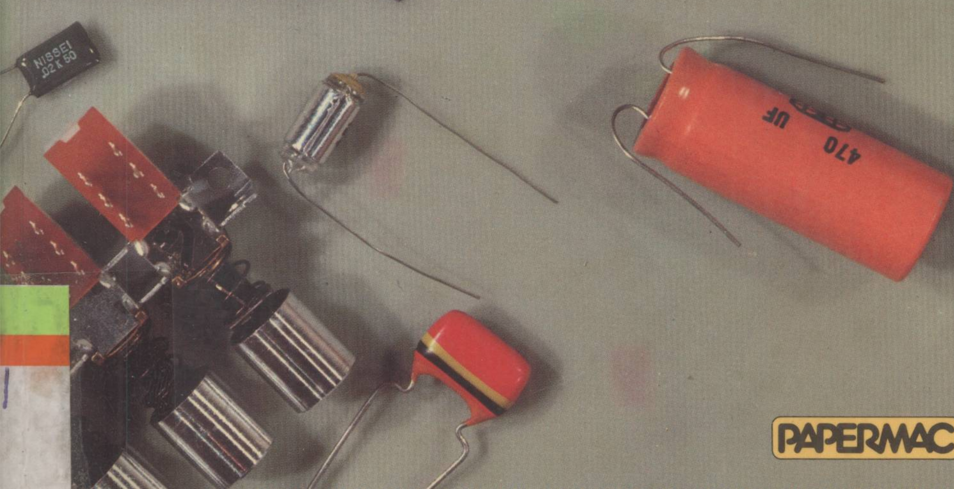
ELECTRONIC PROJECTS 3

AUDIO CIRCUITS AND PROJECTS

Graham Bishop



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Electronic Projects 3

Audio Circuits and Projects

Electronic Projects 1 – Cost-effective Projects Around the Home
John Watson

Electronic Projects 2 – Projects for the Car and Garage
Graham Bishop

Electronic Projects 3 – Audio Circuits and Projects
Graham Bishop

Electronic Projects 4 – Test Gear Projects
Terry Dixon

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Preface

This book is one of a series of electronics hobbies books for electronics constructors at all levels. Almost 100 circuits are described with constructional details given where necessary. The circuits range from those with no integrated circuits or transistors to those with over 15 ICs; all are within the capabilities of the beginner to electronics building.

The first two chapters include some theoretical introduction to the world of audio electronics, where the definitions and jargon are explained. Many hints on construction techniques are also given and the reader is well advised to read these two chapters before plunging into the circuits in the rest of the book. Hints on the purchase of hi-fi equipment and components are included. The circuits of chapters 3 to 7 can be considered as individual modules which can be interlinked to form more complex circuits if desired. A preamplifier and three filters from chapter 3, a power amplifier from chapter 4, a phasing circuit and automatic fade circuit from chapter 5, a sound-to-light unit from chapter 6 and a power supply from chapter 4 together produce a fully comprehensive disco unit with far more facilities than many commercial models.

Neatness of construction and observance of safety precautions are essential. Long wires and shoddy workmanship lead to poor performance, impossible fault-finding and a dangerous piece of equipment, particularly if mains voltages are involved. Constructors need very few tools – a good pair of miniature cutters and pliers with a set of various sized screwdrivers are essential. A 15 or 25 W soldering iron with small bit is required and a simple multimeter for the normal ac/dc ranges are all that are necessary. A signal generator and oscilloscope are useful but not essential – these can often be borrowed.

Components come in all sizes and prices. Bulk packs of resistors, capacitors, transistors, diodes and other components are usually a bargain. Some packs are untested but are quite adequate for the circuits in this book. A diode is tested with an ohmmeter reading about 1k in one direction and several 100k in the other direction. A transistor is tested in the same way, treating the base–emitter and base–collector terminals as two diodes.

Electronics construction is an enjoyable hobby and it should always remain so. It will not be frustrating and expensive if the simple instructions are followed. I hope that you will get much enjoyment out of reading this book and carrying out the circuit-building and testing.

G. D. BISHOP

Publisher's Note

While every effort has been made to ensure the accuracy of the projects and circuits in this book, neither the Publishers nor the author accept liability for any injury or loss resulting from the construction of any of the designs published herein.

