The Audio Handbook

Gordon J. King

THE AUDIO HANDBOOK

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Foreword

by Donald Aldous

Technical Editor, *Hi-Fi News & Record Review*

Do you know the Zanzibar Fallacy? Octogenarian Percy Wilson, the first and most distinguished British reviewer of hi-fi equipment, likes to tell this story: "On the Isle of Zanzibar, there is a town clock at one end and a cannon at the other. Each day, at noon, the clock strikes twelve and the cannon is fired. When a visitor enquired of the keepers of the clock, how the time was set, he was told, 'By the sound of the cannon'. When he asked how the noon time was checked, he was told; 'By the striking of the clock'."

Many of the audio articles and discussions today on hi-fi equipment are of this kind, but Gordon King has always been aware of the dangers of opinions and information based on mistaken logic or lack of observed facts. Continually adding to and upgrading his test equipment, standards and knowledge, Gordon King is in the direct line of the great reviewers. Totally professional in his approach, and in the preparation of material for the Press—as those of us who work with him will know—he retains the enthusiasm of the amateur 'in love' with his hobby.

Towards the end of the war years, Gordon served in India with a Special Communications Unit, and he likes to reminisce on some experiments with a design of 20 watt amplifier using transmitter modulation transformers as output transformers and transmitting valves as power amplifiers. Other areas he devoted time to included the see-saw phase-splitter and primitive wire recorders and disc recorders for sending home Forces' messages.

On returning to civilian life, like many veteran audiophiles, he constructed Williamson amplifiers, and later joined a company directly concerned with the design and manufacture of audio equipment, and making gramophone records.

The full impact of Gordon King's work, both as a development engineer and as a technical writer in the fields of audio, radio and television, is now known throughout the world. This present book I regard as his most important to date on audio. I enjoyed reading it immensely and it cannot but add to his reputation.

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Introduction

A BRIEF HISTORY of this current volume would not be amiss. Back in 1959 I wrote a book called *The Practical Hi-Fi Handbook.* This was published by Odhams Books and was very well accepted, requiring several reprints to satisfy the UK requirements. There was also a Spanish edition. The book was revised and extended in 1969 to take in transistor equipment, tape recording and videotape recording and was published by Newnes-Butterworths under the new title *The Hi-Fi and Tape Recorder Handbook.*

Such has been the advances in the world of high quality audio reproduction that this third design has been demanded. This time, however, the book has been completely rewritten and no longer resembles the earlier volumes, and the generic term 'audio' has been used in the title. At one time equipment with the label 'hi-fi' was several planes higher in design detail and hence in potential quality of reproduction. This is still true to varying degrees even today, but the art of solidstate electronics has evolved such that the distinction between 'basic' audio equipment and equipment carrying the hi-fi tag is diminishing, and from the electronics side, anyway, it appears that before very long there will not be all that much difference between the important parameters of the two classes of equipment.

There will always be a call for the 'super' class of equipment, of course, designed for the enthusiast satisfied only with extra high power at barely measurable levels of harmonic and intermodulation distortion and with facilities and features well in advance of the economics of the basic class.

Electronics aside, some of the greatest differences lie in the signal sources feeding the amplifiers and the loudspeakers fed by them. High quality record playing units with minimal rumble turntables of constant speed and low wow and flutter, with low-tracking magnetic cartridges and partnering low inertia, low bearing friction arms cost money. The same is true of top-flight loudspeaker systems capable of yielding domestic scene sound pressure peaks towards 100 dB at low distortion and with extended frequency response, particularly at the bass end, and minimal coloration.

Tape machines are other items where quality is more reflected by price, and this is particularly true of the latest cassette machines incorporating noise reducing artifices like Dolby B. The low linear velocity of cassette tapes demands highly sophisticated tape transport mechanisms if speed inconstancy and wow and flutter are to be reduced to an acceptable 'hi-fi' level, while highly accurate transducer head engineering is a prerequisite for extended frequency response and minimal crosstalk. The microgaps essential for extended high-frequency response are more prone to wear and distortion by the passage of the tape past the heads than those of some of the earlier reel-to-reel machines, and for this reason more costly head designs—some composed of glass-ferrite—are incorporated in the better class machines. All these features cost more money than the partnering electronic circuits, some of which in replay machines only are not unduly complex. Dolby B, essential for an acceptable signal/noise ratio at the lower tape velocities, is a more complicated circuit addition, though, as also is that of any of the other recently developed noise reducing systems.

