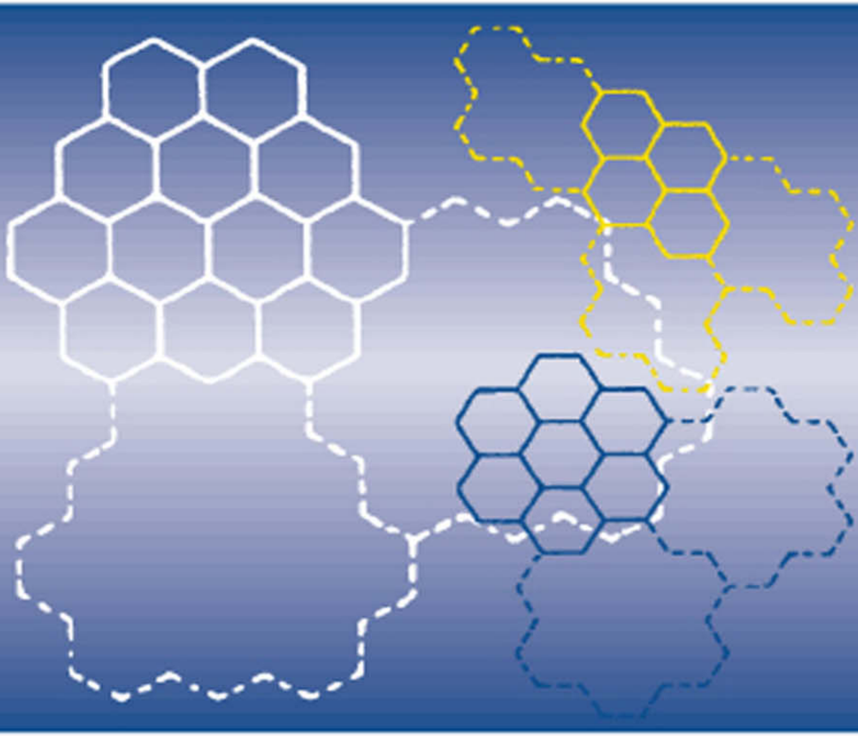
NEWNES

Telecommunications

POCKET BOOK



STEVE WINDER

THIRD EDITION

Newnes Telecommunications Pocket Book

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Newnes Telecommunications Pocket Book

Third edition

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Introduction

The *Telecommunications Pocket Book* provides a summary of current telecommunications systems and working practice. Being a pocket book it cannot give full details of every system, but it will describe them in outline and act as an *aide mémoire*. Since telecommunication techniques are continually changing, the overview given will be useful to practising engineers who wish to update their knowledge.

It also contains data tables that will serve as a useful reference to technicians, engineers and students alike.

This book is structured to describe the signal sources first, followed by the equipment needed to convert these signals into an electrical format suitable for transmission, switching and processing. The transmission medium (radio, copper wire and optical fibre) is then described. Methods of modulating or digitizing the signals for transmission is described. Telephone exchange interfaces and switching is described before signalling and multiplexing are covered. Packet-based networks are described, including the Internet and the numerous protocols such as TCP/IP and HTTP that are associated with it. Standards and organizations are described last.

Telecommunication engineers use abbreviations extensively. This is unfortunate, but necessary to avoid very wordy descriptions of systems and techniques. Since this book will probably be read piecemeal, all abbreviations are described at the end of the book (Chapter 26). Useful formulae and data tables are included in this chapter.

The previous edition of the *Newnes Telecommunications Pocket Book* was published in 1998. Since then there have been many developments, particularly in mobile telecommunications. The growth of Internet traffic has spurned many other developments, such as Wavelength Division Multiplexing (WDM) over optical fibres, which are necessary to carry vast amounts of data. Another significant driver has been the de-regulation of telecommunications in many parts of the world. Regulations in many countries, including the US and most of Europe, have forced incumbent telecommunications companies to allow competition in the local loop (known as 'unbundling'). This, combined with Internet growth, has driven the development of digital line systems (xDSL) that allow high data rates into the home.

I would like to thank my family for their patience while I was editing this book at home during evenings and weekends. I would also like to thank fellow engineers at BT Laboratories for their interest and encouragement.

I have received feedback from readers, including university lecturers and engineers, and this has been welcomed. Please let the publisher know if you feel any topics need to be covered in more detail.

