British Telecommunications Engineering

VOL 3 PART 2 JULY 1984

The Journal of
The Institution of British Telecommunications Engineers

BRITISH TELECOMMUNICATIONS ENGINEERING

VOL 3 PART 2 JULY 1984

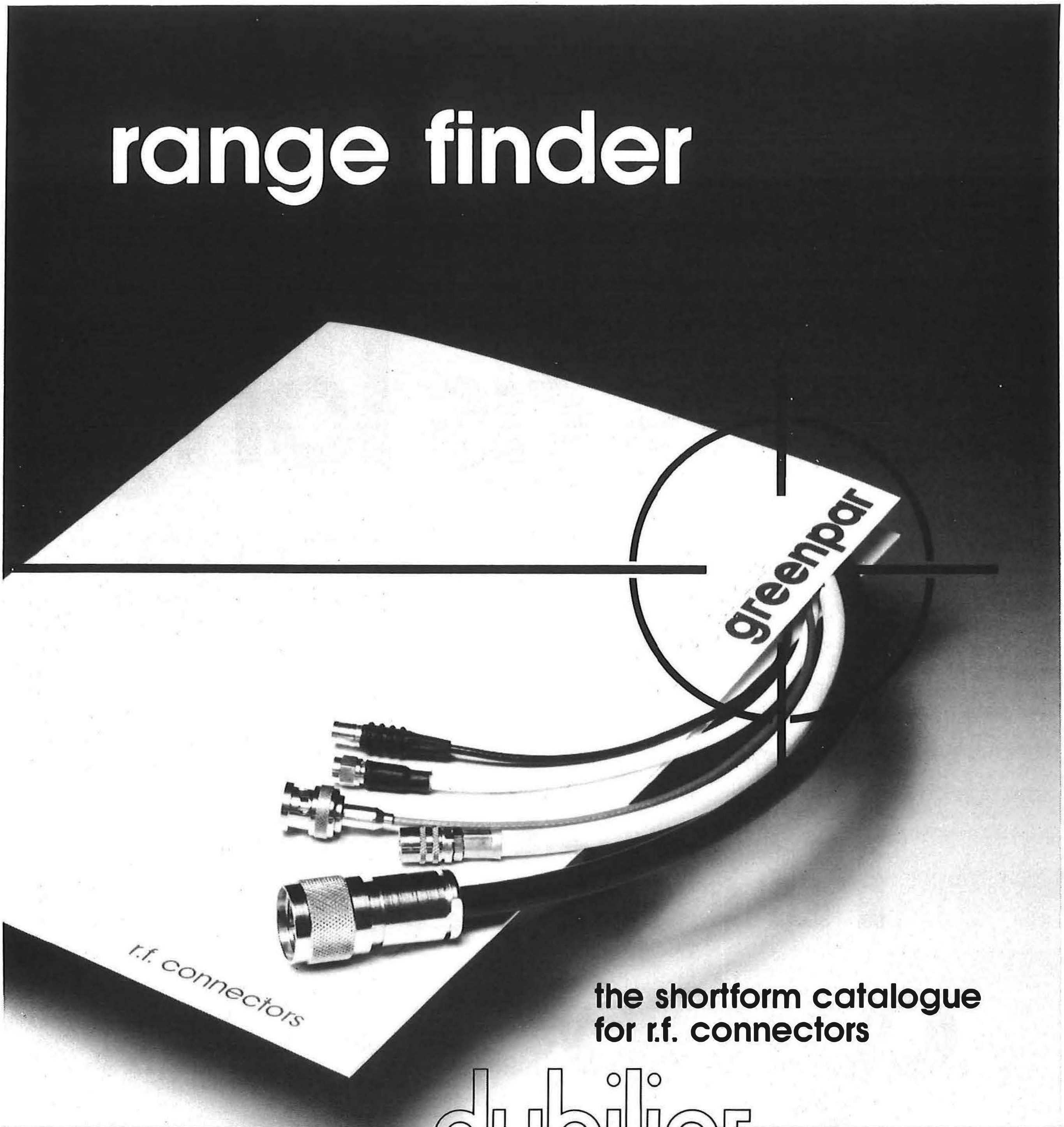
Contents

Published in April, July, October and January by *British Telecommunications Engineering Journal*, 2–12 Gresham Street, London EC2V 7AG.
(Formerly *The Post Office Electrical Engineers' Journal* Vols. 1–74: April 1908–January 1982)
Price: 90p (£1·40 including postage and packaging). Orders by post only.

Annual subscription (including postage and packaging): home and overseas £5·60.
(Canada and the USA $12·00).

Price to British Telecom and British Post Office staff: 48p.

ISSN 0262-401X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British Telecommunications Engineering

EDITORIAL

Twenty years ago, in 1964, two events took place which were to have a major impact on the telecommunications network in the UK. The first event, which marked the beginning of what has become known as the *Information Technology* era, was the introduction of the first datel service (Datel 100). Today, some 100 000 modems are in use in the British Telecom network. The transmission of data is the fundamental concept of Information Technology, but the practical realisation of this task has required considerable effort in the development of operating protocols and standards. The basic principles involved in providing such data communications networks are outlined in a series of articles in this issue of the *Journal*.

The second event was the introduction of digital transmission techniques with the provision of the first field-trial pulse-code modulation system. Digital transmission equipment is now being provided as a matter of course. Since digital transmission, however, is equally suited to data or telephony traffic, the idea of a single general-purpose network, the integrated services digital network (ISDN), was born. But the success of such ISDNs is greatly dependent on the attainment of acceptable levels of performance. Much work has been done in recent years, under the auspices of international organisations, to develop appropriate performance standards for networks and equipment. Three articles in this issue of the *Journal* discuss the objectives for the error performance of digital networks with particular reference to jitter, synchronisation and slip, and take due account of the final meeting of the International Telegraph and Telephone Consultative Committee Study Group XVIII for the current study period (1980–1984).

An Introduction to Data Communications

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UDC 621.394.4 : 681.32

Data transmission over the public switched telephone network was first introduced some 20 years ago. Since then, there has been a rapid growth in data services, both nationally and internationally, often using dedicated data networks. This article introduces some of the concepts behind data communication and discusses the need for international standardisation.

DATA

Data is information, but to define it so is not very revealing. However, a more precise definition depends upon the context in which the word is used. In the present context of data communications, data is information represented by an electrical binary signal typically found at the input and output of a computer or computer peripheral.

The electrical representation of binary data comprises two parts (see Fig. 1): the data signal, which is always at one of two possible states (for example, voltage levels); and a data clock, where a pulse indicates the precise instant the data signal must be read in order to ascertain the binary value (that is, whether the voltage level is high or low). Read in this way, the signal in Fig. 1 would appear to be the bit sequence 10001010, where 0 is one binary state and 1 is the other. It is important to appreciate the dual nature of binary data and how absolutely the interpretation of the data signal depends on the data clock. For instance, if the clock frequency in Fig. 1 was half that shown, then the data signal would be read as either 0000 or 1011 depending on the phase of the clock.

Thus, for meaningful data transfer, the receiving device must be provided with both the data signal and associated clock, and this can be achieved *asynchronously* or *synchronously*.

Asynchronous data transfer is the simplest, since only the data signal is transmitted, the clock being generated locally at the receiver; that is, the receiver is not locked in synchronism to the transmitter. In synchronous data transfer, on the other hand, the receiver and transmitter are locked in synchronism. The receiver receives from the transmitter both the data signal and clock, normally combined as one signal on the line.

A binary signal such as that shown in Fig. 1 is typically present at the input/output of a computer or peripheral because it is compatible with the electronic inner workings of the machine itself; however, while the smallest unit of data is the binary digit (bit) represented by one of two voltage levels, for convenience called *zero* and *one*, data is invariably organised in groups of eight bits. A group of eight bits is known as a *byte* or *octet*, and can be encoded in 2^8, or 256, different ways ranging from 00000000 to 11111111.

In the context of data communications, it is useful to consider data to be 'an integral number of octets' and for data transmission to be 'the process of transferring data from one location to another via a transmission path or circuit either switched or point-to-point (leased line/private circuit)'.

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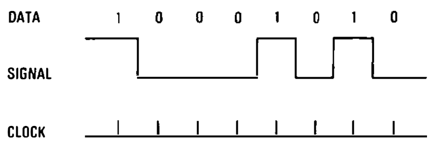


FIG. 1—Binary Data

SEMI-OCTET	HEX CODE
0000	0
0001	1
0010	2
0011	3
0100	4
0101	5
0110	6
0111	7
1000	8
1001	9
1010	A
1011	B
1100	C
1101	D
1110	E
1111	F

FIG. 2—The hexadecimal code

OCTETS

The basic building block of data communications is the octet. Thus, in its simplest form, data transmission consists of establishing an electrical connection, transmitting and/or receiving an integral number of octets, then clearing down the connection. However, before these processes can be discussed in more detail, certain conventions concerning octets must be explained.

Literally, an octet is one of 256 different arrangements of 8 binary digits, and a very convenient shorthand method of representing each is known as the *hexadecimal* (hex) code. Each octet is regarded as two semi-octets, each of four bits, which is given a hex code as shown in Fig. 2. Thus, the 256 possible octets range from 00 to FF, and the octet shown in Fig. 1—10001010—can be more compactly written in hex as 8A, since 8 represents the first four bits and A the second four bits reading from left to right.

In the same way that bits are organised into octets, octets themselves are often organised into groups to serve a particular function.

Depending on the context, groups of octets are known variously as *blocks*, *frames* or *packets*. Each of these structures is merely a group of one or more octets and is usually

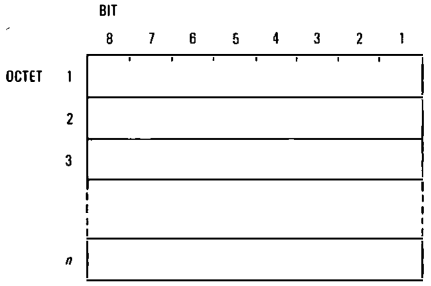


FIG. 3—A group of contiguous octets

represented as shown in Fig. 3, although the octet and bit numbers are often omitted. A group of one or more contiguous bits which perform a particular function within an octet structure is usually referred to as a *field*.

Note that, conventionally, bit 1 of an octet is known as the *least significant bit* and is transmitted to line first, and bit 8 is the *most significant bit* and is transmitted last.

Data transmission networks are invariably transparent to user data, and any combination of all the octets from 00 to FF is delivered safely to the destination. However, for users of switched data transmission networks, there is a stage in every call when a dialogue between the user and the network must take place. In the simplest case, this is to signal to the network the address of the required remote data terminal, although the caller might wish to request some extra service from the network at the same time, for example, reverse charging. Any user/network dialogue must be in a language and to rules (protocol) specified by the network. A language in common use for this purpose is the International Alphabet Number 5 (IA5), which is virtually the same as the American Standard Code for Information Interchange (ASCII). These languages define a way of encoding octets to represent the alpha-numeric control and formatting characters output by the keyboard of a character terminal.

Fig. 4 shows a sequence of three IA5 characters. Each character is always carried as bits 1–7 in a separate octet. Bit 8 of each octet is called the *parity bit* and may be coded ZERO or ONE. As can be seen from Fig. 4, the IA5 code for lower case 'i' is 0101001, and this may be transmitted as either the octet 00101001 (hex 29) or the octet 10101001 (hex A9) depending on the setting of the parity bit. Although the setting of the parity bit makes no difference to the meaning of the IA5 code, it can be used as part of an error detection mechanism. The way it works is that on transmission of an octet containing an IA5 character code, the parity bit is set to ZERO or ONE; that is, whichever makes the total number of ones in the octet EVEN (in the case of *even parity* working) or ODD (if working with *odd parity*). At the receiver, if an octet is received with an odd number of ones when the link is supposed to be working with even parity (or vice versa), an error alarm is raised. Parity working is optional and, therefore, not always used, in which case the eighth bit is fixed at ZERO (space parity) or ONE (mark parity).

BIT	8	7	6	5	4	3	2	1	
OCTET 1	P	1	0	0	1	0	0	0	H
2	P	0	1	0	1	0	0	1	i
3	P	0	1	0	0	0	0	1	!

P : Parity Bit

FIG. 4—Characters 'Hi!' in IA5

IA5 is just one way of encoding octets, and is used in communication to and/or from character terminals where it is necessary for the information carried in one keystroke to be immediately transmitted to line. It is by no means universal even for character terminals, and for many types of data, for example digitised speech, it would be inappropriate.

DATA TRANSMISSION

Data transmission is the process of transferring data in the form of integral numbers of octets from one location to another via a transmission link. Basically, there are two types of transmission: asynchronous and synchronous. Asynchronous transmission is comparatively inexpensive but, as it does not make very efficient use of the circuit, it is used only for low-speed data; for example, up to 1200 bit/s. Synchronous transmission is efficient, and makes maximum use of the bandwidth available; it is expensive, but necessary in order to transmit data at high speed.

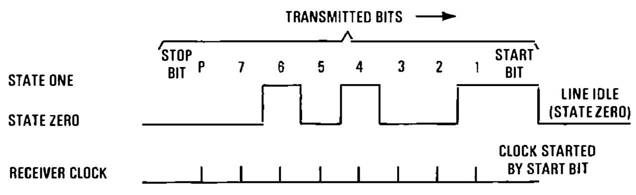
Asynchronous Transmission

With a typical data terminal, each time the keyboard is operated, an IA5 character is generated. These characters are generated asynchronously, which literally means that there is no time relationship between them—their generation is more or less random. Asynchronous transmission involves transmitting these characters as they are generated, but in the form of octets by the addition of a parity bit in the most significant position, with the transmission line returning to an IDLE state between each octet.

Fig. 5 shows the principle of asynchronous transmission. The analogue transmission line is always in one of two states represented, not by voltage levels as at a device interface, but by modulations of a carrier frequency. In the IDLE condition, the line is at ZERO and the receive clock is stopped. When the transmitter has an octet to transmit, it first transmits a *start* bit; that is, it switches the line to state ONE for one bit period, then transmits the eight bits of the octet as a sequence of one bit periods each of either ONE or ZERO. Finally, the transmitter reverts to line IDLE for a minimum of one bit period. At the asynchronous receiver, the clock is started by the start bit and free runs generating a sequence of eight clock pulses indicating the instants at which the incoming line must be read. The receive clock has to remain in synchronism with the received octet only for eight bit periods, so it does not have to be highly stable. At a typical line bit rate of 300 bit/s, an octet is received in 30 ms.

Asynchronous transmission is popular because it is simple and, therefore, comparatively inexpensive. Octets arrive at the receiver separated by IDLE and so octet synchronism is automatically achieved. Also, as the receiver's bit clock is started afresh by the arrival of each octet and has to remain in reasonable synchronism only for the duration of the octet, bit synchronism as described is no problem either. However, despite its many advantages, asynchronous transmission is suitable only for low-speed data for two reasons:

(*a*) the addition of start and stop bits to every octet



P : Parity Bit

FIG. 5—Asynchronous transmission of IA5 character 'i'

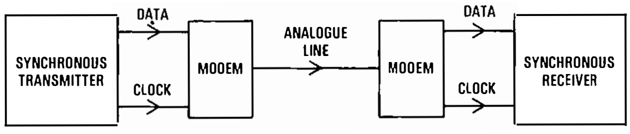
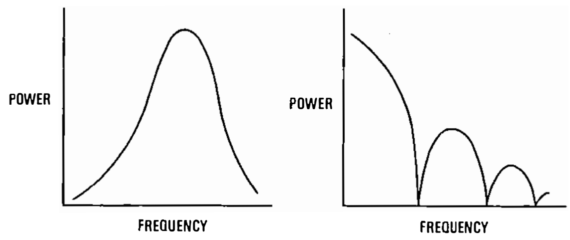


FIG. 6—Synchronous data transmission



(*a*) Telephone signal power (*b*) Data signal power

FIG. 7—Telephone and data signals

introduces an overhead of 25%, and

(*b*) a free running clock at the receiver is a satisfactory synchronisation method only at low speeds.

Synchronous Transmission

Synchronous transmission is so called because the receiver must be kept permanently in synchronism with the transmitter, and to facilitate this, the synchronous transmitter outputs a continuous contiguous stream of octets all the time it is switched on. Even when there is no data to transmit, line IDLE consists of contiguous octets. Synchronous transmission is efficient, no start or stop bits are added; bit 8 of one octet is followed immediately by bit 1 of the next. It is used for buffer-to-buffer transfer of data at the maximum speed the transmission path can support.

Fig. 1 shows data having two components: a data signal and data clock. Asynchronous transmission requires only the transmission of the data signal component plus start and stop bits to control the free-running receive clock. However, a synchronous receiver must be kept permanently in synchronism with the transmitter, a free-running clock is not suitable, and so both components of the original data (that is, the data signal and the clock) must be transmitted to the receiver. The receiver is then locked in synchronism with the transmitter (see Fig. 6).

The data signal and clock signal components are combined in the transmit modem (the modem has another equally important function—see next section), the composite signal is transmitted to line and the two components separated again by the receive modem.

MODEMS

Most data transmission today must take place over analogue-based networks optimised for the transmission of telephony, because digital networks (for example, KiloStream, MegaStream and ISDN) are only now being introduced. Fig. 7(*a*) shows approximately how the power in the output of an analogue telephone instrument is concentrated into a 3 kHz frequency band. Of note is the absence of a DC (zero frequency) component.

Fig. 7(*b*), on the other hand, shows the power/frequency distribution of a typical digital baseband waveform such as that shown in Fig. 1. The power is at a maximum at zero frequency, and is spread over a wide frequency band.

If the data waveform was applied to a telephone connection, it would not only suffer severe distortion, with most of its power being lost as a result of the bandpass filter effect of the telephone circuit, but also the telephone network would suffer interference from the wide range of frequencies injected into it, some of which could possibly coincide with network signalling frequencies in the case of switched connections.

A Modem (*mo*dulator/*dem*odulator) is an electronic device that converts a digital signal into a line signal suitable for transmission over an analogue circuit, and vice versa.

Essentially, the modulator shifts and concentrates the energy of the digital signal into the available bandwidth, the demodulator reverses the process.

Phase-shift keying (PSK) is a technique commonly used to realise the function of the modem and involves transmitting to line one audio frequency to represent binary zero and another to represent binary one. For full-duplex† working, this technique involves four different audio frequencies, two for each direction of transmission, although at any one time there will only be two on the line: one being transmitted and one being received.

Despite having to operate over connections possibly having non-linear electrical characteristics and subject to noise and short breaks in transmission, modems can make the PSTN a very tolerable data network. However, there are disadvantages; modems are relatively expensive and, despite their ever increasing sophistication, cannot reduce the PSTN connection set-up time.

As stated previously and shown in Fig. 6, synchronous modems also have to carry the timing component of the data waveform. The way this is normally achieved is to ensure that, whatever data is being transmitted, the line is always active; that is, always contains transitions from one binary state to another even if the original data is all zeros or all ones. This often means another stage of processing, involving more complication and, consequently, more expense.

Other functions offered by modems can include test loops for local and remote operation; other test facilities, for example a pseudo-random generator; error detection and correction; PSTN call set-up and answering; password generation; encrytion; automatic calling-bit-rate detection and fall-back to lower rate etc.

DATA NETWORKS

A network provides facilities for its users to intercommunicate. By common usage, a network is assumed to be a switched network, although leased-line networks exist. Communicating via a switched network is, of necessity, more complex than communicating directly or via a leased-line (private circuit) because there has to be a set-up and clear-down phase during which communication is with the network not the distant user, and these different phases must be clearly distinguished.

Because data is digital, it would be logical for a data network to be digital too. The user's physical interface to a digital data network would consist of just five major connections: two connections for the data transmit and receive signals, a single clock connection from the network used for both transmit and received binary data; and, of the two remaining connections one would be used by the user to indicate the status of the data being transmitted and the other by the network to indicate the status of the data being received. The status identifies the data as either user data or network signalling. Unfortunately, this interface is possible only if the user's local access circuit is digital and, at

† Full-duplex working involves transmission and reception in both directions simultaneously. In half-duplex working, transmission and reception are possible in both directions, but not at the same time; the circuit can accept traffic in one direction at a time. With simplex transmission, communication is in one direction only.

present, all local access circuits are analogue. And it is worth repeating that an analogue circuit is so called because it is designed to transmit the electrical analogue of speech and, for it to transmit data, modems are required.

Modems considerably complicate the user interface both physically, by increasing the number of interchange circuits required, and procedurally because data transmission cannot proceed until the transmit and receive modems indicate that a transmission path is established between them.

Many Administrations, including British Telecom (BT), have plans to introduce digital local access circuits and, in the UK, the first of these will be provided as part of the ISDN pilot network, which will be linked to the penetration of System X exchanges.

However, in the short and medium term, data networks will continue to offer their users an interface to an analogue local access circuit via a modem.

The PSTN as a Network for Data Transmission

The telephone network is unique, a worldwide network dedicated to providing just one type of circuit between one type of terminal—a nominal 3 kHz circuit between telephone instruments. A data network in the same sense is not possible because of the lack of anything even remotely resembling the data equivalent of the telephone instrument. Data terminals have only one thing in common—the octet. Beyond this, however, because of the wide variety of data transmitters and receivers there can be no consensus as to what frequency the bit clock should be or even whether the octets should be transmitted synchronously or asynchronously.

The PSTN was used initially for data transmission about 20 years ago and, since that time, thanks to advances in modem technology, the data carrying capacity (measured in bit/s) of PSTN connections has steadily increased. BT currently offers 2400 bit/s full duplex and a state-of-the-art 9600 bit/s full duplex modem is on the horizon. There is just one characteristic of the PSTN that lets it down as a data network which even the ingenuity of the modem designer cannot overcome. Call set-up on the analogue-based PSTN takes too long for most real-time data applications. For instance, credit-card evaluation requires an overall response time of 2 or 3 s at most to be viable. The requirement to access remote databases on-line cannot be satisfied by the analogue PSTN: a network with a very short call set-up time is required. Also, the PSTN tariff is duration oriented whereas a volume-based tariff is fairer for data traffic.

Dedicated Data Networks

Many administrations have introduced dedicated data networks as a way of overcoming the limitations imposed by the PSTN. However, unlike the PSTN, which is a circuit-switched network, most public data networks introduced throughout the world to date operate in the packet-switching mode. Packet switching is the dominant technology in worldwide data communications, which explains the amount of space given to it in this article.

A circuit-switched connection, once established, exists between the users until cleared down and is perhaps best for transferring data in bulk at high speed. A packet-switched data network (PSDN), on the other hand, operates on a different principle. A packet is typically up to 128 contiguous octets of user's data, and is accepted by the network for delivery to the remote user some tenths of a second later. A user can transmit a continuous stream of packets into the network and each is switched to the same or different remote terminals as required. Packet switching is an important technique, and networks based on the principle are able to offer their users many valuable extra facilities on top of the basic one of intercommunication. In order to understand why this is so, it is useful to compare circuit and packet switching. But first a word about data network standards.

Data Network Standards

For a terminal to transmit data to and receive data from a network, it must have a compatible interface. Compatibility must extend over three levels: the physical, electrical and procedural. In the same way, for a network to transmit data to or receive data from another network, compatible interfaces must exist.

The body responsible for telecommunications standards is the Consultative Committee on International Telegraphy and Telephony (CCITT). CCITT standards exist for interfaces to both circuit-switched and packet-switched networks.

Circuit Switching

Standards for interfaces to circuit-switched networks are CCITT Recommendations X20, X21 and X22. For reasons which will become apparent later, very few administrations have implemented circuit-switched data networks. However, BT is offering users the physical and electrical levels of X21 and X22 interfaces as part of the KiloStream service. KiloStream circuits are digital leased lines.

Packet Switching

In the early 1970s, several countries implemented PSDNs. But the network/user interface of each was different, particularly at the procedural level, so that each network remained isolated with no interworking possible. Spurred on by this undesirable state of affairs, a group of interested parties from many countries, including the UK, and under the auspices of the CCITT, produced two interface standards for users to access a PSDN: one for packet-mode operation, and the other for non-packet-mode, together with a single packet-mode internetwork interface standard as shown in Fig. 8. These standards are:

(*a*) *CCITT Recommendation X25* This defines an interface for users to access a PSDN in the packet mode. As no data terminal naturally sources or sinks data in this mode, the interface is usually to a user's communications handler or front-end processor, which packetises data on behalf of a number of users' processes.

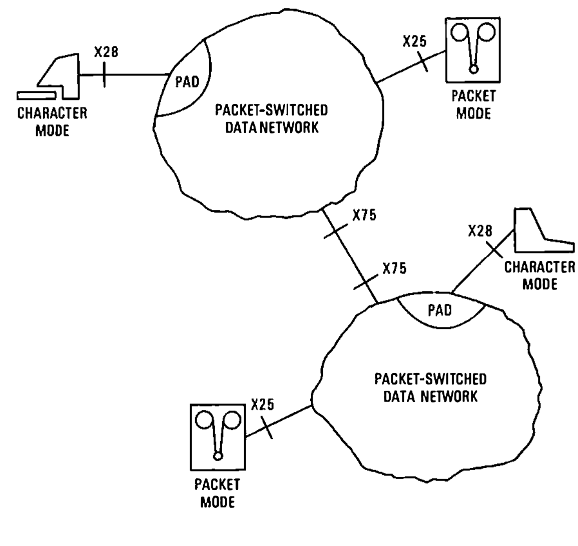


FIG. 8—PSDN interfaces

(*b*) *CCITT Recommendation X28* This defines a simple asynchronous interface suitable for non-intelligent terminals to access a PSDN. The asynchronous terminal accesses a network-based packet assembler/disassembler (PAD) usually via the PSTN, which packetises asynchronous data and vice versa. The PAD permits users with X28 and X25 interfaces to interwork—an important facility for database access. There are two other CCITT Recommendations associated with the PAD which together with X28 are often referred to as *triple X*. These are X3, which defines the facilities a public PAD must offer, and X29, which specifies the small amount of protocol required in addition to that of X25, in order to permit a packet-mode terminal to interwork with a non-packet-mode terminal via a PAD.

(*c*) *CCITT Recommendation X75* This defines a packet-mode internetwork interface (see Fig. 8) and, as such, it is similar to the packet-mode user/network interface X25. Although designed to permit interworking between packet-switching exchanges (PSEs) in different networks, the X75 protocol can also be used, with minor modification, to link PSEs in the same network.

Some measure of the value and acceptability of these Recommendations can be judged by the fact that since their introduction in 1976, they have been used in the 50 or so new public PSDNs, with consequent advantages to users and manufacturers of equipment.

There are problems of course: the Recommendations are not only complex and, therefore, open to various interpretations, but also, as originally introduced, have proved deficient in a number of areas; for example, network user addressing. Also, because wide international agreement has to be obtained, agreed solutions are difficult to find, and even when a solution is agreed, it is implemented urgently on some networks and less urgently on others so that networks may vary in the facilities they offer their users from time to time. Thus, a facility enjoyed by a user on a national call may be unavailable on an international call. Also, international communications are affected by the policies of the Administrations of the various countries and some facilities (for example, reverse charging) are proving difficult administratively rather than technically.

However, despite the problems, the considerable growth of PSDNs conforming to the CCITT Recommendations indicates clear user acceptance of the network interfaces and procedures. Demand from users for advanced data communications facilities continues to grow, but unless these are provided both nationally and internationally, there could be a proliferation of private networks each adopting its own standards. This situation has been avoided in the telephony area, where all telephony networks can interwork via CCITT Recommendations, which is to everyone's advantage. It is equally desirable that data communications follow the same path and, to this end, considerable resources are being devoted by many interested parties to the progressing of both ISO† and CCITT work.

Circuit-Switched Data Networks

Fig. 9 shows diagramatically a user with several data terminals connected to a circuit-switched network, and it can be seen that many individual data circuits to the exchange are required from the user's front-end processor: at least one for each data rate and more if simultaneous calls at the same rate are required. Also of note is that the switch has to switch many different data rates, and inter-node trunks at each rate are also required. A circuit-switched data network is really many separate networks overlayed—one for each data rate supported and is thus not as efficient or economic

† ISO—International Standards Organisation

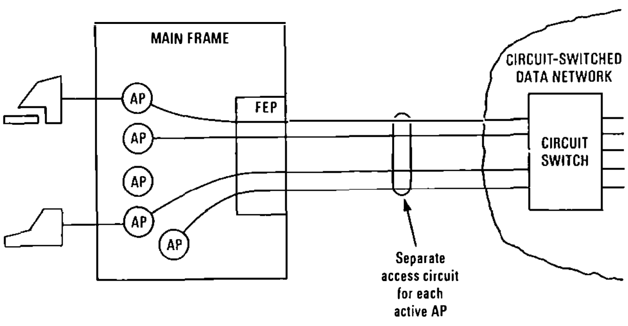


FIG. 9—Access to a circuit-switched data network

as a single network which can carry any signalling rate and which allows interworking between different rates.

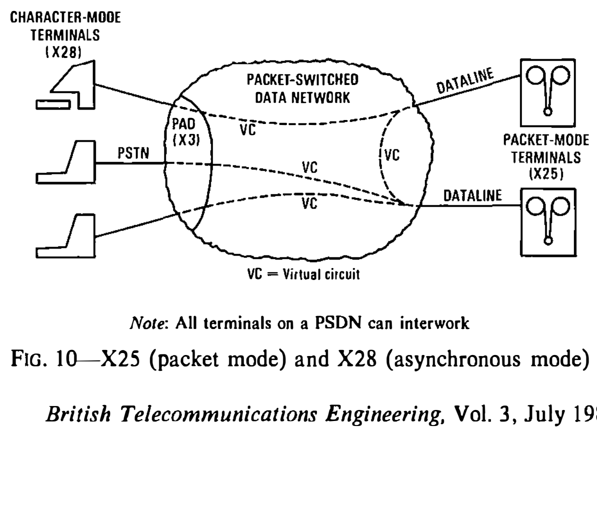
Another characteristic of a circuit-switched network is that any intelligence the network has can generally be used only during call set-up. This is because after call set-up, all the network can offer is a transparent end-to-end circuit at a specified data rate. Because of the real-time nature of the connection, there can be no processing of the user's data. This means that facilities such as reverse charging, closed user groups and call barring are possible, whereas facilities that act on the user's data such as error protection, speed and protocol conversion are not.

One of the advantages of a circuit-switched connection is its simplicity; once the connection has been established, the resources are reserved and the network plays no further active part in the end-to-end communication. A circuit-switched network is not dynamic and, except at call set-up, when congestion is possible, the activities of one user have no way of affecting any other users. Having resources allocated at call set-up time does have disadvantages, however, because should anything go wrong with those resources, the call fails. It is then up to the user to detect this failure and to re-establish the connection.

In a packet-switched network, very few resources are allocated to individual users; instead, resources are shared, and this has the characteristic that, when failure occurs, it is felt by all users as a slight degradation of service, for example, as an increase in packet transit time, instead of a severe failure for just a few.

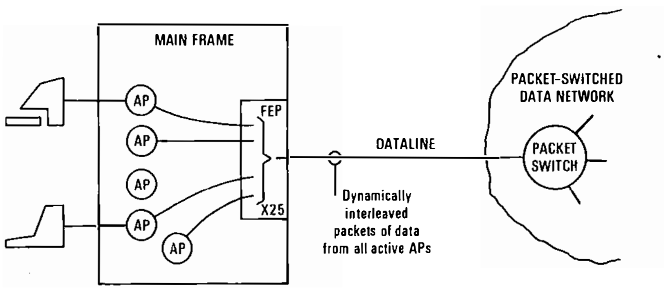
Packet-Switched Data Networks

Fig. 10 gives an overall view of a PSDN showing packet-mode (X25) and non-packet-mode (X28) access. So far, packet-mode terminals always access via direct connections (Datalines) although a protocol for access via the PSTN is in the course of preparation at CCITT. Non-packet-mode terminals, because of their relatively low speed, usually access via the PSTN, although Dataline access is available.



Note: All terminals on a PSDN can interwork

FIG. 10—X25 (packet mode) and X28 (asynchronous mode)



AP : Application process
FEP : Front-end processor

FIG. 11—Packet-mode access to a PSDN

Additionally, non-packet-mode terminals, whatever their method of access, are connected to a PAD. The PAD packetises and depacketises the asynchronous data making it possible for asynchronous terminals to use the PSDN to access high-speed data terminals, for example, databases that have moved from the PSTN in favour of a Dataline connection to the PSDN.

Four different Datalines are available: 2400, 4800, 9600 and 48 000 bit/s. However, as explained below, apart from fixing the maximum aggregate data rate, the Dataline speed is of no importance as the network performs speed matching both between packet-mode terminals and between packet-mode terminals and character terminals connected to a PAD.

To compare the access arrangements of circuit-switched and packet-switched networks, Fig. 11 shows the same application processes of Fig. 9 served by a PSDN, for example, Packet SwitchStream (PSS). The most noticeable difference is that a packet-mode customer of a PSDN requires only a single Dataline regardless of the number of data processes.

To understand the access mechanism and therefore the packet principle, it is useful to assume that only the visual display unit (VDU) in Fig. 11 is generating data. The user's packet terminal accepts data from the VDU at a locally convenient transmission mode and bit rate and stores it in a 128-octet buffer. When the buffer is full, all 128 octets are transmitted to the packet switch, at the speed of the Dataline, together with a 3-octet header, which the network uses to route the packet. If, at this stage, another of the user's processes starts to generate data, this too, quite separately, is stored and then transmitted as a packet, at the speed of the Dataline, to the packet switch. The data from very many separate user's data processes may be packetised by the packet terminal in this way and the resulting packets dynamically interleaved on the Dataline, and sorted out and routed by the packet switch. The interleaving is dynamic because packets are transmitted by the packet terminal when a buffer is full rather than in any particular order.

A second and perhaps even more useful characteristic of a PSDN, speed matching, can best be explained by following the path of a packet across the network. This path is called a *virtual circuit* because no actual physical circuit exists. Nevertheless, in any data transfer session between two users' processes, all packets travel across the network via the same virtual circuit. Although this is not the most efficient transfer mechanism for a dynamic network, it is the simplest method of guaranteeing that packets are delivered in the same order as they were received by the network.

Networks using this packet transfer mechanism are called *connection oriented* and because all presently offered public PSDN operate in this way, this is the type of network described here. Some private PSDN and most local area networks (LANs) use an alternative mechanism where data is transferred as one or more independently routed packets called *Datagrams*. Such networks are called *connectionless*.

When a packet is transmitted to the packet switch, it is transmitted at the speed of the calling user's Dataline. At the local packet switch, the packet is stored briefly, then transmitted across the network, possibly via a transit PSE. On reaching the destination PSE, the packet is again stored briefly, then transmitted to the called packet terminal at the speed of the called user's Dataline. What this amounts to is that all users of a PSDN can intercommunicate regardless of their respective Dataline speeds. Speed matching between users is one of the most useful characteristics of a PSDN, so that, for example, a user with a 48 000 bit/s Dataline can exchange packets with a user with a 2400 bit/s Dataline, or even with a 110 bit/s asynchronous terminal on the PSTN via the PAD, with the average data transfer rate dictated by the lowest speed terminal.

When a user with a 48 000 bit/s Dataline transmits a packet of 128 octets into the network it takes approximately 20 ms. If the called user's Dataline operates at 2400 bit/s, the packet takes approximately 400 ms to be received. Thus a PSDN is vulnerable to congestion caused because it may have to accept packets faster than they can be delivered. An extreme consequence of this would be a PSDN running out of buffers and losing user data. There are two mechanisms designed to ease this situation: logical channels, and flow control.

Logical Channels

Logical channels are subscribed to by users and dictate the maximum number of simultaneous calls that can be set up via that particular Dataline. Logical channels have no physical existence, they are logical entities derived from the Dataline by dynamic multiplexing. Fig. 12 shows one way of illustrating this. For instance, if the user in Fig. 11 has subscribed to only one logical channel then a process, in order to communicate, would have to seize the logical channel, which could not then be seized by any other process. At the end of the communication, that is, when one or more packets have been sent and/or received, the logical channel would be released so becoming immediately available for another process to seize. Thus, with just one logical channel, the full power of the packet-mode of working, the dynamic multiplexing of packets to give the impression of exclusive use of the Dataline to a number of user's processes, cannot be achieved.

If, on the other hand, the user in Fig. 11 has subscribed to 3 logical channels, then any process could seize the logical channel and use the Dataline until all three logical channels were engaged. From this point in time the Dataline is busy, by definition, regardless of the actual activity on the Dataline, until a logical channel is released by one of the processes.

There is virtually no restriction on the number of logical channels a user may subscribe to, but, as each must be supported by expensive resources in both the network and the user's packet terminal, it pays users to keep their demands modest.

There are four types of logical channel and a user sub-

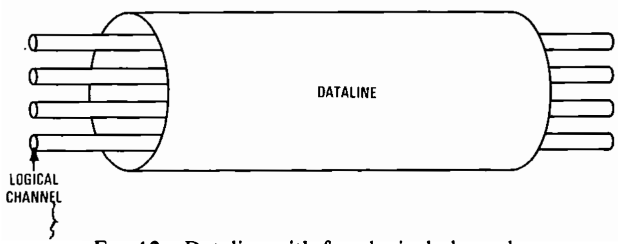


FIG. 12—Dataline with four logical channels

scribes to as many of each as required. One type does not involve switching at all, but supports only permanent virtual circuits (PVCs), a packet network's version of a leased line. The other three types of logical channel support switched virtual circuits (SVCs), and, although all three types offer full duplex transmission, one type can be seized only at the user end of the Dataline (outgoing-only SVC), one type can be seized only at the exchange end (incoming-only SVC), and the final type can be seized at either end (bothway SVC). With the proviso that the user must have at least one logical channel in order to communicate at all, and more than one to take advantage of the simultaneous call capability of a Dataline, the minimum number of logical channels a user must have of each type is zero.

Just as use of a PSDN is organised and disciplined, with logical channels giving the user a framework within which to operate, so the use of each logical channel, whether used for PVC or SVC, must be disciplined and organised with flow control.

Flow Control

When a logical channel is seized and a call established to a remote terminal, the transmitter has authority to transmit a maximum of two packets, then the distant receiver takes control by authorising the transmitter to transmit more packets or not as the case may be. This authority takes the form of acknowledgement of packets received. After the initial two packets, the transmitter is authorised to transmit only as many packets as are acknowledged by the receiver, which will range from zero to a maximum of two. Fig. 13 shows flow control working on logical channel 1 (LC1) as an example. The packets are received at the speed of the Dataline and the average data rate on LC1 can approach the Dataline speed provided that

(*a*) packets are acknowledged promptly, and
(*b*) the other logical channels are inactive.

The average data rate on LC1 can be reduced without limit by the receiver deliberately acknowledging less promptly, or as a result of other logical channels becoming active. These other logical channels contend for the Dataline bandwidth as their associated packets are interleaved with those of LC1. Packets associated with logical channels other than LC1 are not shown on Fig. 13 for the sake of clarity.

The principle of storing packets momentarily within the network, so bringing the network fractionally out of real-time operation, has other advantages apart from making speed matching possible—error protection of user's data for instance. A packet is not erased from a transmit buffer until it has been positively acknowledged as having been received error free by the distant receiver so that, if necessary, it can be retransmitted. Thus, the error rates of the individual transmission links can be hidden from users by this automatic link-by-link error protection.

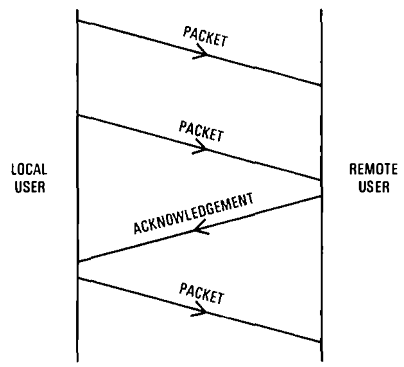


FIG. 13—Flow control on a logical channel

Another difference between a packet-switched and circuit-switched network is the method by which users are charged. Because circuits in the normal sense do not exist in a PSDN, charging is based on time and the volume of data transferred. This is quite different to charging on a circuit-switched network, which is based on time and distance.

Packet Assembler/Disassembler (PAD)

Traditionally, database access has been by non-intelligent terminals via the PSTN. This has been very convenient for users of databases, but has meant that the database itself had to be connected to the PSTN by a large number of lines to permit simultaneous access. Packet switching, as explained, permits simultaneous working on a single Dataline. However, as was realised back in the 1970s, when the access interfaces were being standardised, the database operator would be interested only in the reduction in line and modem costs that a move over to a PSDN would offer if the traditional customers of the database were unaffected. In other words, it was necessary right from the start for PSTN-based asynchronous terminals to be able to interwork with packet-mode terminals on the PSDN. The PAD and the CCITT Triple-X standards permit this, as shown in Fig. 10. The asynchronous terminal accesses the PAD via the PSTN and then uses the procedures laid down in CCITT Recommendation X28 to instruct the PAD to act as a packet-mode terminal on its behalf. The call then proceeds, asynchronously over the PSTN between the PAD and the character-mode terminal, and in the packet mode over the PSDN between the PAD and the database. As far as the database is concerned, this particular call occupies just one of the logical channels of the Dataline and so many such connections, from other PSTN-based terminals, can be supported simultaneously. As far as the database user is concerned, apart from the initial additional X28 protocol to instruct the PAD to set up the virtual circuit, the procedure is the same as when the connection was via the PSTN all the way. The PAD offers a range of facilities to both the asynchronous and packet terminals as laid down in the CCITT Recommendation X3, some of which are mentioned in the section on value-added services below.

THE INTEGRATED SERVICES DIGITAL NETWORK

The integrated services digital network (ISDN) consists of two components: an integrated digital network (IDN) and integrated digital access (IDA). The IDN consists of digital switching nodes interconnected by digital links; for example, System X exchanges and 2048 kbit/s digital transmission systems. IDA will consist of digital local line systems synchronised to the IDN. It must be stressed that, for data, both components of the ISDN are equally important. The IDN provides fast connection set-up and IDA provides the user with a digital interface, so removing the need for modems.

There will be no advantage in dedicating such a digital network to any particular service such as data or telephony because, when reduced to a bit stream, data and digitised telephony are the same. In order to emphasise this point of non-dedication, the all-digital network is being referred to as the ISDN with the stress on integrated services. A pilot service is being offered to some business users in London in the autumn of this year. The pilot ISDN is not an experimental or a field trial system, but the start of a new national digital network. The ISDN, while not dedicated to any particular service, is certainly being designed with the needs of telephony in the forefront. This is not surprising as the demand for non-voice services, despite continued rapid growth, is still small compared with the demand for telephony.

This means that the ISDN offer only circuit-switched connections because, at the present time, a standard packet-switching protocol suitable for digitised speech does not exist. The data packet-switching protocol X25 is unsuitable for telephony for two main reasons. Firstly, the volume of traffic makes impossible demands on network storage and, secondly, packets even between the same 2 terminals are delayed by different amounts. While this is of no consequence to data, it causes intolerable distortion to speech.

However, users of the ISDN will have IDA, which in fact will be a digital link to the local System X exchange. And via IDA, users who want something other than high-quality telephony or circuit-switched data will be able to use IDA to access other networks and services. Interworking to and from terminals on the analogue PSTN, Telex, and packet-switched networks will be available to the pilot ISDN.

VALUE-ADDED NETWORK SERVICES

Most established networks do more than provide facilities for intercommunication. The PSTN, for instance, provides its users with such an extensive range of information and other services that it must be possible to justify a connection for just these alone in certain circumstances.

Value-added network services (VANS) are those services provided by a network over and above the basic intercommunication facility.

Many of these involve changes to users' data in its transit of the network. This type of service can be offered only by a PSDN because the real-time nature of the data-transfer phase of a circuit-switched network precludes in-built services that add to or subtract from users' data. The PAD facility of a PSDN provides some simple VANS (compared to those that will be offered in the future, for example, messaging). As an example, the PAD permits a database to transfer bulk data to a simple asynchronous terminal on the PSTN without needing to know the physical characteristics of the terminal. The PAD is aware of the physical characteristics of the terminals that it supports and is able to insert format effectors at the appropriate intervals, if required, to the stream of IA5 characters received from the remote database. For instance, if the PSTN terminal, a VDU for example, has a line length of 40 characters, the PAD inserts IA5 *carriage return* and *line feed* characters after every fortieth character received from the remote database. Additionally, if the PSTN terminal is an electromechanical printer, then a number of *null* characters are inserted after the *carriage return* and *line feed* to allow time for the mechanism to operate before restarting the character stream. For the VDU, the *null* characters would be unnecessary and be omitted.

When the PSS opened, it offered service only to those terminals able to operate either the X25 (packet-mode) or X28 (character-mode) protocols. It also offered a PAD facility to allow packet and character terminals to interwork. Now, some years later the situation has changed, PSS has an established and growing customer base of sufficient size to make it economic to introduce new facilities, for example, mailbox and interworking with other networks etc. In addition, it is recognised that the growth of the PSS customer base is only part of the increase of the data terminal population generally and that the time is now right to offer some of these other classes of data terminal service on PSS. These extra services and facilities will be VANS. The network will be provided with extra intelligence in order to enhance the service to existing users and to provide service to some new classes of user.

COMMUNICATIONS MODEL

At its simplest, the function of a data network is to transfer an integral number of octets from one geographical location to another. This is exactly analogous to a telephone network transferring the voice of the user. However, there is more to communication than this because, unless the receiving-end user understands the received octets or voice, only partial communication is achieved. Thus, communication is not a simple concept which either exists or not, but a more complex concept built up of separate parts, all of which have to be present before communication can be said to be complete.

This picture of communication being built up of separate parts has proved so helpful that a standard model has been proposed and widely accepted; this is the ISO/CCITT Reference Model of Open Systems Interconnection, which has evolved over the past 10 years or so. The Reference Model pictures communication between users end processes as being in seven independent separate stages called layers, each layer using the services of the next lower level. For instance, layer 1, called the *Physical* layer, uses some physical medium, for example, copper, fibre, or space to transfer a bit stream from one geographical location to another. Layer 2, the *Data-Link* layer, uses layer 1 to transfer blocks of contiguous octets without error. The third layer, called the *Network* layer, with the help of the Data-Link layer or several Data-Link layers in tandem, routes the data blocks through the network. After layer 3, the data has been transferred between the users, and so layers 4–7 are located in the users end systems and are concerned with quality of service, management of the data transfer between the two ends, resource optimisation etc.

There is nothing fundamental about the fact that the Reference Model has seven independent layers. It is just that this is the way the architecture has evolved as the model has been refined over the years. The principle of layering is itself, however, is fundamental and has two main advantages. Firstly, communication between users' processes has proved to be too complex to be comprehended as a whole and so layering is, in fact, the division conceptually into smaller manageable parts. Secondly, each layer has been designed to be functionally independent so that any technological change affecting one layer need not affect any other layer.

SUMMARY

Data transmission is firmly established on the analogue PSTN, although for those users who require greater throughput and/or shorter connection set-up time, BT offers a dedicated PSDN with connections to similar networks overseas. In the future, and starting with the business community, the analogue PSTN will be converted to the ISDN, providing users with the ability to set up connections rapidly for high-quality speech, high-speed data or packet-switched data via the PSDN gateway.

Data volumes will continue to increase as the continuing trend to distributed processing places the emphasis on data communications. This in turn will be supported by in-network intelligence providing value-added services and format conversion.

Biographies

Phil Kelly, until his retirement, was Chief Engineer: Specialised Services, BT National Networks. His responsibilities covered the planning, development and operation of the national non-voice network services, including Telex as well as packet, circuit and leased line data services. He has represented BT at SG VII of the CCITT and at the CEPT Special Committee on Data Transmission (CSTD). He has an honours degree in electrical engineering, and is a member of the IEE and BCS. In March 1981 he was elected a Governor of the International Council for Computer Communications.

Mike Sands is a Head of Group in the National Packet Switching Service Division of BT National Networks Specialised Services. He was responsible for the original PSS User Guide and has been involved in CCITT activities. He is currently working in the VANS area. He is a member of the IEE.

Remote Monitoring of the Pressurised Cable Network

Part 1—Cable Flow Monitoring

J. T. SMITH, B.SC(ENG)., M.SC.†

UDC 621.315.211.4

This article, which is in two parts, briefly describes the remote monitoring of the pressurised cable network. Part 1 describes the monitoring of the air flow into cables. Part 2, which will appear in a later issue of the Journal, will describe the monitoring of the air pressure in the cables, and the cable sheath network surveillance system.

INTRODUCTION

Most local cables and main network and junction cables terminating at telephone exchanges and repeater stations are pressurised with dry air to a maximum of 620 mbar (gauge)[1]. This serves to retard the ingress of moisture and enables maintenance staff, using suitable monitoring devices, to be alerted when a sheath defect occurs[2]. At the present time in British Telecom (BT), these monitoring devices consist of contactors, pressure gauges and flowmeters. The contactors, or pressure switches, are fitted at points along the cable; the gauges and flowmeters are installed at the equipment cable pressurisation (ECP) rack[3]. Contact gauges are also installed at primary cross-connection points (PCCPs), more commonly referred to as *cabinets*, in the local network. The rate of air flow into each cable and the pressure readings at the ECP are noted daily and forwarded to the external plant maintenance control (EPMC). Any increase in flow or fall in pressure indicates a sheath defect, and corrective action is taken. These records give a continuous picture of the state of the cable-sheath network on pressurised cables. This article outlines a microprocessor-based method of remotely monitoring the air flows into cables.

