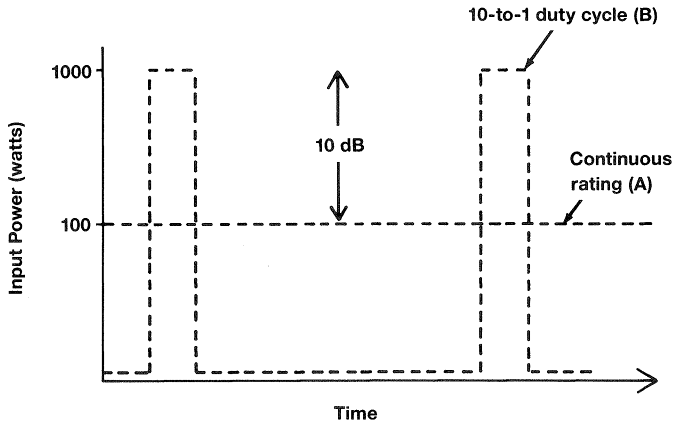
1. Thermal limits:

Under continuous operation, a loudspeaker will heat up to some temperature and remain at that temperature in a state of thermal equilibrium. Heat is removed from the device at the same rate it is generated internally.

Voice coil resistance is the major source of heating, and under normal operation the loudspeaker can withstand temperature rises up to the design limits of the materials and adhesives used in its construction. Temperatures in the range of 200 degrees C (400 degrees F) are not uncommon. When heated, the voice coil resistance rises, bringing with it a host of performance compromises. The resistance rise, however, acts to protect the transducer to a limited degree and does not materially affect the transducer's power rating. Interested readers are referred to Technical Note Volume 1, Number 9, for a discussion of power compression in low-frequency transducers.

Because the internal temperature of the loudspeaker does not rise instantly under sudden current surges, a duty cycle may be established for the device which allows it to see greater input powers than its rated continuous input — for proportionately shorter periods of time. At Figure 6A, we show continuous power input of 100 watts to a transducer whose thermal rating is 100 watts. At 6B, we show the device powered one-tenth the time with ten-times the power, or 1000 watts. The duty cycle here is ten-to-one.

Figure 6. **Loudspeaker duty cycles.**



Most loudspeakers are capable of such operation — as long as the on-cycle is short enough that the voice coil temperature rise is insufficient to cause damage, and the off-cycle is long enough to ensure that sufficient cooling can take place. It is also essential that the bursts of input power be at a sufficiently high frequency so that the transducer is not mechanically stressed in the process.

2. Determining the thermal limit of a transducer:

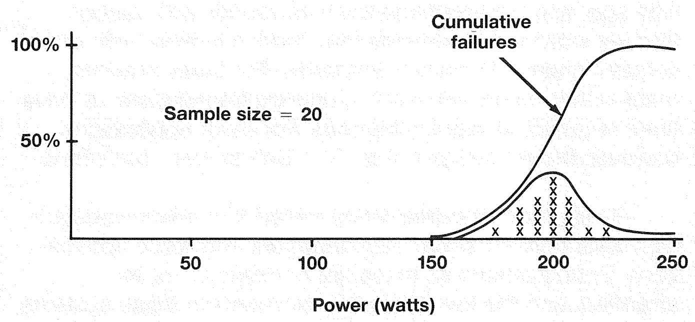
Thermal power limits for a given transducer model are determined statistically. A sufficiently large sample of the transducer in question will be constructed and subjected to increasing input power, typically in 1 dB level increments and allowed to reach thermal equilibrium after each increase. The applied signal is shaped pink noise, and the spectrum is controlled so that the transducers are not unduly stressed mechanically during the entire process.

Power input is carefully monitored by observing rms values of voltage and current, and the test is carried out until all of the samples experience failure due to the burnout. It is clear that not all of the transducers will burn out at exactly the same power input, and what the test usually shows is a normal distribution, or "bell curve," as shown in Figure 7.

In this example, most of the transducers fail in the range of 200 watts. Should this transducer be rated at 200 watts? Not necessarily. To ensure field reliability the manufacturer may want to rate the model at 175 watts since nearly all of the sample can meet this rating.

Such a conservative rating may be good for the manufacturer in the long run, since the demanding user will generally be satisfied. However, in the advertising ratings game, the manufacturer may be at somewhat of a disadvantage when competitive transducers are compared on a unit cost basis.

