DATA COMMUNICATIONS for DEVELOPING COUNTRIES

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Japan International Cooperation Agency

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CONTENTS

Preface

For the first decade of its development, Computer technology was concerned with single, isolated computers. People brought their problems to the computer and carried away the results. Then experiments were tried in which one computer and its remote terminals or two computers interacted; thus making computers accessible from a distance. These were the beginning of DATA COMMUNICATIONS.

Data communications, through connecting of computers, can transmit huge amounts of information, quickly and more accurately than conventional Telegraph and Telephone communications. Moreover, Data communications have the excellent functions of Information straging, processing, calculating which conventional communications do not have.

Since Data communications have been introduced in the world, rapid developments have been carried out, in both hardware and software. Examples, like Deposit Accounts, Money Exchanges, credit cards, home banking, Production controls, Automated Manufacturing (or Robots), Sales and Inventories control of manufacturer, whole sales and Departmental stores, Controlling various kinds of pollutions, traffic controls, personnel registrations, car registrations, urgent medical cares used in social and economic fields of a country.

This book consists mainly of two parts. One for Analog Data transmission technology which is utilizing existing Analog telecommunication plants in the country. The other is for Data communications technology, which adds the value of the data by processing the necessary information to meet the customers demand.

In the developing countries, the top of each government or private organization is seeking the milestone which can guide the introduction of Data communications, utilizing the existing telecommunications facilities efficiently, and combining newly developed technology involving Digital transmission, Packet switching and Optical fibers.

I hope the book will serve as a mile stone and useful reference to those interested in the data communication technology, for the developing countries.

I wish to express my sincere gratitude to Mr. Robert Kittel, Mr. Anwar Altaf, Miss Michel and C.T.R.L. Data comm Lab., staffs, for their valuable assistance.

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Chapter 1 INTRODUCTION

Telecommunication links will bring the capabilities of the computers and the information in data banks to the millions of locations where they can be used. Computers in return will control the immense switching centers and help divide the enormous capacity of the new linkages into usable channels.

Small computers are now being mass-produced in microscopic circuitry. A microprocessor, is small enough to carry in your pocket, but can have as much power as a typical "second generation" processor like the IBM 1401, and thousand times as much memory. Unlike a second generation processor it needs no air-conditioning, no large power supply, and is unlikely to fail. Like printing news-papers, if a large enough quantity is made, the cost of each becomes very low. Now the semi-conductor industry is mass-producing such microprocessor at less than $20 each.

An essential ingredient of economic growth has always been the means of communication or transportation. Those nations which have had the best facilities for distribution have tended to be those which rose fastest to economic power. The first industrial revolution was able to sustain its mushroom-like growth because transportation facilities had been built. Canals were dug, and weary horses spent their lives lugging barges of coal from the new mines to the factories. Many factories clustered round the coal fields and whole segments of the population moved there into gloomy, jerry-built rows of houses. It took many years for the means of transportation to become adequate for the needs of the new industry.

Communications links are a vital part of society's infrastructure. In a computerized society, we look towards electronic communications. Manufacturing will be increasingly run by process control computers and production robots. The paperwork will be handled increasingly by data processing. Management will have an insatiable desire for information and those with the best information will succeed over their competition. Vast data bases will be built up and will serve many locations. Computer network will lace together the corporate and government facilities.

In many ways telecommunications are acting as a substitute for the increasingly expensive physical transportation. There will be no need to send corporate or government mail physically; it will be cheaper to send it electronically.

TELEPHONE CONNECTION

Figure 1.1 illustrates an intercity telephone connection. When the subscriber at the left of the diagram picks up his telephone, hears a dial tone, and is in contact with his local central office, he will be connected directly without any trunk being involved. If he dials a subscriber in a different city, the central office will switch the subscriber loop to a toll-connecting trunk, thus connecting the call to a toll office. The toll offices are interconnected with intertoll trunks, and this toll network establishes the connections between towns. There may be several intermediate switching offices in the path which is set up between the two toll offices.

The long-distance trunks are all four-wire. The term fourwire implies that there are two wires carrying the signals in one direction and two wires carrying them in the other direction.

There are many repeaters (amplifiers) on these circuits, amplifying the entire group of calls in both directions. In reality the circuit may not consist of wire pairs but of higher capacity transmission media, however, the historical term four-wire is still used and implies separate circuits in each direction.

In normal telephone service the local loops are two-wire circuits, on which a single telephone call can be transmitted in both directions over the same pair of wires. The toll-connecting trunk can be either a two-wire or a four-wire line a special circuit is required to join them. For special purposes a four-wire local loop can be used. Some data transmission machines require a four-wire rather than a two-wire connection.

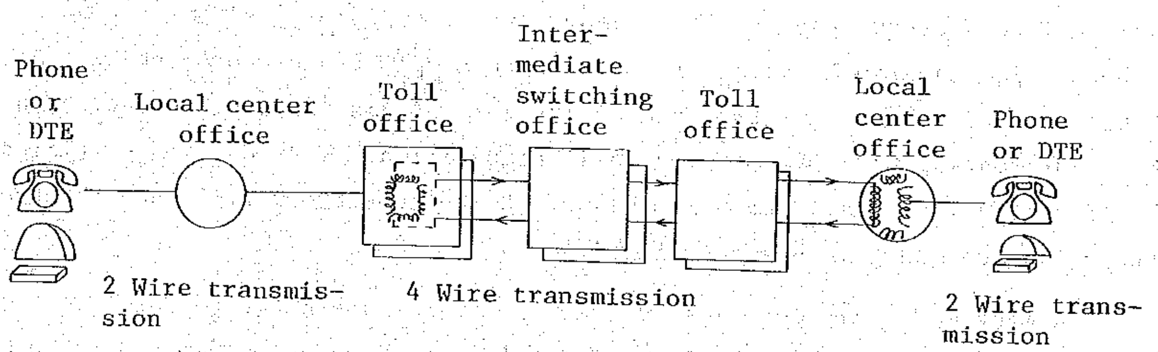


Fig. 1.1 An intercity telephone connection

TELEGRAPH EQUIPMENT

From its earlier days the telecommunications industry has produced a range of machines for telegraphy. These are called teletype machines in North America. The word teleprinter is common elsewhere.

Telegraph machines trasmit at a speed much lower than the capacity of a voice line. Common teletype speeds in North America are 75 and 150 bits per second. Commond speeds elsewhere in the world are 100 and 200 bits per second. The international telex network and many European lines operate at 50 bits per second.

Telegraph signals are formed simply by switching an electrical current on and off or by reversing its direction of flow. The upper part of Fig. 1.2 shows a single-current telegraph signal in which the information is coded by switching the current on and off at particular times. The lower part shows the same signal on a double-current telegraph system in which positive and negative potentials are applied at one end of the line, thus reversing the direction of current flow. The means for producing these current changes are some form of make-and-break contact. The instruments for sending signals, such as teleprinters or paper-tape readers, make and break the circuit at appropriate times. The on and off pulses form a code which is appropriately interpreted by the receiving device. Single-current telegraph signaling is also known as "neutral" or "unipolar" signaling. Double current signaling is called "polar" or "bipolar".

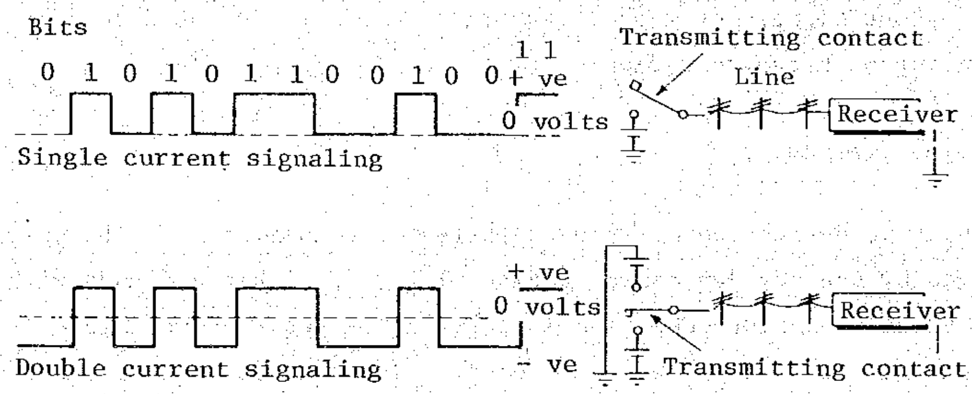


Fig. 1.2

TELEGRAPH CIRCUITS

At one time most telegraph signals were carried on overhead wires like those seen in old Western movies. In Wall Street the sky used to be filled with overhead wires. When telephone developed it soon outgrew telegraphy, and when there were many more telephone circuits than telegraph it became economical to send the telegraph signals over telephone links. Today this trend has gone much farther, and may telegraph signals can travel simultaneously over one path, being manipulated by the complex electronics that are part of today's telephone plant. The simple electromagnetic repeaters that were once used for telegraphy have been largely replaced by the amplifiers and modulators and multiplexing equipment.

A circuit designed for telegraph (not telephone) often had only a single wire spanning the distance. This is called an earth-return circuit. As in the diagrams in Fig. 1.2 each end of the line was connected to earth, and this completed the circuit. One pole of the power supply at each station was connected to earth. This was done by making a connection to the protective lead sheaths of the underground signal cables. The resistance of such an earth-return path is very small.

Other telegraph circuits were obtained as a by-product of telephone circuits. Telegraph signals may be sent down any two telephone channels by a process called "superposing". Each telephone channel ends on a transformer winding. The midpoints of these windings were taken and used as an extra pair of wires down which telegraph signals could be sent without interfering with the Telephone Signals or without the telephone signals could be sent without interfering with the telegraph.

START-STOP INSTRUMENTS

Teletype devices normally use start-stop transmission. The character generated by one key depression begins with a START bit and ends with a STOP bit or bits. The STOP condition is a "mark" or positive voltage on the line. The START bit is a "zero" bit or "space". i.e., no voltage on the line or negative voltage with double current signaling.

The positive voltage of the STOP condition will remain on the line until the next character starts, as indicated in Fig. 1.3 which shows 5-bit and 7-bit telegraphy characters. In other words, the line's idle condition is a "mark" rather than a "space". As soon as a space is detected the receiving device starts and will then be in synchronization with the transmitting device. The STOP condition thus has a minimum duration, which is (1.5 or 2) times the length of the other bits but which could go on indefinately. If the machine is printing or transmitting continuously, there will be no gap between characters, and the STOP condition will last only long enough to clearly separate the characters.

Letter F in 5-bit telegraphy (CCITT Alphabet N°2 or Baudot code)

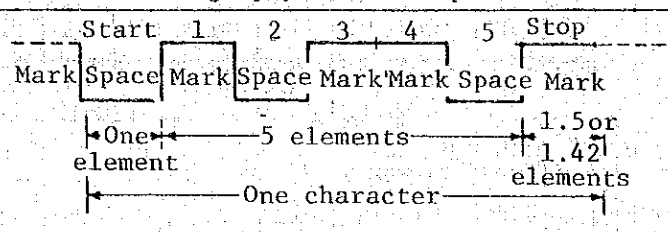


Figure for 8 bit telegraph machines (CCITT Alphabet N°5 or ASC II code)

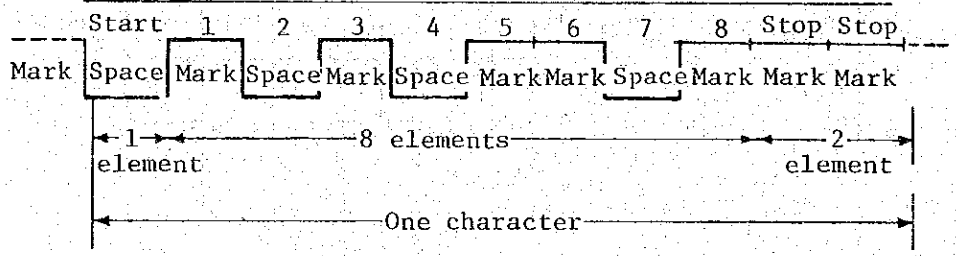


Fig. 1.3 Typical character structures for start-stop (asynchronous) transmission.