CABLE FLOW MONITORING

In BT, the most commonly used type of flowmeter is designated the *Flowmeter No. 2* (see Fig. 1). These flowmeters are used in telephone exchanges and repeater stations to indicate the instantaneous rate of flow of dry air to the individual cables.

Some means of electronic surveillance was required to enable a remote monitoring system to be realised. A transducer was needed to convert the air flow into an electrical signal. The device had to be inexpensive and small enough to be installed easily onto the existing flowmeter. Accurate readings were not essential as long as the results were repeatable over a long period of time. The transducer chosen was the small bead negative-temperature-coefficient (NTC) thermistor.

THE FLOW TRANSDUCER

A thermistor is a device having a resistance that changes with temperature. The resistance of the NTC device decreases with an increase in temperature (see Fig. 2). When current is applied to the thermistor, Joule heating occurs, which raises the temperature of the thermistor bead above ambient temperature. The graph in Fig. 3 shows the logarithmic plot of static voltage across the thermister bead versus the current through the bead, together with the corresponding bead temperature. The graph shows that a current of 10 mA raises the bead's temperature to approximately 150°C, corresponding to a bead resistance of approximately 100 Ω.

To minimise the effects of ambient temperature on the flow metering, the thermistor is self-heated to operate at as high a temperature as possible without the device breaking down. The typical current passed in still air is 12 mA.

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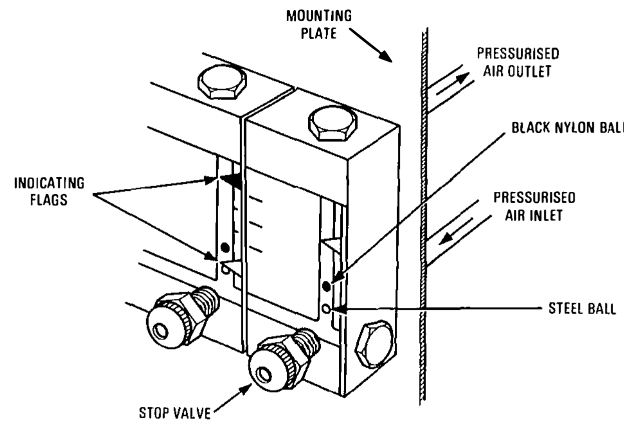
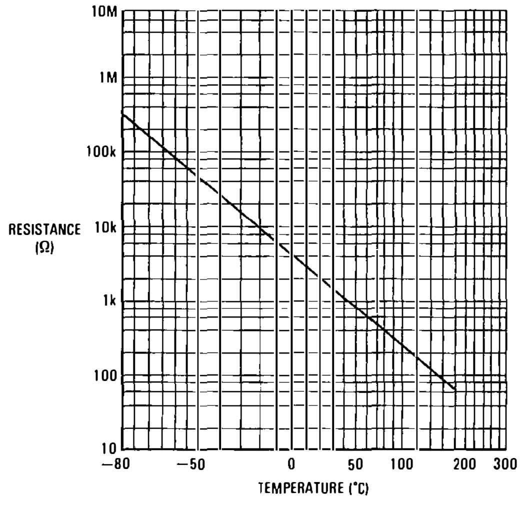


FIG. 1—Drawing of Flowmeter No. 2



Note: The temperature scale (shown in °C for convenience) is based on the reciprocal of the absolute temperature

FIG. 2—Typical thermistor resistance/temperature characteristic.
Diagram courtesy STC Components Ltd. (Thermistor Division)

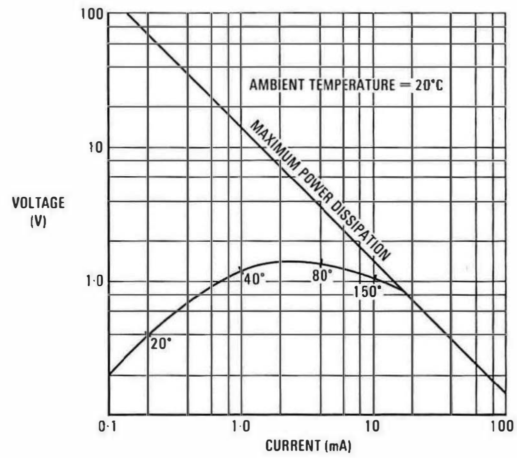


FIG. 3—Graph of voltage against current showing corresponding bead temperatures

However, temperature does have a small effect on the flow readings, and compensation is required; this is achieved by having a separate thermistor to monitor the pressurised air temperature. The monitoring thermistors are mounted on standard TO-5 transistor headers (see Fig. 4). Five such thermistors are mounted in a 5-way thermistor holder installed at the rear of the Flowmeter No. 2 (see Fig. 5).

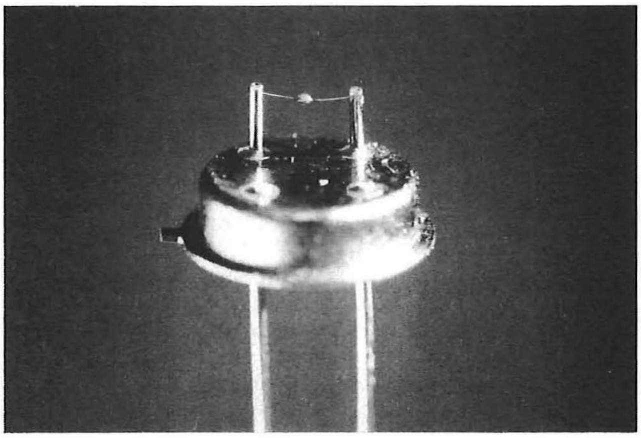


FIG. 4—Thermistor mounted on TO-5 header
Diagram courtesy STC Components Ltd. (Thermistor Division)

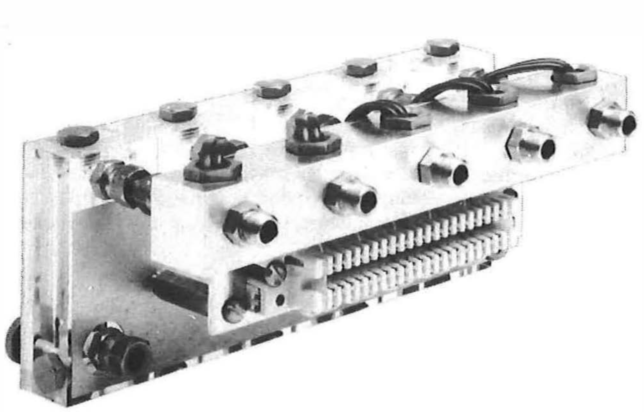


FIG. 5—Five-way thermistor holder installed at the rear of the Flowmeter No. 2

The tab on the TO-5 header acts as a keyway to ensure that the location of the thermistor is in the middle of the airway, and that the mounting pins cause the minimum of turbulence, or mask the air flow. Two neoprene washers give an airtight seal, and the sleeve nut allows the wires to be brought out and connected to a Strips Connection No. 237.

SIGNAL CONDITIONING CIRCUIT

Fig. 6 shows how the thermistor is connected in its signal conditioning circuit. It is based on a circuit for an anemometer application[4]. The thermistor R_t is connected in one leg of a bridge circuit. The current generator applies enough current to enable R_t to self heat until its resistance equals that of R_1. This is the balanced condition, and is monitored by the differential amplifier DA1, whose output regulates the current produced by the current generator. When air flows over the thermistor, heat is dissipated and the bead cools down and thus increases its resistance. This immediately causes the bridge circuit to become unbalanced. The unbalance is detected and the current generator feeds more current into the bridge. The thermistor self heats so that its resistance is reduced until the balanced condition is maintained. The increase in the thermistor current is monitored by the differential amplifier DA2. A typical curve of thermistor current versus flow rate is shown in Fig. 7.

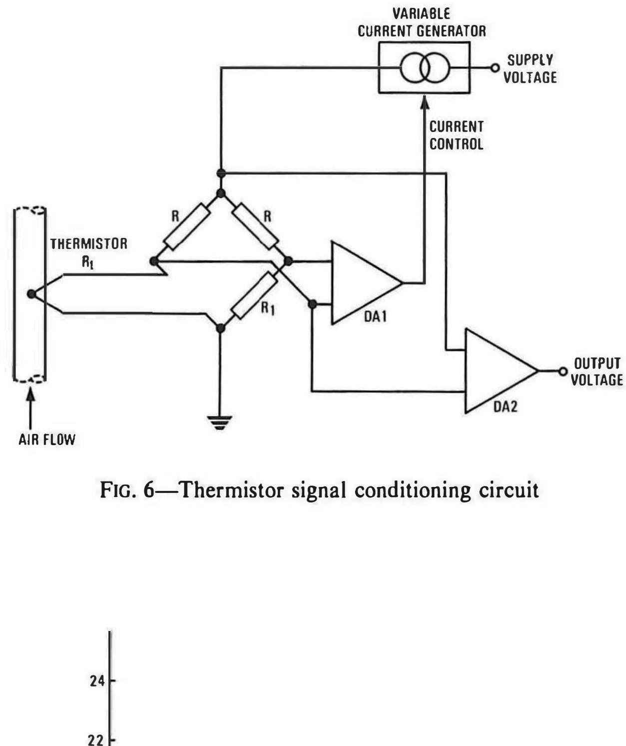
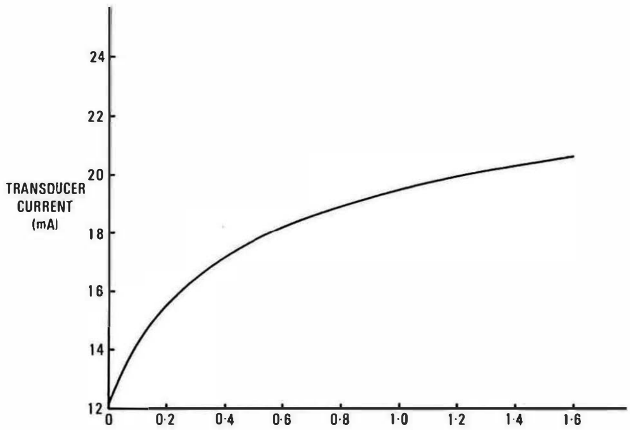
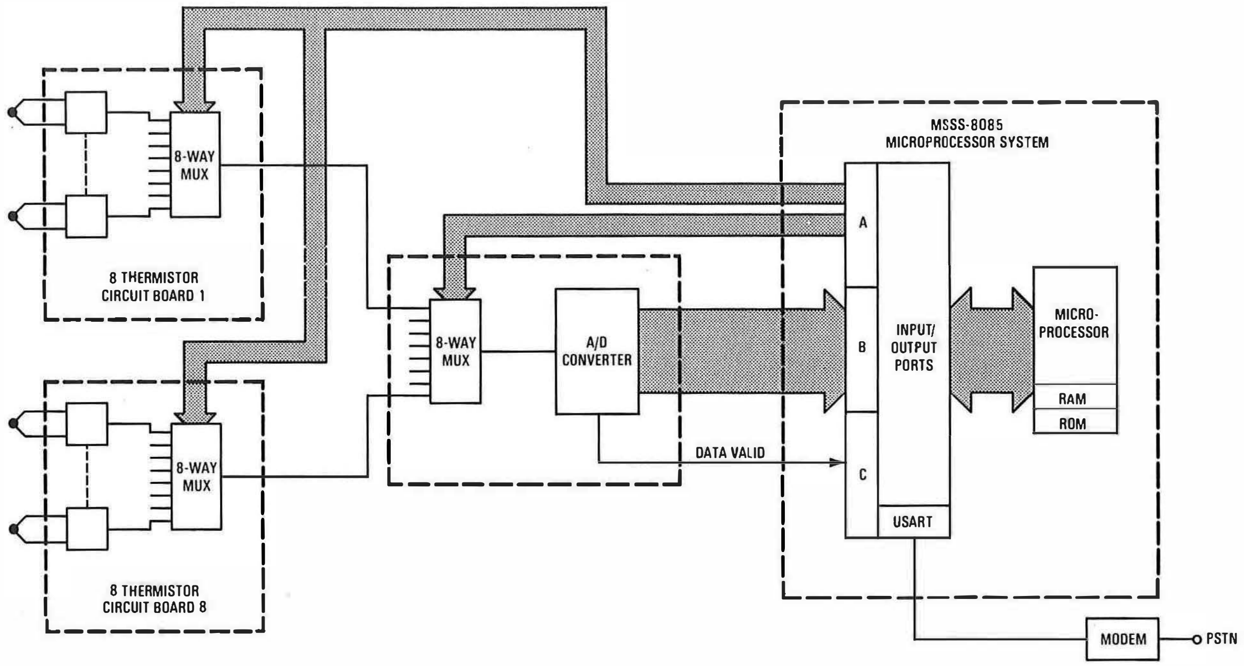


FIG. 6—Thermistor signal conditioning circuit



Note: 1·0 litre/min = 2·12 ft³/h

FIG. 7—Graph of transducer current against flow rate



MUX : Multiplexer
A/D : Analogue-to-digital converter
RAM : Random-access memory
ROM : Read-only memory
USART : Universal synchronous/asynchronous receiver transmitter
PSTN : Public switched telephone network

FIG. 8—Part of remote monitoring unit

In practice, there is a spread of results, the reason being the difficulty of manufacturing each bead to have the same surface area. However, accuracies of ±20% are tolerable because it is not the accuracy of the flow that is required, but the change in flow that must be monitored.

The graph is non-linear, but this does not present a problem because, if the analogue signal is converted to a digital word, a look-up table in the read-only memory (ROM) of the associated microprocessor system can be used to derive a flow value.

DATA ACQUISITION OF THE AIR FLOWS

The above has described a single thermistor circuit. Eight such circuits are mounted on a printed-wiring board.

Fig. 8 shows a block diagram of part of the remote monitoring unit (RMU). Each analogue signal is multiplexed before it is connected to the analogue-to-digital (A/D) converter. When the data is valid, the 8 bit word from the A/D is received by port B of the microprocessor system. Port A is used to address the multiplexers; the three least significant bits are used to address each 8-channel transducer card, while the next three bits are used to address the card multiplexer.

The microprocessor system used is the MSSS-8085 developed by the Microprocessor Systems Support Service at the BT Research Laboratories (BTRL). Once the RMU has received all the values of the air flows, it checks to see whether any have exceeded the alarm threshold. If this is the case, the details are passed to the EPMC by means of a modem link and an associated data terminal printer. The EPMC can also interrogate the RMU and ask for individual or all of the readings. Evaluation exercises have been carried out successfully at a number of exchanges to prove the reliability of the thermistor.

CONCLUSIONS

The thermistor has proved to be a reliable device when used in the field; accuracies of ±20% are good enough for the flow indications and trends required by the EPMC; it is not the actual value, but the change in value that is more important. This has resulted in a low-cost device that can be mounted easily at the rear of the existing flowmeter.

Part 2 of this article in a later issue of this *Journal* will explain the RMU in greater detail.

ACKNOWLEDGEMENTS

The author thanks the Director of Research, British Telecom, for permission to publish this article. He acknowledges the assistance of his colleagues at the External Plant Research Section, BTRL, the Area and Regional personnel who have helped in the evaluation exercises, and Mr. P. Camp of STC Electronics Components (Thermistor Division) for his help and advice.

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Biography

Jim Smith joined the Post Office in 1966 as a Trainee Technician (Apprentice). After a spell of maintenance duties in the Dundee Area, he won a minor/major award to study at Heriot-Watt University where he gained his degree in Electrical and Electronic Engineering. Regraded Executive Engineer in 1978, he joined the External Plant Research Section of BTRL.

Data Services and the ISDN

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UDC 621.395.34 : 621.394.4 : 681.327.8

Telecommunications in the UK will be transformed over the next few years by the widespread introduction of digital switches, interconnected by digital transmission systems. British Telecom is amongst the world leaders in recognising the potential for the extension of digital operation to the customer. This development provides a new range of services for data customers, both over the new network and by interworking with existing networks.

INTRODUCTION

The concept of an integrated services digital network (ISDN) has been aired extensively over the last decade as the next major revolution in telecommunications. But just what is an ISDN, when will one appear, and how will its introduction affect users and, in particular, the data communications user?

This article begins by looking at the basic philosophy behind the ISDN, and briefly outlines the plans for its introduction by British Telecom (BT) in the UK. It compares the service seen by the customer with those provided by existing networks, and shows how, during the evolution and growth of the ISDN, customers will benefit from an enhanced range of services provided in an economic manner. A future article will describe the technical developments behind the ISDN , and give more details regarding its early deployment.

Implementing an ISDN brings advantages to the network operator in terms of economies of scale and ease of administration and, in the long term, may cause the demise of separate networks (national private circuit digital network (NPCDN), packet switched network (PSS) and Telex). However, these networks will continue to exist alongside the ISDN for some time for a variety of reasons:

(*a*) to provide support to existing customers' terminals (for example, Telex),

(*b*) to provide specialised services in advance of their availability in an economic form on the ISDN,

(*c*) to provide service in those areas where they are required before the provision of ISDN exchanges (which have a planned provisioning period extending into the next century), and

(*d*) to support services to dedicated networks in other countries.

The inter-relationship of the ISDN with dedicated networks poses challenging questions both on commercial issues, with the need to develop consistent marketing and pricing policies, and on technical issues, with important interworking requirements to be met to ensure the widest possible communicability between users on the different networks. A major part of this article is devoted to this question of interworking between the ISDN and existing networks.

WHAT IS THE ISDN?

An important distinction must be drawn between two basic forms of integration: integration as seen by the customer, and integration in the network.

As far as the customer is concerned, integration implies the use of a common interface for all services, over which a unified set of control procedures can be operated to obtain a coherent set of basic and supplementary services. The customer does not need to know, and may well not want to know, whether the services are being supported by common plant in the network or by a range of independent networks.

As far as network integration is concerned, the emphasis is placed on the use of common equipment to control, switch, and transmit as many services as possible. With telephony services generating the major proportion of network traffic both now and in the foreseeable future, it is obvious that the network must be optimised largely for telephony. In the past, the nature of the telephony network, with its slow call set-up times, low bandwidth, and primitive signalling systems, presented a major stumbling block to service integration. However, these drawbacks are overcome with the change-over to a digital network for telephony, with sophisticated control processors and inter-exchange signalling system. The extension of digital transmission and sophisticated signalling to the customer's premises provides many of the features required by data users, a typical example being the improvement obtained for circuit-switched data on changing from the analogue network to the ISDN, as shown in Table 1.

TABLE 1
Comparison of Data Service Characteristics on Analogue and ISDN Networks

Feature	Analogue Network	ISDN
Maximum data rate	2·4 kbit/s duplex	64 kbit/s duplex
Call set-up time	5–25 s	1–2 s
Error rate	1 in 10^3 to 10^4	1 in 10^5 to 10^6
Signalling	Slow and primitive	Fast and advanced
Supplementary services	None	Many

IMPLEMENTATION PLANS

In autumn 1984, the first local exchange developed under the System X system enhancement program (SEP) will be brought into service in the heart of London. ISDN features are inherent in the design of this exchange, and have been developed by using the standard System X design and testing arrangements. It was these features that supported the impressive displays of photo-videotex (Picture Prestel) and slow-scan television, which captured the interest of many visitors to the British stand at the World Telecommunications Exhibition in Geneva in October 1983, where a wide range of advanced services were demonstrated supported by an on-site System X concentrator unit.

† Local Network Strategy Department, British Telecom Local Communications Services

The planned build up of SEP local exchanges is extremely rapid, deployment being aimed primarily at business areas and exchanges provided either as overlay units or as replacements for existing exchanges. By the end of the decade, it is expected that over 2000 local exchange units with ISDN capability will be in service.

Although ISDN is inherent in the design of System X, BT is proposing to offer ISDN initially as a pilot service only. The pilot service will be based on four exchanges, with remote multiplexors used to spread the service via out-of-area connections to around 30 major business centres. During this period the pilot service will be closely monitored to ascertain customer reaction to the new service; marketing and operational support aspects will also be reviewed.

Impact of International Standards

The implementation of ISDN features within System X has taken place during a period of intense activity within the worldwide and European telecommunications standards bodies (CCITT† and CEPT*, respectively). A primary study area has been the customer–exchange signalling system, with specifications based on the layered approach outlined by the layered model for Open Systems Interconnection (OSI)[1]. Recommendations for layers 1, 2 and 3 (Physical, Link and Network, respectively) are now emerging in the CCITT I-series Recommendations, but even so there is still some way to go before exchange development can begin, and it is therefore unlikely that the new standards will find widespread support in the network before the end of this decade.

Meanwhile, in order to introduce the ISDN in advance of equipment available to international standards, BT has developed its own customer–exchange signalling system (DASS). In order to shield terminal manufacturers from implementing this standard, BT provides network terminating equipment (NTE) for basic or single-line connections, which present one or more standard X- and V-series interfaces to the customer, as depicted in Fig. 1, and supports DASS to the network.

In this way, BT could, if it wished, replace the NTE with one supporting I-series signalling standards when these are introduced into the network, without any effect on the user. However, there is no reason why DASS and I-series customer–exchange signalling systems cannot co-exist in the network, since they both apply specifically to a single customer–exchange link. It is likely, therefore, that DASS will remain in use until well into the 1990s.

The emerging CCITT standard for a basic ISDN line, specifies the following channel structure:

2 × 64 kbit/s B-channels for voice or non-voice use,
1 × 16 kbit/s D-channel supporting signalling, packet data and telemetry on a shared basis.

TOTAL 144 kbit/s.

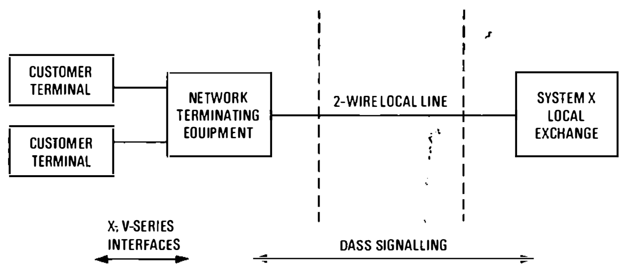


FIG. 1—Role of NTE in decoupling DASS signalling and international standard interfaces

In the early years of the ISDN in the UK, a subset of this structure will be provided as follows:

1 × 64 kbit/s B-channel for voice or non-voice use,
1 × 8 kbit/s B′-channel for non-voice use,
1 × 8 kbit/s D-channel for signalling use.

TOTAL 80 kbit/s.

Methods of rate adaption of user data rates onto the B- and B′-channels have been chosen so that non-voice calls at rates up to 8 kbit/s can be supported by an 8 kbit/s B′-channel at one end of the connection, and a 64 kbit/s B-channel at the other end.

A move to the 144 kbit/s international standard will be considered in due course but, as with DASS and I-series signalling, there are no problems in supporting 80 kbit/s and 144 kbit/s customer lines simultaneously in the network.

CIRCUIT-SWITCHED DATA AND THE ISDN

The introduction of the ISDN is particularly timely in view of the demand for increased data rates, which are pushing customers towards more complex and expensive modems. It also opens the door for the introduction of new services such as high-data-rate photo-videotex (Picture Prestel), fast facsimile, and slow-scan television, which were impracticable with the limitations of the analogue network. While ISDN seems certain to attract customers for high-speed-data applications, it remains a matter of speculation as to what extent low-speed-data users will migrate from Datel, since this will be determined largely by comparative costs. The difference in network costs between the two methods is best summed up as shown in Figs. 2 and 3.

With the modem, analogue-to-digital (A/D) converter, NTE and digital transmission system functions rapidly becoming realised in a handful of integrated circuits, it is probable that ISDN, with its additional range of features, will become attractive at lower and lower data rates.

Pilot Services

For the pilot service, synchronous circuit-switched data at the CCITT recommended rates of 2·4, 4·8, 9·6, 48 and 64 kbit/s (User Classes of Service 4–7 and 30) will be supported, plus the non-standard rate of 8 kbit/s, which makes optimum use of the B′-channel. Synchronous data terminals working at these rates fall into two main categories:

(*a*) Terminals that rely on the user to set up the call from

†CCITT—International Telegraph and Telephone Consultative Committee

*CEPT—Conference of European Postal and Telecommunications Authorities

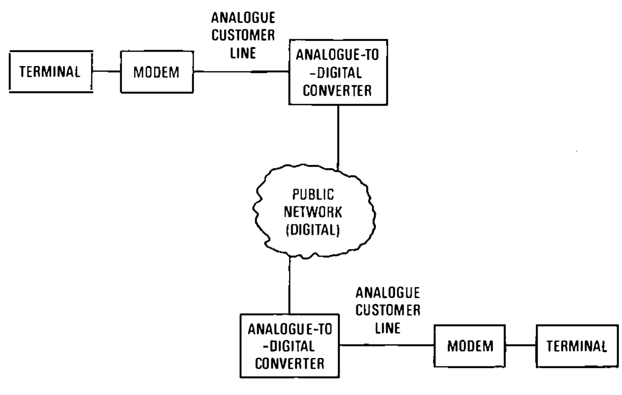


FIG. 2—Elements of a Datel connection

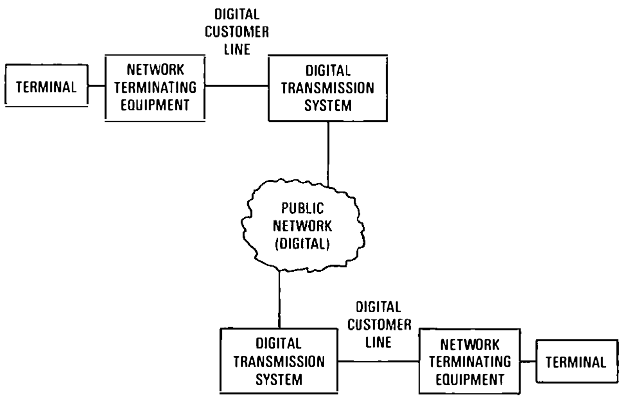


FIG. 3—Elements of an ISDN connection

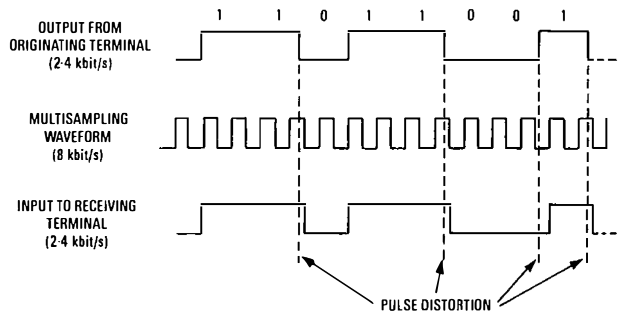


FIG. 4—Effect of multisampling method of rate adaption for asynchronous terminals

the NTE, in much the same way as existing Datel calls must be set up by using a telephone. These terminals use fairly simple call set-up procedures, employing a multitude of control wires across the terminal–NTE interface, with changes in logic levels on the wires to indicate various stages of call progress. Interfaces of this type supported by the ISDN conform to CCITT Recommendation X21*bis* and can have a variety of electrical and physical characteristics[2].

(*b*) Terminals that can instigate call set-up themselves, either by activating a direct-call facility in the NTE, or by providing character-coded call set-up information to the NTE. These more advanced terminals use just four main control wires across the terminal–NTE interface. Interfaces of this type supported by the ISDN conform to CCITT Recommendations X21 and X21 (leased line).

The functions of the NTE involved in setting up circuit-switched data calls are as follows:

(*a*) provision of manual call set-up facilities from the NTE,

(*b*) signalling conversion between in-band or multi-wire terminal interface signalling standards and the out-band DASS on the network side,

(*c*) bit-rate adaption between the user data rate and the channel rate of 8 or 64 kbit/s, and

(*d*) support of ready-for-data alignment procedures to establish when an end-to-end path between terminals is available, so that terminals can be advised when to start transmitting data.

The detailed support of these functions for synchronous terminals has been extensively described elsewhere.[3]

Although synchronous operation is optimum for high data rates, there are many asynchronous terminals available for lower rates. These terminals are supported on the ISDN by using manual call set-up from the NTE, and a rate-adaption scheme termed *multisampling*, which aligns with CCITT Recommendation X52[4]. The effect of multisampling is to introduce pulse distortion into the data stream, as depicted in Fig. 4, and this restricts supportable data rates to 1·2 kbit/s on calls using a B′-channel and 9·6 kbit/s on a B channel. With the large proportion of asynchronous terminals being less than 9·6 kbit/s, these restrictions are not serious, and the method has the advantage of being extremely simple to implement in an NTE.

Call Control and Supplementary Services

The ISDN provides several features specifically for non-voice calls:

(*a*) Messages are provided instead of tones and recorded announcements, to indicate progress of a call. These messages are mapped onto the appropriate X21 indications by the NTE, and also contain text for display to give call-progress guidance on calls being set up manually from the NTE. A wide range of text messages has been incorporated to ensure that non-voice calls can be set up easily. Examples are: NUMBER BUSY and USER HAS CHANGED NUMBER.

(*b*) A guaranteed digital path is provided end-to-end, with protection from operator intrusion. Because BT's policy for the introduction of System X is to install an overlay network rather than isolated pockets of new exchanges, it should, under normal circumstances, always be possible to obtain a digital routeing between System X local exchanges. ISDN call control specifically ensures that such a path is chosen for non-voice calls, protecting the user from abortive calls through the analogue network or to an analogue (telephony) customer. In addition to selection of a digital route, call control also ensures that operators cannot break-in on non-voice calls, which would cause gross errors to be introduced into the data stream.

(*c*) A closed user group (CUG) service is provided. Many data users, especially the financial institutions, require security mechanisms to restrict access (for example, to databases) to authorised users only. The ISDN provides a network-based restriction service based on the CCITT Recommended CUG Service X87[5]. The operation of the service is illustrated in Fig. 5. When the caller requires a CUG call, he indicates this by including the appropriate supplementary service code in the call request.

The network holds a *key* that uniquely identifies the CUG. This key is passed across the network in the form of a unique code, and the destination System X exchange checks that the key fits the *lock* of the called customer. If it does, the call is offered to the called NTE. Various options are available as exchange-held parameters at both originating and terminating ends of the call that determine the action

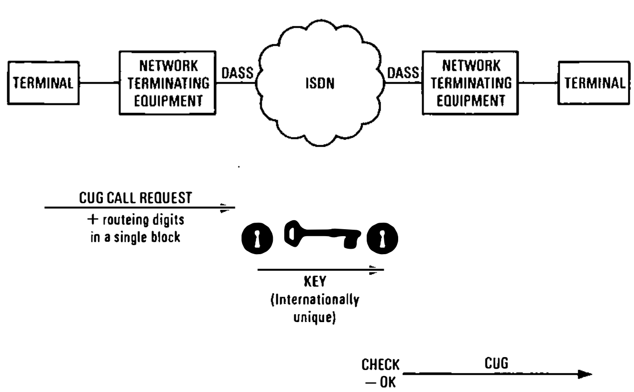


FIG. 5—Closed user group (CUG) call

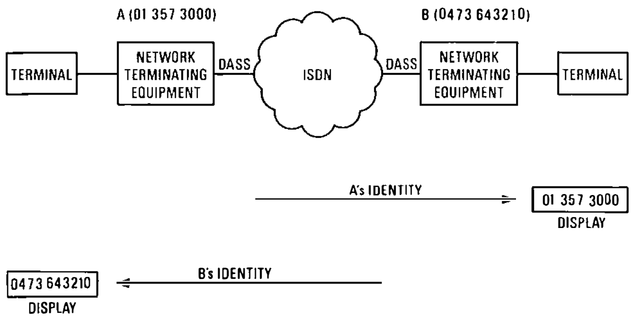


FIG. 6—Calling and called line identification

to be taken on calls between customers who are not in the same CUG.

Members of a CUG can be geographically anywhere on the ISDN and there is no limit on the number who can belong to a given group. BT control the registration of the groups, and customers never know the keys used. In the pilot ISDN, each customer can have up to ten different keys allocated. These are selected by means of a single digit sent at the start of a CUG call.

(*d*) Calling and called line identification is provided. This service is illustrated in Fig. 6, and enables the identity of the caller, A (in the form of a national number), to be given to the called customer, B, and the identity of B to be given to the calling customer. A customer may, if he wishes, request to be ex-directory, in which case, the network withholds his number from the other party on a call.

In addition to the features described above, which are provided specifically for the data user, several of the standard telephony supplementary services will be available, with appropriate variations for the specialist requirements of the data user. In the pilot service these are:

(*a*) abbreviated dialling,
(*b*) repeat last call,
(*c*) fixed destination call, and
(*d*) call barring.

A wide range of further services is under review for later implementation.

Extension of Circuit-Switched Service to KiloStream-Based Customers

The KiloStream network supports data services conforming to CCITT User Classes of Service 4–7, namely 2·4, 4·8, 9·6 and 48 kbit/s, but actually transports customer data in the form of 6+2 envelopes, with two network bits—alignment (A) and status (S) added to every 6 bits of data. The setting of the S-bit, in conjunction with the data bits, has been designed to meet the requirements of X21 terminals, and allows call-control information to be passed across the KiloStream network. Fig. 7 shows the interworking arrangements provided between ISDN and KiloStream for this service. The KiloStream network, in effect, provides a line extender for the normal terminal-to-NTE link, and the KiloStream-based customer has an ISDN number and access to the normal range of data supplementary services. The linker function shown in Fig. 7 is required to provide buffering between the two networks which, although frequency locked to the same national reference timing source, will exhibit phase drift relative to each other because of the operation of their respective clock control mechanisms.

LEASED-CIRCUIT DATA AND THE ISDN

One of the features of System X is that private circuits can be set up through the exchange. These new private circuits will be set up from the System X maintenance console and use paths through the normal route and concentrator switches of System X. The advantage of this approach is that private circuits can be set up, re-arranged, and removed readily; for this reason, they are described as *semi-permanent* in contrast to *permanent* hard-wired types. A further advantage is the continuous monitoring of the connection by the exchange, and this will result in an immediate fault report should failure of the private circuit be detected.

Private circuits set up across the ISDN employ 64 kbit/s paths across the network, and either the 8 or 64 kbit/s bearer channel to each user. Numerous applications can be expected, an obvious one being the use of a private circuit via the 8 kbit/s channel for on-line access to a database or booking system, with simultaneous use of the 64 kbit/s channel for switched telephony.

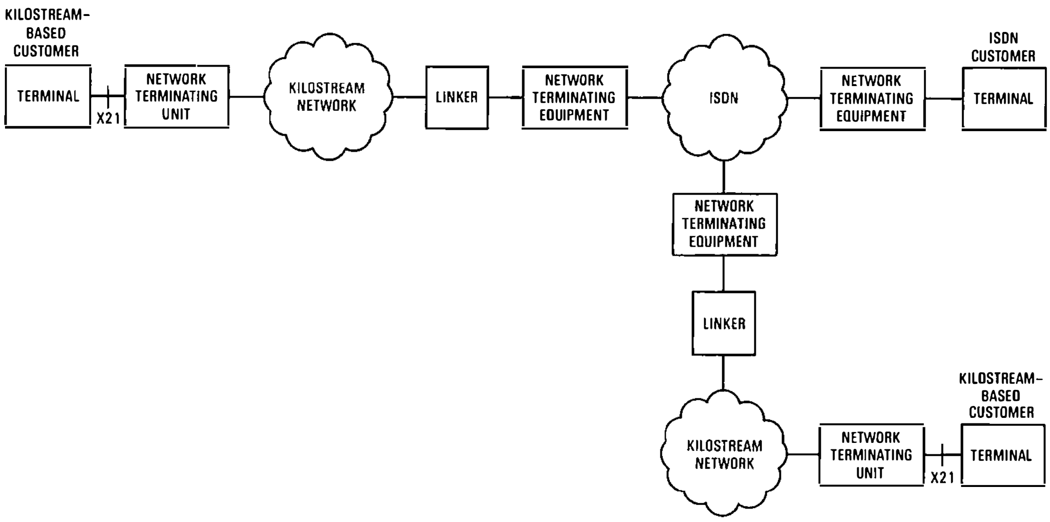


FIG. 7—Interworking of ISDN and KiloStream to provide circuit-switched service to KiloStream-based customers

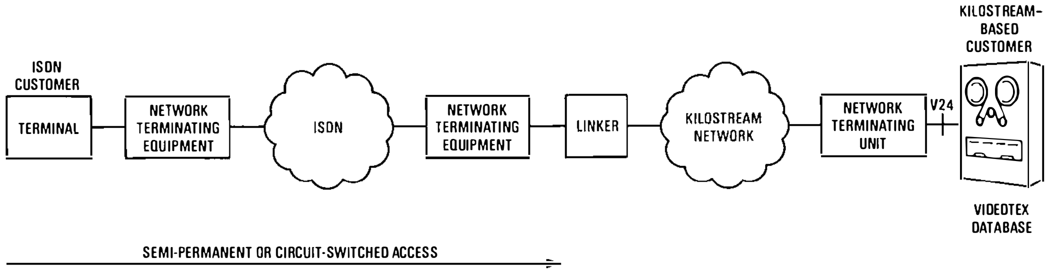


FIG. 8—ISDN–KiloStream interworking: KiloStream-based customer limited to leased-line working

ISDN–KiloStream Interworking

In order to connect leased paths between ISDN customers and KiloStream-based customers, an NTE plus linker combination is used in a similar manner to that described earlier for circuit-switched applications. Fig. 8 shows the resulting arrangement, with the KiloStream-based terminal permanently connected to the interworking NTE, and, in contrast to the circuit-switched case, unable to generate call set-up information into the ISDN. The data rates supportable in this case are 2·4, 4·8, 9·6, 48 and 64 kbit/s; the latter is available since X21-type status and alignment information is not essential for leased-line operation, and so the 6+2 envelope structure required in the circuit-switched case can be dispensed with.

The ISDN customer can choose to adopt a semi-permanent path across the ISDN, or employ circuit-switched access—in effect, a part-time private circuit. If the part-time service is selected, the customer may wish to protect the KiloStream-based terminal from unauthorised access by other ISDN customers, and can make use of the CUG service for this purpose.

PACKET-SWITCHED DATA AND THE ISDN

The growth of packet-switched data traffic will undoubtedly be affected by the introduction of the ISDN, offering 64 kbit/s data capability with low error rates. A continuing demand for packet switched service is nevertheless foreseen for many applications because of the unique features available, namely:

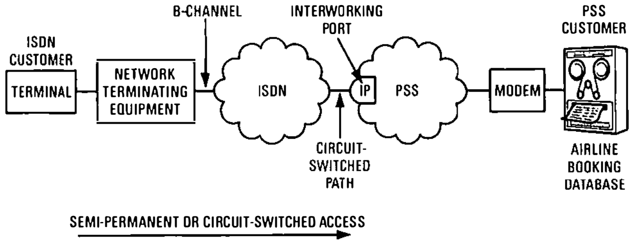
(*a*) the ability to interwork terminals at different rates,
(*b*) the support of a number of virtual calls on one physical access link (valuable for database applications),
(*c*) the use of widely accepted X25 standards,
(*d*) its efficiency for certain types of data traffic (for example, interactive database access), and
(*e*) its low cost on long-distance (for example, international) use, because of dynamic multiplexing onto network paths.

International Standards for ISDN–Packet Network Interworking

Study Group VII of CCITT has been actively studying the techniques for ISDN–packet network interworking for calls between synchronous terminals, in order to establish whether any requirements or restrictions are necessary to enable interworking to operate satisfactorily. The Study Group has formulated Recommendation X31[6] on this issue, outlining two scenarios: minimum and maximum integration.

Minimum Integration

The minimum-integration scenario is illustrated in Fig. 9. Packet calls are handled transparently through the ISDN, whose only function is to provide a transparent channel between the user's X25 terminal adapter and a port on a packet network. This access is possible solely over the circuit-switched B-/B′-channels.



Note: This diagram is a simplification of the X31 Recommendation, which contains greater detail and, in particular, splits the NTE into different functional blocks

FIG. 9—Minimum integration scenario for ISDN–PSS interworking

Where a semi-permanent circuit is used for access, the NTE performs rate adaption only. For switched access from the terminal, the NTE supports ISDN signalling procedures which are used to set up the link to the packet network prior to starting the X25 level 2 and level 3 functions. For calls to the terminal, an interface unit (IP) in the packet network sets up the physical link to the terminal before initiating the packet procedures.

Depending upon the form of access used, terminals have either a packet network number (semi-permanent circuit case), or an ISDN number (switched access case) taken from the normal telephony numbering range.

Maximum Integration

In the maximum-integration scenario, the packet-handling function is provided within the ISDN and all terminals have ISDN numbers. Two means of access are possible: via the circuit-switched B-/B′-channels, or via the D-channel.

Fig. 10 shows the circuit-switched access scenario, in which the packet calls are routed within the ISDN to a packet-handling function which processes the calls. This

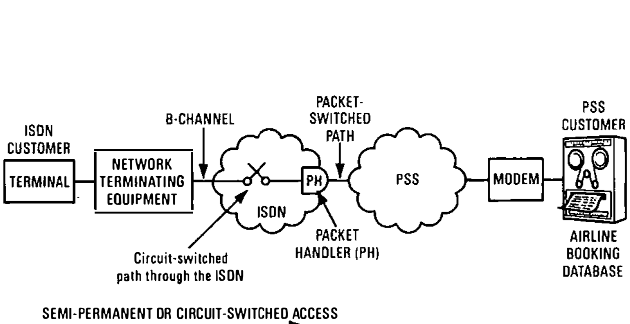


FIG. 10—Maximum integration scenario for ISDN–PSS interworking: B-channel access (simplified diagram)

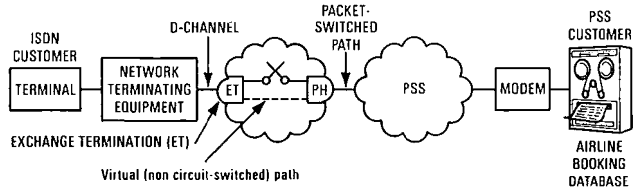


FIG. 11—Maximum integration scenario: D-channel access (simplified diagram)

scenario provides logical channel working whereby multiple calls can be supported simultaneously over a single access channel. The method of accessing the packet-handling function varies in ISDNs having different implementations, but in all implementations a circuit-switched connection is set up to a packet-handling port which must provide X25 functions for level 2 and 3, the path setting up functions for level 1, and any necessary rate adaption.

The scenario for D-channel operation of X25 packet level procedures is shown in Fig. 11. In this case, packets on the D-channel are forwarded to a packet-handling function in the ISDN which must support X25 level 3 procedures.

The network support of X25 level 2 can be either at the packet handler or at the exchange termination, the latter being a more likely choice. Termination of level 2 at the exchange termination allows a choice between continuing transmission of the X25 packets to the packet handler in packet mode by using a virtual path, or forwarding packets via a circuit-switched path, possibly sub-multiplexed with other customers' D-channel packet traffic.

The characteristics of the D-channel operation impose some limitations on the X25 support that is possible by using this access. Only packet terminals of user classes 8 to 10 (2·4 kbit/s to 9·6 kbit/s) can use the D-channel, they must observe a maximum user data field length of 256 octets, and conform to rules concerning multiplexing, the use of numbered information frames and compatibility checking as detailed in Recommendation X31.

Packet Services for UK Pilot ISDN

In the UK, for the initial phase of the ISDN, techniques conforming to the CCITT minimum integration scenario are being adopted. Two methods of interworking between ISDN and PSS for calls from synchronous terminals are being provided, and are illustrated in Fig. 12.

Dial-up Access to PSS via a Packet Network Adapter

The dial-up access service to the PSS utilises a number of dedicated ISDN ports provided on the packet network adapter (PNA), installed principally as a PSS/PSTN gateway for Teletex. The PNA is accessed across the ISDN by using a circuit-switched call, and set-up through the PSS is then completed by using X25 set-up procedures in-band between the terminal and the PNA. Either the 8 kbit/s or 64 kbit/s channel may be used on an NTE, but the user data rate chosen must be compatible with the PNA port. Using this mode of access, a user will be charged ISDN and PSS call charges, plus a PNA charge.

Dial-up/Private Circuit Service via Dedicated NTEs

For customers with a high usage of the PSS, it will be advantageous to BT (by reducing the loading on the PNA) and the user (by avoiding PNA charges) to access the PSS directly, as an alternative to using the PNA.

This service uses between the ISDN and the PSS network an NTE that is dedicated to a particular customer, in order to provide point-of-entry information for charging purposes. Fraudulent use by other customers is barred, either by leased line access, or by utilising the CUG facility on dial-up access.

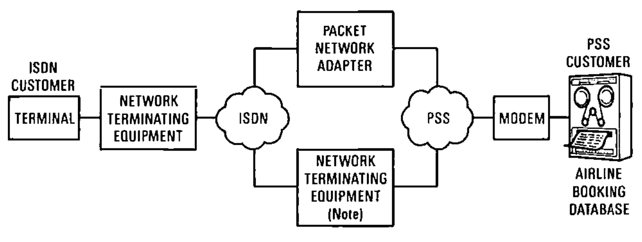
All PSS dataline rates, including 48 kbit/s, will be available by this method.

Asynchronous Terminal Access to Packet Service

Low-speed data users invariably use simple asynchronous terminals with conversion to X25 packet formats in a packet assembler/disassembler (PAD) normally accessed across the PSTN. The introduction of the ISDN retains the ability to access the PSS-based PADs via circuit-switched calls across the ISDN, and the same restrictions (low speed, outgoing calls only) apply. Studies into the implementation of the maximum integration scenario (discussed earlier) indicate that provision of the PAD function in the exchange termination may be practicable in the future, in which case greater flexibility will be available.

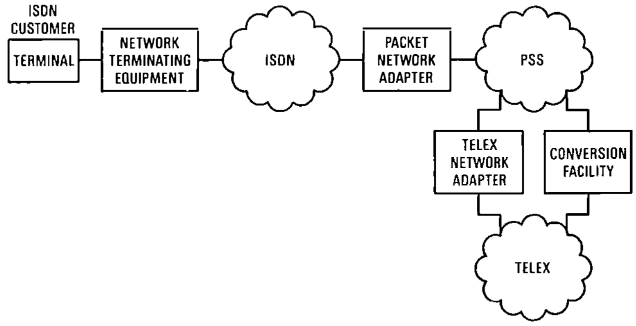
TELEX INTERWORKING

No specific interworking arrangements are being provided for ISDN customers to interwork with Telex customers. However, the two interworking units being provided between the PSS and Telex networks can be accessed via the PSS interworking arrangements, and thus give full communicability between ISDN and Telex customers. The arrangement is shown in Fig. 13. The Telex network adapter (TNA) is provided for general Telex–PSS interworking[6]. The conversion facility (CF)[7] provides additional features specifically to support the new Teletex service. The conversion facility also provides interworking to the PSTN, and consideration will be given to extending this capability to the ISDN at a later date, to allow direct ISDN–Telex interworking for Teletex calls.



Note: Interworking NTE is dedicated to a particular customer

FIG. 12—UK pilot service ISDN–PSS interworking, showing minimum integration approach using PNA or dedicated NTE



Note: ISDN–PSS link may also be via dedicated NTE

FIG. 13—ISDN–Telex interworking

THE FUTURE

The facilities and interworking arrangements described here are just the start for ISDN. Being linked intrinsically with the evolution of a sophisticated and flexible network for telephony, new features and facilities can be introduced and immediately achieve network-wide support. It seems certain that, in future phases of development, additional features will be provided for data customers. These will provide additional facilities on calls within the ISDN, and more flexible and powerful interworking capabilities with dedicated networks.

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Biographies

John Wedlake is Head of Service Planning and Strategy Division in British Telecom's Local Network Strategy Department. He joined the Research Department at Dollis Hill after graduating from London University. Since 1971 he has been involved in network and service definition, initially for circuit and packet-switched data systems and, more recently, for the ISDN. He is currently Vice-Chairman of CCITT Study Group VII: Public Data Networks

Peter Lisle is Head of the Service Strategy Group in British Telecom's Local Network Strategy Department. He joined BT in 1972 as a sponsored university student and graduated from Cambridge University in 1976 with an honours degree in Electrical Sciences. He spent three years in the Systems Strategy Department working on performance aspects of digital networks, before returning to university to obtain an M.Sc. in Computer Science. In 1980, he joined the System X Development Department, where he led a joint BT/contractor team defining the system requirements for the first phase of the ISDN. In 1983, he moved to the Network Strategy Department, where he leads a group defining future service strategy, including the future strategy of the ISDN.

Book Reviews

Optical Communications: A Telecommunications Review. Siemens. John Wiley & Sons Ltd. 220 pp. 296 ills. £19·50.