Steve Winder
2001

1 Telecommunications overview

Telecommunication engineering is about understanding the signals and the transmission medium, and then providing a means of communication in the most efficient way. The following sections describe speech, music, video and data signals. These are the signals we want to transmit from one place to another. Later chapters describe the means to transmit the signals (e.g. the telephone), the transmission path (e.g. optical fibre cable), switching and multiplexing. It is impossible to have dedicated circuits between every communication point, and it would be inefficient. Switching is used to connect points only for the duration of the call, which means that only the line between each communication point and the switch is dedicated–the rest of the telephone network is shared. Transmission paths between switch nodes are made to have a high capacity by multiplexing many circuits together. Instead of transmitting one telephone call along an optical fibre, we can transmit hundreds of calls.

Telecommunications encompasses a wide range of signal sources; it is truly multimedia. The telephone network has being integrated with computer systems to provide the Internet. Data is readily available from many sources and knowledge is king. We have become what has been called the 'information society'.

Historically, the telegraph arrived first; it was invented by Charles Wheatstone in 1837. Telegraph was made more efficient by Samuel Morse in 1844 by the development of the Morse code, which remains in use for some radio signalling. A telegraph code (using a 5-bit code to represent the alphabet) was introduced by Baudot in 1874. However, the typewriter was not invented until 1875, so the telegraph system was quite primitive until Western Union started to use the Baudot code in conjunction with a teletype (a sort of Telex machine) in 1901. The telephone was not invented until 1876 by Graham Alexander Bell, and switching was manual until the invention of the automatic telephone exchange by Strowger.

Many inventions have been developed since that time, the most significant was the computer. The computer has its beginnings during the war years 1939 to 1944, out of necessity to break the German navy's cipher codes. Commercial applications were developed and the US Space Program required small and light computers. The microprocessor and then the personal computer (PC) were developed. The computer explosion, where every company and almost every home has a computer, has resulted in the need to share stored data. Computer data and video picture transmission are now common-place.

1.1 Speech and music signals

Speech requires a minimum of 300 Hz to 3.4 kHz bandwidth for good understanding. Speech is produced by a person when air from the lungs passes through the larynx (voice box) in the throat to cause a tone which is changed in amplitude by the mouth and tongue. Although the change in amplitude occurs at a low frequency, the tone is usually within the 300 Hz to 3.4 kHz bandwidth of a telephone system and no intelligibility is lost. Time varying speech signals and their frequency spectrum are illustrated in *Figure 1.1*.

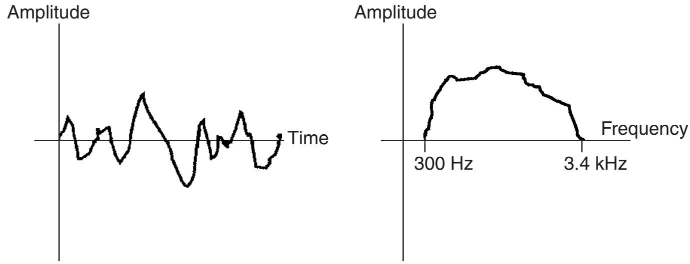

Figure 1.1 Speech in time and frequency domains

Music, on the other hand, requires up to 15 kHz bandwidth. Musical signals are similar to speech, and a passage of music can be identified if the bandwidth is restricted to the speech bandwidth of 300 Hz to 3.4 kHz. However, the quality of music is not appreciated in this limited bandwidth. Drum beats have a significant low frequency content and require a response down to a few hertz. Some string and wind instruments produce significant high frequency content sounds, some beyond the range of human hearing (which is up to 20 kHz in children, but this reduces with age and 15 kHz is a reasonable limit).

1.2 Video signals

Broadcast television signals are complex because they not only contain information about the brightness of the image, they also contain information about the colour and about picture synchronization. Sound signals are often transmitted with the picture too. The system works by the image in a camera being scanned left to right and top to bottom. On the display screen the picture is built up by brightening pixels (picture elements) by scanning them in the same order as the image in the camera. The camera and display scanning must be synchronized and the display refreshed 50–60 times per second. A colour television picture typically requires an analogue bandwidth of d.c. to 6 MHz. If digitized at a 16 MHz sampling rate and using 8-bit encoding, a data rate of 128 Mbit/s would be required.

Time varying video signals and the frequency spectrum for the PAL system is given in *Figure 1.2* . Other video systems such as NTSC or SECAM are similar. Each complete scan of an image generates a frame synchronizing pulse. This is followed by a burst of carrier at about 4.433 MHz to synchronize colour decoders in each receiver. The brightness of the image as it is scanned then determines the amplitude of the signal. Being an amplitude modulated signal, you might expect the spectrum to be double sideband, but it is not. The spectrum is vestigial sideband, which means that the lower sideband is filtered out beyond 1.75 MHz. This reduces the total bandwidth required, but still allows simple detection methods to be employed.