Dolby B, for example, together with the relatively recent high energy (such as CrO_2) tapes, have significantly improved the dynamic range potential of the cassette as a programme signal source, even at the dramatically low velocity of 4.76 cm/sec. The Philips system, which provides stereo (mono compatible) in two-channel pairs (four tracks in all), is the existing 'standard', and most serious cassette machines are correspondingly engineered.

For 'four-channel stereo' or quadraphony both the disc and the standard 6.3 mm ($\frac{1}{4}$ in) tape (cartridge housed) are in current application. At the time of writing no definite four-channel standard has evolved for the cassette, though a two-channel matrix arrangement, using the two-channel pairs of the standard cassette tape would be feasible. This would then be rather similar to the matrix four-channel disc. A four-channel cassette deck has been made by JVC and H-K.

In addition to this type of disc, however, there is also the so-called 'carrier' disc where the rear information constitutes the modulation of a subcarrier recorded in the single groove, which also accommodates the front left and right information in the conventional way. This type of disc is often referred to as four-channel discrete because the four channels are handled from start to finish in essential isolation (i.e. the JVC CD-4 system, where C stands for compatible—meaning that the disc will play in stereo on a stereo system—D stands for discrete, and the 4, of course, indicating four channels).

The number of channels involved in the passage of the information from microphone to loudspeakers, via the recording medium, is commonly given in terms of 4-4-4 for the discrete system and 4-2-4 for the matrix system. The first 4 means that the signals are in four channels to start with (i.e. four microphones). The second number tells how many isolated channels there are in the recording process. In the discrete system, of course, there are four and in the matrix system only two. The third indicates the number of replay channels (i.e. how many separate amplifiers and loudspeakers are used). On this basis, therefore, a two-channel stereo system could be designated 2-2-2.

Pseudo-quadraphony, sometimes called 'surround sound' or 'ambiophony' is designated 2-2-4, which implies that the four loudspeakers obtain their signals from the two-channel source, which was also transmitted or recorded in two channels. The signals for the rear loudspeakers are, in fact, derived from the differential of the left and right stereo signals, so the system is not true four-channel. However, since concert hall ambience is responsible for some of the antiphase or differential components between the left and right channels, the rear loudspeakers respond to this and enhancement of the stereo reproduction is sometimes experienced.

The radio tuner as a programme source is assuming greater importance, particularly now that the audio quality is being improved by the pulse code modulation links between studio centres and the v.h.f./f.m. transmitters, which is also resulting in the stereo service of the BBC being extended over the country. In fact, with a well programmed 'live' transmission the quality of the audio signal from a good f.m. stereo tuner is above that obtainable from any other contemporary source—except, perhaps, from microphones direct!

While the gulf between the detailed electronics of the true hi-fi tuner and the basic counterpart is still quite wide, a well designed f.m.-only basic type with inbuilt decoder is nevertheless capable of providing a very acceptable programme signal when operating under good reception conditions. As with amplifiers, the more expensive models include facilities and features which may not be fundamental to the signal quality. For example, one might pay extra for a.m. bands, for elaborate tuning mechanisms and enclosures, for switchable filters and other technical features which themselves are not directly responsible for the quality of the resultant audio signal.

On the other hand, acceptable reception under adverse conditions, as for example when the aerial signal is weak or when it is required to tune a weak signal which is located close in frequency to a much more powerful one, requires a tuner in which the various circuits have been deliberately tailored to take account of such reception difficulties, and these, of course, have to be paid for.

Since *The Practical Hi-Fi Handbook* was written in 1959, therefore, the standard of basically acceptable quality reproduction has been elevated by innovation and development in almost all associated areas of the art. Apart from television sound and the small transistor radio, stereo reproduction is now the norm. In fact, hi-fi and stereo are often regarded as synonymous in this audio age; but stereo does not change by some magic mediocre quality audio into hi-fi.

