British Telecommunications Engineering

VOL 4 PART 2 JULY 1985

The Journal of
The Institution of British Telecommunications Engineers

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.)

Printed in Great Britain by Unwin Brothers Limited, The Gresham Press, Old Woking, Surrey.

Subscriptions and Back Numbers
Price: £1·00 (£1·50 including postage and packaging). Annual subscription (including postage and packaging): home and overseas £6·00 (Canada and the USA $10·00).

Price to British Telecom and British Post Office staff: 51p

Back numbers can be supplied if available, price £1·00 (£1·50 including postage and packaging).

Orders, by post only, should be addressed to *British Telecommunications Engineering Journal* (Sales), Post Room, 2–12 Gresham Street, London EC2V 7AG.

Remittances for all items (except binding) should be made payable to '*BTE Journal*' and should be crosed '& 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*.

Advertisements
All enquiries relating to advertisement space reservations should be addressed to Mr. A. J. Pritchard, The Adverstisement Manager, *British Telecommunications Engineering Journal*, Room 104, 90–91 Wood Street, London EC2V 7JS.

Communications
With the exceptions indicated, all communications should be addressed to the Editorial Office, *British Telecommunications Engineering*, NN/CMkt 2.2, Room 107, Intel House, 24 Southwark Bridge Road, London SE1 9HJ. (Telephone: 01-928 8686 Extn. 2233.)

Binding
Readers can have their copies bound at a cost of £8·75 per volume, including return postage, by sending the complete set of parts, with remittance, to Press Binders Ltd., 4 Iliffe Yard, London SE17 3QA.

0262-401X/85 $2·00 + ·00

British Telecommunications Engineering

Contents

VOL 4 PART 2 JULY 1985

ISSN 0262-401X

British Telecommunications Engineering

EDITORIAL

Despite continued advances in technology, telecommunication services available to the mobile user in the UK have, in the past, been restricted to about 10 000 users by the small number of radiophone channels available. However, in January of this year, the cellular radio service became operational with a target of providing service covering 95% of the population by the end of the decade and the prospect of over half a million users. The use of the cell concept with low-power transmitters and sophisticated monitoring and hand-over arrangements permits the users to travel between cells without loss of service and enables the available channels to be reused in a different cluster of cells. Although cellular radio is not a new concept, it was first proposed some 40 years ago, technological developments and, in the UK in particular, changes in the regulatory environment have only recently made its introduction possible. An article on p. 62 of this issue of the *Journal* is the first in a series of three on the Cellnet cellular radio network. This article gives a general overview of the system; the subsequent articles will describe the operational features of the system and finally the signal transmission and subscriber equipment.

Time-division multiple-access (TDMA) techniques are shortly to be introduced into some of the satellite communication links operated by the INTELSAT and EUTELSAT organisations. A general outline of TDMA systems, together with details of synchronisation and operating protocols necesary to operate such global systems, is given on p. 85 of this *Journal*.

The Cellnet Cellular Radio Network

Part 1—General System Overview

M. S. APPLEBY, M.A.†, and J. GARRETT, M.A., C.ENG., M.I.E.R.E.*

UDC 621.396.93 : 621.395.4

This article, to be published in three parts, describes the cellular radio network run by Telecom Securicor Cellular Radio Ltd. (TSCR). The TSCR service, the result of a joint venture between British Telecom and Securicor, is one of two cellular radio services that have recently begun operation in the UK. Part 1 begins by discussing the background behind the setting up of cellular radio services in the UK and goes on to describe the basic principles of cellular radio, and the features and operation of the standard that has been adopted for both services—the Total Access Communications System. Parts 2 and 3 will go on to describe the operational features of TSCR's system and the equipment used.

INTRODUCTION

Previous Public Mobile Radiophone Services

The South Lancashire Radiophone Service opened in 1959 and operated from two base stations in the very-high-frequency (VHF) band. This was a pilot system, which used frequency modulation (FM) on 50 kHz spaced channels, and its success gave rise to the opening, in 1965, of the London Radiophone Service. This was also a manually controlled system, which spread to other parts of the country during the 1970s.

The first automatic service, known as *Radiophone System 4*, started in London in 1981 and now covers all the areas previously served by the manual service, plus many more. Well over 10 000 customers are using the service, which now operates in the VHF band on channels spaced at 12·5 kHz.

The ultimate capacity of these systems is limited by the small number of channels available and the relatively slow and limited signalling repertoire. In London, where the demand is highest, the system is unable to support all those who wish to use it and, since 1978, efforts have been made to obtain further frequency allocations to allow the existing service to be expanded or a new one to be opened. No further channels are available in the VHF band, but the 1979 World Administrative Radio Conference allocated a band near 900 MHz that is wide enough to accommodate 1000 channels with 25 kHz spacing. Part of this has been made immediately available in the UK for cellular radio. A further allocation will be made when other users have vacated the band; in addition, there will be a reserved segment for future systems.

Foreign Systems

In the USA, Japan and the Nordic countries, the capacity of their existing VHF and ultra-high-frequency (UHF) services was exhausted and development was proceeding on higher-capacity systems at UHF by adopting cellular techniques. In 1980, trial systems were operating in the USA, Japan and Sweden; these were studied for possible application by British Telecom (BT).

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

* Telecom Securicor Cellular Radio Ltd.

Competition in the Provision of Cellular Radio Services

Whilst BT studied the techniques that could be applied, and awaited the necessary licence for the cellular service, the UK Government moved towards a greater liberalisation in telecommunications. It was decided to license two competing cellular radio operating companies and allocate half of the available channels to each. One operator would be the BT and Securicor joint venture, now called *Telecom Securicor Cellular Radio Ltd.* (TSCR) and trading as *Cellnet*; the other was chosen from a number of applicants and is now known as *Racal Vodafone*. Each operator was required to commence operation by the end of March 1985 and to provide service covering 90% of the population (approximately 60% of the land area) by the end of the decade.

Whatever the merits of competition in speeding up the provision of service and, possibly, in the long run, making for cheaper mobile units, the technical implications of the decision to have two competing services are severe. The division of the small number of available channels into two groups, one for each operator, is not an efficient way of using scarce bandwidth. Not only are double the number of channels allocated as dedicated control channels, but the smaller groups of channels available at each base station cannot be used as efficiently as groups more than twice their size. In the USA, the Federal Communications Committee administers the allocation of radio channels and acts as a central co-ordination point to minimise inter-user interference. There is no such co-ordination for the UK channels and it is possible that neighbouring but competing base stations could cause interference (for example, intermodulation) to each other. To fully avoid such interference, artificial constraints on the allocation of channels would be needed which would have a knock-on effect to further reduce system capacity.

Choice of Standard

None of the existing standards was found to be suitable for direct application in the UK. A detailed technical evaluation of the various systems in service or under development was carried out by BT. Factors considered included speech quality, signalling reliability and system capacity.

The outcome of the technical evaluation was included with other considerations, particularly the time required for system development and a recommendation was made to the Department of Industry (now Department of Trade and Industry (DTI)) by TSCR and Racal that a modified version of the USA standard be adopted for the UK, to be known as the *Total Access Communications System* (TACS). This recommendation was endorsed by the DTI in January 1983. Immediately, a joint committee of BT, Racal and TSCR was set up to define the interfaces necessary to ensure compatibility between the two systems and thus allow a mobile telephone to work on either, subject to suitable commercial arrangements. The committee expanded to include members from the DTI, the Electrical Engineering Association (representing manufacturers of mobile-telephone equipment), the British Approvals Board for Telecommunications (BABT) (the approval agents for mobile telephones), and the BABT-appointed test laboratory, ERA Technology Ltd.

The main areas covered by the joint committee were:

(*a*) the 'air interface', covering the radio specifications, signalling protocols and formats, and audio specifications;
(*b*) the options for the interface to the public switched telephone network (PSTN); and
(*c*) the interface between the cellular systems required to allow full intersystem roaming.

PRINCIPLES OF CELLULAR RADIO

Cellular radio is not a new concept. The fundamental ideas were originally proposed by the Bell Telephone Laboratories soon after the Second World War, although, at the time, the technology available did not allow a system to be implemented. Only recently have working cellular radio systems been introduced into public service. Among the countries currently operating networks are the USA, Japan, the four Nordic countries and, since the start of this year, the UK.

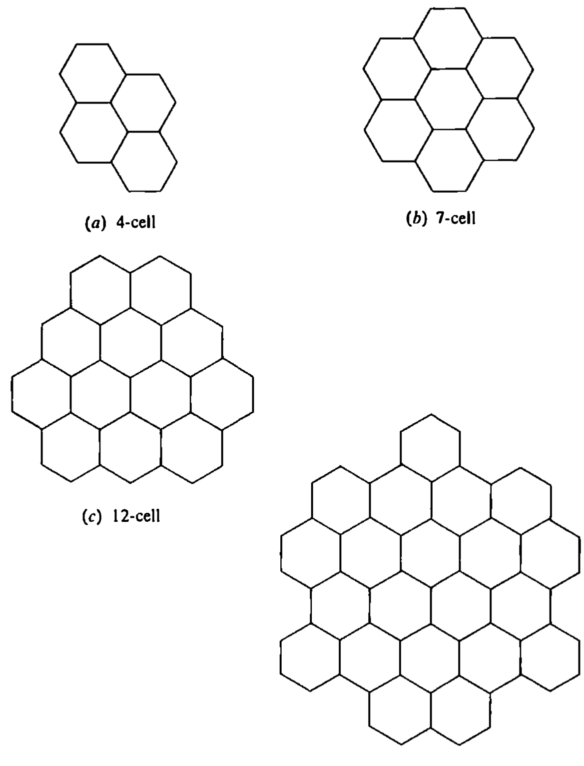
Cell Planning

The main attractions of cellular radio are its ability to cater for a wide range of traffic loading and its ability, ultimately, to handle far more customers than non-cellular systems. The basic principle of cellular radio is to split the required coverage area into a number of smaller areas, or cells, each having its own radio base station. The cells are grouped together into clusters and the radio channels available are allocated to each cluster according to a regular pattern which repeats over the whole coverage area. In this way, each channel is used several times throughout the coverage area in a regular fashion.

The number of cells in a cluster has to be chosen so that the clusters fit together into contiguous areas. Only certain cluster configurations do this, and typical arrangements of interest to cellular radio are groups of 4, 7, 12, and 21 cells (Fig. 1).

The number of cells in each cluster has a significant effect on the capacity of the overall system. The smaller the number of cells per cluster, the larger is the number of channels per cell, and thus the traffic carried per cell is higher. However, there is a penalty to pay for using small clusters. For the same cell size, as the cluster size reduces, the distance between cells using the same channels reduces, and so the interference from adjacent clusters increases. This type of interference, called *co-channel interference*, is one of the most significant factors that has to be taken into account when a cellular network is planned. Appendix 1 shows how co-channel interference is related to cluster size, and how the cluster size is chosen. In practice, the most common size is the 7-cell cluster, which represents a useful compromise between capacity and interference levels.

In the TACS system, signalling between mobiles and base stations for the purposes of setting up calls is carried on different channels from those which carry speech. Since co-channel interference would cause more problems on signalling channels than on speech channels, 21 channels are reserved for signalling, allowing a different cell cluster size to be used. With a cluster size of up to 21, co-channel interference can be kept to low levels.



(*a*) 4-cell (*b*) 7-cell (*c*) 12-cell (*d*) 21-cell

FIG. 1—Cell repeat patterns

As the maximum number of channels in a cell is fixed by the total number of channels divided by the cell repeat pattern, the maximum number of simultaneous calls is correspondingly limited. However, if the size of each cell is reduced, there are more cells in a given area, and so the total number of available channels in the area is increased, and the traffic that can be handled is correspondingly increased. Thus, it is common in urban centres where the demand for service is high to have very small cells. Conversely, in rural areas the traffic demand is comparatively low, and so to provide service economically the cell size is increased, and the number of expensive base stations is kept to a minimum. The size of cells can be controlled by the choice of location for the base station, the height and type of aerial, the power transmitted, and the signal threshold levels for transferring calls dynamically from one cell to another.

System Growth

Initially, the cellular radio networks will be configured with relatively large cells, even in urban areas. However, as traffic demands increase towards the limits that such a configuration can handle, 'cell splitting' will become necessary, particularly in urban areas.

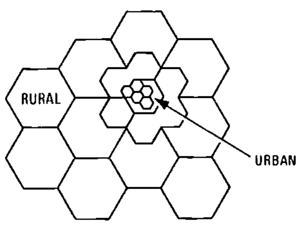


FIG. 2—Cell splitting

Cell splitting consists of adding extra base stations and reconfiguring the channel allocations to give a larger number of smaller cells (Fig. 2). As the cell size is reduced in this way, two problems arise. Firstly, as the total number of base stations becomes large, it becomes increasingly difficult to find suitable sites to house them, and increasingly expensive to equip them. Secondly, propagation considerations mean that the amount of co-channel interference increases, even though the same repeat pattern is used.

The total co-channel interference received by any cell is the result of the signals received from each of the six clusters surrounding the cell (Fig. 3). One way of reducing the interference is to use directional aerials so that interference is received from only one cell. This is called *sectorisation* as the directional aerials split each cell into a number of sectors.

Two methods of sectorisation are commonly used, 3-sector and 6-sector (Fig. 4). With a 3-sector arrangement, the base stations in a 7-cell repeat pattern are fitted with directional aerials to split each cell into 3 areas. The channels of the original cell are allocated accordingly to the new areas. The effect is to produce a larger number of corner-excited cells using the same number of cell sites.

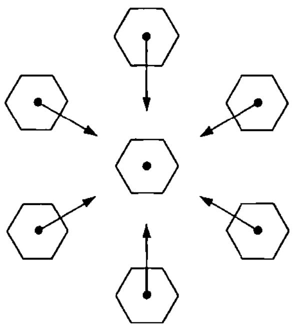


FIG. 3—Co-channel interference is received from six co-channel cells

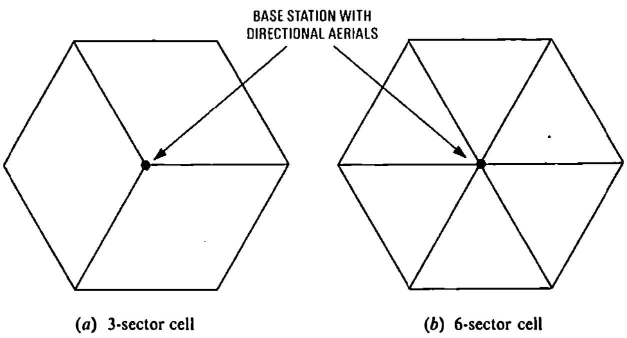


FIG. 4—Sectorised cells

With a 6-sector arrangement, the effect is again to increase the number of cells whilst keeping the same number of sites. However, the 6-sector arrangement also allows the main cell repeat pattern to be reduced to 4, thus increasing the number of channels per cell beyond that of the 7-cell arrangement. Both methods of sectorisation allow smaller cells to be realised by reducing the interference levels.

Network Aspects

A typical cellular network configuration is shown in Fig. 5. Base stations are connected by permanent links to mobile switching centres (MSC), which are computer-controlled telephone exchanges specially designed or adapted for cellular radio service. MSCs are connected to BT's PSTN to give access to and from land customers. MSCs within the cellular radio network are also connected together so that incoming calls to a mobile can be completed wherever it is located. In this way, a complete cellular network can be built up to give continuous radio coverage over a wide geographical area. However, two important features are necessary for the cellular network to function effectively, and these arise because mobiles move from cell to cell as they move through the coverage area.