This book is an English translation of the German original by Siemens' Aktiengesellschaft. If the title gives an impression of a special issue of a manufacturer's house magazine with a series of articles by invited specialists, then this is exactly what one gets. This approach has both strength and weakness, in that each chapter tends to be written as a self-contained article in which the author feels obliged to produce a stand-alone contribution. Unfortunately, therefore, it does not read as a particularly coherent textbook in the normal sense.

The coverage is comprehensive, with sections on basic principles, cable and components, systems, equipment, measuring and testing, quality assurance and future prospects. There is also a useful glossary on optical-fibre terminology.

The quality of presentation is excellent, with the colour photography and diagrams particularly worthy of high commendation. The text, mostly qualitative but adequate, makes the book more useful to the reader wishing to obtain an overview of optical-fibre technology and its applications rather than to the specialist in research and development.

Considering the source of the original publications, it is not surprising that the contents of the book are biased towards German activity in general and Siemens in particular; and to the UK reader this gives an interesting insight into the approach used elsewhere in Europe. Equally, the UK reader must not be too sensitive because he has to search hard to find any mention of optical-fibre activity in the UK. The map on page 108 and the article on projects in Europe completely omit any observation on British Telecom's (BT's) activities. The article on page 206, however, does recognise the UK's existence and Figure 2 shows BT as having been active in optical fibres since 1980. This concentration on German activities could be attributed to the book being a Siemens' in-house publication; conversely, activities in the USA get a chapter to themselves, posssibly because of the tie-up between Siemens' Corporation and Corning's Glassworks in the USA.

Overall, however, the style, presentation and comprehensive coverage of the technology makes this a useful volume for the general engineering reader.

R. D. MARTIN-ROYLE

Telecommunications Primer. Graham Langley. Pitman Books Ltd. viii+148 pp. 132 ills. £4·95.

The objective of this book is to provide newcomers to computing and telecommunications, and people in other disciplines, with a background knowledge of the terminology, technology and principles in this very wide field that can be absorbed quickly. As the title suggests, the book concentrates mainly on telecommunications and has very little to say on computing *per se.*

It is interesting to note that the author has not confined himself only to technical aspects but has included a section on maintenance and operation, covering centralised maintenance systems, telephone tariffs, system viability and forecasting future demands. The rest of the book covers basic principles of line, radio and switching systems; the decibel, waveforms and modulation; cable, radio and transmission, including satellites and mobile radio systems; television; digital services and fundamentals; electronic mail; teletext; viewdata; Prestel; cable television; and a brief look into the future.

The book is well written and presented, and provides a useful overview for those new to telecommunications, provided it is accepted that the wide range of topics covered in 148 pages means that the overall treatment is somewhat superficial. Further reading is recommended at the end of each chapter for those who wish to pursue any topic more thoroughly, and self-test questions are included at intervals during the book to test readers' understanding of terminology.

R. HARVEY

Open Systems Interconnection—An Introductory Guide

P. A. JENKINS, B.TECH. and K. G. KNIGHTSON, M.SC., B.SC., C.ENG., M.I.E.E., M.I.E.R.E.†

UDC 389.6(100) : 621.39

Open System Interconnection (OSI) is the name given to a collection of international standards that specify the means by which information can be exchanged and understood by dissimilar information processing systems. This article gives a brief introduction to the concepts of the hierarchical structure of the seven-layer Reference Model that is the basis of OSI.

INTRODUCTION

Imagine that any computer system could be bought from any manufacturer, connected to any network, and guaranteed to be able to communicate with any other manufacturer's computer system on that network or via any other network—this is the aim of Open Systems Interconnection (OSI) (see Fig. 1).

The key to compatibility is clearly the use of a set of standard data communication protocols by the communicating systems.

During the 1970s, it was becoming increasingly obvious that the plethora and diversity of emerging standards was not in the best interests of the users, and was inhibiting the growth potential of the communication industry. Hence the massive support for the OSI initiative from both large and small manufacturers alike.

What was needed was a framework for co-ordinating the development of data communication standards, and for placing existing standards in perspective. The complexity of computer systems and protocol requirements clearly necessitated some methodology for the organisation of the work into manageable parts. This overall framework is the international standard *IS 7498: Information Processing Systems—Open Systems Interconnection—Basic Reference Model*, published by the International Standards Organisation (ISO).

† Systems Evolution and Standards Department, British Telecom Development and Procurement

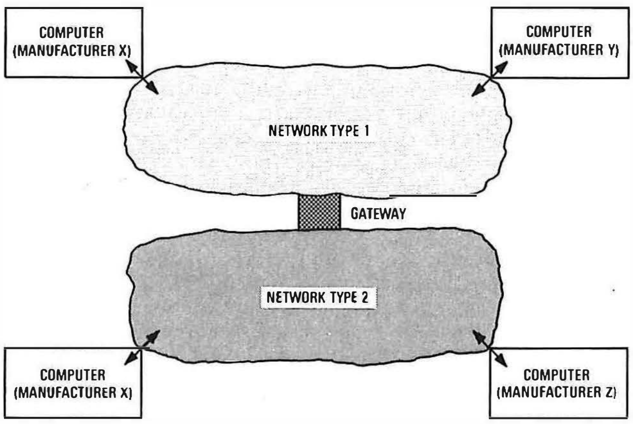


FIG. 1—The purpose of the Open Systems Interconnection

Work began on the OSI Reference Model in 1978 and since then there has been close collaboration between ISO and CCITT* in order that a single common model may be agreed for OSI for both ISO and CCITT defined services. Similar collaboration exists for the particular standards being developed within the framework of the model in areas of common interest jointly within ISO and CCITT.

In theory, subject to certain constraints, any system implementing the OSI standards will be 'open' to all other systems implementing the OSI standards throughout the world.

THE REFERENCE MODEL—THE CONCEPT OF A LAYER

The Reference Model provides an abstract structure for a communicating computer system and comprises a hierarchical seven-layer structure as shown in Fig. 2.

Understandably, its abstract nature is of prime importance since this enables the functional requirements to be specified in an implementation-independent fashion and, hence, does not unnecessarily restrict the freedom of a manufacturer as to the way in which the required external behaviour of a system is achieved. At the same time, it specifies the visible external behaviour of a system (that is, the communications protocol requirements) which, of course, enables the compatible interconnection with other systems made by different manufacturers.

There are two types of standard produced for each of the seven layers within the overall framework. One relates to

* CCITT—International Telegraph and Telephone Consultative Committee

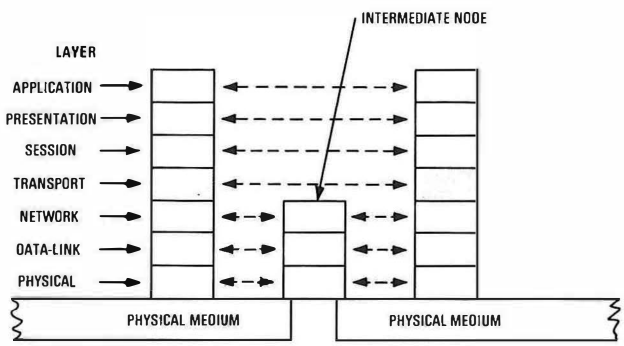


FIG. 2—Structure of the Reference Model

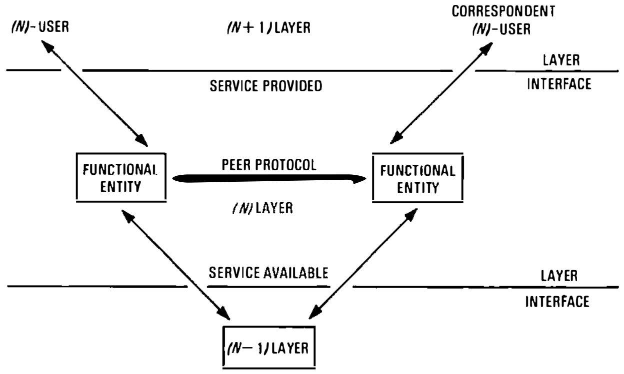


FIG. 3—Layer concepts

the functionality contained within and beneath a particular layer (N)† and represents in abstract terms the 'service' provided to the layer above, (N+1). The other standard specifies the protocol(s) for use within this layer between one system and another.

The concept of a layer is shown in Fig. 3. For a given layer (N), there is knowledge of the service available from layer ($N-1$) and the service to be provided to the layer (N+1), which is the (N) layer service. Generally, the functions within the (N) layer are those required to bridge the service 'gap' between the (N) and ($N-1$) layer services.

The standards for service definition are expressed in terms of elements called *service primitives*. Each service primitive is defined in the form similar to that of a subroutine or procedure call in a computer program; that is, it has a specific name and associated input or output parameters. The sequence of permitted service primitives is defined by the use of state or time-sequence diagrams. State diagrams are used to describe local events within a layer, and are used to show the relationships between events across the local layer service boundary and the consequent events within the layer. On the other hand, time-sequence diagrams are used to describe the relationship of events across the service boundary at one end of a communication with the events across the service boundary at the other.

A simple time-sequence diagram is shown in Fig. 4.

Thus, for each layer of the Reference Model, there is a standard defining the layer service, designated the *service definition*, and one or more standards specifying a protocol that fulfils the requirements of the service definition. The latter are designated *protocol specifications*.

Before each of the seven layers is described, it is necessary

† The brackets denote the Nth layer to distinguish it from *N-layer*, which is used to indicate the Network layer.

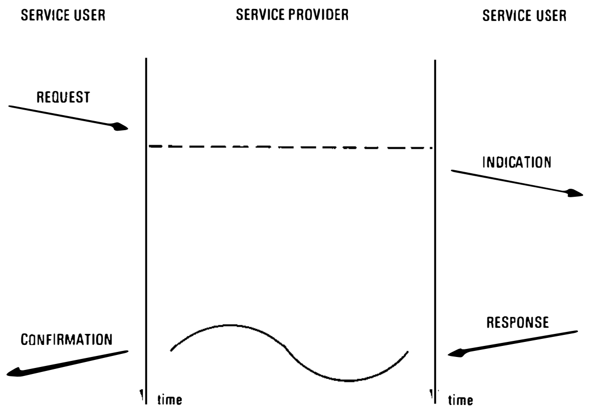


FIG. 4—Simple time-sequence diagrams

to outline the general architectural principles associated with the layering technique and, additionally, the general concept of the mode of operation of a layer; that is, whether it is operating in the connection-oriented mode or the connectionless mode.

The first versions of the Reference Model considered only the connection-oriented mode. The connection-oriented mode is based on classical sign-on, work, sign-off modes of operation. So, for example, the kind of application would be one of reasonable duration, several minutes or longer, involving perhaps a log-in, log-out procedure at the user-computer level, with perhaps a switched connection (real or virtual) between the communicating parties. For this type of operation, the associated (N) layer would clearly have a connection-establishment phase, a data-transfer phase and a disconnection phase. Some 'long-term' association is established between two communicating parties for the duration of the data-transfer phase.

Connectionless mode communication, on the other hand, does not have three phases of operation. There is no connection-establishment or disconnection phase, and no negotiation of the association between communicating parties. This means that a long-term association must exist in advance between the systems, but it is independent of an individual communication. Every time an amount of data is exchanged it is totally self-contained, and thus every piece of data has all the relevant control information associated with it; for example, the calling and called addresses and any other relevant parameters. Datagram networks and transaction applications are typical connectionless mode operations Although there are no public data networks providing Datagram operation, a connectionless mode operation can be gained by using *fast select* packets on X25 networks.

Many of the architectural principles and elements are applicable to both modes of operation, while some may be applicable only to the particular mode. Generally, the principles and associated elements are derived from the basic concept of a layer.

ARCHITECTURAL PRINCIPLES

Functional Entities

The functions in a layer are collected into groups called *entities*. These entities are capable of communicating with other entities in the same layer of different systems by using one or more protocols. An entity provides the functions so that the layer above can use just one, or either, mode of communication. In other words, an entity could just contain the functions to provide a connection-oriented mode of communication, or it could also contain the functions to support a connectionless mode of operation (in which case it provides both connection-oriented and connectionless modes of communication), or it could just contain the functions to provide a connectionless mode of communication. However, there may be more than one entity within a layer in a given system, so the different modes of communication could be available by using different *service access points*.

If more than one protocol is available in a particular layer, then the choice of protocol is made by that layer and not by the layer above, which is using the provided service. The only difference discernable to the (N+1) layer when different underlying protocols are used to provide the same mode of communication is a change in the quality of service (QOS). For example, CCITT Recommendation X25 and X21 interfaces both provide connection-oriented mode operation, but the qualities of the service provided to the user, such as throughput and connection-establishment delay, are quite different. The choice of protocol used by a layer is influenced by:

(*a*) the QOS requirements placed on it by the layer above;
(*b*) a choice of protocol previously agreed with the same

layer in the system being communicated with (particularly for connectionless mode); and

(*c*) the outcome of protocol negotiation with the same layer in the other system during the connection-establishment phase (connection-oriented mode only).

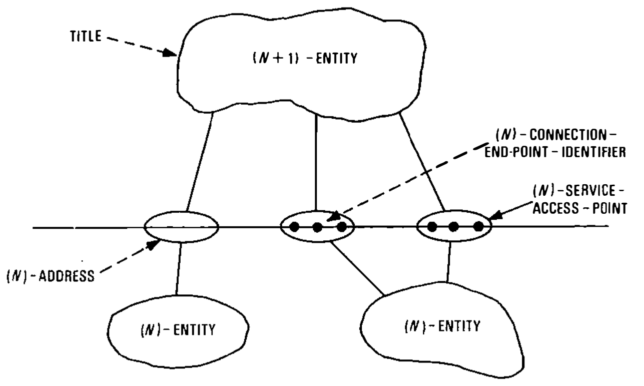
The layers in some systems may not be able to communicate directly with the equivalent layers in all other systems because, for example, no common protocol exists that is implemented in both systems. In such cases, the communication may be routed via one or more intermediate nodes, which can act as protocol converters (see Fig. 2). Within these intermediate nodes, the highest layer of the node contains a routeing and relaying functional block called a *relay*. When a layer contains such a relay, neither the layer above nor the layer below is aware that relaying is being done. The layer above is aware of only a single continuous communication path, while the layer below treats it as two independent communications.

An (N) layer relay is also used when a routeing decision is made by using the (N) address but, in this case, a protocol conversion does not necessarily take place. Thus a packet switching exchange is a layer 3 relay. Transport, Session and Presentation layers are not allowed to contain relays.

Service Access Points

Operations between layers are said to occur at the *service access point* (SAP) for a given instance of communication; that is, for a given connection (out of the set of all connections), or for the transfer of a particular data unit in the case of connectionless mode operation. For a given layer (N), then the (N) SAPs are located on the (N) service boundary, which is the boundary between the (N) layer and the (N+1) layer; for example, the Network SAPs (NSAPs) are located between the Network layer and the Transport layer.

Associated with each SAP is an address. Since only one entity is allowed to reside above a particular SAP, then the address of that SAP can be used to identify the entity (see Fig. 5). This addressing mechanism is used for connectionless mode communication and during the connection-establishment phase of connection-oriented mode communication. Once an (N) connection has been established between two (N) SAPs, and hence between two (N+1) entities, then any information that is passed onto the connection through one SAP will leave the connection through the other SAP; thus no addressing is required. If the SAP has more than one connection associated with it, then the required connection is indicated by the use of a *connection end-point identifier*, which is similar, in principle, to a logical channel number in Recommendation X25.



Note: Dashed arrows refer to identifiers

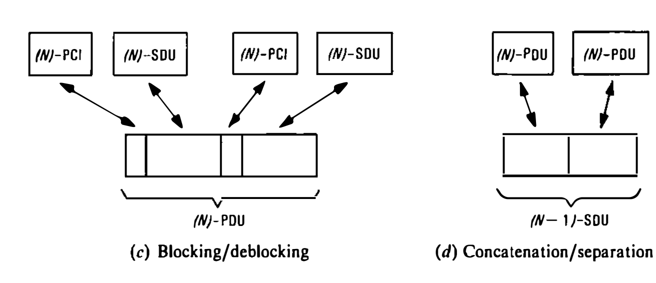
FIG. 5—Entities, service access points and identifiers

Not all the operations between layers are as well defined as this. If a particular inter-layer exchange does not result in a significant change in the external behavior of the system, then it is considered a matter local to the system and, hence, not a proper subject for OSI standardisation. One such subject is flow control.

Flow Control

Two types of flow control are required for the correct operation of the Reference Model architecture. One is explicit flow control between peer layers in communicating systems. This involves protocol exchanges between the systems and hence is well defined by the model and its associated standards. The other type is a more implicit flow control based on *backpressure*. In this case, when a layer has reached its limit for handling data, it asks the layer supplying it with data (that is, the layer above if it is transmitting or the layer below if it is receiving) to stop the flow of data. The layer that is thus stopped can then do the same thing to the layer supplying it with data, and so the flow control moves back along the communication path. Since the flow control exchange is between two layers within a system and does not result in any detectable change in external behaviour, this type of flow control is considered to be local to the system and is not explicitly described by OSI documents, even though the architecture could not operate without it.

Blocks of data that are passed through the service boundary for transmission by the layer are called *service data units* (SDUs). The functions within a layer add any necessary *protocol control information* (PCI) so that the peer entity correctly receives the SDU and delivers it to its intended destination. The SDU and PCI combined form the *protocol data unit* (PDU), which is then passed to the layer below (see Fig. 6(*a*)). Upon passing through the lower service boundary, the PDU becomes the SDU of the lower layer. It is worth noting that an SDU is, therefore, not just data, but also protocol from the layers above; in fact, the SDU may contain no data at all in some cases (for example,



SDU: service-data-unit
PCI: protocol-control-information
PDU: protocol-data-unit

Note: In the case of concatenation, (N)-protocol-data-unit does not necessarily include an (N)-service-data-unit

FIG. 6—Relationship between (N)-service-data-unit, (N)-protocol-data-unit and (N−1)-service-data-unit within a layer

in the case of higher-layer flow control). However, in all layers, the SDU represents a sequence of bits that is uninterpretable at that layer; this sequence of bits is user data where the user is the layer above.

The simple mechanism described above is not always used; for example, it assumes that a whole SDU can be transferred in a single PDU. If this is not true, then segmentation may be used (see Fig. 6(*b*)). For protocol efficiency, it is also possible to transfer more than one SDU in a single PDU (blocking) (see Fig. 6(*c*)) or more than one PDU into an SDU as the service boundary is crossed (concatenation) (see Fig. 6(*d*)). Since integrity of SDUs must be maintained at both ends of a communication path (that is, what comes out at the receiving end is identical to what was put in at the transmitting end), the inverse functions—reassembly, deblocking, and separation—are required to recreate the original SDUs in the peer layers of the receiving system so that they can be correctly delivered to the destination.

DESCRIPTION OF THE LAYERS IN AN OPEN SYSTEM

Outside the Seven Layers

Before the functionality of each layer is described, it is worth looking briefly at what is outside the seven layers, but is, nevertheless, essential to the concepts of OSI.

Above the Application layer is the application process (for example, an operator at a terminal or a program) that is using the underlying protocols to communicate to another application process in a different Open System.

Below the Physical layer and physically connecting Open Systems together is the *physical medium*. This can be a cable pair, an optical fibre, or the ether (for radio communication links).

Physical Layer

The Physical layer provides the means to transmit bits of data across a continuous communication path. If a multiplexing frame is used to allow several communications across a single physical medium connection, this frame would be a Physical layer protocol. The Physical layer is concerned with the electrical and mechanical requirements of a communication and of system activation and deactivation.

A Physical-layer relay can be used to connect different physical media together (for example, a satellite ground station) to form a physical communication path for both connection-oriented and connectionless mode communication.

Data-Link Layer

The Data-Link layer provides framing for data transfer across a Physical connection. It also detects and, where possible, recovers from errors. Where recovery is not possible, the Network layer is notified of the error. However, in the connectionless mode of operation, although error notification can occur, error recovery is not possible. This latter type of operation is exhibited by IEEE† defined local area networks when type 1 logical link control (LLC) is used.

The Data-Link layer may use more than one Physical connection to support a single Data-Link connection. In this case the Data-Link protocol requires a resequencing capability in order to deliver in the correct order data that has been transfered by using different Physical connections to the Network layer. The Network layer is unaware of the Physical connections and is aware only of the Data-Link connection having a large throughput.

Where a continuous physical medium connects together more than two Open Systems (for example, a local area network), the Data-Link layer is able to determine the intended destination of a communication; that is, a primitive routeing function that takes the form of a one-of-*n* selection exists in the Data-Link layer.

Network Layer

The main function of the Network layer is the interconnection of Data-Link communication paths into a global network that connects all Open Systems. This is currently achieved in two, quite distinct, stages.

There are many widely used standards for network layer protocols to interconnect Data-Link communication paths. None of these, however, provides the OSI Network service and are therefore termed *subnetworks* within the OSI environment. The first stage is to take the existing network protocols, such as CCITT Recommendation X25, and define a supplementary network protocol that operates over the top of the existing protocol and thus enhances its functionality so that it meets the requirements of the OSI Network service (see Fig. 7). This provides several individual networks each achieving the OSI Network service but based upon a different subnetwork protocol.

The second stage is to interconnect all these enhanced subnetworks into a single global network. This is done by using Network layer relays. The Network layer is, therefore, ultimately responsible for routeing a communication (in connectionless or connection-oriented mode) between the Open Systems involved. The implication of this is that the NSAP addresses are globally unambiguous (that is, each address identifies only one NSAP, although an NSAP can have more than one address).

Another function important to the Network layer is segmentation/reassembly. Each of the subnetworks being concatenated to form the global OSI network can have a different limit on the maximum size of a data unit that it can transfer in a single protocol exchange. Consequently, when the next subnetwork has a smaller data size, the relay and end Open Systems need to provide segmentation with subsequent reassembly in the receiving Open System.

During a connection, in addition to normal data transfer, the Network layer can provide an 'expedited' data transfer service. Such data is not subject to normal flow control and can arrive at the destination before data that had been sent prior to it (for example, an interrupt packet in X25).

The Network layer is the highest layer in which the conversion between the different modes of operation (connection-oriented subnetwork to connectionless Network and vice versa) is generally available.

Transport Layer

The Transport layer is the lowest layer in which relay is not

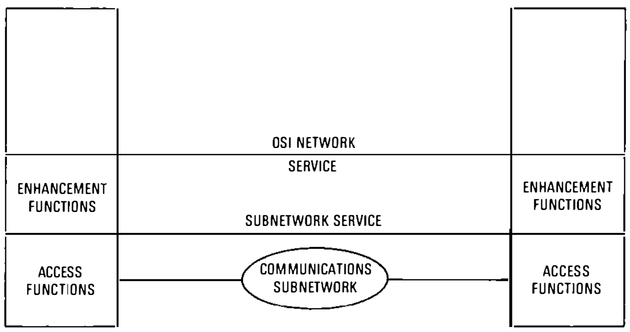


FIG. 7—Enhancement of existing networks to provide OSI Network service

† IEEE—Institute of Electrical and Electronics Engineers, Inc.

allowed and, hence, it is the lowest layer in which the protocol always operates between the communicating end-systems, as opposed to an intermediate point (see Fig. 2). This means that there are no Transport-layer relays to perform protocol conversion and, hence, the systems communicating need to agree on a common transport protocol which they use to communicate. For connectionless communication, this implies a prior agreement between the end systems, but for connection-oriented communication, a protocol negotiation mechanism can be used during call establishment. There are five protocols currently defined for use in the Transport layer and these are designed to provide a connection-oriented Transport service over a connection-oriented Network service (see ISO DIS 8073 or CCITT Recommendation X224). These protocols are negotiable during the connection-establishment phase. Some of these protocols are also defined in other standards; for example, Class O Transport protocol is also defined as the Transport protocol for Teletex in CCITT Recommendation S70.

For a connection-oriented communication, the choice of protocol by the Transport layer is affected by the difference between the QOS required by the Session layer and that achieved by the underlying Network connection. The Transport layer may also try to minimise networking costs by multiplexing multiple Transport connections onto a single Network connection; that is, the Transport PDUs of several Transport connections between the same systems are transmitted and received by using the same Network connection. The presence or absence of multiplexing will also affect the choice of protocol. The Transport layer can use multiple Network connections in parallel to increase the throughput apparent to the Session layer, and this too would affect the choice of protocol.

The connection-oriented Transport service has an expedited data transfer facility that is similar to the one described for the Network layer.

Addressing is of less importance in the Transport layer than in the Network layer, as it affects only the routeing of a communication within an Open System, and hence an unambiguous Transport address is easier to achieve. The address provides a one-of-*n* selection where *n* Transport SAPs (TSAPs) can be reached by the same NSAP; that is, by selecting the correct end-point as the communication passes through the Transport layer, having reached the correct Open System. (See Fig. 8.)

Session Layer

The Session layer is the lowest layer of the Reference Model that deals explicitly with the communication of the Open Systems as opposed to the interconnection of the Open Systems. Like the Transport layer, the Session layer protocol is end-to-end and, for connection-oriented communication, it is negotiated during the connection-establishment phase. However, the mechanism for negotiation is different, in that the Session layer protocol contains a set of optional protocol functional units, from which a subset is chosen that forms the Session layer protocol for that Session connection. The choice of Session protocol affects the service made available

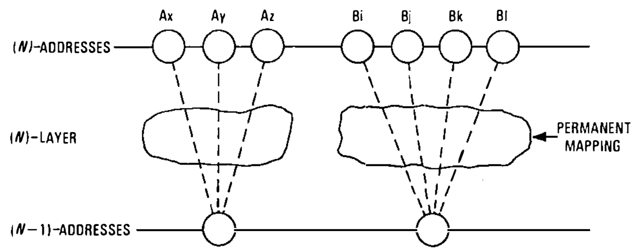


FIG. 8—Hierarchical (*N*)-address-mapping

to the Presentation layer which, consequently, has a greater influence on the choice of the Session protocol than the Session layer has on the choice of the Transport protocol. If certain groups of these functional units are chosen, then the resultant Session protocol can be the same as protocols defined in other documents; for example, one subset of the functions can be selected to provide the Session protocol defined for Teletex (CCITT Recommendation S62).

The Session layer provides interaction management for the Presentation layer by negotiating the type of interaction to be used (for example, two-way simultaneous, two-way alternate, or one-way), and then by controlling which system's presentation layer is allowed to invoke particular Session services according to the agreement reached. The Session layer can also provide a data synchronisation service (that is, during text transfer, the Session layer can provide synchronisation marks at, for example, page boundaries of text) and an expediated data service to the Presentation layer if the relevant protocol functions are included in the chosen subset.

The Session layer may reuse a Transport connection for several Session connections sequentially (that is, one at a time) or several Transport connections sequentially for one Session connection, but at any instant of time there is, at most, a one-to-one correspondence between Session and Transport connections. The Session layer also tries to recover from Transport layer signalled errors. The Session layer that attempts the recovery is determined by the interaction management.

Presention Layer

The Presentation layer is concerned with making the information transferred comprehensible to the receiving system.

Each system has a particular method for storing each data type, defined by the encoding rules and the data structures used in the system (termed the *concrete syntax* in OSI). Systems of the same type use the same concrete syntax, but since OSI aims to encompass all types of systems, a major task for OSI is to allow communication between application layers in systems using different concrete syntaxes. The Presentation layer is primarily concerned with this task.

During the connection-establishment phase, the two Presentation layer entities negotiate to determine the extent of the syntax translation to be undertaken by each entity. If both systems use the same concrete syntax, then no translation is required. Otherwise, both Presentation layers can be involved in translating from one concrete syntax to another for the duration of the communication. In this case, a third concrete syntax is used to transfer the information between the systems using the OSI protocols; such a syntax is called a *concrete transfer syntax*.

After the completion of the connection-establishment phase, further syntax negotiation may occur during a communication if, for example, a new data type is required for which a concrete transfer syntax had not previously been negotiated.

Application Layer

The Application layer provides the interface between the communications environment and the application process using it. In some cases, this is an indivisible part of the application process itself, in which case the process is modelled as partly residing within the Application layer. Otherwise, the application process can use an interface which it constructed by selecting a subset of the set of Application service elements made available by the Application layer. In this case, the application process is seen as existing wholly outside the Reference Model.

TABLE 1
Standards Relating to Open System Interconnection

	CCITT	ISO
Reference Model	X200	ISO 7498
Network Service	X213	DIS 8348
Transport Service	X214	DIS 8072
Session Service	X215	DIS 8326
Transport Protocol	X224	DIS 8073
Session Protocol	X225	DIS 8327

X = CCITT Recommendation produced by Study Group VII
ISO = Agreed International Standard
DIS = Draft International Standard

STANDARDS ACTIVITY

As mentioned in the introduction, both CCITT and ISO are working on the standards and both are producing documents for OSI. The close co-operation between the groups in each organisation that deal with the same areas has meant that the current documents are in extremely close alignment, the major differences being editorial in nature. The documents that are currently stable are given in Table 1.

There are several existing standards which are suitable as protocol specifications for layers below the Network layer (for example, high-level data link control (HDLC) is suitable as a Data-Link protocol). There are no suitable standards currently available that could be used to specify a Network protocol. All current specifications (including Recommendations X21 and X25) fall short of the required Network layer functionality and, hence, can be considered only as subnetworks. This has resulted in an activity to develop enhancement protocols to provide the OSI Network service using these subnetworks, and other work to enhance the protocols themselves so that they can eventually be considered as OSI Network protocols. This latter activity has resulted in many changes to the encoding of the facilities field of the *call establishment* packets as defined in CCITT Recommendation X25.

In the UK, the Department of Trade and Industry has produced a set of intercept recommendations to encourage British industry to develop products by using working documents in important areas of standardisation activity which are not yet standards, but are stable enough to use. The draft standards listed in Table 1, and the working documents that currently define the enhancement protocols for providing the OSI Network service over X21 and X25 subnetworks, are the subject of some of these intercept recommendations.

Biographies

Paul Jenkins graduated with a first-class honours degree in Electrical and Electronic Engineering from Bradford University in 1979. He was then employed for three years by Marconi Research Laboratories in Great Baddow, Essex, where he was involved with the design of microprocessor-controlled test and inspection equipment. Since joining British Telecom (BT) in 1982 he has worked in the OSI Standards Studies Section within the Communication Standards Division of BT Development and Procurement. He is a regular member of the British Standards Institution (BSI) working groups studying the Data-Link, Network and Transport layer standards, and has been part of the UK delegation to three international meetings studying these areas.

Keith Knightson is the head of the OSI Standards Studies Section of the Communication Standards Division of BT Development and Procurement. He graduated from Central London Polytechnic in 1973 with a B.Sc. in Applied Computing, and from Essex University in 1974 with an M.Sc. in Computer Science. During his career with BT, he has been involved in the development of CCITT packet-switching Recommendations as well as the specification and implementation of BT's packet-switched service, value added services, and the UK's earlier experimental packet-switched service. He actively participates in the CCITT and ISO work on OSI and is currently the CCITT Special Rapporteur for the Transport layer and Network service. In addition, he is chairman of the BSI working group studying the Network layer. He has recently been appointed to manage the UK Government Information Technology Standards Unit's initiative in the connection, communication and switching areas.

Book Review

Electronic Inventions and Discoveries. Third revised edition. G. W. A. Dummer. Pergamon Press Ltd. ix+233 pp. 33 ills. £9·95.

This is the third edition of a book that comprehensively records the history of development in the field of electronics. This edition has been revised and enlarged, and now describes over 500 inventions and includes over 1000 references.

Several chapters have been devoted to descriptions of early discoveries, the development of components and circuits, and their application, including electronics in industry. Further chapters provide concise histories of audio and sound reproduction, radio and telecommunications, radar, television, computers, information technology, and industrial applications.

The book also contains a list of inventions by subject (under 13 separate headings), a list of inventors, and a list of books on both inventors and inventions; however, some of the older books listed are now out of print, or not easily obtainable.

The main part of the book describes, in historical order, each invention listed, and gives further references, including abstracts, patents and, in some instances, diagrams. Some of the descriptions are very generous; for example, the scanning electron microscope (1935) is given almost a page and a half. It is interesting to note the very early dates of some inventions which were not able to be properly exploited until later technologies became available. Telecommunications history is divided between 'Audio and sound reproduction' and 'Radio communications'. Packet switching is mentioned under both headings but, while reference is made to pulse-code modulation, optical fibres, and 65 separate communication satellites, none is made to frequency-division multiplexing and time-division multiplexing; nor to the less glamorous developments in land-cable transmission systems, which in fact form the major part of so many transmission networks.

There are one or two occasions when an idea might have been more clearly expressed; for example, 'the latest video recorder tapes can record up to 18 octaves for colour television as against 9 octaves for normal sound recording'. But this is a minor quibble, and the book can be highly recommended both to the specialist as a work of reference, and to the amateur as general-interest reading about discoveries which are permeating so many different aspects of social and economic life today.

K. E. PARISH

Error Performance Objectives for Digital Networks

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UDC 621.391.88 : 621.395.49

The liability of a digital signal to error is one of the most significant transmission impairments experienced in a digital communication network. Consideration is given to the activities and status of studies within the CCITT. The sources of interference causing errors are discussed and an introduction is given to hypothetical reference connections. The internationally agreed end-to-end error performance specification for a digital connection is reviewed, along with the methodology for allocating this overall objective to constituent transmission equipment operating over different media. A number of mathematical models are identified which might be adequately representative of the error characteristics to be found in real networks. Finally, measurement aspects are discussed.*

This article is a revised version of a paper which first appeared in The Radio and Electronic Engineer‡ and is reproduced by permission of the Institution of Electronic and Radio Engineers. This updated version takes account of the final CCITT SGX VIII meeting during the current study period (1980–84).

INTRODUCTION

Telecommunication networks throughout the world are currently in the process of being digitalised and one of the main tasks of international standard-setting organisations, such as the CCITT, is to formulate performance objectives which ensure that even the most complex international connections are acceptable to users. Users need to know the quality of service provided by telecommunication networks so that judgements can be made as to their suitability for planned applications. Quality-of-service objectives encompass a wide range of parameters concerned with availability, reliability, call processing as well as the classical transmission impairments.

One of the most significant transmission impairments in digital networks is the liability to error. The purpose of this article is to discuss the activities and status of studies within CCITT concerning error performance. Significant progress has been made in establishing overall end-to-end objectives for networks as a whole, as well as for constituent parts.

BACKGROUND INFORMATION

Error Performance Parameters

Early theoretical studies of error performance were based on the assumption that practical error distributions could be described successfully in terms of a constant bit-error probability, numerically equal to the long-term mean error ratio (LTMER) (that is, the ratio of the number of bits received in error to the total number of bits transmitted). The justification for such a choice came out of the belief that a significant proportion of error events in a real network arise as the result of random processes and, consequently, it was assumed that the resultant error characteristics could be adequately described by using a simple Poisson model. This traditional method of describing error performance has been found to be deficient, in that it does not, for many types of system, provide an accurate characterisation of the distribution of errors with time. For users to plan effective communication services, a more useful and meaningful measure of performance needs to include information about how error events are distributed with time.

Based on operational experience within British Telecom and other administrations, there is now considerable evidence to show that errors tend to occur in 'bursts' or 'clusters' to varying degrees[1, 2, 3, 4]. Earlier assumptions that errors are randomly distributed in time seem not to be valid and neither, therefore, is a simple Poisson model.

In retrospect, this is fairly obvious because of signal processing devices commonly found in digital networks. For example, self-synchronising scramblers usually provide three errors in the descrambled signal for every single error on the transmission link. Similarly, line coding techniques often give rise to error multiplication effects. Furthermore, many sources of interference which cause errors are impulsive in nature.

The occurrence of a few, but very intense, error bursts can have a significant impact on overall measured mean bit error ratio, but such an occurrence might not seriously affect a particular service. Consequently, it is desirable to establish an alternative means of defining error performance in a way that more directly relates to the effect on services. The philosophy emerging from CCITT is to use two types of measure:

(*a*) the proportion of error-free time intervals, which is thought to be useful for many data-type services; and

(*b*) the proportion of time that a specified short-term bit-error-ratio threshold is exceeded, which is thought to be useful for voice and some other services.

The Importance of Error Characteristics

In deciding which error performance parameters should be adopted, it is necessary to realise that different services are affected by errors in different ways. For example, one service might be insensitive to a low background error ratio exhibiting a random distribution, but sensitive to error bursts, while another service may react in the opposite way. This sensitivity of a service to the distribution of errors needs to be taken into account. For example, with typical data services, information is assembled and transmitted in blocks, and the systematic inclusion of error control bits in each block enables the receiving terminal to check the validity of the received information. With this type of arrangement, the actual number of errors in a block is usually immaterial, since any error will require a retransmission of the entire block. For this type of service, for a given number of errors,

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* CCITT—International Telegraph and Telephone Consultative Committee

‡ McLINTOCK, R. W., and KEARSEY, B. N. Error performance objectives for digital networks. *The Radio and Electronic Eng.*, Feb. 1984, **54**, pp. 79–85.

it is preferable that they occur in bursts, since this will result in fewer retransmitted blocks and thus a greater throughput efficiency.

In contrast, with video services, the effect of a single error may persist for some time. Of course, error correcting codes can be used to deal with single and sometimes small bursts of errors but not usually large bursts.

Sources of Errors

Although there are many sources of interference which cause errors, it is possible, at least conceptually, to classify them into two distinct types: predictable and unpredictable sources.

Predictable Sources

These sources of interference relate to known system design limitations, which the designer can take into account. Examples of some interferences that are significant for particular systems are

(*a*) thermal noise in the case of high-speed digital line systems operating on coaxial cable,
(*b*) the combined effect of crosstalk from signals on other pairs in the same cable for digital line systems that operate on symmetric-pair-type cable, and
(*c*) propagation effects in accordance with a model for radio and satellite systems.

Unpredictable Sources

These sources of interference, sometimes referred to as *network effects*, are often both unpredictable in nature and unquantifiable in magnitude. They are much more difficult and sometimes impossible to take into account during system design. Examples of some typical sources of this type of interference are

(*a*) equipment imperfections (for example, dry joints and intermittent connectors),
(*b*) electrical interference from external sources (for example, impulsive noise from analogue exchange switching equipment, electric trains, lightning discharges, switching of heavy power currents),
(*c*) manual intervention (for example, during maintenance and repair), and
(*d*) freak radio propagation effects not covered by standard propagation models.

The totality of error incidence can therefore be regarded as the combined effect of the underlying random processes, generally stemming from known and predictable design limitations, and the rather less predictable error-burst phenomena.

TYPES OF ERROR PERFORMANCE OBJECTIVE

Before the details of CCITT error performance studies are discussed, it is useful to review the meaning of the two terms *network performance objectives* and *system* or *equipment design objectives*. See Fig. 1.

Network Performance Objectives

Network performance objectives are set for complete connections through the network. They are important from the network user's point of view because they give him an indication of the actual performance that he can expect under real operating conditions. These objectives take account of all sources of errors.

Equipment Design Objectives

Equipment design objectives are of use to system designers and manufacturers. They give a list of performance require-

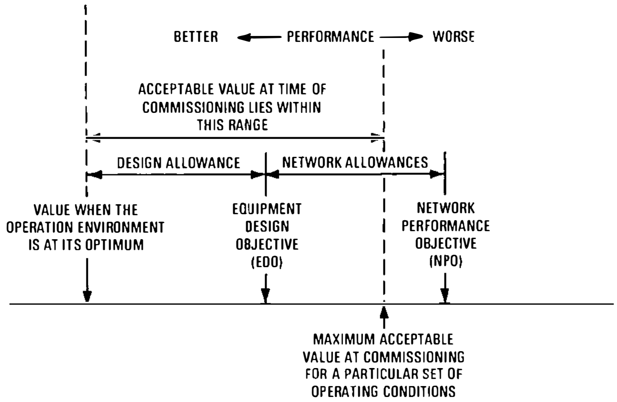


FIG. 1—Illustration of the significance of network performance objectives and equipment design objectives (based on CCITT Recommendation G102)

ments that should be met under specified working conditions. In principle, the performance limits should be met even when all the working conditions are simultaneously at their most adverse values. Generally, equipment can be readily evaluated against design objectives under laboratory test conditions. In terms of error-ratio performance, design objectives usually permit a very small error ratio, recognising that a truly zero error-ratio performance cannot usually be attained in the presence of predictable disturbing signals, some of which are unavoidable.

Operational Error Performance

When systems are installed into a real network situation, the actual performance achieved is likely to be different from that quoted as the limiting design objective, for two reasons:

(*a*) the actual working conditions are unlikely to be all at the most extreme values quoted as part of the design objective and, for this reason, the actual performance tends to be significantly better than the design objectives and, in many cases, systems appear to be error free, except for
(*b*) the fact that, under real operating conditions, there are interfering signals present which it is not possible to fully quantify (unpredictable sources). Ultimately, it might be possible to quantify such sources of interference. By a combination of improved installation practices (under the control of the operating administration) and by greater immunity of system design, errors attributable to these causes may be reduced. However, it is unlikely to be economic, or even necessary from the user's point of view, to eradicate all errors due to these effects.

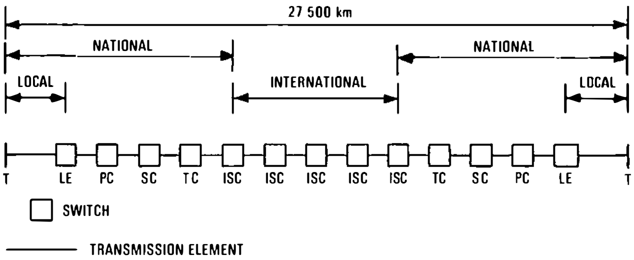
Summary

It is useful to summarise at this point:

(*a*) Network performance objectives are of interest to users of networks. Both equipment design objectives and installation practices/operating environment affect network performance.
(*b*) Equipment design objectives are of interest to equipment designers. The onus is on the person who prepares the specifications to ensure that the requirements accurately reflect the real working conditions in the network to the maximum extent possible.

HYPOTHETICAL REFERENCE CONNECTION (HRX)

A digital HRX is a model with which studies relating to



T: T-reference point. See CCITT Recommendation I411. The T-reference point is a CCITT defined subscriber/interface ISDN interface.
LE: Local exchange
PC: Primary centre
TC: Tertiary centre
ISC: International switching centre

FIG. 2—Digital hypothetical reference connection (longest length)

overall performance can be conducted, and thereby facilitating the formulation of standards and objectives. In order to initiate studies directed at the performance of an integrated services digital network (ISDN), an all digital 64 kbit/s connection has been used. The model often used by the CCITT is one which is tending towards the longest envisaged international connection and is described in CCITT Recommendation G104[6] and draft Recommendation G81x[7, 12]. Only a few real connections will be longer. The diversity of national and international networks in respect of country size, circuit implementation, terrain difficulties, national policies etc. naturally preclude the accommodation of all the significant features into a single or even a few simple network models. Essentially, the longest standard HRX provides a uniform basis for the study of digital impairments, providing a common reference datum for the specification of international performance standards. Obviously, national administrations may need to develop their own representative network models reflecting the features of their own evolving national digital networks, in order to validate prima facie compliance with international standards.

The error performance objectives given in CCITT Recommendation G821 (see later) are expressed in relation to a model based on this longest standard HRX (see Fig. 2).

Two other shorter HRXs have been defined in CCITT Recommendation G104 and draft Recommendation G81x to facilitate studies of typical performance levels for more commonly occurring lengths of international connection.

ERROR PERFORMANCE SPECIFICATION

CCITT Recommendation G821

(Error Performance of an International Digital Connection forming part of an Integrated Services Digital Network)

The original proposed objective for a digital network was in terms of a limiting value for the long-term mean error ratio of a longest-length international 64 kbit/s HRX. An overall value of 1×10^{-5} represented a level of impairment which was considered 'just discernable on low-level speech' when pulse-code modulation (PCM) was used. However, this objective was never adopted as a formal CCITT Recommendation.

The current CCITT position on error performance objectives can be found in the proposed revision of Recommendation G821[7]. Although the original version of G821 was published in 1980, some important amendments and extensions of the Recommendation are under consideration and the final results of these deliberations will be published later this year. The following paragraphs relate to these latest considerations.

Because the objectives are aimed at network users, it is necessary to express the performance in a way that gives the most information about the effects on services. Based on

TABLE 1
Error Performance Objectives for an International ISDN HRX at 64 kbit/s

Performance Classification	Objective
(*a*) Degraded minutes (notes 1, 2)	Fewer than 10% of 1-minute intervals to have a bit error ratio worse than 1×10^{-6}
(*b*) Severely errored seconds (note 1)	Fewer than 0.2% of 1-second intervals to have a bit error ratio worse than 1×10^{-3}
(*c*) Errored seconds (notes 1, 3)	Fewer than 8% of 1-second intervals to have any errors (equivalent to 92% error-free second (EFS))

Notes: 1 The terms *degraded minutes*, *severely errored seconds* and *errored seconds* are used as a convenient and concise performance-objective identifier.
2 The 1-minute intervals are derived by removing unavailable time and severely errored seconds from the total time, and then consecutively grouping the remaining seconds into blocks of 60. The basic 1-second intervals are derived from a fixed time pattern.
3 The 1-second integration period was chosen in the belief that it represented a short enough time interval to display sufficient detail of the error structure, yet long enough to encompass most block lengths that data users may choose for error control.
4 The time interval T_L over which the percentages are to be assessed has not been specified since the period may depend upon the application. A period of the order of any one month is suggested as a reference.

the considerations given earlier in the Background Information section of this article, the error performance requirements of an ISDN are defined in the following way:

> 'The percentage of averaging periods each of time interval T_0 during which the bit error ratio exceeds threshold value. The percentage is assessed over a much longer time interval T_L.'

Although the objectives are expressed to suit the needs of different services, during the original formulation two important classes of user were identified, namely *telephony users* and *data users*. Consequently, the latest formulation reflects this influence. The three-part network performance objective for the longest international 64 kbit/s HRX from customer to customer is shown in Table 1, all parts to be met concurrently.

The network performance objectives, although expressed in a way to suit the needs of different services, are intended to represent, to a first approximation, a single level of transmission quality, although for some types of system one part of the objective might tend to be dominant. Other envisaged ISDN usages (for example, facsimile, videophone) may require the formulation of additional requirements expressed in other ways but which are consistent with this notion of a single quality.

In formulating these objectives, there was no definitive starting point, and it was necessary to reach a compromise between a desire to meet service needs and a need to realise transmission systems, taking into account economic and technical constraints.

It should be noted that total time (T_L) is split into two parts, namely that time for which the connection is deemed to be available and that time when it is unavailable. The objectives refer to the period when the connection is available. When the bit error ratio in each second is worse than 1×10^{-3} for periods equal to or exceeding ten consecutive seconds, the connection is considered to be unavailable.

Allocation of Overall Network Performance Objectives

The error performance objectives aim to serve two main functions:

(*a*) to give the user of future national and international digital networks an indication as to the expected error performance under real operating conditions, thus facilitating service planning and terminal equipment design; and (*b*) to form the basis upon which performance standards are derived for transmission equipment and systems used to support connections.

To satisfy this latter aim, it is necessary to subdivide the overall end-to-end objective to constituent parts of networks and even to individual equipment and systems.

The overall apportionment philosophy involves the use of two slightly different strategies, one applicable to the degraded-minute and the errored-second requirements (objective parts (*a*) and (*c*)) and the other applicable to the severely-errored-second requirement (objective part (*b*)).

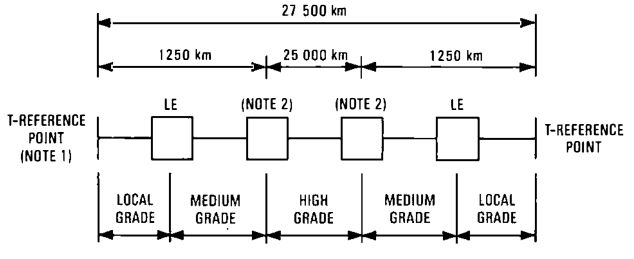
Basic Apportionment Principles[7, 8]

An important feature of the basic approach is that apportionment is based on the assumed use of transmission systems having qualities falling into one of a limited number of different classifications rather than the allocation of fixed allowances to the national and international portions. Based on these considerations, three distinct quality classifications, representative of practical digital transmission systems, have been identified. These classifications are termed *local*, *medium* and *high grade* and their usage generally tends to be dependent on their location within a network (Fig. 3). The three circuit quality classifications for transmission systems are as follows:

(*a*) *Local Grade* This embraces systems assumed to be operating between customers' premises and local exchanges (LEs), and will typically be low-digit-rate transmission systems operating at a rate below the primary levels of 1544 and 2048 kbit/s over a diverse range of cable types and other media.

(*b*) *Medium Grade* This embraces systems assumed to be operating between local exchanges (LEs) and beyond into the national part of the connection. The actual distance covered by such systems will vary considerably from one country to another, but under most circumstances will not exceed 1250 km. For example, in large countries, this distance might reach only the primary centre while in smaller countries, it may go as far as the secondary centre, tertiary centre or the international switching centre. For this reason, transmission systems assigned to this classification will exhibit a variation in quality falling between the local- and high-grade classifications. In many cases, the variation in quality relates more to network planning and maintenance aspects, rather than to intrinsic differences in system quality. Typically, these systems will operate at low, medium or high digit rates and use a variety of media.