From two channels we are now graduating to four; but the practice here is little more advanced than was stereo when *The Practical Hi-Fi Handbook* was launched. Technically, however, developments abound, though there is still some way to go before universal four-channel standards can be expected. In spite of this, both the hardware (the mechanics and electronics) and the software (discs and tapes) are available for those wishing to join the early pioneers.

What is happening, therefore, is that hi-fi in the strict term is embracing new areas and that audio which a decade or two back would have been regarded as hi-fi is now almost 'domestic quality'.

In writing this new book emphasis has been removed from the topic of servicing (since from the audio aspects this is covered in my companion volumes *Radio and Audio Servicing Handbook* and *F.M. Radio Servicing Handbook*, both by the same publisher as this book) and applied more to the new arts and the sciences relating to them, thereby making it a book of wider appeal not only to the service technician and hi-fi dealer, but also to the audiophile and music lover desirous of keeping abreast of the changing face of home audio reproduction.

In conclusion, I should like to acknowledge the encouragement given by colleagues throughout the audio industry from those concerned with making the records to those whose job it is to design the electronics, including the editors of the hi-fi and audio magazines in which area I work. Special thanks are offered to my friend Donald Aldous for his helpful suggestions, for reading the proofs and for contributing the Foreword.

Brixham, Devon Gordon J. King

CHAPTER 1

Audio Fundamentals

EQUIPMENT DESIGNED for high quality sound reproduction differs from the basically 'domestic' type of equipment in that extra special attention is given to a large number of small points of detail. This is what costs the extra money. Minor changes, either as the result of alteration in value of a component or components or unskilled servicing, will disturb the critical design balance and in some way or other impair the quality of reproduction. Distortion, for example, may increase or the frequency response may suffer.

The service technician insensitive to the hi-fi equation may not be aware of these shortcomings; but the hi-fi perfectionist, whose equipment it is, will certainly notice a difference. The technician is thus obliged to employ accurate audio test equipment to judge the standard of the equipment before and after repair or adjustment. Listening tests waste time and lead to frustration. It is the technician's job to ensure that the equipment matches the parameters of its specification.

Experience has shown that hi-fi types fall into three main categories. There is the music lover whose primary desire is to play his favourite records with the least apparent distortion. This type is generally the least technically exacting since fair quality reproduction is adequate to re-create in his mind the atmosphere of the concert hall—slight distortion and other minor shortcomings—thus go unnoticed.

Then there is the technical perfectionist whose senses are sharply focused on the various technical parameters of the equipment. This type may not possess a highly developed aesthetic interest in music, but he is often able to judge with uncanny accuracy whether there is more distortion than there should be; whether extra damping would improve the bass delivery of the loudspeakers; whether there is a droop or peak in the overall frequency response; and similar technical matters. He secures satisfaction from listening to loud reproduction at low distortion, and when *he* says that the distortion, etc. is higher than it should be it is prudent for the technician not to disagree with him—until he can prove otherwise more objectively, of course!

The third type is a mixture of the previous two, and here is represented the large majority of hi-fi enthusiasts and audiophiles. This type is an enthusiast because he is both technically interested and at the same time a true music lover.

The audio technician, therefore, should be aware of these three shades of enthusiast. The technician should desirably have an interest in music and attend live concerts if only to acquaint himself with the real thing! The technician practising in the high quality audio field needs to be something of a psychologist as well as an engineer, and it is just as well to know one's subjects. Sound as it is heard is a purely subjective function, so from here let us proceed.

SOUND

The source of any sound is always in some state of vibration. This is clearly demonstrated by the piano string, the tuning fork, the cone of a loudspeaker unit, etc. The vibration may be so slight or so rapid as not to be visible, while it may be large and relatively slow as to be clearly visible, as in the case of a very loud mains hum emanating from the loudspeaker. There is no future in removing the vibration in the latter case by glueing the speech coil to the magnetic pole pieces—a condition once observed by the author when investigating a system for lack of output! When questioned, the owner was true in principle by commenting 'but I got rid of the terrible hum which was being caused by this cone thing vibrating' (a true story, incidentally).

