SOUND SYSTEM ENGINEERING

4th Edition

Don Davis
Eugene Patronis, Jr.
Pat Brown
Sound System Engineering
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Fourth Edition

Don Davis
Eugene Patronis, Jr.
Pat Brown

Edited by
Glen Ballou
The fourth edition of *Sound System Engineering* is dedicated to

*Carolyn Davis*

who is the catalyst that made it happen
and is the glue that held it all together.
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There are two worlds in audio—one of wave equations, Fourier, Hilbert, and Laplace transforms, and the other of Ohm’s Law, Sabine and Hopkins Stryker. Eugene Patronis, Jr., straddles both like a colossus, as he is able to theorize in Quantum Mechanics and design, build, and service, with his own hands, all components used in audio.

Pat Brown is our new co-author and brings to this volume unique tools he has developed in the course of his loudspeaker testing, particularly directivity measurements, into the twenty-first century. He has taught Syn-Aud-Con seminars all over the world in person as well as through his internet training programs. He is a longtime friend of both Dr. Patronis and Don Davis, and like them, a man who delights in fully sharing his knowledge of sound system engineering with others. We welcome his participation in this volume.

The authors come from two quite different backgrounds: one is academic, the others are industrial and field oriented. “The lion is known by his claw” was said of Newton, whereas the technician approach uses a broad brush to get a workable, if not elegant, answer. Therefore, we have identified each author’s contribution separately. It’s your privilege to select the approach most applicable to your need. With today’s generation of computer users and the wealth of available software it’s you, the reader, who chooses the boundaries of your interests and academic skills. It is our wish that whatever background you bring to the subject you will find new tools for that level and hints of the next.

Sound System Engineering is a widely sold, widely used text on sound system design. The first editions were oriented toward those planning systems from components available in the existing marketplace, i.e., they were treated as boxes on a diagram. The first editions ignored component design and analysis other than their interconnecting parameters.

When Don and Carolyn Davis, the authors of the first two editions, sought specific advice on component design and in-depth analysis of given components they turned to their long time friend and mentor, Eugene Patronis, Jr. to provide the in-depth analysis he excels in. You will find in this edition both approaches, allowing newcomers to operate efficiently while providing the more experienced an opportunity to achieve a more advanced viewpoint. We know that one can start reading on one level, but as our experience and expertise develops, we are grateful for the more advanced approach. What we read as our learning process starts is much different years later and we become very grateful for the more advanced material.

Those who have benefited from a rigorous and thorough academic background will find that Eugene Patronis’ work is a succinct summary of all you should have absorbed intellectually whereas the less sophisticated approach may contain useful nuggets that have surmounted “gray” areas in system compromises. This dual approach provides some seemingly uneven interconnects but benefits from the diverse experience of the authors.

The authors have retained their own mental images of who they are writing for, often a combination of both approaches. We hope that you will find this volume useful in pursuit of our mutual goal of truly engineered rather than merely assembled sound systems.

Thanks to Glen Ballou

Our special thanks to Glen Ballou, who transcribed our material into a publishable format. These simple words can’t begin to describe the agony he has endured.

Pat Brown,
Don Davis,
Eugene Patronis, Jr.
Chapter 1

Why Sound System Engineering?

by Don Davis

1.1 Prerequisites ................................................................. 3
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1.9 Fields ................................................................. 4
“Sound” has over the centuries been associated with human hearing (i.e.: “Is there a sound if a tree falls in the forest without a listener present?”) According to Webster, “The sensation perceived by the sense of hearing.” Also from Webster: “Audio, on the other hand, has largely been associated with electrical communication circuits.”

“System” is a word we use to describe any “experience cluster” that we can map as a set of interacting elements over time. Typically a system is mapped by identifying the pathways of information flow, as well as possibly the flow of energy, matter, and other variables. But the flow of information is special; because only information can go from A to B while also staying at A. (Consider: photocopy machines would be useless if one didn’t get to keep the original). Digital systems, analog systems, acoustic systems, etc. should be regarded by a system engineer as so many “black boxes” that need to be matched, interconnected, and adjusted. The internal circuitry should be the interest of the component designer/manufacturer.

1.1 Prerequisites

What kind of background should an aspiring sound system engineer possess is an often asked question. A list of desirable experiences would include:

1. Some basic electrical training.
2. An interest in mathematics.
3. A good ear (a love of quality sound and acute aural senses).
4. Skill with basic tools.
5. Some appreciation of the perils of rigging.
6. Good reading and writing skills.
7. A genuine appreciation for the art, philosophy, and science of sound.

1.2 Basic Electrical Training

Time spent as an apprentice electrician is not wasted. In many cases, large sound systems deal with separate power systems, and safety springs from knowledge of the power circuits that are involved. Conduits, cable sizes, and types of grounding and shielding can be complex even at power frequencies. Knowledge of the electrical codes is a necessary fundamental tool.

1.3 Mathematics

From Ohm’s law to the bidding process, an ability to quickly learn new algorithms both speeds up processes and ensure profits. In today’s markets “cut and try” is too expensive of both time and money to allow avoidance of basic computer skills; the use of programs such as Mathcad for both technical and financial calculations is important. Knowing what the formulae actually used are doing is essential. In order to trust any computer program, having done it first on paper the hard way, provides knowledge and confidence in the fast way and leaves you capable of detecting unexpected anomalies that might occur. Yes! You do need more than arithmetic.

1.4 Hearing Versus Listening

We all hear. But what we listen to depends to a large degree on our previous listening experiences. I have often stood in the center of an acoustic anomaly such as a reflection from an undesirable angle, distance, and level, that was destroying speech intelligibility, and watched the startled expression on the face of a person sitting in the pew as a piece of acoustical material is passed between his ears and the reflection, which restored intelligibility.

Once experienced, your eyes, ears, and brain, can recognize such problems by simply walking through them. Sensitive listening is a great plus in sound system work, and it is a sufficient reason to hear as many venues as possible under normal usage conditions. I am always surprised when I see engineers trying to design a church sound system from a set of drawings without ever having attended a service to see what their actual needs are versus what they’d like to provide them.

Because all sound system design starts in the acoustic environment and works back from there to the input, failure to experience the normal use of the space can be fatal to the ultimate end result. On one occasion I was listening in a mammoth cathedral from a position behind the altar, when asked by the administrator, if our design could solve their intelligibility problem. The priest about to conduct the service spoke to me, and because of a combination of a speech defect and in a foreign accent, I was unable to understand him to sufficiently comprehend his message. I had to tell the administrator that our system could only raise the priest’s audio level, not his intelligibility.

Watching successful ministers, politicians, and other public figures use microphones reveals a world of problems unaddressed by the most compe-
tent engineer. In one case the engineer was asked if he could “put more soul in the monitor.”

1.5 Craftsmanship

Possession of a guitar does not make one a musician nor do tools make a craftsman. Skill with basic tools manifests itself in clean solder joints, orderly cabling, careful labeling on panels and terminals. Construction of successful loudspeaker arrays is a challenge to both artistry and craftsmanship. In my experience craftsmanship is a direct expression of character.

1.6 Rigging

Rigging, in itself, is a business as complex and difficult as engineering the sound system and often behooves sound contractors to seek out professional assistance when required to hang large, heavy, and expensive loudspeaker arrays.