The Publishers will, however, be pleased to hear from readers who have corrections to the text or genuine queries, and will refer any such queries to the author.

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Why Amplifiers?

You may remember seeing old 'His Master's Voice' record labels with a dog listening to the sounds emerging from the horn of an early phonograph. In those days there were no valves or transistors, so manufacturers had to rely on mechanical 'amplification'. The gramophone could produce sounds which, although limited in quality, could fill a medium-sized room. The large horn was a form of acoustic impedance-matching device, its shape being carefully designed (like many brass musical instruments) to throw the maximum amount of sound outwards. Such horns are still used today in horn-speakers for home use and for public-address systems; their output, watt for watt, is far greater than that of the conventional cone type.

The modern electronic amplifier and speaker are far smaller, and can produce louder sounds which are variable both in quality and volume. This chapter describes the required properties of the electronic amplifier so that the amplifier designs of later chapters can reproduce a sound that, to the human ear, is 'hi-fi'.

1.1 The Human Ear

An analysis of the relevant parts of the human ear follows, including frequency response, construction, loudness sensitivity and other aspects which make certain demands on a hi-fi audio system.

1.1.1 Frequency Characteristics

The **audio frequency range** extends from 30 Hz to 20 KHz. The lower limit is difficult to define because, at low frequencies, sound and feeling become difficult to separate. The upper limit is also difficult to define – as we get older, higher frequencies become inaudible. Very loud sounds will damage the ear temporarily or permanently – a three-hour disco session may result in several hours' deafness afterwards.

The central **reference** frequency against which most measurements are made is called the **mid-band frequency** of 1 kHz. The frequency response of figure 1.1 shows the normal plot of amplitude (vertically, measured in decibels) against frequency (horizontally, measured in hertz). The decibel (dB) scale is used because logarithmic scales are shorter and the values can be added and subtracted for a complex system, rather than multiplied and divided on a linear scale. Appendix I gives further details of the mathematics involved in decibel usage.

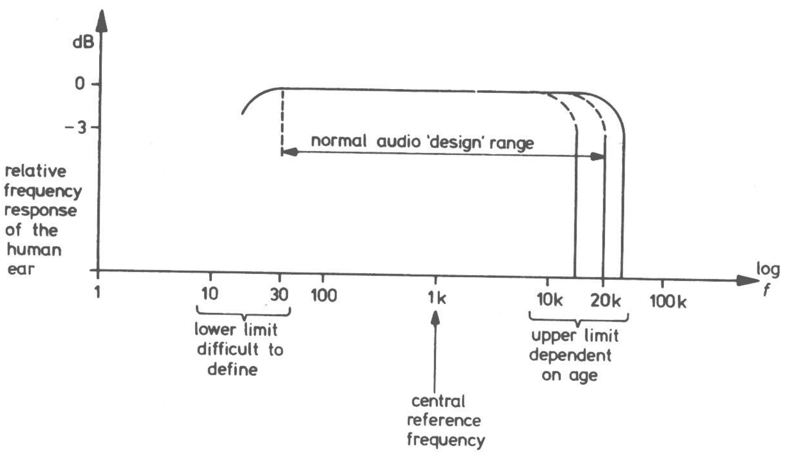

Figure 1.1 The ideal relative frequency response of the human ear

Table 1.1 gives decibel values for various voltage and current ratios, using the mathematics of appendix I.

Some hi-fi amplifiers quote noise and hum figures of 60 dB relative to the signal – known as the **signal-to-noise** dB ratio. This indicates that the signal is one million times stronger than the noise

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Table 1.1

dB Value	*Voltage Gain* $R_{in} = R_{out}$	*Current Gain* $R_{in} = R_{out}$	*Power Gain*
100	100 000	100 000	10^{10}
80	10 000	10 000	10^8
60	1000	1000	10^6
40	100	100	10^4
20	10	10	10^2
10	3.2	3.2	10
6	2	2	4.1
3	1.4	1.4	2
0	1	1	1, reference point
−3	1/1.4 = 0.707	1/1.4 = 0.707	1/2, half-power point
−6	1/2	1/2	1/4.1
−10	1/3.2	1/3.2	1/10

or hum at 1 kHz; thus a signal of 1 V has 1 μV of noise superimposed on top – a very small amount, inaudible to the listener (unless he is one of those perfectionists who converts his living room into a multi-million pound recording studio – of which more later). These hi-fi figures also show that, at frequencies lower or higher than 1 kHz, the noise level could be far worse, particularly at 10 kHz and above where it can be more disturbing – statistics can be misleading

The frequency is also plotted on a logarithmic scale, with equal divisions between 1, 10, 1000 and all powers of 10 Hz. Thus a very wide range of frequencies can be shown. Frequency response is often plotted on special **log–log paper**. Hi-fi systems normally concentrate on the **design range**, as shown in figure 1.1, with extension to higher frequencies if necessary (see p. 5).