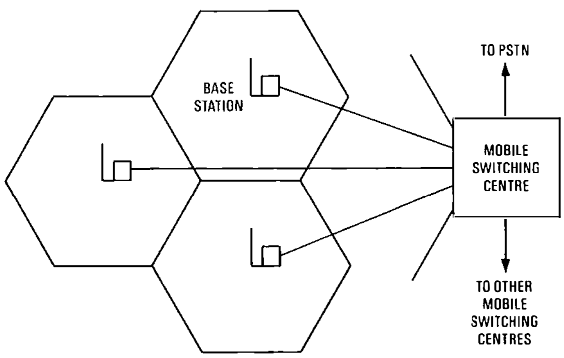


FIG. 5—Cellular radio network

The first of these features is mobile tracking, or location. When a call is received for a mobile from the PSTN, the cellular network needs to find which base station the mobile is nearest to so that the call can be connected successfully. In a network consisting of several hundred base stations and hundreds of thousands of mobiles, it is not feasible to transmit the call to the mobile on every base station. If this were the case, the capacity of the signalling channels would quickly become exhausted, and the overall capacity of the network would be limited. Instead, the cellular network is organised into a number of traffic areas, each consisting of a group of cells. The MSCs keep a record of the current location of all mobiles, which is updated by a process known as *registration*. Whenever a mobile is not making a call, it constantly listens to one of the common signalling channels. As part of the information transmitted on the signalling channels, the base stations generate a code identifying the traffic area. If the mobile starts to receive errors in the data stream, indicating that the signal has dropped below a usable level, it rescans the signalling channels to search for a higher signal level. If a change in traffic area code is detected, indicating a new traffic area, the mobile automatically registers its new location by calling the new base station and identifying itself. The network then ensures that the location information is updated.

The second feature is *in-call hand-off*. When a mobile is engaged on a call, it often moves from the coverage area of one base station into another. So that the conversation is not interrupted, the call must be handed-off automatically to the next base station. The base station constantly checks the signal level received from all mobiles in communication and, when the level drops below a threshold, it informs the MSC that a hand-off may be required. The MSC commands all the surrounding base stations to measure the signal strength of the mobile, and then chooses the best cell to transfer the call to. Once the new base station has been informed of the hand-off request, and the radio channel allocation has been made, the original base station is commanded to send a control message to the mobile to move to the new channel. This all happens automatically within a few seconds, and the user of the mobile is aware of only a very brief break in transmission (about 400 ms) when the hand-off proper takes place.

TACS RADIO FEATURES AND PARAMETERS

The cellular system chosen for adoption in the UK, TACS, is an adaptation and enhancement of the Advanced Mobile Phone System (AMPS) adopted in North America. Adapta-

TABLE 1

TACS and AMPS Radio Parameters

Parameter	TACS	AMPS
Frequency band Mobile transmitter Base transmitter	 890–915 MHz 935–960 MHz	 825–845 MHz 870–890 MHz
Maximum number of channels	1000	666
Channel spacing	25 kHz	30 kHz
Peak frequency deviation Speech Signalling SAT	 9·5 kHz 6·4 kHz 1·7 kHz	 12 kHz 8 kHz 2 kHz
Signalling data rate	8 kbit/s	10 kbit/s

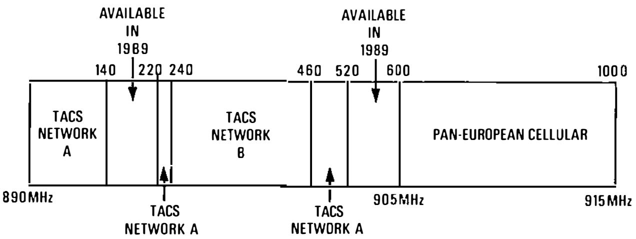
SAT: Supervisory audio tone

tion of AMPS was necessary as the regulatory position in the UK was different from that in the USA, particularly with regard to radio frequency bands and channel spacing (Table 1).

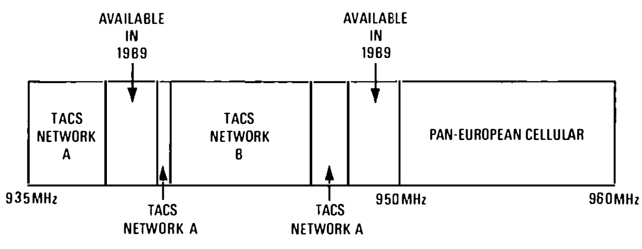
Radio Channels

The TACS system operates in the CEPT† 900 MHz band and uses up to 1000 radio channels with a 25 kHz spacing between channels. Each TACS channel comprises a pair of frequencies spaced 45 MHz apart; the higher frequency is used for transmission from the base station to the mobile (forward direction), and the lower from the mobile to the base station (reverse direction). The TACS system allows for two entirely independent networks to share the band, each being allocated its own exclusive channels. In the UK, the two competing networks will, by 1989, be allocated a total of 600 channels (300 to each operator), with the remaining 400 being held in reserve for a future pan-European system. However, the allocation of channels has been complicated by the need to allow time for services currently using the band to move frequency. Fig. 6 shows the current allocation to the two networks.

† CEPT—European Conference of Posts and Telecommunications



(*a*) Mobile transmit



(*b*) Mobile receive

FIG. 6—UK 900 MHz cellular radio frequency band (1985)

The TACS system has two types of radio channel: control and speech. For each of the two networks, 21 channels are reserved for control and cannot be used as speech channels. Of the remaining channels, some may be used for control where traffic demands extra control channels; otherwise, they can all be used as speech channels (Fig. 7).

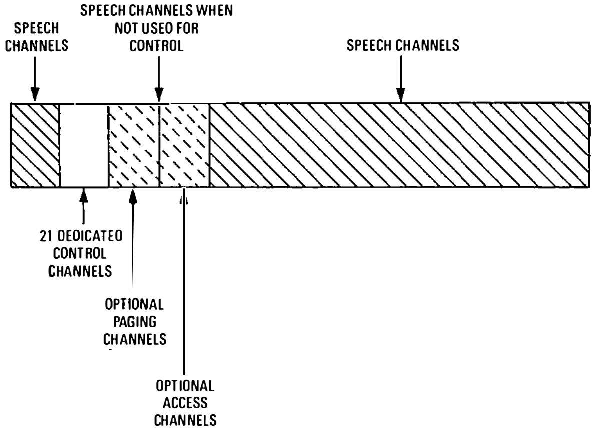


FIG. 7—TACS control and speech channels

Control Channels

The TACS control channels are used for the co-ordination of mobiles and for all call set-up procedures. Functionally, there are three types of control channel: dedicated control channels (DCCs), paging channels and access channels. In practice, all three functions may be combined and carried on the same channels. As the network grows and the traffic load increases, some functions may be allocated separate blocks of channels to avoid overload.

The DCCs are the most fundamental of the three types of control channel. All mobiles are permanently programmed with the channel numbers of the DCCs for both TACS networks and scan these channels at switch-on. The DCCs carry basic information about the network and inform the mobiles about the channel numbers of the paging channels.

The paging channels are used both to transmit messages to specific mobiles (for instance, to alert them to incoming calls), and general network information such as traffic area identity, channel numbers of the access channels, access methods to be used by mobiles etc.

The access channels are used by mobiles for accessing the network to initiate outgoing calls, to register their location and to respond to paging calls.

All three types of control channel carry status information in the sequences of data blocks called *overhead messages*. The type of overhead message depends on the function of the control channel. Where the functions of dedicated control channel, paging channel and access channel are combined, the overhead messages contain all the relevant information as one message train.

Speech Channels

All voice communication is carried on the speech channels; in addition, some signalling (for hand-off, power control, cleardown etc.) is carried.

The speech signal is carried on the speech channels by means of analogue FM with a peak frequency deviation of 9·5 kHz. By comparison, previous radiophone systems with the same channel spacing of 25 kHz have used a frequency deviation of only 5 kHz, in order to minimise interference problems in the adjacent channels. The use of this wider deviation in the TACS greatly improves the rejection of unwanted signals on the same frequency (co-channel interference). Co-channel interference is the most significant limiting factor that determines the cell repeat pattern used, and thus the ultimate capacity of the system. Reducing the effects of co-channel interference by increasing the deviation thus results in an overall increase in the maximum number of customers.

Increasing the deviation to 9·5 kHz increases the interference to the adjacent channels and, if this effect is too great, the benefit from the use of a wider deviation is negated. However, the adjacent-channel interference can be reduced by careful channel allocation; that is, by ensuring that adjacent channels are never allocated in the same cell, and only to a limited extent in the adjoining cells. It can be shown that, by following these rules and by restricting the deviation to 9·5 kHz, interference can be kept to acceptable levels.

Signalling

Signalling between the mobile and the base station for call set-up, hand-off and other similar control functions is carried digitally by using frequency-shift keying (FSK) of the radio-frequency carrier with a deviation of 6·4 kHz. The basic data rate used is 8 kbit/s, but, to enable the clock to be extracted at the receiving end, the data is Manchester encoded before transmission to give a data rate of 16 kbit/s.

It is necessary for all signalling information to be carried reliably even under poor signal conditions with substantial amounts of fading. A robust form of error protection is therefore used which has the combined techniques of repeated transmission with majority decision, and forward error correction.

The message to be sent is coded into a block, and a parity word is generated from it by using Bose-Chaudhuri-Hocquenghem (BCH) coding. The parity word is appended to the message to form a signalling block. A complete signalling frame is then formed which starts with a bit- and word-synchronisation sequence, followed by the combined message and parity word repeated several times. On the signalling channels, the base station repeats the message five times; but, on the speech channels, where greater levels of interference may be expected, the message is repeated 11 times. All messages sent by the mobile to the base station are repeated five times, whether on the signalling or speech channel. When the message is received, a bit-by-bit majority decision is carried out on the repeated blocks, and many of the errors that may have occurred are corrected. The parity word is then used to correct up to one remaining error, and to detect if more than one error is present. If there are two or more errors remaining after majority decision, they cannot be corrected and the message is rejected.

Since the number of data bits required for signalling messages in each of the four cases where signalling is used (base-to-mobile, mobile-to-base on both control and speech channels, see Fig. 8) is different, the format of the signalling frames is adapted for each case. Table 2 lists the number of bits in the message, the number of parity bits and the number of repeats of each block for each of the four cases, and Fig. 9 shows the four signalling-frame formats.

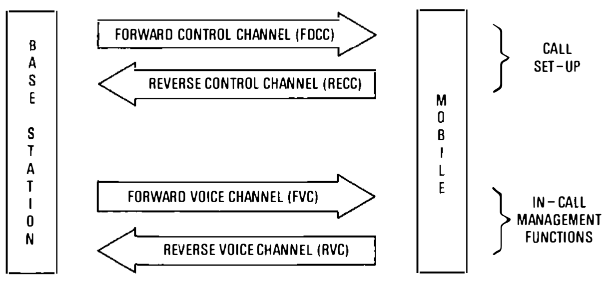
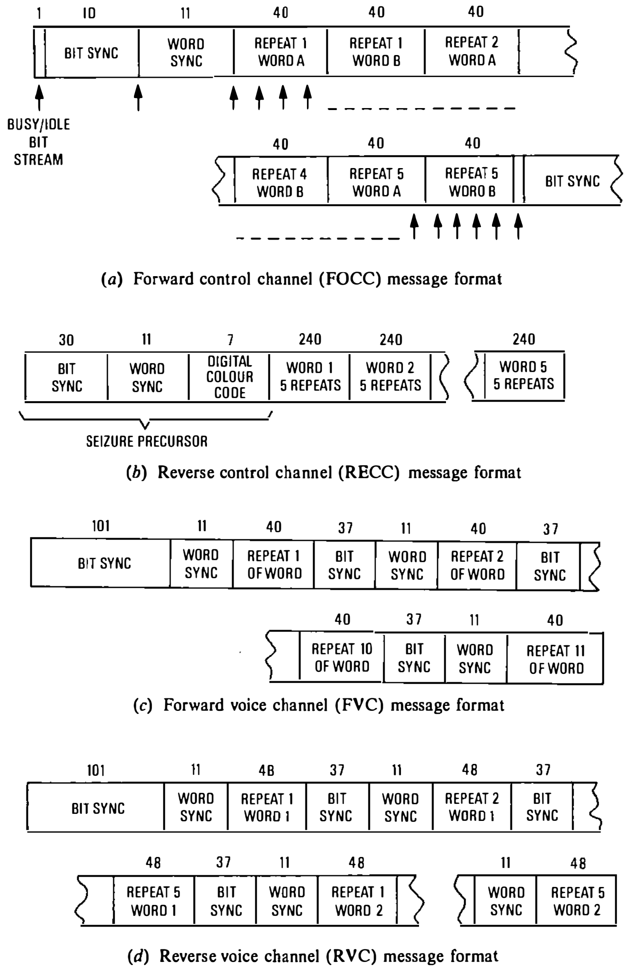


FIG. 8—TACS signalling paths



SYNC: Synchronisation

FIG. 9—TACS signalling frame formats

TABLE 2

Composition of Signalling Frame

Signalling Channel	Data bits	Parity bits	Repeats
Base-to-mobile (forward) control channel	28	12	5
Mobile-to-base (reverse) control channel	36	12	5
Base-to-mobile (forward) voice channel	28	12	11
Mobile-to-base (reverse) voice channel	36	12	5

The signalling format used from the base to the mobile on the control channels is of particular interest since it contains three data streams multiplexed together. The complete data stream consists of blocks of 463 bits of information. Bit synchronisation enables mobiles to lock onto the bit stream, and word synchronisation indicates the start of the information frame. The remainder of the block consists of two 40 bit messages (word A and word B) repeated five times each, multiplexed with a BUSY/IDLE bit stream. Each mobile responds only to either the word A or word B stream, depending on whether the identity number of the mobile is odd or even. The BUSY/IDLE stream is used by the base station to indicate whether the reverse signalling channel is free or not, and is checked by mobiles before the system is accessed.

During some phases of call set-up, the mobile needs to inform the base station of the ON/OFF-HOOK status (for example, whilst a mobile is being rung, or during cleardown). Instead of sending a complete signalling frame to indicate simply this status, the mobile transmits an 8 kHz signalling tone when the mobile is required to indicate that it is ON-HOOK.

Supervision

Whilst a call is in progress, a supervisory audio tone (SAT) is transmitted by the base station and looped back by the mobile. Both base station and mobile require the presence of the SAT on the received signal to enable the audio path. Three different frequencies are used for the SAT, all around 6 kHz. During call set-up, the base station informs the mobile which SAT to expect on the speech channel. If the SAT is incorrect, the mobile does not enable the audio path, but starts a timer which, on expiry, returns the mobile to stand-by. Similarly, the base station expects to see its transmitted SAT returned and takes this as confirmation that the mobile is operating on the correct channel.

The three SAT frequencies are allocated to the cells so that the clusters form a three-cluster repeat pattern. Thus a cell in the adjoining cluster using the same radio channel uses a different SAT. The effect of this is to reduce the possibility of a co-channel interfering signal, which, for some reason, is stronger than the wanted signal, being overheard by a mobile.

TACS METHOD OF OPERATION

Mobile Scanning

Mobiles are programmed with the channel numbers of the DCCs of both networks, together with a mark to indicate the primary network. When a mobile is first turned on, it scans the 21 DCCs of its primary network looking for the channels with the strongest and second strongest signals. It then tunes to the strongest channel and attempts to receive the overhead messages being transmitted. If successful, it receives information indicating the channel numbers of the paging channels. If it does not receive the information correctly, it tunes to the second strongest channel and tries again. If it is still unsuccessful, it is likely that the mobile is not within range of its primary network, and so it repeats the scanning procedure using the DCCs of the secondary network.