I was involved in a consulting job for a major public arena venue where the owner intended to hang the new array from the previous array’s rigging. (A complicated system of cables and drums for raising and lowering the arrays). I insisted on their hiring a notable rigging authority who went up into the rigging with a camera and came down with a dozen photographs of impending disasters, such as grooves worn in the drums by the cables, frayed cables, unsafe connectors, and a lack of safety cables, to cite but a few of the problems. There are recorded fatalities from falling arrays. It is not a business for amateurs.

1.7 Literacy

This would seem obvious, but is often a weak link in an otherwise successful background experience. Sales presentations, bid offers, instruction manuals for the operators of your systems, all require reading and writing skills. Communications with customers, suppliers, and consultants needs to be thoughtfully and concisely written. For example, the contractor should be on record telling the customer that the design will function properly only if the HVAC contractor meets the specified noise criteria that is provided in the Specification. Failure to do so can be disastrous. A memo on file with the owner can save the sound contractor and/or consultant from having to take the blame.

1.8 The Art, Philosophy, and Science of Sound

The design of well-engineered sound systems stands on the shoulders of the giants who created the communication industry. “Art precedes science” is an axiom that is eternally true. Prof. Higgins as portrayed in the film, “My Fair Lady,” exemplified the majesty of language, the science of studying its proper sounds, and meanings, and the engineering systems used in that earlier day. Even today the most difficult sound systems to design, build, and operate are those used in the reinforcement of live speech. Systems that are notoriously poor at speech reinforcement often pass reinforcing music with flying colors. Mega churches find that the music reproduction and reinforcement systems are often best separated into two systems.

1.9 Fields

From my first view of the rainbow depiction of the electromagnetic spectrum from dc to gamma ray I have striven to gain a conceptual mental view of various fields, Fig. 1-1. Physical science, during the past century, has come to the conclusion that the Universe is some sort of field. The nature of this universal field remains controversial—is it matter which has mass? Or something more ethereal such as information?

Michael Faraday, 1831, said “Perhaps some force is emanating from the wire.”

A Cambridge man said “Faraday, let me assure you, at Cambridge our electricity flows through the wire.”

Oliver Heaviside, 1882, from his book, Electrical Papers, Vol. 1:

Had we not better give up the idea that energy is transmitted through the wire altogether? That is the plain course. The energy from the battery neither goes through the wire one way nor the other. Nor is it standing still, the transmission takes place entirely through the dielectric. What, then, is the wire? It is the sink into which the energy is poured from the dielectric and there wasted, passing from the electrical system altogether.

John Ambrose Fleming in 1898 wrote:

It is important that the student should bear in mind that, although we are accustomed to speak of current as flowing through the wire in one direction or the other, this is a mere form of
words. What we call the current in the wire is, to a large extent, a process going on in the space or material outside the wire....


Heaviside is the only one who considers the nature of the sources as well as the boundary effects both for the initial buildup or transient behavior and for the steady-state condition. He is the first also, to consider the leakage through the insulation, in view of which the true significance of the inductance parameter may be appreciated.... His work is a first approximation only as compared with other, more rigorous treatments. For the engineer, however, this first approximation is usually sufficient....

Further

The concept of guided waves, before Maxwell, the physical picture of the propagation of electricity through a long circuit was more or less that which is frequently presented in elementary textbooks, where the hydraulic analogy to an electric circuit is given for purposes of visualization. That is, the seat of the phenomenon was taken to be within the conductor. What occurred outside the conductor could be neither definitely formulated nor described. The electrical energy was thought of as being transmitted through the conductor which, therefore, became of prime importance. In fact, if we accept this point of view altogether, it becomes impossible to conceive of a flow of electrical energy from one point to another without the aid of an intervening conductor of some sort. It has been the writer’s experience that many students are quite wedded to this point of view, so much so, in fact, that to them the propagation of energy without wires (wireless transmission) becomes a thing altogether apart from other forms of transmission involving an intervening conducting medium.

An appreciation of Maxwell’s theory of electromagnetic wave propagation brings the so-called wireless and wired forms of transmission under the same roof, so to speak. They merely appear as special cases of the same fundamental phenomenon.... The presence of a conductor merely causes the field be broken up into various components, some of which are assigned to the conductor itself, others to the surrounding medium, and still others to the surface separating the two media.

From the *Standard Handbook for Electrical Engineers* by Donald G. Fink and H. Wayne Beaty.... There is a section entitled, “Electromagnetic Wave Propagation Phenomenon.”

The usually accepted view that the conductor current produces a magnetic field surrounding it must be displaced by the more appropriate one that the electromagnetic field surrounding the conductor produces, through a small drain on the energy supply, the current in the conductor. Although the value of the latter may be used in computing transmitted energy, one should clearly recognize that physically this current produces only a loss and in no way has a direct part in the phenomenon of power transmission.

Ralph Morrison’s website has some comments on electromagnetic laws.

The laws I want to talk about are the basic laws of electricity. I’m not referring to circuit theory laws as described by Kirchhoff or Ohm but the laws governing the electric and magnetic fields. These fields are fundamental to all electrical activity whether the phenomenon is lightning, electrostatic display, radar, antennas, sunlight, and power generation, analog or digital circuitry. These laws are often called Maxwell’s equations. Light energy can be directed by lenses, radar energy can be directed by waveguides and the energy and power frequencies can be directed by copper conductors. Thus we direct energy flow at different frequencies by using different materials. For utility power the energy travels in the space between conductors not in the conductors. In digital circuits the signals and energy travel in the spaces between traces or between the traces and the conducting surfaces. Buildings have halls and walls. People move in the halls not the walls. Circuits have traces and spaces, signals and energy moves in the spaces not in the traces.

Scanning the Electromagnetic Spectrum Chart from dc through radio waves, light itself, out to gamma rays we can see that electromagnetic fields play a key part in our lives, Fig. 1-1.

Researchers studying human consciousness are finding electromagnetic phenomenon in addition to the previously known electrical phenomena. When EMI (electromagnetic interference) occurs in audio systems RF spectrum analyzers can be useful tools.
Figure 1-1. Electromagnetic spectrum chart.
References

3 Chapter 3 Sound and Our Brain


5 Chapter 5 Digital Theory


J. R. Stuart, Coding High Quality Digital Audio.

6 Chapter 6 Mathematics for Audio Systems


Table 6-10. (cont.) Points of Figs. 1-19 and 1-20

<table>
<thead>
<tr>
<th>7.5</th>
<th>-2.840</th>
<th>-2.850</th>
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<th>0.809</th>
<th>12.5</th>
<th>-4.420</th>
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<td>17.5</td>
<td>-5.500</td>
<td>-5.590</td>
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<td>-6.200</td>
<td>30.0</td>
<td>-0.309</td>
<td>32.5</td>
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<td>-5.590</td>
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<td></td>
</tr>
<tr>
<td>72.5</td>
<td>0.309</td>
<td>6.100</td>
<td>6.200</td>
<td></td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
Time $\times 10^5 \times 10^6 \times 10^3 \times 10^3 \times 75.0 \times 0.000$

77.5 6.180 6.200 80.0 0.304 82.5 5.580 5.590 85.0 0.588
97.5 0.980 0.982 1.000 1.000 Table 6-10. (cont.) Points of
Figs. 1-19 and 1-20 Time $\times 10^5 \times 10^6 \times 10^3 \times 10^3 \times 75.0 \times 0.000$

Chapter 7 Using the Decibel

Sound pressure water: 1 dyn/cm² air: 0.0002 dyn/cm² or 0.00002 N/m² SPL or L_P

Sound intensity 10⁻¹⁶ W/cm² 10⁻¹² W/m²

Sound power 10⁻¹² W (new) 10⁻¹³ W (old) PWL or L_w

Audio power 10⁻³ W dBm

EMF 1 V dBV

Much earlier, but valuable, literature used 10⁻¹³ W as a reference. In that case, the L_P value approximately equals the L_w value at 0.282 ft from an omnidirectional radiator in a free field (i.e., the number values are the same but, of course, different quantities are being measured). For 1 W using 10⁻¹² W at 0.283 m, L_w ≈ L_P = 120 dB. For 1 W using 10⁻¹³ W at 0.282 ft, L_w ≈ L_P = 130 dB as found with the equation: (7-30)

where,

L_w is 10log the wattage divided by the reference power 10⁻¹³,

r is the distance in meters from the center of the sound source.