Figure 7. **Distribution curve for power failure.**



3. Factors affecting thermal capacity of transducers:

There are two key points here: materials must be capable of withstanding high temperatures, and heat must be removed as efficiently as possible. While the search for high-temperature adhesives is ongoing, the effective removal of heat remains the major design challenge for transducer engineers everywhere.

Henricksen (2) describes in detail a number of techniques for maximizing heat sinking away from voice coils. In general, large diameter voice coils dissipate heat better than smaller ones, since there is more wire in the voice coil, and a greater area of the coil is exposed to the top plate and pole piece of the magnetic structure.

We often see fins molded into the outer portion of the magnet structure. This gives the impression that there is significant convection cooling effort under way, but such is rarely the case.

4. When do thermal ratings apply?

The thermal power rating of a transducer applies any time it can be assured that voice coil motion will be restricted to its linear range. In low-frequency applications, this implies that the transducers are mounted in properly ported or horn-loaded enclosures and that their response is electrically rolled off below enclosure resonance or horn cutoff.

For mid- and high-frequency cone transducers the nature of program virtually ensures that voice coil excursions will be small enough to be quite linear. For high-frequency compression drivers excursion limits must be carefully noted, and the high-pass slope should be at least 12 dB/octave. In cases of bi-amplification it will be essential to place a blocking capacitor in series with the driver in order to avoid the effects of very-low-frequency turn-on transients. These can easily destroy a compression driver diaphragm by causing it to undergo very large excursions.

Since compression drivers are normally padded down relative to low-frequency system requirements, a resistor, normally equal to three times the nominal impedance of the driver, can be shunted across the driver, affording added protection by shunting the reactive components of the impedance peaks below horn cutoff.

5. Displacement power ratings:

For constant applied voltage the displacement of a cone will double for each halving of the driving frequency. A limit occurs at the resonance of the driver, below which the displacement will remain constant for all frequencies.

If an unbaffled low-frequency transducer is driven with a signal in the range of its primary resonance, its displacement can be quite large. If driven with a power input approaching its thermal rating, damage may be done due to mechanical stresses. The suspension may be deformed, and the voice coil made to rub against the pole piece or top plate. Once these conditions occur, the transducer is likely to fail abruptly.

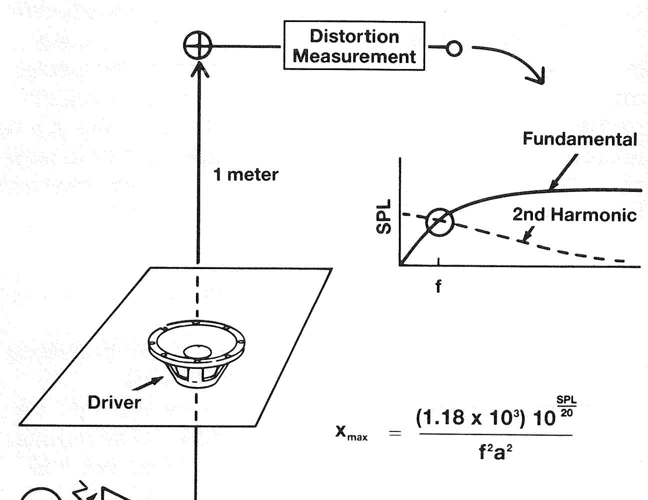
Thus, the transducer designer must impose certain power drive limits on low-frequency drivers based on the frequency of the input signal. But long before the transducer has reached the danger point in its excursion, it will have reached an undersirable level of distortion, and it is this which we want to identify as a displacement limit for the transducer.

By general consent in the professional sound industry, a departure from voice coil motion linearity of 10% is considered an upper limit for acceptable system performance. Thus, for each model driver we must identify the maximum excursion, in each direction from the rest position, at which the nonlinearity of the moving system has reached 10%.

There are several ways of doing this. Some manufacturers measure current through the voice coil, noting the peak amplitude of the cone at which distortion in the current waveform has reached 10%. This is the method used in the AES standard (3).