With organ pipes and other wind instruments the source is air vibration. The vibration can often be experienced by placing a finger over the pipe, etc. or on the string or loudspeaker cone. It is surprising how sensitive the finger can be in detecting vibration. In fact, some engineers check for slight mains hum—which may be inaudible—by placing a finger lightly on the cone.

A sounding source causes the surrounding air to be alternately compressed and rarefied in sympathy with the vibrations. A pulse of high pressure is thus followed by a pulse of low pressure and so on.

LONGITUDINAL WAVES

Sound waves in air are known as *longitudinal* waves, which implies that the molecules of the air travel to and fro in a path which is parallel to the direction of the wave propagation, and that each molecule executes the same motion as the one before it but a small interval of time later.

Waves associated with the electromagnetic family—radio, light, etc.—are known as *transverse* waves because the particles of the medium travel in a path perpendicular to, instead of parallel to, the propagation of the wave.

SOUND VELOCITY

The speed at which a sound wave travels—called the velocity of propagation—depends on the physical properties of the medium, the relationship when the medium is a gas being

$$c = \sqrt{\frac{\gamma p}{\rho}} \tag{1.1}$$

where c is the velocity, γ the ratio of specific heats (which is a constant and about 1.4 for atmospheric air), p the pressure (which is nominally 10^6 dynes/cm^2 corresponding to 1 bar or, in SI units, to 10^5 newtons/m^2 and ρ the density (which for air is close to 1.2 kg/m^3).

The velocity through atmospheric air at sea level and 20 °C is close to 344 m/s (1 130 ft/s). The velocity in a given gas medium and at constant temperature is thus dependent on density and 'compressibility', but because the density is proportional to the pressure, the velocity is independent of pressure over a wide range, but changing temperature changes the velocity.

Velocity is also related to the frequency f and the wavelength λ of the sound by

$$c = \lambda f \tag{1.2}$$

so that the wavelength can be found by dividing the velocity by the frequency. For example, the wavelength of a sound of 50 Hz (the mains frequency) is 6.88 m (22 ft 7 in). At 20 Hz the wavelength is 17.2 m (56 ft 4 in), while at 5 kHz it is 6.9 cm (2.7 in). Thus the wavelength diminishes as the frequency increases. A knowledge of the wavelength can be useful when investigating for standing waves (eigentones) in the listening room, as well as for other aspects of the audio art.

To summarise, therefore, any gas, such as air, consists of a large number of molecules moving rapidly at random. A pressure is thus experienced by an object within the gas of a value dependent on the number of molecules per unit volume and on their kinetic energy; in other words, on the barometric pressure of the gas (air) and on its temperature.

SOUND PRESSURE

The normal atmospheric pressure is about 10^6 dynes/cm^2, which in contemporary term corresponds to 1 bar or to 10^5 pascals (Pa), equivalent to 10^5 N/m^2 in SI units. Sound waves cause variations within this normal pressure over the range from about 20 μPa (micro indicating one-millionth) to 60 Pa—from the quietest sounds round the threshold of hearing to sounds of intensity into the threshold of pain.

A sound wave is thus characterised by an oscillatory (depending upon the nature of the sound) variation in the air pressure above and below the prevailing atmospheric pressure and by a corresponding to and fro velocity of the gas molecules about the random gaseous velocity. The *power* propagated by a sound wave is the product of the sound particle velocity about the random velocity and the acoustical pressure about the normal atmospheric pressure.

SOUND POWER

The mean power per unit area can thus be expressed as

$$W_a = pu \tag{1.3}$$

where p is the r.m.s. gas pressure and u the r.m.s. particle velocity. From the electrical point of view, p can be regarded as the analogue of voltage and u as the analogue of current, their product giving the power.