Fig. 7-6 requires that you either know the distance from the source or assumes you are in the steady reverberant sound field of an enclosed space. L_P readings without one of these is meaningless.

Fig. 7-7 shows typical power and L_w values for
various acoustic sources. 7.12 The Equivalent Level \( L_{EQ} \) in Noise Measurements Increasingly, acoustical workers in the noise control field are erecting an interesting edifice of measurement systems. A number of these measurement systems are based on the concept of average energy. Suppose, for example, that we have some means of collecting all of the A-weighted sound energy that arrives at a particular location over a certain period of time such as 90 dBA for 3.6 s (this could be a series of levels that lasted seconds, hours, or even days). We can then calculate the decibel level of steady noise for, say, 1 h that would be the equivalent level of the dBA for 3.6 s. That is, we wish to find the energy equivalent level for 1 h: (7-31)

\[
L_{eq} = \frac{P_A}{P_0} \cdot 3600 \, s
\]

This integration reduces to

Table 7-8. Recommended Descriptor List

<table>
<thead>
<tr>
<th>Term</th>
<th>A-Weighting Alternative*</th>
<th>A-Weighting Other Weighting †</th>
</tr>
</thead>
<tbody>
<tr>
<td>Unweighted</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sound (pressure) level**</td>
<td>( L_A ) ( p_A ) ( L_B ), ( L_{pB} ) ( L_P )</td>
<td></td>
</tr>
<tr>
<td>Sound power level ( L_{WA} ) ( L_{WB} ) ( L_W )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Max sound level ( L_{max} ) ( A_{max} ) ( B_{max} ) ( p_{max} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Peak sound (pressure) level ( L_{Apk} ) ( B_{pk} ) ( L_{pk} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Level exceeded x% of the time ( L_x ) ( X_{A} ) ( X_{B} ) ( L_{px} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Equivalent sound level ( L_{eq} ) ( A_{eq} ) ( B_{eq} ) ( L_{peq} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Equivalent sound level over time (T) # ( L_{eq(T)} ) ( A_{eq(T)} ) ( L_{Beq(T)} ) ( L_{peq(T)} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Day sound level ( L_{d} ) ( A_{d} ) ( B_{d} ) ( p_{d} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Night sound level ( L_{n} ) ( A_{n} ) ( B_{n} ) ( p_{n} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Day-night sound level ( L_{dn} ) ( A_{dn} ) ( B_{dn} ) ( p_{dn} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Yearly day-night sound level ( L_{dn(Y)} ) ( A_{dn(Y)} ) ( B_{dn(Y)} ) ( p_{dn(Y)} )</td>
<td></td>
<td></td>
</tr>
<tr>
<td>Sound exposure level ( L_{S} ) ( L_{SA} ) ( L_{SB} ) ( L_{Sp} )</td>
<td></td>
<td></td>
</tr>
</tbody>
</table>
| Energy average value over (nontime,
domain) set of observations $L_{eq(e)}$ $L_{Aeq(e)}$ $L_{Beq(e)}$ $L_{peg(e)}$ Level exceeded $x\%$ of the total set of (nontime domain) observations $L_x(e)$ $L_{Ax(e)}$ $L_{Bx(e)}$ $L_{px(e)}$

Average $L \times \frac{x(e)}{A_x(e)} \frac{B_x(e)}{L px(e)}$

**“Alternative” symbols may be used to assure clarity or consistency.**

† Only B-weighting shown. Applies also to C, D, E weighting.

**The term “pressure” is used only for the unweighted level.**

‡ Unless otherwise specified, time is in hours (e.g., the hourly equivalent level is $L_{eq(1)}$). Time may be specified in nonquantitative terms (e.g., could be specified as $L_{eq(WASH)}$ to mean the washing cycle noise for a washing machine).

$L \rho L \omega \rho 10^{4\pi r^2} L_{eq} = \frac{L}{10^{3600s}} \frac{P_A}{L^{2 P_o 2}} \frac{td^0}{3.6s} \int \frac{\delta}{\omega} \frac{\omega}{\delta \omega}$ in decibels log-

Thus, 1.0 hour of noise energy at 60 dBA is the equivalent energy exposure of 90 dBA for 3.6 s. $L_{DN}$ (day-night level), CNEL (community noise level), etc., all follow similar schemes with variation in weightings for differing times of day, etc. It is of interest that shooting a 0.458 magnum 174.7 L P (peak) for 2.5 ms translates into:

of steady sound for 1 h. OSHA allows only 15 min of exposure to levels of 110dBA–115dBA. As Howard Ruark’s African guide, Harry Selby, remarked after Ruark had accidentally set off both barrels at once of a 0.470 express rifle while being charged by a Cape buffalo, “One of you ought to get up.”
7.13 Combining Decibels

7.13.1 Adding Decibel Levels

The sum of two or more levels expressed in dB may be found as follows: (7-32) If, for example, we have a noisy piece of machinery with an $L_P = 90$ dB, and wish to turn on a second machine with an $L_P = 90$ dB, we need to know the combined $L_P$. Since both measured levels are the result of the power being applied to the machine, with some percentage being converted into acoustic power, we can determine $LT$ by using Eq. 7-33. Therefore: Doubling the acoustic power results in a 3 dB increase. An alternative dB addition technique is given through the courtesy of Gary Berner. (7-33) Example If we wish to add 90 dB to 96 dB, using Eq. 7-33, take the difference in dB (6 dB) and put it in the equation: Input signals to a mixing network also combine in this same manner, but the insertion loss of the

\[
L_{EQ} = 10 \log \left( \frac{10^{90} + 10^{90}}{10^{90} + 10^{90}} \right)
\]

\[
L_{EQ} = 10 \log \left( \frac{10^{90} + 10^{90}}{10^{90} + 10^{90}} \right)
\]

\[
L_T = 10 \log \left( 10^{90} + 10^{9} \right)
\]

\[
L_T = 10 \log \left( 10^{90} + 10^{9} \right)
\]

Figure 7-6. Typical A-weighted sound levels as measured with a sound level meter. (Courtesy GenRad) 50 hp siren (100 ft) Jet takeoff (200 ft) Riveting machine* Cut-off saw* Pneumatic hammer* Textile weaving plant* Subway train (20 ft) Pneumatic drill (50 ft) Freight train (100 ft) Vacuum cleaner (10 ft) Speech (1 ft) 140 130 120 110 100 90 80 70 60 50 40 30 20 10 0 Large transformer (200 ft) Soft whisper (5 ft) At a given distance from the source Decibels re: 20 mN/m^2
Environmental Casting shakeout area Electric furnace area
Boiler room Printing press plant Tabulating room Inside
sport car (50 mph) Near freeway (auto traffic) Large store
Accounting office Private business office Light traffic
(100 ft) Average residence Minimum levels - residential
Areas in Chicago at night Studio (speech) Studio (sound
recording) Threshold of hearing youths-1000 to 4000 Hz
*operator position \( L \ T \ 10 \ 10 \ \text{diff in dB} \)- \( 10 \)
---------------------------------------------------------------------\( 1+ \) \( \frac{\log\text{smallest}}{\log 90+} \) dB= 96.97

network must be subtracted. Two phase-coherent

sinewave signals of equal amplitude will combine to
give a level 6 dB higher than either sinewave.