1.1.2 The Human Ear

The basic components of the ear are shown in figure 1.2: the **pinna**, directed towards the front of the head, the **eardrum** and its three bones which conduct the sound to the sensitive **inner ear**, where the frequency and intensity of the sound are converted into nerve pulses which travel to the brain for processing. The way in which very low and very high frequencies are heard is very complex, the skull playing an active part, along with the eustachian tube that connects the ear to the back of the throat – suffice it to say that the ear works!

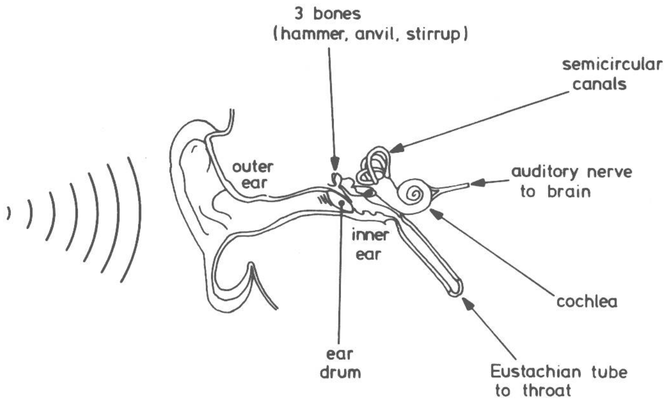

Figure 1.2 Main components of the human ear

Two important points to be made are firstly, that humans have **two** ears, one on the left and one on the right; secondly, the ears are positioned at the main *centre* of communications in the head. These may seem obvious, but the brain can tell the direction of sounds very accurately by decoding the phases and amplitudes of the signals from both ears and distinguishing between different frequencies and sound intensities to either ear. Musicians have a very keen sense of hearing, sometimes as the result of training, and can identify the individual frequencies of a complex mixture of sounds. Try playing a chord on a piano and then whistle or sing the individual notes. Also try closing your eyes and then identifying where a faint sound is coming from in a room – this is relatively simple to do provided neither ear is damaged.

Stereo reproduction uses the ears to give the impression of a 'sound stage' of instruments across about 90°; careful phasing of the left and right signals deceives the ear into this illusion. Mono records are sometimes processed electronically using phasing circuits (see chapter 5) which mislead the ear into thinking that the performers surround the listener. Quadraphony, or surround-sound, represents true 360° using three or four sound sources.

1.1.3 Sound Intensities

Table 1.2 shows the power intensities of sounds (in dB) relative to the **threshold of hearing**. This threshold can only be determined in absolute silence in, say, an anechoic chamber – a room lined with material that will absorb all sounds. Also listed are the sound inten-

Table 1.2 Sound Levels of the Ear

Sound Level (dB)	*Intensity* ($\mu W/m^2$)	*Level*	*Type of Sound*
600	10^{14}	Dangerous	Saturn rocket take-off
120	10^6	Very loud	Threshold of pain –
110	10^5		Concorde take-off;
100	10^4		thunder; jet take-off;
90	10^3		Underground train
80	10^2	Loud	Loud motor horn;
70	10		pneumatic drill; heavy
60	1		traffic
50	10^{-1}	Moderate	Ordinary conversation;
40	10^{-2}		inside train or car
30	10^{-3}		
20	10^{-4}	Quiet	Turning page of newspaper;
10	10^{-5}		faint whisper; breathing
0	10^{-6}	Silence	Threshold of hearing

sities (in $\mu W/m^2$) of these same sounds, which are related to the dB figures by the mathematical formulae given in appendix I.

Sound levels up to 120 dB give no discomfort, but beyond this level pain and damage to the ear will result. Some amplifiers described later in the book can produce sound levels in excess of 120 dB for amplification in large areas.