The specific radiation impedance Z is given by

$$Z = \rho c \qquad (1.4)$$

where c is the velocity of propagation and ρ the gas density. Thus W_a can also be expressed as

$$W_a = Zu^2 \qquad (1.5)$$

or as

$$W_a = \frac{p^2}{Z} \qquad (1.6)$$

Z is often given as 40.7 acoustical ohms in c.g.s. units or 407 acoustical ohms in SI units, so by using the appropriate value in, say, expression 1.6 it can be calculated that a sound pressure of 20 μbars (2 Pa) gives W_a a value of 9.828 ergs/cm^2 or 98 280 $\times 10^{-7}$ joule/m^2, equal to 98 280 ergs/m^2. Taking these respectively as 10 ergs/cm^2 and 100 000 ergs/m^2, and since 1 watt is equal to 10^7 ergs (1 joule) per second, we derive unit powers of 1 μW/cm^2 and 10 mW/m^2.

SPHERICAL WAVES

When the wavefront is at right-angles to the direction of propagation the waves are called plane. Under most practical conditions, however, the wavefront expands non-uniformly, such that in unrestricted free space, the waves radiate outwards as an expanding sphere (spherical waves). The *power* per unit area then falls inversely as the square of the distance and the *pressure* inversely as the distance.

The mean power W_a per unit area due to a spherical wave is given by

$$W_a = \rho c u^2 \frac{2\pi^2 r^4}{\lambda^2 d^2} \qquad (1.7)$$

while the r.m.s. pressure p (when the distance is greater than the wavelength) is given by

$$p = \rho c u \frac{\sqrt{(2)}\pi r^2}{\lambda d} \qquad (1.8)$$

where in both expressions ρ is the gas density, c the sound velocity, u the r.m.s. particle velocity, r the radius of the wavefront, d the distance from source and λ the wavelength.

AMPLITUDE OF SOUND WAVE

The loudness of a sound is governed by the amplitude of the wave and hence on the energy carried by it to the ear, since sound is ultimately perceived by our faculty of hearing.

The r.m.s. amplitude of a plane wave is given by

$$a = \frac{u}{2\pi f} \quad \text{or} \quad \frac{p}{\rho c 2\pi f} \qquad (1.9)$$

and of a spherical wave by

$$a = \frac{u}{2\pi f}\frac{\sqrt{(2)}\pi r^2}{\lambda d} \tag{1.10}$$

when the distance is greater than the wavelength, where a is the amplitude, u the r.m.s. particle velocity, f the frequency, r the radius of wavefront, d the distance and λ the wavelength.

CHARACTERISTICS OF SOUND

The pitch of a sound is determined by the frequency and hence wavelength, while the quality or timbre, which distinguishes between notes of the same pitch sounded by different instruments, is determined by the harmonic composition of the sound, which can be regarded for the present as subsidiary vibrations whose frequencies are multiples of the fundamental vibration.

As the frequency is reduced, the note eventually becomes resolved into the separate impulses of which it is composed. As the frequency is increased, the note becomes very shrill, and around 15 kHz it tends to fade into a very high pitched hiss.

The high-frequency limit of audibility is related to age. With young people it is around 20 kHz, falling possibly to 6 kHz or less in old age. Some young people are uncomfortably conscious of the line timebase whistle of television receivers, while older people are not at all disturbed. The squeak of a mouse is commonly inaudible to people in their fifties yet dramatically apparent to the younger person.

TRANSIENTS

Nevertheless, a person whose hearing is failing at the high-frequency end of the spectrum due to normal age is still able to appreciate music containing harmonic components extending above his cutoff frequency. It is thus still necessary for audio equipment used by such a person to be capable of responding to high audio frequencies. One reason for this is that parts of music waveforms consist of steep, rapidly occurring wavefronts, called *transients*. These can be resolved into a large number of harmonic components, so attenuating the high-frequency response has the effect of reducing the rate of rise of the wavefronts and of diminishing the amplitude of the transients. Since transients are responsible for the 'attack' of music reproduction, impairing these such that their acceleration and amplitude are reduced is obvious equally to persons with and without extended high-frequency hearing response.

THE DECIBEL

The pressure and energy (power) of a sound wave are often expressed as a decibel (dB) unit, so we must now see what is meant by this.