When the mobile has received the channel numbers of the paging channels, it scans these channels, again looking for the channels with the two strongest signals. It tunes to the strongest paging channel and attempts to receive the overhead messages being transmitted on that channel. If successful, it receives information about the traffic area in which it is operating, and a number of parameters about the network configuration. As for the DCCs, if the mobile does not successfully receive the information on the strongest channel, it tries the second strongest. If this fails, it restarts the whole scanning operation.

From switch-on to being locked to a paging channel typically takes a mobile between 5 and 10 seconds. However, if the mobile needs to revert to the second stronger DCC or paging channel, this time could be lengthened to about 17 seconds.

When the mobile has found a suitable paging channel, it enters its IDLE state and remains on that channel monitoring the messages being transmitted. If the signal level falls as a result of the mobile moving away from the base station it is receiving, the messages cease to be received correctly. When this happens, the mobile re-enters the scanning routine to find a paging channel from a more suitable base station.

Registration

Registration is used by mobiles to announce their current location and to enable the cellular network to direct incoming calls to the appropriate cells. The TACS system has two forms of registration, forced registration and periodic re-registration. Either, or both, can be enabled by means of parameter messages transmitted on the paging channels.

With forced registration, mobiles are required to register every time they cross a traffic area boundary. Whenever a mobile rescans to find a new paging channel, it compares the traffic area information received on the paging channel with that received on the previous paging channel. If a difference is detected, the mobile must have crossed a traffic area boundary, and so registers its location with the network. When it has registered, it updates its internal memory to the new traffic area.

With periodic re-registration, the mobile maintains a list of the last four traffic areas visited, together with a numeric indicator for each, which is used to tell the mobile when to re-register. The base stations transmit a registration identity number on a regular basis. This number is incremented every time it is transmitted. The mobile compares the number received with the number in its memory appropriate to the current traffic area and, when the two are equal, initiates a registration with the network. When it has registered, the mobile increments the number stored in its memory and awaits the next registration period. The network can control the rate at which mobiles re-register by varying the time between transmitting registration identity messages, or by changing the number by which mobiles increment their stored number. If a mobile finds that it is in a traffic area for which it contains no entry in its memory, it creates a new entry (deleting the oldest) and immediately registers with the network in the same way as for forced registration.

To register with the network, the mobile performs a system access and sends a registration message. System access is covered below under 'Call Origination'.

Call Origination

When the user wishes to make an outgoing call, the number required is keyed into the mobile, or the required number is extracted from the short-code memory in the mobile. The user then initiates the call, normally by pressing a SEND key. This causes the mobile to perform a system access so that it can transmit its message to the network.

To perform a system access, the mobile first scans the network's access channels. The channel numbers of the access channels are transmitted as part of the overhead information on the paging channels. In the same way that other scanning operations are carried out, the mobile chooses the two channels with the highest signal level and attempts to receive the overhead messages being transmitted. Again, if the mobile fails on the strongest channel, it tries on the second choice. Contained in the overhead messages are parameters to tell the mobile about the required access procedure. Once the mobile has received the parameters, it checks the status of the BUSY/IDLE bit stream being sent by the base station. If this indicates the IDLE condition, the mobile waits a random time, turns on its transmitter, starts to send its message and continues to monitor the BUSY/IDLE bit stream. When the base station receives the start of the message from the mobile, it sets the status of the BUSY/IDLE stream to BUSY. The mobile then checks the time between the start of its message and the point when the status changes to BUSY. If this occurs at the correct time, the mobile continues to send its message. If it occurs too early or too late, the mobile assumes that the transition is due to some other mobile, not itself, and immediately aborts the message. It then waits a random time and attempts to transmit its message again. In this way, clashes between mobiles attempting to access the network at the same time are reduced.

When the mobile has sent its message to the network, it turns off its transmitter and remains on the access channel awaiting a message from the base station. For call originations, the message is normally a speech-channel allocation and contains the channel number and the SAT code. On receipt of the message, the mobile tunes to the required channel and starts to transpond the SAT. If the SAT received is correct, the audio paths are enabled and the user can hear the call being set up.

If the access was as a result of a registration, the message received on the access channel is normally a registration confirmation. On receipt of this message, the mobile returns to the IDLE condition.

Call Receipt

When an incoming call for a mobile is received, the MSC checks the current location of the mobile that has been obtained through the registration procedures. A paging call is then transmitted on the paging channel of all base stations in the mobile's current traffic area.

When the mobile receives a paging call, it accesses the network in the same way as for call originations, but the message sent to the base station informs the network that the access is as a result of receiving a page. The mobile receives a speech-channel allocation from the base station, tunes to the new channel, and checks the SAT received. The base station then transmits an *alert* message to the mobile, causing the mobile to alert the user to the incoming call and to transmit a continuous 8 kHz signalling tone. When the user answers the call, the signalling tone is turned off, the audio paths are enabled and the call proceeds.

Hand-Off

Whenever a mobile is operating on a speech channel, the base station monitors the signal level received. If the signal falls below a given threshold, the base station informs the MSC that hand-off to a nearby cell may be necessary. The MSC commands each of the surrounding base stations to measure the signal strength of the mobile by using their special measuring receivers. When the MSC receives the results of the measurements, it decides whether any cell is receiving a better signal from the mobile than the present cell. If there is a better cell, the relevant base station allocates a speech channel, and the MSC commands the original base station to order the mobile to the new channel. A short signalling message is sent to the mobile on the speech channel giving the new channel number, and the mobile tunes to the required channel. The user notices only a brief period of quiet (about 400 ms) during the hand-off process.

Power Control

If a mobile moves close to a base station, the high signal level could result in intermodulation in base station receivers causing interference to other users of the network. To avoid this happening, the signal level is monitored and, if it is found to be above a given threshold, a message is sent to the mobile to reduce its transmitter power level. On receipt of the message, the mobile acknowledges by sending a message to the base station and selects the appropriate power level.

Invoking Facilities

During a call, a user may wish to invoke some special facilities (for example, call a third party for a three-way conference call). The user keys in the appropriate code and presses the SEND key. The mobile sends a short burst (0·4 s) of 8 kHz signalling tone to the base station, which responds with a digital signalling message on the voice channel requesting the mobile to send its information. The mobile transmits the information to the base station in the form of a digital signalling message on the voice channel, and then returns to the conversation mode whilst the network processes the request for the facility.

Call Termination

When the mobile user has finished a call and replaced the handset, the mobile transmits a long burst (1·8 s) of 8 kHz signalling tone to the base station and then re-enters the control-channel scanning procedure.

If the originator of a call on the PSTN clears down, a *release* message is sent to the mobile, which responds by sending a burst of 8 kHz signalling tone, after which it re-enters the control-channel scanning procedure.

CONCLUSION

The UK TACS has, in a very short time, been defined and brought into service, with two competing networks operating to the same overall standard. TACS was derived from the USA AMPS, which has a proven track record in trial systems, although has itself only recently been put into full public service. The specification of a system as complex as TACS has involved the participating parties in substantial amounts of effort, and, whilst it is still possible that there may be some inconsistencies in the overall specification, early experience with working networks has created confidence in the integrity of the specification.

This article has explained the basic technical details of cellular radio and the operation of TACS. The second article in this series will describe TSCR's network in greater detail.

APPENDIX 1
Relationship between Cell Radius (*R*) and Repeat Distance (*D*)

Fig. 10 shows two 7-cell clusters. The radius of the hexagonal cells is R, and the distance between the centres of the clusters is D.

Each cluster can also be represented by a larger hexagon with identical area and radius R'. The distance between the centres of the two large hexagons is also D.

Clearly, the ratio of the areas of the large and small hexagons is equal to the number of cells in the cluster, that is, 7. As the

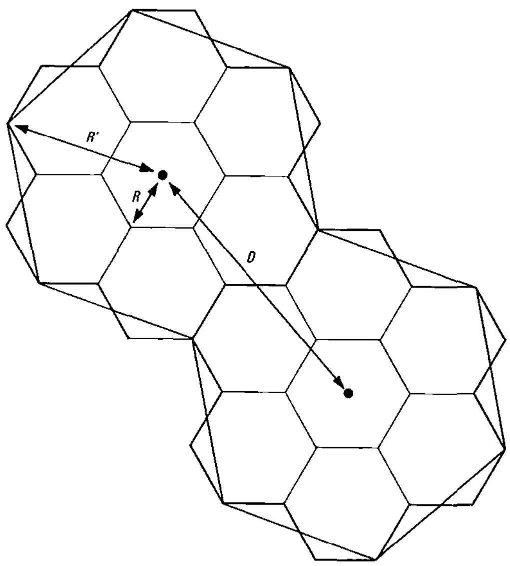


FIG. 10—Calculation of D/R

area of a hexagon is proportional to the square of its radius, the radii of the large and small hexagons is related by

$$R'/R = \sqrt{7}.$$

The distance between the centres of the two large hexagons, by simple trigonometry, is given by

$$D = R'\sqrt{3}.$$

Thus D and R are related by

$$D = R\sqrt{3}\sqrt{7}.$$

So the re-use ratio

$$D/R = \sqrt{3}\sqrt{7}.$$

The number 7 arises because the cluster contains 7 cells. This relationship can be shown to hold in general cases where the cluster size is N. Then D and R are related by

$$D/R = \sqrt{(3N)}.$$

Derivation of Carrier-to-Interference Ratio and Repeat Patterns

If there are two co-channel cells of radius R and distance D apart (Fig. 11), a mobile receives signals from both base stations

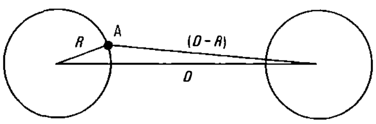


FIG. 11—Co-channel interference model

with a wanted-carrier-to-interfering-carrier ratio of C/I. The mobile receives the strongest signal from the interfering base station, together with the weakest signal from the wanted base station, when it is at about point A. Thus at this point the C/I is at its lowest.

At point A, the mobile is a distance R from the wanted base station, and a distance $D-R$ from the interfering base station.

The signal received from the wanted base station is

$$C \propto 1/R^x,$$

where x is the propagation law, normally between 3 and 4.

The signal received from the interfering base station is

$$I \propto 1/(D - R)^x.$$

Thus the carrier-to-interference ratio is

$$C/I \propto \{(D - R)/R\}^x \propto \{D/R - 1\}^x.$$

Above a certain C/I (typically around 10 dB), the effect of the interfering signal becomes negligible because of the FM capture effect. This therefore sets a lower limit to C/I, and thus a lower limit to D/R. D/R has been shown above to be related to the number of cells in a cluster; so the C/I limit effectively fixes the smallest cluster size that can be realised.

In practice, a moving mobile does not receive a constant signal from either the wanted base station or the interferer. The signals suffer two types of fading, slow (called *log-normal fading*) and fast (called *Rayleigh fading*). The effect of fading is to increase the required D/R, and thus increase the minimum cluster size. In addition, interference, in general, is received from more than one cell, further increasing the required D/R. More complex propagation models, verified by measurements, are necessary to be able to finally decide the repeat pattern and cell sizes to be adopted.

Biographies

Malcolm Appleby is Head of the Radiophone Systems Enhancements Group at the BT Research Laboratories, Martlesham. He joined the then Post Office in 1970 as a Post Office Student, and, from 1971 to 1974, read Engineering and Electrical Sciences at Cambridge University. After leaving university, he joined the Research Department and spent several years in the digital switching divisions during the development of System X. He moved to the radiophone field in 1981 and was concerned with testing and proving the equipment forming Radiophone System 4. Since 1983, he has been working on the standards and specification of TACS, and has played an active part in a number of the JRTIG committees. He has since been involved in the testing and acceptance of TSCR's TACS network.

John Garrett joined the Electronic Switching Division of the then Post Office Research Department in 1969 and worked on stored-program control switching, high-level telephony programming languages and function distribution in switching systems. After a period in System X software support, he moved into the area of mobile communications, including the development of Radiophone System 4 and the selection of the cellular radio system to be used in the UK. As a Head of Section in BT Enterprises, he is currently on secondment to TSCR as System Manager of Cellnet.

Fall Arrestors—A New Approach to Dynamic Performance Testing

K. J. CLARK, C.ENG., A.M.I.E.R.E., A.M.I.O.S.H.†

UDC 621.315.66 : 614.89

This article describes a new approach to dynamic performance testing of fall arrestors—safety devices used by British Telecom aerial riggers when working at height on radio masts and towers. The article begins by describing the types of fall arrestor available and where they are typically used. It outlines the new testing arrangements that were adopted and the rather alarming results obtained. The corrective action taken for a particular type of fall arrestor is detailed and mention is made of consequential changes to be included in British and International Standards.

INTRODUCTION

Fall arrestors, devices to prevent the fall of persons when working at height, are used by British Telecom (BT) aerial riggers and are covered by the requirements of British Standard BS5062. This specification requires that fall arrestors must pass tests including those to prove their ability to arrest falling loads.

The findings show that some types of fall arrestor, although fully meeting the requirements of the British Standard, could under certain conditions fail to arrest the fall of a person. These findings have been presented to the appropriate BSI* Committee and as a result the British Standard for fall arrestors is to be amended.

FALL ARRESTOR ENVIRONMENT

Fig. 1 shows a typical radio tower to be found at National Networks' (NN) microwave radio stations on the trunk radio network. The towers are provided with fixed safe means of access, and working places in accordance with the Construction (Working Places) Regulations 1961, in the form of vertical hooped ladders to BS4211 and fixed platforms with handrails and toeboards.

In order to carry out maintenance work or to install a variety of aerials, for example, dipoles, Yagis and paraboloids, it is often necessary for the riggers to leave the working platform and climb the steelwork to reach their place of work. This activity is obviously hazardous and, unless specific precautions are taken, one false move or slip could prove fatal, particularly when typical working heights of 80 m are involved.

BT POLICY

BT operates a policy of 'permanent attachment' for its riggers whenever they leave a place of safe access; that is, hooped vertical ladder or permanent working platform. This means that they must be attached to the structure at all times in such a way that they are prevented from falling any appreciable distance. There are a number of ways of meeting this requirement, and various approved methods are available so that the most suitable for a particular application can be used.

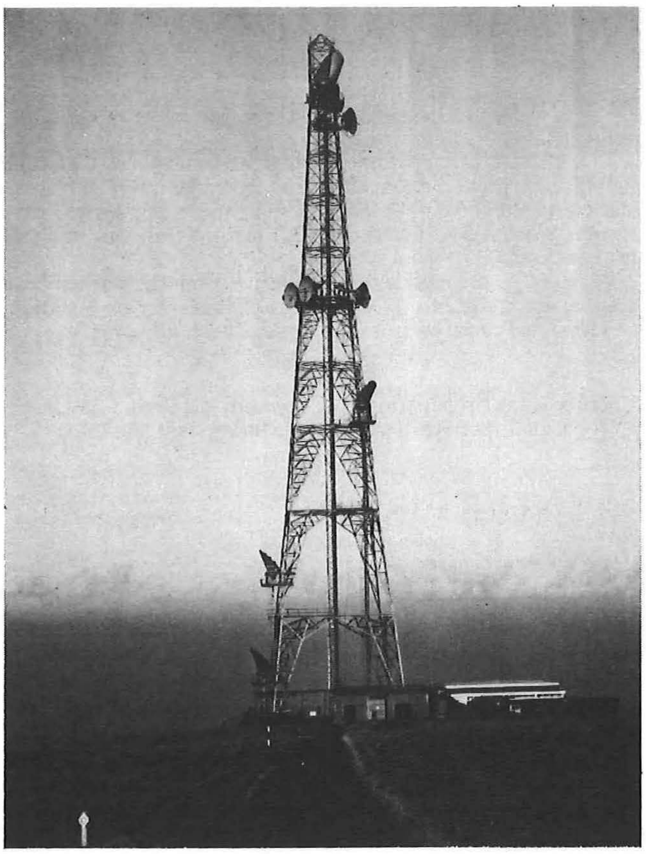


FIG. 1—Typical microwave radio tower

Table 1 lists some of the currently available methods and typical applications. Figs. 2 and 3 show some of the devices in actual use. It is important to note that these devices and other approved access methods can be used only in association with a full safety harness (BT Belt Safety No. 4 or No. 7). The Belt Safety No. 7 is a new development and has the dual facility of front and rear attachment points for fall arrestors. The Belt Safety No. 4 has only the rear attachment.