(*c*) *High Grade* This embraces systems used for realising transmission over both long-haul national and international parts of a connection. Typically, such systems operate at medium or high digit rates, utilising all types of media (for example, metallic, optical, radio and satellite). Of the total HRX length (that is, 27 500 km), a significant proportion will be effected using high-grade plant.



Notes: 1 The T-reference point is a CCITT defined subscriber/network interface.
2 This point may be at the LE, PC, SC, TC, or ISC depending on country size. (See section on basic apportionment principles (*b*).)

FIG. 3—System quality demarcation for longest HRX

TABLE 2
Proportionate Allocation for Degraded-Minute and Errored-Second Objectives

Quality Classification	Proportionate Allocation of Degraded-Minute and Errored-Seconds Objectives
Local grade (two ends)	15% block allowance to each end
Medium grade (two ends)	15% block allowance to each end
High grade	40% (equivalent to conceptual quality of 0·0016% per kilometre for 25 000 km)

These quality classifications for different parts of the connection represent what is believed to be the situation for a large proportion of real international connections. However, it is recognised that there will be exceptions to this rule in some countries. If administrations configure networks using transmission systems with the appropriate quality for their area of application, the vast majority of real connections will experience a level of performance consistent with, or better than, the recommended performance objectives. There will be a small number of real connections that are longer than the 27 500 km HRX. By definition, the additional connection length will be carried over high-grade plant and, hence, the amount by which such connections exceed the overall objective for the HRX will be proportional to the amount by which the 25 000 km portion is exceeded.

The following general assumptions apply to the apportionment strategy that is described below:

(*a*) In apportioning the overall objective to the constituent elements of a connection, it is the 'percentage of averaging periods that is permitted to exceed the threshold values' that is subdivided.

(*b*) An equal apportionment of the overall Recommendation G821 objective applies for both 'percentage of degraded minutes' and 'percentage of errored-seconds'.

(*c*) The error-ratio thresholds are not themselves subdivided.

(*d*) The error contributions from both digital switching elements and digital multiplex equipments are assumed to be negligible in comparison with the contribution from transmission systems.

Apportionment Strategy for the Degraded-Minute and Errored-Second Requirements

Table 2 illustrates the basis on which the overall permissible degradations are allocated to the constituent portions of the HRX.

The local- and medium-grade apportionments are considered to be block allowances, regardless of length. It does not seem appropriate further to subdivide these objectives on the basis of length. In practice, because of the many variables associated with transmission in these parts of the network, shorter connections will not necessarily yield better error performance than longer ones. Furthermore, it does not seem appropriate to set different standards for different customers, depending upon the distance to their local exchange (except in extreme cases).

To introduce an element of flexibility, administrations are permitted to allocate the overall block allowance for the local- and medium-grade portions as necessary up to the first 1250 km of a connection from the customer's premises.

The overall high-grade objective is subdivided on the basis of length, resulting in a conceptual per kilometre allocation, which can be used to derive a block allowance for particular lengths of system.

Based on the foregoing, the derived network performance objectives are given in Tables 3 and 4.

If account is taken of the fact that the error performance

TABLE 3
Allocation of Percentage of Degraded-Minute and Errored-Second Objectives

Quality Classification	Network Performance Objective at 64 kbit/s	
	Percentage of degraded minutes	Percentage of errored seconds
Local grade (block allowance for each end)	< 1·5	< 1·2
Medium grade (block allowance for each end)	< 1·5	< 1·2
High grade (for 25 000 km)	< 4·0	< 3·2

TABLE 4
Allocation of Percentage of Degraded-Minute and Errored-Second Objective for High-Grade Transmission Systems

High-Grade System Length	Network Performance Objective at 64 kbit/s	
	Percentage of degraded minutes	Percentage of errored seconds
280 km	< 0·0448	< 0·03584
420 km	< 0·0672	< 0·05376
2500 km	< 0·4	< 0·32

TABLE 5
Severely-Errored-Seconds Objectives

Quality Classification	Network Performance Objectives at 64 kbit/s
	Percentage of severely errored seconds
Local grade (block allowance to each end)	<0·015
Medium grade (block allowance to each end)	< 0·015
High grade (for 25 000 km)	< 0·04

Note: This table includes only the partial allocation and does not take into account the additional media-dependent allowance described in the text.

of a satellite system is largely independent of the terrestrial distance covered, a block allowance of 20% of the total permitted degraded-minute and errored-seconds objectives is allocated to a single satellite link employed in the high-grade portion of the HRX. If the high-grade portion of a connection includes a satellite system and the remaining distance included in this classification exceeds 12 500 km, or if the high-grade portion of a non-satellite connection exceeds 25 000 km, then the objectives given in Table 1 may be exceeded. The occurrence of such connections is thought to be relatively rare.

Apportionment Strategy for Severely Errored Seconds

The apportionment strategy outlined in the previous section is considered to be inappropriate for the severely-errored-seconds objective. The total allowance of 0·2% is subdivided between each classification (that is, local, medium and high) in the following manner.

(*a*) 0·1% is divided between the three circuit classifications in the same proportions as adopted for the other two objectives. This results in the allocation given in Table 5.

These allowances are media independent, and aim to take account of both adverse network effects (for example, external interference) and basic design limitations. Transmission systems covered by the high-grade classification should not contribute more than 0·004% for each 2500 km portion. The relative significance of these two factors is dependent, in the main, on the type of transmission medium. For example, terrestrial radio-relay and satellite systems are subject to adverse propagation conditions. It is normally not economic to design such systems to provide a good error-ratio performance for 100% of the time, but to accept that, occasionally, owing to extreme propagation conditions, the performance will be degraded and, by design, will impinge on the severely-errored-seconds allowance. On the other hand, cable-based transmission systems do not generally suffer from these intrinsic limitations and a large part of the contribution towards this objective is due to unpredictable network effects.

(*b*) In recognition of the difficulty experienced in controlling the occurrence of severely errored seconds in propagation-limited systems, particularly during worst-month conditions, the remaining 0·1% is allocated to those types of system falling into the medium- and high-grade classifications. On the basis that adverse propagation conditions are unlikely to be experienced simultaneously over all parts of the HRX, a further 0·05% is allocated to a 2500 km hypothetical reference digital path for radio-relay systems which can be used in the high-grade and the medium-grade portion of the connection. Similarly, an additional 0·01% is allocated to a satellite link.

Based on these considerations, the apportionment to individual transmission systems reflects the practical difficulty of satisfying the severely-errored-seconds objective. The allocation to transmission systems falling into the high-grade classification is as follows:

(*a*) *Radio Systems Experiencing Worst-Month Propagation* 0·054% for a 2500 km radio-relay system on the basis that a maximum of two such radio systems will experience worst-month propagation conditions at the same time. Under normal operating conditions, which prevail for a large proportion of the time, the performance will be significantly better.

(*b*) *Satellite Systems* 0·03% for a satellite link independent of the equivalent terrestrial distance. Similarly, as with radio systems, the performance for a large proportion of the time will be significantly better.

(*c*) *Cable Systems* 0·004% for a 2500 km system. It is assumed that a major contribution towards this objective will be the unpredictable network effects. This parameter is not very meaningful in terms of influencing the design of these types of system.

MATHEMATICAL MODELS FOR DESCRIBING ERROR DISTRIBUTIONS

The network performance objectives in Recommendation G821 are expressed in a way to suit the needs of a range of different types of network users. For other users, it might be of more interest to know the quality of transmission expressed in another way. To achieve a convenient means of conversion into more meaningful terms, it is an essential prerequisite to find a mathematical model which is adequately representative of the error characteristics to be found in a future ISDN. Whether it will be possible to find a single model, adequately representative of the range of transmission systems used, is open to doubt. In an attempt to progress this aspect, many researchers are investigating the properties of a number of mathematical models that appear to be potential candidates based on operational experience to date [4, 9, 10].

The following models are currently under consideration: simple Poisson, Markov, Neyman, log-normal, gamma Poisson, and error-burst probability distribution function.

Of these, the Neyman model has looked to be a promising candidate and has provoked much debate within international fora. This model, which requires only two parameters to describe it completely, was developed by J. Neyman and used to model the distribution of lavae in a field. It was first applied in this field of engineering by W. T. Jones and G. Pullum of Standard Telecommunication Laboratories Ltd. The model can be used to describe events that occur in bursts (clusters), which themselves have a Poisson distribution with parameter M_1, and the numbers of events in bursts also have a Poisson distribution, but with parameter M_2.

MEASUREMENT CONSIDERATIONS

The objectives given above in the section on error performance specification represent the error performance as observed at the 64 kbit/s level. In practice, it is very desirable to make measurements at higher-bit-rate interfaces. Based upon certain assumptions it will be necessary to relate such measurements to the resulting performance at the 64 kbit/s level.

For example, over a long measurement period, of the order of one month, it might be realistic to assume that the percentage of 1 minute intervals with an error ratio exceeding 1×10^{-6} at the 64 kbit/s level would be the same as the percentage of 1 minute intervals exceeding 1×10^{-6} when measured at the higher bit rate.

In the case of the percentage EFS objective, the performance resulting at the 64 kbit/s level is likely to be at least as good as the percentage EFS measured at the higher bit rate. By way of example, taking the case of percentage EFS measurements at the 140 Mbit/s level, it might be appropriate to estimate the number of errors affecting each errored second and, in relating performance to the 64 kbit/s level, to weight each errored second accordingly. For example, an errored second measured at the 140 Mbit/s level and estimated to contain six binary errors could only cause, in the worst case, errored seconds in six out of the 1920 64 kbit/s signals. On the basis of similar worst-case assumptions, Fig. 4 illustrates the relationship between percentage EFS as measured at the 140 bit/s level and the number of errors contained within each errored second in order to achieve a 92% EFS objective at the 64 kbit/s level. The assumptions are only likely to be valid when averaged over long periods of time such as one month.

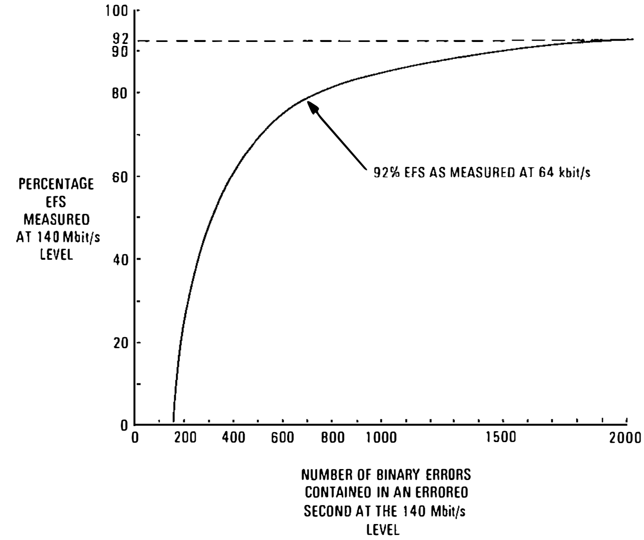


FIG. 4—Relationship between errored seconds as measured at the 140 Mbit/s level and the number of errors contained within each errored second in order to achieve a 92% EFS objective at the 64 kbit/s level

From this, it is clear that to translate accurately an error performance measurement at one bit rate to another requires, as a prerequisite, a detailed knowledge of the error characteristics. These aspects are currently the subject of study in international fora and emphasise the importance of mathematical modelling.

In formulating the error objectives in G821, it was necessary to take into account the performance characteristics of different transmission systems operating over a diverse range of media (metallic, optical fibre, radio and satellite). This was particularly crucial in the case of radio and satellite systems, in that it was necessary to recognise the infrequent occurrence of unfavourable propagation conditions, which often cause severe degradations in error performance. In cognisance of this behaviour, the error performance objectives relate to a long observation interval of the order of one month and are expressed in a manner suitable for long-term validation. Consequently, to validate whether a real connection is satisfying these objectives, it is necessary to provide monitoring for a commensurate observation period. For this reason, the performance specification method adopted (see the section on error performance specification) is clearly not suitable for the short-term (order of minutes) assessment of a connection during normal operational use. Short-term error performance assessment techniques, to judge whether a connection is performing to an acceptable standard laid down by maintenance and service personnel, are currently the subject of much debate within telecommunication administrations and CCITT.

On existing digital equipment, the 'on-board' in-service error detection facilities do not directly monitor the performance in accordance with Recommendation G821. This is not feasible because the latter is based on very long measurement periods of the order of a month. Many of the error detectors used within systems necessarily operate on a sampling basis, and have rather limited accuracy and long measurement times when implemented to detect very low error ratios. However, the evidence to date indicates that digital systems do not hover at a marginal error ratio, but tend to crash through the highest threshold whenever a fault occurs. Thus, it is believed that in the majority of cases, faults that cause the performance objectives to be exceeded would result in an alarm indication being given at the faulty system. It is envisaged that routine checks will be made, on a sample of real network connections, to confirm that the network as a whole is meeting the performance objectives that are set.

The standards for activating error alarms on equipment are based primarily on the needs of telephony, and the thresholds agreed internationally are generally as follows:

(*a*) 1×10^{-3} (this level of degradation would require immediate remedial action and the system is usually automatically removed from service), and

(*b*) 1×10^{-5} or 1×10^{-6} (this level of degradation does not necessarily require immediate attention, but gives a warning that the performance is falling below a higher standard).

As well as the performance monitors provided on individual transmission systems, an overall assessment of the performance of cascaded transmission systems can be obtained by monitoring the frame alignment signal of the transmitted signal, provided a standard frame structure is being employed. This monitoring can take place in equipment which terminates the digital link and/or in portable test equipment which can be connected at appropriate monitor points. To localise a fault which manifests itself as a high error ratio, successive measurements on maintenance entites would eventually identify the source. To achieve this objective, a comprehensive range of maintenance test equipment is available. Suitable test apparatus is discussed in Reference 11.

FUTURE STUDIES

The direction in which the CCITT studies are heading is as follows:

(*a*) Information about the performance of real digital systems is urgently needed and it is to be expected that administrations will initiate comprehensive measurement programmes to collect data in a manner which accords with, and can be compared with, the requirements of Recommendation G821.

(*b*) Network performance objectives for constituent parts of the network will have to be established; for example, a 140 Mbit/s line system. Account will need to be taken of the fact that Recommendation G821 is concerned with the performance at the 64 kbit/s level. Some means will need to be found to relate this to the requirement as measured at the 140 Mbit/s level in the case of a 140 Mbit/s line system.

(*c*) Equipment design objectives will have to be determined so that the network operator is reasonably sure that, when the equipment is installed into a real network, the equipment will meet the required network performance.

(*d*) Short-term error performance evaluation techniques for use by service and maintenance personnel will have to be formulated in order to provide a fast and efficient means of deciding whether a connection is performing to an acceptable standard.

ACKNOWLEDGEMENT

Acknowledgement is made to the Chief Executive, Trunk Services, British Telecom, for permission to make use of the information contained in this article.

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Biography

Brian Kearsey worked initially for some seven years in heavy electrical engineering with Enfield Standard Power Cables and GEC Rectifiers in the field of power distribution for the national grid and for railway traction. During that time, he obtained a first-class honours degree in electrical and electronic engineering at The City University, London. On graduating in 1973, he joined British Telecom and became a member of the division concerned with the development of digital transmission systems and equipment. Since 1979, he has been active in digital transmission network standards involving participation at CCITT, CCIR and CEPT.

Robert McLintock was awarded a first-class honours degree in electronic and electrical engineering by Manchester University in 1972. He joined the then Post Office Telecommunications Development Department as an executive engineer and worked, initially, on the development of 24-channel and, subsequently, on 30-channel PCM transmission systems. Since 1978, he has headed the group responsible for digital network transmission standards. These duties have included regular participation in some of the international standardisation committees of the CCITT and CEPT. He is currently chairman of a CEPT group concerned with the standardisation of speech encoding techniques.

Editorial note: Subsequent to ratification at the October 1984 CCITT Plenary Assembly, Draft Recommendation G81x will be renumbered Recommendation G801.

Synchronisation and Slip Performance in a Digital Network

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UDC 621.394.423 : 621.395.49

The timing of signals and processes within integrated digital networks requires careful control in order to minimise signal degradation through the occurrence of slip. This article discusses the slip mechanism and the alternative methods of its minimisation; that is, the use of high-accuracy clocks working in a plesiochronous mode or the phase locking of nodal clocks to a reference clock.

The process of relevant international standardisation activities in CCITT is discussed, particularly in relation to the specification of reference clocks, limits for tolerable slip performance and equipment design limits to cater for wander and jitter in the network.

The approach taken to timing control within the international network and to synchronisation techniques adopted by certain administrations with respect to synchronisation methodology and network topology, clock configurations and interconnecting timing links are also given.

This article is a revised version of a paper which first appeared in The Radio and Electronic Engineer and is reproduced by permission of the Institution of Electronic and Radio Engineers. This updated version takes due account of the final CCITT SG XVIII meeting during the current study period (1980–84).*

INTRODUCTION

Integrated digital communication networks comprise user terminals of various kinds, processing nodes which provide various functions including multiplexing and switching, and interconnecting transmission links. If a digital signal emanating from a particular terminal is to be faithfully transported through the multiplexing and/or switching processes to its selected destination then, not only must the physical network be carefully designed in terms of signal loss and dispersion with respect to thermal and other extraneous forms of noise, but, since digital processing is conducted in the time domain, particular attention must be paid to the timing of the components of the various processing functions with respect to the timing of the signals being transmitted.

Fig. 1 illustrates an example of signal degradation introduced when an incoming digital signal is processed by sampling with a timing signal with which it is not synchronous. Clearly, errors are introduced by the asynchronous sampling processes and bit mutilation is present, but, more importantly, the number of signal elements has been changed, in this case reduced.

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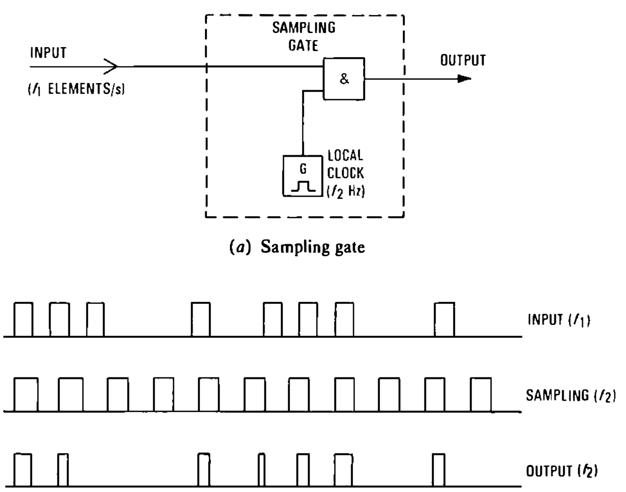
* SMITH, R., and MILLOTT, L. J. Synchronization and slip performance in a digital network. *The Radio and Electronic Eng.*, Feb. 1984, **54**(2), pp. 87–96.



(*a*) Sampling gate

(*b*) Timing waveforms

Note: Pulse mutilation and deletion of some elements in (*b*)

FIG. 1—Simple example of degradation due to timing error

Users have learnt to tolerate the presence of introduced errors in the received signal and efficient error detection and correction schemes have evolved. However, digital signals for almost all purposes, numerical or text data, digitally-encoded facsimile, speech or video and particularly error-control schemes, are arranged in blocks which are usually of fixed length. Framing signals are inserted in the stream to enable the start of the block to be determined and the various time-slots within the block are identified by maintaining a bit count as the signal progresses. The deletion or insertion of bits in the stream by processing at intermediate nodes upsets the counting process and causes framing to be lost. The recovery of valid transmission requires both the recognition that framing has been lost and its re-establishment, and the consequent loss or retransmission of all data transmitted within the intervening period.

The process of deletion or insertion of bits (or groups of bits) within a digital stream due to timing inequalities or imperfections is known as *slip*. Ideally, slip cannot be eliminated, but can be controlled by appropriate design as to the moment at which it occurs and the magnitude of the slip; that is, the number of bits lost or inserted. Its frequency can be regulated either by the use of highly accurate clocks, which minimise slip frequency, or by ensuring that all signals and processes within the network are synchronous which, in an ideal situation, eliminates slip.

This article discusses the topic of slip and its various methods of control in some detail. Sections are devoted to the slip mechanism, the effect of slip on various services, and the description and use of precision clocks and methodologies of establishing synchronised networks. A section is devoted to the status of international standardisation activities in this area within the CCITT†.

Mentioned here, but not further discussed in the article, is the concept of plesiochronous (asynchronous) multiplexing. This is a process in which the digital streams of only

† CCITT—International Telegraph and Telephone Consultative Committee

approximately the same rate can be multiplexed together to form a single high-data-rate stream for economic transmission on a wideband point-to-point bearer. The technique involves adding identifiable dummy bits to each input stream to obtain modified streams that are indeed synchronous—a process known as *justification*. These justified streams may then be processed to form a composite higher-order stream without mutilation. On reception, the high-rate stream is disassembled and the justification bits removed to recover the original data streams, each with its own independent timing. This process is useful only in the above context. Where it is desired to process sub-units of each input stream, for example, 64 kbit/s channel time-slots within a primary multiplex stream, the justification technique is inappropriate.

With the growth of network synchronisation expertise, a significant new vocabulary has been added to the lexicon of the communications engineer. To assist the reader, a glossary of selected terms has been included in an Appendix.

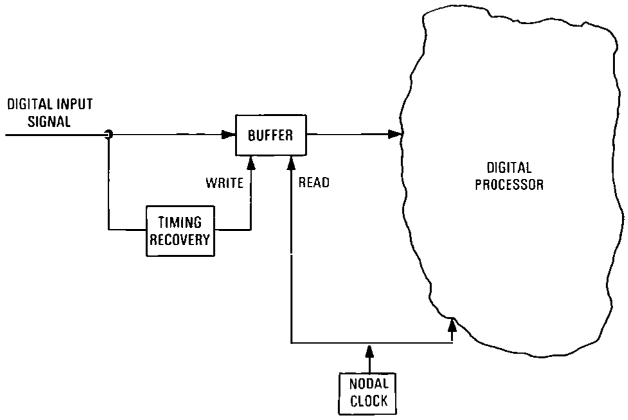


FIG. 2—Buffer store at the input of a digital processor

SLIP

As already indicated, bit mutilation can occur in a digital network when a nodal processor clock is not accurately aligned with the timing of an input bit stream. Bit mutilation is not appropriate in a practical digital network and it can be eliminated by adding a buffer store to the input of nodal processors, as shown in Fig. 2. A (sampling) clock signal is extracted from the input bit steam and is used to write incoming bits into the buffer with no possibility of bit mutilation. The (local) clock controlling the nodal processor reads bits sequentially from the store and provides the processor with a bit stream free of mutilation, whether the nodal clock is aligned with the input bit stream or not.

While buffer stores eliminate bit mutilation from digital networks, they cannot eliminate the occurrence of slips. As outlined above, slip is the repetition (insertion) or deletion of bits from a digital stream and is a result of a finite buffer underflowing or overflowing—in either case, the slip restores the state of fill of the buffer so that its normal operation can continue. The capacity of a buffer store determines the maximum number of bits that constitute a slip and, in a multiservices digital network, it is usual to design buffers so that only one or two types of slip occur[1].

(*a*) A single slip of 8 bits from a 64 kbit/s digital stream is known as an *octet slip* and represents the slip of one sample in a pulse-code modulation (PCM) encoded analogue signal[2, 3].

(*b*) A slip of one frame from a primary level digital stream is known as a *frame slip*. For networks operating with a primary rate of 2048 kbit/s, a frame slip represents simultaneous slips of one sample (that is, octet slips) in 30 PCM-encoded analogue signals[2] and for networks operating at 1544 kbit/s, a frame slip represents sample slips in 24 encoded analogue signals[3].

The choice between these two types of slip can be made by equipment designers without consideration of digital network topologies or services because the average impacts of both type of slip, in terms of bits gained or lost per unit time in any 64 kbit/s channel, is identical. The remainder of this article assumes that slips in 64 kbit/s digital channels are of octets, without explicit reference to the structure of equipment producing the slips.

The sources of slip in a digital network are varied and several sources may be present at one time. The most obvious source of slip is the use of plesiochronous clocks, which lead to the repetitive filling or emptying of buffers. Propagation delay changes in transmission systems (due to ageing or temperature variations) cause timing variations known as *wander* and are also a source of slip. Wander adds or subtracts directly to the state of fill of a buffer and, therefore, may cause a buffer to overflow or underflow and produce a slip; wander is discussed in more detail later in this article. A third source of slip is network re-arrangement and fault condition; either event can lead to transmission breaks and/or step changes in transmission delay and a slip can be generated as a result of a step change in propagation time.

The impact of slip on services supported by a multiservices digital network is independent of the source of slip, but does depend critically on the nature of the affected service[4]. Slips disrupt the framing of multiplexed digital signals and destroy bit count integrity of all digital messages, as well as causing errors. The impact of these effects varies with the details of particular services, but Table 1 summarises typical effects for the major telecommunication services. It can be seen that telephony is relatively insensitive to slips while facsimile services using simple terminals are seriously impaired. The effect of slip on data services is also shown in Table 1, a distinction being made between digital data and PCM-encoded voiceband data. Encoded voiceband data are rela-

TABLE 1
Effect of Slip on Services and Possible Slip Rate Objectives

Service	Effect of Slip	Possible Slip Rate Objective
Telephony	Only 5% of slips lead to audible clicks	1 slip per minute
PCM encoded voiceband data	Isolated errors or bursts of errors, but no data slips	1 slip per 4 minutes from data performance objectives
Digital data	For fixed block length data, two data blocks will be lost: about 80 ms. For variable block length data, up to 6 s may be lost	1 slip per hour for fixed block length data and 1 slip per 3 hours for variable block length data
Facsimile	For non-error processing systems, a slip causes fatal degradation of a page. When error processing is employed, up to two scan lines may be replaced with the preceding one	1 slip per 6 hours for non-error processing systems and 1 slip per 2½ minutes when error processing is used

tively insensitive to slip, because slip causes an analogue impairment that leads to errors but not slip in the data stream[5], whereas the sensitivity of digital data to slip can approach that of simple facsimile services. Consideration of the sensitivity of data and facsimile services to slip leads to the end-to-end performance objectives for slip in multiservices digital networks, and these are discussed later in this article.

When slip is considered in the context of network performance objectives, it is convenient to recognise two classes of slip: controlled and uncontrolled slip. A controlled slip is one in which the size of the slip and the instant of its occurrence are determined by a nodal processor. In contrast to this, uncontrolled slips are not initiated by nodal processors; they may occur at any instant and may involve the deletion or repetition of an arbitary number of bits. Network rearrangements and the switching of redundant transmission links that are not time-delay equalised are potential sources of uncontrolled slips. It is controlled slips that are the subject of international performance objectives[6].

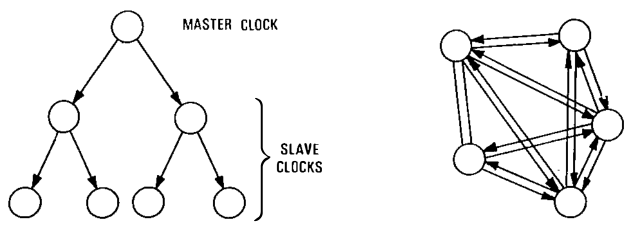
SLIP CONTROL AND METHODOLOGY

The Introduction intimates that slip rates in digital networks can be controlled by suitably restricting the range of frequencies and phases that nodal clocks can assume. It is now relevant to consider the methods available to achieve this aim and the areas of application of the various methods.

The simplest method of maintaining low slip rates on digital connections is to operate nodal clocks plesiochronously, each with a suitably small frequency departure. Clocks of this type operate independently and, therefore, do not burden network operators with the need to control and co-ordinate clocks. However, because any node in a digital network may participate in an international connection, all plesiochronous clocks must satisfy the requirements established for the international digital network. These requirements are discussed later and it is shown that plesiochronous clocks must be extremely reliable and have a long-term frequency departure of the order of 1×10^{-11}, or less. The only type of clock that at present meets these requirements is the caesium (atomic) clock and, in practice, these must be operated in groups to obtain the necessary reliability. Caesium clocks are expensive and require specialised operation and maintenance; they are, therefore, unattractive for wide-spread application in digital communications networks and their use is kept to a minimum.

A convenient alternative to plesiochronous operation is the use of synchronised clocks. With this approach, relatively low-cost phase-locked loops (PLLs)[7] based on quartz crystal oscillators function as nodal clocks and are synchronised (phase-locked) to an external reference. It is common to derive these references from a single high-accuracy source so that all clocks in the digital network adopt a common frequency, and slips arising from frequency differences are theoretically eliminated. Because phase-locked nodal clocks can provide performances approaching that of their high-accuracy references at low cost, they have a major role in digital communications networks, where they can be easily co-ordinated. Notice, however, that a penalty for using synchronised clocks is the need to establish and operate a network to interconnect the clocks. This obligation is discussed in detail later.

There are two fundamental methods of synchronising nodal clocks and these are shown in Fig. 3—they are the *master–slave* and *mutual synchronisation* techniques. The master–slave synchronisation system has a single master (reference) clock to which all other clocks are slaved (synchronised). Synchronisation is achieved by conveying a timing reference from one clock to the next via a suitable transmission system and phase-locking the slave clock to the reference. Hierarchies of clocks can be established with some



(*a*) Master–slave synchronisation (*b*) Mutual synchronisation

FIG. 3—Synchronisation techniques using phase-locked nodal clocks

clocks being slaved from higher-order clocks and in turn acting as master clocks for lower order ones. The topology of master–slave clock networks is based on an orderly hierarchy of the type shown in Fig. 3. This arrangement ensures that the network is stable (by preventing any clock receiving a reference derived from its own output) and generally matches the topology of the network of clocks with the hierarchy of exchanges in communications networks.

A mutual synchronisation system is also shown in Fig. 3 and it contrasts with the master–slave system in that all clocks are interconnected; there is no unique reference clock and there is no underlying hierarchical structure. In this system, each clock is phase-locked to a reference derived from several other clocks (and indirectly from its own output)[8]. It can be shown[8] that mutually synchronised clock systems are stable under some, but not all, conditions and that all clocks will establish a common operating frequency; this frequency is a function of the transmission delay in the network and is, therefore, difficult to predict precisely. The key features of mutually synchronised clocks of interest to communications network operators are their democratic nature (that is, they do not require a master), and their potential robustness in the presence of transmission faults.

A practical synchronisation strategy can combine mutual and master–slave techniques within the one network by the use of structured hierarchies; clocks may be mutually synchronised at one level (for example, the upper level) of the hierarchy and master–slave controlled at the lower levels. Several practical arrangements are discribed later.

INTERNATIONAL STANDARDISATION

The major forum for the development of international standards or recommendations in this field is the CCITT, specifically its Study Group XVIII, which has a study question devoted to the synchronisation of digital networks. The formal output of the study so far comprises two published Recommendations dealing with timing inaccuracies at international nodes and the establishment of slip performance objectives[1, 6]. In addition, amendments to these Recommendations and two further Recommendations dealing with the required tolerance to jitter and wander at network nodes are currently being prepared for ratification at the 1984 Plenary of CCITT[9, 10, 35, 36].

These Recommendations have thus evolved over a number of years, Recommendation G811, dealing with the specification of high-accuracy clocks at nodes terminating international plesiochronous links, being initially ratified in 1976. However, they have evolved, in the main, in anticipation of network development and, consequently, without the benefit of actual network experience. It is only now that a number of telecommunication authorities are designing and beginning to implement multiservices digital networks, and this is leading to a further round of improvement to the Recommendations so that they can be more easily interpreted and be more consistent with each other. Nevertheless, the content of the existing texts forms a useful basis for ongoing development within Study Group XVIII.

Plesiochronous Clocks for International Operation

Recommendation G811 was initially produced in the days when the integration of services within a single ubiquitous network was seen by many to be extremely futuristic. Specialised networks, notably for computer communication purposes, were being rapidly developed and, for those concerned with such networks, the presence of slip was an anathema. Considerable effort was put into the study of mutual synchronisation in the hope that a global network of mutually synchronised clocks could be established. This would have been politically acceptable, in that no country or administration would have been subservient to another in the matter of network timing, and technically acceptable to the data communicators and others in that slip theoretically would be eliminated. In the event, the doubts concerning the technical viability of an extensive global mutually synchronised network, combined with an acceptance of limited slip occurrence, led to the recommendation that, at least for the present, international communication links would be operated plesiochronously.

Recommendation G811 specifies the timing to be associated with nodes terminating international links. It is based upon the performance available from the reference clock using caesium technology, either located at the node, or which remotely synchronises a slaved clock located at the node. The latest amendment[36] specifies the performance of the reference clock itself and also the timing of the signal outgoing from the nodes in terms of the absolute time-interval-error (TIE) of the signals. It provides for a mean long-term slip rate per plesiochronous link, in any 64 kbit/s channel, of not more than one slip in approximately 70 days as implied by a long-term frequency departure of 1×10^{-11}. The current specifications are illustrated in Fig. 4.

The second major content of Recommendation G811 deals with the reliability of reference and derived clocks. Some tentative values for unavailability and degradation of clocks are given, but these are only provisional and require further study. (Indeed, they may be more severe than is necessary.) The major difficulty is that the primary performance parameter as seen by the user is slip rate, which is directly related to the frequency departure of the clock, but which in turn (under degraded or unavailable conditions) is a function of a number of interrelated parameters, including mean-time-between-failures (MTBF), mean-time-to-repair (MTTR),

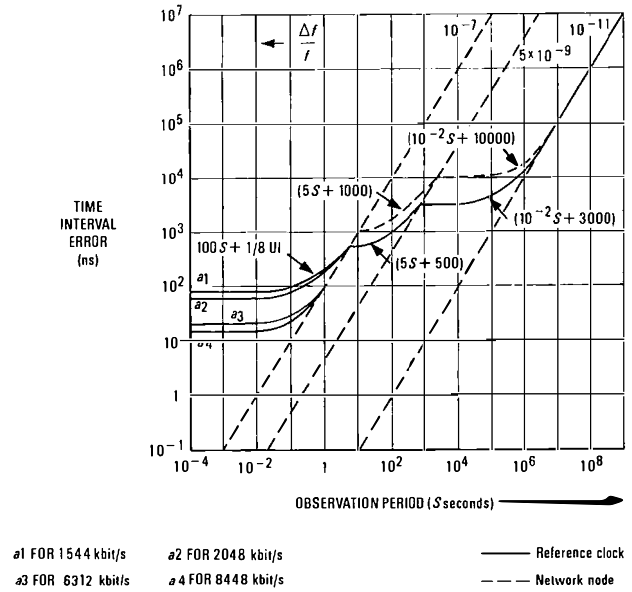


FIG. 4—Graph of permissible time interval error (TIE) against observation period for reference clocks and output signals of international nodes

the natural stability of the clock and the use of memory facilities in the clock control circuitry. Various combinations of values of these parameters would produce a satisfactory performance objective. However, some administrations see these parameters as relating to national maintenance philosophy and are, therefore, matters of individual national selection.

CCITT studies into the reliability of clocks have been considerably aided by Study Group 7 of the CCIR†, which has responsibilities for the provision of standard time and frequency reference signals around the world. In particular, a useful quantity of statistical information on the reliability of clocks using various technologies has been recently supplied to the CCITT[11].

Recommendation G811 also refers to the timing to be used in digital satellite links using time-division multiple-access (TDMA) techniques. It recommends that the satellite system be operated in a plesiochronous mode by using high accuracy timing that is either provided locally or derived from a suitable national network timing reference. This method of working has been accepted as preferable by CCIR Study Group 4.

Some doubt remained for a while concerning the timing arrangements for satellites providing on-board circuit switching. This has now dissipated as it is believed that sufficiently accurate on-board clocks can be established.

International Slip Rate Performance Objectives

Recommendation G822 attempts to specify performance objectives for controlled slip rate on an international digital connection. In most cases, it is expected that international connections in the future will comprise one or more plesiochronous links interconnecting two separately synchronised national networks. Assuming that there will be a maximum of, say, four cascaded international links, and having regard to Recommendation G811, a nominal overall slip rate of about one slip in 17 days would be attained. If both national networks are also plesiochronous, the slip rate might increase to, say, one slip in about six days. However, in a practical end-to-end connection, the slip rate may significantly exceed the value computed above, owing to various design, environmental and operational conditions in both the international and the national portions. These include:

(*a*) configuration of the international digital network,
(*b*) national timing control arrangements,
(*c*) wander due to extreme temperature variations,
(*d*) operational performance characteristics of various types of switches and transmission links (including diurnal variations of satellite facilities), and
(*e*) temporary disturbances on transmission and synchronisation links (network rearrangements, protection switching, human errors, etc.).

A threshold of satisfactory slip performance is a suitable compromise between desired service requirements and normally achievable performance. Provisional values of slip performance are contained in Recommendation G822 and are reproduced in Table 2. In performance category (*a*), all services should be transmitted without significant impairment. In the range of category (*b*), transmission of some services may be considered degraded, while other services, for example, speech, will be essentially unimpaired. Category (*c*) reflects an unacceptable performance level for all services.

Recommendation G822 also attempts to subdivide the end-to-end performance of the international connection between its constituent parts. At present, the agreed subdivision is:

† CCIR—International Radio Consultative Committee

TABLE 2
Performance Objectives for Slip Rate on a 64 kbit/s International Connection

Performance Category	Mean Slip Rate	Proportion of Time
(*a*)	≤ 5 slips in 24 hours	> 98·9%
(*b*)	> 5 slips in 24 hours and ≤ 30 slips in 1 hour	<1·0%
(*c*)	> 30 slips in 1 hour	<0·1%

Total time ≥ 1 year

(*a*) subscriber to primary centre (local portion),
(*b*) primary centre to international gateway (national transit portion), and
(*c*) international portion.

Values for each portion have been included.

Jitter and Wander Tolerance Specifications for Equipment

The third major activity of CCITT Study Group XVIII in the area of network synchronisation is the development of Draft Recommendations G82x and G82y*. These recommendations deal with the amount of wander and jitter which may be reasonably encountered within digital networks. Of specific interest as far as synchronisation is concerned is the determination of the amount of wander that network nodes must tolerate without introducing slip.

Within a synchronised network, input circuitry at nodes should accommodate short-term clock variations, together with any jitter and wander introduced by the transmission plant. The input equipment, therefore, will be similar to that described earlier when discussing slip and will include, *inter alia*, timing-recovery circuitry of relatively low time constant to generate a sampling clock (which, in effect, will follow wander and low-frequency jitter imposed on the input signal and, therefore, allow satisfactory sampling), and a following buffer store which should be sufficiently large to absorb the low-frequency wander and jitter. Consequently, under normal synchronised conditions, slip will not occur.

The level of wander and jitter which, according to the draft Recommendation G82x and G82y, must be accommodated is given in Fig. 5 and Table 3. These levels are an interesting mixture of equipment capability at the higher frequencies and network requirement at the very low frequencies. Specifically, the value A_2 relates to the effective horizontal eye opening at the sampling point of digital line equipment required for satisfactory performance, f_3 relates to the cut-off frequency of the timing-recovery circuitry, and the 20 dB/decade slope immediately below f_3 reflects the typical single-pole nature of that cut-off. Similarly, the values A_1, and f_1 represent equipment limitations found in asynchronous (or justification type) demultiplexers. On the other hand, at the low-frequency end, A_0 is a value chosen to exceed the wander likely to be introduced by seasonal environmental variations. While some calculations indicate that a value of 18 μs is insufficient for A_0, wander exceeding this value will cause only an additional two slips per year. Little is documented about the values of wander at the mid-frequencies encountered in the network (although a value of 1·5 unit interval seems conservative).

* Subsequent to formal ratification at the October 1984 CCITT Plenary Assembly, these Draft Recommendations will be renumbered G823 and G824, respectively.

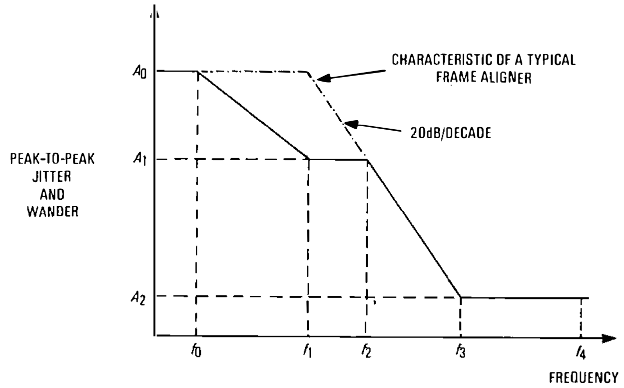


FIG. 5—Mask of peak-to-peak jitter and wander which must be accommodated at the input of a node in a digital network

TABLE 3
Values for the Mask of Fig. 5

	2048 kbit/s	8448 kbit/s	1544 kbit/s
A_0 (μs)	18	18	18
A_1 (UI)	1·5	1·5	2
A_2 (UI)	0·2	0·2	0·05
f_0 (Hz)	$1 \cdot 2 \times 10^{-5}$	$1 \cdot 2 \times 10^{-5}$	$1 \cdot 2 \times 10^{-5}$
f_1 (Hz)	20	20	10
f_2 (Hz)	$2 \cdot 4 \times 10^3$	400	200
f_3 (Hz)	18×10^3	3×10^3	8×10^3
f_4 (Hz)	100×10^3	400×10^3	40×10^3

Note: UI = Unit Interval
For 1544 kbit/s systems 1 UI = 648 ns
For 2048 kbit/s systems 1 UI = 488 ns
For 8448 kbit/s systems 1 UI = 118 ns

For any of the reasons mentioned earlier, in the discussion of Recommendation G822, the TIE between the incoming signal and the local clock may exceed the capacity of the buffer store and a controlled slip will result. This leads to two further design factors. Firstly, the buffer store itself, or the subsequent equipment, must have sufficient additional storage capacity at least equal to the magnitude of the slip required; that is, octet or frame. Secondly, once a slip has occurred, it is desirable that short-term reversal of the TIE shortly afterwards should not cause another slip. This possibility can be recognised by considering a buffer having a storage capacity exactly equal to the magnitude of a slip. Immediately after a slip, the buffer will be empty if the TIE had been increasing, and any decrease in TIE will immediately cause another slip as the buffer refills to accommodate the change. This situation contrasts with that resulting from a continuing increase in TIE, where a TIE change at least equal to the buffer capacity must occur between successive slips. In order to prevent such a slip, the equipment should incorporate a suitable hysteresis for this phenomenon. The size of this hysteresis has been set at a value equal to A_0.

The special case of the wander introduced by a satellite link is interesting. The time delay of a transmission path via a geostationary orbit may have a daily peak-to-peak cyclic variation in the order of 0·15 to perhaps 1·5 ms, owing to orbital perturbations. Longitudinal variation of the satellite about its nominal position can cause additional longer-term cyclic variations of about the same order of magnitude. CCIR Study Group 4 has consequently indicated the possibility of slipping in multiframe blocks (16 frames) rather than in single frames[12]. The magnitude (2 ms) of the necessary buffer stores should more than cater for the satellite path variation and, because of this large buffer, could reduce the frequency of slips to less than that of predicted circuit outage[12].

SYNCHRONISATION OF NATIONAL DIGITAL NETWORKS

Each digital network operator has considered slip control and, while each national network is unique, there is a commonality of requirements that leads to similar national slip control strategies. To the knowledge of the authors, all national digital telecommunications networks are, or are planned to be, synchronised; that is, they are timed by synchronised clocks. Some of the common features of these networks are outlined below.

National Reference Clocks

National digital networks require only a small number of caesium reference clocks that meet international requirements[1], as all the other clocks in the network can be phase-locked to them. Many networks, including those in the USA[14], the UK[15], Italy[16], and Australia[17], have a single national reference clock (NRC) of the type illustrated in Fig. 6. The clock is (usually) triplicated to provide high reliability because of the relatively short lifetimes of caesium oscillators[18]. The three oscillators operate independently and each normally provides an output suitable for timing international digital communications. The output from one of these oscillators (suitably processed by a frequency synthesiser) is selected by a switch (which may be redundant itself) to provide the clock output. The operation of the switch is determined from a combination of measurements on oscillator status, relative oscillator frequencies and phases. An array of frequency synthesisers usually follows the switch to generate reference signals suitable for transmission in a communications network.

Several national digital networks have more than one reference clock. The Swiss network, for example, is to be operated as three synchronised regions with a reference clock controlling each region independently of the others[19]. Each of the reference clocks is to be based on a single caesium oscillator with triplication being provided geographically to enhance the reliability of network timing. Canada is another country to use more than one reference clock; its size and the interaction of its operating companies makes two NRCs appropriate. An East and a West synchronised region will be established by operators belonging to the TransCanada Telephone System and each region will be timed by independent reference clocks[20]; the clocks will incorporate three and two caesium oscillators respectively.

As the design and application of NRCs varies amongst digital networks, so does the environment in which they operate. Many reference clocks are operated as integral parts of selected digital exchanges while, in other cases, they are operated from special laboratories. This later approach offers the advantages of a highly stable environment for the caesium oscillators and ready availability of skilled staff, and allows digital network timing to be controlled by facilities providing other precision timing and frequency control functions in national communications networks. Examples of special 'time and frequency laboratories' can be found in Australia[21], the Federal Republic of Germany[22] and the USA[23].

National Synchronisation Network Topologies

National communications networks have hierarchical structures and it is appropriate for the network of synchronised clocks controlling these networks to be generally similarly arranged. A network of clocks and interconnecting transmission links is known as a *synchronisation network* and, to the knowledge of the authors, all national synchronisation networks have the hierarchical structure shown in Fig. 7. The network is divided into hierarchical layers with all clocks in any layer having an equal status (which is reflected in their reliability and accuracy), which is above that of clocks in the layer below and below that of clocks in the layer above. The hierarchy is headed by an NRC and is completed by numerous low-cost clocks in individual digital equipments. Master–slave synchronisation is employed between layers of the clock hierarchy in strict hierarchical order and all timing is ultimately derived from the NRC. The details of how the clocks are synchronised vary from network to network, but there are only two major techniques.

The master–slave synchronisation technique can be used directly to slave a clock in one hierarchical layer from another in the same layer or from a clock with higher status. However, the simple network of Fig. 3 is vulnerable to faults in the transmission links interconnecting clocks; outages cause frequency errors and, therefore, slips. This limitation can be overcome by replicating the links interconnecting the clocks so that at least two simultaneous transmission system failures are required to disrupt the operation of a synchronisation network. This practice is used in the upper hierarchical layers of all national synchronised networks to achieve a high reliability. Fig. 8 shows an example of a practical network with mainly duplicated transmission links, following diverse routes where possible, between the clocks. Only one of the synchronisation references available to any clock is selected as the working reference and the details of this process vary from network to network. A common practice is to select from the available references, which have been ranked in order of priority by the network designer, on the

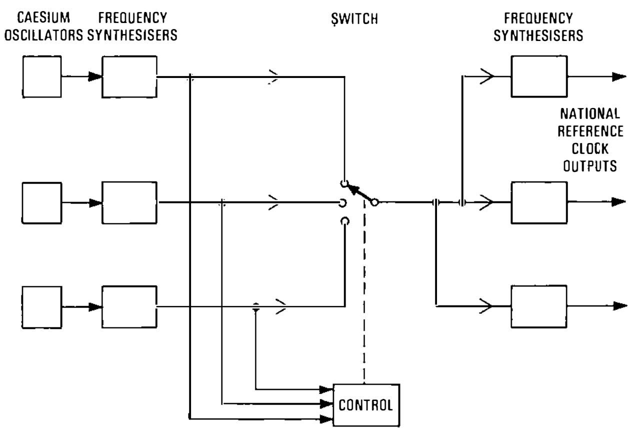


FIG. 6—Typical realisation of a national reference clock (NRC)

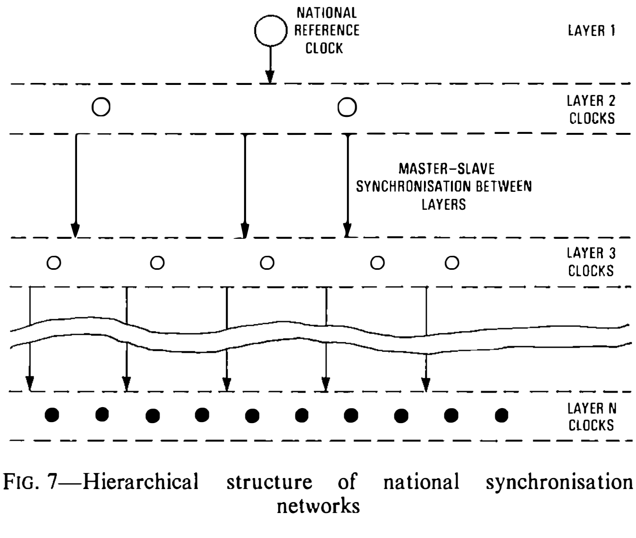


FIG. 7—Hierarchical structure of national synchronisation networks

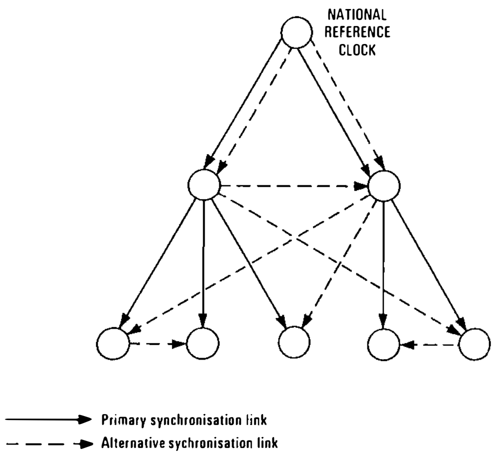


FIG. 8—Practical national synchronisation network using master–slave techniques combined with alternative synchronisation links

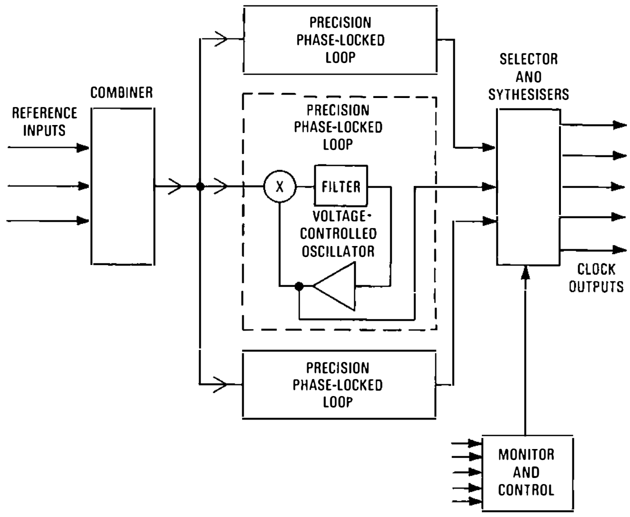


FIG. 9—Typical topology of a synchronised clock with high national status

basis of knowledge derived only from the received signals; for example, their presence or absence, relative frequency differences, etc.