JBL uses a more complex method, one relating more clearly to actual use of the transducers in the field (4). The method is shown in Figure 8 and is described as follows:

Figure 8. **Calculating X_{max}.**



$$X_{max} = \frac{(1.18 \times 10^3)\, 10^{\frac{SPL}{20}}}{f^2 a^2}$$

Where: f = frequency of 10% distortion
a = cone radius (mm)

The low-frequency transducer is mounted in a large sealed baffle, and the measurements of fundamental, 2nd, and 3rd harmonic distortion are made. In JBL's usual data presentation, the distortion components are raised 20 dB as shown on the graph. Thus, at any point on the graph where either distortion curve crosses the fundamental, we know that the actual distortion is 20 dB lower, or one-tenth the amplitude of the fundamental. This corresponds to 10% nonlinearity, and we can find the corresponding cone displacement by solving the following equation:

$$X_{max} = \frac{(1.18 \times 10^3)\, 10^{\frac{SPL}{20}}}{f^2 a^2}$$

where: SPL = level of fundamental where distortion is 10%
f = frequency of fundamental where distortion is 10%
a = radius of cone in mm

The limiting excursion may be defined by either 2nd or 3rd harmonic, and it is important to note the smallest excursion value at which the 10% limit has been reached.

6. Dependence of cone excursion on enclosure parameters:

Low-frequency cone excursion is highly dependent on enclosure parameters. Specifically, properly designed ported enclosures tend to minimize cone excursion in the octave above the enclosure tuning, and it is the goal of ported systems to allow the low-frequency transducer to maintain its thermal power rating down as low as the enclosure cut-off frequency. Below this frequency it is essential to roll off the signal applied to the system, since the "unloading" of the enclosure may lead to excessive cone motion.

All JBL systems, whether low-frequency or full range, have been carefully designed to minimize cone excursion, and the published input power ratings take into account X_{max} as well as thermal limits.

In recent years JBL has adopted what are referred to as progressive suspensions in low-frequency transducers. Paradoxically, their design involves the use of non-linear inner suspensions which work in opposition to the electromagnetic nonlinearity resulting from driving the voice coil partially outside the magnetic gap. The result of this is a net reduction of distortion and an overall improvement in large-signal transducer performance. Details of this are shown in Technical Note Volume 1, Number 9.

7. Failure modes of low-frequency systems:

It is clear that in high-level operation a low-frequency transducer may be stressed both thermally and mechanically, and under extended operation there will be some interaction between the two possible failure modes. Specifically, excess heating of the voice coil causes it to expand, or possibly buckle, and this may aggravate mechanical rubbing of the voice coil against internal parts of the magnetic structure. This in turn can lead to electrical shorting problems between adjacent voice coil turns, and then to electrical failure.

While all low-frequency transducers will alter their performance characteristics over time, most of those installed in well-designed theater and studio systems will continue to perform almost indefinitely. Cone materials may break down with age, and polyurethane foam surrounds and various viscous surround treatments may harden in time, with consequent change in system response. For these reasons, many critical users will want to change transducers, or have them reconed, at regular intervals. For most applications, however, the old adage holds: "if it isn't broken, don't fix it."

By far, most low-frequency transducer failures occur in high-level concert sound reinforcement and disco applications. Such systems as these are normally bi- or tri-amplified, and the low-frequency elements in these systems, as we noted earlier, receive signals with relatively low crest factors. This implies that the transducers are working "full out" most of the time, exercising both their thermal and mechanical capabilities. Often, the low-frequency transducer complement in such systems may be operated very close — often too close — to actual design limits, and any carelessness on the part of an operator may push the system into real stress. The onset of amplifier clipping lowers the crest factor even further, and operators have been known to drive subwoofers virtually with square waves!

Analysis over the years at JBL's warranty service centers confirms that voice coil burnout is the major failure mode of low-frequency transducers. Next in order is separation of voice coil from its former due to failure of adhesives operating at temperatures beyond design limits.

8. Mid- and high-frequency devices:

Because voice coil excursions are not large with mid- and high-frequency drivers, the predominant failure mode is voice coil burnout. Below the range of effective horn loading, the voice coils in compression drivers may be subject to high excursions, and fracturing of the dome-voice coil structure can occur. Materials such as phenolic impregnated linen cloth and titanium are less prone to fracture than other materials. Aluminum, in particular, simply "grows old" after long periods of use and fractures, even when there has been no particular stress given to it.

9. Giving the transducer a meaningful power rating:

The dilemma for the manufacturer is to find a rating method, or rather a specific test signal, which gives realistic guidelines to the professional user — and yet is not so conservative as to put the manufacturer at a disadvantage relative to competitive ratings. The manufacturer's prime responsibility here is to the professional user, and "specmanship" for its own sake is, in the long run, good for no one.