1.1.4 High Fidelity

It is most people's ambition to have a true **high fidelity** (hi-fi) system in their home, but what is true hi-fi?

The aim of a hi-fi audio reproduction system is to reproduce perfectly in the home, without any distortion at all, the sounds created in the concert hall or recording studio. However, this aim is unattainable. Figure 1.3a shows the frequency response of the ear, with a defined **bandwidth** between the 3 dB points (also called the **half-power points** – see table 1.1) of 20 kHz. The ear is also able to detect far above 20 kHz, not as direct audio frequencies but as *indirect* audio sounds. Listen to a live concert performance and then to a recording of it – you will notice a distinct loss of quality in the recording. It is difficult to identify what is missing in the recording until you hear a professional recording played back on a system that is sensitive up to 100 kHz – you will immediately sense additional sounds that were missing from the recording up to 15 or 20 kHz.

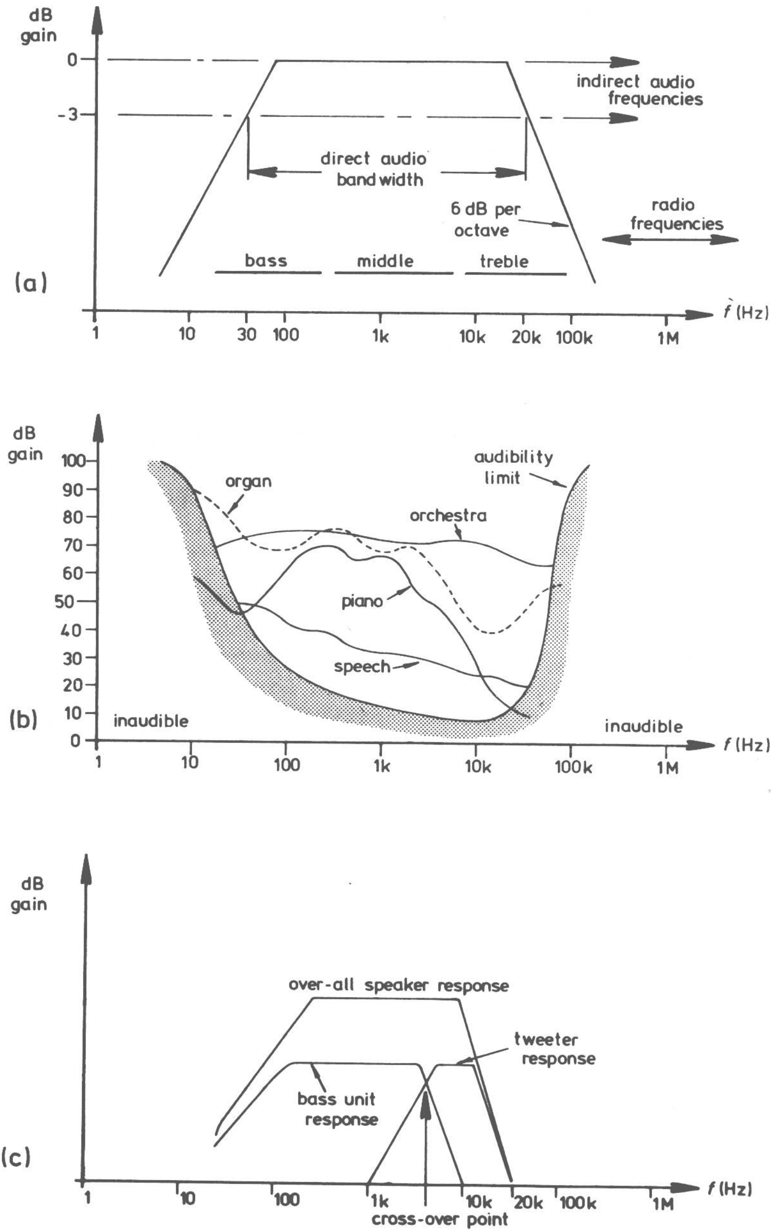

Figure 1.3 Amplifier frequency responses

Unfortunately, an amplifier that is sensitive up to 100 kHz is capable of picking up radio frequencies, which start to appear at 100 kHz (longwave lower limit), so then radio interference becomes a problem. Public-address systems and good hi-fi systems often pick up radio signals, either because they are amplifying 100 kHz signals picked up directly at the amplifier input, or because higher radio frequencies are picked up on the loudspeaker leads – these radio signals travel to the amplifier which detects them and then amplifies them along with the audio signal. Strong taxi transmitters and 'radio ham' transmitters cause much of this interference, and special input filters are needed to filter it out.