Author describes basic knowledge about data transmission. Data transmission is very similar with Telex or telegraph, for dealing with digital ON/OFF signals, but data code in data transmission has a difference as follows.

DATA CODES

It is desirable to have internationally agreed-upon code alphabets for transmission of data so that different computers and terminals can intercommunicate.

During the 1960s an alarming proliferation of data transmission codes developed. Some of these codes have now dropped out of use, and two codes predominate, each with minor deviations for local needs. The first is standardized by the CCITT as their International Alphabet No. 2: This is a 5-bit code based on the Murray code used decades ago for telegraph and now used for telex transmission around the world. It is commonly (and incorrectly) known as the Baudot code in the United States. The second is standardized by the CCITT and also by the ISO (International Organization for Standardization) known as the International Alphabet No. 5. This is a 7-bit code of the U.S. ASCII code (American Standard Code for Information Interchange).

Bit positions 5, 6, 7:

Bit positions 1, 2, 3, 4:		000	100	010	110	001	101	011	111
		0	1	2	3	4	5	6	7
0000	0	NUL	DLE	SP	0	@	P	`	p
1000	1	SOH	DC1	!	1	A	Q	a	q
0100	2	STX	DC2	"	2	B	R	b	r
1100	3	ETX	DC3	#	3	C	S	c	s
0010	4	EOT	DC4	$	4	D	T	d	t
1010	5	ENQ	NAK	%	5	E	U	e	u
0110	6	ACK	SYN	&	6	F	V	f	v
1110	7	BEL	ETB	'	7	G	W	g	w
0001	8	BS	CAN	(	8	H	X	h	x
1001	9	HT	EM	)	9	I	Y	i	y
0101	10	LF	SUB	*	:	J	Z	i	z
1101	11	VT	ESC	+	;	K	[	k	{
0011	12	FF	FS	,	<	L	\	l	|
1011	13	CR	GS	-	=	M	]	m	}
0111	14	SO	RS	.	>	N	^	n	~
1111	15	SI	US	/	?	O	_	o	DEL

Fig. 1.4 The U.S. ASCII code. This is the U.S. national version of CCITT alphabet number 5. The control characters are explained next.

NUL (Null): No character. Used for filling in time or filling space on tape when there is no data.

SOH (Start of Heading): Used to indicate the start of a heading which may contain address or routing information.

STX (Start of Text): Used to indicate the start of the text and so also indicates the end of the heading.

ETX (End of Text): Used to terminate the text which was started with STX.

EOT (End of Transmission): Indicates the end of a transmission, which may have included one or more "texts" with their headings.

ENQ (Enquiry): A request for a response from a remote station. It may be used as a "WHO ARE YOU?" request for a station to identify itself.

ACK (Acknowledge): A character transmitted by a receiving device as an affirmation response to a sender. It is used as a positive response to polling messages.

BEL (Bell): Used when there is need to call human attention. It may control alarm or attention devices.

BS (Backspace): Indicates movement of the printing mechanism or display cursor backwards in one position.

HT (Horizontal Tab): Indicates movement of the printing mechanism or display cursor forward to the next preassigned "tab" or stopping position.

LF (Line Feed): Indicates movement of the printing mechanism or display cursor to the start of the next line.

VT (Vertical Tab): Indicates movement of the printing mechanism or display cursor to the next of a series of preassigned printing lines.

FF (Form Feed): Indicates movement of the printing mechanism or display cursor to the starting position of the next page, form, or screen.

CR (Carriage Return): Indicates movement of the printing mechanism or display cursor to the starting position of the same line.

SO (Shift Out): Indicates that the code combinations which follow shall be interpreted as outside of the standard character set until a SHIFT IN character is reached.

SI (Shift In): Indicates that the code combinations which follow shall be interpreted according to the standard character set.

DLE (Data Link Escape): A character which shall change the meaning of one or more contiguously following characters. It can provide supplementary controls, or permits the sending of data characters having any bit combination.

DC1, DC2, DC3 and DC4 (Device Controls): Characters for the control of ancillary devices or special terminal features.

NAK (Negative Acknowledgment): A character transmitted by a receiving device as a negative response to a sender. It is used as a negative response to polling measages.

SYN (Synchronous/Idle): Used as a synchronous transmission system to achieve synchronization. When no data is being sent a synchronous transmission system may send SYN characters continuously.

ETB (End of Transmission Block): Indicates the end of a block of data for communication purposes. It is used for blocking data where the block structure is not necessarily related to the processing format.

CAN (Cancel): Indicates that the data which precedes it in a message or block should be disregarded (usually because an error has been detected).

EM (End of Medium): Indicates the physical end of a card, tape or other medium, or the end of the required or used portion of the medium.

SUB (Substitute): Substituted for a character that is found to be erroneous or invalid.

ESC (Escape): A character intended to provide code extension in that it gives a specified number of contiguously following characters an alternate meaning.

FS (File Separator):	Information separators to be used in an optional
GS (Group Separator):	manner except that their hierarchy shall be FS (the
RS (Record Separator):	most inclusive) to US (the least inclusive).
US (United Separator):	

SP (Space); A nonprinting character used to separate words, or to move the printing mechanism or display cursor forward by one position.

DEL (Delete): Used to obliterate unwanted characters (for example, on paper tape by punching a hole in every bit position).

The U.S. ASCII CODE; The Control Characters

Chapter 2 SYSTEM WHICH USE DATA TRANSMISSION

Data transmission systems are built for a wide variety of purposes and differ accordingly in the way they function.

Probably the most common from of system in the future will be one in which people at various kinds of terminals communicate with a distant huge computer. The most of the terminal computer will usually respond to them quickly. Often a dialogue takes place between the terminal user and the remote computer, when the job is complicated and required various data which the terminal computer can't deal with.

ON-LINE AND OFF-LINE SYSTEM

Many systems are not interactive in nature but are required merely to move a quantity of data from one point to another. The communication links in this case may be on-line to a computer or off-line. On-line means that they go directly into the computer, with the computer controlling the transmission. Off-line means that telecommunication data do not go directly into the computer but are written onto magnetic tape or disk or are punched into paper tape or cards for later processing.

Most transmission from human operators at terminals are interactive: in fact, it is a bad design not to give a response to an operator and to leave him wondering whether his input has reached the computer or not.

For noninteractive systems or systems that give a very rudimentary response, the data will flow in one direction only. The transmission system is not normally designed to be entirely one-way because a small trickle of control signals going in the other direction is needed. Occasionally, in telemetry, the use of radio makes two-way transmission difficult, and a purely one-way link is used. An interactive system, on the other hand, can have a high flow of data in both directions.

TIME FOR TRANSMISSION

Sometimes it is necessary to transmit the data quickly. The speed required depends on the system. A system for relaying one-way messages such as telegrams may be required to deliver the message in an hour or so. It would be convenient to have it done faster, but no major economic need to do so exists. When batches of data are sent for batch processing on a distant computer, a delivery time longer than 1 hour is sometimes acceptable. However, where a man-computer dialogue is taking place, the responses must be returned to the man sufficiently quickly so as not to impede his train of thought. Response times between 1 and 5 seconds are typical. In real-time systems in which a machine or process is being controlled, response times can vary from a few milliseconds to many minutes.

AIRLINE RESERVATIONS

There are many systems using computers and telecommunications for special functions such as banking, transferring funds electronically, and booking hotels, and spectacular use of telecommunications is found in worldwide airline reservation systems. Some of these have a thousand or more terminals, all obtaining very quick responses on communication links that circle much of the world. They employ most of the different communication facilities.

The real-time terminals of Japan Airline are on leased voice-grade lines with many terminals connected via concentrators to one line. The voice-grade lines cover America and Europe, Asia from Tokyo. To take them further than this would have been uneconomical. The rest of the world is reached, therefore, by leased telegraph lines and telex. These lower speed circuits often travel their long distances by radio and have a much higher proportion of data errors than the voice-grade lines or land telegraph lines. The telegraph circuits are used for administrative messages of all types, as well as for reservations. They go through message-swtiching centers some of which are manually operated or semiautomatic, and this means that there is a delay in some reservation signals reaching the computer. Special procedures therefore have to be programmed for the offices that do not have conversational terminals.

There are three main objectives in designing a reservation system:

1. To improve the service given to passengers and potential passengers.
2. To minimize staff in sales offices and in control offices where reservations are processed and space on aircraft controlled.
3. To improve the load factor on flights.

When a system handles cargo reservations, the objectives here are to minimize staff and to optimize the loading of cargo. Many airline systems carry out functions other than reservations, including crew scheduling, passenger check-in at airports, load and trip calculations prior to a plane's takeoff, and scheduling of maintenance services.

INTERLINKED SYSTEMS

When many on-line systems exist for functions such as banking and ticket reservations, the next logical step is for separate systems to be interlinked. An airline agent should be able to interrogate the reservation systems of other airlines so that multi-airline journeys can be planned and booked. A travel agent should have access to systems which book hotels, cars, theater tickets, boats, trains, and air-line seats. An electronic fund transfer system should interlink the computers of many banks. Computers doing corporate payrolls should be able to send the money electronically to the computers of employees' banks, via a clearing system. Computerized cash registers in supermarkets and stores have been connected by telecommunications to bank computers. A society is evolving which will employ vast networks of machines connected by telecommunications.

TIME-SHARING SYSTEMS

Most systems with manually operated terminals are time-shared, meaning that more than one user is using them at the same time. When the machine pauses in the processing of one user's item, it switches its attention to another user.

The term "time sharing," however, is commonly used to refer to a system in which the users are independent, each using the terminal as though it were the console of a computer and composing, testing and executing programs of his own at the terminal. Each user feels as though he were the only person using the system. The programs of one terminal user are quite unrelated to those of other users.

In many systems the users do not program the system at the terminal, and neither are the users independent. They are each using the programs in the computer in a related manner. They may, for example, be insurance or railroad clerks possibly using the same programs and possibly the same file areas. In many time-shared systems, however, the computing facility is being divided between separate users who can program whatever they wish on it, independently of one another. Three categories of on-line systems, differing by the degree of independence of the users, are in common use.

1. Systems that carry out a carefully specified and limited function for example, banking systems or airline reservation systems in which all terminal users can update the same files.

2. Systems for a specific limited function in which the user has personal independent files or in which shared files can be read but not updated by general terminal users.

3. Systems in which programmers can program anything they wish at the terminals, providing they all use the same language, and interpreter or compiler for this being in the machine. Normally each user has the same type of terminal.