JBL's power ratings have traditionally been realistic ones, and measurement methods have been consistently applied over the years. We will now explain in detail how these ratings are determined:

Program and sine wave ratings: A low-frequency transducer carries a continuous program rating, and the specification sheet defines this 3 dB greater than, or twice, the continuous sine wave power rating of the transducer.

Taking an example, the model 2225 transducer is given a rating of 400 watts, continuous program, and there is the following qualification: "Continuous program power is defined as 3 dB greater than sine wave power and is a conservative expression of the transducer's ability to handle typical speech and music program material."

The corresponding sine wave rating is thus 200 watts, and that rating has been rigorously and conservatively determined by JBL design engineering and certified by production quality assurance on samples from each production run. It would thus be quite safe to operate each 2225 in a system with 400 available watts if non-distorted operation were ensured. Actually, in studio monitoring applications, it would be safe to power each 2225 with 800 watts, assuming that operating levels will be carefully monitored. In a later section of this Technical Note we will present specific case studies, outlining for the reader all the pertinent calculations and assumptions which must be made in properly matching loudspeakers and amplifiers.

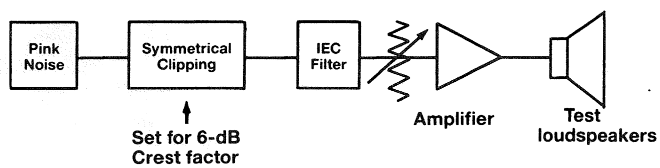
10. AES testing method (3):

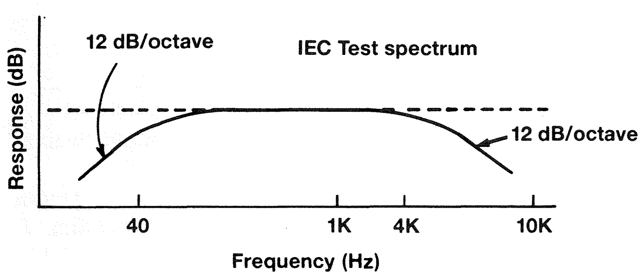
The AES (Audio Engineering Society) has issued testing procedures for components used in sound reinforcement work, and for low-frequency transducers a one decade wide band of pink noise (6 dB crest factor) is applied and incremented upward in power. The lower limit of the noise decade is the lowest frequency the manufacturer recommends for the transducer in question. While such a test makes it possible to compare similar transducers from several manufacturers, it does not relate at all to how these devices will perform in typical ported enclosures. Thus, it is not a practical test in the sense of providing the user guidelines in systems engineering.

11. Power ratings for systems:

JBL has adopted the IEC (International Electrotechnical Commission) method for rating assembled systems with passive dividing networks (5). This method is shown in Figure 9.

***Figure 9.* IEC power measurement.**





Note that the pink noise signal is symmetrically clipped so that it maintains a crest factor of 6 dB. This means that when the average power applied to a test system is, say 100 watts, the instantaneous peak power applied to it is 6 dB greater, or 400 watts. Thus, the test relates well to real world conditions. Power is incremented upward and the system allowed to reach thermal equilibrium after each increase. Rated power for the system is the maximum power the system can accommodate for two hours with no permanent change in response.

12. Power ratings for compression drivers:

JBL compression drivers are tested according to the AES standard for that device class. The test uses a pink noise signal (6 dB crest factor), limited at the low-frequency end of the range by a sharp cutoff filter of at least 12 dB/octave slope. Power input to the driver is calculated according to the minimum value of impedance over the operating bandwidth. Continuous program ratings are defined as three dB greater than the pink noise rating. Since the signals which can be accommodated produce large diaphragm excursions, we will see such ratings as follows:

"70 watts continuous program above 800 Hz"
"100 watts continuous program above 1.2 kHz"

This is the power rating given to the 2426 small format compression driver. While the difference in the 800 Hz and 1.2 kHz ratings is only 1.5 dB, it is an important consideration in system design and should be carefully observed. If the manufacturer recommends a minimum crossover slope for a given mode of operation, that value should be specified or exceeded.

Real World Applications:

By now, the reader has gained an appreciation of how complex both amplifier and loudspeaker power ratings may be, taken individually. When we put them together, things become more complex yet, and the situation will simplify itself only when we go back to basics in the analysis of system requirements.