The general case equation for adding either

sound pressure, voltages, or currents is: (7-34)

Table 7-9 shows the effects of adding two equal

amplitude signals with different phase together

using Eq. 7-36.

7.13.2 Subtracting Decibels

The difference of two levels expressed in dB may be

found as follows: (7-35) 7.13.3 Combining Levels of

Uncorrelated Noise Signals When the sound level of a

source is measured in the presence of noise, it is

necessary to subtract out the effect of the noise on the

reading. First, take a reading of the source and the noise

combined (\( L \ S \ + \ N \)). Then take another reading of the noise

alone (the source having been shut off ). The second

reading is designated \( L \ N \). Then: (7-36) To combine the

levels of uncorrelated noise signals we can also use the

chart in Fig. 7-8 as follows

Figure 7-7. Typical power and \( L \ W \) values for various

acoustic sources. 170 160 150 140 130 120 110 100 90 80 70
60 50 40 30 100,000 10,000 1000 100 10 1 0.1 0.01 0.001
0.0001 0.000 01 0.000 001

0.000 000 1
0.000 000 01
0.000 000 001 Power

(watts) Power Level dB re: 10^-12 W Source

20 to 40 million 195 Saturn rocket Ram jet Turboprop engine with afterburner Turbojet engine (7000 lb thrust) 4 jet engine 75-piece orchestra Piano organ Small aircraft engine Large chipping hammer Piano BB tuba Blaring radio Centrifugal ventilating fan (15,000 CFM) 4-ft loom Auto on highway Vane axial ventilating fan (1500 CFM) Voice-shouting (average long-term rms)

Voice—conversational level (average long-time rms)
Voice—very soft whisper Peak rms values in 1/8 s intervals Peak rms values in 1/8 s intervals}

Combined L P =

\[ 20 \log \left( \frac{10^{L_{P1}}}{10^{L_{P2}}} \right) + \log \left( \frac{a_1}{a_2} \right) \cos(\theta) + \log \frac{10^{L_{S1}}}{10^{L_{S2}}} \]

Table 7-9. Combining Pure Tones of the Same Frequency but Differing Phase Angles

<table>
<thead>
<tr>
<th>Signal 1 Amplitude, L P in dB</th>
<th>Signal 1 Phase, in Degrees</th>
<th>Signal 2 Amplitude, L P in dB</th>
<th>Signal 2 Phase, in Degrees</th>
<th>Combined Signal Amplitude, L P in dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>90</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>96.02</td>
</tr>
<tr>
<td>90</td>
<td>+90</td>
<td>95.72</td>
<td>+90</td>
<td>95.89</td>
</tr>
<tr>
<td>90</td>
<td>+90</td>
<td>95.48</td>
<td>+90</td>
<td>95.72</td>
</tr>
<tr>
<td>90</td>
<td>+90</td>
<td>94.29</td>
<td>+90</td>
<td>94.77</td>
</tr>
<tr>
<td>90</td>
<td>+90</td>
<td>93.71</td>
<td>+90</td>
<td>93.80</td>
</tr>
<tr>
<td>90</td>
<td>+90</td>
<td>93.01</td>
<td>+90</td>
<td>92.18</td>
</tr>
<tr>
<td>90</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>90.00</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>89.38</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>88.54</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>87.60</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>86.70</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>85.83</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>85.01</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>84.28</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>83.54</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>82.81</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>82.08</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>81.38</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>80.76</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>80.14</td>
</tr>
<tr>
<td>80</td>
<td>+90</td>
<td>90</td>
<td>+90</td>
<td>79.53</td>
</tr>
</tbody>
</table>

7.13.4 To Add Levels

Enter the chart with the numerical difference between the two levels being added (top of chart).

Follow the line corresponding to this value to its intersection with the curved line; then move left to read the numerical difference between the total and
larger levels. Add this value to the larger level to determine the total.

Example

To add 75 dB to 80 dB, subtract 75 dB from 80 dB; the difference is 5 dB. In Fig. 7-8, the 5 dB line intersects the curved line at 1.2 dB on the vertical scale. Thus the total value is 80 dB + 1.2 dB, or 81.2 dB.

7.13.5 To Subtract Levels

Enter the chart in Fig. 7-8 with the numerical difference between the total and larger levels if this value is less than 3 dB. Enter the chart with the numerical difference between the total and smaller levels if this value is between 3 dB and 14 dB. Follow the line corresponding to this value to its intersection with the curved line, then either left or down to read the numerical difference between total and larger (smaller) levels. Subtract this value from the total level to determine the unknown level. Example Subtract 81 dB from 90 dB; the difference is 9 dB. The 9 dB vertical line intersects the curved line at 0.6 dB on the vertical scale. Thus the unknown level is 90 dB - 0.6 dB, or 89.4 dB.

7.14 Combining Voltage

To combine voltages, use the following equation: (7-37) where, $E_T$ is the total sound pressure, current, or voltage, $E_1$ is the sound pressure, current, or voltage of the first signal, $E_2$ is the sound pressure, current, or voltage of the second signal, $\alpha_1$ is the phase angle of signal one, $\alpha_2$ is the phase angle of signal two.

7.15 Using the Log Charts

7.15.1 The $10\log x$ Chart

There are two scales on the top of the $10\log 10x$ chart in Fig. 7-9. One is in dB above and below a 1 W reference level, and the other is in dBm (reference 0.001 W). Power ratios may be read directly from the 1 W dB scale. Example How many decibels is a 25:1 power ratio? 1. Look up 25 on the Power-watts scale. 2. Read 14 dB directly above the 25.
Figure 7-8. Chart used for determining the combined level of uncorrelated noise signals. 2.0 1.2 0 1 2 3 4 5 6 7 8 9 10 11 12 13 Numerical difference between two levels being added - dB 3 4 5 6 7 8 9 10 11 12 13 14 Numerical difference between total and larger levels - dB 3.0 1.0 0.6 0 Numerical difference between total and smaller levels - dB E T E 1 2 E 2 2 2 2E 1 E 2 a 1 a 2 -( )cos[ ] + = Figure 7-9. The 10log 10 x chart. 1000 400 200 100 60 40 20 10 6 4 2 1 0.6 0.4 0.2 0.1 0.04 0.02 0.01 0.004 0.002 0.001 30 20 10 0 -10 -20 -30 60 50 40 30 20 10 0 Decibels above and below a one-watt reference level dBm Power - watts

Example

We have a 100 W amplifier but plan to use a 12 dB margin for “head room.” How many watts will our program level be?