Chapter 3 THE STRUCTURE OF NETWORKS

Basically, there are two ways in which information of any type can be transmitted over telecommunication media: analog or digital.

Analog means that the amplitude of the transmitted signal varies over a continuous range. Oscillating signals are normally transmitted and the frequency of the oscillation can also vary over a continuous range. Both the sound you hear and the light you see are analog signals spread over a range of frequencies.

The telephone company, in the interest of economy in telephone line, transmits a range of frequencies that may vary from about 300 to 3,400 Hz only. This is enough to make a person's voice recognizable and intelligible. When telephone signals travel over lengthy channels, they are packed together, or multiplexed, so that one channel can carry as many such signals as possible. To do this your voice might have been raised in frequency from 300 - 3,100 to 60, 300 - 63. 100 Hz. Your neighbour's voice might have been raised from 64, 300 to 67, 100. In this way, they can travel together without interfering with one another, but both are still transmitted in an analog form - that is, as a continuous signal in a continuous range of frequencies.

Digital transmission means that a stream of on/off pulses are sent like data travelling in computer circuits. The pulses are referred to as bits. It is possible today to transmit an extremely hight bit rate.

Most telephone channels today are analog channels, capable of transmitting a certain range of frequencies. If we send computer data over them, we have to convert that digital bit stream into an analog signal using a special device known as a DATA-modem. This converts the data into a continuous range of frequencies - the same range as the telephone voice. In this way, we can use any of the world's analog channels for sending digital data.

On the other hand, where digital channels have been constructed, it is possible to transmit the human voice over them by coverting it into a digital form. Similarly, any analog signal can be digitized for transmission in this manner. We can convert hi-fi music, television pictures, temperature readings the output of a copying machine , or any other analog signal into a bit stream.

Almost all the world's telephone plant grew up using analog transmission. Much of the transmission will remain so for years one because of the multi-billions of dollars tied up in such equipment. However, many millions of channel miles of digital channels are now operating, mostly designed to carry digitized speech. Digital technology is rapidly evolving, and major advantages in digital transmission are beginning to emerge.

Chapter 4 AC SIGNALLING AND BANDWIDTH

Light, sound, radio waves, and ac signals passing along telephone wires are all described in terms of frequencies. In all these means of transmission of the instantaneous amplitude of the signal at a given point oscillates rapidly, Rate of oscillation is referred to as the frequency and described in terms of Hertz.

With light we see different frequencies as different colors. Violet light has a higher frequency than green, green has a higher frequency than red. With sound the higher frequencies are heard as higher pitch.

The human voice consists of a jumble of different frequencies, when we see a red light it is not one frequency but a collection of frequencies which combine to give this particular shade of red. The same is true with the electrical and radio signals of telecommunications. We will not usually be discussing one single frequency but a collection, or a band of frequencies occupying a given range.

THE SPEECH SPECTRUM

The human ear can detect sounds over a range of frequencies; in other words, it can hear sounds of different pitch. A sensitive ear can hear sounds of frequencies ranging from about 30 Hz upto 20,000 Hz, though most people have a range somewhat less than this.

When we refer to a sound of a given frequency, we mean that the air is vibrating with that number of oscillations per second. To transmit this sound the microphone of a telephone converts the sound into an equivalent number of electrical oscillations per second. The telephone channels over which we wish to send data are, then, designed to transmit electrical oscillations of a range equivalent to the frequencies of the human voice, although these frequencies are often changed for transmission purposes.

In fact, the telephone circuits do not transmit the whole range of the human voice. It was found that this was unnecessary for the understanding of the speech and the recognition of the speaker. Figure 4.1 illustrates the characteristics of human speech and shows that its strength is different at different frequencies. Most of the energy is concentrated between the frequencies 300 and 3,100 Hz, and each telephone channel is designed to transmit only this range. This is a decision based upon economics. It permits the maximum number of telephone conversations to be sent at one time over the various physical media while still making the human voice intelligible and the recognizable.

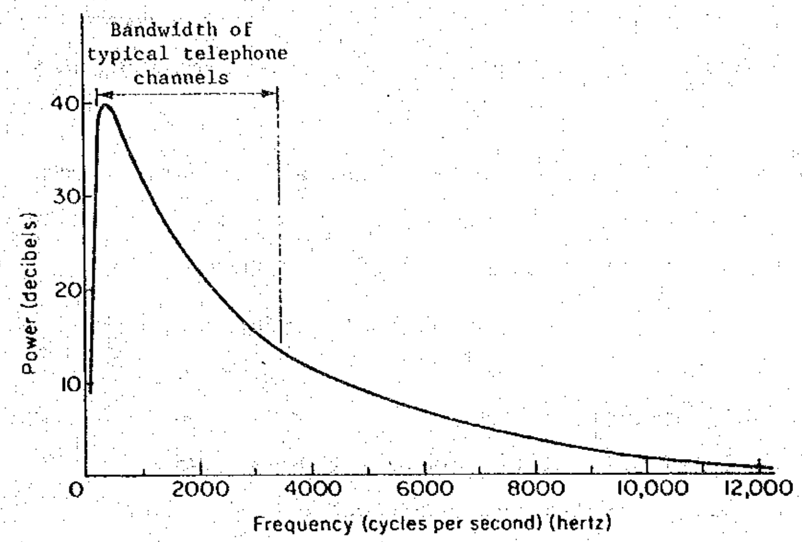


Fig. 4.1 Spectrum of human speech. In order to transmit speech so that the speaker is recognizable and understandable only the range indicated need be sent.

Figure 4.1 may be described as a spectrum diagram. The reader would become familiar with such a form of diagram. Sound spectra or electrical signal spectra are broadly equivalent to light spectra.

A spectral line of electrical waves relates to transmission at one frequency only. For a single frequency of the transmission may be represented by the equation

$$a = A \sin 2\pi ft$$

a: instantaneous amplitude

t: function of time t

A: the maximum amplitude

We shall refer many times to this sine wave. It is illustrated Fig. 4.2.

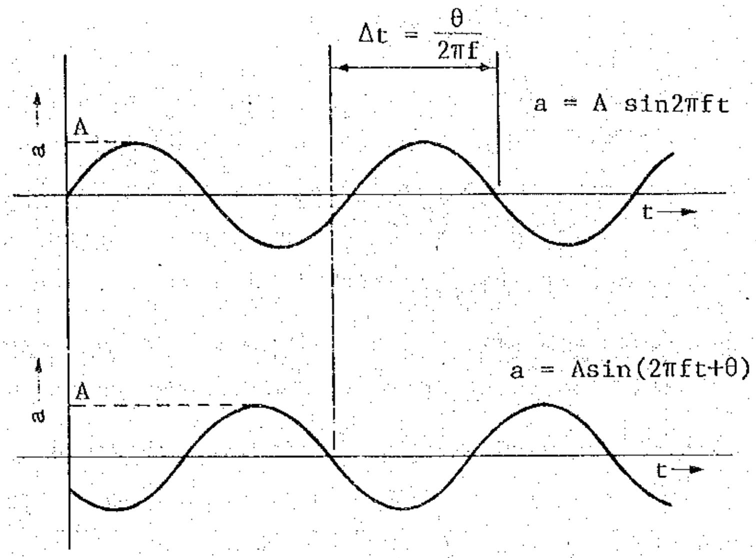


Fig. 4.2 Two sine waves of frequency f with phase difference θ

There are f peaks of the type in this diagram occurring every second - that is what we mean when we say the "frequency" is f Hz. There are f complete cycles per second, two of which are illustrated in each diagram in Fig. 4.2.

PHASE

Figure 4.2 shows two sine waves both frequency f, which are displaced from each other in time so that their peaks do not occur simultaneously. They are therefore said to be different in phase. If one wave is described by $a = A \sin 2\pi ft$, then the other is described by $a = A \sin (2\pi ft + \theta)$, where θ is said to be the phase difference.

The time for one cycle of a sine wave, in other words, time $1/f$, is equivalent to a phase difference or angle of $\theta = 360°$ (or 2π radians). Two waves of the same frequency but differing in phase by 360° are identical. The maximum that the phase can be varied for this purpose is less than 360° because a 360° change would be indistinguishable from the original.

Two sine waves differing in time by Δt have a phase difference of $\theta = 2\pi f \Delta t$ radians. The time difference Δt, therefore, as indicated in Fig. 4.2 is $\theta/2\pi f$.

DECIBLES

The unit normally used for expressing signal strength in telecommunications is the decible. It is also used to quote gains and losses in signal strength. Spectrum diagrams like that in Fig. 4.1 have decibels on the vertical axis. The unit measures differences in signal strengths, not the absolute strength of a signal, and it is a logarithmic unit, not a linear one.

Both of these facts sometime cause confusion. Box 4.1 gives the information the reader should know about decibels and similar units.

The decibel was first used as a unit referring to sound. It made sense to refer to sound levels by a logarithmic unit because the response of the human ear is proportional to the logarithm of the sound energy, not to the energy itself. If one noise sounds twice as great as another, it is not in fact twice the power but it is approximately 2 decibels greater. The sound energy reaching your ears in the subway may be 10,000 times greater than in the room where you are reading this book, but it does not sound 10,000 times greater. It sounds about 40 times greater-you have to shout 40 times harder to make yourself heard to a person the same distance away. Ten thousand times the sound energy is called "40 Decibels greater".

Box 4.1 Decibels, dBv, dBm, and nepers

DECIBELS

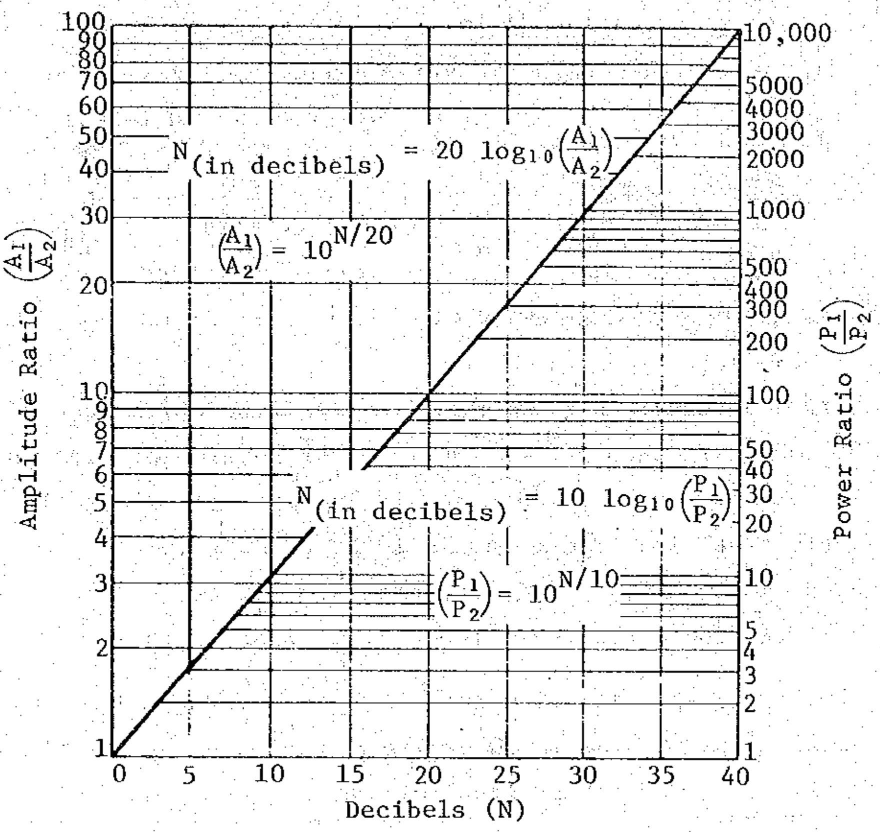
The unit which is normally used for expressing differences in signal strengths in telecommunications is the *decibel*. The decibel is a unit of power *ratio*. It is not an absolute unit but a unit which is employed to compare the power of two signals. Signal-to-noise ratio is normally quoted, for example, in decibels.