The following steps are essential:

For transducers, determine:

a. Power bandwidth requirements
b. Acoustical headroom requirements

For amplifiers, determine:

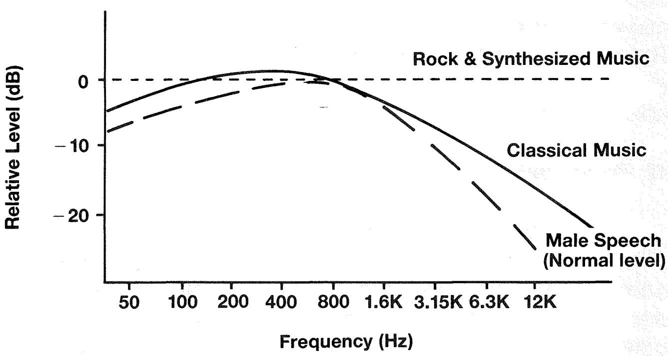
a. Single versus bi- or tri-amplification
b. Electrical headroom requirements

1. Selecting the right loudspeaker:

All matters of proper coverage aside, the designer must ensure that the class of components chosen for a given job can produce the needed acoustical levels over the required bandwidth. While the system may be equalized so that it is substantially flat in its direct field response, there may or may not be the requirement that it be capable of delivering the same maximum level at all frequencies.

Power bandwidth is the consideration here, and the curves shown in Figure 10 indicate the typical requirements for various types of program.

Figure 10. **Power bandwidth requirements.**



The designer must first determine the kind of spectrum to be reinforced. Then, calculations are made to determine average acoustical levels and the absolute peak requirements for headroom over these average requirements. Calculations may have to be made for several parts of the frequency range in order to cover all requirements.

These decisions will determine the class and number of transducers which will be required to do the job. At the same time, considerations of coverage will dictate the number of components required.

2. Selecting the right amplifier:

The first important decision is whether to go with a single amplifier or to bi-amplify the system. At the present time, there is so much general favor for bi-amplification that it has become the normal design method. Only the smallest speech reinforcement systems and low powered distributed systems should be candidates for single amplification.

Power requirements are then calculated to produce the desired average acoustical levels in the space, and then sufficient electrical headroom is provided, at a very minimum, to accommodate the acoustical peak requirements. Many designers will increase the electrical headroom margin so that it is at least three dB greater than the acoustical peak requirement. This will ensure that the electrical system has reserve margin so that undistorted operation can generally be maintained. (It also implies that there is the potential for damaging the acoustical components if the system is improperly operated.)

3. Useful signal processing:

In addition to equalization, filtering, and time delay, compressors and limiters are useful adjuncts in system design. A compressor can be placed in a signal chain in such a way that it will protect the system from prolonged signals that could cause damage. At the same time, depending on the design of the compressor, short term peaks that will do no damage can be passed.

4. Some design examples:

In working out the following examples, we will solve only for direct field levels, ignoring for the most part the effects of reverberation. Our answers then will be "worst case," since we know that reverberation will maintain levels in those parts of the room where the reverberant field predominates. In other words, the designer should not rely on the reverberant field in achieving a given level in the room, but rather simply take the benefit as it presents itself.

A. A simple sound reinforcement system:

We desire a capability for average speech levels of 85 dB, with peaks levels some 12 dB higher, or 97 dB. The farthest listener is 25 meters away from the array.

Calculating the inverse square attenuation from 1 meter to 25 meters:

Loss = 20 log (25) = 28 dB

Adding this to 97, we get 125 dB

Assume that the HF horn/driver we are using has a sensitivity of 113 dB (1 watt at 1 meter), we can calculate the electrical power (WE) required to reach this level:

125 - 113 = 12 dB

WE = $10^{(12/10)}$ = 16 watts

The driver can easily handle this amount of input power, and we now want to decide how much electrical headroom to design into the system. Adding electrical headroom of 6 dB would bring the power required up to 64 watts, and we would "round up" this figure to 75 watts, still well within the power rating of a 2445J driver.

Now we perform a similar analysis for the LF part of the system. Again, we have the same maximum acoustical level requirement at 25 meters of 97 dB. Relating this by inverse square to a distance of one meter gives, as before, a level of 125 dB.

We will use the 4648 LF system. With its sensitivity of 100 dB, we will require electrical power of:

WE = $10^{(25/10)}$ = 316 watts

For additional electrical headroom of 3 dB, we would specify 630 watts. Rounding up to 800 watts (which is the program rating of the 4648) would give us headroom of 4 dB.