The frequency response of the ear is seen to be divided (but not definitively) into **bass, middle** and **treble** ranges. The response also has the normal rise and fall of 6 dB per octave at the lower and upper ends – an **octave** is a frequency of 2 times the **fundamental**, thus 1 kHz has octaves at 2 kHz (the 'second harmonic'), 4 kHz (the 'fourth harmonic'), 8 kHz (the 'eighth harmonic'), and so on. A fall of 6 dB per octave means an attenuation of 6 dB (half gain) from frequency f to frequency $2f$.

Figure 1.3b shows the audible range of the ear relative to the frequency ranges and volumes of speech, a piano, an organ and an orchestra. A hi-fi system should be able to produce all these frequencies and volumes perfectly, otherwise distortion will be heard. This distortion is not necessarily the characteristic distortion heard when an amplifier is overloaded – any deficiency of a waveform, such as added or subtracted frequencies, is regarded as distortion.

The design of electronic circuits which amplify frequencies from 10 Hz to 20 kHz is relatively simple, most modern transistors can amplify frequencies above the audio frequency range. The output is limited mainly by the **loudspeaker,** which converts the high quality electrical audio signal into a mechanical output, or by the input pick-up, which converts the mechanical vibrations of the stylus or diaphragm into electrical signals. Section 1.2 shows why these limitations occur.

One final factor is the *room*. Most living rooms are furnished with carpets, curtains, furniture and people – a variety of sound-absorbent surfaces. Each surface absorbs different frequencies so that the resultant sound heard by the listener bears no relationship to the sound leaving the speaker. The sound also changes as the listener moves about the room, unless a properly designed studio lounge is created with calculated absorbence figures for each surface. True hi-fi cannot therefore be heard in most homes, and for this reason tone controls are incorporated to adjust the amplifier response to suit the surroundings.

1.2 Microphones, Pick-ups and Speakers

The loudspeaker has the worst frequency response of any component in the system. The low frequencies are limited by the construction of the speaker **cone** and the high frequencies by the over-all **damping** of the whole speaker; damping in this case is the inability of the heavy cone to move quickly enough to reproduce the high frequencies. Figure 1.3c shows the simplified response of a bass unit with low response at 10 kHz. Most speaker units compromise with a single unit, in low-priced record players, a double unit (as shown), in medium-fi audio systems, or a treble unit, in hi-fi audio equipment. The contradictory requirements of the bass and treble frequencies can be satisfied in a multi-speaker system, where each speaker is fed with its appropriate range of frequencies from a **cross-over unit** (see chapter 4). The over-all output is the sum of the individual speaker responses, as shown – the low frequency unit is called the **woofer** and the high frequency unit is called the **tweeter**.

The bass unit requires the speaker cone to move relatively slowly over several millimetres or centimetres, while the treble unit requires the cone to move very fast indeed; medium-range units compromise between the two. The speaker shown in figure 1.4e is an electro-mechanical device with an input current fed to the speech coil which is supported between the poles of a very strong magnet. A combination of current and magnetic field causes the coil to move in and out, the coil being physically connected to the speaker cone so that the sound is transmitted outwards. The cone is made of light-weight paper or plastics and is supported flexibly round its edge.

Speaker enclosures are not 'electronic' so they are not described in great detail here, but it must be mentioned that a power amplifier can deliver several amperes of current to the loudspeaker. The speaker speech coil should be designed to carry this current, otherwise smoke output will be substituted for sound output! Always choose a speaker with a **power rating** to match the amplifier and whose **impedance** also matches the amplifier for **maximum power transfer**. Loudspeakers with a **lower** impedance than the amplifier should never be used; speakers with a *higher* impedance can safely be used but the output power will be lower. Series and parallel combinations of speakers can also be used provided the over-all equivalent resistance matches that of the amplifier. A 100 W 8 Ω amplifier, for instance, can drive *four* 25 W 2 Ω speakers connected in series, the sound power being distributed between the individual units, thereby allowing low power speakers to be used rather than the expensive 100 W types.

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