A decibel is equal to 10 times the logarithm (to base 10) of the power ratio:

$$\text{number of decibels} = 10 \log_{10} \frac{P_1}{P_2}$$

where P_1 is the larger power (normally) and P_2 is the smaller.

The decibel also used to be defined as the unit of attenuation caused by 1 mile of standard No. 19 gage cable at a frequency of 866 Hz, though this definition is now regarded as obsolete. 1 decibel attenuation means that a signal has dropped to 0.794 of its original power. 1 decibel gain means that



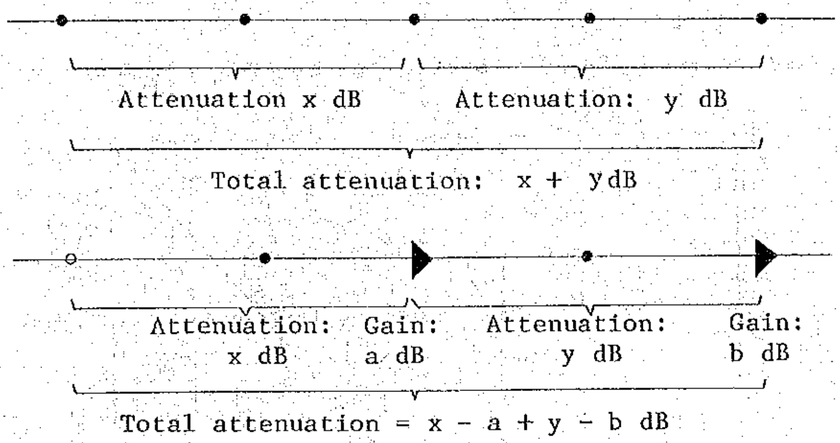
a signal has increased to 1.259 of its original power.

Voltage and current ratios are also quoted in decibels. Power is proportional to the square of the amplitude of signal. A power ratio of 100, say, is equivalent to an amplitude ratio of 10. Therefore, where the two curent levels are a_1 and a_2, or the two voltage levels are V_1 and V_2, we have

$$\text{number of decibels} = 20 \log_{10} \frac{a_1}{a_2} \text{ or } 20 \log_{10} \frac{V_1}{V_2}$$

Decibels are used to express such quantities as gain in amplifiers, noise levels, losses in transmission lines, and also differences in sound intensity. The decibel is a valuable unit for telecommunications because losses or gains in signal strength may be added or subtracted if they are referred to in decibels as shown in the figure below.

Box 4.1 Continued



Suppose that a signal is transmitted over a line which reduces it in power in a ratio 20 to 1. It then passes over another section of line which reduces it in a ratio 7 to 1. The net reduction is in the ratio 140 to 1. Expressing this in decibels, the first reduction is $10 \log_{10} 20=13.01$ decibels, and the second reduction is $10 \log_{10} 7 = 8.45$ decibels. The net reduction is the sum of these: 21.46 decibels ($10 \log_{10} 140 = 21.46$ decibels).

Similarly, if we say that line loss is 2 decibels per mile, then the loss at the end of 25 miles of line is 50 decibels. We therefore need an amplifier of gain 50 decibels to produce a signal of the original power.

BANDWIDTH

While the frequency range 300 to 4,000 or even 300 to 3,000 is satisfactory for voice transmission, music would sound poor because it would be clipped of the higher and lower frequencies which give it its quality. To faithfully reproduce the deep notes of percussion or double bass we need to go down to 30 Hz, and to reproduce the high harmonies which make instruments sound realistic, a frequency up to 15,000 or better, 20,000, is desirable. It is toward these extremities that the high-fidelity enthusiasts trive.

AM radio transmits sound frequencies up to 5 kHz and thus it is capable of reproducing music that does not sound too distorted but is not high fidelity. FM radio can produce the whole range needed for high-fidelity reproduction.

BANDWIDTH OF A TELEPHONE CHANNEL

A telephone call, as we shall see, may pass through many multiplexing processes. The multiplexing is engineered to precise standards however, and the properties of the channels which the users perceive are remarkably similar.

Figure 4.3 shows the properties of a typical channel which has passed through multiplexing equipment. It shows how the signal strength varies with frequency. There is little attenuation of the signal between 300 and 3,100 Hz, but outside those limits its strength falls off rapidly. The main energy of human speech lies within this frequency range.

The electronics of the telephone plant have deliberately chopped the signal up in the shape of Fig. 4.3 so that it fits completely into 4 kHz slices. There is some wastage between the slices, leaving a comfortable gap between channels. The gap is needed to minimize interferences or "crosstalk" between channels.

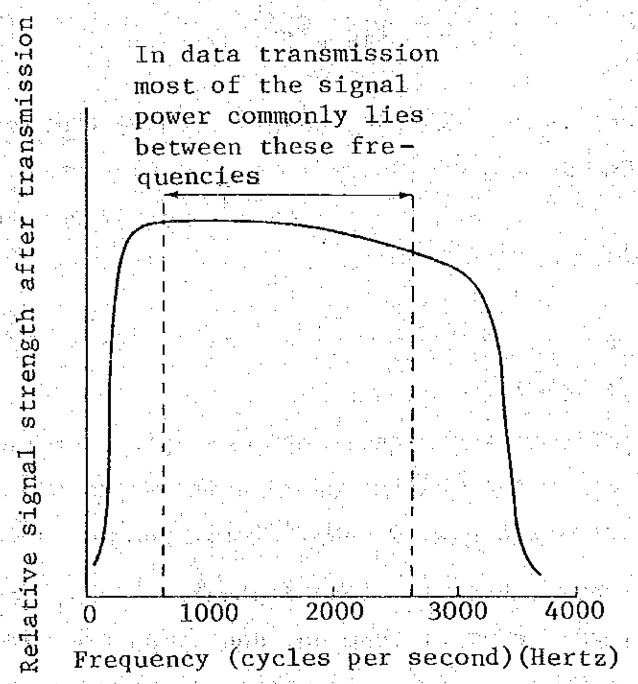


Fig. 4.3

LOADING

Loading is a means of descreasing the attenuation of a wire-pair line and holding it more, nearly constant over a given frequency range.

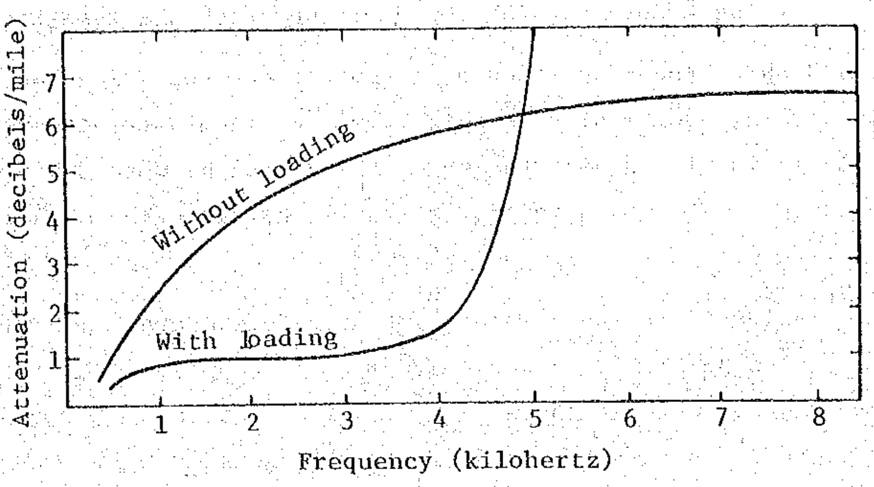


Fig. 4.4

If high frequencies or high data rate are to be transmitted over a subscriber loop, the telephony loading coil must be removed.

TWO-WIRE AND FOUR-WIRE CIRCUITS

A four-wire circuit has two amplifiers in its repeaters, one for each direction of transmission.

A two-wire circuit may have one amplifier which serves both directions. This is made possible by suitably arranged coils which act like transformers. But most two-wire repeaters use two amplifiers, one for each signal direction.

A four-wire circuit, used for long-distance trunks, may not necessarily have four actual wires. Many of them do, and these are referred to as physical four-wire circuits. However, two conductros can form an equivalent four-wire circuit. (The computer profession would probably call it a "virtual" four-wire circuit). The two directions of transmission use different frequency bands. The east-west and west-east signals are changed in frequency by different amounts and can then travel on the same two physical wires without interfering with each other. Similarly, in data transmission signals it can be made to travel over two wires if their frequencies are changed so that they do not interfere with each other. The two directions are separated in frequency rather than space.

Whether they are physical four-wire or virtual four-wire circuits depends on how important it is to minimize the number of physical paths. With open-wire pairs, for example, a physical four-wire circuit would mean doubling the number of wires stretched between the telephone poles. This is undesirable, and usually virtual four-wire transmission is used, needing one pair of wires only. On the other hand, high-capacity intercity trunks carrying large numbers of conversations at once usually separate the directions of transmission physically. The electronic equipment is designed to pack many channels all going in the same direction into one coaxial cable or microwave facility.

HYBRID COILS

Where the two-wire line from your telephone joins the four-wire trunk, a connecting circuit is needed. An outgoing signal on the two wires is transferred to the appropriate pair of the four-wire line by this, and an incoming signal on the other pair of the four-wire line is transferred to the two-wire line.

The essence of this junction circuit is the hybrid coil, shown in Fig. 4.5. The signal on the two-wire line travelling west to east in the diagram is picked up by the coil entering the upper amplifier, amplified, and transmitted down the west-east half of the four-wire line. A signal traveling east to west is amplified by the lower amplifier and enters the two-wire line. It enters the hybrid coil at its center, and hence the signals induced into the uppermost part of the coil cancel out. If the hybrid coil were perfect and the balancing network precisely duplicated the line section it faces, no signal would enter the upper amplifier. On a two-wire line, two amplifiers are often used rather than the one.

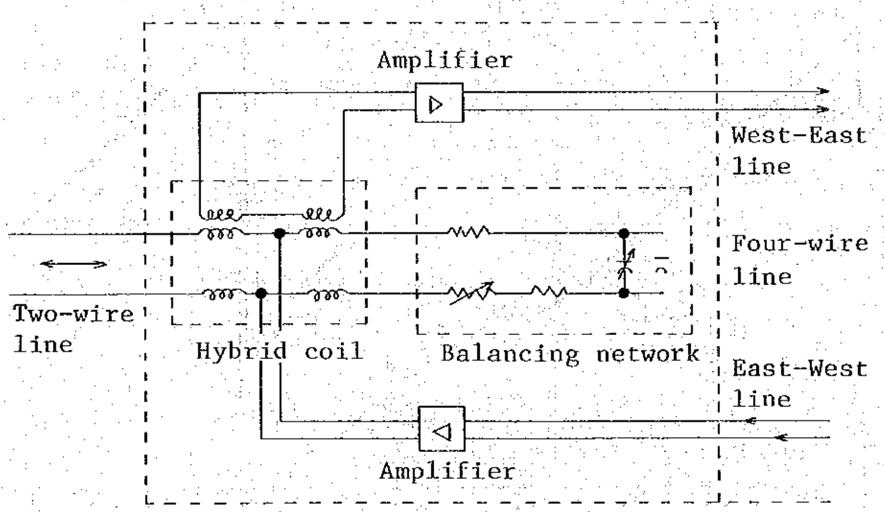


Fig. 4.5 A hybrid coil circuit used to convert two-wire signals into four-wire signals and vice versa

ECHOES

Unfortunately, the hybrid circuit is not perfect. When a signal travels east to west a small portion of it does find its way into the upper amplifier. The signals induced into the two halves of the upper coil do not cancel out exactly. The balancing network is chosen to minimize this unbalancing.