† Trunk Services Planning and Works, British Telecom National Networks

* BSI—British Standards Institution

FIG. 2—Type 1 fall arrestor in use

FIG. 3—Type 2 fall arrestor in use

TABLE 1

Fall Arrestor Methods and Typical Applications

Fall Arrestor	Typical Application
Fall Safe Rope Grip	To provide temporary safe access over any distance, but where the climber is not involved with much horizontal movement
Railok	To provide permanent safe access over any distance on unhooped ladders or other vertical climbs such as the Mast No. 1
Evalok Steel 50	General temporary safe access on towers or masts. Limited to a distance of 15 m. Can be used other than vertically; for example, on roofs
Sala Block	As Evalok Steel 50, but limited to 5·5 m

BRITISH STANDARD BS5062

Two distinct types of fall-arresting device are described in BS5062. These are classified as *Type 1* and *Type 2* devices.

Type 1 devices (Fig. 2) travel on a fixed line or rail and are attached to the rigger's safety harness by a short connecting lanyard. These devices travel with the rigger as he ascends and descends the structure, but, if he should fall, the increase in velocity causes the device to operate and lock onto the safety line or rail.

Type 2 devices (Fig. 3) consist of a safety line which is unreeled from a spring-loaded drum as the rigger descends, and automatically rereels as he ascends.

These devices are velocity sensing and, should a fall occur, a brake engages to lock the wire and arrest the fall immediately a critical speed is reached.

Testing for Approval

As part of the process of obtaining BSI approval, devices must pass stringent tests.

Both Type 1 and Type 2 fall arrestors are subjected to an endurance test whereby they are moved 10 000 times over a distance of not less than 30 cm, the last 1000 times of which the devices are caused to lock. Type 1 devices are then supported in an unlocked position and released—they should lock positively. Type 2 devices are tested to ensure that they lock at a line velocity of not more than 4·5 m/s.

All devices then undergo a number of conditioning tests, in which they are subjected, in order, to dust, cold, wetness and, for Type 2 devices only, oil. After each conditioning, they are subjected to a drop test with a load of 136 kg attached to the device by a 1 m length of line. For Type 1 devices, the load is released from a position as near as possible 1 m above the device and allowed to fall freely until arrested. On arrest, the device itself must not have dropped more than 1 m. For Type 2 devices, the load is attached to the device and then released. In this case, the load itself should not drop more than 1 m.

THE INVESTIGATION

Although BSI approval tests can confirm the strength and durability of a device, they do not necessarily show how it would perform in actual man-fall situations. It was considered that testing such devices under man-fall conditions was a fundamental requirement that ought to be included in the approval procedures. In the absence of any evidence of earlier work on the subject, it was decided to conduct an in-house investigation.

It was of prime importance that the tests should resemble as closely as possible what would actually happen if a person should fall, but for obvious safety reasons a man could not be used for the tests.

The next best thing to a man for this type of application was used, namely an anthropometric dummy. This is a fully articulated, jointed and limbed dummy which has a weight distribution closely resembling that of an 'average' man. The dummy was equipped with the usual rigger's gear; namely, safety helmet, safety boots, protective clothing and safety harness. The dummy complete with gear weighed 76 kg.

In choosing the testing arrangements to be adopted, consideration was given to the situations likely to result in an actual fall rather than a temporary loss of control or stability which is likely to be recoverable. It was concluded that a fall was most likely to result when a loss of grip from both hands occurred in a non-recoverable manner; that is, with the body moving away from the steelwork or ladder. Loss of foot contact by either or both feet is unlikely to result in a non-recoverable situation, provided one or both hands retain a grip. The next step was to simulate the fall condition where sudden loss of grip from both hands occurred, but one or both feet remained initially in contact with the structure. The arrangement adopted is shown in Fig. 4(*a*).

Testing Arrangement

The structure used for testing was a Mast No. 1A, which is a triangular lattice steel mast of height 57 m with built-in climbing steps. This structure is in common use in BT International's radio stations and uses Type 1 fall arrestors to provide a safe means of access.

Before each test, the dummy was manually set up in a position such that the hands were loosely taped to the mast and one or both feet located on the climbing steps. A winch and lifting rope were connected via a quick-release mechanism to an eye-bolt in the dummy's head. On release of the dummy, the hands fell free of the mast and the body pivoted outwards, therefore simulating the non-recoverable fall situation.

Each test was recorded by using high-speed filming techniques (8 × normal) so that the results obtained could be studied in detail. The Type 2 fall arrestors performed perfectly well, each time arresting the fall in a minimal distance. The results with Type 1 devices were not so reassuring. One particular design, consisting of a carriage moving on a fixed rail, failed on two occasions to prevent the dummy falling to the ground. Figs. 4(*a*), (*b*) and (*c*) show one such occasion when the dummy's fall was not arrested.

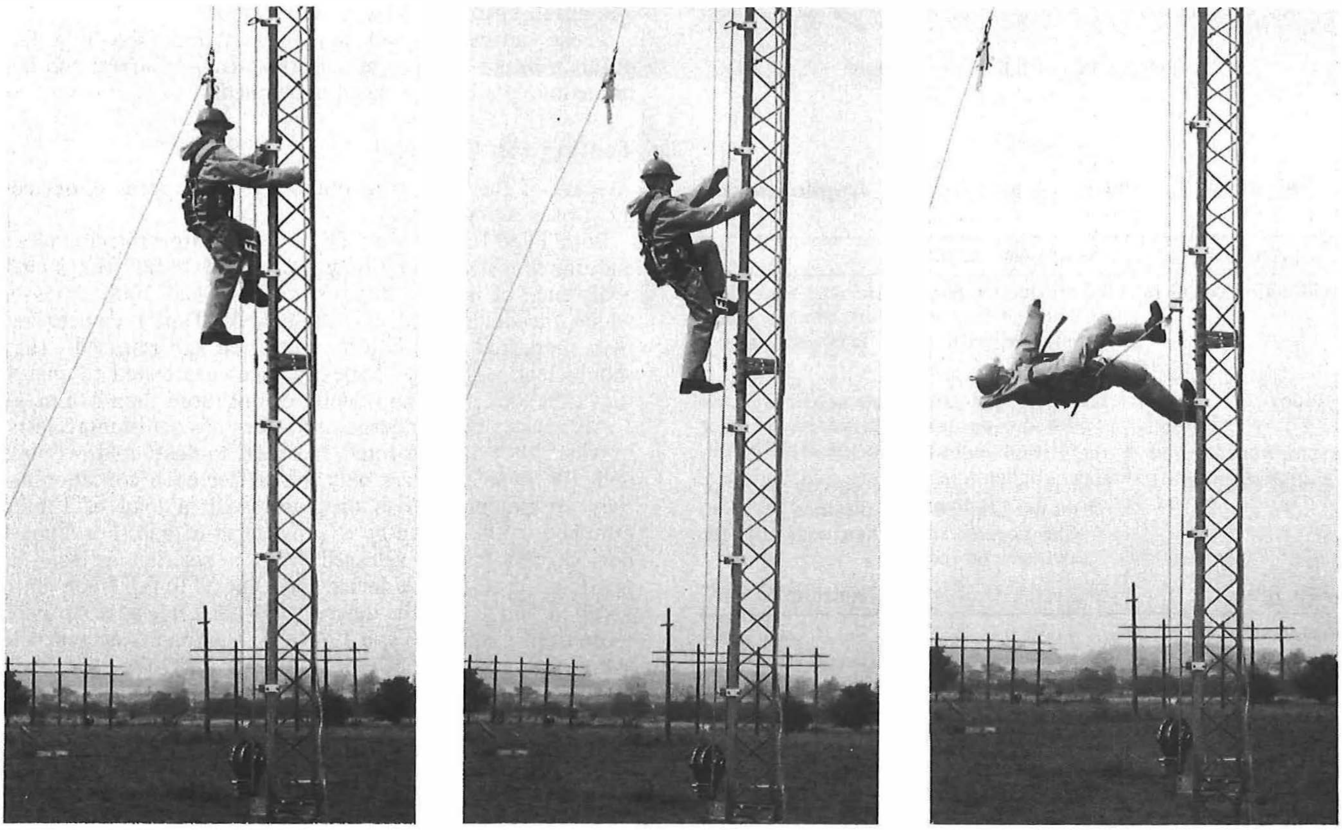
RESULTS

From Figs. 4(*a*), (*b*) and (*c*), the pivoting action of the body away from the mast (a non-recoverable situation) can clearly be seen. This pivoting action imposes a horizontal force on the carriage which is not representative of a free-fall situation (downward movement only), and so the carriage either does not lock on the rail, or having initially locked is then subsequently caused to release, permitting the dummy to fall.

Corrective Action

As soon as this mode of non-operation was realised, the particular fall arrestor was temporarily withdrawn from use in BT pending further investigations. The manufacturer was also informed of the findings.

If the pivoting action of the body could be eliminated or reduced to an acceptable minimum, it seemed likely that this would markedly improve the performance of these fall



(*a*) (*b*) (*c*)

FIG. 4—Stages in the fall of a non-arrested dummy

arrestors. After several unsuccessful arrangements were tried, the desired effect was achieved:

(*a*) by using a connecting lanyard of length not greater than 260 mm;
(*b*) by changing the angle at which the rail is connected to the mast through 45° for ease of climbing with the now shorter connecting lanyard; and
(*c*) by making a small modification to the fall arrestor itself, whereby the angle between the attachment dee-ring and the carriage was changed.

This arrangement resulted in a vast improvement in the performance of the fall arrestor with insignificant movement of the carriage down the rail during repeated tests.

The film and findings were presented to the BSI Committee concerned with BS5062 and, as a result, the British Standard is to be amended to include performance tests using an anthropometric dummy. As it is essential that the tests are repeatable, the dummy used for the tests will have locked knee joints.

The film has also been shown to a meeting of the ISO†, and resulted in a resolution being passed to specify the use of an anthropometric dummy in performance tests in international standards.

CONCLUSIONS

From the results of the tests and examination of the film, it has been shown that the length of connector between a harness and fall arrestor can dramatically affect the performance of the fall arrestor. It has also been shown that, in order to evaluate the ability of a fall arrestor to arrest the fall of a man, the tests used should simulate as closely as possible an actual man-fall situation. An anthropometric dummy has been shown to provide a safe and efficient method of achieving this.

The results and recommendations have been accepted by both the BSI and ISO, who will be including the use of dummies in their specifications for performance testing of fall arrestors.

† ISO—International Standards Organisation

ACKNOWLEDGEMENTS

The author wishes to thank Mr. J. P. Hartley for his assistance in producing this article, Mr. A. Mousley and his rigging gangs for their assistance during testing, and the staff of BT Reprographics for producing the visual records.

Biography

Ken Clark joined the then Post Office in 1963 as a Youth-in-Training with the London Test Section at Studd Street. He spent some time at contractors' works involved in the acceptance testing of 4 GHz analogue microwave radio systems for the Conversion of Rural Manual Exchanges (CRMX) project for the Highlands and Islands scheme in Scotland, before moving, on promotion, to the Technical Liaison Branch of Supplies Division. Since 1973, he has worked in the Microwave Radio Works Section, becoming the Head of Group Radio Safety Officer in 1981. On the formation of NN, he became the NN Safety Officer, and is now responsible for advice on safety matters, both technical and legal, on all aspects of NN's activities.

Book Reviews

Analog and Switching Circuit Design. J. Watson. Adam Hilger Ltd. xv+602 pp. 235 ills. Hardback: £40·00; Softback: £18·50.

The author has provided a broadly based introduction to bipolar and field-effect devices described in terms of their electrical properties and, assuming only a basic knowledge of circuit theory, algebra and calculus, he establishes the basic principles whereby devices may be combined to produce useful circuit functions. The text is supplemented by numerous worked examples which, together with an appendix containing problems and their solutions, makes the book particularly suitable for undergraduates and engineers seeking an introduction to analog- and switching-circuit design. The book places emphasis upon applications in control and instrumentation and, from a telecommunications engineering viewpoint, the value of the work is somewhat limited by the virtual exclusion of any treatment of digital or high-frequency applications and the restriction of integrated-circuit design to quite low levels of complexity.

D. BAKER

Introduction to Adaptive Filters. Simon Haykin. MacMillan. xii+217 pp. 59 ills. £35·95.

Over the next few years, a plethora of books will appear on the subject of adaptive filters, which are capable of modelling unknown processes, particularly as these devices become more widely used and their power and versatility is appreciated by engineers. Currently, I know of two other books which are in some stage of production. As to how Professor Haykin's contribution will rank in the final list of books remains to be determined, but in terms of organisation and clarity of explanation it will certainly be hard to improve upon in the market for which the book is obviously aimed. Should the subject appear on the curriculum of engineering degree courses (probably the final year), a reference would be required which is relatively simple, yet detailed enough to provide a deep understanding of the mechanisms of adaptation. The book fits admirably into this category.

Coverage is restricted to filters that work in the sampled-data domain and that use only arithmetic (linear) operations. The class of filters described is limited to the time domain and have a finite-impulse response (FIR). Thus, two very complex categories are excluded, recursive structures (in the time domain) and transform techniques (for example, the frequency domain). All the theory is treated in matrix algebra, which is not particularly difficult, although occasionally one may be forced to a reference for clarification; many of the matrix properties used are described in the text in simple terms. It may be noted that this is not the only approach one can take, and a better understanding of certain characteristics can be obtained by statistical analysis.

An important feature in introducing a subject is the organisation of the material, and here an excellent formula has been achieved. Initially, one is taken through optimum filters and linear prediction, using FIR and lattice structures. Good use is made of examples to describe the transition from single- to multi-stage filters. The remainder of the book deals with solving for the filter coefficients by using recursive algorithms for the two principal types, FIR and lattice. Included at the end of each chapter are notes describing the historic development of the principles covered in the text and giving references to papers which have extended the work.

Only one minor criticism concerns some typographical errors, but these are not serious and will become obvious to anyone reading the book.

C. R. SOUTH

The Development of Single-Mode Fibre Transmission Systems at BTRL

Part 1—Early Developments

R. C. HOOPER, B.SC., C.ENG., M.I.E.E., D. B. PAYNE, B.SC., M.SC., A.M.I.E.E., and M. H. REEVE, B.SC., C.ENG., M.I.E.E.†

UDC 621.391.63

In this first of two articles, the development of single-mode fibre systems at British Telecom Research Laboratories from 1979 to 1982 is charted with particular emphasis on the field-trial results and the impact upon the evolution of the trunk network. The second part of this article will describe the developments to date and discuss the future potential of single-mode fibre applications.

INTRODUCTION

The origins of single-mode (monomode) fibre as a transmission medium stretch back to the seminal ideas[1] of Kao and Hockham in the mid-1960s. The enormous bandwidth potential of light frequency ($\approx 10^{14}$ Hz) carriers was one of the main attractions of this medium and stimulated an intensive research programme at the British Telecom Research Laboratories (BTRL). It was recognised that, if solutions to the many practical difficulties could be found, then the communication engineers' dream of virtually unlimited bandwidth would be realised. The translation of this dream into reality has taken a more tortuous path than perhaps had first been imagined. However, over the past four years, as this realisation has taken shape, the long-term prospects have come more into focus to indicate areas of application not considered in the early days.