These simple schemes do not require communication between the clocks and examples of their application can be found in Canada[20], the USA[24], Australia[17], Japan[25], France[26], and Switzerland[19]. A more sophisticated approach to the selection of references from replicated transmission links can be found in Italy[16] where some clocks in the upper hierarchical layers of that network exchange information about link status and select links on the basis of global network knowledge. This network is therefore relatively complex, but offers the potential to adapt automatically to unforeseen circumstances.

Another approach to overcoming the weaknesses of a simple master–slave national synchronisation network has been adopted in the UK[27]. Master–slave synchronisation is used between network layers, but some layers consist of a mutually synchronised mesh of clocks. This combination of master–slave and mutual synchronisation techniques is claimed to improve clock accuracy and the ability of the network to withstand transmission faults.

Synchronised Clocks for National Networks

Synchronised clocks used for timing digital networks are based on PLLs, incorporating quartz crystal oscillators. The quality of the oscillators, and the degree to which they, and other clock components, are replicated, depends on the status of the clock in the national synchronisation network. Clocks with a high status are used to time large segments of the digital network and, therefore, are likely to contain several precision oscillators, while clocks at the bottom of a clock hierarchy consist of a single low-cost phase-locked oscillator.

A likely topology for a clock with a high status in a national synchronisation network is shown in Fig. 9. The available incoming timing links are combined by using a switching or phase-adding technique to produce a single reference for the following PLLs. Each PLL incorporates a precision quartz crystal oscillator, operated in a carefully controlled environment, a digital phase comparator and a digital low-pass filter; the oscillators typically have a fractional ageing rate of 1×10^{-10}/day and the filter has a time constant of 1000 s or so. The PLL outputs are then selected so that the clock outputs can be generated by frequency synthesisers. Triplication of clock components is shown in Fig. 9, and this degree of replication is popular because it is the minimum that allows majority voting in signal selection processes. However, clocks using two and even four oscillators are presently being applied in digital networks. Examples of precision synchronised clock designs can be found elsewhere[25, 28, 31, 32, 33].

The complex precision clock shown in Fig. 9 can be contrasted with the simple clock, suitable for the bottom of a national synchronisation hierarchy, shown in Fig. 10. A single timing reference controls a simple PLL, incorporating an elementary phase detector, an analogue low-pass filter and a low-cost quartz crystal oscillator operated without environmental stabilisation. This type of clock is normally designed and supplied as a minor part of digital communications equipment and is subject to only simple specifications[2,3].

Between the extremes discussed above, clocks for digital networks have been developed with varying degrees of precision and redundancy. In all cases, the clocks are designed to meet the requirements imposed by respective hierarchical national synchronisation networks.

TRANSMISSION LINKS FOR SYNCHRONISATION

Timing links between clocks in a synchronisation network can be provided on either analogue or digital transmission facilities. Analogue facilities are utilised by having a reference tone inserted (or two tones where frequency translations must be overcome) onto a transmission system and filtering it out at a remote location where it may be used to phase lock a clock[21]. When used in this way, analogue transmission

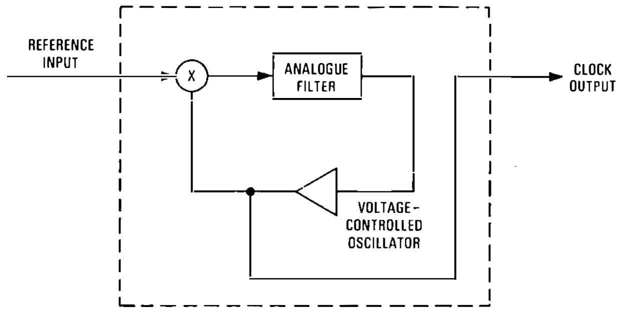


FIG. 10—Typical topology of a synchronised clock with low national status

systems must have low noise and wander performance, and must be very reliable. Examples of analogue links in synchronisation networks can be found in the USA[29] and the Federal Republic of Germany[28]. Digital transmission facilities can also be used to convey timing references between synchronised clocks and the most widely used technique is to use the reference signal to time a digital transmission system (via a network node) and to extract this timing information at a remote location[30]. This method of timing distribution does not interfere with the traffic-carrying function of digital transmission systems and is used to some extent in all national synchronisation networks. Digital transmission systems for this application are selected for high reliability, and low wander and jitter performance.

All transmission systems, whether analogue or digital, are sources of wander, which can be a serious impairment to the performance of synchronisation networks. The need to limit wander was outlined earlier and the magnitude of allowable wander eliminates simple satellite transmission systems as bearers for timing distribution. However, sophisticated and very precise time transfer schemes using satellites have been developed[34], and may find application in synchronisation networks in the future. Representative wander performances for terrestrial transmission systems are listed in Table 4, and this data is used by synchronisation network designers to guide the selection of transmission facilities so that wander is minimised. One obvious point from this Table is that pair cable is a major source of wander and special care should be taken if it is to be used for timing links to international communications facilities[9].

CONCLUSIONS

This article has reviewed developments relating to slip and slip control in digital communications networks, and it is reasonable to conclude that this is now a maturing field of endeavour.

The mechanisms leading to slip are well understood, as are the effects of slip on communications services. The recognised method of limiting the occurrence and the effects of slip is to install buffer stores at the inputs to digital nodes and to time these nodes, either from precision clocks, or from simpler clocks that are synchronised to them. In both cases, the relevant issues are being effectively dealt with: synchronised clocks of various types are in production, the suitability of transmission systems for interconnecting clocks has been demonstrated, and the CCITT has addressed slip control on international digital connections. Some work remains in the fields of national and international slip control and sychronisation, particularly in finalising performance objectives, but the progress to date and the increasing understanding of the issues suggests that this work should move to completion.

On the basis of developments to date, it can be expected that very low and well-controlled slip rates on connections in digital networks will be the norm. This is an essential step towards the establishment of multiservices digital networks and should expedite the introduction of these future communications services.

TABLE 4
Typical Wander from Terrestrial Transmission Systems

Transmission system	Bearer	Equipment
Pair Cable Plastic insulation Paper insulation	 0·3 ns/°C/km 3 ns/°C/km	0·5 ns/°C for Primary Level Digital Repeaters
Coaxial cable (2·6/9·5 mm)	0·02 ns/°C/km	0·075 degree/°C for analogue repeaters. $1 \cdot 5 \times 10^{-3}$ bit interval/°C for digital repeaters
Optical fibre (Single Mode)	0·08 ns/°C/km	As for coaxial cable with digital repeaters
Radio	Small <10 ns over 40 km hop due to severe weather	As for coaxial cable

ACKNOWLEDGEMENT

The permission of the Director, Telecom Australia Research Laboratories, to publish this article is acknowleged.

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Biographies

Roger Smith graduated with first-class honours from Adelaide University in 1956 and was awarded the degree of Master of Engineering from the same university in 1959. He has been a member of the Research Laboratories of Telecom Australia since 1958, mostly in the digital transmission field, although he included three years in digital switching studies. He is at present Deputy Director of the Research Laboratories. Mr. Smith has been involved with CCITT activities (Study Groups XV and XVIII) since 1966 and is currently the Special Rapporteur within Study Group XVIII responsible for co-ordination of the study of digital network synchronisation.

John Millott was educated at Monash University, Australia, receiving the degrees of B.Eng. in 1975 and M.Eng.Sc. in 1978 for work on data transmission. Since joining the Telecom Australia Research Laboratories in 1977, he has worked on transmission systems research, particularly in the fields of digital line systems, data transmission and digital network synchronisation.

APPENDIX
Glossary of Terms

Network synchronisation has spawned a number of terms, which may be new to the non-specialist. They have specific meaning and, in many cases, the difference between them is somewhat subtle. The following terms have been extracted from the CCITT Vocabulary[13] to assist those interested with this text and with further reading.

TIMING SIGNAL A cyclic signal used to control the timing of operations.

CLOCK Equipment providing a time base used in a transmission system to control the timing of certain functions such as the control of the duration of signal elements, the sampling etc. It is assumed that where replicated sources are used for security reasons, the assembly of these is regarded as being a single clock.

REFERENCE CLOCK A clock of high stability and accuracy that is used to govern the frequency of clocks of lower stability. The failure of such a clock does not necessarily cause loss of synchronism.

MASTER CLOCK A clock that generates accurate timing signals for the control of other clocks and possibly other equipment.

RETIMING Adjustment of the intervals between corresponding significant instants of a digital signal, by reference to a timing signal.

TIMING RECOVERY (TIMING EXTRACTION) The derivation of a timing signal from a received signal.

ISOCHRONOUS A signal is isochronous if the time interval separating any two significant instants is theoretically equal to the unit interval or to an integral multiple of the unit interval.
Note: In practice, variations in the time intervals are constrained within specified limits.

SYNCHRONOUS Signals are synchronous if their corresponding significant instants occur at precisely the same average rate. (This property had previously been defined as *mesochronous*.)

HOMOCHRONOUS Signals are homochronous if their corresponding significant instants have a constant, but uncontrolled, phase relationship with each other.

PLESIOCHRONOUS Signals are plesiochronous if their corresponding significant instants occur at nominally the same rate, any variation in rate being constrained within specified limits

HETEROCHRONOUS Signals are heterochronous if their corresponding significant instants do not necessarily occur at the same rate.

SYNCHRONISED (SYNCHRONOUS) NETWORK A network in which the corresponding significant instants of nominated signals are adjusted to make them synchronous.
Note: Ideally, the signals are synchronous, but they may be mesochronous in practice. By common usage, such mesochronous networks are frequently described as synchronised.

MUTUALLY SYNCHRONISED NETWORK A synchronised network in which each clock exerts a degree of control on all others.

DEMOCRATIC (MUTUALLY SYNCHRONISED) NETWORK A mutually synchronised network in which all clocks are of equal status and exert equal amounts of control on the others, the network operating frequency (digit rate) being the mean of the natural (uncontrolled) frequencies of the population of clocks.

HIERARCHIC (MUTUALLY SYNCHRONISED) NETWORK A mutually synchronised network in which some clocks exert more control than others, the network operating frequency being the weighted mean of the natural frequencies of the population of clocks.

DESPOTIC (SYNCHRONISED) NETWORK A synchronised network in which a unique master clock exists with full power of control of all other clocks.

OLIGARCHIC (SYNCHRONISED) NETWORK A synchronised network in which control is exercised by a few selected clocks, the remainder being controlled by these.

CONTROLLED SLIP The controlled irretrievable loss or gain of a set of consecutive digit positions in a digital signal to enable the signal to accord with a rate different from its use.
Note: Where appropriate, the term may be qualified; for example, controlled octet slip, controlled frame slip.

UNCONTROLLED SLIP The uncontrolled loss or gain of a digit position or a set of consecutive digit positions resulting from an aberration of the timing process associated with transmission or switching of a digital signal.

JUSTIFICATION (PULSE STUFFING) A process of changing the rate of a digital signal in a controlled manner so that it can accord with a rate different from its own inherent rate, usually without loss of information.

Jitter in Digital Telecommunication Networks

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UDC 621.391.88 : 621.395.49

Unless proper control is exercised in limiting the amplitude of jitter in future digital networks, serious degradations can arise. This article describes the important sources of jitter and discusses the manner in which jitter accumulates in a digital network, with particular emphasis being given to accumulation along transmission systems. A brief review is given of the current status of jitter studies within international organisations (CCITT and CEPT), and the jitter control and specification philosophy for digital networks and equipment based on the 2048 kbit/s hierarchy is outlined.*

This article is a revised version of a paper which first appeared in The Radio and Electronic Engineer‡, and is reproduced here by permission of the Institution of Electronic and Radio Engineers. This updated version takes due account of the final CCITT SG XVIII meeting during the current study period (1980–84).

INTRODUCTION

Jitter is an impairment affecting digital signals in which individual signal elements are displaced from their ideal time positions. More formally, the CCITT definition is 'Jitter: Short term variations of the significant instants of a digital signal from their ideal positions in time'. The significant instant referred to in the definition can be any fixed arbitrary point on a digital signal that is clearly identifiable, such as the leading or trailing edges, or the mid-points of pulses[1]. It is often convenient to consider jitter as an apparent phase modulation of the timing signal that determines the transitions of a digital signal; in fact, the impairment is often termed *phase jitter*. By using this concept, the modulating waveform represents jitter as a continuous time function (see Fig. 1), and such a waveform can be extracted and characterised by suitable test equipment.

It should be noted that all the systems considered in this article are self-timed, in that they rely for their operation on extracting a timing signal from the incoming digital signal itself. A frequency component corresponding to the digit rate is not usually present and the digital signal has to be operated on by a non-linear process, often full-wave rectification and slicing, in order to generate a component, which can then be extracted by a narrow-band filter.

The units in which jitter amplitudes are specified and measured depend mainly on convenience and personal choice, and can be in units of absolute time, or in terms of parts of a unit interval expressed as a fraction, a percentage or a phase in degrees. For example, an instantaneous jitter amplitude of 0·01 ms on a 1 kHz timing signal is equivalent to a jitter amplitude of 3·6° or 0·01 unit interval.

Unlike some other impairments affecting digital signals, such as errors and slips**, jitter can generally be reduced to any desired level by suitable processing, which will be described later.

THE IMPORTANCE OF CONTROLLING JITTER

If proper control is not exercised in limiting the amplitude of jitter in a digital network, then, under certain circumstances, jitter can accumulate to such an extent that the following degradations can arise:

(*a*) an increase in the probability of introducing errors in digital signals at points of signal regeneration as a result of the extracted timing signals being displaced from their optimum positions in time,

(*b*) a degradation of digitally-encoded analogue information as a result of the phase modulation of the reconstructed samples in the digital-to-analogue conversion device at the end of the connection, and

(*c*) the introduction of slips into digital signals resulting from the overflow or underflow of storage devices, or the overload of phase comparators that are used in certain types of terminal equipment (for example, jitter reducers, digital exchanges and certain digital multiplex equipment).

In degradation (*a*) outlined above, the equipment designer should take the necessary precautions to ensure that the displacement of the decision instant, determined by the extracted timing signal, does not significantly impair the noise margin. Interestingly, the most significant feature of the operation of the retiming circuitry within a regenerator

† Trunk Services, British Telecom National Networks

* CCITT—International Telegraph and Telephone Consultative Committee

CEPT—European Conference of Posts and Telecommunications Administrations

‡ KEARSEY, B. N., and McLINTOCK, R. W. Jitter in digital telecommunication networks. *The Radio and Electronic Eng.* Feb. 1984, **54**(2), pp. 70–78.

** Slip—Slip is the loss or gain of a digital position or set of consecutive digit positions, resulting from an aberration of the timing process associated with transmission or switching of a digital signal.

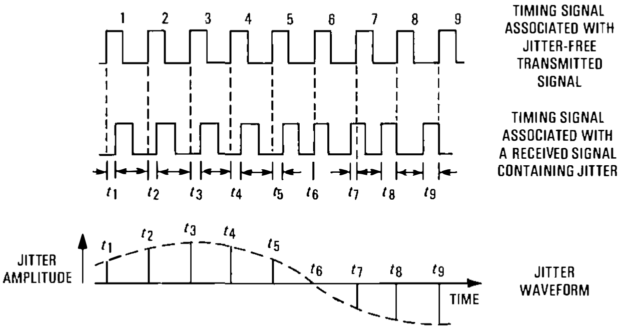


FIG. 1—Illustration of jitter and its representation as a continuous time function

is that it is not the absolute magnitude of the jitter affecting a digital signal that determines whether or not an error occurs, but rather the difference between jitter on the incoming signal and the extracted timing signal that matters; this relative measure is referred to as *alignment jitter.* Thus, to avoid errors, it is the alignment jitter that must be restricted to an acceptable level. For example, with a timing recovery circuit realised by using a simple LC tuned circuit, this can be achieved by restricting the Q-factor and the maximum frequency instability of the tuned circuit, restricting the pulse density variation of the line signal and by judicious choice of the equalisation strategy. It is generally assumed that, when the regenerators in a chain are of similar design, especially with regard to the characteristics of their timing recovery function, the jitter produced at the output of a particular regenerator can be 'tracked' by the succeeding regenerator without introducing digital errors, although the absolute jitter amplitude increases along the chain.

For the second degradation (*b*), it is necessary to limit the jitter amplitude to a level that is subjectively acceptable for most of the analogue services likely to be transported over a digital network in a digitally-encoded form. Thus, when networks are designed, it is necessary to decide upon an appropriate level, taking into account the sensitivity of particular services to jitter. For example, telephony is tolerant to fairly large jitter amplitudes, whereas television and sound broadcast-quality signals are much more sensitive. In fact, the two latter requirements are so demanding that it is not normally economic to design networks to achieve this level of jitter performance directly and special provisions are made for reducing jitter in terminal equipment associated with such services. The overall network approach generally adopted for general-purpose telecommunications networks is to constrain jitter amplitudes to a level subjectively acceptable for the telephone service.

The last degradation (*c*), is potentially the most difficult to restrict to negligible proportions, since it involves a thorough understanding of both jitter generating mechanisms and jitter accumulation laws for different types of equipment that form a digital network. To minimise the occurrences of slips, it is necessary to develop an overall network jitter control philosophy, incorporating a specification strategy for individual digital equipment. Compliance with this philosophy should result in a satisfactory performance even for the most complex connections.

SOURCES OF JITTER

Jitter observed in a digital network originates from a variety of sources, each one exhibiting its own particular characteristics.

Jitter Produced by Electronic Components

Two mechanisms account for the presence of jitter on timing signals originating from oscillators. The first is the intrinsic phase noise generated by components and is associated with contact and surface irregularities in the materials used. This noise tends to be dominated by low-frequency components and has an amplitude distribution which is Gaussian. It is often referred to as *contact noise*, *excess noise*, *flicker noise*, or *1/f noise*, the latter because of its peculiar increase towards very low frequencies[2]. The second mechanism is the phase noise in logic circuits arising as a result of signal transition uncertainties. Although both sources of jitter are measurable, they are of minor importance because of their low amplitude in comparison with other sources. A typical value for this type of jitter is a few nanoseconds RMS.

Jitter Produced by Digital Regenerators

Significant Pattern-Dependent Sources

Most of the significant sources of jitter produced by digital regenerators are strongly dependent on the pattern content of the digital signal being transmitted [3,4,5,6]. These effects, known as *pattern-dependent jitter* effects, all arise from imperfections in the different parts of a regenerator brought about by the need to produce cost-effective designs, and the need to compromise between a wide equalised channel bandwidth for good jitter performance and a narrow bandwidth for good noise immunity. These particular effects are potentially problematic because a similar degradation of the transmitted signal tends to occur at each regenerator in the chain and the jitter produced in this way can be significantly cumulative. The most significant sources of pattern-dependent jitter are caused by intersymbol interference resulting from the characteristics of the equalisation, the so-called *finite pulse width effect*, and the amplitude-to-phase conversion in the limiting amplifier associated with timing-signal extraction, all of which produce fluctuations in the time interval between zero crossings on the timing signal. The pattern-dependent jitter produced by each regenerator is restricted by design to ensure that the displacement of the timing instant does not significantly impair the noise margin.

A brief summary of the three most important pattern-dependent sources follows.

(*a*) *Intersymbol Interference* Deficiencies in the ability of a regenerator to equalise perfectly result in intersymbol interference, the extent of which depends on the pattern content of the signal. The application of a mis-equalised pulse sequence to the non-linear element of the timing recovery circuit causes a positional displacement of each pulse. Such a displacement is reflected as a phase variation in the extracted timing signal.

(*b*) *Finite Pulse Width Effects* It can be shown[3] that, following non-linear processing, the signal used to excite the tuned circuit should pass through zero when the output from the tuned circuit passes through zero if no jitter is to be introduced. Otherwise, jitter, which is dependent on both the pulse shape and the pattern content of the signal used to excite the tuned circuit is generated. This is known as *finite pulse width effect* and is a further source of timing jitter in a practical regenerator.

(*c*) *Amplitude-to-Phase Conversion* Ideally, the limiting amplifier which follows the tuned circuit should produce a squared-up timing signal that is completely independent of the amplitude of the applied input signal. Since most practical circuits exhibit voltage offsets, due to ageing and temperature effects, the output phase and pulse width are affected to a small extent by the amplitude of the applied input signal, which itself can vary according to the pulse density of the input digital signal.

Based on operational experience within British Telecom, values for pattern-dependent jitter for a single regenerator fall typically in the range 0·4 to 1·5% of a unit interval RMS, for all digit rates.

Other Sources

Jitter sources within a regenerator that are not strongly dependent on the transmitted signal are often termed *random jitter sources*. The most important sources of random jitter result from tuned-circuit mistuning (caused by environmental and ageing effects or incorrect initial adjustment), signal crosstalk from other digital systems operating in the same cable, and differential pulse delay in the output drive circuitry of regenerators.

Usually, the jitter produced at each regenerator by these mechanisms is not highly correlated and, although cumulative, the rate of accumulation is less than for pattern-dependent jitter.

The three most important sources of random jitter are summarised below.

(*a*) *Differential Pulse Delay* In the output stages of regenerators, it is common for positive and negative pulses to go through separate physical paths. The output drive circuitry usually involves the driving of transistors into saturation, the turn on/turn off delay times being primarily a function of junction capacitance. Consequently, any variation in junction capacitance produces positional asymmetry between the positive and negative pulses at the output of the regenerator.

The effect of this impairment is to reduce the noise margin at the next regenerator in the chain. However, this jitter is mainly at high frequency and is therefore considerably attenuated by subsequent timing circuits[7].

(*b*) *Mistuning Jitter* (for timing recovery circuits using an LC tuned circuit). Mistuning of the tuned circuit causes timing displacement in two distinct ways[5, 6]. The first is a static phase shift, which is an inherent characteristic of this type of timing element, and its amplitude relates directly to the Q-factor and the degree of mistuning. This shift effectively causes a change in the transmission delay through the regenerator and, assuming the degree of mistuning is restricted, the resultant degradation is insignificant. In addition to the steady-state displacement, there is a dynamic displacement of the timing signal. When the transmitted pulse sequence is random, the RMS value of this jitter produced at each regenerator is proportional to the product of the square root of the Q-factor and the degree of mistuning. It has been reported in Reference 6 that the accumulated RMS value increases to only about twice the amount at the first regenerator along a very long system. If, as is most likely, some of the regenerators in the chain are mistuned in opposite directions or to a smaller degree, the jitter at the end is smaller than that indicated. Thus, mistuning jitter is not thought to be a serious factor in respect of accumulation if the transmitted signal is random.

(*c*) *Crosstalk* Signal crosstalk from other digital line systems operating in the same cable can introduce phase shift in the regenerator's timing signal. Since the crosstalk coupling is different from one regenerator to another, the jitter produced within each regenerator is uncorrelated and is, consequently, considered to be one of the less significant sources.

Jitter Produced by Multiplex Equipment Using Justification Techniques

With a synchronous multiplexer, the several input tributary signals and the multiplexed signal are all controlled by timing signals that are synchronised to each other, and the process of combination to form a composite multiplexed signal can be implemented without jitter being introduced at the output of the demultiplexer. However, this is not the case with multiplexers that use justification techniques since jitter can arise as a consequence of the justification process. Justification is the means of bringing together the plesiochronous input tributaries to a common digit rate and timing relationship by the controlled addition of supplementary digits in each multiplex frame. These digits can either contain information or padding bits as necessary. This process, together with the necessity of using a frame structure with justification control digits and a frame-alignment signal, causes the demultiplexed output signal to exhibit a characteristic jitter[8, 9, 10].

Two types of jitter are produced. The first is a fixed-frequency component caused by the systematic removal of the frame alignment signal and the justification control digits at the demultiplexer. The second is *justification jitter*, of which a variable low-frequency component (termed *waiting-time jitter*) is the most troublesome. Minimisation of these components at the demultiplexer output is invariably effected by using a phase-locked loop (PLL) having a narrow loop bandwidth of a few tens of hertz. Notwithstanding this, the waiting-time jitter may have components at frequencies within the pass band of the PLL and appear at the demultiplexer output. The maximum amplitude of waiting-time jitter can be shown to be the reciprocal of the denominator of the justification ratio when the latter is expressed as an irreducible ratio of two integers. For both 2048 kbit/s and 8448 kbit/s tributaries using standard multiplex equipment, a maximum waiting-time jitter of 1/7 unit interval is predicted.

Wander

One effect of temperature variation on cables is to change their propagation delay. For a symetrical pair cable over a few hundred kilometres, the apparent phase change may be of the order of some tens of unit intervals peak-to-peak during the course of a year. Clearly, this type of phase variation occurs very slowly, and the effect is more properly considered as one of *drift*, or *wander*. Satellite systems operating in the geostationary orbit can also be a source of significant amounts of wander resulting from the movement of the satellite relative to the earth over a 24-hour period.

Wander is defined as 'the longer-term variations of the significant instants of a digital signal from their ideal positions in time'. The precise distinction between jitter and wander is obscure as there is no strict definition in terms of frequency, and often the two are distinguishable by cause. Wander magnitudes can be quite large and it is often difficult to reduce it. Certain types of transmission equipment, such as digital line systems and digital multiplex equipment using justification techniques, are effectively transparent to these very low frequency changes in phase. However, it does need to be accommodated at the input of certain other digital equipment such as digital switches and synchronous multiplex equipment where buffer or elastic stores are provided for this purpose. A companion article in this issue† elaborates further on these types of equipment.

JITTER ACCUMULATION

Digital Line, Radio and Optical-Fibre Systems

With digital line, radio and optical-fibre transmission systems, the rate of jitter accumulation is strongly dependent on the pattern content of the transmitted signal, the physical implementation of timing recovery, and the inclusion, or otherwise, of a pattern-transformation device (for example, a scrambler). A number of jitter accumulation relationships are identified.

Since a digital regenerator, as far as its effect on jitter is concerned, can be shown to be equivalent to a low-pass filter, the low-frequency content of the pattern-dependent jitter injected by each regenerator accumulates at successive regenerators. Byrne *et al*[11] were able to derive the following useful relationships, based on the use of the idealised Chapman model, giving the RMS jitter accumulated along a chain of regenerators when a signal with a random signal content is transmitted.

$$J_N = \sqrt{\phi B P(N)}, \qquad \ldots\ldots (1)$$

where J_N is the RMS jitter present on the signal after N regenerators,

ϕ is the low-frequency mean square jitter amplitude injected at each regenerator,

B is the half bandwidth of the timing recovery circuit ($B = \omega_0/2Q$) where ω_0 is the angular resonant frequency, and

† SMITH, R., and MILLOT, L. J. Synchronisation and Slip Performance in a Digital Network. *Br. Telecommun. Eng.*, July 1984, **3**, p. 99.

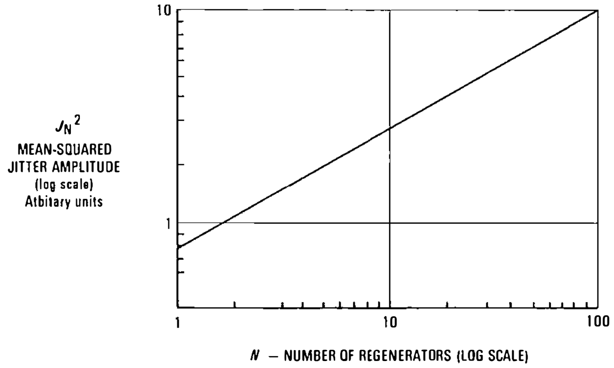


FIG. 2—Mean square jitter amplitude versus the number of cascaded regenerators

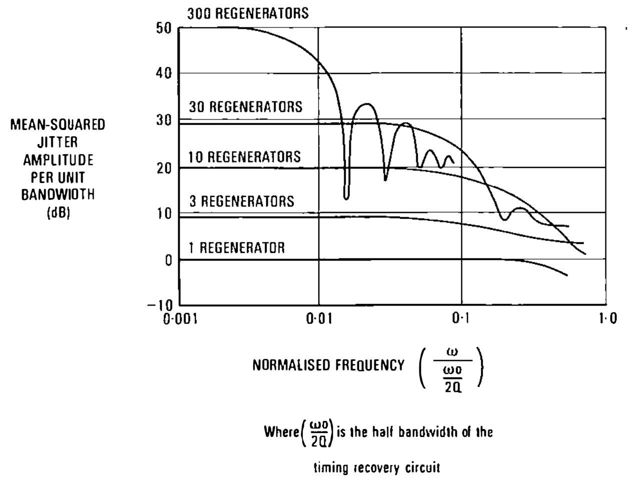


FIG. 3—A typical jitter spectrum produced by cascaded regenerators when a random digital signal is being transmitted

$P(N)$ is a factor dependent on the number of cascaded regenerators.

The use of a single-LC tuned timing recovery circuit is assumed. This relationship is illustrated in Fig. 2.

For a chain of more than 100 regenerators the following approximate relationship is valid:

$$J_N \approx \sqrt{\frac{\phi BN.}{2}} \qquad \ldots\ldots (2)$$

Equation (2), derived from Byrne's paper, is often expressed in the following familiar terms:

$$J_N \approx J\sqrt{2N}, \qquad \ldots\ldots (3)$$

where J is the effective RMS jitter injected by each regenerator.

Note: The word *effective* is used to distinguish it clearly from the absolute RMS jitter injected by a single regenerator, which is obtained by a direct wideband-frequency measurement of a single regenerator. The former primarily takes into account the predominant low-frequency pattern-dependent jitter mechanisms that play an important part in jitter accumulation, whereas the latter also includes additional contributions from the less significant high-frequency sources of jitter.

Therefore, the total RMS jitter due to the predominant pattern-dependent sources increases as the square root of N. If the jitter contributed by each regenerator is truly random, as distinct from pattern-dependent, then it has been shown that the total RMS jitter, J_N, present on the digital signal after N regenerators accumulates at a rate approximating to the fourth root of N [3, 4]. This assumes that the jitter contribution at each regenerator is uncorrelated.

$$J_N \approx J_R \sqrt[4]{N}, \qquad \ldots\ldots (4)$$

where J_R is the effective RMS jitter from a single regenerator due to uncorrelated jitter sources (random-jitter sources).

Studies based on extensive operational experience have shown that the actual RMS accumulation along transmission systems using simple LC timing-recovery circuits lies somewhere between the square root and the fourth root of the number of cascaded regenerators. This is because real digital paths are configured by using digital transmission systems originating from different manufacturers, and these systems, therefore, exhibit different characteristics. Secondly, regenerators within the same transmission system tend not to introduce an identical jitter magnitude. The resulting rather more favourable accumulation arises because each transmission system is generating jitter that is not well correlated. It is possible to take advantage of this behaviour in controlling jitter magnitudes in operational networks.

When a signal with a random information content is transmitted, the mean-square jitter amplitude per unit bandwidth is typically as shown in Fig. 3[11].

The implementation of timing recovery using a PLL causes the rate of accumulation to be marginally greater because of the effects of the low-frequency jitter gain associated with this type of timing device, and is given by the approximate relationship[12]:

$$J_N \approx J\sqrt{2NA}, \qquad \ldots\ldots (5)$$

where A is a factor dependent upon both the number of regenerators and the damping factor of the phase-locked loops.

Generally in this application, A has an amplitude only marginally greater than unity. However, the jitter J, generated by such a regenerator, tends to be smaller than that produced by a regenerator using an LC tuned circuit, by virtue of the much higher effective Q-factor.

A signal with a random information content generally produces jitter that is random in nature because pattern-dependent mechanisms predominate. Similarly, when a pseudo-random binary sequence (PRBS) is transmitted, a pseudo-random jitter waveform is produced; a typical jitter waveform is shown in Fig. 4.

The implementation of timing recovery using a transversal surface acoustic wave filter produces a rate of accumulation approaching that obtained for uncorrelated jitter sources[13]. This favourable jitter accumulation arises because of the large inherent delay of these filters, which reduces the correlation between the recovered timing signal and the input digital signal. Pattern-dependent jitter is, therefore, effectively randomised and tends to accumulate in a manner similar to that obtained from uncorrelated jitter sources. The only noticeable side-effect is a marginal degradation in the alignment jitter. This favourable jitter accumulation is not exhibited by surface acoustic wave resonators because of their different mode of operation.

Similarly, for regenerators incorporating circuitry involving a signal transformation, a favourable accumulation is achieved by virtue of the resultant signal randomisation, which effectively causes de-correlation of the jitter sources at each regenerator[14]. For example, a signal transformation based on the modulo-2 addition of a signal and its delayed version causes the RMS jitter to accumulate approximately with the fourth root of the number of regenerators.

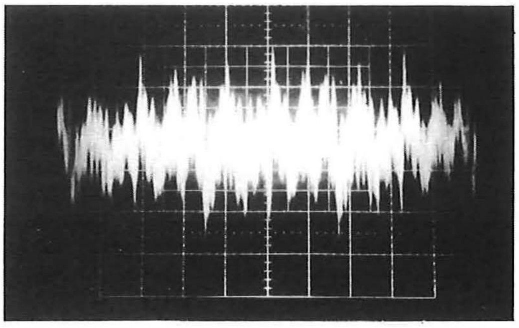


FIG. 4—A typical jitter waveform obtained when a $2^{15}-1$ PRBS is being transmitted over a 2048 kbit/s digital line system

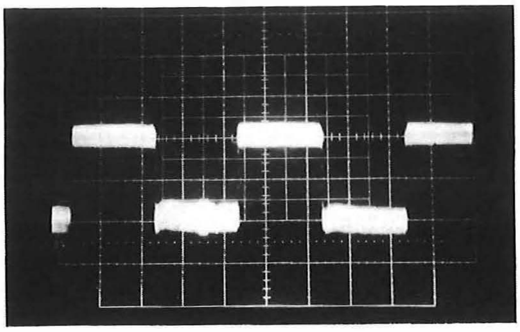


FIG. 5—A typical jitter waveform obtained when a signal, comprising two repetitive patterns (1111 and 1010), alternating at 100 Hz, is being transmitted over a 2048 kbit/s digital line system

Equations (3) and (4) demonstrate two important results:

(*a*) pattern-dependent jitter accumulates more rapidly than non-pattern-dependent jitter, and

(*b*) the amplitude of jitter produced by a chain of regenerators increases without limit, as the number of regenerators is increased.

The jitter produced by a random pattern is itself likely to be random in nature, and have an amplitude probability distribution close to Gaussian[11,15]. Hence, for a given RMS amplitude, the probability of exceeding any chosen peak-to-peak amplitude can be estimated. A peak-to-peak to RMS ratio of between 12 and 15 is often assumed for specification purposes; this is a ratio that has a low probability of being exceeded.

In contrast, when the signal being transmitted is composed of two repetitive patterns, alternating at low frequency, the jitter appears as a low-frequency repetitive wave, having an amplitude proportional to the number of regenerators. This situation can lead to very large amplitudes of jitter[11]. In such instances, the total peak-to-peak jitter amplitude (J_N) is described by the following relationship:

$$J_N = dN, \qquad \ldots\ldots (6)$$

where d is the peak-to-peak jitter produced by a single regenerator when subjected to the alternating repetitive pattern and N is the number of regenerators.

This relationship assumes that the repetition rate is sufficiently low that steady states are attained. The actual value of d is dependent on the patterns used. A jitter waveform measured under these repetitive-pattern conditions is illustrated in Fig. 5.

This situation is unlikely to occur in normal operation because the signal transmitted is generally made up of traffic from a number of different sources, although not necessarily so at the primary line rate, together with a frame alignment signal and justification control digits etc. Furthermore, the probability of fixed patterns occuring can be reduced still further by the use of digital scramblers, which are often used to randomise the transmitted signal.

The effect of the inclusion of a scrambler/descrambler combination in a digital transmission system needs to be considered when systems are connected in cascade. In such situations, the total jitter contributed by each system is uncorrelated and is, therefore, thought to accumulate in accordance with the fourth root of the number of cascaded systems. On this basis, the RMS jitter, J_M, present on the digital signal after M digital transmission systems is given by the approximate relationship:

$$J_M \approx J_S \sqrt[4]{M}, \qquad \ldots\ldots (7)$$

where J_S is the RMS jitter from a single system.
(*Note:* This relationship assumes that each system contributes an identical jitter amplitude.)

The corresponding generalised solution in which each transmission system contributes a different jitter amplitude is given by the approximate relationship:

$$J_M \approx \sqrt[4]{(J_{S1})^4 + (J_{S2})^4 + \ldots + (J_{SM})^4}, \qquad \ldots\ldots (8)$$

where J_{Sx} is the RMS jitter from the xth system

Where individual jitter reducers are provided in addition to scramblers, the same accumulation relationship is believed to apply, except that the value for J_S is likely to be significantly reduced. In such circumstances the RMS jitter, J_S, is given by the following approximate relationship:

$$J_S \approx 2NJ\sqrt{\frac{f_c}{B}} \qquad \text{for large } N, \qquad \ldots\ldots (9)$$

where J is the effective RMS jitter from a single regenerator,
N is the number of cascaded regenerators,
f_c is the cut-off frequency of the jitter reducer, and
B is the half bandwidth of the timing signal extraction circuitry of a single regenerator ($B = \omega_0/2Q$) where ω_0 is the angular resonant frequency of the timing signal extraction circuitry.

Note: The validity of the relationships (7), (8) and (9) requires further study, particularly in the case where jitter reducers are incorporated, because the degree of randomisation, produced by the length of scrambler commonly considered acceptable, might not be sufficient to ensure that the jitter components falling within the pass-band of the jitter reducer are uncorrelated to the extent that fourth root accumulation is dominant.

Multiplex Equipment Using Justification

With multiplex equipment that uses justification, the only type of jitter that is likely to accumulate to any significant extent is the variable low-frequency waiting-time jitter, which may have components at frequencies within the pass-band of the demultiplexer's PLL[16]. The expectations are that the accumulation of waiting-time jitter is at a rate between the fourth root of Y and the square root of Y, where Y is the number of cascaded multiplexer/demultiplexer pairs. Further study is required to determine a more exact relationship.

STATUS OF INTERNATIONAL STUDIES

International organisations such as CCITT and CEPT are actively engaged in studies concerning the jitter specification of digital equipment and the control of jitter in digital networks. The outcome of studies to date is enunciated in a new draft CCITT Recommendation G823 (The control of jitter and wander within digital networks which are based on the 2048 kbit/s hierarchy)[17, 18, 19]. The objective is to define a set of jitter specifications for individual systems so that they may be freely interconnected without the overall network performance being impaired.

The jitter control and specification philosophy formulated thus far for networks and equipment, based on the 2048 kbit/s hierarchy, is as follows:

(*a*) a maximum network limit has been defined that should not be exceeded at any hierarchical interface, and

(*b*) a consistent specification framework for individual digital equipment (for example, digital multiplex, and digital line, radio and optical-fibre systems) has been established.

Network Limit for the Maximum Output Jitter at any Hierarchical Interface

The limits given in Table 1 represent the maximum permissible levels of jitter at hierarchical interfaces within a digital network. The limits should be met for all operating conditions and regardless of the amount of equipment preceding the interface. These network limits are compatible with the minimum tolerance to jitter that all equipment input ports are required to provide. It should be noted that the limits for the 64 kbit/s interface are provisional and only applicable to the co-directional interface described in CCITT Recommendation G703[22].

A maximum network limit for wander at an hierarchical interface has not been defined. Actual magnitudes of wander, being largely dependent of the fundamental propagation characteristics of transmission media and the ageing of clock circuitry, can be predicted. Studies have shown that, provided input ports can tolerate wander in accordance with the input tolerance requirements discussed later, then the slip rate, introduced as a result of exceeding the input tolerance, will be insignificant.

It is assumed that, within a synchronised network, digital equipment provided at nodes will accommodate permitted phase deviations on the incoming signal together with jitter and wander from transmission plant; that is, under normal synchronised conditions, slip will not occur. However, it should be recognised that, as a result of some performance degradations, failure conditions, maintenance actions or other events, the relative time interval error (TIE) (see CCITT Recommendation G811) between the incoming signal and the internal timing signal of the terminating equipment may exceed the wander and jitter tolerance of the equipment, which will result in a controlled slip.

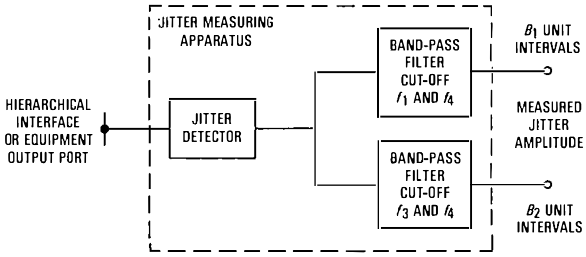


FIG. 6—Measurement arrangements for output jitter from a hierarchical interface or an equipment output port

The arrangements for measuring output jitter at a digital interface are illustrated in Fig. 6. The CCITT has defined the detailed characteristics of suitable test equipment in CCITT Recommendation O171[20]. The specific jitter limits and values of filter cut-off frequencies for the different hierarchical levels are given in Table 1. The frequency response of the filters associated with the measurement apparatus should have a roll-off of 20 dB/decade.

In this article, space does not permit a full explanation as to why the specification philosophy and associated limits were formulated. An insight can be obtained in Reference 21.

Jitter Specification Philosophy for Digital Equipment

For individual digital equipment, the approach adopted is to specify the jitter performance in three ways:

(*a*) the jitter tolerance of digital input ports,

(*b*) the maximum output jitter in the absence of input jitter, and

(*c*) the jitter transfer characteristic measured between input and output ports.

Jitter and Wander Tolerance of Digital Input Ports

To ensure that any equipment can be connected to any recommended hierarchical interface within a network, it is necessary to arrange that the input ports of all equipment are capable of accommodating levels of jitter up to the maximum network limit defined in Table 1.

For convenience of testing, the required tolerance is defined in terms of the amplitude and frequency of sinusoidal jitter which, when modulating a test pattern, should not cause a deterioration of system performance below specified

TABLE 1
Maximum Permissible Jitter at a Hierarchical Interface

Digit rate (kbit/s) \ Parameter value	Network limit		Measurement filter bandwidth Band-pass filter having lower cut-off frequency f_1 or f_3 and an upper cut-off frequency f_4		
	B_1 unit interval peak-to-peak	B_2 unit interval peak-to-peak	f_1	f_3	f_4
64†	0·25	0·05	20 Hz	3 kHz	20 kHz
2048	1·5	0·2	20 Hz	18 kHz (700 Hz)‡	100 kHz
8448	1·5	0·2	20 Hz	3 kHz (80 kHz)‡	400 kHz
34368	1·5	0·15	100 Hz	10 kHz	800 kHz
139264	1·5	0·075	200 Hz	10 kHz	3500 kHz

† For co-directional interface only (provisional).
‡ The frequency values shown in parentheses only apply to certain national interfaces used by some other Administrations.

TABLE 2
Parameter Values for Input Jitter and Wander Tolerance

Digit rate (kbit/s) \ Parameter value	Peak-to-peak amplitude in unit interval			Frequency					Pseudo-random test signal
	A_0*	A_1	A_2	f_0	f_1	f_2	f_3	f_4	
64†	1·15 (18 μs)	0·25	0·05	$1{\cdot}2\times10^{-5}$ Hz	20 Hz	600 Hz	3 kHz	20 kHz	$2^{11}-1$
2048	36·9 (18 μs)	1·5	0·2	$1{\cdot}2\times10^{-5}$ Hz	20 Hz	2·4 kHz (93 Hz)‡	18 kHz (700 Hz)‡	100 kHz	$2^{15}-1$ (Rec. O 151)
8448	152 (18 μs)	1·5	0·2	$1{\cdot}2\times10^{-5}$ Hz	20 Hz	400 Hz (10·7 kHz)‡	3 kHz (80 kHz)‡	400 kHz	$2^{15}-1$ (Rec. O 151)
34368	×	1·5	0·15	×	100 Hz	1 kHz	10 kHz	800 kHz	$2^{23}-1$ (Rec. O 151)
139264	×	1·5	0·075	×	200 Hz	500 Hz	10 kHz	3500 kHz	$2^{23}-1$ (Rec. O 151)

× Values under study.
* The value for A_0 (18 μs) represent a relative phase deviation between the incoming signal and the internal local timing signal derived from the local clock. This value for A_0 corresponds to an absolute value of 21 μs at the input to a node (that is, equipment input port). The difference of 3 μs corresponds to the 3μs allowed for long-term phase deviation in the national reference clock (Recommendation G811 paragraph 3(*c*)).
† For co-directional interface only.
‡ The frequency values shown in parentheses only apply to certain national interfaces used by some other Administrations.

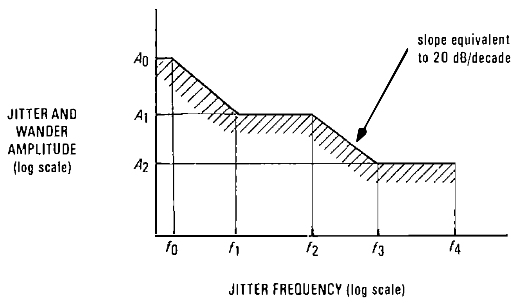


FIG. 7—Lower limit of maximum tolerable input jitter and wander

levels. It is important to recognise that the test condition is not, in itself, intended to be representative of the type of jitter to be found in practice in a network. However, the test does ensure that the Q-factor associated with timing signal recovery is not excessive and, where necessary, that an adequate amount of buffer storage has been provided.

Thus, all digital input ports of equipment should be able to tolerate a digital signal having electrical characteristics in accordance with the requirements of CCITT Recommendation G703[22] and modulated by sinusoidal jitter having an amplitude/frequency relationship defined in Fig. 7. Table 2 indicates the appropriate limits for the different hierarchical levels.

In principle, these requirements should be met regardless of the information content of the digital signal. For test purposes, the equivalent binary content of the signal with jitter modulation is usually a pseudo-random bit sequence as defined in Table 2.

In deriving these limits, the wander effects are considered to be predominant at frequencies below f_1, and many items of transmission equipment, such as digital line systems and asynchronous muldexes using justification techniques, are effectively transparent to these very low frequency changes in phase. Notwithstanding this, it does need to be accommodated at the input of certain equipment (for example, digital switches and synchronous muldexes). The requirement below f_1 is not amenable to simple practical evaluation, but account should be taken of the requirement at the design stage of the equipment.

Unlike that part of the mask between frequencies f_1 and f_4, which reflect the maximum permissible jitter magnitude in a digital network, that part of the mask below the frequency f_1 does not aim to represent the maximum permissible wander that might occur in practice. Below the frequency f_1, the mask is derived such that, where necessary, the provision of this level of buffer storage at the input of an item of equipment facilitates the accommodation of wander generated in a large proportion of real connections.

Maximum Output Jitter in the Absence of Input Jitter

It is necessary to restrict the amount of jitter generated within individual equipment. CCITT Recommendations dealing with specific systems will ultimately define the maximum levels of jitter that may be generated in the absence of input jitter. The actual limits applied depend upon the type of equipment. They should be met regardless of the information content of the digital signal. In all cases, the limits never exceed the maximum permitted network limit. The arrangement for measuring output jitter is illustrated in Fig. 6.