LOCATION OF REPEATERS

The repeaters are spaced at intervals sufficiently close to prevent the signal from being attenuated to a level at which it will be too small relative to the possible sources of noise. For voice transmission it is generally desirable to maintain an overall signal-to-noise ratio of 30 decibels or better.

On many good-quality circuits, when the signal has traveled a distance such that it has fallen in power by a factor of the order of 100, in other words 20 decibels, it is boosted back to its original value. Thus on a coaxial cable system with an average attenuation coefficient of 5 decibels per mile, repeaters may be installed every 4 miles. On the other hand, on open-wire pairs with an average attenuation of say, 0.4 decibel per mile, the repeaters may be placed at distances of about 50 miles.

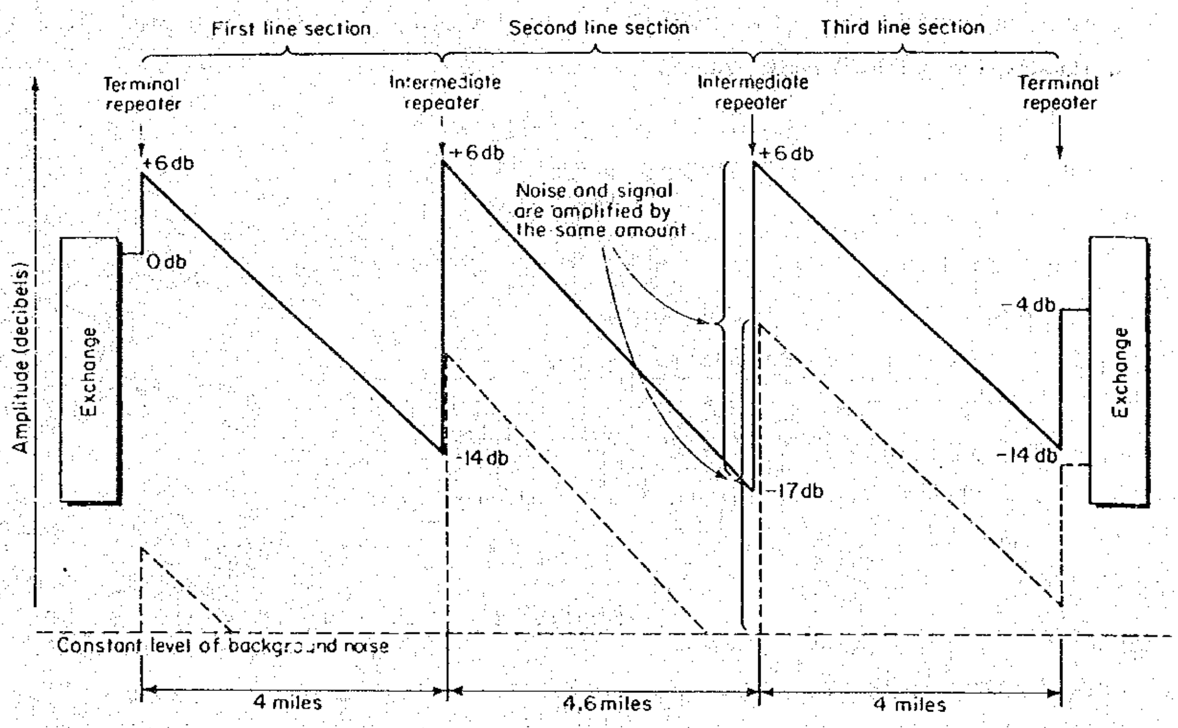


Fig. 4.6 Attenuation and amplification on a line with a coefficient of 5 decibels per mile

Figure 4.6 illustrates the use of repeaters on a 12.6 mile length of cable with an attenuation constant of 5 decibels per mile. If there were no repeaters on this line, the total drop in signal amplitude would be 63 decibels. This is too great a drop as the signal level would fall below the level of the background noise. When the signal is amplified the noise is amplified with it, as shown by the dotted line in Fig. 4.6. If the signal falls too low, then the ratio of signal strength to that of background noise becomes low and never improves because the two can never be separated. On the second line section of Fig. 4.6 the repeater spacing is slightly greater than on the first section. This allows the signal to fall slightly closer to the background noise level. This closer spacing of signal and noise remains until the signal reaches the exchange. It can be seen from Fig. 4.6 that if the repeaters had been closer together, the signal-to-noise ratio would have remained at a better figure. The one repeater interval that is greater than the others causes more than its fair share of degradation of the signal, and so it is advisable to have all the repeaters the same distance apart.

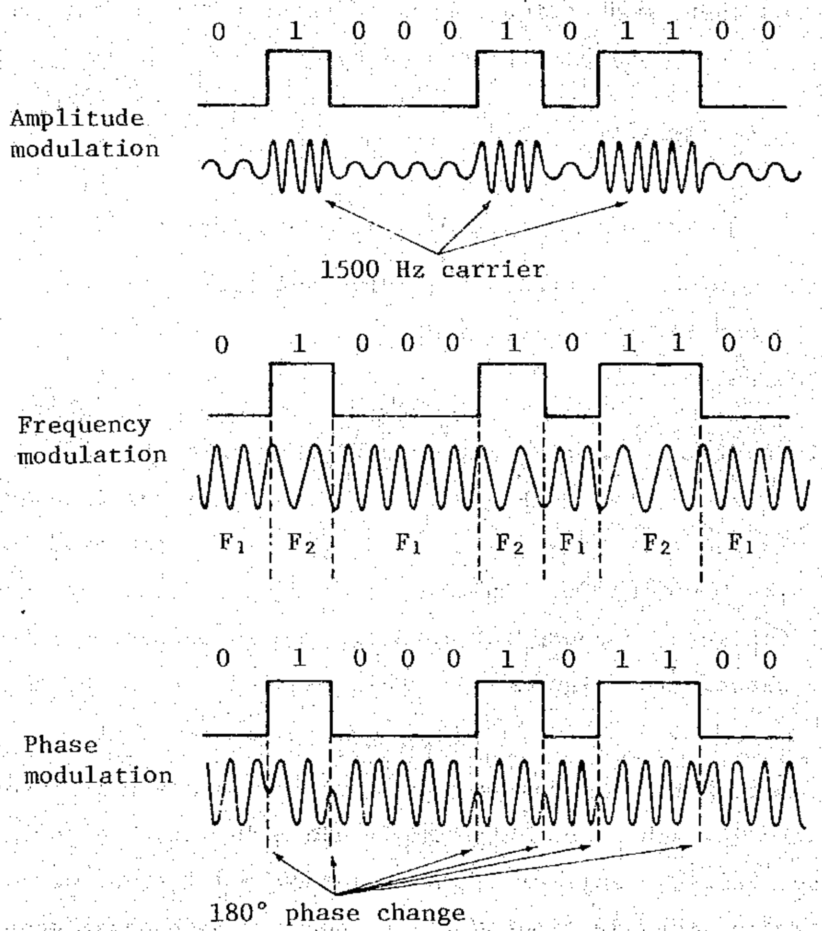


Fig. 4.7 The three basic methods of modulating a sine wave carrier (a simplified diagram showing only binary signals). The amplitude, the frequency, or the phase, can be changed to carry the data

We are therefore seeking a workable compromise between the quantity of data that can be packed into the transmission and the ability of the modem to decode it correctly in the presence of noise and distortion.

For correct decoding an accurate replica of the original carrier must be given to the demodulating circuit. There are a number of ways of obtaining this. In some cases it is sufficient to generate the replica independently in the demodulating equipment. A reference frequency may be generated by a high precision quartz oscillator and used for decoding frequency modulation. It is in phase modulation that it is most difficult to obtain a reference. The demodulator can have no absolute sense of phase.

THREE DIFFERENT TYPES OF MODULATION

When we employ a sine-wave carrier to convey data, it has three parameters which we could modulate; its amplitude, its frequency, and its phase. There are thus three basic types of modulation in use; amplitude modulation, frequency modulation, and phase modulation. Each of these methods is in common use today. The sine-wave carrier may be represented by

$$a_c = A_c \sin (2\pi f_c t + \theta_c)$$

where a_c is the instataneous value of carrier voltage at time t,

A_c is the maximum amplitude of carrier voltage,
f_c is the carrier frequency, and
θ_c is the phase.

The values of A_c, f_c, or θ_c may be varied to make the wave carry information. This is illustrated in Fig. 4.6. A sinusoidal carrier wave of, say, 1,500 Hz. a midfrequency of a voice channel band, is modulated to carry the information bits 01000101100. In this simplified diagram the channel is being operated inefficiently because far more bits could be packed into the carrier oscillations shown. The tightness of this packing determines the speed of operation.

Furthermore, here we modulate the carrier by placing it in one of two possible states in each case. With amplitude modulation we could send several different amplitudes. Similarly with frequency modulation, we could use several frequencies rather than just the F and F_2 shown. With phase modulation Fig. 4.6 illustrates only 180° phase changes, we could use phase changes which are multiples of 90°, giving four possible states, or 45° giving eitht, and so on. Increasing the number of states of the carrier that are used increases the complexity of the decoding or demodulation circuits and considerably increases the susceptibility of the transmission to noise and distortion.

The original carrier may be reconstructed from information in the signal. This may be done by transmitting a separate tone of narrow bandwidth along with the signal, or it may possibly be obtained from the modulated signal itself. Sometimes the signal is briefly interrupted at intervals to give information about the carrier.

Chapter 5 MODULATION

1. AMPLITUDE MODULATION

In amplitude modulation the amplitude of the carrier wave is varied in accordance with the signal to be sent. In its simplest form the carrier is simply switched on and off to send 0 and 1 bits as in Fig. 4.6.

In general, the signal to be sent is multiplied by the carrier wave,

$$ac = Ac \sin (2\pi fct + \theta c)$$

This results in a signal which contains the original carrier plus two side bands, one higher in frequency that the carrier and the other lower. If the signal being transmitted has a frequency fm (modulation frequency), this will give an upper side band with frequency $fc + fm$ and a lower side band with frequency $fc - fm$. It is in these two side bands that the information is carried.