We have now specified a system which has more than enough electrical power available to drive it to its design acoustical limits, and yet can safely handle that power. Because of the rolled off spectrum in speech signals, it would not be necessary to make another analysis at a higher frequency.

B. A motion picture theater system:

Here, we can use the stock JBL model 4675 systems, bi-amplified, and arrive at levels at 25 meters which are exactly as in the previous example. But there would be three or five of these systems, depending on the size of the theater and the kind of films exhibited there. Since reference acoustical level of 85 dB is established at full modulation of the recording medium, we would probably not need the 12 dB acoustical headroom — but it wouldn't hurt.

What would work to our advantage is the extra channels. Three channels will play about 5 dB louder than one, and 5 channels will play about 7 dB louder.

The added requirement in this installation is subwoofers, and we will now make those calculations. First, we must determine how loud the subwoofer system is to play. Equal loudness contours in the region below 50 Hz indicate that, in order to match a mid-band level of 85 dB, levels on the order of 100 dB will be needed. Let us set that as our goal at a distance of 25 meters and proceed with the calculations:

A single 4645 has a sensitivity of 95 dB and a program power rating of 600 watts.

At a reference distance of 1 meter, a single 4645, powered with 600 watts, will produce a level of:

$95 + 10 \log (600) = 122$ dB

At a distance of 25 meters, this will be diminished by 28 dB, giving us a level of 94 dB.

If we use two 4645's we will pick up 3 additional dB in terms of added power capability, but we will also pick up the advantage of mutual coupling, which will give us an efficiency doubling, or another 3 dB.

Therefore, a pair of 4645's each fully powered, would give us 100 dB, while a total of four of them would give us the capability of 106 dB, and eight systems would give us the capability of 112 dB.

The system installed in the Motion Picture Academy has eight 4645 systems, and the maximum level, at a distance of about 25 meters from the array, was measured at 112 dB, which is in agreement with the calculations made here.

It is customary to equalize motion picture systems for flat electroacoustical response, realizing that this response will be rolled off to some degree by HF screen losses and air losses (at large distances). These cumulative losses approximate the desired "house curve" specified in ISO Bulletin 2969; however, the electrical boost required to achieve flat electroacoustical response does carry the potential for damage if the system is improperly operated.

Typically, a HF driver mounted on a uniform coverage horn will need about 8 to 10 dB of boost at 10 kHz, relative to 1 kHz, in order to meet the overall equalization requirement in the house.

Recalling that it took only 16 watts power input to reach a level of 97 dB at a distance of 25 meters, we can calculate the difference in level between 16 watts and the driver's program rating of 100 watts:

$10 \log (100/16) = 8$ dB

Thus, we barely have enough headroom in the HF driver to accomplish our equalization goals. But things are not as borderline as they might seem, since we are not likely to make full use of the 12 dB acoustical headroom over 85 dB, which the system was originally designed for.

In other words, we have more than enough acoustical headroom for the job at hand.

C. A disco system:

Disco installations are normally multi-channel, and there will be as many individual systems in the total installation as required for proper coverage. Let us assume that a given system in a disco installation will be used to cover a target area no greater than a distance of 8 meters. Each system will be configured as 4-way, and flat power bandwidth will be an absolute requirement.

Our analyses will be made in the center of each band at frequencies of 50 Hz, 200 Hz, 2 kHz, and 10 kHz. Our target will be for levels of 110 dB at a distance of 8 meters, and this requirement must be met by each of the four frequency sections of each system.

Beginning with the HF section, we will specify a 2380/2445J combination, with its sensitivity of 113 dB.

Thus, the level at 1 meter will be:

$110 + 20 \log (8) = 128$ dB

The electrical power required for this will be:

$W_E = 10^{((128-113)/10)} = 32$ watts

Moving on to the LF part of the system, we will use a combination of four 2123 drivers. The combination has a net sensitivity of about 105 dB and an overall program power rating of 1000 watts.

The power required for a level of 110 dB at a distance of 8 meters is calculated:

$110 + 20 \log (8) = 128$ dB

$W_E = 10^{((128-105)/10)} = 200$ watts

The rated program power of 1000 watts will give us 7 dB of acoustical headroom.