Jitter Transfer Characteristic

When jitter is present at the digital input port of an equipment, in many cases, some residual jitter is transmitted to the corresponding digital output port. Many types of digital equipment inherently attenuate the higher-frequency jitter components present at the input, but some equipment, notably those incorporating PLLs, tend to amplify low-frequency jitter. CCITT Recommendations dealing with particular equipment will ultimately define limiting values for the jitter transfer characteristics. To control jitter accumulation in operational situations, it is particularly important to restrict the value for the maximum jitter gain.

Jitter Specification for Digital Transmission Systems

At the May 1984 meeting of CCITT Study Group XVIII, an agreement was reached for the appropriate limits for output jitter and jitter transfer characteristic for digital line and radio systems. These limits were derived on the basis that only two digital sections would be connected in cascade and, moreover, no account was taken of jitter originating from asynchronous multiplexing equipment. However, in real network configurations likely to be encountered in the UK, it will be necessary to have more digital sections in

TABLE 3

The Maximum Output Jitter in the Absence of Input Jitter for a Digital Section up to the Length of a HRDS†

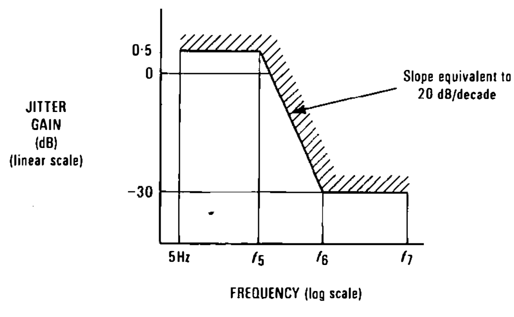
Digit rate (kbit/s)	HRDS length (km)	Maximum output jitter for digital section up to the length of a HRDS: Low frequency limit (f_1–f_4) unit interval peak-peak	High frequency limit (f_3–f_4) unit interval peak-peak	Measurement filter bandwidth Band-pass filter having a lower cut-off frequency f_1 or f_3 and an upper cut-off frequency f_4: f_1	f_3	f_4
2048	50	0·4	0·2	20 Hz	18 kHz (700 Hz)‡	100 kHz
8448	125	0·4	0·2	20 Hz	3 kHz (80 kHz)‡	400 kHz
34368	280	0·4	0·15	100 Hz	10 kHz	800 kHz
139264	280	0·4	0·075	200 Hz	10 kHz	3500 kHz

† Measurements are made in accordance with the method shown in Fig. 6.
‡ The frequency values shown in parentheses only apply to certain national interfaces used by some other Administrations.

cascade along with many asynchronous digital multiplexers. For effective jitter control in these situations, it will be necessary to satisfy more demanding limits and/or to use other means of minimising jitter (for example, by the use of jitter reducers and/or the inclusion of digital scramblers in individual systems to ensure a more favourable rate of jitter accumulation). The limits given below are those adopted by British Telecom and proposed to CCITT/CEPT[23].

To achieve favourable jitter accumulation properties it is preferred that digital scramblers should form an integral part of each system. The input jitter tolerance requirement is consistent with that detailed in Table 2. The jitter transfer characteristics should meet the requirements of Fig. 8. The maximum permissible output jitter in the absence of input jitter is given in Table 4. It is intended that these limits for digital sections should be met by all sections, regardless of length and the number of regenerators, provided that they are not longer than the hypothetical reference digital section (HRDS) models, which are assumed to have lengths as shown in Fig. 8. For longer sections, which are likely to be rare, larger limits might be necessary.

During equipment design and validation, it is important to note that the limits are intended to be met regardless of the transmitted signal. In many situations, it is often convenient to evaluate an equipment by using a specific deterministic test signal, such as a standard readily-available pseudo-random sequence.



Digit rate(kbit/s)	Frequency: f_5(Hz)	f_7(kHz)	Length of HRDS (km)
2048	40	100	50
8448	100	400	125
34368	300	800	280
139264	500	3500	280

FIG. 8—Jitter transfer characteristic for a digital section up to the length of a HRDS

METHODS OF MINIMISING JITTER

There are essentially three basic methods of minimising jitter generation and accumulation in a digital connection[13, 14, 24, 25]. The first method involves the use of a pattern-transformation device (for example, a digital scrambler), which, by effectively randomising the signal, ensures minimal jitter generation and accumulation due to the predominant pattern-dependent mechanisms within a regenerator. The second method involves the use of a jitter reducer, which is basically a retiming circuit with a jitter pass-band that is small (for example, a few tens of hertz) compared with the jitter bandwidth of the digital signal. The occurrence of jitter at frequencies within the pass-band of the jitter reducer is not usually too troublesome because many digital equipment are transparent to it or else, in the case of digital exchanges, buffer stores (that is, frame aligners) are able to cope with it. The third method involves the choice for realising timing-recovery circuits in regenerators and the significance of this factor has been illustrated earlier in the section on jitter accumulation.

THE EFFECT OF JITTER ON DIGITALLY-ENCODED ANALOGUE SIGNALS

A large proportion of services that will be carried by a digital telecommunications network originate in an analogue form; for example, telephony, sound-programme and television signals. The presence of jitter on the digital signal applied to a digital-to-analogue converter causes the decoded analgue signal samples to deviate from their ideal time positions, thus introducing distortion into the reconstructed signal. An assessment of the degree of degradation likely to be experienced by an analogue signal is complex if one takes into account the jitter characteristics likely to be encountered in practice. Recognising the mathematical difficulties of analysing this consequential distortion, most evaluation studies tend to be subjective in nature. Notwithstanding this, a useful relationship for evaluating the distortion in digitally-encoded frequency-division multiplex (FDM) assemblies has been developed[5]. The application of this relationship to pulse-code modulation (PCM) telephony and 625-line PAL colour television signals gives results that compare favourably with those derived by direct subjective testing[10].

The seriousness of this effect varies widely with the type of analogue signal. To give an idea of the sensitivity of a particular analogue signal to jitter, the values that follow have been extracted from a number of references.

For telephony signals, encoded by PCM, a permissible level of jitter of 1·4 μs RMS (approximately 3 unit interval RMS at 2048 kbit/s) has been reported[10].

The subjective effect of jitter on digitally-encoded FDM assemblies (for example, supergroups, hypergroups) is critically dependent on the frequency spectrum of the jitter. For example, jitter components above about 2 kHz can be troublesome because they can cause distortion products to fall into adjacent channels, causing noise even though a signal may not be present in that channel. Jitter components below about 1 kHz are less troublesome, as distortion products are contained within the channel and the jitter distortion will occur only in the presence of a signal. Analytical studies indicate permissible RMS jitter magnitudes of 35 ns, 6·9 ns and 0·95 ns for group, supergroup and hypergroup assemblies, respectively[10].

In the case of broadcast-standard sound programmes and television signals, the effect of jitter on quality have been investigated by the British Broadcasting Corporation (BBC) using both sinusoidal and random jitter[26,27]. Since, for both types of signal, the subjective effect depends significantly on the nature of the programme material being conveyed, the establishment of a firm value for the permissible jitter can be difficult. However, the following levels have been suggested for 'critical' sound-programme signals—a maximum jitter amplitude of 35 ns RMS for jitter frequencies above 2 kHz, a limit of 3·5 ms RMS at very low frequencies (below 0·02 Hz), and a limit inversely proportional to jitter frequency in the intermediate frequency range 0·02 Hz to 2 kHz. As far as television signals are concerned, the part of the system most sensitive to jitter is the PAL colour decoder in the television receiver. The degradation manifests itself in terms of errors in colour saturation and hue. Assuming that the PAL decoder is perfectly aligned, a permissible level of peak-to-peak jitter is 20 ns for random jitter. As a target specification, 5 ns peak-to-peak for random jitter has been suggested by the BBC.

FINAL COMMENTS

This article has attempted to give an introduction to the digital transmission impairment known as *jitter*. To date, there has been the advent of skeletal digital transmission usage throughout the world and, as time goes, on these will develop into wholly digital networks. The study of jitter in digital networks is undeniably a complex subject, and more operating experience is required to confirm the validity of the jitter accumulation laws that have been derived. Although Telecommunication Administrations are actively investigating these aspects, it will be many years before a thorough understanding of jitter accumulation in a real network is satisfactorily concluded.

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Biographies

Brian Kearsey worked initially for some seven years in heavy electrical engineering with Enfield Standard Power Cables and GEC Rectifiers in the field of power distribution for the national grid and for railway traction. During that time, he obtained a first-class honours degree in electrical and electronic engineering at The City University, London. On graduating in 1973, he joined British Telecom and became a member of the division concerned with the development of digital transmission systems and equipment. Since 1979, he has been active in digital transmission network standards involving participation at CCITT, CCIR and CEPT.

Robert McLintock was awarded a first-class honours degree in electronic and electrical engineering by Manchester University in 1972. He joined the then Post Office Telecommunications Development Department as an Executive Engineer and worked, initially, on the development of 24-channel and, subsequently, 30-channel PCM transmission systems. Since 1978, he has headed the group responsible for digital network transmission standards. These duties have included regular participation in some of the international standardisation committees of the CCITT and CEPT. He is currently chairman of a CEPT group concerned with the standardisation of speech encoding techniques.

A New Microprocessor Controller for Parcel Sorting Machines

E. HEARN†

UDC 656.851 : 681.187 : 681.31-181

A new controller for parcel sorting machines, based on microprocessor technology, which has been developed by Post Office staff, is described in this article.

INTRODUCTION

The implementation of the Parcel Post Plan, during the decade 1966–1976, saw the phased introduction of mechanised parcel concentration offices (PCOs). The tilted-belt parcel sorting machine (PSM) has been used in almost all of the offices, being replaced only recently by tilting-slat and tilting-tray systems[1], which are capable of carrying larger volumes of traffic.

The period during which PCOs were commissioned was also one of rapid development in electronics. Equipment control systems installed in PCOs accurately mirror these developments. Early controllers used electromechanical components, combined with discrete semiconductors. These were followed by designs based on magnetic core stores which, in turn, gave way to systems using integrated circuits. During the late-1970s, controller designs used re-circulating shift registers, based upon large-scale integrated (LSI) circuits and read-only memories (ROMs).

Despite being adapted to the innovations in technology, the functional specification of the PSM controller remained unchanged; it continued as a piece of dedicated hardware, providing sorting controls and limited traffic information in the form of electromechanical counters. Those PCOs requiring special control arrangements, because of operational or building constraints, were provided with one-off designs. In some instances, machines were enhanced retrospectively to convert the input to two-man operation. This involved installing additional controller equipment, because conversion was too costly and operationally disruptive.

The introduction of the microprocessor provided the opportunity to develop a low-cost flexible controller, which could be programmed to meet the specific requirements of PCOs. Furthermore, the new systems could simplify the storage and transmission of data and, therefore, enable PSM traffic reporting to be easily implemented. In view of the potential benefits, the design of a microprocessor-based PSM controller was undertaken. The design was the work of a Post Office (PO) team, and construction of a prototype was carried out by Engineering Department laboratory staff. In this way, total control over the design was retained and the resulting system meets all design requirements.

PARCEL SORTING MACHINES

Machine Description

The tilted-belt PSM comprises a straight conveyor, on which mail is placed by operators and discharged at sortation points along its length. At the input end, the conveyor belt is horizontal, but a twist section in the bed plate causes the conveyor to tilt at an angle of 37°. Along the length of the tilted conveyor, there is a 150 mm high side wall, into which doors are set. The doors can be opened through 90°, allowing parcels to slide, under gravity, from the conveyor into storage chutes. Each of these chutes is a machine selection.

Machine operators have a sorting keyboard, with keys for each selection, one of which is depressed as the item is placed on the machine. A synchronising photobeam registers receipt of the parcel onto the machine and initiates the sorting control process.

Typical PSM power control circuits consist of 50 V DC actuators and relays. PSM keyboards are also designed to operate at 50 V DC. This voltage level is used to provide a sufficient margin against electrical noise in what is a highly hostile electrical environment.

Existing Controllers

Early designs of the tilted-belt controller were based upon pin-wheel mechanisms; they had transistorised circuits for driving electromechanical solenoids to set pins on the edge of wheels, which are rotated at a speed proportional to the speed of the conveyor[2] (see Fig. 1). Later designs of con-

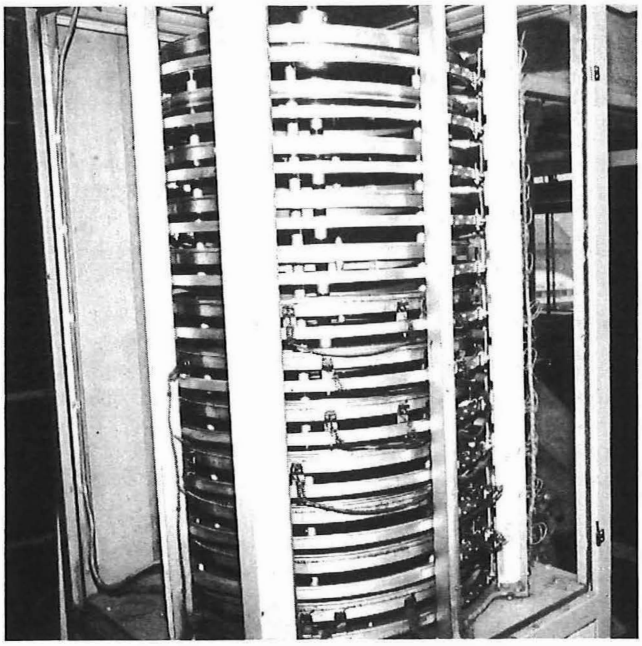

FIG. 1—Electromechanical pin-wheel controller

† Engineering Department, The Post Office

troller were based on an electronic shift-register system. Parcel sorting was achieved by stepping codes through stages of a shift register; each stage represented a 1 m section of the conveyor. In this way, parcel travel was tracked along the conveyor's length. The shift register used integrated circuits.

Further developments of controllers were tried on a single office basis. For example, a centralised controller, for an installation of 13 PSMs at Birmingham PCO, was based upon standard PO logic boards and a magnetic core-store memory. Difficulties have been experienced with this type of design, because faults have the effect of disrupting all machines and paralysing the office. To improve PSM utilisation, a two-position injection unit dual-injection feed (DIF) was developed and installed in a number of offices. Where these were fitted retrospectively, a separate control cubicle was installed.

Difficulty has been experienced by maintenance staff in servicing the older controllers, because components and board assemblies are no longer in production. Their constructional techniques make access to boards and interconnections difficult. They also lack adequate test and diagnostic facilities, and overall reliability is low.

DESIGN OBJECTIVES

In view of the need to replace existing controllers, the main requirement was that the new system must interface with existing machines. The new equipment had also to meet all the relevant requirements of the standard specification for PSM controllers.

The following additional objectives were also set for the controllers:

(*a*) Provision should be made for the control of a DIF system, which might be added retrospectively to an installation.

(*b*) Where a traffic information system (TRIPOS)[3] is not provided, facilities should be provided for a printed traffic report for supervisory staff, giving traffic handled during a work period.

(*c*) An interface should be provided for the serial transmission of traffic data to a remote computer-based traffic monitoring system[3].

(*d*) Maintenance aids should be provided to enable faults to be located to printed-wiring-board (PWB) level within 15 minutes.

(*e*) The electronic section of the controller should be modular to facilitate complete replacement.

(*f*) All PWBs should be designed for servicing at the Postal Engineering Service Centre (PESC) using automatic test equipment.

The requirement for a modular electronic crate was suggested by maintenance staff, who were consulted during the design stage. Experience with control systems has shown that problems occasionally occur in PWB crates, which have resulted in long repair times due to their poor serviceability.

OPERATIONAL DESCRIPTION

The new controller has three modes of operation, one for each of the operational functions that are performed: operational mode, supervisory mode and engineering mode.

Operational Mode

When power is first applied, the controller enters the operational mode; in this mode, it awaits signals from the PSM. When the PSM START button is pressed, the machine keyboard is energised and the door-solenoid drive circuits are initialised. This mode remains set until either a PSM *stop* signal is received, or a 5-minute timer, which is reset by signals from the synchronising beam, times out.

TRAFFIC REPORT FOR PSM NUMBER FOUR
****

SORTING PLAN:......

REPORT TIMED AT: 09:38   DATE: 21/03/84   SERIAL NO.: 1

DOOR	SORTED	RECIRCULATION	DOOR	SORTED	RECIRCULATION
1	2	0	2	3	0
3	1	0	4	4	0
5	1	0	6	2	0
7	2	0	8	2	0
9	3	0	10	8	0
11	0	0	12	2	0
13	1	0	14	4	0
15	3	0	16	3	0
17	2	0	18	5	0
19	2	0	20	3	0
21	1	0	22	2	0
23	3	0	24	2	5
25	2	0	26	2	0
27	2	0	28	3	0
29	2	0	30	1	0
31	1	0	32	2	0
33	2	0	34	3	0
35	0	4	36	3	0
37	1	0	38	2	0
39	3	0	40	2	0
41	1	0	42	5	0
43	5	0	44	5	0
45	2	0	46	2	0
47	2	0	48	2	4

TOTAL SORTED: 116   COUNT TO RECIRCULATION: 9

TOTAL TO TRANSFER: 4   HOURS RUN: 0:11

TOTAL RECIRCULATION: 13

UNCODED PARCELS: 6

TOTAL PARCELS 139

****

FOR FURTHER COPIES PRESS PRINT BUTTON

(WARNING: DESELECTING UPDATE AFTER GETTING REPORT DESTROYS TOTALS)

FIG. 2—Example of traffic report print-out

Supervisory Mode

Selection of the supervisory mode is by means of a switch on a small control box mounted in the central control office. The control box is also connected to a teletype printer, operating with an RS232 interface. The controller senses the switch signal and then responds to *report request* signals. Traffic reports are printed on 80-column wide paper (see Fig. 2).

Engineering Mode

In the event of controller faults, maintenance routines can be accessed from the engineering mode. This mode is selected by means of a key switch in the controller. Software-controlled diagnostics are accessed by means of a hand-held teletype or visual display unit terminal plugged into a socket in the controller. Diagnostics are available to test all input, output and microprocessor functions.

Sample diagnostic messages are shown in Fig. 3.

SOFTWARE

Pilot studies of the control algorithm for the PSM indicated that assembly-language programming was most suitable, in order to meet the stringent timing requirements; by using this, rather than a high-level language, efficiency is improved. Development of software was performed by using a microprocessor development system, with the facility to download programs directly to erasable-programmable read-only memory (EPROM) integrated circuits. Versions of the control program were loaded into the controller and tested. Revisions of the program, or enhancements, could be rapidly tested by using these devices.

```
       DIAGNOSTICS
       ***********
       COMMANDS ARE AS FOLLOWS:-1)REPORT, TYPE R
                                2)DIAGNOSTICS, TYPE  D1 TO D6
D2
       D2-I/P BUFFER CHECKS
       TYPE A FOR K/B CODE          TYPE B FOR CANCEL
         "  C  "  SYNC BEAM           "  D  "  START
         "  E  "  D-STOP.             "  F  "  BELT PULSES
         "  G  "  RECIRC BEAM         "  H  "  R-RECIRC
         "  I  "  UPDATE BUTTON       "  J  "  PRINT BUTTON
         "  K  "  F-ROUTE             "  L  "  G-ROUTE
         "  M  "  H-ROUTE             "  N  "  PLAN SWITCH
         "  O  "  BLOCK CHUTE         "  Q  "  QUIT
C
       TO TEST SYNC BEAM, BLOCK IT  THEN TYPE T
T
       SYNC NOT RECEIVED
C
       TO TEST SYNC BEAM, BLOCK IT  THEN TYPE T
T
       SYNC RECEIVED
A
       TO TEST K/B, PRESS A BUTTON  THEN TYPE T
T
       CODE REGISTERED :-000110
Q
       QUITTING I/P BUFFER TEST
D4
       D4-O/P DRIVER CHECK
       TYPE 1 FOR LAMPS & TOTALS METER
         "  2  "  COD-CON
         "  3  "  DOOR TEST
         "  4  "  PSM ROUTINE
         "  Q  "  QUIT
1
       LAMPS TESTED
2
       CODE-CON TESTED
3
       TO TEST A DOOR, TYPE ITS NUMBER THEN C-R
       TO QUIT, TYPE Q ONLY
5

       DOOR TEST FINISHED
Q
       YOU ARE QUITTING DOORS TEST
4
       PSM ROUTINED
Q
       YOU ARE QUITTING D4
```

FIG. 3—Sample of diagnostic messages

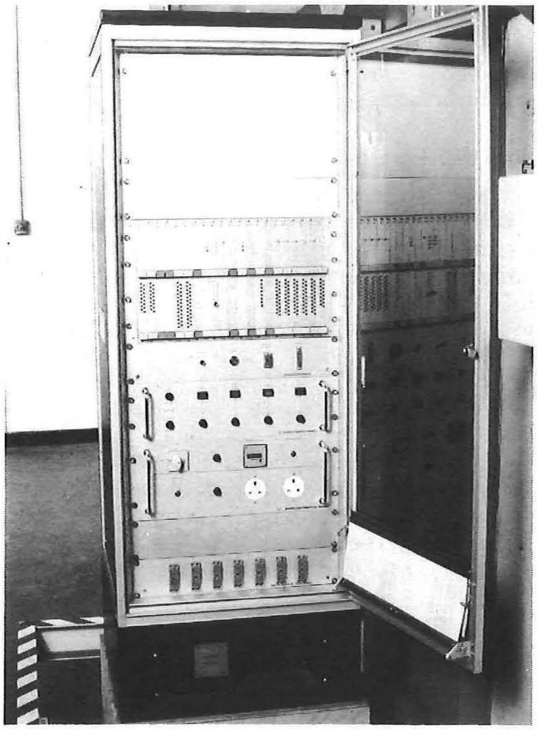

FIG. 4—Controller cubicle

A modular development approach was adopted for the complete system software comprising four parts:

(*a*) an executive module which, at switch on, performs the various initialisation operations required and then selects one of the three operating modes, each of which is a self-contained module;

(*b*) the operational module, which contains the control program for servicing the machine controls, the sorting operation and data gathering;

(*c*) the supervisory module, which deals with all management information software, printing reports etc; and

(*d*) the diagnostics module, which deals with the engineering maintenance routines that are available.

The modular approach to the software design has the added benefit of making the software easier to document. To this end, extensive comments in the program listings are also used.

HARDWARE DESIGN

In view of the relatively small numbers of controllers required at any one time, it was not economic to design a dedicated cubicle or racking system. A commercially available cubicle was selected (see Fig. 4), compatible with the industry-standard 19 inch racking system. A range of optional fittings is available for tailoring the cubicle design for different applications. Despite this, however, it proved necessary to design some parts in-house to achieve a cubicle that met PO requirements.

The cubicle selected has enough spare capacity to accommodate an additional rack of PWBs. To date, this space has not been utilised and will be dispensed with in later models. To simplify construction and maintenance, the various functions in the cubicle are constructed on a modular basis. The power supplies are mounted on shelves, which can be withdrawn on telescopic runners. Supply fuses and indicators are mounted on the front panel of the shelf.

Printed-Wiring Boards

Each PWB is of the standard Eurocard type, having dimensions of height 233·4 mm and depth 220 mm, and carries two 64-way connectors of the DIN 41612 type (see Fig. 5). This large number of connection points was chosen to facilitate automatic testing of the boards, test points being extended to spare pin positions. All boards have double-sided artwork and use plated-through holes.

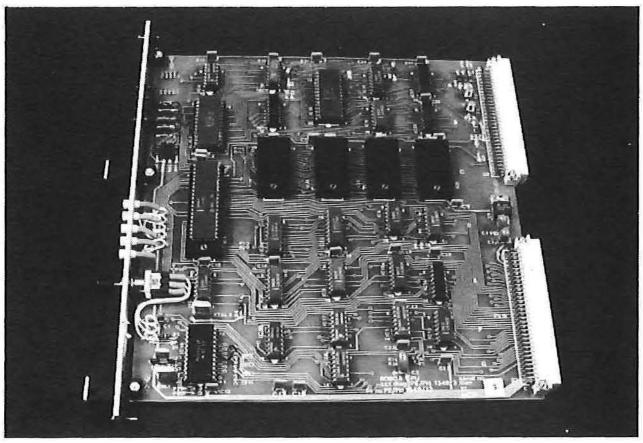

FIG. 5—Typical printed-wiring board

Metal front panels, which have the function identification legend printed on them, are attached to each board. Light-emitting diodes (LEDs) are fixed into the panels to provide status indicators for servicing.

Printed-Wiring Boards Crate

A modular construction is used, the entire crate being removable. All connections are made by means of plugs and sockets, mounted on a hinged rear panel. Signal connections are hard wired from the PWB connector pins to the sockets on the hinged panel.

The boards are grouped functionally within the crate. All input PWBs are grouped at the left-hand end, the central section contains the microprocessor boards and all output boards are situated on the right. The precise ratio of input and output boards can vary according to the number of doors to be controlled and the facilities required.

All microprocessor boards are wired by using a common 64-way bus. Several of the bus rails carry power distribution to the boards, and these use a solid copper strip to minimise voltage drop.

Engineers Panel

Sited beneath the PWB crate is the engineers panel, on which is mounted a keyswitch, push-button and two connectors. These enable service personnel to enter into the diagnostic and routining mode, and to obtain traffic reports at the cubicle. Full tests can be performed from this panel, which is particularly useful when the controller is sited remotely from the PSM.

Power Supply and Mains Distribution Panels

Below the engineers panel is a sliding tray containing the switched-mode power-supply units used by the controller. Fuses and indicators are mounted on the panel. Outputs from the power supplies are wired by using flexible cables and terminated in a multi-pin plug, which connects directly into the PWB crate.

Immediately beneath the power-supply tray is the mains distribution tray, which contains two mains supply conditioners. On the front panel is an ON/OFF isolator, hours-run meter and two mains socket outlets.

Input/Output Test Panel

At the base of the cubicle is a fixed panel, on which is mounted a group of test sockets. The pins of the socket are wired to the rows of cable terminals in the base of the cubicle to give access to all inputs and outputs. A portable tester is used, which can generate test input signals and monitor output signals.

FIELD TRIAL

Two production prototype controllers were installed at Sheffield for field trial in October 1981, in time for the Christmas pressure period. Commissioning trials involved the use of dummy mail, processed by staff working as a sorter-facer pair. Test sorting rates of 2400 items per hour were achieved, with a 100% successful sorting rate.

To provide back-up during the field trial the existing pin-wheel controller was left in circuit, but in a by-passed mode. In the event of failures of the new controller, the keyboard and door driver plugs could be re-connected to the pin-wheel controller.

Initial problems were experienced on one of the controllers, which were found to be due to a faulty microprocessor socket; this caused misoperations of the control program. Certain operational problems, specific to the Sheffield installation, were encountered during commissioning, one of which was due to the operation of the overflow door that was two doors back from the end of the machine. This meant that two selections required the closure of the overflow door, while all other selections would initiate its opening. These problems were quickly resolved by re-programming the controller on site.

A formal 6-months field trial of the two controllers was initiated in July 1982 and was monitored by the Maintenance Division of the PO Engineering Department. The results showed that the design objectives were adequately met. To date, the controllers have been operational for a total of 21 000 hours. During this time, only two component failures have occurred—on a power supply and an integrated circuit socket—giving a mean-time-between-failure (MTBF) of 10 500 hours.

Further controllers have now been installed at Birmingham PCO, and these have operated without component failure for over a year. This installation required that additional control facilities be provided for the transfer routes between groups of PSM and multi-sectioned doors. By adaptation of the control program, these additions were easily incorporated into the controller.

CONCLUSIONS

A new PSM controller based upon microprocessor technology has been successfully tested, and has demonstrated its reliability and flexibility. Future developments will enable it to be integrated into a network in which traffic information will be gathered on a regional and national basis, thus facilitating improved management of parcel traffic.

The modular design of the controller enables it to be adapted for the control of tilting-slat and tray sorting machines. Controls for multi-injection feed systems can also be integrated in to designs.

The controller meets the design objectives of low cost in comparison with proprietary controllers, high reliability and low mean-time-to-repair (MTTR). Its built-in interface to TRIPOS provides an economic solution to the installation of TRIPOS at existing PCOs.

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Biography

Eric Hearn joined the Post Office as a Youth-in-Training in 1960 and worked in London Telecommunications Region on exchange power plant installation and maintenance as well as accommodation services. After becoming a graduate member of the IERE in 1966, he transferred to Postal Engineering Laboratory in what was then the Engineer-in-Chief's Office. While there, he worked on early versions of letter-sorting-machine translators and other digital systems. Since then, most of his career has been spent on the design and development of control systems for parcel sorting and handling equipment. He is currently working in the computer systems hardware evaluation group in the Post Office Engineering Department.

Fault Location Techniques on the Inland Microwave Radio-Relay Network

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UDC 621.396.66 : 621.37.029.6

British Telecom has introduced more efficient and cost-effective methods of maintenance into the inland microwave radio-relay network. This article describes the new techniques and the aids that have been devised to help maintenance staff locate faults on both analogue and digital microwave radio-relay equipment.

INTRODUCTION

Maintenance techniques used on the UK inland microwave radio-relay network have evolved since the first multi-hop system was opened to traffic in December 1949 between London and Birmingham to the fully integrated network of today.

By 1970, extensive expansion of the network had taken place. This expansion, continuing until the present day, reflects the considerable improvement in equipment development in this field. Fig. 1 shows the analogue microwave network in the UK to date.

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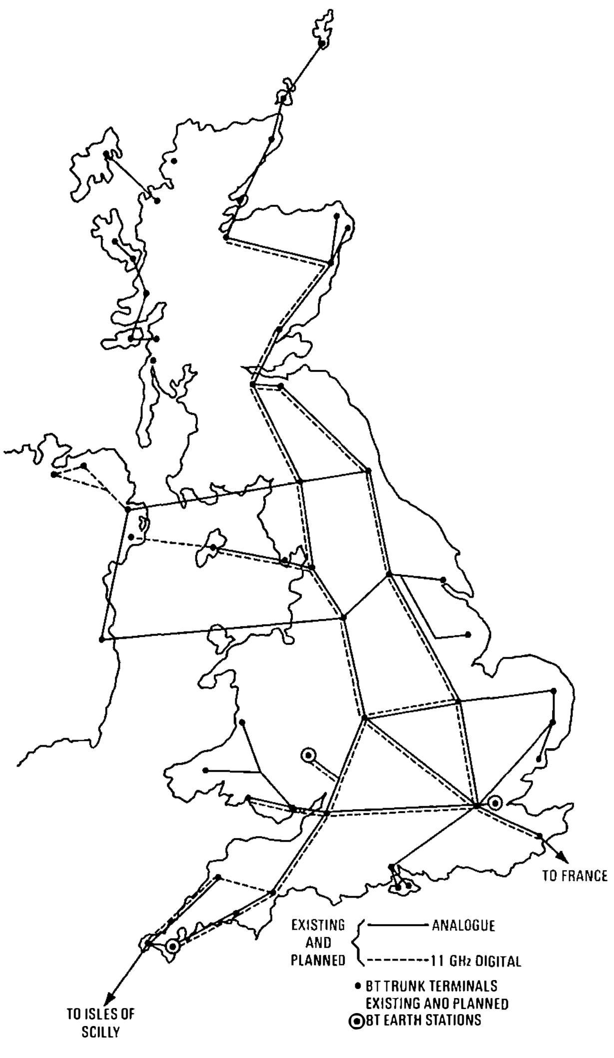


FIG. 1—British Telecom microwave radio-relay network

More recently, British Telecom (BT) has been concerned with the introduction of 11 GHz digital microwave radio-relay equipment to meet the requirements of industry and commerce for digital telecommunication facilities. This new digital equipment differs in design from the analogue microwave equipment. The former requires the further training of staff in order to maintain the quality of service expected by the customer.

The first microwave radio-relay system provided during 1949 carried television signals for the British Broadcasting Corporation between London and Birmingham. As television coverage increased, the number of BT microwave channels needed to carry these programmes increased, culminating in the establishment of the Independent Broadcasting Authority's Channel 4 television network. Currently, television programmes are carried by BT analogue microwave links throughout the UK, and to the earth stations at Goonhilly Downs, Madley and London Docks for transmission internationally.

BT also provides an extensive network of analogue microwave radio-relay channels for carrying inter-city telephony and data traffic. All forms of traffic carried on the microwave network require not only quality, but also continuity of service. This aim is assisted by the provision of stand-by facilities on all microwave systems. Further support is provided by a service protection network, which is available to re-route traffic if required. This continuity and quality of service also depends on the efficiency of maintenance techniques used by staff on the microwave radio-relay network.

Maintenance of microwave radio-relay systems falls into two categories: routine and corrective. Routine maintenance, until the early-1970s, involved the periodic inspection and testing of individual panels and racks at all stations on a system, irrespective of whether the overall performance was degraded or not. This required extensive use of the protection facilities and prolonged staff involvement, which was costly in both equipment and manhours. This method also endangered the remaining traffic-carrying channels while the protection facilities were in use for maintenance purposes.

Corrective maintenance, which deals with faults either indicated by equipment alarms or reported by a customer, followed much the same approach as routine maintenance. After service was restored by switching to a protection facility, the investigation into the cause of the fault involved detailed testing and inspection of the radio equipment at all the stations on the channel in question. Once started, this work was out of the control of the terminal station responsible for the eventual restoration of service. This approach was costly in channel time and in manhours. It was also extremely inefficient because few, if any, clues to the cause of the original complaint were in the hands of the staff concerned with the work being carried out. Much time was spent on tests that had no relevance to the problem in question; moreover, in many cases, other faults were caused by the unnecessary testing of equipment that previously

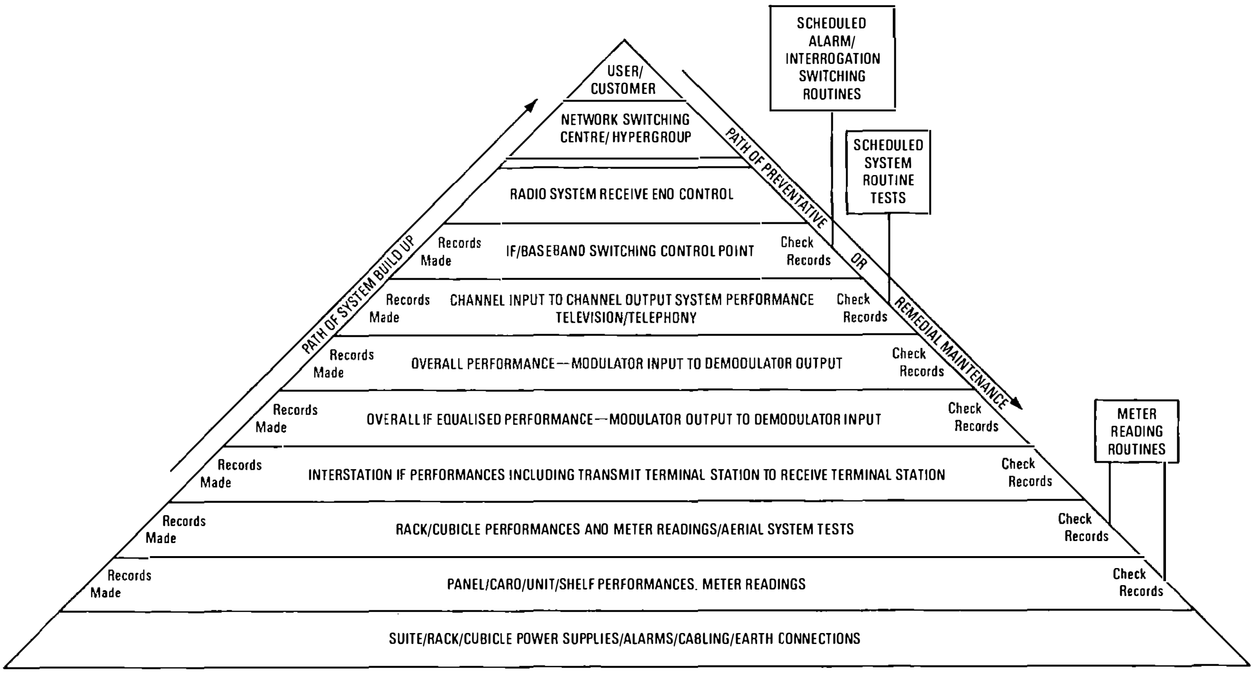


FIG. 2—System engineering philosophy

operated correctly. This work depended very much upon the expertise, experience and judgement of the staff involved. But experience and judgement are fallible and can often cause, rather than cure, problems.

In 1973, the group in the Headquarters of the British Post Office concerned with the maintenance of the microwave radio-relay network in the UK decided that a more efficient and cost-effective method of maintenance must be introduced. This new approach was to be introduced nationally under the auspices of this group, which also monitored its progress.

SYSTEM ENGINEERING ON THE ANALOGUE MICROWAVE NETWORK

The new maintenance approach was given the title of *system engineering*, because it involved diagnostic techniques based on the overall performance of the system. This diagnostic approach is concerned with initially inspecting the overall performance parameters of the system. Where any one or more of these parameters is outside the required limits, this information can be used to help decide the likely cause of any degradation, and where on the radio channel it is likely to be found. Various charts of information were designed to assist in this diagnostic work and, once this information had been proved by practical field tests, these charts were issued to all stations on the network. However, because the changes were of such a radical nature, it was decided to introduce this printed information and the new techniques to the maintenance staff and Telephone Area management through the medium of lectures. Some 25 of these lectures were given, and were attended by over 400 staff of various grades.

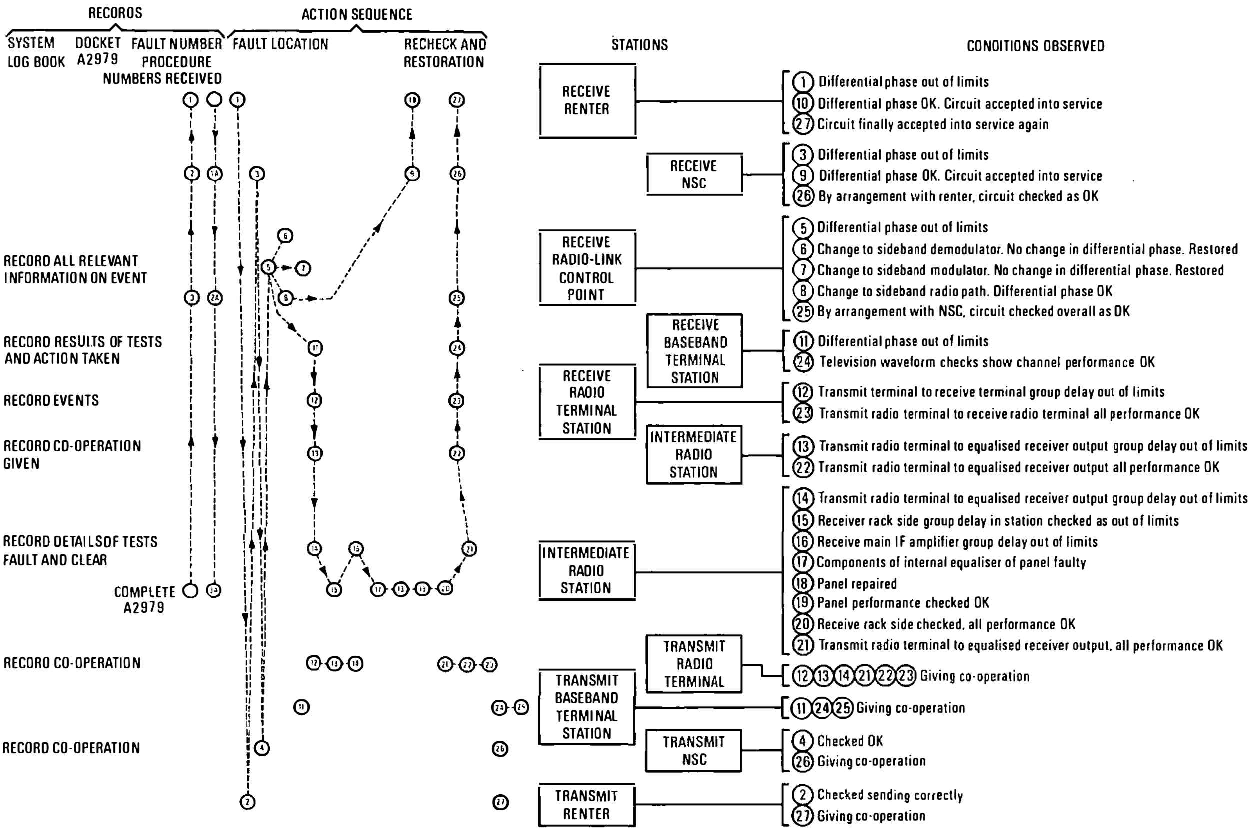
The printed material first produced is shown in Fig. 2; this was produced in poster form, and in pocket-book size for personal use. On the left-hand side of the triangle, from base to apex, are shown the steps normally taken by a manufacturer to produce a radio channel from the initial testing of panels in the factory to the final overall tests over the completed channel in the field. The details of the steps are shown across the centre section of the triangle. On the right-hand side of the triangle are shown the steps, from apex down to base, that should be taken to establish the cause of degradation of one or more of the system parameters. This sequential testing and location requires orderly recording of the results of the tests carried out at each stage, and controlled acceptance of responsibility during such an operation. In the lectures given to the staff, two examples of such work were explained in detail by reference to two charts issued on a personal basis at each lecture.

One such example is shown in Fig. 3; this is of a fault reported by a television customer where differential phase is out of limits. The numbered steps to be taken and the stations concerned are shown in the action sequence section of the chart. Explanation of the steps is shown on the right of the chart; on the left, are shown the records to be made during the operation, including the entries to be made in the system log book. The log book is issued on a system basis at each station, to provide a station record of the complete history of the system concerned. In the case of the television signal fault shown in Fig. 3, the cause of the degradation in the parameter was traced to an amplifier at an intermediate radio station in step 17.

Although these charts show the steps to be taken in investigating such faults, the information is in itself no guide to the diagnostic techniques to be used in such a procedure. Therefore, two charts were produced to show the relationship between cause and effect of degradations of performance on radio channels carrying television and telephony traffic. It is necessary to recognise that the basic parameters on which the overall performance of a microwave radio-relay channel depends are the same parameters which, if distorted, are the cause of a degraded performance; there are no hidden causes or mysteries at the root of any problem that may arise. Figs. 4 and 5 show the direct links between cause and effect of any degradation of overall performance.

Fig. 4 shows the overall television system parameters and the causes of degradation of such tests. For example, on the left-hand side of the chart is information about monochrome waveform distortion. The basis of this test is a waveform consisting of a T pulse of 0·1 μs duration, and a bar signal of 25 μs duration, each having an amplitude that should be 0·7 V above black level. The column headed 'Effect' shows various distortions of this pulse-and-bar signal and, to the right, in general terms, the state of the overall channel gain/frequency response that would give rise to such a waveform distortion. To the right again, under the heading 'Likely Causes', is shown information on each possible cause of these degradations.

If the waveform headed 'High-Frequency Loss' is considered, it is seen that the high-frequency (HF) end of the



NSC: Network switching centre

FIG. 3—Standard fault location procedure—example for a fault reported by a television customer

band has less gain than the low-frequency (LF) end. The information under 'Likely Causes' shows that, if the problem is on the intermediate-frequency (IF)/radio-frequency (RF) section of the radio channel, it is due to distorted responses of the character depicted. Whereas the 25 μs bar needs an effective bandwidth of only approximately 2 MHz for transmission without distortion, the T pulse needs at least 5 MHz. In order to maintain the gain at the HF end of the band, the IF/RF gain/frequency response needs to have as much sum total power in the sidebands at the edges of the response as it has in the centre. If the power is less at the edges of the IF/RF bandwidth, the pulse is unable to rise to the same level as the bar. By using this to diagnose the cause of the pulse being low on the channel overall, the cause of the degraded television signal parameter can be predicted. Because of the problems associated with making measurements of IF/RF gain/frequency response through equipment that has automatic gain control (AGC) and where limiting is present, some further information is needed; this is given in Fig. 6, which, together with Fig. 7, is issued to staff in the field. Similarly, the diagnosing of problems with the television signal parameters of differential gain and phase shown in Fig. 4 is made much easier by referring to the information in Figs. 6 and 7. Usually, when one overall parameter is beyond the limits allowed, it is not the only parameter affected. For this reason, all allied parameters should be checked when a fault is found and all tests should be repeated after a fault has been dealt with, to ensure that no other problem has arisen as a result of the work carried out (see Fig. 3).

The chart shown in Fig. 5, again supported by the information in Figs. 6 and 7, contains information on channels carrying telephony traffic.

PRACTICAL DEMONSTRATION OF FAULTING TECHNIQUES ON ANALOGUE SYSTEMS

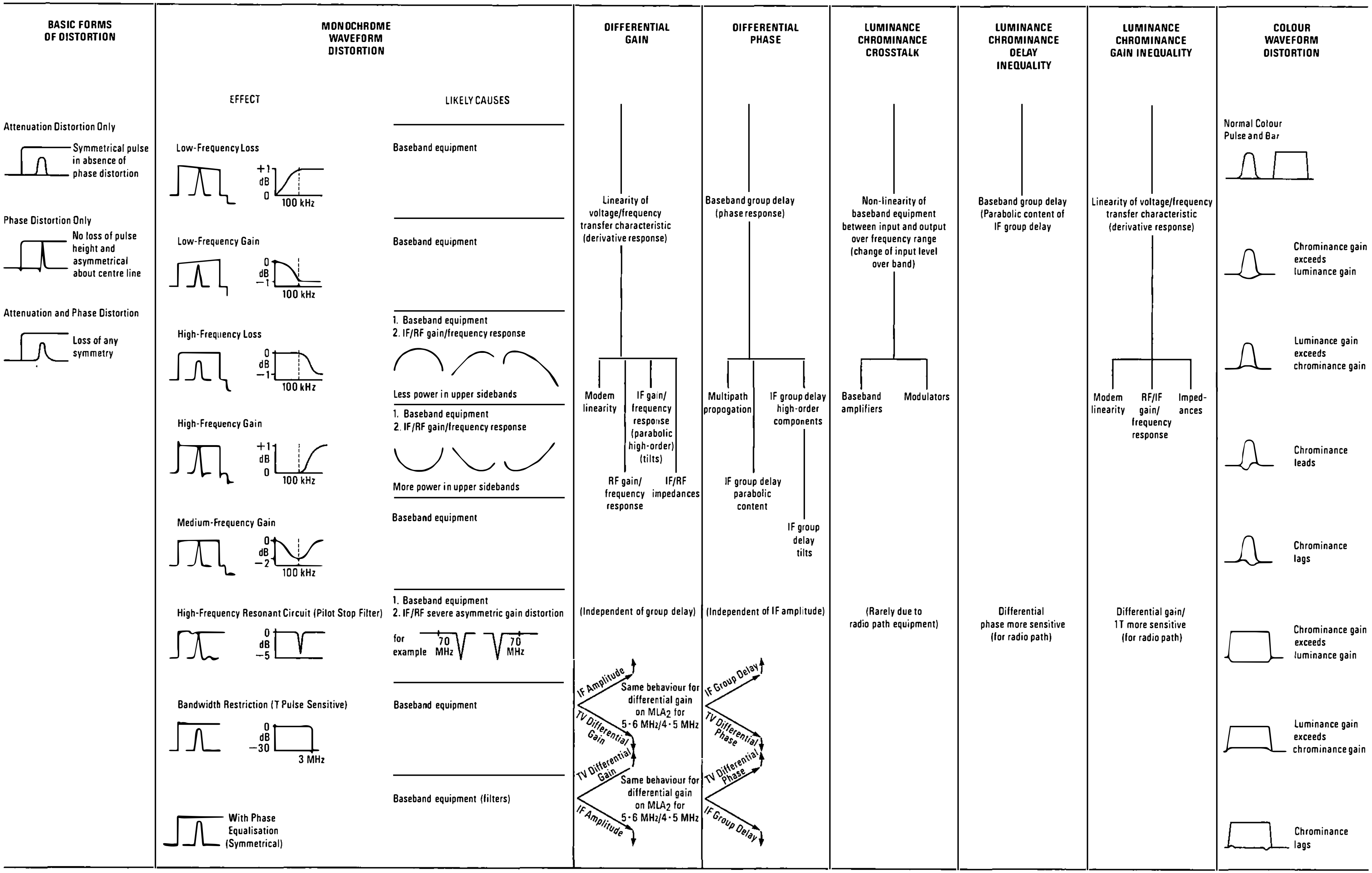
After the introduction of system engineering, it was decided that some practical demonstration of the validity of this new information was necessary so that this philosophy could be successfully applied.

In 1976, an outside broadcast vehicle that was no longer in use was obtained and fitted out for demonstrations. The equipment included a closed-circuit microwave link and the test equipment normally found in every radio station. The behaviour of the radio link under various fault conditions was demonstrated and staff were shown how to apply the information issued to the stations. Although not ideal for the work in hand, the vehicle was used for some four years and, during this time, many two-day demonstrations were given to staff in the field. These demonstrations consisted of the introduction of various distortions of radio performance and the detection by inspection of the overall television or telephony parameters.