1. Above 100 W find +50 dBm.
2. Subtract 12 dB from 50 dBm to obtain +38 dBm.

Just below +38 dBm find approximately 6 W.

Example

A 100 W amplifier has 64 dB of gain. What input level in dBm will drive it to full power?

1. Above 100 W read +50 dBm.
2. +50 dBm - 64 dB gain = -14 dBm.

Example

A loudspeaker has a sensitivity of L P = 99 dB at 4 ft with a 1 W input. How many watts are needed to
have an L P of 115 at 4 ft?

1. 115 L P − 99 L P = +16 dB.

2. At +16 on the 1 W scale read 39.8 W.

7.15.2 The 20Log x Chart

Refer to the chart in Fig. 7-10. A 2:1 voltage, distance, or sound pressure change is found by locating 2 on the ratio or D scale and looking directly above to 6 dB.

Example

A loudspeaker has a sensitivity of L P = 99 dB at 4 ft with 1 W of input power. What will the level be at 100 ft?

1. Find the relative dB for 4 ft (dB Rel = 12 dB).
2. Find the relative dB for 100 ft (dB Rel = 40 dB).
3. Calculate the absolute dB (40 dB − 12 dB = 28 dB).
4. L P = 99 dB − 28 dB = 71 dB.

Example If we raise the voltage from 2 V to 10 V, how many decibels would we increase the power?

1. Find the relative dB for a ratio of 2 (dB Rel = 6 dB).
2. Find the relative dB for a ratio of 10 (dB Rel = 20 dB).
3. Absolute dB change = 20 dB − 6 dB = 14 dB.

Since a dB is a dB, the power also changed by 14 dB.

Finding the Logarithm of a Number to Any Base

In communication theory, the base 2 is used. Occasionally other bases are chosen. To find the logarithm of a number to any possible given base, write: (7-38) where, x is the number for which a logarithm is to be found, b is the base, n is the logarithm. Then write: (7-39) and (7-40) x b^n = xlog n blog = xlog blog ---------- n= Figure 7-10. The 20log 10 x chart.

20 25 30 35 40
45 50 55 60 1
2 4 6 8 10 20
40 50 80 100 200 400 600
1000 0 -10 -20
-30 -40 1.0 0.6 0.4 0.2
0.1 0.06 0.04 0.02 0.01 ΔD-dB Ratio or D-feet Ratio or ΔD-dB Suppose we want to find the natural logarithm of 2 (written ln2). The base of natural logarithms is
e = 2.718281828. Then:

To verify this result:

To find log 2 of 26:

The general case is: (7-41)

7.17 Semitone Intervals

Suppose that we need (the semitone interval in
music). We could write: (7-42)

Therefore,

This is the same as multiplying 1.05946 by itself 12
times to obtain 2. 10 0.02508 is called the antilog of
0.02508. The

antilog is also written as log -1 , antilog 10, or 10 exp.

All these terms mean exactly the same thing.

7.18 System Gain Changes

Imagine a noise generator driving a power amplifier

and a loudspeaker, Fig. 72-11. If the voltage out of

the noise generator is raised by 6 dB, what happens? This
means that, in a linear system, a level change ahead of any
components results in a level change for that same signal
in all subsequent components, though it might be measured
as quite different voltages or wattages at differing
points. The change in level at any point would be the same.
We will work with this concept a little later when we plot
the gains and losses through a total system. 7.19 The VU
and the VI Instrument Volts, amperes, and watts can be
measured by inserting an appropriate meter into the
circuit. If all audio signals were sinewaves, we could
insert a dBm meter into the circuit and get a reading that
would correlate with both electrical and acoustical
variations. Unfortunately, audio signals are complex
waveforms, and their rms value is not 0.707 times peak but
can range from as small as 0.04 times peak to as high as
0.99 times peak, Fig. 7-12. To solve this problem,
broadcasting and telephone engineers got together in 1939
and designed a special instrument for measuring speech and music in communication circuits. They calibrated this new type of instrument in units called VU. The dBm and the VU are almost identical, the only difference being in their usage. The instrument used to measure VU is called the volume indicator (VI) instrument. (Some users ignore this and incorrectly call it a VU meter.) Both dBm meters and volume indicator instruments are specially calibrated voltmeters. Consequently, the VU and dBm scales on these meters give correct readings only when the measurement is being made across the impedance for which they are calibrated (usually 150 Ω or 600 Ω). Readings taken across the 2 log e

--- 0.30103 0.43425 = 0.69315 =

2^0.69315 = 2

--- 1.41497 0.30103 = 4.70044 =

log 10 of the number log of the base

--- log base of the number = 2 12

--- 2^12 = Voltage Electrical Power L P L W Doubled Quadrupled Doubled Quadrupled +6 dB +6 dB +6 dB +6 dB

Figure 7-11. Voltage, electrical power, P, and sound pressure, compared. Reads twice the voltage +6 dB Four times the power +6 dB Loudspeaker ac high 2 meter Noise generator Power amplifier Twice the S.P. +6 dB Sound level meter

design impedance are referred to as true levels,

whereas readings taken across other impedances are called apparent levels.

Apparent levels can be useful for relative frequency response measurements, for example.

When the impedance is not 600 Ω, the correction factor of 10log (600 / new impedance) can be added
to the formula containing the reference level as in

the following equation: (7-43)

The result is the true level.

7.19.1 Crest Factor

The crest factor (CF) is the ratio of the peak output
to the average output. It is typically graphed in terms
of the output power and is expressed in dB. For example,
the CF of a sine wave is 3 dB. The CF of music may vary
between 6 dB and 24 dB. Crest factor is defined as ten
times the logarithm of peak power out divided by average
power. The actual measurements are often made using
voltage. In that case we divide the peak voltage squared by
the rms voltage squared. The root mean square integral is;

(7-44) The rms voltage squared divided by the impedance is
average power. Using voltage implies that both
measurements were made across identical impedances or (open
circuit). Crest factor applied to voltage waveforms is a

common practice but not the defined term. Powers are chosen
because when impedances vary widely over the bandwidth of
interest, and consequently the bandwidth is divided into
components, the powers can be added to obtain the overall
crest factor. The Texas Instruments “Guidelines for
Measuring Audio Power Amplifier Performance” page 25
displays actual measurements and the usefulness of power
measurements. When we remember that power describes heat
dissipation in power amplifiers then CF as a ratio of
powers becomes evident. 7.19.2 The VU Impedance Correction
When a VU instrument is connected across 600 Ω and is
indicating 0 VU on a sinewave signal, the true level is 4
dB higher, or +4 dBm, instead of 0 dBm or zero level. The
reason this is so is shown in Fig. 7-13. The VU inst t
rument uses a 50 µA

Figure 7-12. Sinewave voltage values. The average
t voltage of a sine wave is zero. rms Peak Peak to peak Max
neg One alteration One cycle