Any signal that is to be sent (voice or data) can be represented by a series of sine waves using Fourier analysis. We are then modulating a sine-wave carrier by another sine waves. This can be represented mathematically as follows;

The carrier $ac = Ac \sin 2\pi fct$ is to be modulated by a wave $am = Am \sin 2\pi fmt$. The resultant wave is

$$\begin{aligned}
amc &= (Ac + am) \sin 2\pi fct \\
&= (Ac + Am \sin 2\pi fmt) \sin 2\pi fct \\
&= Ac \sin 2\pi fct + Am (\sin 2\pi fmt) \sin 2\pi fct \\
&= Ac \sin 2\pi fct + \frac{Am}{2} \cos 2\pi f(fc - fm)t - \frac{Am}{2} \cos 2\pi f(fc + fm)t \\
&= Ac \sin 2\pi fct + \frac{Am}{2} \sin [2\pi f(fc - fm)t + \frac{\pi}{2}] \\
&= + \frac{Am}{2} \sin [2\pi f(fc + fm)t - \frac{\pi}{2}]
\end{aligned}$$

This contains the three components, the carrier at frequency fc, and two side bands at frequencies $fc - fm$ and $fc + fm$ which contain information.

thus carrier : $Ac \sin 2\pi fct$

lower side band: $\frac{Am}{2} \sin [2\pi (fc - fm)t + \frac{\pi}{2}]$

upper side band: $\frac{Am}{2} \sin [2\pi (fc + fm)t + \frac{\pi}{2}]$

Am/Ac is referred to as the modulation factor or modulation index.

The maximum practical value of the modulation index is 1, and often it is less. If the amplitude of the modulating wave became greater than that of the carrier, giving a modulation index greater than 1, the resultant wave would have an envelope with more peaks than the modulating wave, and the original signal would not be recovered. This is shown in Fig. 5.1, and it is referred to as overmodulation.

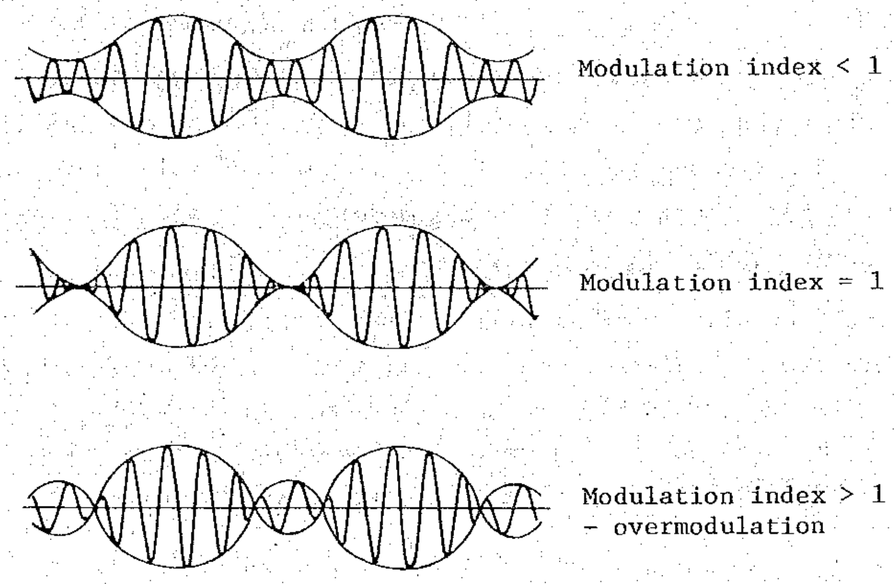


Fig. 5.1 Different values of the modulation index Am/Ac

If the modulating signal consists in effect of several sine waves, then we shall have that number of frequencies both in the upper and lower side bands.

In voice transmission, for example, if a carrier of 60 kHz is modulated by speech filling a band of 300 to 3,000 Hz, the resulting transmission will be of an upper side band of 60,300 to 63,000 Hz, a lower side band of 57,000 to 59,700, and the original carrier of 60,000.

The carrier wave contains no information and may be suppressed. The lower side band duplicates the information that is in the upper side band, and therefore some systems transmit only one sideband. Further, not all the components shown are needed to extract the information. It could be extracted from a relatively narrow band of frequencies where the side-band amplitude is largest.

2. FREQUENCY MODULATION

When frequency modulation was developed it was used to replace amplitude modulation where better performance in the presence of impulse noise and voltage level changes was needed. The signal is transmitted at constant amplitude and so is resistant to changes in amplitude. However, a larger bandwidth is needed. Although amplitude modulation techniques are still used extensively in the common carrier's plant, frequency modulation is now being used more and more for modems.

In frequency modulation, the frequency of the carrier wave varies in accordance with the signal to be sent. This is illustrated in a simplified form in Fig. 4.6. The frequency of the carrier assumes one value for a 1 bit and another for a 0 bit. This type of on/off modulation is sometimes called frequency-shift keying (FSK).

The modulation can also be a continuous analog process, the input signal being any waveform, which again we may regard as a collection of sine waves. The unmodulated carrier, as before, may be represented by

$$ac = Ac \sin 2\pi fct$$

If its frequency fc is modulated by a sine wave of frequency fm, we have

$$acm = Ac \sin 2\pi\ (fc + \Delta fc \sin 2\pi fmt)t$$

where Δfc is the maximum frequency deviation that can occur.

The ratio $\Delta fc/fm$ is referred to as the modulation index.

In general, the spectrum resulting from frequency modulation is much more complex than the equivalent amplitude modulation. If the modulating waveform is a simple sine wave, it can be shown that the resulting wave will contain side bands at frequencies $fc + fm$ and $fc - fm$, and also at $fc + 2fm$, $fc - 2fm$, $fc + 3fm$, $fc - 3fm$, $fc + 4fm$, $fc - 4fm$, and so on. In other words, there are an infinite number of side bands spaced at equal to the modulating frequency.

It will be seen that the spectrum lines carrying information are concentrated into a narrower range of frequencies when fc/fm is small. The transmission fm can therefore occur over narrower bandwidths.

The spectra could be considerably more complicated for actual data transmission because, as we have discussed above, the data consist in effect not of one sine wave but of many. One relatively simple case may be considered, however, and that is where data bits are sent strictly at two frequencies fc and $fc + \Delta fc$.

3. PHASE MODULATION

Just as frequency modulation for data transmission to a large extent replaced amplitude modulation because of its better resilience to noise, so phase modulation is now to some extent replacing the others,especially high speed data transmission method.

In phase modulation the phase of the carrier is varied in accordance with the data to be sent. A sudden phase change of +180° cannot be differentiated from a change of -180°. Therefore, the maximum range over which the phase can be varied is ±180°. As small changes in phase cannot be transmitted and detected with accuracy. Phase modulation is not normally used for the transmission of speech and music, for which frequency modulation and amplitude modulation are commonly used. The small range of variations can be used, however, to code the 2 bits of binary transmission, or 4 bits, 8 bits, or possibly even more when multiple-level codes are used. With four phases in use each interval carries 2 bits of information (a "dibit") and with eight phases, 3 bits.

The unmodulated carrier may, as before, be represented by

$$ac = Ac \sin 2\pi Fct$$

If its phase is modulated by a sine wave of frequency fm, we have

$$acm = Ac \sin (2\pi Fct + \Delta\theta m \sin 2\pi fmt)$$

where $\Delta\theta m$ is the maximum change in phase and is here called the modulation index.

The instantaneous frequency of the wave is $(1/2\pi)$ x (the rate at which its angle is changing at that instant), in this case

$$\frac{1}{2\pi} \times \frac{d}{dt}(2\pi Fct + \Delta\theta m \sin 2\pi fmt) = Fc + fm\Delta\theta m \cos 2\pi fmt$$

Thus the instantaneous frequency is Fc, the carrier frequency + a term $fm\Delta\theta m\cos\ 2\pi fmt$. This is equivalent to frequency modulation of the carrier frequency fm by a wave of frequency fm.

Δf. the maximum frequency deviation is $fm\theta m$.

Thus the resulting wave will contain an infinite number of side bands spaced at intervals equal to the modulating frequency, i.e., side bands at frequencies $Fc \pm fm$, $Fc \pm 2fm$, $Fc \pm 3fm$, and so on.

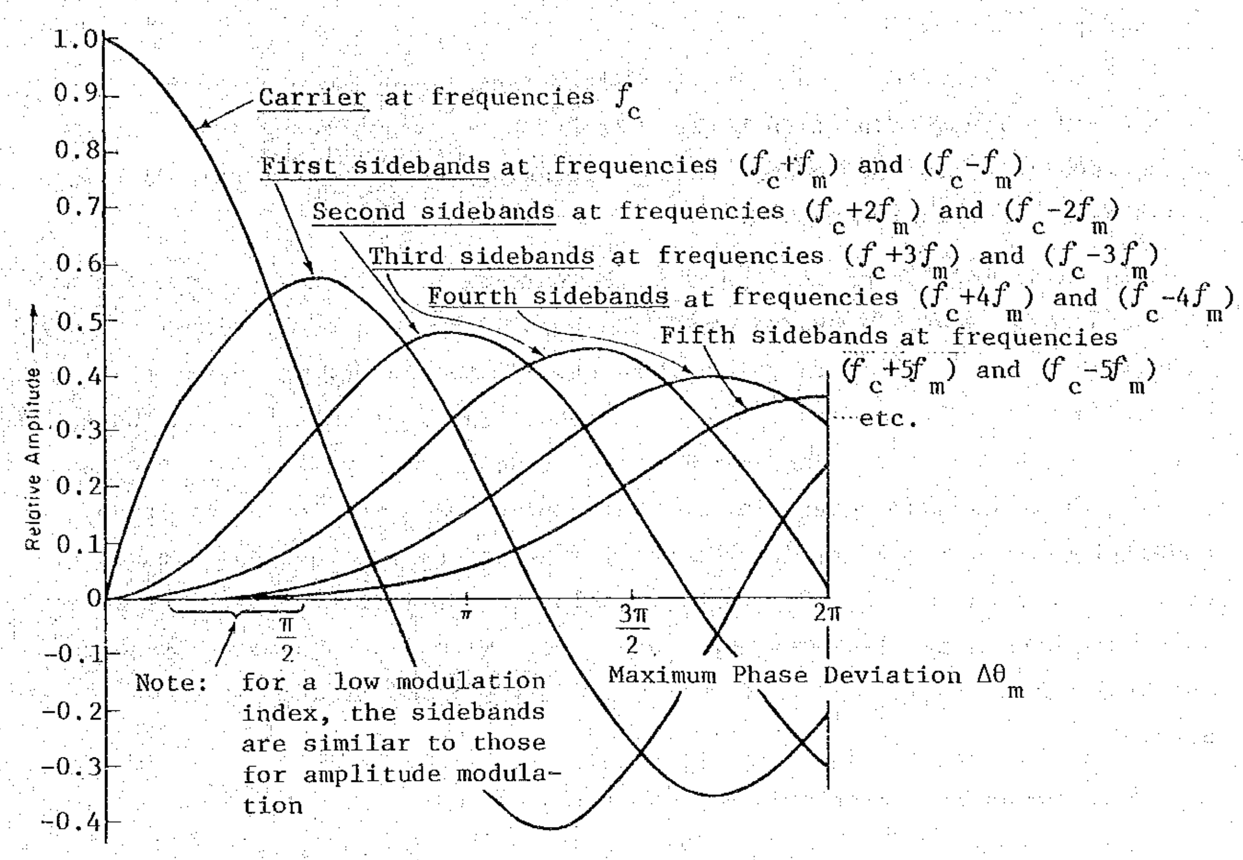


Fig. 5.2 Spectral components of a carrier of frequency fc, phase modulated by a sine wave of frequency fm

The spectra resulting from phase modulating a carrier with a sine wave would be similar to those for frequency modulation. Generally phase modulation uses a smaller bandwidth than frequency modulation, or conversely, more information can be sent in a given bandwidth. The highest transmission speeds on a given bandwidth have thus often been obtained with phase modulation, dispersive noisy medium, but, instead of becoming more and more distroted until eventually signals are unrecognizable, it is repeatedly reconstructed. Of course, an exceptionally large noise impulse may destroy one or more pulses so that they cannot be reconstructed by the repeater stations.