The practicalities of fabricating, splicing and coupling low-loss single-mode fibres, with core dimensions of only a few microns, were identified at an early stage as being among the major problems to be solved. The requirement in the early-1970s for a rapid evolution of fibre technology to the systems demonstration stage meant the temporary abandonment of single-mode fibre for multimode fibre. This fibre, with its large core diameter of several tens of microns, offered simpler splicing and coupling than single mode, albeit with low bandwidth due to the dispersion caused by the variation in the velocity of the many propagating modes. In consequence, a concentration of effort at low bit rates (8 Mbit/s) ensued and, in 1975, at the opening of BTRL at Martlesham Heath, an 8 Mbit/s system operating over a 6 km multimode fibre link was demonstrated.

In 1977, two significant field-trial systems using cabled multimode fibre installed in duct in the Martlesham area were demonstrated[2, 3]. Although originally planned as a step-index multimode experiment, graded-index fibre cable having superior bandwidth performance, due to its mode velocity equalisation property, became available and allowed experiments to be carried out at 140 Mbit/s as well as 8 Mbit/s. These experiments gave the first indications that fibre cables could be successfully installed in duct, spliced (jointed) and operated over repeater sections much longer than conventional copper-media based systems.

During the late-1970s, silica-based fibre, with losses of less than 5 dB/km in the 800–900 nm wavelength region, was beginning to dominate the fibre scene. Reliability performance of the opto-electronic devices (light-emitting diode (LED) or laser and photodiode) was improving and there was the growing prospect of operational systems being required by BT. However, in the mid to late-1970s, research establishments throughout the world were drawing attention to the possibility of lower-loss fibre, perhaps as low as 1 dB/km, at 1300 nm wavelength. Also, the chromatic dispersion, normally a significant bandwidth reducing factor at 800–900 nm with modulated LED and laser sources having wide optical spectra, is virtually zero in silica fibre at 1300 nm wavelength.

Opto-electronic device development for the 1300 nm wavelength region required new materials technology and, in the late-1970s, research programmes were gradually shifted to producing suitable semiconductor lasers and photo diode detectors.

The new wavelength region stimulated a re-awakening of single-mode technology since yet lower loss still, $<0\cdot5$ dB/km, could be achieved, and work began in earnest at BTRL in 1978. Since modal dispersion disappears with single-mode propagation and chromatic dispersion had been virtually removed by the shift in wavelength, the fibre now offered almost limitless bandwidth. The development had come full circle, but with the prospect of repeater spacings at least of an order of magnitude greater than for equivalent copper-media systems.

In 1979, BT embarked on a major ordering programme for fibre systems for trunk and junction routes. These orders were for systems operating at 800–900 nm wavelength. Clearly, the future ordering programme would increasingly gravitate towards 1300 nm operation as confidence grew in the device technology for the longer wavelengths. The question then posed was should the network opt for a single-mode or multimode solution. The advantages of the single-mode solution were so attractive for the trunk network that BTRL stepped up the programme of research begun in 1978 to attempt to solve the practical problems that had remained since the early-1970s. In 1980, a demonstration of a 37 km unrepeatered system operating at 140 Mbit/s and 1300 nm during the first Martlesham Open Week indicated the potential of the single-mode approach. This experiment also demonstrated for the first time that the 30 km maximum separation of power feed stations in the UK trunk network could be bridged without a repeater. The components for this demonstration were all developed at BTRL initially for the inland network, but it had been realised that systems using single-mode fibre were well suited to undersea applications where maximising repeater spacing was a prime concern. In 1980, a 10 km link working at 140 Mbit/s and using identical components was demonstrated at Loch Fyne in collaboration with Standard Telecommunication Laboratories (STL). These and other demonstrations of single-mode capability encouraged the development of solutions to the cabling, splicing, coupling and systems problems.

† Research Department, British Telecom Development and Procurement

The culmination of the initial phase of research activity on single-mode fibre systems was the demonstration of field installed cables having low splice losses and carrying digital signals at up to 650 Mbit/s over links up to 60 km without intermediate repeaters. Several demonstrations of this type were undertaken during 1982, with further upgrading experiments performed subsequently, some of which took place in the lower-loss window at 1550 nm.

SINGLE-MODE FIBRE

The description of propagation in single-mode fibre has been dealt with elsewhere[4]; note that the ray-path approach used with multimode structures is no longer applicable and that electromagnetic (EM) field theory must be used. In single-mode fibre, the core diameter can be only 8 μm with a cladding diameter of about 125 μm.

Current single-mode fibres are based on silica, doped with other materials in order to produce the refractive index difference between core and cladding necessary to give guidance or propagation. Various materials-related loss mechanisms in these glasses produce absorption or scattering of the optical power in the propagating mode, and thus limit the available system repeater spacing. At shorter wavelengths, the ultra-violet absorption edge dominates, while at long wavelengths, the infra-red absorption from the oxides of the glass is the major loss effect. In between these two regions, Rayleigh scatter (due to particles of size comparable to the wavelength of the light transmitted) is, for the most part, the dominant loss mechanism. However, at wavelengths corresponding to the oxygen–hydrogen (O–H) bond overtone absorption lines (950 and 1370 nm) and at associated combination bands, absorption losses can be dominant. These effects result in two low-loss windows, at 1300 and 1550 nm. Attenuation in these windows is due to Rayleigh scatter, remnant O–H absorption and absorption from combination bands of the fibre dopants with O–H. Unfortunately, dopants added to the silica to produce the necessary fibre properties tend to increase the Rayleigh scatter. The need to keep processing temperatures as low as possible during fabrication militates against the use of pure silica for either the core or the cladding (in spite of the attraction of such structures on loss grounds). The necessary processing temperatures reduce with increasing dopant levels. These loss mechanisms and the best combination of dopants to use for the core and cladding for lowest loss have been the subjects of extensive study at BTRL[5, 6]. The resulting solution uses a core region uniformly doped with germania (GeO). The cladding region uses both fluorine and phosphorus as dopants. These two dopants have opposite effects on the silica refractive index, allowing the cladding index to be matched to that of silica whilst enabling the processing temperature to be lower than that for pure silica. The dopant concentration in the cladding is arranged to be a maximum furthest from the core, reducing to nearly zero in the most loss-sensitive region near the core. In this way, losses can be reduced to minima of around 0·3 dB/km and 0·2 dB/km in the 1300 and 1550 nm windows, respectively.

Whilst the single-mode fibre eliminates modal dispersion, the small core region presents a challenge for splicing or jointing, since the cores must be accurately aligned, perhaps to within $\pm 0 \cdot 5$ μm. The launching of light from a source such as a semiconductor laser, with its divergent asymmetric radiation pattern, into a small-core fibre is also more demanding in alignment accuracy. These factors were partly responsible for single mode being put to one side in the early to mid-1970s in favour of multimode technology.

In single-mode fibre, chromatic dispersion is the major bandwidth limiting effect. (Polarisation dispersion due to any slight ellipticity of the core also occurs, but this effect can usually be neglected.) Since most light sources, including lasers, do not always emit just a single wavelength, then the dependence of the fibre material refractive index (and hence velocity of propagation) on wavelength leads to dispersion effects. The semiconductor laser often emits a broad spectrum or 'comb' of wavelengths with each 'tooth' having a slightly different propagation velocity and, therefore, transit time along the fibre. The resulting dispersion, which produces intersymbol interference (ISI) in a digital system, can be obviated by reducing the spread of wavelengths (linewidth) emitted by the source and/or choosing the wavelength region of operation to be where the dispersivity is a minimum. Fortuitously, silica exhibits zero chromatic dispersion at 1300 nm, a wavelength at which the fibre loss is invitingly low ($<0 \cdot 5$ dB/km).

A typical spectral-loss plot for single-mode fibre is shown in Fig. 1. The two low-loss windows are evident; one around 1300 nm and the other centered on 1550 nm. A large part of the loss is due to Rayleigh scatter in the silica. Since this loss reduces with the fourth power of wavelength, the fibre loss is lower at 1550 nm than at 1300 nm. However, the chromatic dispersion at 1550 nm is not zero and narrow linewidth laser sources will be needed to exploit effectively the lower loss.

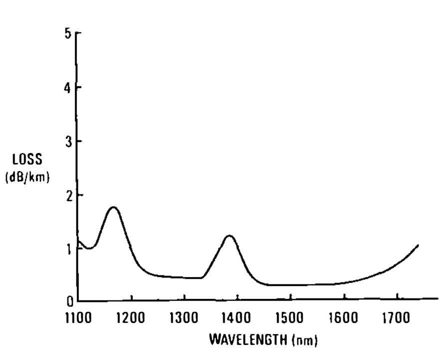


FIG. 1—Spectral loss of a single-mode fibre

Single-Mode Fibre Design and Specification

A more detailed investigation of the design of the single-mode fibre is necessary to assess properly the above properties. One of the main aims was to arrive at a fibre design that would work well in existing cable structures and allow 'standard' cabling techniques in manufacture and installation to be used for both multimode and single-mode fibre.

The need to minimise overall link loss produces several detailed constraints on the fibre design:

(*a*) *Cut-off wavelength* This is the wavelength which marks the transition between multi- and single-mode operation; above this wavelength the operation is single mode. In general, the cut-off wavelength must be below the operating wavelength.

(*b*) *Intrinsic fibre loss* This is the combination of Rayleigh scatter and losses due to dopants introduced into the silica to form the fibre core and/or to water (O–H ions) in the fibre.

(*c*) *Incremental cable loss due to microbending* This is superficially similar to the same phenomenon in multimode fibre, but here concerns minute bends in the fibre, coupling only one guided mode out of the core.

(*d*) *Splice loss* This is due to a combination of concentricity error, EM field mismatch and splicing techniques used.

Work at BTRL in the late-1970s was concerned with experimentally investigating these factors to arrive at the optimum design. The fibre design is specified in terms of the two measured parameters: the mode cut-off wavelength and

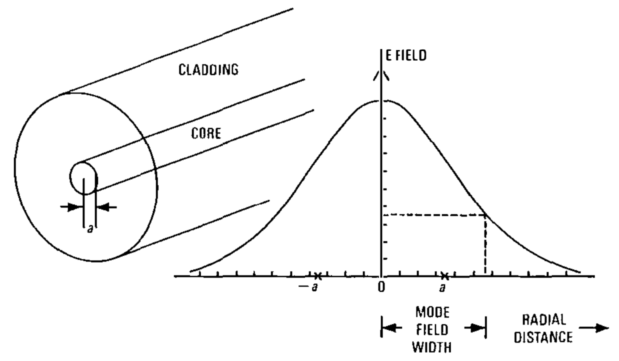


FIG. 2—Approximation to the electric field (mode field) distribution in a single-mode fibre

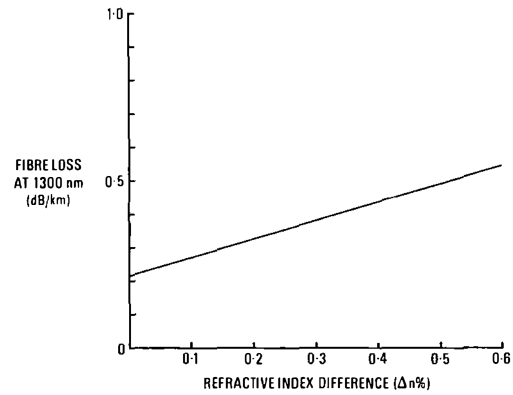


FIG. 4—Dopant level dependent loss as a function of index difference (Δn%)

some measure of the mode field width (Fig. 2). Care must be taken to link these experimentally-determined quantities to the measurement technique used, an important technique being that of the offset joint[7].

If a plot with these two quantities as axes is produced, the fibre specification appears as an area on the plot, with the constraints as lines bordering the specification 'box'. This is shown in Fig. 3. Taking each constraint in turn:

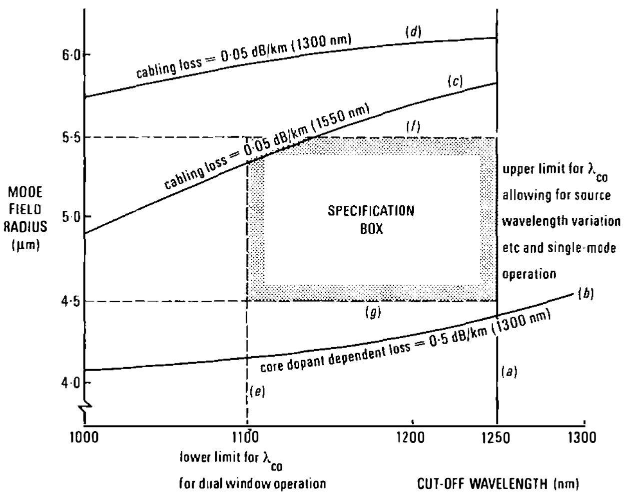


FIG. 3—Fibre design diagram

Cut-Off Wavelength

From a consideration of source wavelength variations, it is desirable to specify the fibre loss and to operate the systems over a wavelength range of 1275–1325 nm in the 1300 nm window. To avoid any possibility of bi-moded fibre at an operating wavelength, an upper value of 1250 nm is placed on the cut-off wavelength—this is line (*a*). A lower limit of 1100 nm has been adopted in order to minimise the effects of cabling loss and dispersion—this is line (*e*).

Intrinsic Fibre Loss

Fibre loss increases with doping level (germanium and fluorine are added to the fibre core to raise its refractive index above that of the cladding) and with O–H content. The dopant-level dependent loss places an upper bound on the fibre refractive index difference, leading to a lower bound on the field width for any particular cut-off wavelength. Fig. 4 shows empirical data for this loss at 1300 nm as a function of index difference, after the O–H contribution has been removed. If 0·5 dB/km is taken as a maximum for this loss, boundary line (*b*) in Fig. 3 is produced. The O–H contribution can be expected to add less than 0·05 dB/km to this figure.

Incremental Cabling Loss

In step-index single-mode fibre, the mode becomes less well guided as the wavelength is increased away from the cut-off, and eventually becomes susceptible to microbending loss from the cable structure. A fibre for operation in both windows must have the cut-off below 1300 nm and the microbending loss edge beyond 1550 nm. For any particular fibre core diameter, this places a lower limit on the fibre refractive index difference, corresponding to an upper limit on the mode field width. Since various types of cable are used by BT, it becomes a matter of some difficulty to determine the level of microbending to which any particular fibre will be subjected. Empirical data for a loose-tube coating is shown in Fig. 5. The worst-case situation for this cable type corresponds with the fibre left under some tension against the tube wall. Fig. 6 shows similar data for a tight secondary fibre coating. In spite of the very different cabling situation, a similar lower limit on index difference (Δn) is obtained in each case. Assuming a limit of 0·05 dB/km incremental loss from microbending, the loose-tube data is plotted in Fig. 3 as line (*c*). If operation at 1300 nm only is required, the data can be converted theoretically to show the same limit at 1300 nm as line (*d*). Fibre within the

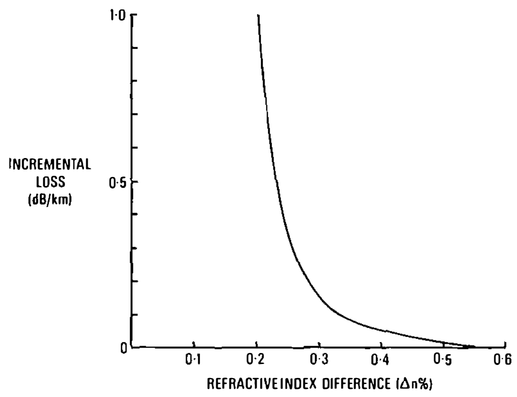


FIG. 5—Incremental loss at 1550 nm for a loose-tube fibre versus index difference (Δn%)

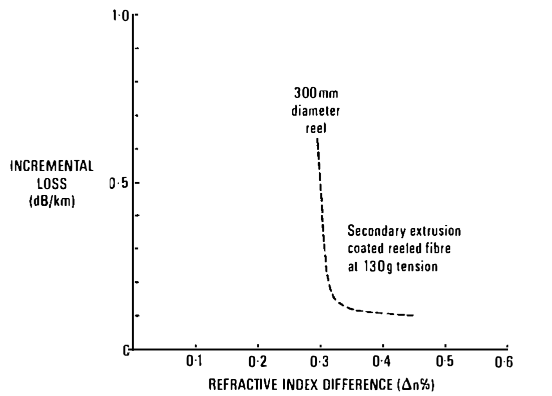


FIG. 6—Incremental loss at 1550 nm for a tight secondary coating versus fibre index difference (Δn%)

specification box could thus be expected to have average losses of around 0·45 dB/km at 1300 nm and 0·3 dB/km at 1550 nm, and could be cabled by using existing cable types with negligible loss increase at either wavelengths.