Max

pos 90° 180° 0° 270° 960° rms = 0.707 × peak
voltage rms = 0.3535 × peak to peak voltage
peak = 1.414 × rms voltage peak-to-peak = 2.828 × rms/vue
D’Arsonval movement in conjunction with a copper-oxide bridge-type rectifier. The impedance of the instrument and rectifier is 3900 Ω. To minimize its effect when placed across a 600 Ω line, it is “built-out” an additional 3600 Ω to a total value of 7500 Ω. The addition of this build-out resistance causes a 4 dB loss between the circuit being measured and the instrument. Therefore, when a properly installed VI instrument is fed with 0 dBm across a 600 Ω line, the meter would actually read -4VU on its scale. (When the attenuator setting is added, the total reading is indeed 0 VU.) Presently, no major U.S. manufacturer offers for sale a standard volume indicator that complies with the applicable standard (C16.5). The standard requires that an attenuator be supplied with the instrument and none of the manufacturers do so. What they are doing requires some attention. The instruments (usually high-impedance bridge-types) are calibrated so as to act as if the attenuator were present. When the meter reads 0 VU (on a sinewave for calibration purposes), the true level is +4 dBm.
This means a voltage of 1.23 V across 600 Ω will cause the instrument to read an apparent 0 VU. Note that when reading sine wave levels, the label used is “dBm.” When measuring program levels, the label used is “VU.” The VU value is always the instrument indication plus the attenuator value. Two different types of scales are available for VI meters, Fig. 7-14. Scale A is a VU scale (recording studio use), and scale B is a modulation scale (broadcast use). On complex waveforms (speech and music), the readings observed and the peak levels present are about 10 dB apart. This means that with a mixer amplifier having a sinewave output capability of +18 dBm, you are in danger of distortion with any signal indicating more than +8 VU on the VI instrument (+18 dBm − [+10 dB] peaking factor or meter lag equals +8 VU). Fig. 7-15 shows an example of commercially available VI instrument panels used in the past that included the VI instrument and 3900 Ω attenuator, which also contains the 3600 Ω build-out resistor. 7.19.3 How to Read the VU Level on a VI Instrument A VI instrument is used to measure the level of a signal in VU. In calibration: 0 VU = 0 dBm and a 1.0 VU increment is identical to a 1.0 dB increment. The true level reading in VU is found by: (7-45) or where, Apparent level = Instrument indication + Attenuator or sensitivity indicator. Thus, we can have: 1. A direct reading from the face of the instrument (zero preferred). 2. The reading from the face of the instrument plus the reading from the attenuator or other sensitivity adjustment—normally a minimum of +4 dB or higher. When the instrument indicates zero, the apparent level is the attenuator setting. 3. The correction factor for impedance other than the reference impedance. 600 Ω is the normal impedance chosen for a
reference, but any value Figure 7-14. VI instrument scales. 
A. Recording and test equipment. B. Broadcast monitoring. 
Figure 7-15. Examples of commercial-type VI instrument 
panels. True VU level Apparent level Impedance correction 
+= True VU level Instrument indication 10 600 Z act 
---------[ ] log +=

can be used so long as the voltage across it
results in 0.001 W, Fig. 7-16.

Example

We have an indication on the instrument of −4 VU.
The sensitivity control is at +4. We are across 50 Ω 
(a 100 W amplifier 70.7 V output). Using Fig. 7-16, 
our true VU would be −4 VU + (+4 VU) + 10.8 
correction factor = 10.8VU.

7.19.4 Calibrating a VI Instrument

The instrument should be calibrated to read a true 
level of zero VU when an input of a 1000 Hz 
steady-state sinewave signal of 0 dBm (0.001 W) is 
connected to it. For example, typical calibration is 
when the instrument indicates −4, the attenuator 
value is +4, and it is connected across a 600 Ω 
circuit. Levels read on a VI instrument when the 
source is the aforementioned sinewave signal should 
be stated as dBm levels.

7.19.5 Reading a VI Instrument on Program 
Material

Because of the ballistic properties of VI instruments, 
they exhibit what has been called “instrument lag.”
On short-duration peak levels, they will “lag” by approximately 10 dB. Stated another way, if we read a true VU level of +8 VU on a speech signal, then the level in dBm becomes +18 dBm. This means that the associated amplification equipment, when fed a true VU level of +8 VU, must have a steady-state sinewave capability of +18 dBm to avoid overload. Rule Levels stated in VU are assumed to be program material and levels stated in dBm are assumed to be steady-state sinewave. 7.19.6 Reading Apparent VU Levels VI instruments can be used to read apparent or relative levels. If, for example, you know that overload occurs at some apparent level. You can use that reading as a satisfactory guide to the system’s operation, even though you do not know the true level. When adjusting levels using the instrument to read the relative change in level, such as turning the system down 6 dB, you do not need to do so in true level readings. Instrument indication serves effectively in such cases. When being given a level, be sure to ascertain whether it is: 1. An instrument indication. 2. An apparent level. 3. A true level. 4. A relative level. 5. A calibration level. 6. A program level. 7. None of the above but simply an arbitrary meter reading. Special Note: Well-designed mixers have instruments that indicate the available input power level to the device connected to its output. Such levels are true levels. 7.20 Calculating the Number of Decades in a Frequency Span To find the relationship of the number of decades between the lowest and highest frequencies, use the following equations: (7-46) therefore: (7-47) or

Figure 7-16. Relationship between circuit impedance and the dB correction value. Circuit impedance-Ω

30
25
20
15
10 5 0
dB
correct
for
10 2 5 10 20 50 100 200
500 1K True VU = Instrument indication + attenuator setting
+ 10log (600/Z act ) H.F. L.F. --------10 1 1 decade =
H.F. L.F. --------10 x decade = H.F. ln L.F. ln- 10ln
------------------------------------x decades = (7-48)

Further: (7-49)
and (7-50)

Example

How many decades does the bandpass 500 Hz to
12,500 Hz contain? Using Eq. 7-48:
If we had 12,500 Hz as a H.F. limit and wished to
know the low frequency that would give us
1.4 decades, we would calculate:
If we had the L.F. limit and wished to know the
H.F., then:
7.21 Deflection of the Eardrum at Various
Sound Levels
If we make the assumption that the eardrum displacement is the same as that of the air striking it we can write: (7-51)

or (7-52)

where, $D_{in}$ is the displacement in inches (the rms amplitude) of the air, $D_{cm}$ is the displacement in cm, $f$ is the frequency in Hz, $L_P$ is the sound level in decibels referred to $0.00002 \text{ N/m}^2$. Example What is the displacement of the eardrum in inches for a tone at 1000 Hz at a level of 74 dB? Using Eq. 7-51: which is a displacement of approximately one-one-millionth of an inch ($0.000001 \text{ in}$). 7.22 The Phon Fig. 7-17 shows free-field equal-loudness contours for pure tones (observer facing source), determined by Robinson and Dadson at the National Physical Laboraory, Teddington, England, in 1956 (ISO/R226-1961). The phon scale is of equal loudness level contours. At 1000 Hz every decibel is the equivalent loudness of a phon unit. For two different sounds within a critical band (for most practical purposes, using $1/3$ octave bands suffices) they are added in the same manner as decibel readings. $H.F. \ln L.F. \ln 10 \times 10^\text{x decades( }\times\ln$:

$$H.F. e \times \text{ decades( } \times 10\ln( )\times L .F .\ln + =$$

$$L.F. e H .F .\ln x \text{ decades } 10\ln( )-[] = 12,500\ln 500\ln -10\ln$$

$$----------------------------------------------1.39794 \text{ decades=}$$

$$L.F. e 12,500\ln 1.4 \text{ decades } 10\ln( )-[] = 497.63 \text{ Hz= }$$

$$H.F. e 1.4 \text{ decades } 10\ln( ) 497.63\ln + = 12,500 \text{ Hz( )=}$$

$$D_{in} 3 10 7-10 L P 20 ----f \times--( ) \times\ln$$

$$D_{cm} 39 10 3-0.0002 10 \times L P 20 ----f$$

$$---------------------------------------------- \times$$

Figure 7-17. Equal loudness contours. $D_{in} 3 10 7-10 L P 20 \times-1000 \times-100 100 80 60 40 200 -10 50 100 300 500 1k 3k 5k 10k 20k \text{ Sound pressure level - dB }$

Frequency-Hz Loudness Level-Phons 100 120 110 90 80 70 60 50 40 30 20 10 Minimum audible (7-53)

where,
L P1 and L P2 are the individual sound levels in dB.