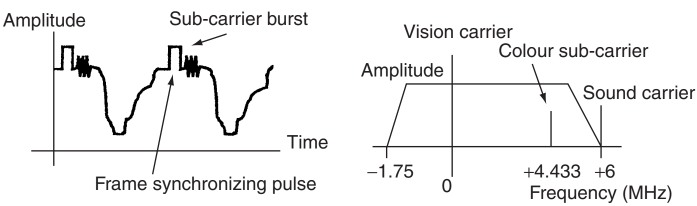

Figure 1.2 Video signals in time and frequency domains

Picture digitizing systems can reduce the bandwidth required. JPEG and MPEG encoding can reduce the transmission bandwidth of a television picture to 2 Mbit/s. Pre-recorded programmes can be digitized and stored. Decoding these signals takes place at the television in a set-top box. The encoding of ‘live’ television pictures is now commercially available.

Slow scan television is used for security surveillance systems. Generally these are monochrome pictures with no sound and are used to monitor slowly changing events, door entry systems, etc. By transmitting each picture slowly the bandwidth is much reduced. This system requires a picture store system at both the camera and the display terminal. A snap-shot is taken by the camera and the image is digitized and stored. Data from the store is transmitted slowly over the transmission system and is stored at the display terminal.

Video-conferencing systems only transmit the difference between one image and the next. In the usual situation, where there is little movement, the amount of data needed to be sent is perhaps just 1% compared to the full picture (i.e. 1.28 Mbit/s). If the display is not refreshed often and video compression techniques are used, a bandwidth of only 64 kbit/s can be used.

1.3 Data signals

Computer data is a binary (two level) representation of numbers, text, images, etc. The binary number is either 1 or 0, often referred to as logic 1 and logic 0. A logic 1 is at or near the positive supply rail of the digital circuits. A logic 0 is at or near the 0V rail. The data word, or ‘byte’, is normally 8 bits long, but is broken down into two ‘nibbles’ so that the word can be expressed in hexadecimal form (0000 = 0, through to 1111 = F). Thus 7E = 01111110 in binary.

Data has a d.c. content. A string of logic 1s or logic 0s is clearly d.c. The maximum fundamental frequency occurs when the data string is 10101010, and the frequency is half the data rate. However, data is pulses which have significant harmonic content and a bandwidth of at least five times the data rate is required. Although data cannot be directly transmitted in a speech bandwidth of 300 Hz to 3.4 kHz, low speed data can be used to modulate a signal within that bandwidth. Techniques to remove the d.c. content of data do allow long range transmission. In one method, the data is converted into alternating positive and negative pulses to remove the d.c. content.

Figure 1.3 shows a data stream in both time and frequency domains. There is no spectral content in the signal at a frequency equal to the data rate, or at an integer multiplier of the data rate. If the data rate is 1 Mbit/s, so that each pulse is 1 μs long, the maximum fundamental frequency is 500 kHz. Since a square wave has no even

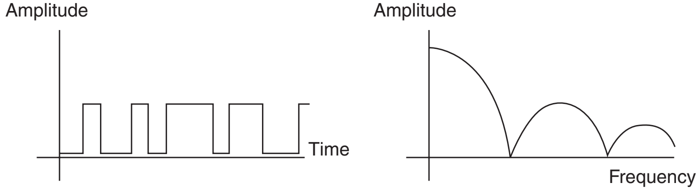

Figure 1.3 Data signal in time and frequency domains

harmonic content, there is no spectral power at 1 or 2 MHz. There are, however, peaks in the spectral content at 1.5 and 2.5 MHz.

Data rates for transmission are expressed as either bits per second or bytes per second. The higher the data rate, the lower the time required for transmission and the processor can begin another task. If a transmission link is charged on a per second basis, the higher data rate also means less cost. For both of these reasons, modem speeds have increased as the technology has permitted and are close to their theoretical limit.

Another reason for increased data rates is the use of interactive systems. Automatic teller machines which dispense cash through a ‘hole in the wall’ are an example of interactive systems, however the data rate can be quite low without seriously affecting performance. Where data rate is very important is in Internet applications. The Internet response time is notoriously slow, with some pages taking a few minutes to download.

2 Telephone equipment

2.1 Constraints

This chapter deals with the terminal equipment that can be connected to the local distribution cables of a public switched telephone network (PSTN), and therefore covers equipment that can be installed at domestic and small business premises, or extensions on a PBX using local lines.