Dispersion

The nominal step-index profile fibre design adopted by BT has a zero chromatic dispersion point around 1325 nm. Near the dispersion minimum, broad spectrum lasers can be used without incurring a serious penalty. As operation moves from the zero dispersion point, the penalty increases, and ultimately requires the use of narrow spectrum sources.

The dispersion characteristics of step-index fibre meeting the BT specification are shown in Fig. 7. Curve 1 is for a

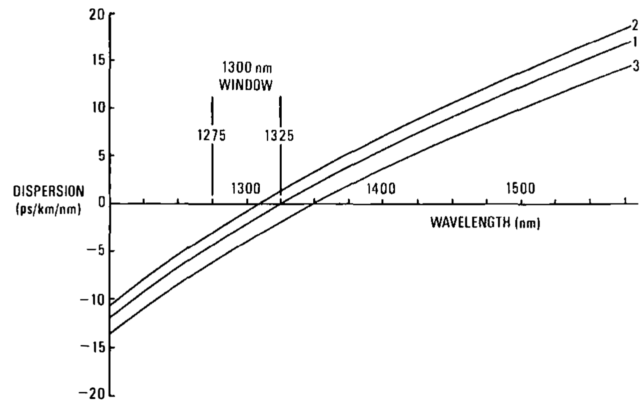


FIG. 7—Dispersion versus wavelength for single-mode fibres

typical fibre with dispersion minimum at 1325 nm and the curves either side show the extremes that are expected for fibre falling within the specification box. Curve 2 corresponds to a fibre in the top-right-hand corner of the box and curve 3 to a fibre in the lower-left-hand corner. It is fibres in this region of the specification box that will first give rise to dispersion penalties and, therefore, is a further incentive to restrict the lower cut-off wavelength bound to 1100 nm.

Splice Loss

The main splicing technique for single-mode fibre is currently electric-arc fusion. Core concentricity and mode field width mismatch form the main fibre-related loss contributors. The effect of any given concentricity error on splice loss reduces as the field width is increased, making it desirable to operate towards the top of the area between the cabling (*c*) and cut-off (*a*) lines. Unfortunately, this reduces the available cut-off range; a good compromise is a field width centre of 5 μm. At this field width, a concentricity error of 0·5 μm in the two fibres could produce a maximum splice loss of around 0·2 dB due to this effect. Fibre production spreads and joint alignment statistics give an average splice loss considerably less than this figure. A field width range of ± 0·5 μm around the 5 μm centre (lines (*f*) and (*g*) in Fig. 3) gives a possible maximum splice loss due to field width mismatch of 0·17 dB. Again, an average value much less than this would be expected when the spread of fibre parameters and the statistical aspects of splicing are taken into account.

SPLICING TECHNIQUES

There are several well-documented techniques for the splicing of single-mode optical fibres and these can be generally classified on the following basis:

(*a*) fully-fused splices using cladding alignment only (optical power monitoring is not used);

(*b*) partially-fused splices using core alignment by optical power monitoring, to reduce losses due to core concentricity errors;

(*c*) fully-fused splices where core offset is minimised by using optical power monitoring and fibre rotation; and

(*d*) glued splices where core alignment by optical power monitoring is used to reduce core-concentricity errors, and core deformation is eliminated by avoiding the fusion process.

The fusion process involves butting together the two fibres to be spliced in the vicinity of some heating device, usually an electric arc. The fibres are then simply melted or fused together. In order to obtain low losses, however, this process has to be very carefully controlled, usually by a microprocessor. The glue technique bonds the fibres together by using some form of transparent adhesive.

The choice of technique to be used depends on the fibre specification, design and its application.

Splice Performance Comparison

The fundamental performance criterion is mean splice loss. In general, the mean splice loss in single-mode fibres can be considered to consist of three components:

(*a*) field width mismatch;

(*b*) core displacement due to core concentricity error; and

(*c*) core deformation due to the fusion technique.

Table 1 summarises the contributions from these components to the total splice loss for the four splicing classifications given above.

Field Width Mismatch

The loss contribution from field width variation is a fundamental splice loss limit. If Gaussian statistics for the distribution of field widths and the limits mentioned above (5 μm ± 0·5 μm) are assumed, then the mean loss introduced is less than 0·03 dB.

Core Concentricity Error

The question of which splicing technique to adopt hinges on the effect of core concentricity on splice loss. The allowable loss that can be allocated for concentricity effects will in turn determine the geometrical parameters in the fibre specification required for each of the splicing techniques discussed. The fibre specification will then determine fibre yields from given manufacturing processes and bear directly on the fibre's cost.

In order to assess the allowable allocation of splice loss arising from the effects of concentricity error alone, it is

TABLE 1
Splice Performance Comparisons

Loss Mechanism	Method (*a*) Fusion Cladding Alignment	Method (*b*) Fusion Core Alignment	Method (*c*) Fusion Fibre Rotation	Method (*d*) Glue Splicing
Field width mismatch	Same for all techniques			
Core displacement due to concentricity error	Depends critically on fibre specification	Does not depend significantly on fibre specification. Limit depends on alignment accuracy	Depends on fibre specification, but may be relaxed compared with method (*a*)	Independent of fibre specification. Limit given by alignment accuracy
Core deformation	Is minimised by machine parameters. <0·08 dB for standard system fibre	Additional deformation due to surface tension forces trying to align claddings. Minimised by short-duration fusion cycle	Same as for method (*a*)	None

necessary to examine system requirements and power budgets. A detailed statistical analysis was used to obtain a value for the mean system splice loss that can be tolerated for a 30 km 140 Mbit/s system. Then, by using the distributions of the component losses (field width mismatch, core deformation and concentricity error), values for the maximum tolerable concentricity error can be determined. (Less than 0·7 μm should give a mean loss below 0·08 dB, which is adequate for this purpose.)

Core Deformation

Splice loss due to core deformation occurs to some degree in all fusion-splicing processses and arises from three major causes.

(*a*) Surface-tension forces act to align the claddings of the fibre at the splice point. The effect, therefore, of any initial cladding misalignment is to introduce a shear force across the faces of the fibres during fusion.

(*b*) End-face angle on the fibres being fused also causes core deformation. When a single-mode fusion splicing machine is set up and operated correctly, end-angle effects are probably the dominant cause of core deformation.

(*c*) Contamination of the fibres by foreign material in the splice region, for example, dirt or primary coating material not being completely removed, is an often underestimated contribution to poor-quality fusion splicing.

Fusion-splicing techniques suffer from about the same amount of core deformation. Method (*b*) using power monitoring during core alignment will be prone to surface-tension forces increasing with core concentricity error. The effect of this may be mitigated by a short-duration heating cycle optimised for this method. On the other hand, method (*a*) uses these surface-tension forces for the final alignment of the fibre claddings. It is found that the initial visual cladding alignment under a low-power microscope is generally accurate to the order of 1 μm and that this residual misalignment produces insignificant core deformation. The glued-splice technique produces no core deformation and, in principle, could produce the lowest loss splices. Note, however, that there may be a small loss due to the refractive index mismatch (Fresnel loss) at the fibre/glue interfaces.

Splice Strength

High-strength fusion splices may be achieved with care and attention to absolute cleanliness in the preparation of the ends of the fibre. Post-etching in hydrofluoric acid increases the strength of the fusion splices dramatically. This high-strength splicing may be achieved with techniques (*a*) and (*c*). Splices made by using technique (*b*) are likely to be weaker because of the discontinuity in the outside diameter and because a full fusion does not take place. The strength of glued splices is likely to be lower than those achieved by using fusion techniques. In general, for land-based systems, strain relief for the fibres and splices is provided in the joint housing and very-high-strength splices are not essential. The major application for high-strength splices is in undersea systems, where production-line high-strength splicing is required in order to produce long cable lengths without full-joint housings.

Complexity of Splicing Technique

There are two distinct categories of splicing techniques described above: those where optical power monitoring is required and those where it is not. In technique (*a*), the splicing-machine operator visually aligns the fibre claddings to an accuracy in the order of 1 μm by using micromanipulators and a two-axis microscope. The fibres are butted together and automatic fusing takes place under the precise control of a microprocessor. Full fusion occurs, thus allowing the final very precise alignment of the fibres to be brought about by surface-tension forces.

The simplest power-monitoring system requires optical power to be launched into the remote end of the fibre to be spliced and either a distant or local receiver to feed back a signal related to the power coupled into the spliced fibre. The fibres are then aligned either by the operator or automatically so that the coupled power is maximised prior to splicing.

Power-monitoring systems can require an extra team at the launch site and possibly a further team at the receive site if local monitoring is not used. In addition, sequential splicing along a route may often be necessary. This could be disadvantageous since installation practices and local road traffic problems may restrict splicing to times outside normal working hours on difficult or busy sections of the route.

If a local injection/detection scheme is used to avoid the above problems, the length of stripped fibre at the splice could be fairly large, and this increases the risk of damage to the fibre and makes subsequent mechanical protection more difficult. However, injection/detection through the coatings can be used in order to avoid long lengths of uncoated fibre, but this is dependent on the choice of materials used for the primary and secondary coatings.

As a result of the detailed studies of splicing techniques undertaken at BTRL during the late-1970s and early-1980s, BT has adopted fusion splicing without power monitoring as the initial splicing technique for the trunk network modernisation programme. The main reason for this is the simplicity of the technique for the field environment. As the use of single-mode fibre moves into the junction and local networks, the need for simple and low-cost splicing techniques increases. The fusion process adopted offers the greatest potential for simple low-cost, yet high-quality, splicing.

SINGLE-MODE FIBRE SYSTEMS TRIALS

The production of quantities of experimental single-mode fibre at BTRL with loss less than 0·6 dB/km at 1300 nm wavelength led to a series of system tests beginning in 1980. The development of suitable transmitter and receiver modules at BTRL had been proceeding during the late-1970s, initially for graded-index multimode systems, but subsequently for single-mode fibre applications

Transmitter and Receiver Modules

The basic opto-electronic devices for operation at 1300 nm, that is, lasers and photodetectors, were incorporated into hybrid packages, usually in dual-in-line format, with integral fibre tails for connection to the main transmission fibres. A transmitter module typically comprised an optical source, often a semiconductor laser chip, and the necessary optical coupling components such as miniature lenses and a copper submount alignment jig to achieve an efficient launch of optical power into the single-mode fibre tail[9]. This package, shown in Fig. 8, was also able to incorporate electronic

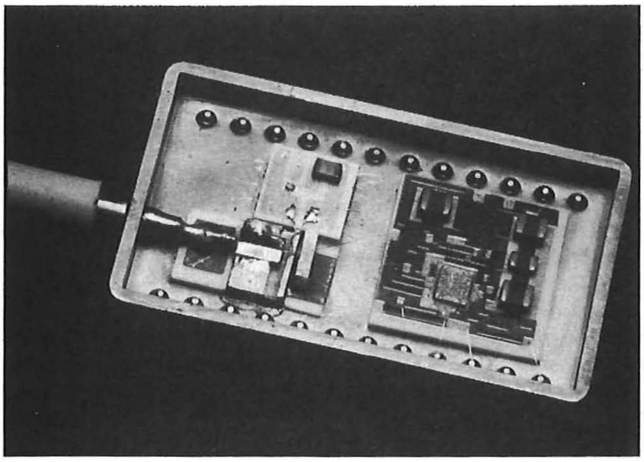


FIG. 8—Laser transmitter module showing fibre (left), submount, laser chip and feedback photodiodes (centre left) and IC controller chip (right)

interfaces in the form of drive and bias circuits. The stabilisation of the laser operating point against temperature variations and ageing was of major importance since it was recognised that 1300 nm lasers were likely to be more temperature sensitive than their 850 nm counterparts. Successful operation at 140 Mbit/s and above in a practical systems environment depended on the effect of laser switch-on delay[10] being minimised without other performance criteria being compromised. The development of the module therefore included a rear laser facet monitoring photodiode and specially developed BTRL integrated circuit for the control of the bias and drive signals[9].

The hybrid packaging concept became vitally important to the development of receivers at 1300 and 1550 nm. Fundamental changes to receiver design became necessary when the new wavelength region was adopted, since the silicon avalanche photodiode (APD) that had given such good performance at 850 nm could not be made to detect beyond about 1100 nm. The APD can be regarded as a device that detects incoming optical signals, and converts them to equivalent electrical current signals which undergo amplification within the device itself by a carrier avalanching effect; the gain is dependent upon the applied voltage. The effective noise figure of such a device is dependent in a complex fashion upon the gain and the level of the input signal. The commercially available silicon APD could be operated with a noise figure much less than that of a conventional transistor amplifier. However, commercially available APD devices operating at 1300 and 1500 nm were germanium based with effective noise figures much higher than those for silicon and with a significant temperature dependency as well.

A number of research laboratories undertook development of APD devices in materials related to those being used for laser development; that is, gallium indium arsenide (GaInAs) and gallium indium arsenide phosphide (GaInAsP) III-V based semiconductor alloys. However, at BTRL, it was believed that any long-wavelength APD development would be on a long time-scale since these devices are structurally complex and therefore difficult to fabricate (true even in silicon, which is a well-characterised material). For this reason, the development of a simple non-avalanche GaInAs PIN structure photodiode was initiated by BTRL both internally and at Plessey Research. In order to achieve adequate sensitivity performance, it proved necessary to ensure that both the photodiode and following transistor preamplifier had low capacitance and that the input transistor was a low-noise device. Only by using hybrid integration techniques was it possible to achieve this first objective. The receiver, shown in Fig. 9, comprised a PIN

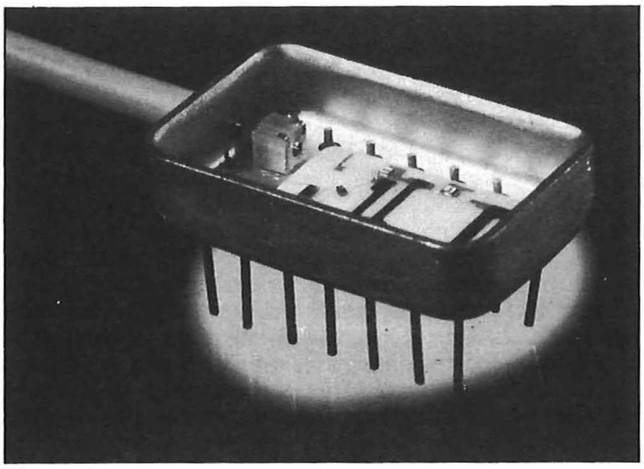


FIG. 9—PINFET receiver module showing fibre entry (left), quartz cube submount for photodiode (centre left) and preamplifier circuit (centre and right)

photodiode connected to a low-noise GaAs field-effect transistor (FET) device forming the first stage of amplification. Further amplification and buffering was also provided in the package by using additional bipolar transistor devices. The resulting module, known as a *PINFET* receiver, had sensitivity performance better than commercial germanium APDs in 1979[11]. More recently, both III-V and germanium-based APDs have been developed in the USA and Japan. These detectors have shown improved sensitivity so that performance is now comparable with PINFET receivers, which have also advanced in this period. Operational considerations are now, and in the future, likely to play a more dominant role in the choice of receiver technology.