The object was to prove, by practical tests, that the information given in the printed handout material was of practical use in the location of faults. Some time was also spent on the use of the various items of test equipment. Where staff had any doubts about either the results achieved or the tests made, they were invited to try the tests themselves.

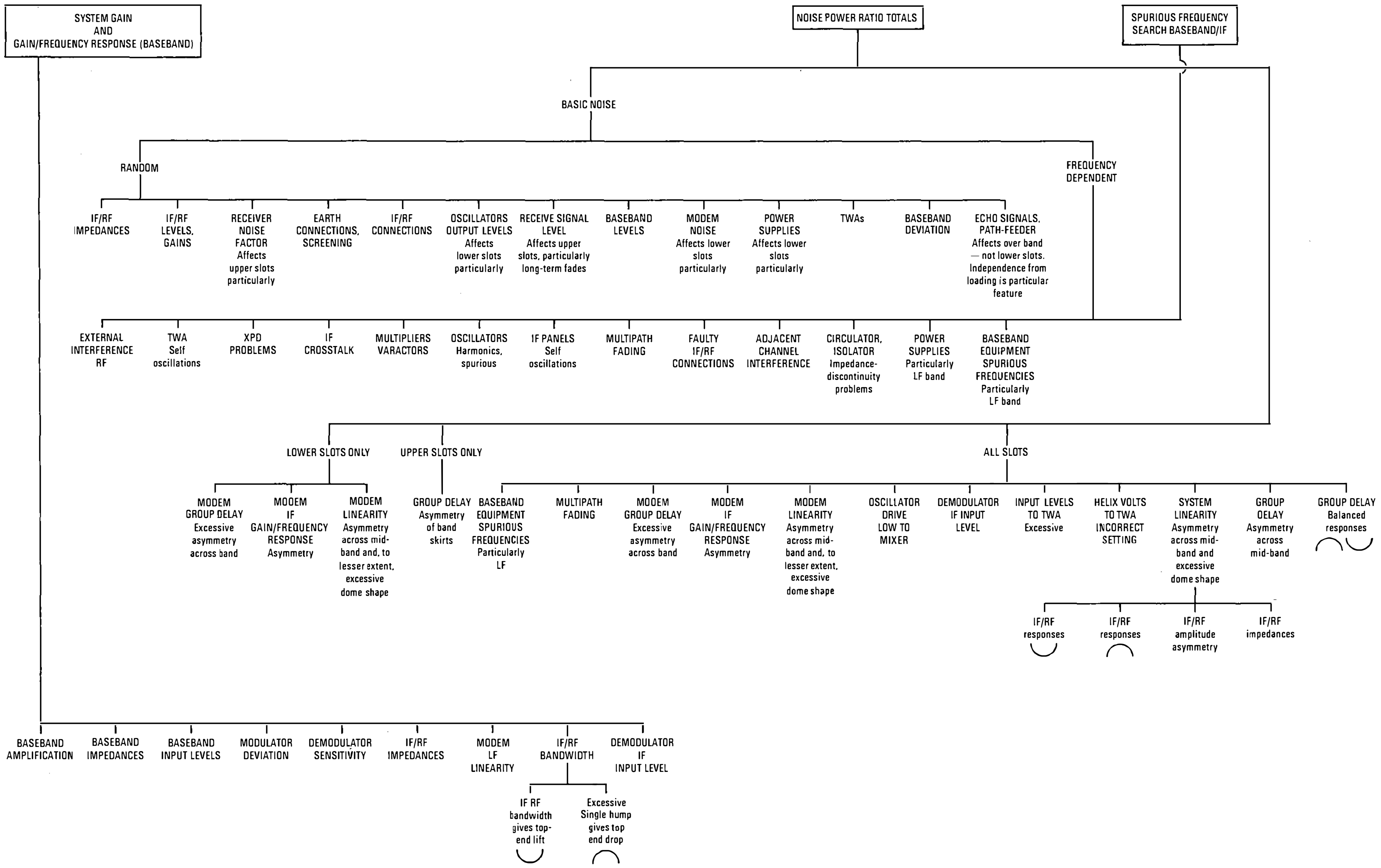
In 1980, a new custom-built vehicle was provided and fitted out with more modern radio equipment, with which it was possible to produce a much more effective series of demonstrations. Unlike the previous vehicle, the equipment in this one was fitted on shock-absorbing mounts and the accommodation for the staff attending the demonstrations was more generous. The equipment is shown in Fig. 8; the test equipment is mounted on the main and upper shelves, with the radio equipment mounted under the main shelf. Also shown are more recently fitted items for demonstrating digital techniques which are referred to later in this article. This new vehicle continues to be used for giving demonstrations on analogue systems, and 114 such demonstrations have now been given to 850 staff.

Although all the information has been issued and the demonstrations have been given, problems have still existed



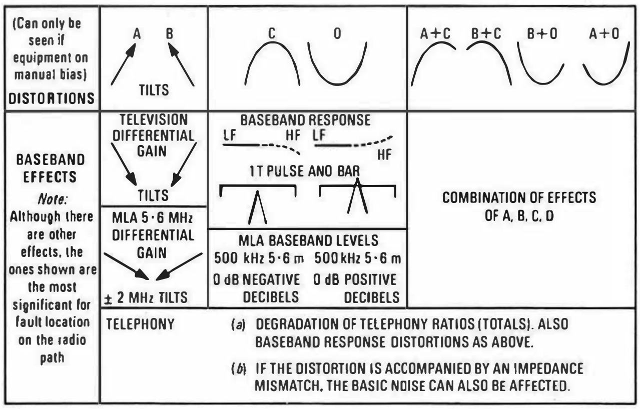
MLA: Microwave link analyser

FIG. 4—Radio system parameter relationships (television)



TWA: Travelling-wave amplifier
XPD: Cross polarisation

FIG. 5—Radio system parameter relationships (telephony)



MLA: Microwave link analyser

FIG. 6—Effects of IF/RF gain/frequency-response distortions on the radio path

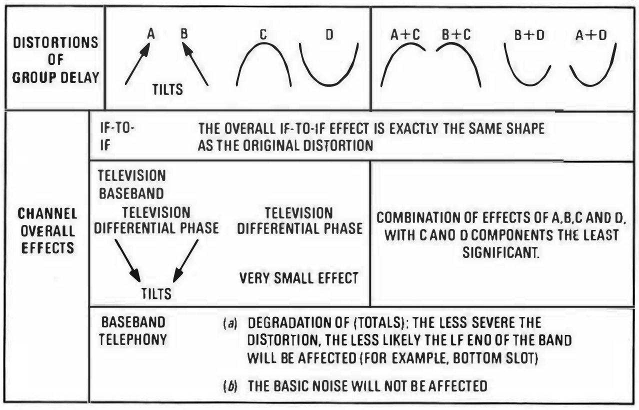


FIG. 7—Effects of phase-response distortions on the radio path

because, in some cases, the procedures that have been introduced have not being implemented. This has led to extended periods being needed to locate and clear faults. A means of controlling such activities was obviously necessary before the benefits of the system-engineering philosophy could be fully realised on the analogue microwave network. Two fault-location forms were therefore produced to ensure that the work of fault location was carried out in the manner described. Fig. 9 shows one of these forms which, when completed, must be placed in the appropriate system log book. This form is for an investigation into the steps taken and the results achieved. The information issued, when followed, is certain to provide the solution to the problem in

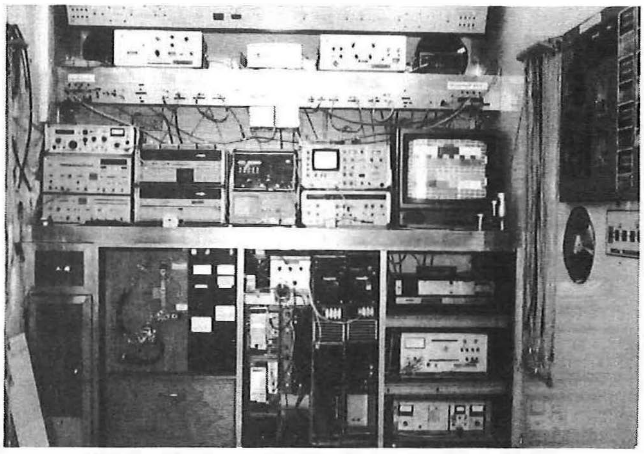


FIG. 8—Equipment in the demonstration vehicle

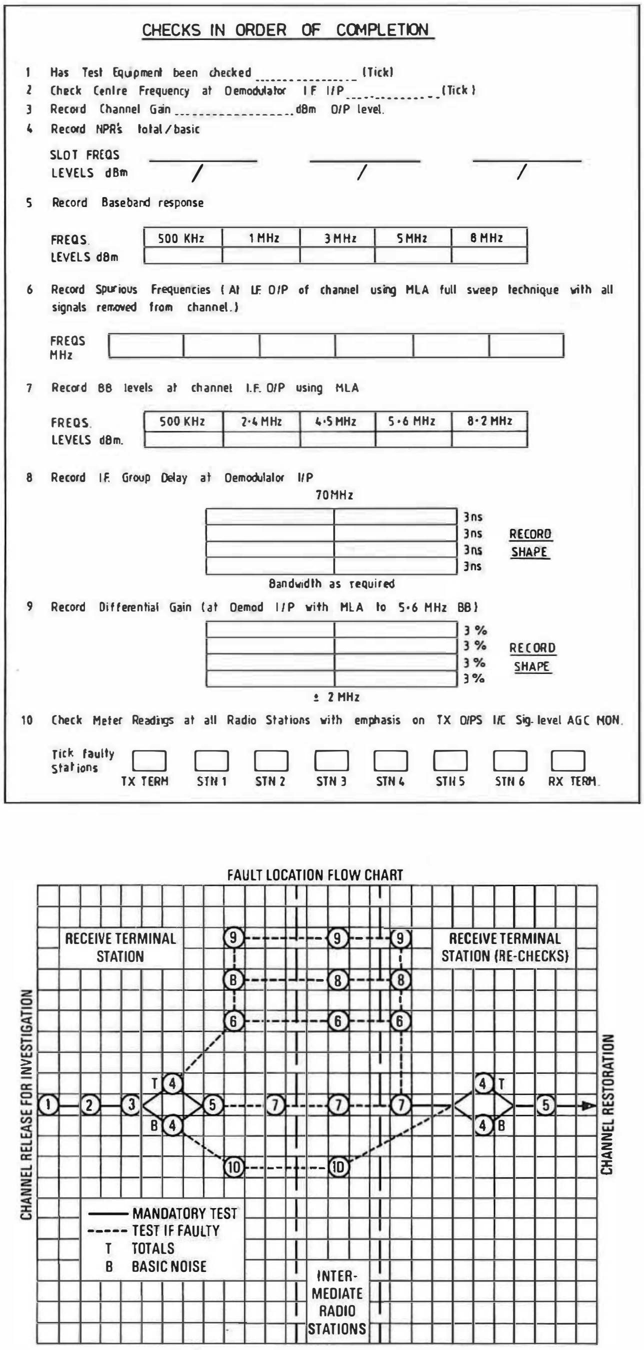


FIG. 9—Fault location form and flow chart

hand. These forms are now to be issued to the stations in the network and their use is to be demonstrated shortly to District staff.

FAULT LOCATION TECHNIQUES ON DIGITAL RADIO SYSTEMS

In 1981, digital microwave radio-relay equipment was being installed as an overlay facility over existing analogue routes, and it became necessary to arrange advance training, in demonstration form, before the maintenance of each route was finally taken on. This training needed to be of a practical nature so that techniques for fault recognition and location could be provided. A temporary digital facility was installed in the demonstration vehicle which, although not ideal, was sufficient to show some of the problems associated with this new equipment. To support this facility, information had to be produced to assist the maintenance staff during the early life of this digital radio equipment. The demonstration vehicle has now been fitted with an 11 GHz 140 Mbit/s radio installation; this is shown under the main test equipment shelf in Fig. 8.

The new equipment allows virtually all the problems that could be encountered on the digital radio equipment currently being installed or already in maintenance to be demonstrated. The exceptions are those problems associated with intermittent errors that occur at a rate not catered for on the normal radio monitoring facilities. The digital demonstration also includes a video tape that describes the 140 Mbit/s radio equipment.

On analogue microwave radio-relay systems, fault location is carried out by diagnosing causes of the degradation of overall performance, closely inspecting system parameters and using various information aids. On the digital equipment, these methods are not possible. The recognition, location and correction of faults on the 11 GHz digital system fall into two distinct categories. Faults that are of immediate significance are almost always indicated by light-emitting-diode (LED) displays on the equipment and, in some cases, by changes in equipment meter readings. Routine faults are those whose existence are not indicated until they are discovered by a routine test; even then, the LEDs are no clue to the possible cause of an indicated deterioration of performance. These categories of faults are now discussed in detail.

Immediate Faults

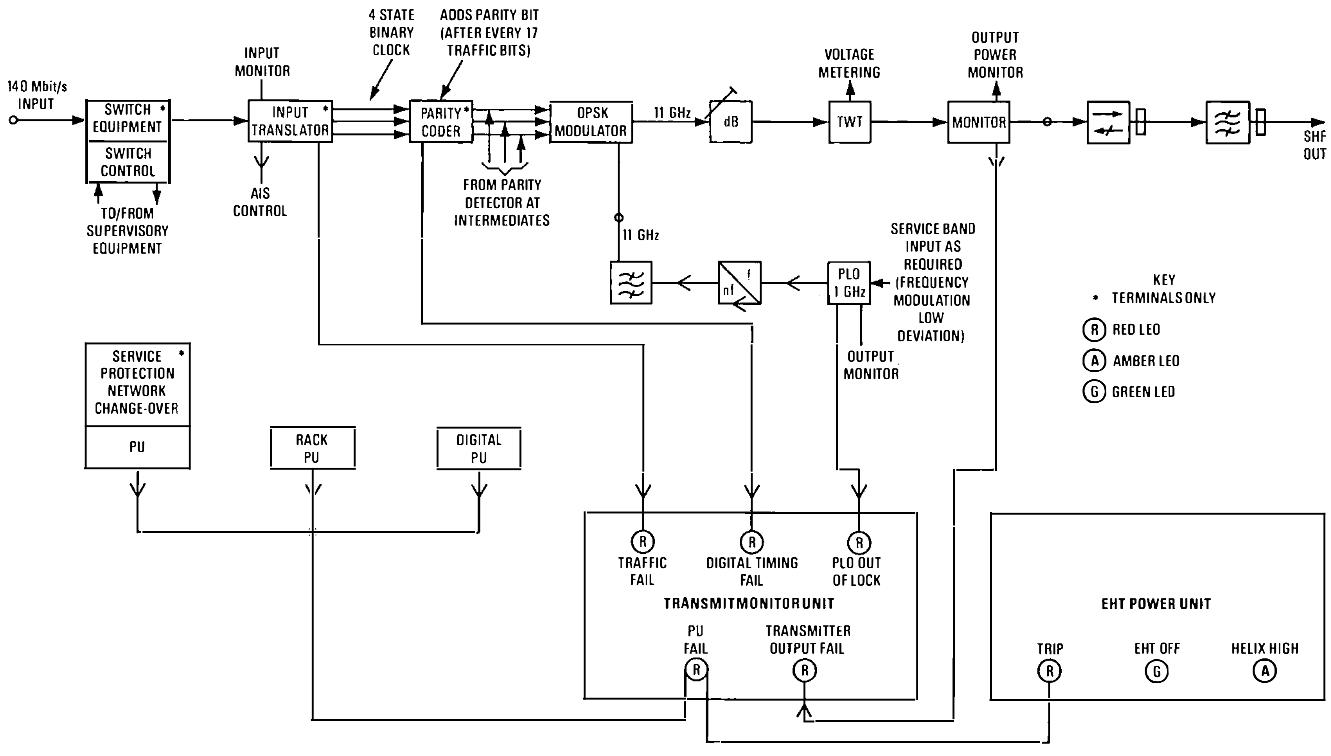
There are several LEDs on the radio equipment that indicate faults. Some expansion of the existing information is necessary to illustrate what these LEDs are monitoring, and the origin of the conditions that cause them to indicate the existence of a problem. This information is vital to the maintenance staff to enable them to understand the significance of the LED display. In order to assist staff to interpret the LED display under each fault condition, two block diagrams were produced: one for the transmitter and another for the receiver at all radio stations (see Figs. 10 and 11).

All radio stations, whether intermediate or terminal, having this type of equipment have identical radio equipment. The difference lies in the digital processing, and this is reflected in Figs. 10 and 11, where items to be found only at terminal stations are marked with an asterisk. With minor exceptions, the LED displays are the same at all stations, and it is these that are so important for fault location. Although the final judge of the suitability of a channel to carry traffic is the indicated error rate, several other LEDs provide further clues to the state of the radio channel, and it is the recognition of the significance of the complete display under fault conditions that is the key to successsful corrective maintenance.

Although Figs. 10 and 11 provide very useful information, application is difficult in the form in which it is presented when it is used as a diagnostic tool, as a variety of LEDs can be illuminated during a fault condition. To understand this information, some other aid was needed.

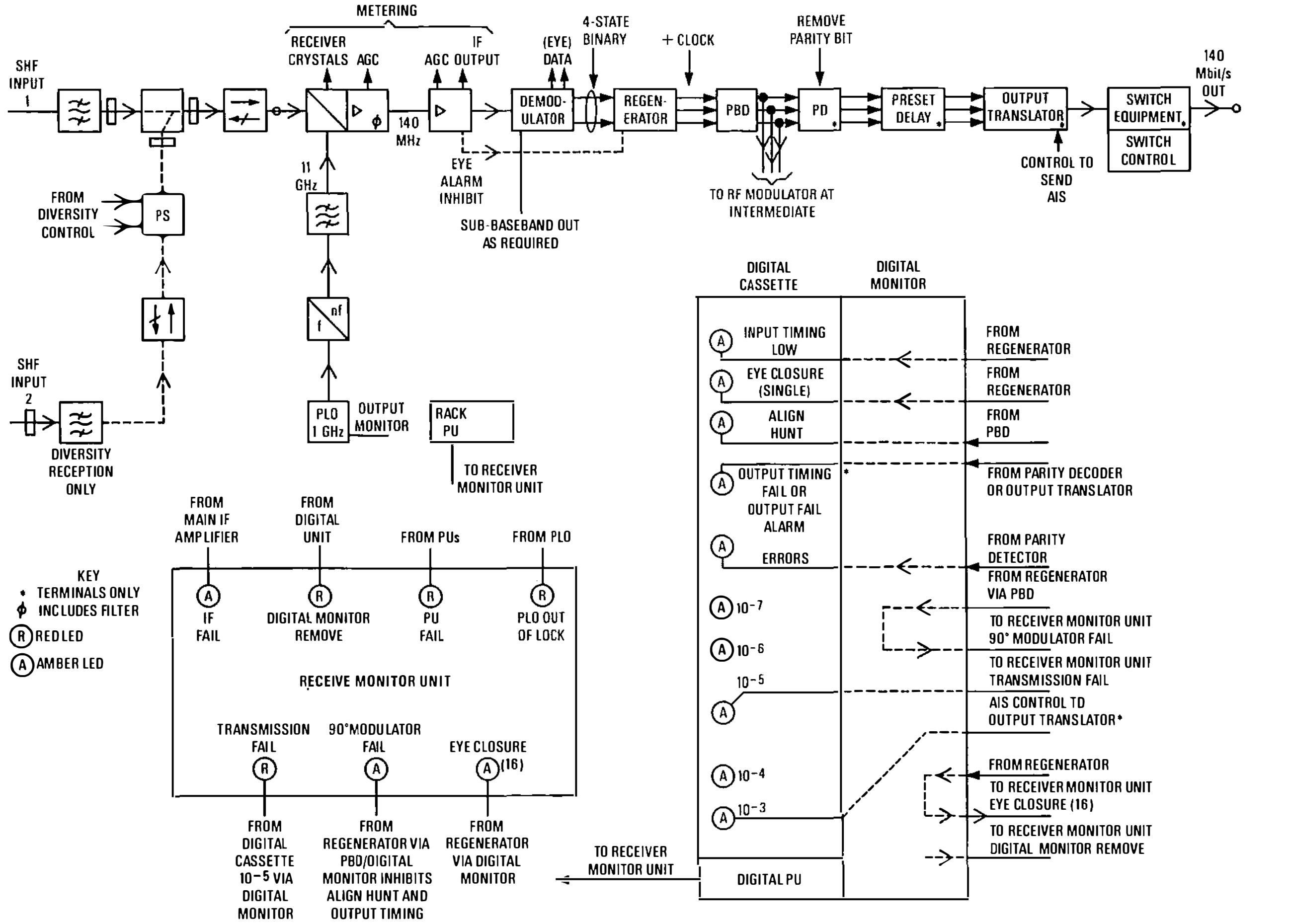
Therefore, an exercise was carried out to establish the meaning of each pattern of LEDs. On an intermediate station of the London–Birmingham 11 GHz 140 Mbit/s system, which was still under trial, a series of carefully tabulated faults was introduced onto a radio channel and the corresponding pattern of LEDs for each condition recorded. Some 50 conditions were recorded, and the information gained was made into a chart, part of which is shown in Fig. 12.

When the station staff see a particular pattern of LEDs on the equipment, they can refer to this chart to find the fault condition causing that display. In some cases, faults occur that are not displayed by the LEDs on the equipment originating the fault, and such incidents are the reason for the block of information on the far right of the chart. Because it would not be possible to predict or apply under



AIS: *Alarm inhibit* signal
EHT: Extra-high tension
PLO: Phase-locked oscillator
PU: Power unit
TWT: Travelling-wave tube
QPSK: Quaternary phase-shift keying
SHF: Super-high frequency

FIG. 10—Block diagram of 11 GHz 140 Mbit/s transmitter



AIS: *Alarm inhibit* signal
AGC: Automatic gain control
PBD: Parity-bit detector
PD: Parity decoder
PLO: Phase-locked oscillator
PS: Phase shifter
PU: Power unit
RSA: Remote station alarm
SHF: Super-high frequency
TWT: Travelling-wave tube

FIG. 11—Block diagram of 11 GHz 140 Mbit/s receiver

test conditions all of the possible patterns that could be caused by fault conditions, a second chart, similar to that shown in Fig. 12, was produced on which no fault conditions are entered. Whenever such different displays occur, staff can record the patterns for future reference. Such extra information is passed on to the group in National Networks Head Office concerned with maintenance of the microwave network so that all the network can be informed. Twelve extra incidents have already been recorded, and the information for these has now been passed on.

Location of faults indicated by LED displays, with the added information provided by the equipment meter readings, is then a logical procedure and eliminates the need for guesswork. It is possible for a comparative newcomer to this

FAULT CONDITIONS	FAULT ORIGINATING EQUIPMENT																								FOLLOWING RADIO STATION EQUIPMENT LEDs								
	RADIO RACK (RECEIVER) LEDs															TRANSMITTER LEDs								RSA								RSA	
	INPUT TIMING LOW	EYE CLOSURE (SINGLE)	ALIGN HUNT	ERRORS	10−7 10−6	10−5 10−4 10−3	IF FAIL	DIGITAL MONITOR REMOVE	PU FAIL	PLO OUT OF LOCK	TRANSMISSION FAIL	90° MODULATOR FAIL	EYE CLOSURE (16)	OUTPUT TIMING FAIL-	OUTPUT FAIL ALARM	TRAFFIC FAIL	DIGITAL TIMING FAIL	PLO OUT OF LOCK	PU FAIL	TRANSMITTER OUTPUT FAIL	TRIP	EHT OFF	HELIX HIGH	EXTENDED ALARMS	ALIGN HUNT	TRANSMISSION FAIL	90°	ERRORS 10−3	ERRORS	INPUT TIMING LOW	EYE CLOSURE (16)	EXTENDED ALARMS	INTRODUCTION OF AIS
COMPLETE INCOMING RADIO SIGNAL FAIL			A	A		A	A				R													R									X
RECEIVER PRE-AMPLIFIER OUTPUT FAIL	A		A	A		A	A				R													R									X
MAIN IF AMPLIFIER OUTPUT FAIL	A		A	A		A					R	A												R									X
SINGLE DEMODULATOR OUTPUT FAIL	A		A	A		A					R													R									X
BOTH DEMODULATOR OUTPUTS FAIL	A		A	A		A					R	A												R									X
INCOMING RADIO SIGNAL LOW (INITIAL)				A	A																												
LOW PRE-AMPLIFIER OUTPUT				A	A																												

PU: Power unit
EHT: Extra-high tension
PLO: Phase-locked oscillator
RSA: Remote station alarm

FIG. 12—Fault location chart for 11 GHz 140 Mbit/s digital radio equipment

field to deal with such incidents, as well as staff who have some experience of microwave maintenance. This depends entirely on staff being persuaded to use the information provided on the charts, and this point led to further development in the information made available to all staff on the network.

It seemed necessary to display the information on the charts in a way that would be both attractive to the user, portable, and could be used by each member of the staff on a personal basis at any time of the day without recourse to outside facilities. The answer to the problem is shown in Fig. 13, which depicts the fault locator specifically designed for use on the 11 GHz 140 Mbit/s 10/55 radio equipment. This locator consists of two discs, fixed at a central point, in such a way that one can rotate on top of the other. The top disc has, printed over an arc next to an open window, all of the heading information to be found on the charts previously discussed; that is, details of the LEDs found on the radio equipment. On the edge of an open window is a row of blank squares that the maintenance staff can use to enter, with a water-soluble pen, the pattern of LEDs observed for a particular fault condition. On the back disc, part of which can be seen through the window of the top disc, are rows of squares radiating from the centre on which are marked all the patterns of LEDs recorded on the fault chart earlier described. Also, against each radial set of squares is printed the fault condition that causes each pattern of LEDs. The operator, having recorded on the top disc the pattern of LEDs for a given fault, turns the top disc round until a similar pattern appears in the window. This then shows the cause of the fault.

Because, as previously mentioned, extra incidents not already shown on the charts and the locator are likely to occur, the facility shown in Fig. 13 is repeated on the reverse of the locator. The only difference is that the lines of squares radiating from the centre on the fixed disc are blank to allow such extra information to be recorded. The retention of the charts is necessary, as immediate recording of such incidents is more easily carried out on those, rather than on the locator.

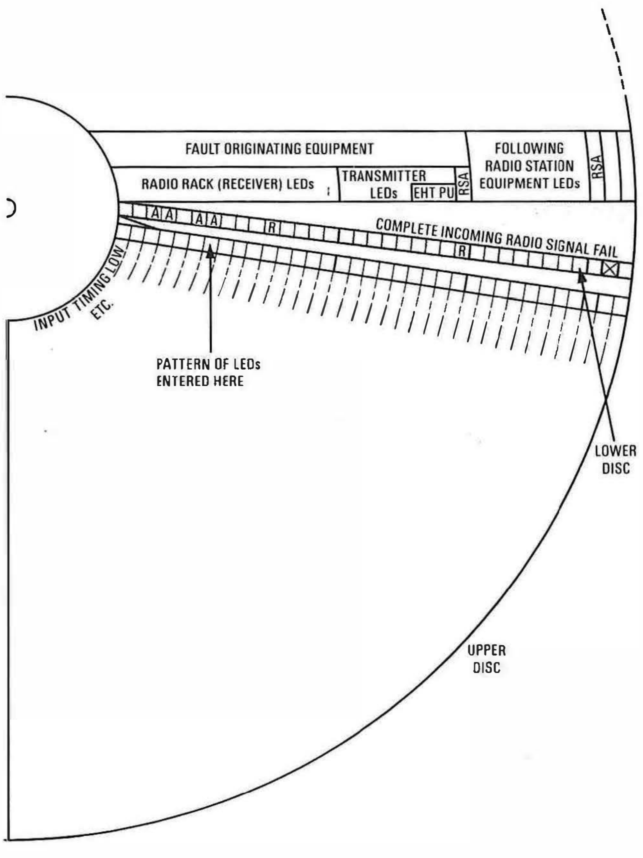


FIG. 13—Fault locator

Routine Faults

Certain degradations arise that are not sufficiently obvious to cause LED displays under normal propagation conditions. These reduce the margin available under fade conditions before they affect the traffic being carried on the channel.

One important routine maintenance test that has to be carried out at regular intervals by the maintenance staff is known as the *receive signal level and fade margin test*. The incoming signal level at the input to the receive frequency changer is first checked. If this level is as originally recorded, then a variable attenuator is introduced into the circuit at the point of measurement. The attenuator is then adjusted to reduce the level at the input to the receive frequency changer until LEDs start to indicate error rates. At an error rate of 10^{-5}, and again at 10^{-3}, the number of decibels introduced by the attenuator is noted and compared with a record held in the system log book. If either of these figures differs from the record, this indicates that the margin has changed; depending on the amount of change, and its stability, the reason for the change has to be investigated. Because there are no LEDs illuminated and no changes in the equipment meter readings, there are no obvious clues to the cause of the change recorded. Some other means of identifying the cause of the change has to be used in order to correct the problem before it affects the traffic on the channel.

This is where the eye diagram is most useful in diagnostic work. It should be emphasised that the basic radio-equipment parameters that control the performance of the channel are exactly the same as those on any other microwave radio-relay equipment, whatever traffic is being carried. The difficulty lies in the identification of the state of these parameters on the digital equipment. An example of a typical eye diagram displayed on an oscilloscope is shown in Fig. 14. This particular eye is for an undistorted channel; the overall voltage is within limits. The eye aperture is in excess of the minimum percentage allowed; the two halves of the eye above and below the cross-over point are symmetrical and are clear of noise. Under conditions where the margin previously referred to is changed under the test conditions applied, one or more of the basic radio parameters is degraded. As a result, one or more of the features of the eye diagram is affected. It is the interpretation of these eye changes that is the key to the identification of the radio parameter changes causing the problem.

Before the eye diagram can be used as a diagnostic tool, not only must a record be made of the fade margin figures

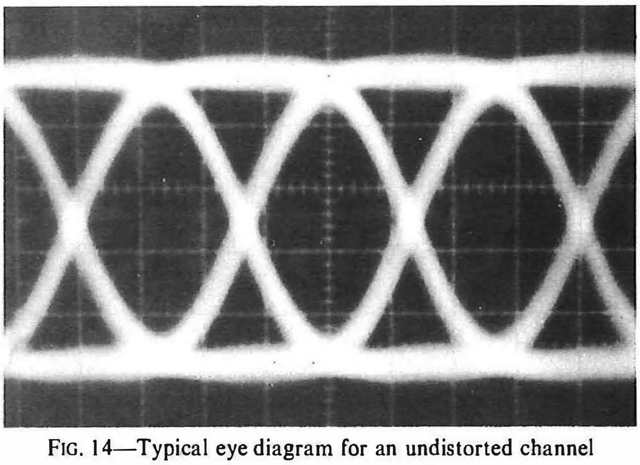
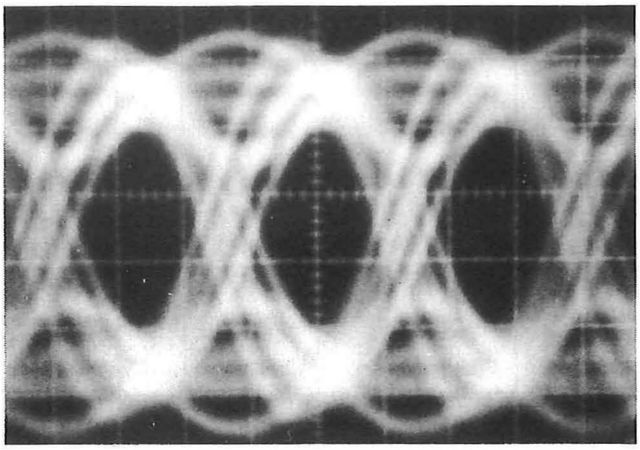


FIG. 14—Typical eye diagram for an undistorted channel

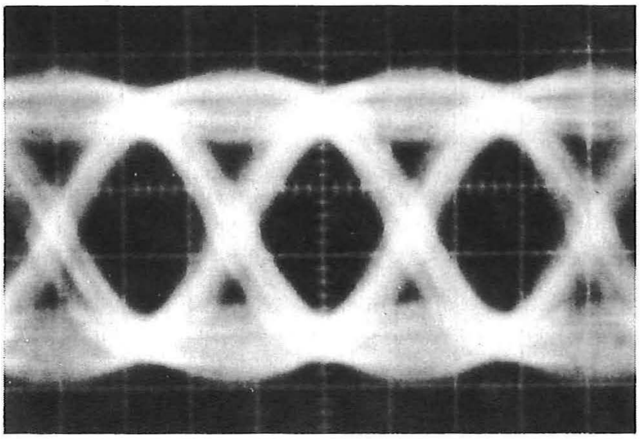


Note: Fade margin at an error rate of 1 in 10^5 is 37 dB

FIG. 15—Eye diagram as a result of multipath propagation on the preceding radio path

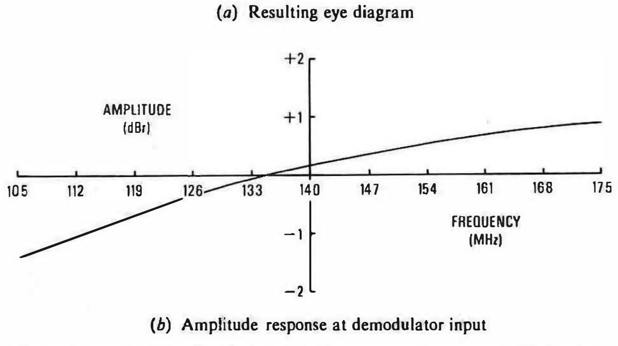
on hand-over to maintenance, but also photographs of the eye diagram and the interstation responses must be recorded at the same time. The initial use of the eye is to identify change rather than absolute values. Tests have now been carried out to show that changes can be detected, and that, in many cases, the causes of these changes can be positively identified.

Two examples of this eye behaviour and the causes are shown in Figs. 15 and 16 The fade margin for the eye diagram shown in Fig. 14 is 44·75 dB at an error rate of 10^{-5}. The diagram shown in Fig. 15 is as a result of multipath propagation on the preceding radio path and the fade margin in this case is 37 dB at an error rate of 10^{-5}. The fade margin test shows a marked change under these conditions, although, until the routine test was carried out, there was no indication on the equipment of any fault being present. This particular fault is not one that can be rectified but, in the second example shown in Fig. 16, the fault can be located and dealt with. The fault condition introduced into the channel is an IF/RF gain/frequency response distortion at the demodulator input (see Fig. 16(*b*)), and is not accompanied by a change in IF level. The fade margin under this fault condition has now changed to 40 dB at an error rate of 10^{-5}. Comparison of the eye diagram with the original eye in Fig. 14 shows that the eye has changed in two ways. Firstly, the overall voltage of the eye is reduced, and this is almost exclusively associated with the behaviour of the demodulator. Secondly, the eye trace has thickened, not as a result of noise, but as a result of multitraces close together, unlike those in Fig. 15. From this evidence, it is reasonable to centre the investigation on the demodulator. But these diagnostic techniques depend entirely on an adequate record of the original eye diagram being available to make a comparison.



Note: Fade margin at an error rate of 1 in 10^5 is 40 dB

(*a*) Resulting eye diagram



(*b*) Amplitude response at demodulator input

FIG. 16—Effect of IF/RF gain/frequency response distortion at the demodulator input

The eye diagrams shown in Figs. 15 and 16 are only two of a series of examples to be issued to maintenance staff on the digital radio network. The intention is that under the test circumstances described, or under other fault conditions difficult to diagnose, the eye diagram can be inspected and compared with the record of that channel, and with the set of behaviour examples. Further tests, as shown in Fig. 16(*b*), can then be carried out, if required, to clear the fault.

CONCLUSION

The object of maintenance on the microwave radio-relay network should be to provide quality and reliability of service to the customer in the most cost-effective manner. To assist in the achievement of this aim, it is necessary that fault recognition, location and correction should be carried out methodically and accurately; this is the purpose of the methods described in this article, and introduced into the network. There will, of course, be exceptions, but as long as the information supplied is used, serious difficulties or delays can be kept to a minimum.

BT National Networks has now produced a book which incorporates all the printed material referred to in this article, both for analogue and digital equipment, which, together with the digital fault locator, is now being issued to all staff concerned with the maintenance of the microwave radio-relay network.

Biographies

Cliff Dorkings joined the British Post Office (BPO) in 1966 as a Trainee Technician (Apprentice) in the Canterbury Telephone Area. After studying for his B.Sc., he moved to Telecommunications Headquarters as an Assistant Executive Engineer in 1972 to work on the maintenance of microwave radio systems. In 1977, he gained his M.Sc. in Electronics after three years of part-time study. On his promotion to Executive Engineer, in 1979, he worked on the maintenance of closed-circuit and cable television systems. In 1981, he transferred to his present job in the Trunk Networks Maintenance and Technical Support Division of British Telecom National Netorks, where he is involved with special fault investigations on the microwave network.

Arthur Hickson joined the BPO in 1937 as an external Youth-in-Training. Between 1940 and 1946, he served in the Royal Corps of Signals in the UK, Europe and West Africa. On his return to the BPO in 1946, he worked as a Technical Officer on transmission maintenance, and at the Charwelton Microwave Station. Between 1967 and 1972, he was a Regional Supervising Officer in the Midland Region, where he supervised the installation and testing of microwave systems. In 1973, he joined his present group in the now Trunk Networks Maintenance and Technical Support Division of British Telecom National Networks, where he has been concerned mainly with microwave-system overhauls and special faults. Since 1977, he has been responsible for demonstrating, in a mobile van, maintenance techniques on both analogue and digital microwave equipment throughout the UK. So far, over 1000 BT staff have attended these demonstrations.

British Telecom Press Notices

PRESTEL RECEIVES QUEEN'S AWARD FOR TECHNOLOGICAL ACHIEVEMENT

Prestel, British Telecom's (BT's) viewdata system, has been awarded the Queen's Award for Technological Achievement. The accolade goes to the Videotex Section of BT's laboratories at Martlesham Heath in Suffolk and to Prestel Headquarters in the City of London. Prestel, the world's first, largest and most comprehensive viewdata network, developed by BT, has opened up new horizons for home and business users.

Frank Burgess, Prestel's General Manager, said: 'We are very proud to receive this honour. It recognises the original development of viewdata and the continuing technical development of Prestel within our organisation over the years. Thanks to the hard work and dedication of everyone concerned with Prestel, our service remains the world's leading system, not only in technological achievement, but also in the applications which we have developed and which are now proving successful.'

Keith Clarke, now Deputy Director of the Martlesham research laboratories, was in charge of the early development of Prestel. Commenting on the award, he said: 'Under Sam Fedida, BT's research scientists, then based at Dollis Hill, London, made the important breakthrough in using a television set for purposes other than straightforward entertainment. They adapted new semiconductor technology to turn science fiction into economic possibility by using ordinary telephone lines.'

The first public demonstration of the system was made at Heathrow, London, in 1975, shortly after another important development—the adoption of a joint technical standard for the display format with the BBC and IBA. Prestel entered public commercial service in September 1979.

Now 98% of the world's viewdata and teletext systems in 28 countries are on the British standard, which has now been incorporated into the new European standard of the Confederation of European Post and Telecommunications Authorities. Britain is also consolidating its leading position through further research. Photovideotex is a new development by BT that will open the way for a variety of Prestel and cable-televison applications that require illustrations; for example, the display of house pictures by estate agents.

The Prestel network now has more than 42 000 terminals connected to it, and is growing by 1500 every month. The biggest growth has been in residential users. Compared with 18% in 1982, nearly 40% of all terminals are in homes.

Telephone lines are at the heart of the Prestel system. They link home or office terminals—televisions with inexpensive adaptors, home computers or special Prestel sets—to powerful computers in the network. These are either Prestel's own computers, on which an array of information sources—currently about 1200—place their data, or third-party computers linked through one of the 16 Prestel Gateways.

Unlike teletext services broadcast on television channels, Prestel allows users to talk back to computers, and to call up information of their choice instantly. This sophistication has paved the way for practical developments such as home banking, home shopping and a host of personalised services. A recent example of these practical applications is Farmlink, a service for farmers in the south-west of England, launched in April of this year. Like all Prestel services, Farmlink operates 24 hours a day. Other Prestel services which have helped to make the network such a success are listed below.

Pictured at the presentation of the Queen's Award for Technological Achievement are (left to right) David Merlo, Director of Research; The Lord Mayor of London, Dame Mary Donaldson; and Frank Burgess, the General Manager of Prestel.

Homelink A co-operative venture between BT, the Nottingham Building Society and the Bank of Scotland. Customers can check their building society and bank accounts from home, transfer funds, pay bills and shop from the comfort of their armchairs.
Micronet 800 A joint venture between Prestel and the Telemap subsidiary of the East Midlands Allied Press. It offers a comprehensive library of software and information to microcomputer users. It was launched in March 1983.
Skytrack An automated airline reservation service giving travel agents access to most of the world's airline companies' own computers. Hotel and car-hire bookings can be made instantly. About 95% of all members of the Association of British Travel Agents have Prestel sets.
Mailbox An electronic message service and Telex link, enabling users to send and receive correspondence. At present available on one of Prestel's London computers, Mailbox will be extended to the rest of the UK during 1984.

A recent expansion of Prestel's network means that 94% of Britain's telephone subscribers are within reach of the service at local telephone charge rates.

COMPUTERS TO AID CUSTOMERS

British Telecom (BT) has announced that it is going ahead with a major computing development programme to provide new business information systems for use in its local offices. The systems will be designed primarily to give customers a more responsive and effective service; therefore, they will encompass most aspects of the operations of the Telephone Areas of BT's Local Communications Services.

Other developments within BT, including the restructuring of Telephone Areas into Districts, have emphasised the need to replace existing systems to match the new organisation and its competitive environment.

The new systems, to be known as *customer services systems (CSS)*, will be integrated fully and controlled locally, to reflect the devolved nature of the new organisational structure. Each District will have its own computing centre, and there will be about 30 in all.

The initial development of the systems is to be undertaken by a joint team of staff from BT and Logica, with support from McKinsey and Company. The first phase, which will include on-line billing facilities and an order handling system, will be introduced in some Districts next year. All phases are planned to be fully operational within four years.

To maintain BT's policy of multiple sourcing of computing equipment, the systems will be developed for use with either ICL's VME operating system, or IBM's OS-MVS operating system. The proportion of Districts using each will depend on the competitive performance of the respective suppliers.

The Development of a Computer-Aided Contact Resistance Measurement System

B. WILTSHIRE, M.SC., PH.D., C.ENG., M.I.MECH.E., M.I.M.†, and J. W. COLLINS*

UDC 621.317.73 : 681.31

This article describes a computerised test system for measuring the resistance of electrical contacts. In operation, the test equipment is providing quick and accurate test results. In addition, many benefits have been derived from the system in terms of speed, accuracy and repeatability. Finally, a data-management software package has been developed to process the accumulated data. This program has enabled the results to be presented systematically.

INTRODUCTION

Continuous research is underway at British Telecom Research Laboratories to develop connection systems for the network. During development trials, connectors are normally subjected to a series of environmental tests to assess their service characteristics. Connector performance is measured mainly by monitoring changes of contact resistance. This method is used universally[1] and has been proved to give accurate indications of service performance. A contact resistance measurement system is, therefore, an important part of the contact evaluation process, particularly if the system has a good resolution and can deal with large batches of contacts.

This article describes the development of a computer-aided system for measuring contact resistance. The individual measurements are made by using the well-known 4-wire configuration, under dry-circuit conditions, where the current is fixed and the open-circuit potential is limited. The system has been designed so that all operations are carried out under software control, to ensure maximum speed of operation and accuracy of measurement.

CONTACT MEASUREMENT

The classical 4-wire measuring system is, in concept, simple. By applying a constant current through an unknown resistance R_x (that is, the test contact), and measuring the voltage drop, then Ohm's law can be applied to calculate the resistance R_x (Fig. 1). The term *4-wire* is used because two connections are used to supply current, and a further two connections are used as voltage probes. The resistance measurement occurs between the voltage probe tips, and so total lead lengths are not important. In fact, this makes it possible to carry out resistance measurements remote from control equipment.

The primary requirement for dry-circuit testing is that the voltage and current applied to the test sample must not exceed certain values, otherwise the interfacial conditions in the contact may be altered and a false measure of resistance obtained. These values can vary depending on the contact design and application. However, for contacts used in telecommunications, there is general agreement[2, 3, 4] that, with 100 mA applied current and open-circuit potential limited to 20 mV, the interfacial conditions will remain unchanged; thus, these values have been used as test parameters.

It is virtually impossible to make direct measurements of contact resistance because of the physical shape of connectors. Instead, measurements are normally made of the joint resistance between fixed positions. The resulting resistance value is, therefore, a combination of contact resistance and bulk resistance. However, during environmental testing, the bulk resistance cannot alter when measured at constant temperature, and so any changes in measured resistance must be caused by changes in contact resistance. Thus, the important performance criteria in contact evaluation is the change of resistance ΔR.

For test purposes, connectors are mounted, 20 at a time, on printed-wiring boards (Fig. 2). This configuration has been chosen to permit the test board to be retrieved from the testing schedule and plugged directly into the measurement equipment via an edge connector. The board is designed so that each connector position has an individual 4-wire circuit. The resistance is measured between the two inner pins, which are the voltage probes. This measured value therefore comprises: the contact resistance, the connector bulk resistance and the resistance of approximately 30 mm of wire.

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* Research Department, British Telecom Development and Procurement

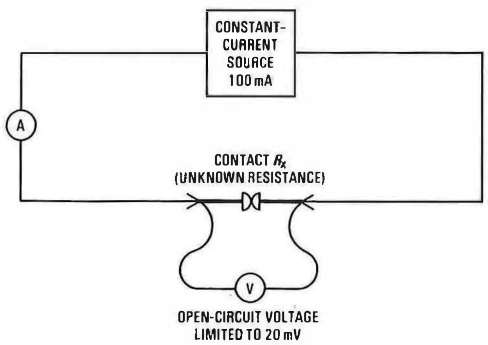


FIG. 1—Block diagram of 4-wire resistance measurement

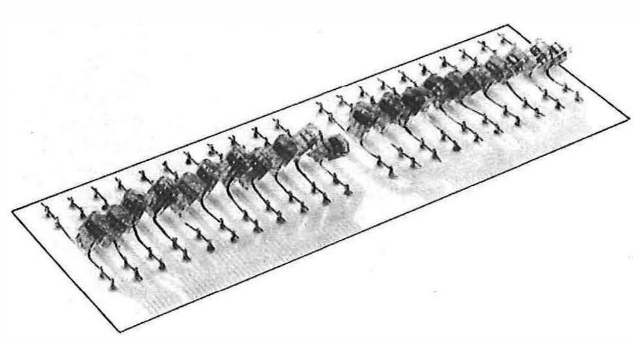


FIG. 2—Connector test board

For telephony contacts, the individual resistance values, for connectors wired in the configuration shown in Fig. 2, are normally in the range 5–10 mΩ. Of this value, the contact resistance contribution is approximately 0·1 mΩ. Experience has shown that, during environmental testing, the increase of resistance, for a well designed contact, will be less than 1 mΩ. However, for a defective contact, the rise in resistance could be in excess of 10 mΩ. It is important to note that, for any batch of identical contacts, the increase in resistance normally follows a small statistical distribution. In this situation, therefore, a small sample size, of say 20 contacts, can be justified. However, with this small sample size, the individual joint resistances must be measured on equipment that has a good resolution, in the micro-ohm range.

SYSTEM DESIGN

The resistance measurement system is based on a 32 Kbyte Commodore Business Machines Ltd. desk-top computer, with attached dual floppy-disc drive and printer. The associated measuring equipment consists of a micro-ohmmeter, a digital current monitor, a programmable data-logger and a tape punch/reader (Fig. 3). The system is designed so that the computer controls all of the equipment via an IEEE† -488 bus. The IEEE-488 bus allows full talker/listener communication protocols, so that the computer not only controls the test equipment, but also acts as a data highway to accept the measured resistance results. All of the equipment is powered from an uninterruptable power supply; this consists of a battery back-up supply that provides 10 min regulated AC supply at mains voltage and frequency if mains failure occurs.

The equipment is housed in a temperature-controlled room, which is maintained at 20°C to minimise any changes in resistivity due to temperature variations occuring during measurement. The resistance temperature coefficient for copper and aluminium, for example, is approximately 0·4% per degree centigrade, and so variations of several degrees would shift the measured joint resistance significantly.

† Institute of Electrical and Electronics Engineers, Inc.