Public Telephone Operator (PTO) monopolies are being broken worldwide to enable competition from privately run companies. This, and the rapid advance of integrated circuits, has resulted in a wide choice of both facilities and suppliers. This is the lowest level at which connections can be made to the PSTN, and imposes a number of constraints on the equipment:

(a) The available frequency band is 300 to 3400 Hz. This constraint is imposed by the 8 kHz sampling rate of analogue to digital converters and the limitations of the transmission bridges used in the local exchanges. A DC path exists for signalling purposes but not transmissions.

(b) The −50 volt exchange battery supply is fed out to the local lines. This is provided primarily to give a DC loop calling facility, and to energize a carbon microphone. It does, however, provide a small amount of available power for the operation of terminal equipment, and with low-power ICs enables a considerable amount of processing to take place in the terminal equipment without the provision of local power. The available power is limited by the line loop resistance of up to 1250 ohms. Constant current feeds are supplied from some digital exchanges.

(c) It is an offence to connect unapproved equipment to the public network. The requirements of telephone apparatus to meet certain standards depends on the local telecommunications regulator. Generally equipment has to meet the EMC and safety standards of the host country. The equipment must meet specified requirements for that class of equipment, and be tested to the appropriate specifications by an authorized independent body before approval is granted. If the equipment derives power from the mains supply, stringent safety conditions apply.

2.2 Telephones

In its simplest form a telephone consists of:

(a) a microphone and earpiece, normally combined into a single hand-held unit;
(b) means of signalling the exchange when a call is to be made;
(c) means of sending the 'address' information of the called subscriber to the exchange;
(d) an incoming call alarm;
(e) means of signalling the exchange that the call has been answered.

The requirements of (b) and (e) are met by completing a DC loop when the handset is removed from its rest. The DC path is then via a carbon transmitter, providing the necessary polarizing current for the transmitter. Carbon microphones developed for the telephone provide an adequately good frequency response for speech at high sensitivity, avoiding the need for amplification. Less sensitive microphones with amplifiers are now more common having longer life and avoiding the problem of granules breaking down and causing noise.

The incoming call alarm (d) has in the past been provided by a bell operating from a 17 Hz supply from the local exchange. This is being largely replaced by an audible tone from a small sounder or loudspeaker, a choice of tone being available in some equipment to allow a number of phones in a restricted area to be identified. This is referred to as a 'tone caller' and on some equipment the earpiece is used for this purpose. In such cases the calling signal must start at a low level and progressively increase as a safety precaution.

The 'address' of any subscriber is represented by a series of digits, the first group in the series providing the routing information to the local exchange, and the second group, the subscriber's individual number in the exchange. Since the advent of automatic exchanges the series of digits has been provided by a dial which produces an impulse sequence, breaking the DC path. The impulses in the train are 33 $\left(\frac{1}{3}\right)$ ms on, 66 $\left(\frac{2}{3}\right)$ ms off, providing an impulsing rate of 10 per second. The speed restriction was imposed by the switching rate of the mechanical selectors and not by the transmission path; hence the introduction of electronic switching has made possible a much faster switching rate, and reduced the time required to set up a call.

Press-button switches have therefore replaced dials on modern telephones. When used in areas still served by mechanically switched exchanges, the keypad is quicker and more convenient to use, but has to store the number and send it to line as a 10 ips signal; hence the actual time required to set up the call is not substantially reduced. When operated into a digital exchange, however, multifrequency coding is employed, each digit being represented by two audio tones. DTMF signalling allows 10 digits, plus * and # symbols to be coded as shown in *Table 2.1*.

Table 2.1 DTMF code

Digit	*Frequencies (Hz)*	
1	697	1209
2	697	1336
3	697	1477
4	770	1209
5	770	1336
6	770	1477
7	852	1209
8	852	1336
9	852	1477
0	941	1336
*	941	1209
#	941	1477

Many of the telephones now in production will produce either type of signal with a simple changeover to enable them to operate with either type of exchange.

The logical extension to the keypad is number storage, and a wide range of telephones incorporate both last number redial and a store enabling the most used numbers to be called by depressing a single press button. Again the maximum advantage is obtained from digital exchanges, since exchanges requiring impulse dialling cannot take advantage of the quick-call set-up obtained with MF coding. Further facilities now available include an LCD display, which shows the number being called, and can also provide a clock and call timer, and a loudspeaker to allow handsfree operation.

The digital exchange is also increasing the services available to the customer. These include:

Charge advice: enabling the cost of a call to be advised at the end of the call.
Reminder call: enabling an alarm call to be made at any required time.
Call diversion: enabling all incoming calls to be diverted to another telephone.
Call barring: enabling certain types of call (e.g. overseas or trunk) to be barred.