Initial System Tests

The development of the transmitter and receiver modules at BTRL was undoubtedly a key factor in the successful demonstrations that took place during the 1980s over long lengths of single-mode fibre. In the first of these experiments, a prototype laser transmitter operating at 1275 nm was modulated with a 140 Mbit/s $2^{10}-1$ non-return-to-zero (NRZ) pseudo-random binary sequence (PRBS) test signal. The mean output power coupled from the laser into the fibre without lensing was 80 μW, or −11 dBm—an efficiency of about 10%. A continuous 37 km length of single-mode fibre coated, but not packaged or cabled, was butt connected to the fibre tail of the transmitter module. The loss of this fibre was about 0·7 dB/km at 1275 nm, slightly higher than the 1300 nm figure. The short multimode fibre tail of the receiver was connected to the main fibre via a lens connector. The sensitivity of the PINFET receiver was measured to be 50 nW, or −43 dBm, at 140 Mbit/s, with an error rate of 10^{-9}. The photodiode quantum efficiency, the proportion of incoming light that contributes to useful current flow, was 40%. The low-level output signal from a PINFET receiver, usually an integrated version of the received optical signal, was equalised by a differentiator and fed a regenerative repeater.

The repeater comprised linear and limiting wideband amplifiers, a clock extraction circuit and a bistable decision gate. The repeater also included an equaliser, which, in the case of single-mode fibre systems, is usually just a low-pass filter. The gain blocks and digital circuits were implemented by using integrated circuit components; no discrete devices were used. The system block diagram is shown in Fig. 10. This system was successfully operated over the full 37 km unrepeatered length with a system margin of 3 dB. A further experiment was undertaken in the 1550 nm low-loss window where the fibre attenuation averaged about 0·4 dB/km. The laser transmitter was changed for a prototype 1510 nm laser chip mounted on a package header. A BTRL photodiode capable of operating at both 1300 and 1550 nm had been used in the receiver module enabling the receiver/regenerator combination to remain unchanged. The reduced fibre loss enabled an extra 12 km of fibre to be added to the link to form a continuous 49 km length. This system again operated successfully with about a 3 dB system margin. However, some important differences were noted between the two experiments. At 1300 nm, the performance was loss limited, in other words the link length achievable was set only by the laser coupled power, the receiver sensitivity and the fibre loss. On the other hand, at 1510 nm, when the receiver sensitivity was measured through the 49 km length, an additional penalty of about 0·7 dB resulting from the interaction of the broad spectrum of the laser and the fibre dispersion was discovered. The power budgets for these tests are given in Table 2.

TABLE 2

Power Budgets

Wavelength	1275 nm	1510 nm
Link length	37 km	49 km
Bit rate	140 Mbit/s	140 Mbit/s
Path loss	29 dB	24 dB
Transmitter coupled power	−11 dBm	−15·4 dBm
Received power	−40 dBm	−39·4 dBm
Receiver sensitivity	−43·5 dBm	−44·2 dBm
Laser extinction penalty	0 dB	1 dB
Dispersion penalty	0 dB	0·7 dB
System margin	3·5 dB	3·1 dB

Dispersion and Laser Spectrum

The dispersion of the fibre in the 1550 nm window was known to be in the region of 15 ps/nm/km; that is, after 1 km of fibre a launched impulse will have been transformed to a Gaussian pulse with 1/e full width of 15 ps for every nanometre of source linewidth, the linewidth being a measure of the spread of wavelengths emitted by the source. The 1510 nm laser was estimated to have an effective linewidth of 4 nm so that the dispersion amounted to about 3 ns, just under half the bit time for a 140 Mbit/s system. Since the transmitter launched full-width pulses, that is, 7·14 ns wide pulses, and not impulses, the dispersion of the fibre spreads these pulses into their adjacent time-slots, and causes ISI. About 1 dB of ISI penalty results when the dispersion is equivalent to half a time-slot and 5 dB when it is a full time-slot. This initial experiment at 1510 nm was nearing dispersion limited operation.

It was appreciated that if the 1550 nm window of this fibre was to be fully exploited with conventional single-mode fibre, then some measures to narrow the source linewidth needed to be taken. Subsequent 1550 nm source developments were aimed towards achieving this end (at 1300 nm the linewidth problems were of less concern). The spectrum of a semiconductor laser, as mentioned earlier, is often a comb of discrete wavelengths generated by the multiple cavity resonances. When modulated, this spectrum undergoes changes, with the first part of the pulse usually exhibiting the highest degree of instability of spectrum. Under DC conditions, or during the latter part of a long pulse, the spectrum stabilises and, in certain devices, can exhibit nearly single wavelength operation. A possible evolution of the wavelength spectrum of a semiconductor laser is shown in

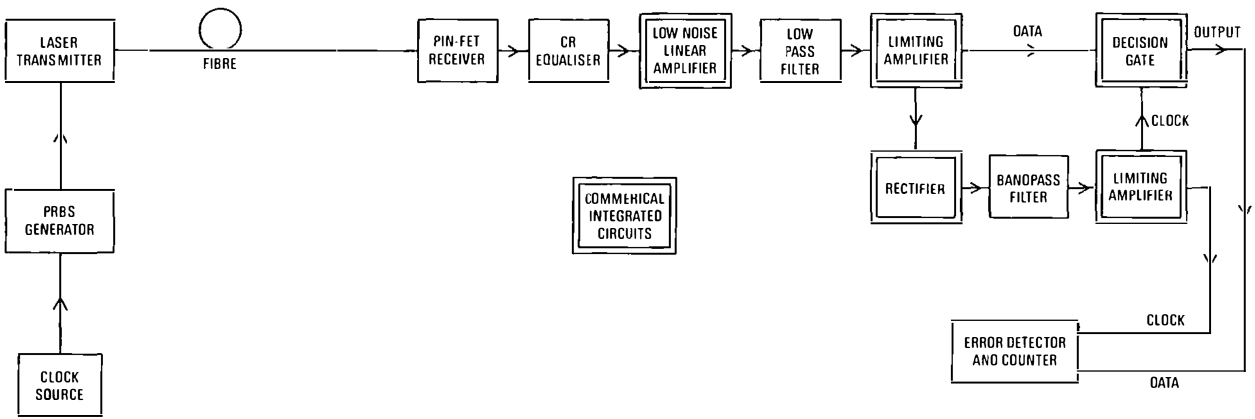


FIG. 10—Block diagram of fibre system under test, showing receiver regenerator configuration

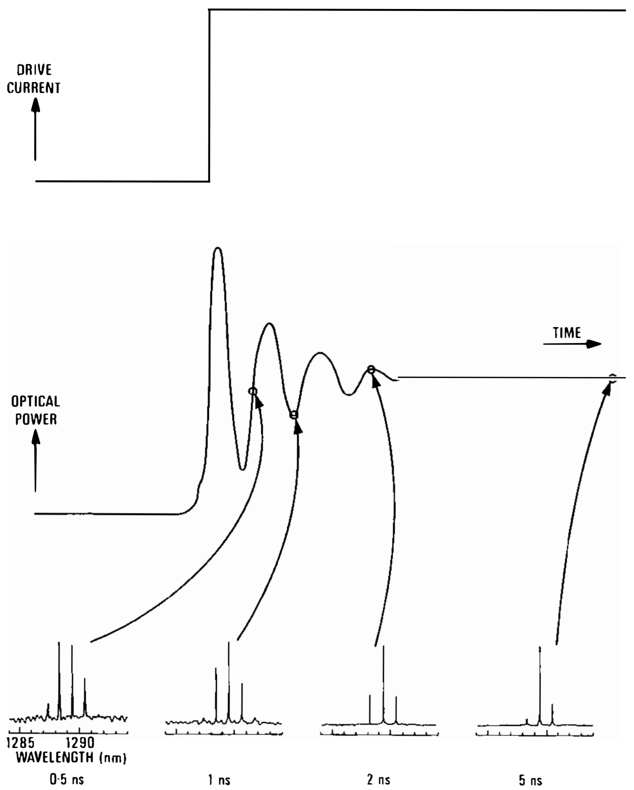


FIG. 11—Evolution of the mode spectrum of a nominal 1300 nm wavelength semiconductor laser

schematic form in Fig. 11. Note the transient ringing of the optical pulse, a distinctive feature of a pulsed semiconductor laser. A spectral line is referred to as a *longitudinal mode* and the presence of just one line referred to as a *single longitudinal mode* (SLM); the term *mode* here means cavity mode rather than propagating mode. The envelope of the comb of wavelengths bring about the dispersion penalty mentioned before, but additional effects were predicted even when an SLM was predominant.

During the life of a laser device, temperature and ageing effects may change the position of the SLM. It is possible that before this happens the device may hop between possible SLM positions. If this is coupled with the dispersion of the fibre, it is possible that with very long lengths, that is, many tens of kilometres, pulses may appear and disappear—a gross ISI effect. This effect leads to the error-rate performance becoming independent of input optical power above a certain threshold level and may result in the target system error rate being unachievable. These considerations led to the development of devices and techniques (external cavity control and distributed feedback (DFB) lasers) to produce more stable SLM operation under modulation for the 1550 nm wavelength region.

Laboratory-Based Cable Tests

Whilst the performance at 1550 nm clearly held out great potential, the development of a laser transmitter with the necessary spectral confinement and stability at 1550 nm was much further away than a suitable 1300 nm laser. The next stage of systems development was therefore centred on 1300 nm systems with the object of demonstrating a more realistic system using a laboratory-based cabled fibre link of about 30 km length. If a system at 140 Mbit/s could be operated over this length in the field without a repeater, then it would remove the need for dependent power-fed repeaters and metallic conductors in the cable for almost all routes in the national trunk network.

Towards the end of 1981, an experimental 31·6 km single-mode fibre link was set up in the laboratory in order to simulate as closely as possible an operational cable route[12]. In this experiment, 14 fibres, each 2·25 km long were cabled in three 2·25 km cables using a loose-tube technique[13]. All the fibres were fully characterised prior to splicing. The three cables were spliced together by using the fusion splicing technique without power monitoring to form a continuous fibre length of 31·6 km. Splice losses were determined by two methods: one was to use an optical time-domain reflectometer (OTDR)[14] to measure the individual splice losses, and the other was to measure the total link loss and then subtract the previously measured cabled-fibre losses to give a figure for the total splice loss. For this system there was good agreement between the two measurements, which indicated that the mean splice loss was between 0·18 and 0·23 dB at 1300 nm and approximately 0·12 dB at 1550 nm. The reduced loss at 1550 nm is due to the increasing field width with longer wavelengths; this helps to reduce the splice losses associated with concentricity error and core deformation. The total link loss for this experimental cable was 17·5 dB at 1300 nm, and permitted several system experiments to be performed in the 1300 nm window.

Transmitter and receiver modules at 1300 nm for operation over the cabled single-mode link were made available for a series of systems tests at 140, 280 and 565 Mbit/s. The transmitter module comprised a GaInAsP 10 μm oxide insulated stripe laser[10] with nominal operating wavelength of 1300 nm. The device was multi-longitudinal mode, with three main modes separated by 1 nm, to give an effective linewidth of about 2 nm. The optical power from the laser facet was launched into a single-mode fibre tail using simple butt coupling. The launched power was 80 μW (−11 dBm) mean when pulsed; no additional power could be coupled into the fibre by increasing the drive current since the main lobe of the laser output radiation saturated at this level and any additional power obtained was lost to uncoupled side lobes. The transmitter module was temperature controlled by a Peltier-effect thermoelectric cooler. This allowed stable transmitter operation to be achieved with these early 1300 nm devices. The threshold current at 16°C was 280 mA and a drive current of 50 mA was required for full modulation of the optical output power. It was envisaged that for most applications, reductions in threshold or bias current and improvements in module design might eventually enable the temperature controller to be dispensed with. Further development of the receiver, particularly with fibre coupling, had taken place. The PIN photodiode was now mounted on a quartz block with an alignment hole for the fibre inputs. The receiver sensitivity at 140 Mbit/s and 1300 nm was now −46 dBm. The principal results of the system tests are given in Table 3.

TABLE 3
Results of System Trials

Wavelength	1280 nm	1280 nm	1280 nm
Link length	31·6 km	31·6 km	31·6 km
Bit rate	140 Mbit/s	280 Mbit/s	565 Mbit/s
Total path loss†	20·5 dB	20·5 dB	20·5 dB
Transmitter power	−11 dBm	−11 dBm	−11 dBm
Received power	−31·5 dBm	−31·5 dBm	−31·5 dBm
Receiver sensitivity	−46 dBm	−41·5 dBm	−36·5 dBm
System margin	14·5 dB	10 dB	5 dB

† The increased loss over the 17·5 dB fibre and splice loss is mainly attributed to the two prototype lens connectors used at each end of the system.

The system margin of 14·5 dB at 140 Mbit/s was very encouraging and was judged sufficient to cover the additional performance degradations usually experienced in transferring laboratory-based systems demonstrations to the working environment. At 565 Mbit/s, the system margin reduces to 5 dB because of the reduction in receiver sensitivity with bit rate. However, it was believed that additional system margin would be obtained by improvements to laser performance and transmitter design. For example, the use of a fibre tail with a specially designed lens[9] fabricated on its end face, as shown in Fig. 12, was predicted to yield perhaps up to a factor of four improvement in coupling efficiency.

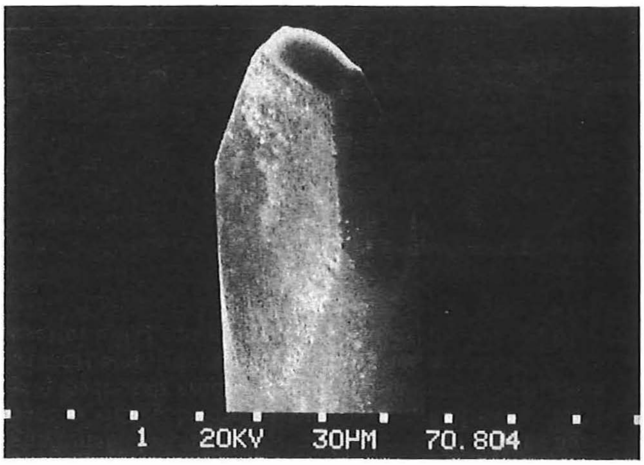


FIG. 12—Single-mode fibre with asymmetric hemispherical lens

The results signified that 140 Mbit/s systems without dependent repeaters were feasible and that such systems, once having been installed, could be upgraded to at least 565 Mbit/s at the same repeater spacing whenever necessary[16]. Indeed, it was becoming evident that repeater spacings greater than 30 km were eventually going to be viable, and this realisation stimulated still further the development of single-mode fibre systems for undersea applications. Some initial tests at 140 Mbit/s using broad linewidth lasers were also undertaken in the 1550 nm window confirming that this wavelength option can be exploited. It remained to show the complete viability of single-mode technology by demonstrating systems over cables installed in the field.

The MIME and Woodbridge System Trials

During the early part of 1982, two major field trials of single-mode fibre technology were carried out. The first of these was a co-operative venture between BTRL and Telephone Cables Limited and involved the installation of a 7.5 km cable link between Martlesham and the nearby town of Woodbridge[17]. The other trial was in association with STL and involved the installation of a 15 km cable route between Martlesham and the town of Ipswich, this latter link being known as the *Martlesham Ipswich Monomode Experiment* (MIME)[18]. A map of the two routes is shown in Fig. 13.