Consider a telephone wire pair under the streets of a city, with analog transmission it can carry a channel group (12 voice channels). Now suppose that we transmit digitally over the same wire pair. The digital signal becomes distorted as it is transmitted as shown in Fig. 6.1. We catch it before it becomes too distorted to recognize whether a bit is 0 or 1. The bit stream is then reconstructed, retimed, and retransmitted. The faster the bit stream is transmitted, the greater will be the distortion and the closer the spacing of repeaters necessary to reconstruct the signal correctly.

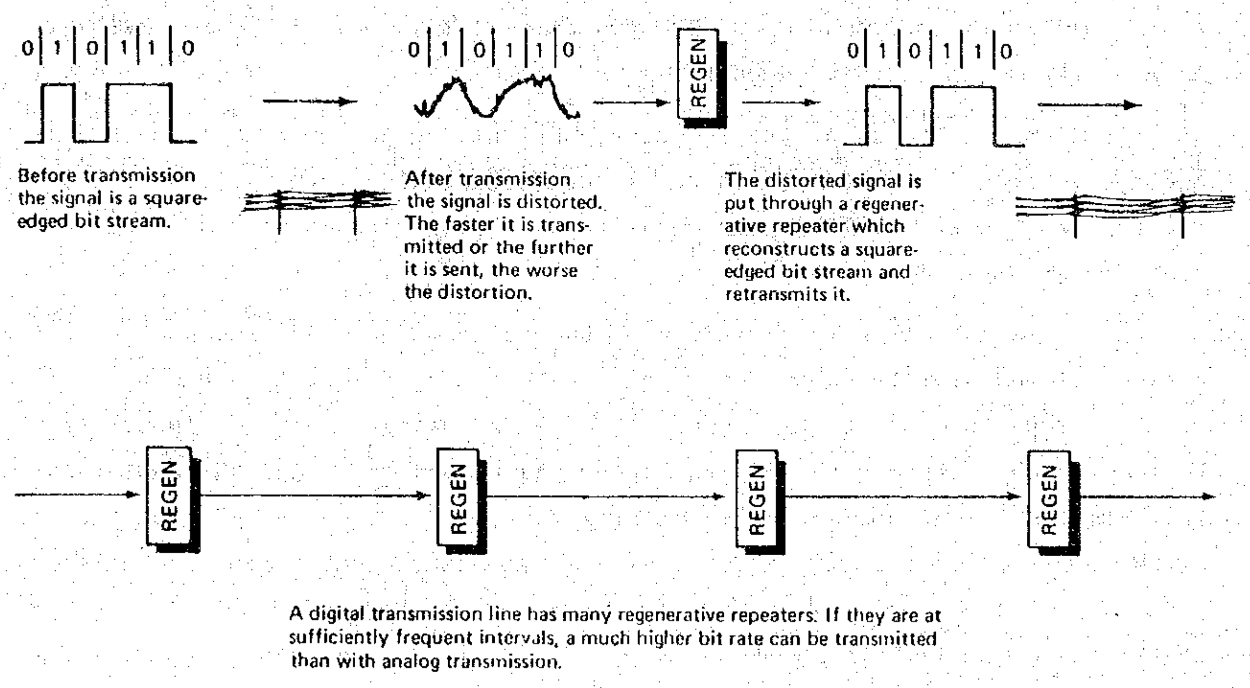


Fig. 6.1

Chapter 6 ANALOG - DIGITAL

Analog information such as the sound of the human voice needs to be converted to digital form before it can be transmitted over digital links. This conversion is done with a device which is called a codec. Just as "modem" is a contraction of the words modulated and demodulate, so "codec" is a contraction of the words code and decode.

At present instead of the computer users having to convert its data with a modem to travel over the telephone lines in an analog form, in near future the telephone users will have to convert its analog signals with a codec to travel over digital lines.

A telephone call when digitized by the telephone company requires 64,000 bits per second for transmission. This rate is much higher than the 4,800 or 9,600 bits per second at which data travel over analog telephone lines. At the balance of cost between the digital telephone transmission and the analog data transmission is, at present substantially in favor of the analog, but it will be the digital due to evolving of digital, computer, fiber optics technologies.

ADVANTAGES OF DIGITAL TRANSMISSION

What are the advantages of transmitting the telephone voice in digital form? With analog transmission whenever the signal is amplified, the noise is amplified with it. As the signal passes through its many amplifying stations, so the noise is cumulative. With digital transmission, however, each repeater station regenerates the pulses. New, clean pulses are reconstructed and sent on to the next repeater where another cleaning-up process takes place. And so the pulse train can travel through a

ECONOMIC FACTORS

The economic circumstances have been favoring digital transmission system from several factors.

First, the cost of digital circuitry produced in large quantities is dropping fast due to LSI (Large-scale integration) technology. Telecommunication systems employ large quantities of LSI circuit and so can be benefited from the mass-production economics of LSI.

Second, for analog transmission, it is becoming economical to build channels of higher bandwidth like the micro wave link or satellite communication links, but a high level of multiplexing is needed to make use of high-capacity channels. When many calls are packed in a high bandwidth, the cost of voice channel per kilometer drops greatly, but the cost of the increasing number of multiplexing and routing operations becomes high. Analog frequency-division multiplexing uses fairly expensive circuit components such as filters.

When thousands of telephone conversation travel together over coaxial cable or microwave links, they must be demultiplexed, switched and then multiplexed together again at each switching point. As the channel capacities increase, so the multiplexing and switching costs assume a greater proportion of the total network cost.

Digital circuitry, on the other hand, is dropping in cost at a high rate. With the maturing of LSI techniques, it will drop even more. Where digital trnasmission is used, this increasingly low-cost circuitry handles the multiplexing and switching. So the telecommunication networks becomes like a vast digital computer.

Third, the use of digital transmission makes it possible to operate on links with a high noise-to-signal ratio. On links with a wide-ranging tradeoff between noise-to-signal ratio and bandwidth, a somewhat larger number of voice channels can be derived by PCM techniques than by analog techniques. This is the case on today's wire-pair telephone lines and satellite channels, for example.

Fourth, digitization of voice channels can be done with more complex techniques than PCM, which will permit the speech to be encoded into fewer bits, as one chip CODEC which is recently developed.

Last, an additional economic factor is the rapidly increasing use of data transmission. Although data transmission still employs only a small proportion of the total bandwidth in use, it is increasing much more rapidly than other uses of the telecommunication networks. Data can be transmitted over digital voice links with a total equipment cost that can be as low as a tenth of that for transmission over analog links with modems.

In terms of the immediate economics of today's common carriers, pressed for increasing capacity, digital transmission is appealing for short-distance links because with relatively low-cost electronics it can substantially increase the capacity of existing wire pairs. This is particularly important in the congested city streets.

An important long-term advantage is the fact that all signals (voice, television, facsimile, and data) become a stream of similar-looking pulses. Consequently, they will not interfere with one another and will not make differing demands on the engineering of the channels. In an analog signal format, television and data are much more demanding in the fidelity of transmission than speech and create more interference when transmitted with other signals.

Chapter 7 MODEM & DATA TERMINALS

The signal we send through a communication line can travel to its destination by a wide variety of possible means. Whatever way the signal has traveled, original telephone signals are engineered through the telephone trunk to curve of attenuation against frequency. (See Fig. 4.3). It is therefore into this frequency range that we must fit our signal. The primary function of the line termination equipment (modem) is to fit data signal into transmission space (0.3k $\sim$ 3.4kHz).

FULL-DUPLEX VERSUS HALF-DUPLEX

Over a given physical line, the terminal equipment may be designed so that it can transmit either in both directions at once full-duplex transmission or else in either direction but not both at the same time half-duplex. All four-wire facilities are capable of full-duplex working (through sometimes one finds them used in a half-duplex manner). Some two-wire facilities can operate only in a half-duplex mode, through over many two-wire links full-duplex operation is possible.

A terminal or a computer line adapter will work somewhat differently depending on which of these possibilities is used. Where full-duplex transmission is employed it may be used either to send data streams in both directions at the same time or to send data in one direction and control signals in the other. The control signals would govern the flow of data and would be used for error control. Data at the transmitting end would be held until the receiving end indicated that they had been received correctly. Control signals would ensure that no two terminals transmit at once on a line with many terminals and would organize the sequence of transmission.

As we commented earlier, simultaneous transmission in two directions can be obtained on a two-wire line by using two separate frequency bands. One is used for transmission in one direction and the other for the opposite direction. By keeping the signals strictly separated in frequency, they can be prevented from interfering with each other.

The two bands may not be of the same bandwidth. A much larger channel capacity is needed for sending data than for sending the return signals which control the flow of data.

If, therefore, data are to be sent in one direction only, the majority of the line bandwidth can be used for data. Some modems provide a data channel of several thousand bits per second and a return channel of 75 bps, or less. Both channels can reverse their direction of transmission simultaneously.

FUNCTIONS OF LINE TERMINATION EQUIPMENT

The equipment at the ends of a telephone line on the user's premises carries out several functions:

1. It must handle the initial setting up of the connection (handshaking").
2. It must transmit and receive.
3. It must convert the digital signals into a form suitable for transmission and must convert them back again after transmission.

All these functions could be performed by the modem or line terminal equipment which constitutes an interface between the line and the data-processing machines which use the line. Often, however, the modem carries out only the last two (2, 3) functions. Dialing the call may be done by the operator. The error detection is done by the data-processing equipment. And automatic diagnosis of failures is too often not done at all.

There is a wide variety of different modems.

The main criteria for choosing between different types of design are:

1. Speed of transmission.
2. Cost.
3. Numbers of errors in transmissions.
4. Line turnaround time.

MODEM STANDARDS

Most of the countries nowaday adapt CCITT Modem Standard Recommendations. Following Box 7-1 shows CCITT MODEM Standards.

Box 7-1 CCITT recommendations for modem standards (1)

CCITT	Recommendation
	The interface between the modem and data processing terminal.
CCITT	Recommendation
V.24	Definition of interchange circuits between modems and data terminal equipment
V.25	Automatic calling and/or answering equipment and disabling of echo suppressors
	Modem standards for use on the switched telephone network
CCITT	Recommendation
V.21	200-baud modem
V.23	600/1,200-baud modem
V.26b	2,400/1,200-bps modem
V.30	Parallel data transmission modems
V.22	Standardization of data-signaling rates
V.25	Automatic calling and/or answering equipment and disabling echo suppressors
V.15	Use of acoustical coupling
	Modem standards for use on leased telephone line
CCITT	Recommendation
V.26	2,400-bps modem
V.27	4,800-bps modem
V.22b	Standardization of data-signaling rates
V.35	Data transmission at 48,000 bps using 60-108-kHz (channel group) circuits

A STANDARD 4,800-bps. MODEM

As an example of a modem standard we will discuss CCITT Recommendation V.27 (1) for a 4,800-bps modem.