The dB expresses the ratio of two powers logarithmically—ten times the common logarithmic ratio in fact, so mathematically we have

$$\text{dB} = 10\log_{10}(W/W_o) \tag{1.11}$$

where (W/W_o) is the ratio of the two powers concerned. Because the dB value implies a *ratio*, the reference power W_o must always be stated or at least clearly understood.

Now, since sound power is proportional to the sound pressure *squared*, the number of decibels between two pressures is given by

$$\begin{aligned} dB &= 10 \log_{10}(p/p_o)^2 \\ &= 20 \log_{10}(p/p_o) \end{aligned} \tag{1.12}$$

where (p/p_o) is the ratio of the two pressures, the implication again being that the reference pressure p_o must be stated or understood.

By international agreement, the reference sound pressure is 0.0002 dyne/cm^2 or 0.0002 microbar, and the SI equivalent is 20 micronewtons per sq. m (20 μN/m^2) which corresponds, again by international agreement, to 20 micro-pascal whose symbol is μPa. Thus 0 dB, the reference level, corresponds to 20 μPa or 0·0002 microbar, which is the sound pressure approximating to the threshold of hearing.

When the pressure of a sound wave exceeds about 120 dB a tickling sensation and ultimate pain is experienced by a listener. 120 dB corresponds to a pressure ratio of 10^6:1, which means, then, that the pressure at this level is 200 μb (20 Pa). Thus, from the practical point of view the human ear is able to accommodate a pressure range from 0.0002 (20 μPa) to 200 μb (20 Pa) for the softest sounds (threshold of hearing) to the loudest sounds (approaching the threshold of pain). In actual fact, the upper limit is more like 130 dB, corresponding to a sound pressure of just over 600 μb (60 Pa), but then the sound experience is so intense as to be really painful and damaging.

Table 1.1 gives some examples of sounds, their pressures in μb and the dB equivalents.

PER UNIT SOUND POWER

We have seen (expressions 1.3 and 1.7) that power is propagated by a sound wave as a function of the sound pressure and particle velocity. At 0 dB (0.0002 μb pressure) the power is 10^{-12} W/m^2, which has international acceptance. Thus at 100 dB the mean power per m^2 is 10^{-2} watt (10 mW). At 120 dB the power is 1 W/m^2 and at 130 dB 10 W/m^2. Thus from the threshold of hearing (0 dB) to 120 dB the power or energy range is a massive 10^{12}:1, which is also shown in Table 1.1. The pressure range is 10^6:1, as we have seen.

All this may give thought as to why such high power amplifiers are used for domestic sound reproduction. The reason for the need for high electrical power will become apparent later.

THE DECIBEL AND RATIOS OF ELECTRICAL UNITS

Before we leave the subject of decibels, mention must be made of the fact that they are used also to represent ratios of power, voltage and current in electrical circuits.

For electrical power ratios expression (1.11) is applicable and for voltage ratios expression (1.12) is applicable, but then the ratio is (V/V_o), where V_o is the reference voltage. Expression (1.12) is also applicable to current ratios, the ratio then being (I/I_o), where I_o is the reference current.

However, it is important that the resistance R in which current I or voltage V operates is common to both sides of the ratio in the expressions. When R

Table 1.1. EXAMPLES OF SOUNDS, THEIR PRESSURES, RELATIVE ENERGIES AND DECIBEL EQUIVALENTS

Noise	*Decibels*	*Relative energy*	*Pressure microbars*	*Typical examples*
	120	1 000 000 000 000	200	
Deafening	110	100 000 000 000		Jet aircraft at 150 m (500 ft) Inside boiler-making factory 'Pop' music group Motor horn at 5 m (16 ft)
———	100	10 000 000 000	20	———
Very loud	90	1 000 000 000		Inside tube train Busy street Workshop Small car at 7.5 m (24 ft)
———	80	100 000 000	2	———
Loud	70	10 000 000		Noisy office Inside small car Large shop Radio set—full volume
———	60	1 000 000	0.2	———
Moderate	50	100 000		Normal conversation at 1 m (3 ft) Urban house Quiet office Rural house
———	40	10 000	0.02	———
Faint	30	1 000		Public library Quiet conversation Rustle of paper Whisper
———	20	100	0.002	———
Very faint	10	10		Quiet church Still night in the country Sound-proof room Threshold of hearing
	0	1	0.0002	