For example, suppose that within the same critical band we have two tones each at 70 phons. Using Eq. 7-53:

An interesting experiment in this regard is to start with two equal level signals 10 Hz apart at 1000 Hz and gradually separate them in frequency while maintaining their phon level.

They will increase in apparent loudness as they separate. This is one of the reasons a distorted system sounds louder than an undistorted system at equal power levels. One final factor worthy of storage in your own mental “read only memory” is that in the 1000 Hz region most listeners judge a change in level of 10 dB as twice or half the loudness of the original tone.

Fig. 7-18 is a chart of frequency and dynamic range for various musical instruments and the upper and lower frequency range of the average young adult.

7.23 The Tempered Scale

The equal tempered musical scale is composed of 12 equally spaced intervals separated by a factor of . All notes on the musical scale (excluding sharps and flats) however, are not equally spaced.
This is because there are two ½ step intervals on the scale: that between E and F, and that between B and C. The 12 tones, therefore, go as follows: C, C#, D, D#, E, F, F#, G, G#, A, A#, B, C, see Table 7-10.

7.24 Measuring Distortion

Fig. 7-19 illustrates one of the ways of measuring harmonic distortion. Two main methods are employed. One uses a band rejection filter of narrow bandwidth having a rejection capability of at least 80 dB in the center of the notch. This deep notch "rejects" the fundamental of the test signal (usually a known-quality sinewave from a test audio oscillator) and permits reading the noise voltage of everything remaining in the rest of the bandpass. Unfortunately, this also includes the hum and noise as well as the harmonic content of the equipment being tested, Fig. 7-20. The second method is more useful. It uses a tunable wave analyzer. This instrument allows the measurement of the amplitudes of the fundamental and each harmonic, as well as identifying the hum, amplitude, and the noise spectrum shape, Fig. 7-20. Such analyzers come in many different bandwidths, with a 1/10 octave unit allowing readings down to 1% of the fundamental (it is −45 dB at 2f). By looking at Fig. 7-20, it is easy to see that harmonic distortion appears as a spurious noise. Today tracking filter wave analysis allows nonlinear distortion behavior to be “tracked” or measured.

7.25 The Acoustical Meaning of Harmonic Distortion

The availability of extremely wide-band amplifiers with distortions approaching the infinitesimal and the gradual engineering of a limited number of loudspeakers with distortions just under 1% at usable levels (90-100 dB SPL at 10 -12 ft) brings up an interesting question: “How low a distortion is really needed?”

P T 10 10 L P1 10 --------10 L P2 10 --------+ ⎝ ⎟ ⎛ ⎞ 
log= phons =

P T 10 10 70 10 ----10 70 10 ----+ ⎝ ⎟ ⎛ ⎞ 
log= 73 phons=

2 12 Table 7-10. Tempered Scale Note Frequency Ratio

7.25.1 Calculating the Maximum Allowable Total Harmonic Distortion in an Arena Sound System

The most difficult parameter to achieve in the typical arena sound system is a sufficient signal-to-noise ratio (SNR) to ensure acceptable articulation losses for consonants in speech. It must be at least 25 dB. In that case, the total harmonic distortion should be at least 10 dB below the 25 dB Figure 7-10.

Audible frequency range. Lower limit of organ scale Upper limit of ordinary piano scale Upper limit of concert piano scale 20k 10k 5k 2k 1k 500 40 60 100 20 Lower limit of ordinary piano scale Cymbals Snare Drum Bass Drum Kettle Drum Violin Piano Cello Bass Violin Piccolo Flute Oboe Soprano Saxophone Trumpet Clarinet French Horn Trombone Bass Clarinet Bassoon Bass Saxophone Bass Tuba Female Voice Male Voice Handclapping Footsteps Frequency range necessary for understanding speech Upper limit of organ scale Lower limit of audibility Upper limit of audibility Frequency-Hz

SNR to avoid the addition of the two signals. If both signals were at the same level, a 3 dB increase in level would occur. Therefore, (-25 dB) + (-10 dB) means that the total harmonic distortion (THD) should not exceed -35dB. (7-54)

Therefore, we could calculate:
This is why carefully thought-out designs for use in heavy-duty commercial sound work have a THD of 0.8 to 0.9:
Since the 0.8% already represents (100 - 99.2), we can write
Now, suppose an amplifier has 0.001% distortion.
What sort of dynamic range does this represent?

That is a power ratio of: We can conclude that if such a figure were achievable, it would nevertheless not be useful in arena systems. 7.26 Playback Systems in Studios Assume that a monitor loudspeaker can develop $L_P = 110$ dB at the mixer’s ears and that in an exceptionally quiet studio we reach $L_P = 18$ dB at 2000 Hz (NC-20). We then have (7-55) which is equal to 92 dB. Adding 10 dB to avoid the inadvertent addition of levels gives 102 dB. The distortion now becomes: In this case, extraordinary as it is, the previously esoteric figure becomes a useful parameter. 7.26.1 Choosing an Amplifier As we pointed out earlier, the loudspeaker will establish equilibrium around 1% with its acoustic distortion. To the builder of systems, this means that extremely low distortion figures cannot be used within the system as a whole. Therefore, systems-oriented amplifier designers have not attempted to extend the bandpass to extreme limits. They know that they must balance bandpass, distortion, noise, and hum against stability with all types of loads, extensions of mean time-before-failure characteristics. Most high quality sound reinforcement amplifiers incorporate an output transformer, giving us 70 V, 25 V, and 4 Ω, 8 Ω, and 16 Ω outputs. In fact, connecting across the 4 Ω and 8 Ω taps yields a 0.69 Ω output. Example: Let the rms speech value be $L_P = 65$ dB at 2 ft in the 1000–2000 Hz octave band, Fig. 7-21. Let the ambient noise level be $L_P = 32$ dB with the air conditioning on and 16 dB with the air conditioning off in the same octave band, Fig. 7-22. With the air conditioning on the signal to noise ratio (SNR) is:

Figure 7-19. Measurement of harmonic distortion.