This modem is intended for synchronous transmission on leased telephone circuits. It is capable of operating in half-duplex fashion or faull-duplex, meaning that a backward channel of up to 75 bps can transmit simultaneously with the forward transmission of 4,800 bps and that the direction of both of these channels can be reversed.

The modulation rate is 1,600 bauds (i.e., 1,600 separate line conditions per second). The data are divided into groups of three consecutive bits (tribits). Each tribit is transmitted as one change in line condition, thus giving 4,800 bps. The bit rate and hence the modulation rate are held constant to ±0.01%.

Each tribit is encoded as a phase change relative to the phase of the immediately preceding tribit. The encoding is as follows:

Tribit	Phase Changes(%)
001	0
000	45
010	90
011	135
111	180
110	225
100	270
101	315

The receiving modem detects these changes and converts them into the appropriate bits.

On a bad-quality line it is sometimes difficult to detect phase changes to the required level of accuracy. During periods of noisy transmission it would be of value to change to modulating dibits with 90% phase differences rather than tribits with 45% differences.

A data stream 0 0 1 0 0 1 0 0 1 ... could result in no phase changes, and this could result in loss of synchronization between the transmitting and receiving modems. Certain other repetitive bit patterns might also cause problems. To avoid the transmission of repetitive bit patterns the bit stream is scrambled before modulation and unscrambled by the receiving modem. The procedure for scrambling is specified in the CCITT recommendation.

The reverse channel of up to 75 is an optional feature of the modem. The reverse channel is organized like a standard voice-frequency telegraph channel with frequency-shift modulation: 390 Hz represents a 1 bit or mark; 450 Hz represents a 0 bit or space. The use for this reverse channel is to permit control signals, especially error control signals, to be sent simultaneously with the transmission of data in the opposite direction.

ECHO SUPPRESSORS

Echos bounce back from points in a telephone line at which its impedance changes. If the line is sufficiently long that the echo of a talker would reach him 45 milliseconds or more after he speaks, the echo must be removed. The receive path is attenuated so that a talker cannot hear the echos.

A device which suppresses echos would also suppress data, and so when full-duplex transmission is used the echo suppressors must be disabled.

INTERFACE BETWEEN MODEMS AND MACHINES

A standard interface between modems and business machine is specified in CCITT Recommendation V24. The American Electronic Industry Associations Standard RS-232C is the same. CCITT Recommendation V.25 gives standards for automatic calling.

Fig. 7.2 shows CCITT V.24

Fig. 7.3 shows CCITT V.25

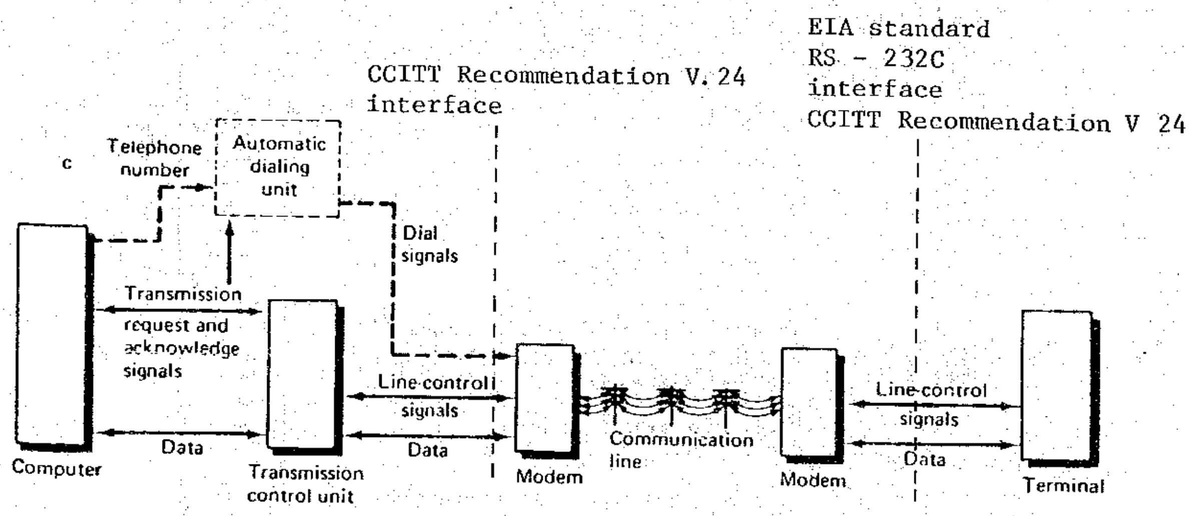


Fig. 7.1 The interface between the modem and terminal

Fig. 7.2 shows the detail signal interface between the modem and the data terminal Equipment (DTE).

Circuit No.	Function	
Grounds		
101	Protective ground or earth	—
102	Signal ground or common return	—
Data		
103	Transmitted data	→
104	Received data	←
118	Transmitted backward channel data	→
119	Received backward channel data	←
Control		
105	Request to send	→
106	Ready for sending	←
107	Data set ready	←
108/1	Connect data set to line	→
108/2	Data terminal ready	→
109	Data channel received line signal detector	←
110	Signal quality detector	←
111	Data-signaling rate selector	→
112	Data-signaling rate selector	←
116	Select standby	→
117	Standby indicator	←
120	Transmit backward channel line signal	→
121	Backward channel ready	←
122	Backward channel received line signal detector	←
123	Backward channel signal quality detector	→
124	Select frequency groups	←
125	Calling indicator	→
126	Select transmit frequency	→
127	Select receive frequency	→
129	Request to receive	→
130	Trasmit backward tone	→
132	Return to nondata mode	→
133	Ready for receiving	→
134	Received data present	←
191	Transmitted voice answer	→
192	Received voice answer	←
Timing		
113	Transmitter signal element timing	→
114	Transmitter signal element timing	←
115	Receiver signal element timing	←
128	Receiver signal element timing	→
131	Received character timing	←
Automatic calling		
201	Signal ground or common return	—
202	Call request	→
203	Data line occupied	←
204	Distant station connected	←
205	Abandon call	←
206	Digit signal (2^0) 4 bits giving the number dialed	→
207	Digit signal (2^1)	→
208	Digit signal (2^2)	→
209	Digit signal (2^3)	→
210	Present next digit	←
211	Digit present	→
212	Protective ground earth	—
213	Power indication	←

Kev: From the terminal ——
To the terminal ——

Fig.7.2 A complete list of the circuits in the CCIT V.24 interface between modem and data machine.

	Unit actions			
1.	Checks initial conditions			On hook
2.	Puts circuit 220 ON	CALL REQUEST ON →		
3.	Puts circuit 108/2 ON	DATA TERMINAL READY ON →		
4.	Puts circuit 105 ON	REQUEST TO SEND ON →	Puts line "off-hook"	Off hook
5.		← DATA LINE OCCUPIED ON	Puts circuit 203 ON	
6.				Telephone network responds with a dial tone
7.		← PRESENT NEXT DIGIT ON	Puts circuit 210 ON	
8.	Presents first dial digit and puts circuit 211 ON	Circuit 206 → Circuit 207 → Circuit 208 → Circuit 209 → DIGIT PRESENT ON →		
9.		← PRESENT NEXT DIGIT OFF	Modem dials the digit and takes circuit 210 OFF	Telephone number is dialed →
10.	Takes cirucit 211 OFF	DIGIT PRESENT OFF →		
11.	Steps 7 to 10 are repeated with all of the digits to be dialed and then with a (4-bit) end-of-number code			
			Modem places a calling tone on the line. The calling tone is interrupted at intervals to permit the disabling of suppressors	980 or 1,300 Hz calling tone →
12.			When the 2,100 Hz response is recorgnized for 450 to 600 milliseconds, the calling tone is discontinued	← 2,100 Hz response (which disables echo suppressors)
13.		← DATA SET READY ON	Puts circuit 107 ON	
		← DISTANT STATION CONNECTED ON	Puts circuit 204 ON	
		← READY FOR SENDING ON	Puts circuit 106 ON	
14.	The computer sends data	Data on circuit 103 →	The modem modulates and transmits the data	Data transmission →

Fig. 7.3 The sequence of events when a transmission control unit initiates the automatic dialing of a call on the telephone network and transmits data when the connection is made. This illustration shows the use of the interface circuits and frequencies conforming to CCITT Recommendation V.25.

Box 7.2 Circuits commonly used in the interface between a modem and a data terminal*

	CCITT Circuit No.
1. Data signals	
(a) Transmitted data (to the modem). Data generated by the terminal for transmission.	103
(b) Received data (to the terminal). Data received by the modem for the terminal.	104
2. Timing signals	
(a) Transmitter signal element timing. Two connections are defined. One sends signal element timing information from the transmitting terminal to its modem. The other sends timing information from the transmitting modem to its terminal.	113, 114
(b) Receiver signal element timing. Two connections are defined. One sends signal element timing information from the receiving terminal to its modem. The other sends timing information from the receiving modem to its terminal. The timing signal connections are optional. A modem for start-stop transmission does not use them.	115, 128
3. Control signals	
(a) Request to send (to the modem). Signals on this connection are generated by the transmitting terminal when it wishes to transmit. The modem's carrier signal is transmitted during the ON condition of this connection. (With half-duplex operation, the OFF condition of this connection holds the modem in the receive-data state).	105
(b) Clear to send (to the terminal). Signals on this connection are generated by the transmitting modem to indicate that it is prepared to transmit data. They are a response to the Request to Send signal from the transmitting device. (With full-duplex operation the modem is in the transmit state at all times.)	106

* CCITT Recommendation V.24 and EIA Standard RS-232-C.

		CCITT Circuit No.
(c)	Data set ready (to the terminal). Signals on this connection are generated by the local modem to indicate to the transmitting machine that it is ready to operate.	107
	(The following control signals are optional)	
(d)	Data terminal ready (to the modem). When the terminal sends the ON condition on this connection it causes the modem to be connnected to the communication line. The OFF condition causes it to be disconnected, in order to terminate a call or free the line for a different use.	108
(e)	Ring indicator (to the terminal). A signal on the connection informs the terminal that the modem is receiving a ringing signal from a remote location.	125
(f)	Data carrier detector (to the terminal). A signal on this connection indicates to the terminal that the carrier (the sine wave that carries the signal) is being received. If the carrier is lost because of a fault condition on the line, the terminal will be notified by an OFF condition in this connection.	109
(g)	Data modulation detector (to the terminal). An ON condition on this connection informs the terminal that the signal is being demodulated correctly by the modem. When the quality of demodulation drops below a certain threshold the terminal may take corrective action such as requesting retransmission or requesting that a lower transmission rate be used.	110
(h)	Speed selector. There are two speed selector connections, one to the modem and one to the terminal. Using them, the transmission rate may be changed.	111, 112
4.	Grounds	
(a)	Protective ground. Attached to the machine frame and possibly to external grounds.	101
(b)	Signal ground. Establishes the common ground reference potential for the circuits.	102

HIGH SPEED LINE TERMINATION

Lines with a bandwidth higher than that of a Telephone can be leased. CCITT Recommendation V.35 is for a 48,000 bps Modem operating in the 60 to 104 kHz band. Suppressed carrier AM modulation of a 100 kHz carrier is recommended.

With the spread of PCM and digital circuits now operate over digital lines and hence not need modems.