The VLF part of the system will use as many 2240's as required. For a level of 110 at 8 meters, the level at 1 meter will be 128 dB. A single 2240 powered with one watt will produce 98 dB at one meter. Powered to its program rating of 600 watts, a single 2240 will produce a level of 125 dB, referred to one meter. A pair of 2240's will give us 6 dB more (increased power plus mutual coupling), and our specification will be met.

The UHF requirements will be met by a group of ring radiators. Let us choose the 2404H, with its program rating of 40 watts and sensitivity of 105 dB.

Again, the requirement is for a level of 128 dB at a distance of 1 meter. A single 2404, powered by its rated 40 watts, will produce a level of 121 dB, some 7 dB short of the goal.

Generally, we can estimate that each doubling of ring radiators will give us a net 3 dB increase, assuming that the array does not become more directional. Therefore, a total of four 2404's would give us 126 dB, still 2 dB short of the mark. Doubling it further to eight would give us, conservatively, capability of 129 dB on continuous program, and this would meet the requirement, with a total power availability of 320 watts into the UHF array.

In a system such as this, it would be appropriate to allot power to the four sections based solely on the sine wave capability of the drivers. The reason is that the system is likely to be stressed much of the time; any additional electrical headroom capability would represent a hazard.

What we have shown here is only a small part of the total design process for a properly engineered disco system. Such considerations as component layout and orientation, crossover frequencies and slopes, impedances, electrical interface, and signal processing would take far more time. What we have ensured is that the system is in fact power flat and that all four sections of it can produce the required 110 dB at a distance of 8 meters.

D. A studio monitor system:

For the smaller monitors (4406, 4408, 4410, and 4412) we recommend they be powered at twice their IEC rating, since this will allow them to accept short-term mid- and high-frequency transients of sufficiently low duty cycle for overall cleaner output. However, if there is any question of competence of the operating engineers, it would be best to specify no more than the IEC rated input power.

The Bi-Radial monitors (4425, 4430, and 4435) have detailed specification sheets which clearly spell out the maximum input power bandwidth of the systems. These curves represent continuous sine wave input, and if that mode of operation is likely to occur, then it would be best to power the loudspeakers accordingly. However, for normal program applications we would recommend that the systems, in bi-amplified mode, be powered with amplifiers twice the LF and HF IEC ratings, since this would provide cleaner mid- and high-frequency transients.

E. Musical instrument applications:

For the performer on an electric instrument, the amplifier and loudspeaker are extensions of the instrument itself. Amplifier distortion and loudspeaker overload may be used to musical advantage, but these are difficult parameters for a loudspeaker manufacturer to identify and attach numbers to.

JBL's muscial instrument loudspeakers are tested and rated the same way that other transducers are. However, recognizing that amplifier distortion is a valid musical tool, we recommend that a musical instrument loudspeaker be powered with an amplifier of one-half the loudspeaker's sine wave rating. The reason for this is as follows:

When an amplifier is driven into hard clipping, it produces virtual square waves at its output. An amplifier rated at, say, 100 watts for sine wave output will produce a 200-watt square wave. And it can do this pretty much on a continuous basis as long as it can be adequately cooled.

Since the crest factor of a square wave is 0 dB, the full 200 watts would be delivered to the loudspeaker on a continuous basis.

As a typical example, let us assume that a guitar player has two E130 loudspeakers. The sine wave rating for a single E130 is 150 watts, and for the pair used in a system, the sine wave rating would be 300 watts. We would recommend that the pair be driven with a 150-watt amplifier, realizing that the amplifier could produce continuous output of 300 watts when driven into clipped output.

Conclusions:

Well engineered systems rarely fail, and the slight cost disadvantage incurred in making a marginal system a "comfortable" one is often well worth it in long term reliability and customer satisfaction. The general observations and rules presented in this Technical Note must be assimilated over a long period of time, but the education and insight it provides will be well worth the time invested.

As a final note, let us state the working "rules of thumb" which come out of this Technical Note:

1. For carefully monitored applications where peak transient capability must be maintained, a system should be powered with an amplifier capable of delivering twice the sine wave rating of the loudspeaker system.

2. For routine application where high continuous, but non-distorted, output is likely to be encountered, a system should be powered with an amplifier capable of delivering the sine wave rating of the loudspeaker system.

3. For musical instrument application, where distorted output may be a musical requirement, the loudspeaker should be powered with an amplifier capable of delivering only one-half the sine wave rating of the loudspeaker system.

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5. IEC Standard 268-5

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