differs from side to side, the following expressions should be used

$$\mathrm{dB} = 20\log_{10}(V/V_o) + 10\log_{10}(R/R_o) \tag{1.13}$$

$$\mathrm{dB} + 20\log_{10}(I/I_o) + 10\log_{10}(R/R_o) \tag{1.14}$$

where R_o is the resistance of the circuit referring to the reference voltage or current and R is the resistance of the circuit in which the other side of the ratio is measured.

When the dB number is known, the power, current or voltage ratio can be found from

$$(W/W_o) = \text{antilog}\, \text{dB}/10 \qquad (1.15)$$

$$(I/I_o) = \text{antilog}\, \text{dB}/20 \qquad (1.16)$$

$$(V/V_o) = \text{antilog}\, \text{dB}/20 \qquad (1.17)$$

where dB is the decibel *number*.

To save the toil of calculation dB tables have been evolved, samples of which are given in Tables 1.2 and 1.3. Figures not given in these can easily be calculated. For example, when two dB numbers are added their ratios are multiplied.

Table 1.2. CONVERSION OF DECIBELS TO POWER AND VOLTAGE/CURRENT RATIOS

dB	*Power Ratio*	*Voltage Ratio*	*dB*	*Power Ratio*	*Voltage Ratio*
1	1.26	1.12	15	31.6	5.62
2	1.58	1.26	20	100	10
3	2.0	1.41	30	1 000	31.6
4	2.51	1.58	40	10^4	10^2
5	3.16	1.78	50	10^5	316
6	3.98	2.0	60	10^6	10^3
7	5.01	2.24	70	10^7	3 160
8	6.31	2.51	80	10^8	10^4
9	7.94	2.82	90	10^9	31 600
10	10	3.16	100	10^{10}	10^5

Table 1.3. CONVERSION OF POWER RATIOS TO DECIBELS

Power Ratio	*dB*	*Power Ratio*	*dB*	*Power Ratio*	*dB*	*Power Ratio*	*dB*
1.0	0.000	3.3	5.185	5.6	7.482	7.9	8.976
1.1	0.414	3.4	5.315	5.7	7.559	8.0	9.031
1.2	0.792	3.5	5.441	5.8	7.634	8.1	9.085
1.3	1.139	3.6	5.563	5.9	7.709	8.2	9.138
1.4	1.461	3.7	5.682	6.0	7.782	8.3	9.191
1.5	1.761	3.8	5.798	6.1	7.835	8.4	9.243
1.6	2.041	3.9	5.911	6.2	7.924	8.5	9.294
1.7	2.304	4.0	6.021	6.3	7.993	8.6	9.345
1.8	2.553	4.1	6.128	6.4	8.062	8.7	9.395
1.9	2.788	4.2	6.232	6.5	8.129	8.8	9.445
2.0	3.010	4.3	6.335	6.6	8.195	8.9	9.494
2.1	3.222	4.4	6.435	6.7	8.261	9.0	9.542
2.2	3.424	4.5	6.532	6.8	8.325	9.1	9.590
2.3	3.617	4.6	6.628	6.9	8.388	9.2	9.638
2.4	3.802	4.7	6.721	7.0	8.451	9.3	9.685
2.5	3.979	4.8	6.812	7.1	8.513	9.4	9.731
2.6	4.150	4.9	6.902	7.2	8.573	9.5	9.777
2.7	4.314	5.0	6.990	7.3	8.633	9.6	9.823
2.8	4.472	5.1	7.076	7.4	8.692	9.7	9.868
2.9	4.624	5.2	7.160	7.5	8.751	9.8	9.912
3.0	4.771	5.3	7.243	7.6	8.808	9.9	9.956
3.1	4.914	5.4	7.324	7.7	8.865	10.0	10.000
3.2	5.051	5.5	7.404	7.8	8.921		