Figure 7-20. Methods of measuring distortion. Use if available Sound system loudspeaker Sound level meter

Sound system power amplifier

Sinewave oscillator 1/3 Octave wave analyzer Graphic level recorder

$R$ $e$ $s$
Ambient noise "Band rejection"

distortion analyzer "Band pass" wave analyzer f = 0 dB
down 5 dB 3f = 25 dB down 2f = 36 dB down 500 1k
2k 5k Frequency–Hz

Percentage 100 10 dB± 20 ------------×=
100 10 35– 20 --------
× 1.78%=

20 100 ± x% 100 ------------------------log dB change=
20 0.8 100 ------------log 42 dB= 

20 0.001 100 ----------------log 100= 10 100 10
--------10,000,000,000= L P Diff L P Total L P Noise = 100
10 102– 20 ------------× 0.00078% (7-56)

and with the air conditioning off:

For a harmonic to be equal to -33 dB, its percentage
would be:

For a harmonic to be equal to -49 dB, its percentage
would be:

7.27 Decibels and Percentages

The comparison of data in decibels often needs to be

expressed as percentages. The measurement of THD compares
the harmonics with the fundamental. After finding out how
many dB down each harmonic is compared to the fundamental,
sum up all the harmonics and then compare their sum to the
fundamental value. The difference is expressed as a
percentage. The efficiency of a loudspeaker in converting
electrical energy to acoustic energy is also expressed as a
percentage. We know that: Therefore, a signal of -20 dB is
1/10 of the fundamental, or $100 \times \frac{1}{10} = 10\%$. A signal of -40 dB is $1/100$ of the fundamental, or $100 \times \frac{1}{100} = 1\%$. A signal of -60 dB is $1/1000$ of the fundamental, or $100 \times \frac{1}{1000} = 0.1\%$. We can now turn this into an equation for finding the percentage when the level difference in decibels is known. For such ratios as voltage, SPL, and distance: (7-57) For power ratios: (7-58)

Occasionally we are presented with two percentages and need the decibel difference between them. For example, two loudspeakers of otherwise identical specifications have differing efficiencies: One is 0.1% efficient, and the other is 25% efficient. If the same wattage is fed to both loudspeakers, what will be the difference in level between them in dB? Since we are now talking about efficiency, we are talking about power ratios, not voltage ratios. We know that: and so forth. A 0.1% efficiency is a power ratio of 1000 to 1, or -30 dB. We also know that -3 dB is 50% of a signal, so -6 dB would be 25%; (-6) - (-30) = 24 dB. In other words, there would be a 24 dB difference in level between these two loudspeakers when fed by the same signal. Some consumer market loudspeakers vary this much in efficiency.

Figure 7-21. Male speech, normal level 2 feet from the microphone.

Figure 7-22. Ambient noise levels.

<table>
<thead>
<tr>
<th>SNR 65 dB</th>
<th>32 dB</th>
<th>33 dB</th>
</tr>
</thead>
<tbody>
<tr>
<td>SNR 65 dB</td>
<td>16 dB</td>
<td>49 dB</td>
</tr>
</tbody>
</table>

100  10  33– 20 --------× 2.24%

100  10  49– 20 --------× 0.355% = 90  80  70  60  50  40  30  L P – d B 16 31.5 63 125 250 500 1k 2k 4k 8k 16k

Frequency–Hz Peak rms 50 40 30 20 10 0 -10 L P – d B 16 31.5 63 125 250 500 1k 2k 4k 8k 16k

Frequency–Hz Air conditioning "on" Instrument threshold Air conditioning "off" 20 10 log 20 dB = 20 100 log 40 dB = 20 1000 log 60 dB = Percentage 100 10 dB± 20 --------×=

Percentage 100 10 dB± 10 --------×= 10 10 log 10 dB = 10 100 log 20 dB = 10 1000 log 30 dB =

7.28 Summary

The decibel is the product of the greatest engineering minds in communications early in the last
When it is combined with the work of Oliver Heaviside and others on impedance at the turn of the 20th century, we are equipped to handle audio levels. The concepts of dB, Ω, and dBm are the tools of the professional as well as their language.


Research Council of the Academy of Motion Picture Arts and Sciences, Motion Picture Sound Engineering. New York: Van Nostrand, 1938.

Chapter 8 Interfacing Electrical and Acoustic System


_____. “The Ultimate Noise,” db Magazine (June 1969)


Chapter 9 Loudspeaker Directivity and Coverage


Chapter 10 The Acoustic Environment


Figure 10-16. Comparison of direct, early, and reverberant sound fields in an auditorium (reflection angles adjusted for purposes of illustration). Source 1 2 2 2 3 3 3 3 3

1 Direct field

2 Early field

3 Reverberant field Figure 10-17. Graphic representation of near field, free field, and reverberant field. log r S P L

Chapter 11 Audio and Acoustic Measurements


12 Chapter 12 Large Room Acoustics


______. “Sabine’s Reverberation Time and Ergodic Auditorium,” J. Acousti. Soc. Am., Vol. 58 (1975), pp. 643-655. Figure 12-14. (cont.) Complete study of a sound system where the direct-to-reverberant ratio is degraded as each loudspeaker is added. Prepared by Rollins Brook of BBN. 6 dB 6 dB 6 dB 6 dB 1 2 3 4 6 dB 6 dB 6 dB 6 dB 1 2 3 4 Row 26 Row 31 Direct to reverberant ratio: 8.55 dB Direct to reverberant ratio: 6.17 dB Direct to reverberant ratio: 4.17 dB Direct to reverberant ratio: 7.51 dB Direct to reverberant ratio: 4.29 dB Direct to reverberant ratio: 0.87 dB Direct to reverberant ratio: 1.17 dB


14 Chapter 14 Designing for Acoustic Gain


15 Chapter 15 Designing for Speech Intelligibility


Chapter 16 What is Waving and Why


\[ p \omega \rho \omega^2 \sum_{\theta} \frac{1}{d \cos(\theta)} = \]

Figure 16-28. Polar plot of the normalized directivity of an acoustic dipole. 90 60 30 0 330 300 270 240 210 180 150 120 1 0.8 0.6 0.4 0.2
17 Chapter 17 Microphones


P. M. Morse, Vibration and Sound, 2nd ed. New York: Mc-Graw-Hill, 1948. \[ p \gamma P \theta A \varepsilon \gamma m 2\pi ft(\cos \theta) \]

\[ \frac{V \theta}{\sqrt{\frac{1}{2}}} = p_{rms} \gamma P \theta 2 \quad \quad \quad \frac{A \varepsilon m}{\gamma} \]

\[ \{ \} = CP \theta = \]
Chapter 18 Loudspeakers and Loudspeaker Arrays


21 Chapter 21 Signal Delay and Signal Synchronization


22 Chapter 22 Signal Processing


Figure 22-86. Attack behavior of a limiter. Dynamics Attack
\[ \text{Time-seconds} +2 \]
\[ +1.5 \]
\[ +1.0 \]
\[ +0.5 \ 0 \]
\[ -0.5 \]
\[ -1.0 \]
\[ -1.5 \]
\[ -2.0 \]
\[ V \]
\[ o \ l \]

\[ t \ s \] Figure 22-87. Release behavior of a limiter. Dynamics Release
\[ \text{Time-seconds} +1.5 +1.0 +0.5 0 -0.5 -1.0 -1.5 V o l t s \]
\[ 1 0.5e \ t 0.005-( ) \tau/- -( )±= \]
24 Chapter 24 Sound System Equalization


Chapter 25 Putting It All Together

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AES-3id-2001

AES42-2001


Figure 25-28. Ethernet with CobraNet®. A A A A A A CD
CK Ur A = Audio CD = Control data Ck = clock Ur =
Unregulated data E = Ethernet E