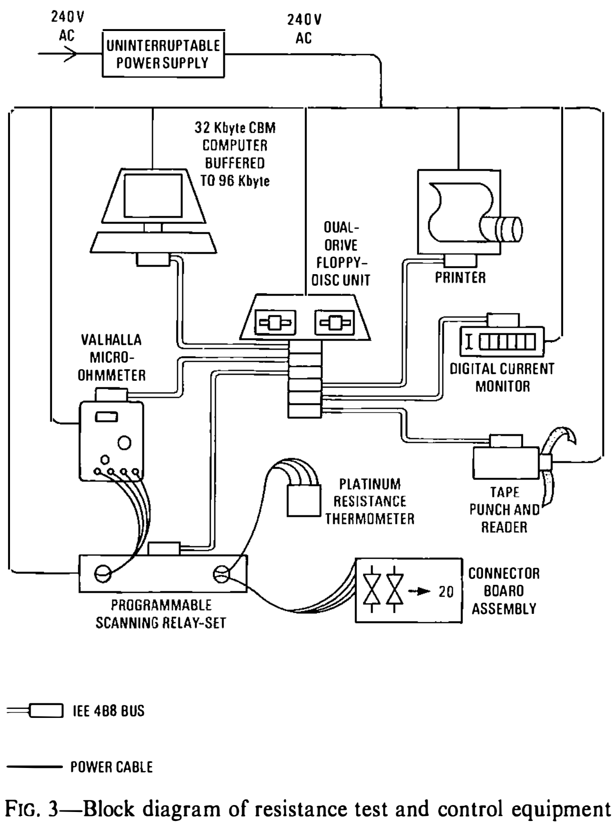


FIG. 3—Block diagram of resistance test and control equipment

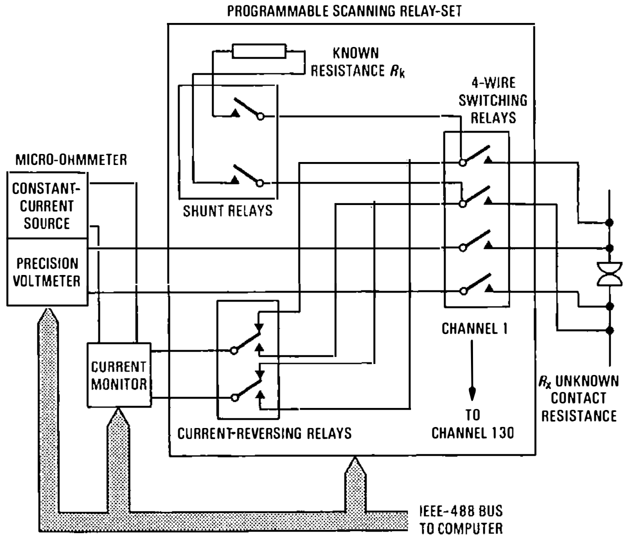


FIG. 4—Programmable relay operation

HARDWARE

The heart of the measurement system is the Valhalla micro-ohmmeter, which is a composite assembly consisting of two separate circuits: one for current and the other for voltage. Both of the circuits are electrically isolated, except under test conditions. The design of the constant-current circuit is such that it is capable of generating 100 mA in the unknown contact resistance R_x, regardless of its value. The internal voltmeter is used subsequently to measure the potential drop across R_x when current is applied. When the equipment is used in the 0–200 mΩ range, which is appropriate for most designs of telecommunications contact, the measurement resolution is 10 μΩ. The programmable scanner consists of sets of latching-relay cards, which can provide switching for up to 130 separate contacts, and is designed to switch sequentially through a batch of contacts. Each relay can be individually addressed from the computer via the interface, and commanded to close or open.

Fig. 4 shows that there are 3 types of relay function in the scanning set, as follows:

(*a*) *Current-reversing relays* These are operated to enable 2-way measurement of resistance to be made.

(*b*) *Shunt relays* These are used to connect a known resistance R_k across the unknown R_x. In equipment such as this, the output of the constant-current source is normally shunted by the fixed resistance R_k to limit open-circuit voltage during switching and initial joint resistance measurement.

(*c*) *Four-wire switching relays* These relays are of the 4-pole cradle type, and are used to extend 4-wire measurement out to the test contact.

SOFTWARE MEASUREMENT PROGRAM

The ASTM† standard for measuring the resistance of low-current contacts[1] sets down a number of measurement criteria, some of which have been described earlier. One further important factor is that the resistance of the contact must be measured in both directions, to cancel out any thermocouple or semiconductor effects that may occur. The final resistance result is taken to be the average of the two readings. In the system being described, some additions are made to this requirement; further readings are taken in each direction, and their values compared to determine whether they are within a given tolerance. This is done, both to ensure greater accuracy of measurement, and to cancel out

† American Standard for the Testing of Materials

any problems due to airborne or circuit-borne spikes, which have caused problems in the past.

The software program is designed to carry out the measurement of resistance, as shown in Fig. 5. The necessary steps are described for carrying out the standard test of contacts having a resistance in the range of 0–200 mΩ, with open-circuit potential limited to 20 mV. For clarity, the sequence of operations for measuring a single contact is described.

Firstly, the contact, having the unknown resistance R_x, is connected in parallel with the internal known resistance R_k, to ensure that no current is applied across an open circuit; the current source of 100 mA is then applied to the circuit. If zero current is detected by the digital current monitor, which is connected in series with R_x, then NC (not connected) is logged in the computer memory and the system moves on to the next contact. If R_x is in place, measurement can proceed.

The shunt relay is opened and a current of 100 mA is applied to resistance R_x. Subsequently, the voltage drop across R_x is measured and the resistance calculated. If the resistance is over 200 mΩ, a ***high resistance*** indication is given and the system is moved to the next contact. The original channel number is logged in the memory and the contact will, eventually, be measured under the non-standard program.

If R_x is below 200 mΩ, a resistance reading is taken in one direction (R_{x1}), followed by a second reading (R_{x2}), taken in the same direction. The two readings are compared and, if they are within a given tolerance of (5% + 0·5 mΩ), the value of R_{x1} is logged. This composite tolerance value has been chosen to give realistic comparisons of sequential readings at both low and high values of joint resistance. Alternatively, if the two readings are outside the tolerance, a third reading (R_{x3}) is taken, and compared with R_{x1} and R_{x2} in succession. If one pair of readings falls within the above tolerance, the appropriate reading is logged. If none of the readings is within tolerance, the program is stopped and an error notation given.

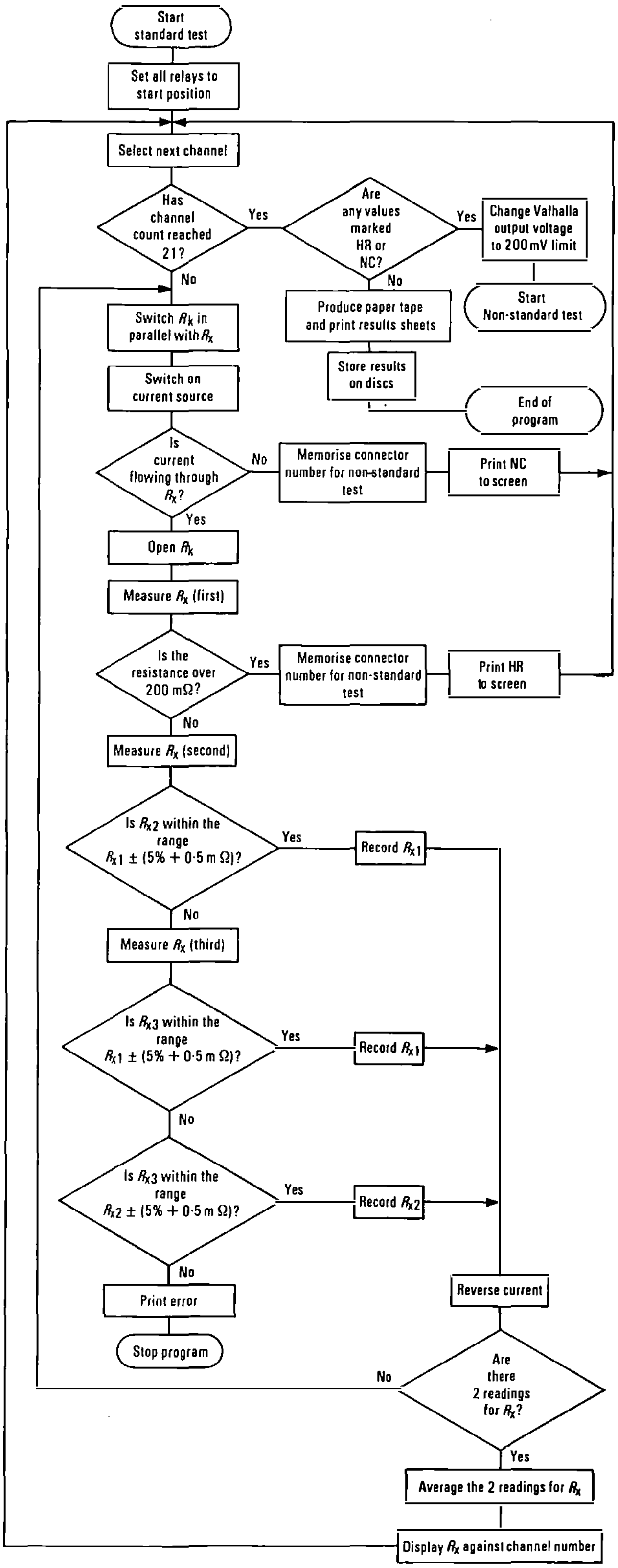
Once a successful reading has been logged, the current-reversing relays are operated and the same sequence of measurement is carried out with the current flowing through R_x in the reverse direction. Finally, the average of the 2-direction readings is calculated and printed out as the resistance of the joint.

Once the 20 connections have been measured by the standard program, the system is switched to the non-standard measurement program (see Fig. 6). This does not conform to the dry-circuit specification of the ASTM standard, but, nevertheless, allows a reasonably accurate resistance measurement to be made. In this part of the program, the contacts, which have a resistance in the range 200 mΩ to 2 Ω, are selected in numerical order and measured. The test sequence is the same as for the standard run and uses the same current (100 mA), but the open-circuit voltage limit is increased to 200 mV.

SYSTEM MANAGEMENT

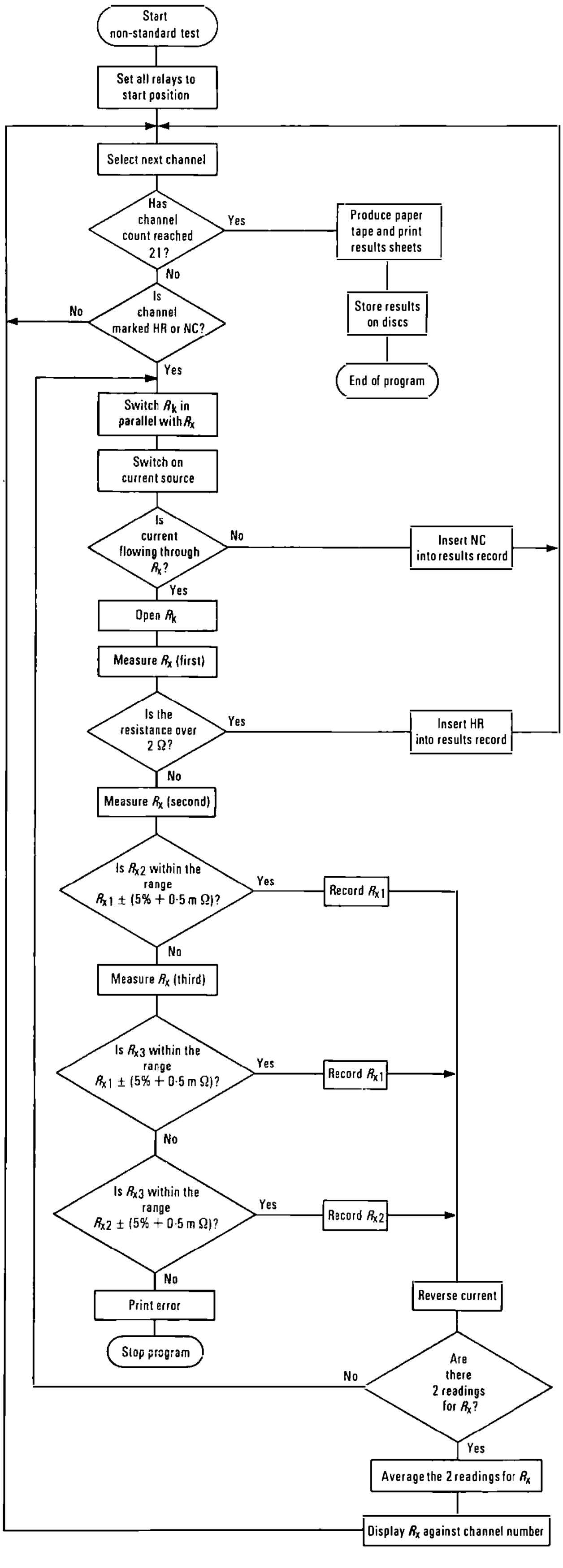
The complete dual computer program, for both measuring contacts and collating data, has been designed to be simple in operation. Operator interaction sequences have been kept as simple as possible, so that no particular computer skill is required to operate the system. The tests and routines can be applied with minimal operator input; indeed, in many cases, the selection of one option from a menu of operational options is all that is required.

The system is designed to be used on a daily basis. For each daily use, there are some disc insertion routines to be carried out, as follows. Initially, a pre-programmed disc must be loaded, to buffer the computer memory from 32 Kbyte to 96 Kbyte of random-access memory (RAM). Secondly, the contact-measurement and data-management program has to be loaded. The computer is now set up to carry out all required contact resistance measurement and data-handling processes. It remains only to insert two results discs. For security purposes, two discs are used to receive identical results information, in case of disc damage.



NC: Not connected
HR: High resistance

FIG. 5—Standard resistance measurement software program



NC: Not connected
HR: High resistance

FIG. 6—Non-standard resistance measurement software program

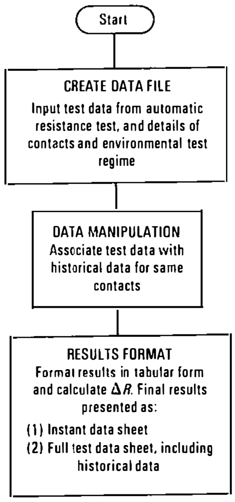


FIG. 7—Data-management software program

A further security measure is provided in the form of punched tape. Each time a test board is measured, a punched tape is generated as a record of the results. This tape can then be fed back into the computer, should any of the results discs be damaged.

The data-management program, shown in Fig. 7, not only assembles the collected data, but also collates it with historical data for the scanned set of contacts. Details of the data-management program are not discussed here, but, basically, it involves the use of standard file-handling techniques. The nature of environmental testing is such that it is necessary to accumulate data about a set of contacts over a number of months. Therefore, the historical results need to be readily accessible for the system to work efficiently; this is achieved by consigning the results for a number of test boards to a disc (or pair of discs).

With a capacity of 256 Kbytes per disc, the complete history of environmental testing results for perhaps 50 or more boards can be accommodated on each disc. Therefore, when a board resistance is being read, it is simply necessary for the operator to retrieve from the store, the pair of discs pertaining to that board.

For this project, it was decided to produce tabular results in two forms. The first set of results (sheet 1) contains only the instant results from a measurement session. Included on this sheet is extra data such as the applied current to a resolution of 0·1 mA, the room temperature measurement, the forward resistance, the reverse resistance, and the average 2-way resistance. The second form of result (sheet 2), shown in Fig. 8, contains the historical results for each contact, but, more importantly, shows the change of resistance parameter ΔR.

CONCLUSION

A computer-aided resistance measurement system has been developed for evaluating the dry-circuit performance of electrical contacts. Each contact is measured by using the classical 4-wire system, to ensure an accurate reading and to conform to international measuring standards. In addition, a data-management software package has been developed to handle the accummulated test data.

The results show that the computer-controlled test equipment provides quick and accurate resistance test results. In addition, many benefits have been derived from the measurement system in terms of speed, accuracy and repeatability. Finally, the data-management software program has made possible the efficient and systematic presentation of results.

B.T.RESEARCH LABS.R.1.2.2
CONNECTOR: XXX
CONDUCTOR: XXX
TOOL: XXX
NOTES: XXX
: XXX
: XXX
: XXX
TESTS: temp cy - heat sk - ind atm - vib tst - slt spr - hum con

BOARD NUMBER 9999
KEY:R=mohms.Vmax(Ø-2ØØmohms)=2ØmV
I=1ØØmA.Vmax(Ø.2-2ohms)=2ØØmV
N.C.=No Circuit.H.R=over 2ohms

date	Ø1jan83	Ø8jan83	14jan83	2Øjan83	14feb83	1Ømch83	Ø4apr83	29apr83	Ø5may83
test	initial	temp cy	temp cy	temp cy	heat sk	heat sk	heat sk	heat sk	ind atm
data	reading	1ØØ cys	2ØØ cys	3ØØ cys	25 dys	5Ø dys	75 dys	1ØØ dys	7 dys
Ø	1Ø.86	1Ø.85	1Ø.87	1Ø.87	1Ø.86	1Ø.86	1Ø.84	1Ø.85	1Ø.88
1	1Ø.69	1Ø.68	1Ø.69	1Ø.68	1Ø.67	1Ø.69	1Ø.7	1Ø.68	1Ø.7
2	11.17	11.19	11.19	11.18	11.19	11.19	11.19	11.19	11.18
3	1Ø.88	1Ø.87	1Ø.87	1Ø.89	1Ø.88	1Ø.87	1Ø.88	1Ø.89	1Ø.87
4	1Ø.9	1Ø.88	1Ø.91	1Ø.89	1Ø.9	1Ø.92	1Ø.91	1Ø.89	1Ø.9
5	1Ø.69	1Ø.69	1Ø.68	1Ø.69	1Ø.68	1Ø.68	1Ø.69	1Ø.69	1Ø.72
6	1Ø.42	1Ø.43	1Ø.42	1Ø.43	1Ø.42	1Ø.43	1Ø.41	1Ø.41	1Ø.43
7	1Ø.49	1Ø.48	1Ø.48	1Ø.49	1Ø.48	1Ø.49	1Ø.5	1Ø.48	1Ø.48
8	1Ø.85	1Ø.84	1Ø.83	1Ø.83	1Ø.83	1Ø.84	1Ø.83	1Ø.83	1Ø.84
9	1Ø.69	1Ø.68	1Ø.71	1Ø.71	1Ø.7	1Ø.7	1Ø.69	1Ø.68	1Ø.69
1Ø	1Ø.66	1Ø.65	1Ø.64	1Ø.67	1Ø.66	1Ø.66	1Ø.67	1Ø.65	1Ø.68
11	1Ø.89	1Ø.9	1Ø.89	1Ø.89	1Ø.87	1Ø.88	1Ø.89	1Ø.89	1Ø.9
12	11.Ø7	11.Ø6	11.Ø6	11.Ø6	11.Ø5	11.Ø6	11.Ø5	11.Ø7	11.Ø6
13	1Ø.91	1Ø.9	1Ø.91	1Ø.91	1Ø.9	1Ø.92	1Ø.92	1Ø.91	1Ø.9
14	1Ø.98	1Ø.97	1Ø.97	1Ø.98	1Ø.99	1Ø.99	1Ø.98	1Ø.97	1Ø.99
15	11.26	11.24	11.24	11.24	11.26	11.24	11.24	11.25	11.26
16	11.Ø5	11.Ø3	11.Ø6	11.Ø3	11.Ø2	11.Ø3	11.Ø5	11.Ø4	11.Ø3
17	11.22	11.22	11.22	11.23	11.23	11.23	11.23	11.22	11.21
18	11.13	11.13	11.12	11.15	11.13	11.14	11.14	11.13	11.13
19	11.11	11.Ø9	11.11	11.Ø9	11.11	11.11	11.1	11.1	11.13
2Ø	11.34	11.33	11.33	11.35	11.33	11.31	11.32	11.33	11.35
aver	1Ø.9	1Ø.89	1Ø.89	1Ø.9	1Ø.89	1Ø.9	1Ø.9	1Ø.89	1Ø.9

date	12may83	19may83	2Ømay83	27may83	Ø3jne83	1Øjne83	16jne83	22jne83	change
test	ind atm	ind atm	vib tst	slt spr	slt spr	slt spr	hum con	hum con	of
data	14 dys	21 dys	6 hrs	7 dys	14 dys	21 dys	6 dys	12 dys	resist
Ø	1Ø.84	1Ø.86	1Ø.84	1Ø.86	1Ø.85	1Ø.87	1Ø.84	1Ø.87	.Ø1
1	1Ø.68	1Ø.69	1Ø.7	1Ø.7	1Ø.69	1Ø.69	1Ø.71	1Ø.67	-.Ø2
2	11.21	11.18	11.18	11.18	11.2	11.2	11.19	11.2	.Ø3
3	1Ø.89	1Ø.88	1Ø.89	1Ø.88	1Ø.88	1Ø.88	1Ø.87	1Ø.87	-.Ø1
4	1Ø.89	1Ø.9	1Ø.89	1Ø.93	1Ø.91	1Ø.91	1Ø.89	1Ø.92	.Ø2
5	1Ø.69	1Ø.69	1Ø.68	1Ø.71	1Ø.69	1Ø.68	1Ø.68	1Ø.67	-.Ø2
6	1Ø.44	1Ø.42	1Ø.42	1Ø.43	1Ø.42	1Ø.41	1Ø.43	1Ø.42	Ø
7	1Ø.48	1Ø.49	1Ø.48	1Ø.5	1Ø.49	1Ø.5	1Ø.49	1Ø.48	-.Ø1
8	1Ø.84	1Ø.84	1Ø.86	1Ø.86	1Ø.85	1Ø.85	1Ø.83	1Ø.83	-.Ø2
9	1Ø.7	1Ø.7	1Ø.68	1Ø.68	1Ø.69	1Ø.68	1Ø.69	1Ø.7	.Ø1
1Ø	1Ø.66	1Ø.67	1Ø.68	1Ø.65	1Ø.66	1Ø.65	1Ø.67	1Ø.64	-.Ø2
11	1Ø.88	1Ø.89	1Ø.88	1Ø.9	1Ø.89	1Ø.91	1Ø.88	1Ø.89	Ø
12	11.Ø8	11.Ø6	11.Ø8	11.Ø9	11.Ø7	11.Ø8	11.Ø6	11.Ø7	Ø
13	1Ø.91	1Ø.93	1Ø.91	1Ø.93	1Ø.91	1Ø.91	1Ø.92	1Ø.89	-.Ø2
14	1Ø.98	1Ø.98	1Ø.98	11	1Ø.97	1Ø.97	1Ø.98	1Ø.98	Ø
15	11.25	11.24	11.25	11.27	11.27	11.26	11.23	11.25	-.Ø1
16	11.Ø6	11.Ø5	11.Ø4	11.Ø6	11.Ø3	11.Ø4	11.Ø5	11.Ø3	-.Ø2
17	11.23	11.23	11.22	11.21	11.21	11.21	11.22	11.22	Ø
18	11.13	11.14	11.14	11.15	11.14	11.13	11.13	11.13	Ø
19	11.11	11.11	11.1	11.11	11.1	11.11	11.11	11.1	-.Ø1
2Ø	11.34	11.34	11.35	11.33	11.35	11.34	11.32	11.34	Ø
aver	1Ø.9	1Ø.9	1Ø.9	1Ø.9	1Ø.9	1Ø.9	1Ø.89	1Ø.89	-.Ø1

FIG. 8—Full test data result sheet

ACKNOWLEDGEMENTS

The authors wish to thank all those who have assisted in the project, in particular, David Butler and Bill Bellhouse, who carried out the early development work on which the present system is based; and PPM Ltd., who compiled the software and supplied the programmable scanning unit.

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[3] JEDYNAK, L. Instrumentation for Measuring Dry-Circuit Contact Resistance. Proceedings, 1974 Holm Conference on Electrical Contacts, Chicago, USA.

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Biographies

Bruce Wiltshire is with Bell Northern in Canada. After receiving his Ph.D. degree in Metallurgy at Cambridge University, he joined British Telecom Research Laboratories (BTRL) in 1980 and worked in the External Plant Research Section. He is currently specialising in the design and development of connectors for the telephone network. He has published a number of research papers on contact physics and connector technology.

John Collins is an Assistant Executive Engineer (AEE) in the External Plant Research section at BTRL. He joined BT in 1964 as a Youth in Training in the London West Telephone Area. He progressed to Technical Officer in external planning and development and, in 1973, joined Research Department, where he became an AEE. He has worked on pulse-code modulation terminal equipment and datel modem equipment, and is currently engaged on the design and development of external connector systems. He is responsible for the commissioning and everyday running of the computer-controlled measuring equipment.

Profiles of Senior Staff

MANAGING DIRECTOR LOCAL COMMUNICATIONS SERVICES

I. D. T. VALANCE M.SC., B.A.

Iain Valance was born in London in 1943 and spent much of his early life in Scotland, where his father was Director of the Post Office.

After receiving his B.A. degree from Brasenose College, Oxford, he joined the North West Region of the British Post Office (BPO) in Manchester as an Assistant Postal Controller. Two years later he moved to Postal Headquarters in London. In 1970, he became a postgraduate of the London Business School. After being awarded an M.Sc. in 1972, he joined the Financial Policy Division; a year later he was appointed Personal Assistant to the Chairman of the BPO.

He was made head of the Financial Policy Division in 1975; the following year he had the distinction, held to this day, of becoming the youngest director ever to be appointed by the BPO, when, at 32 years of age, he became Director of Central Finance.

In 1978, he became Director of Telecommunications Financial Planning, a newly created post which he held until he took over the directorship of Procurement Executive's Materials Department in 1979.

He joined the Board of British Telecom (BT) in 1981 and was made responsible for co-ordinating BT's organisation and business systems. After a spell as Assistant Managing Director of the then Inland Division, he was appointed to the post that he currently holds, that of Managing Director of Local Communications Services, BT's largest division.

He is a Trustee of BT's Staff Superannuation Scheme, a director of Postel, and a Vice-President of the Institution of British Telecommunications Engineers.

DEPUTY ENGINEER-IN-CHIEF

D. M. LEAKEY, B.SC.(ENG.), PH.D., D.I.C., F.C.G.I., F.ENG., F.I.E.E.

David Leakey studied for his first degree at Imperial College, London University, and graduated in 1953. After a two-year apprenticeship with GEC Coventry, he returned to Imperial College on a GEC scholarship and was awarded a doctorate in 1958. From 1958 to 1963 he worked at the GEC Hirst Research Centre on the design of electronic telephone exchanges. In 1963, he returned to GEC Coventry to take charge of the Electronic Exchange Development Group. In 1965, he became Technical Manager of the Public Exchange Division in charge of all the development and contract engineering of the Division.

In 1966, he left GEC to become an academic, but returned in 1967 to take up the post of Head of Advanced Product Planning, and then to become Technical Director of GEC Telecommunications Ltd., a position he held until the end of 1983. During this period, he also became a Director of other GEC Companies, including Marconi Electronic Devices Ltd. and GEC Software Ltd.

In 1979, he became a Fellow of the Fellowship of Engineering. He is also an active member of the Institution of Electrical Engineers (IEE), having been elected a Member in 1963 and a Fellow in 1969. He has been a member of Council, Chairman of the Electronics Division, Chairman of the Management and Design division, and a Member of several of the Council's Boards and Committees. With effect from 1 October 1984, he will be a Vice-President of the IEE. He is, and has been, involved with several Government Committees, including the Focus Committee concerned with standards for data communication systems.

He joined British Telecom as the Deputy Engineer-in-Chief in January of this year.

MANAGING DIRECTOR BRITISH TELECOM ENTERPRISES

C. CROOK F.ENG., M.I.E.E., M.I.E.E.E., M.I.E.R.E., M.A.C.M., DIP.EE., ACT.(HONS) LIV.

After working for NASA on the guidance system for the Apollo moon landing, Colin Crook returned to the UK and worked for Plessey while obtaining his degree in computers and control theory.

Subsequently, he worked for Motorola in Geneva as the head of their computer laboratory for Europe, and then as Computer Marketing Manager. He was promoted to the post of Product Marketing Manager for their large-scale integration operation in the United States; he was the first non-American to hold a senior position in the company. As Group Operations Director, he was responsible for Motorola's microcomputer business. including the development of the MC6800 and the MC68000 series of products.

In 1979, he returned to the UK and took up the position of Managing Director of Rank Precision Industries; in 1980, he initiated a new trading division—Zynar BV—of the Rank Organisation, specialising in local area networks and software.

He joined British Telecom in October 1983 as Board Member and Managing Director of British Telecom Enterprises. He is a Vice-President of the Institution of British Telecommunications Engineers.

DIRECTOR RESEARCH

D. MERLO, B.SC.(ENG.), C.ENG., F.I.E.E.

David Merlo joined the Research Department, when it was situated at Dollis Hill, as a Youth-In-Training in 1948 and later transferred to the Radio Branch. In 1954, he obtained a first class honours degree in electrical engineering at London University, mainly by part-time study at Northampton Polytechnic, while pursuing his first main job of assisting in the development of the Australian speaking clock. His appointment as an Assistant Engineer involved him for a short time on the design of test equipment for the first transatlantic cable.

He was promoted to Executive Engineer in 1955 and joined the Radio Branch (also at Dollis Hill) where he worked on filter and network design, a task that gave him experience of the frequency band from long waves to microwaves. He designed networks for, and worked on, several radio systems, including microwave links, and in particular the London to Isle of Wight system. He then became heavily involved in part of the work of planning, constructing and installing the first aerial at the Goonhilly Downs satellite earth station, and its subsequent commissioning and operation during its early experimental phases. During this period, he was appointed Senior Scientific Officer with additional responsibilities for the development of coaxial and microwave measurement techniques and standards.

In 1965, he was promoted to Principal Scientific Officer to lead the Microwave System Studies section where the work, over a period of a few years, changed in character from satellite and analogue microwave systems to digital terrestrial microwave systems. In 1974, he was promoted to Head of the Millimetric Waveguide Division, a post in which he was responsible for the completion of work leading to the successful field trial in Suffolk of a 14 km low-loss circular waveguide system with a potential transmission capacity of half a million simultaneous 2-way telephone conversations. For a time, he also concurrently led a division concerned with human factors, customers' equipment and transmission performance.

He was appointed Deputy Director of Research in 1977, in which capacity he had responsibility for the divisions concerned with radio, satellite and speech communications, videotex, human factors and the centre for visual telecommunications. He was appointed Director of Research in October 1983.

Institution of British Telecommunications Engineers

(formerly Institution of Post Office Electrical Engineers)

General Secretary: Mr. J. Bateman, National Networks Strategy Unit (NNSU1.4.2), Room 620, Williams National House, 11–13 Holborn Viaduct, London EC1A 2AT; Telephone: 01-357 3858.
(Membership and other local enquiries should be directed to the appropriate Local-Centre Secretary as listed in the October 1983 issue.)

CHURCHILL TRAVELLING FELLOWSHIPS—THE CHANCE OF A LIFETIME

Applications for the 1985 Churchill Travelling Fellowships are now being invited. All UK citizens irrespective of age or occupation can apply for these awards and, as no educational or professional qualifications are needed, they are of special interest to people who are not eligible for other types of grants. The object of the awards is to give men and women from all walks of life the chance to gain a better understanding of the lives and work of people overseas and to acquire knowledge and experience for the benefit of their work and the community. The only requirement is that candidates must be able to show that they can make effective use of the knowledge and experience they have obtained abroad.

Grants are offered in different categories each year; anyone whose trade, profession or interest falls within these categories can propose a project that they wish to carry out in countries outside the UK. About 100 awards are made annually, and there are now over 1700 Churchill Fellows. Categories for 1985 include the micro-electronics industry and 'the Churchill connection' — open to anyone with a project related to Sir Winston Churchill's varied interests: history, journalism, painting, animals, bricklaying, etc.

Those wishing to apply should send a stamped addressed envelope between August and mid-October this year to the Winston Churchill Memorial Trust, 15 Queen's Gate Terrace, London SW7 5PR. The Trust will send an explanatory leaflet and a form to complete, which must be returned to the Trust Office by 31 October 1984. Applications received after this date will not be accepted, and allowance should be made for postal delays.

The final selection will be made by interview in London in January 1985. Successful candidates will be expected to start their travels during that year, making their own plans and arrangements within the scope of the grants. The grant will cover return air fare, and all travel and living expenses abroad for about eight weeks.

IBTE CENTRAL LIBRARY

The books listed below have been added to the IBTE library. Any member who does not have a copy of the 1982 edition of the library catalogue can obtain one on loan from The Librarian, IBTE, 2–12 Gresham Street, London EC2V 7AG. Library requisition forms are available from the Librarian, from Local-Centre and Associate Section Centre Secretaries and representatives. The form should be sent to the Librarian. A self-addressed label must be enclosed.

5359 *Microprocessors: Your Questions Answered.* A. Wood (1982).

This book covers the subject of microprocessors from binary arithmetic to hardware and software in plain down-to-earth language.

5360 *The Sinclair ZX81—Programming for Real Applications.* R. Hurley (1981).

With this book, the reader is able to learn, a step at a time, how complex programs are constructed and written for practical applications such as finance and education.

5361 *Digital Telephony.* J. C. Bellamy (1982).

This book is both a reference work for telecommunications engineers and a text for graduate-level engineering and computer-science students. The book provides an introduction to all aspects of digital communication, with emphasis on voice applications, voice digitalisation, digital transmission, digital switching, network synchronisation, network control and network analysis.

5362 *Computing Using Basic—An Interactive Approach.* T. Cope (1981).

5363 *Solar Cells—Operating Principles, Technology and Systems Applications.* M. A. Green (1982).

This book gives in-depth descriptions of the basic operating principles and design of solar cells, and examines the techniques currently used to produce solar cells.

5364 *Materials and Processes in Electronics.* C. E. Jowlett (1982).

This book describes the best candidate materials, processes and fabrication methods for a given design of modern electronic device.

5365 *Electronic Equipment Reliability.* J. C. Cluley (1981).

This book describes the principles of assessing the reliability of electronic equipment, including the mathematical background and the methods that can be used to improve reliability.

5366 *Heating, Ventilating and Air Conditioning—Analysis and Design.* F. C. McQuiston, and J. D. Parker.

This book takes the reader step by step through all the basic procedures of designing heating, ventilating and air-conditioning systems.

5367 *Electrical and Electronic Applications.* N. Morris (1982).

This book provides basic information on light-current and heavy-current applications, and covers the requirements of a first technician course.

5369 *The Restoration of the Tidal Thames.* L. B. Wood (1982).

This fascinating book gives a definitive account of the water quality in the Thames tideway from early times to the present day.

5370 *Advanced Engineering Mathematics.* L. D. Kovach (1982).

5371 *Engines, Energy and Entropy—A Thermodynamics Primer.* J. B. Fenn (1982).

5372 *Plumbing and Heating Calculations.* F. Hall (1982).

This book covers topics such as pipe size, heat loss, boiler power and pump duty; it is not only ideal for students but also suitable for the practical plumbing or heating engineer.

5373 *Building Design and Construction Handbook,* fourth edition. F. S. Merritt (1982).

5374 *Digital and Microprocessor Engineering.* S. J. Cahill (1982).

This book presents the topics of random and microprocessor logic in an unusually unified manner, with particular emphasis on design stratagems based on current integrated circuits.

5375 *The Design of Electrical Services for Buildings.* F. Porges (1982).

This book covers methods of wiring, schemes of distribution and protection for lighting, and power installations for buildings of all types, including large commercial and industrial installations.

5376 *Site Surveying and Levelling—Level 2.* H. Rawlinson (1982).

This book is an introduction to elementary surveying and surveying techniques. It is specifically designed to cover the topics in the Business and Technician Education Council's (BTEC's) level-2 standard unit Site Surveying and Levelling, and most of the City and Guilds of London Institute's craft supplementary course Surveying and Levelling.

5377 *Mathematics for Technicians—Level 2.* D. J. Hancox (1982).

This is the second in a series of three books by the same author designed to provide the mathematics for BTEC Certificate courses. It covers the following three level-2 half-units, which were all revised in 1980: Mathematics II, Mensuration II and Analytical Mathematics II.

5378 *Applying Mathematics—A Course in Mathematical Modelling.* D. N. Burghes, I. Huntly, and J. McDonald (1982).

5379 *Integrated Optics: Theory and Technology.* R. G. Hunsperger.

This book is an introduction to the theory and technology of integrated optics for graduate students in electrical engineering, and for practicing engineers and students who wish to improve their understanding of the principles of, and applications in, this relatively new and rapidly growing field.

5380 *Contact with the Stars.* R. Brewer (1982).

This fascinating book details man's search for extraterrestrial intelligence and the chances of communication with other civilisations in this galaxy.

5381 *Students Must Write—A Guide to Better Writing in Course Work and Examinations.* R. Barrass (1982).

5382 *Building—Finishes, Fittings and Domestic Services.* R. Chudley (1982).

This book covers basic non-structural elements, finishes above ground level, and simple domestic services installations, and discusses these topics in relation to current concepts in construction technology. The text is complemented by numerous illustrations.

P. W. SALTER
Librarian

Forthcoming Conferences

Further details can be obtained from the conferences department of the organising body.

Institution of Electrical Engineers, Savoy Place, London WC2R 0BL.
Telephone 01-240 1871

CONFERENCES

IFIP Conference on Human-Computer Interaction
4–7 September 1984
Imperial College, London

Mobile Radio Systems and Techniques
10–13 September 1984
University of York

Computers in Communication and Control (EUROCON 84)
26–28 September 1984
Brighton

Computer-Aided Engineering
10–12 December 1984
University of Warwick

HF Communication Systems and Techniques
26–27 February 1985
Institution of Electrical Engineers

Telecommunication Transmission
18–21 March 1985
Institution of Electrical Engineers

Advances in Command Control and Communication Systems: Theory and Applications
16–18 April 1985
Bournemouth

VACATION SCHOOLS

Satellite Communication Systems Planning
2–7 September 1984
University of Surrey

Industrial Digital Control Systems
9–14 September 1984
Balliol College, Oxford

Integrated-Circuit Testing
9–14 September 1984
Brunel University

The Evolving Local Telecommunication Network
16–21 September 1984
University of Aston

Institution of Electronic and Radio Engineers, 99 Gower Street, London WC1E 6AZ.
Telephone 01-388 3071

Telesoftware
27–28 September 1984
Cavendish Conference Centre, London

Custom VLSI for Control and Instrumentation
6–7 November 1984
Cavendish Conference Centre, London

Colour in Information Technology
25–29 March 1985
University of Surrey

Digital Processing of Signals in Communications
22–25 April 1985
University of Loughborough

Oyez Scientific and Technical Services Ltd., Bath House (Third Floor), 56 Holborn Viaduct, London EC1A 2EX.
Telephone 01-236 4080

Mobile Communication Systems
18–19 October 1984
London

British Telecom Press Notice

CALMS

Britain's householders may soon have British Telecom (BT) to help them fight rising energy bills, if trials begun in March of this year are successful. Householders in the trials can save money by allowing certain electrical appliances to be switched off automatically for short periods to reduce peak loads on the network. This is accomplished by means of information sent cheaply over householders' telephone lines when they are not being used for telephone calls.

They will also be able to pay their gas, electricity and water bills automatically by telephone. They will be able, simply by pressing a button, to send instructions over the telephone line to a computer telling the gas, electricity or water board to transfer money from their bank accounts.

This revolutionary development in information technology, known as *Credit and Load Management Systems (CALMS)*, has been developed by South Eastern Electricity Board engineers. The CALM unit—*Energy Minder*—is a push-button microchip metering and control unit.

Trials are now being held in 300 homes at Brookwood near Guildford, Ilkeston near Derby and Kingswinford near Dudley, by the South Eastern, East Midlands and Midlands Electricity Boards. Each householder's Energy Minder is connected over BT's telephone network to special equipment provided by BT at the local exchanges. This communicates with the electricity board's computers by private circuits. Once a quarter, the Energy Minder is called up automatically over the telephone line by the electricity authority's computer to collect the recorded electricity consumption and to calculate the customer's bill.

If the customer agrees, the computer can be programmed to send a signal telling the Energy Minder to switch off, for short periods, some domestic appliances (such as immersion heaters and freezers). This will help the electricity boards to cut the short-term expensive methods of generation that they have to use during the peak periods of demand. The boards will be able to charge less for the electricity provided on this 'interruptible' basis.

The Energy Minder has also been designed so that it can monitor gas and water supplies. CALMS does not interfere with the use of the telephone. Information is passed to and from the unit only when the line is not being used for telephone calls. This 'quiet line' use of the telephone line is seen by BT as a significant new way of exploiting more fully its nationwide network of local lines connecting customers to telephone exchanges. A wide range of new facilities for the domestic and small business user are planned: as well as improving service to the customer, the extra revenue from greater use of the network will help BT fight inflation and keep prices stable.

The CALMS trial is the first application of telemetering—based on low-speed, low-volume data transmission at low cost. A new pilot BT service, to be called *BitStream*, is aimed at meeting a variety of needs beyond the telemetry requirements of energy management, distant meter reading and remote control. It will be able to cater for home banking, home computer applications, home security systems and provide access to databanks such as Prestel. These services will be available to BT's customers without the need for special circuits to the exchange. They offer a way of serving the 'electronic home' (and the electronic small business) of the future. BitStream will complement the more sophisticated services (based on much higher data speeds) which will be available eventually from the integrated services digital network and interactive services which may be provided over systems for cable television.

OPENING OF COMPUTERISED INTERNATIONAL OPERATOR CENTRE

British Telecom's new computer-based international operator centre, the first to enter public service in Britain, was officially opened early in May of this year by Lady Jefferson, wife of BT's Chairman, Sir George Jefferson.

Located at Mondial House in the City of London, the centre is the first of two to be provided by Thorn Ericsson Telecommunications under a contract, valued at £3·5M, which includes the provision of an associated operator switching unit.

The new centre will handle customers' requests for operator-assisted calls drawn from an area stretching from the Isle of Wight to Folkstone, Kent, and requests for assistance from overseas operators requiring help in contacting UK customers.

The centre is equiped with visual display terminals served by minicomputers; these will be connected initially to a computer-controlled Ericsson crosspoint exchange (known as *AKE 13*), also located at Mondial House.

The opening marks the first phase of British Telecom International's (BTI's) modernisation programme to replace the older 'plug-and-cord' international operator centres.

Customers will enjoy several benefits introduced by the new centre. These are:

(*a*) overseas calls are connected faster and general enquiries handled quicker;

(*b*) calls are presented in their order of arrival to the first operator who becomes free and, if no operator is free, calls are queued in order of arrival;

(*c*) calls that have to be deferred are stored in the electronic memory of the exchange, and are re-presented automatically; and

(*d*) call duration and charge are calculated automatically, and, when requested, are presented to an operator to advise the customer.

Initially, 28 visual display terminals will be operational, and an additional 30 will come on line next spring.

Despite the growth of international direct dialling, residential and business telephone customers still have to use operators when ringing some 70 countries around the world. Other customers require certain types of call that are available only via an operator, such as transfer-charge calls, or calls made by using a telephone credit card, Other customers make use of the advice-of-duration-and-charge (ADC) service.

The provision of the centre marks the completion of BTI's programme to provide all UK customers with direct access to an international operator. The new centre is currently handling approximately 2500 calls and enquiries every day.

The second phase of the programme for modernising international operator services will be the introduction next spring of a new international operator centre at Brighton, and this will replace the present old plug-and-cord unit. A further new operator centre is planned for service in late 1986.

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Full membership of FITCE in the UK is available only through IBTE. Members and Affiliated Members of IBTE who hold a University science degree or who are Chartered Engineers may join through the FITCE Group of IBTE. The annual subscription for 1984/85 has been fixed at £5·00; this covers local administration expenses as well as the *per capita* contribution to FITCE funds, and thus ensures that no charge proper to FITCE affairs will fall upon the general membership of IBTE. Membership forms are available from your Local-Centre Secretary (see p. 228 of the October 1983 issue of this *Journal*) or direct from the Assistant Secretary (FITCE), Mr. P. A. P. Joseph, BTHQ/TES 3.1.3.2, Room 314, Broad Street House, 55 Old Broad Street, London EC2M 1RX; Tel: 01-588 8970.

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Notes and Comments

INCREASE IN THE PRICE OF THE *JOURNAL*

The Board of Editors regrets that the special price of the *Journal* to British Telecom and British Post Office staff will increase from the October 1984 issue; the new special price will be 51p per copy.

CONTRIBUTIONS TO THE JOURNAL

Contributions to *British Telecommunications Engineering* are always welcome. In particular, the Board of Editors would like to reaffirm its desire to continue to receive contributions from Regions and Areas, and from those Headquarters departments that are traditionally modest about their work.

Anyone who feels that he or she could contribute an article (short or long) of technical, managerial or general interest to engineers in British Telecom and the Post Office is invited to contact the Managing Editor at the address given below. The editors will always be pleased to give advice and try to arrange for help with the preparation of an article, if needed.

GUIDANCE FOR AUTHORS

Some guiding notes are available to authors to help them prepare manuscripts of *Journal* articles in a way that will assist in securing uniformity of presentation, simplify the work of the *Journal*'s editors, printer and illustrators, and help ensure that authors' wishes are easily interpreted. Any author preparing an article is invited to write to the Managing Editor, at the address given below, to obtain a copy.

All contributions to the *Journal* must be typed, *with double spacing between lines*, on one side only of each sheet of paper.

As a guide there are about 750 words to a page, allowing for illustrations, and the average length of an article is about six pages, although shorter articles are welcome. Contributions should preferably be illustrated with photographs, diagrams or sketches. Each circuit diagram or sketch should be drawn on a separate sheet of paper; neat sketches are all that is required. Photographs should be clear and sharply focused. Prints should preferably be glossy and should be unmounted, any notes or captions being written on a separate sheet of paper. Good colour prints and slides can be accepted for black-and-white reproduction. Negatives are not required.

It is important that approval for publication is given at organisational level 5 (that is, at General Manager/Regional Controller/BTHQ Head of Division level) and authors should seek approval, through supervising officers if appropriate, before submitting manuscripts.

Contributions should be sent to the Managing Editor, *British Telecommunications Engineering*, LCS/P5.1.1. Room 704, Lutyens House, Finsbury Circus, London EC2M 7LY.

British Telecommunications Engineering

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The *Journal* is published quarterly in April, July, October and January, at 90p per copy (£1·40 per copy including postage and packaging); annual subscription: £5·60; Canada and the USA: $12·00.

The price to British Telecom and British Post Office staff is 48p per copy.

Back numbers will be supplied if available, price 90p (£1·40 including postage and packaging). At present, copies are available of all issues from April 1974 to date with the exception of the April and October 1975; January, April and October 1976; and January 1982 issues.

Orders, by post only, should be addressed to *British Telecommunications Engineering Journal*, 2–12 Gresham Street, London EC2V 7AG.

Employees of British Telecom and the British Post Office can obtain the *Journal* through local agents.

Binding

Readers can have their copies bound at a cost of £8·75, including return postage, by sending the complete set of parts, with a remittance, to Press Binders Ltd., 4 Iliffe Yard, London SE17 3QA.

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Remittances for all items (except binding) should be made payable to *'BTE Journal'* and should be crossed '& Co.'

European customers can pay by means of a direct bank mail transfer. Customers using this method of payment should instruct their bankers to remit payments to the *Journal's* bankers—Midland Bank plc, 2 Gresham Street, London, EC2V 7JD, England—specifying that the beneficiary is to be *British Telecommunications Engineering Journal*.

Distribution and Sales

Correspondence relating to the distribution and sales of the *Journal* should be addressed to *British Telecommunications Engineering Journal* (Sales), Post Room, 2–12 Gresham Street, London EC2V 7AG.

Communications

With the exceptions indicated, all communications should be addressed to the Editorial Office, *British Telecommunications Engineering*, LCS/P5.1.1, Room 704, Lutyens House, Finsbury Circus, London EC2M 7LY (Telephone: 01-357 4313).

Advertisements

All enquiries relating to advertisement space reservations should be addressed to Mr. A. J. Pritchard, The Advertisement Manager, *British Telecommunications Engineering Journal*, Room 608, 2–12 Gresham Street, London EC2V 7AG.

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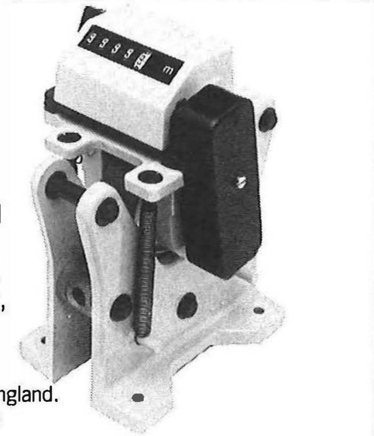

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He won't care tuppence about its state-of-the-art telephone technology. It's just much more efficient to dial directory numbers at the touch of a button. And to have the number redialled when he wants.

APPROVED for use with Telecommunications Systems run by British Telecommunications in accordance with the conditions in the instructions for use.

He's not likely to think of it as an on-line data terminal. But receiving Viewdata without moving from his desk will help him become much better informed, much more quickly.

In short, nothing about Executel (and there's more) is going to stop him wanting one. Especially after we've spent £500,000 in a national advertising campaign directed at top executives.

We're launching Executel right now.

And it's being centrally purchased through BT Consumer Products.

At STC Telecommunications, we've got a raft of technical data and product information about Executel. But don't let that put you off calling us now on Burnham (06286) 65411.

Executel NOT JUST ANOTHER TELEPHONE.

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