The fibre for the Woodbridge trial was produced on an experimental basis by BTRL and GEC Optical Fibres Ltd. (GECOF), and was not tightly toleranced. The fibre came from both BTRL and GECOF sources and derived from many different and varied experimental preforms. The fibre for the MIME trial was was tightly toleranced and was produced by STL/STC in a pilot manufacturing facility. As a consequence, the MIME fibre was more representative of

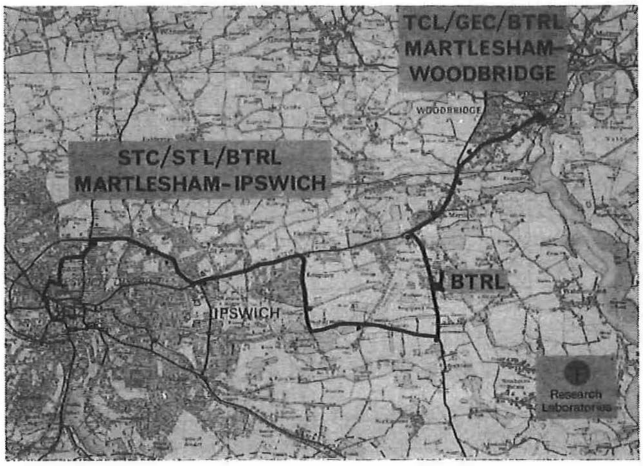


FIG. 13—MIME and Woodbridge single-mode fibre cable installations

the ultimate quality that could be expected from a dedicated production environment. The effect of fibre tolerances clearly manifested itself on the splicing results from the two links. Nevertheless, both links showed encouraging splicing performance.

The links were spliced by using the fusion-splicing technique without power monitoring. The Woodbridge link was spliced by BTRL, and the MIME link by STC field staff trained by STL. The splice-loss distributions from the two links are shown in Fig. 14. The mean splice loss of the Woodbridge link was 0·22 dB. The careful and extensive measurement programme associated with this link allowed calculations to be performed showing that this loss was made up of 0·08 dB due to core deformation, 0·07 dB due to concentricity error and a further 0·07 dB due to field width mismatch. The mean splice loss of the MIME link, however, was only 0·11 dB, indicating the possible level of splice loss that could be achieved with tightly toleranced fibre. Both of these cable installations were then used as test beds for many system experiments and, indeed, in 1985, they still form the major test beds for such experiments.

These cabled single-mode fibre installations gave the first opportunity to test, under field conditions, the operation of systems at 140 Mbit/s and above over repeater spacings of 30 km and greater. Laser transmitters with improved laser designs and lensed fibres were used in this series of experi-

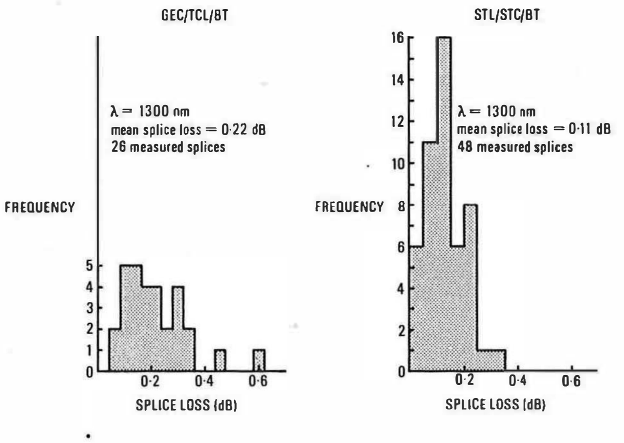


FIG. 14—Splice loss histograms for MIME and Woodbridge field installations

ments at 1300 nm. The transmitter and receiver were connected to the transmission fibres using shaped memory effect ferrule connectors[19]. The receiver and regenerator combination remained essentially the same as in previous experiments. The Woodbridge cable was operated at 140 Mbit/s initially and subsequently at 650 Mbit/s—the highest bit rate then available from test equipment. The MIME cable was operated over a range of bit rates with a final test at 565 Mbit/s over the full 61·3 km of the installation. The power budgets for the 1300 nm experiments are given in Table 4.

TABLE 4

Power Budgets for 1300 nm Experiments

Wavelength	1290 nm	1290 nm	1290 nm
Link length	31·5 km	31·5 km	61·3 km
Bit rate	140 Mbit/s	650 Mbit/s	565 Mbit/s
Path loss	21·9 dB	21·9 dB	30·5 dB
Transmitter power	−7 dBm	−8·7 dBm	−4·5 dBm
Received power	−28·9 dBm	−30·6 dBm	−35 dBm
Receiver sensitivity	−45 dBm	−34 dBm	−36 dBm
System margin	16·1 dB	3·4 dB	1 dB

The system margins achieved in this series of experiments were greater than in previous tests and were large enough to leave virtually no doubt about the eventual practicality of the 30 km spacing up to 650 Mbit/s. Fibre, cable and splicing data from these trials formed the basis for current BT single-mode fibre specifications. Research effort was then put into demonstrating the capabilities of systems operating at 1550 nm.

The 1550 nm System Tests

The initial attempts to stabilise laser devices for narrow line operation at 1550 nm utilised techniques involving external stabilising components. In the first of these experiments, injection locking[20], see Fig. 15, was used wherein a laser was operated continuous wave to give a stabilised SLM. The output from this laser (LD1) feeds a second laser (LD2) forcing it to oscillate exclusively at this same wavelength The second laser (LD2) was modulated in the conventional manner. The technique allowed the transmission of a 140 Mbit/s signal over a 102 km link of conventional fibre on reels without a repeater[20].

A second experiment using a transmitter module with an external cavity comprising a reflector placed at a controlled distance from the rear facet of the laser, see Fig. 16, allowed operation again at 140 Mbit/s and 102 km[21]. The SLM could be tuned in this case by altering the position of the reflector by applying heat via the heating coil to expand the metal tube on which the reflector is placed.

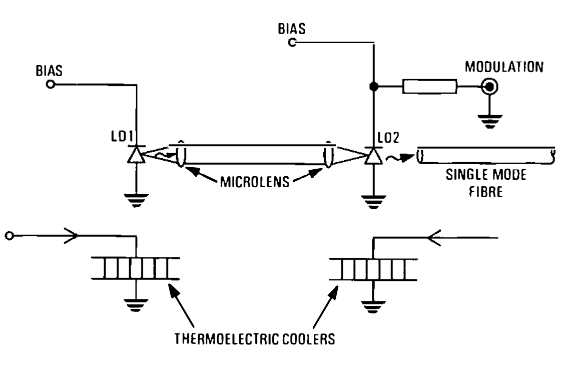


FIG. 15—Injection-locked laser transmitter arrangement

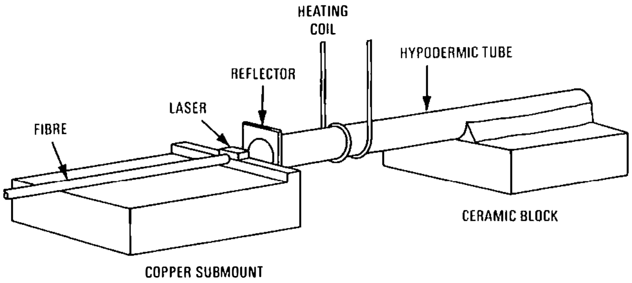


FIG. 16—External cavity laser transmitter arrangement

This latter transmitter module was employed in a further systems test in which all of the installed cable was linked together to form a 90·5 km length. Successful operation was achieved at 140 Mbit/s with a margin of about 2·5 dB. This feasibility demonstration, with a field-installed single-mode route, of a system operating over three times the then required unrepeatered span for the trunk network, set the scene for potentially very long unrepeatered systems at 1550 nm of great interest for submerged applications.

The power budgets for these systems tests are given in Table 5.

TABLE 5

1550 nm System Test Power Budgets

Wavelength	1520 nm	1520 nm	1520 nm
Link length	102 km†	102 km†	90·6 km‡
Bit rate	140 Mbit/s	140 Mbit/s	140 Mbit/s
Path loss	34 dB	36 dB	35·7 dB
Transmitter power	−8 dBm	−6 dBm	−7·8 dBm
Received power	−42 dBm	−42 dBm	−43·5 dBm
Receiver sensitivity	−45·7 dBm	−45·7 dBm	−46 dBm
Laser extinction	1 dB	0 dB	0 dB
System penalty*	1·6 dB	0·5 dB	0 dB
System margin	1·1 dB Note 1	3·2 dB Note 2	2·5 dB Note 2

† On reels
‡ MIME + Woodbridge installed fibre
* Miscellaneous Impairments
Note 1: Injection-locked transmitter
Note 2: External-cavity transmitter

CONCLUSION

This article has described the development of single-mode fibre systems at BTRL up to about 1982 and indicated the significant aspects of fibre design, cabling, splicing and systems design that evolved during this period. These developments have had a profound effect upon the ensuing applications of the technology in the trunk and junction networks, and on undersea routes. In the second part of this article, some of the developments that have occurred since the MIME field trials will be described as well as an indication of what the implications are for the future exploitation of the growing single-mode fibre network.

Acknowledgement

The authors wish to acknowledge the contributions made by their many colleagues throughout the optical project at BTRL and elsewhere; in particular that of Dr. C. J. Todd, Head of the Optical Communications Technology Division at BTRL for helpful discussions.

References

[1] KAO, K. C., and HOCKHAM, G. A. Dielectric Surface Waveguide for Optical Frequencies. *Proc. IEE*, 1966, **113**, p. 1151.

[2] BRACE, D. J., and RAVENSCROFT, I. A. Optical-Fibre Transmission Systems: The 8·844 Mbit/s Feasibility Trial. *Post Off. Electr. Engs. J.*, Oct. 1977, **70**, pp. 146–153.

[3] BERRY, R. W., and RAVENSCROFT, I. A. Optical-Fibre Transmission Systems: The 140 Mbit/s Feasibility Trial. *ibid.*, Jan. 1978, **70**, pp. 261–268.

[4] DYOTT, R. B. The Optical Fibre as a Transmission Line. *ibid.*, Oct. 1974, **67**, pp. 164–168.

[5] AINSLIE, B. J., DAY, C. R., FRANCE, P. W., BEALES, K. J., and NEWNES, G. R. Preparation of Long Lengths of Ultra-Low Loss Single-Mode Fibre. *Electron. Lett.*, 1979, **15**(14), pp. 411–413.

[6] AINSLIE, B. J., DAY, C. R., FRANCE, P. W., BEALES, K. J., and NEWNES, G. R. Single Mode Low Loss Silica Fibre Prepared in Long Lengths. Proceedings 5th ECOC, Amsterdam, 1979, pp. 3.6-1– 3.6-4.

[7] MILLAR C. A. Applications of ESI Profiling to a 31·6 km Monomode Fibre System. OFC 82, Phoenix, Paper thEE3.

[8] PAYNE, D. B., and MCCARTNEY, D. J. A comparative study of monomode splicing techniques. ICC 84, Amsterdam.

[9] SMITH, D. W., and MATTHEWS, M. R. Laser transmitter design for optical fibre systems. IEEE Journal on selected areas in communications, Apr. 1983, **SAC–1**(3), pp. 515–523.

[10] NEWMAN, D. H. Sources for Optical-Fibre Transmission Systems. *Post Off. Electr. Engs. J.*, Oct. 1974, **67**, pp. 208–212.

[11] SMITH, D. R., HOOPER, R. C., AHMAD, K., JENKINS, D., MABBITT, A. W., and NICKLIN, R. Pin-FET hybrid optical receiver for longer wavelength optical communications systems. *Electron. Lett.*, Jan 1980, **16**(2), pp. 69–71.

[12] PAYNE, D. B., and MCCARTNEY, D. J., and HEALEY, P. Fusion splicing of a 31.6 km monomode optical fibre system. *ibid.*, 1982, **18**(2), pp. 82–84.

[13] JACKSON, L. A., REEVE, M. H., and DUNN, A. G. Optical fibre packaging in loose fitting tubes of orientated polymer. *Optical and Quantum Electronics*, 1977, **9**, pp. 493–498.

[14] HEALEY, P. OTDR in monomode fibre at 1·3 μm using a semiconductor laser. *Electron. Lett.*, 1981, **17**(2), pp. 62–64.

[15] MACDONALD, B. M., and SAUNDERS, A. G. Coupling of an optical fibre to a substrate entry photodiode. *ibid.*, Oct. 1981, **17**(22), pp. 832–833.

[16] STANLEY, I. W., HOOPER, R. C., SMITH, D. W., and MATTHEWS, M. R. The performance of experimental monomode transmission links. Proceedings of the 8th European Conference on Optical Communications, ECOC, Cannes, France, p. 447

[17] STERN, J. R., *et al.* Field installation of a 31.5 km monomode optical fibre system operated at 140 Mbit/s and 650 Mbit/s. *Electron. Lett.*, 1982, **18**(14), pp. 631–632.

[18] STERN, J. R., *et al.* The characterisation of monomode fibre links installed in operational ducts. Technical Digest of the Symposium on Optical Fibre Measurements, Boulder USA, 1982.

[19] MALLINSON, S. R. A monomode connector using shape memory effect metal. Proceedings 9th European Conference on Optical Communications, Geneva, 1983.

[20] MALYON, D. J., and MCDONNA, A. P. 102 km unrepeatered monomode fibre system experiment at 140 Mbit/s with an injection locked 1·52 μm laser transmitter. *Electron. Lett.*, May 1982, **18**(11), pp. 445–446.

[21] CAMERON, K. H., CHIDGEY, P. J., and PRESTON, K. R. 102 km optical fibre transmission experiments at 1·52 μm using an external cavity controlled laser transmitter module. *ibid.*, July 1982, **15**, pp. 650–651.

Bibliography

MIDWINTER, J. E. Studies of monomode long wavelength fiber systems at the British Telecom Research Laboratories. *IEEE J. Quantum Electron.*, June 1981, **QE–17**(6), pp. 911–918.

AINSLIE, B. J. *et al.* Interplay of design parameters and fabrication conditions on the performance of monomode fibres made by MCVD. *ibid.*, **QE–17**(6), pp. 854–857.

WHITE, K. I., *et al.* Characterisation of single-mode optical fibres. *The Radio and Electron. Eng.*, July/Aug. 1981, **51**(7/8), pp. 385–391.

REEVE, M. H. Optical fibre cables. *ibid.*, pp. 327–332.

NEWMAN, D. H. *et al.* Sources for optical fibre communications. *Telecommun. J.* (Geneva), Nov. 1981, **48**, pp. 673–680.

MATTHEWS, M. R., and SMITH, D. R. Photodiodes for Optical Fibre Transmission Systems. *Post Off. Electr. Engs. J.*, Jan. 1974, **67**, pp. 213–217.

Semiconductor Laser Development and Production. *ibid.*, Apr. 1979, **72**, pp. 19–26.

HOOPER, R. C., MIDWINTER, J. E., SMITH, D. W., and STANLEY, I. W. Progress in Monomode Transmission Techniques in the United Kingdom. *Journal of Lightwave Technology LT1*, Dec. 1983, pp. 596–611.

Biographies

Raymond Hooper joined the Post Office as a Trainee Technician Apprentice in London North Area. After graduating with an honours degree in Electrical Engineering from the Middlesex Polytechnic in 1970, he joined the optical-fibre communications project at the Research Department. He bas been involved in all the major BTRL field demonstrations of optical-fibre transmission systems and was responsible for the field systems measurements with the installed single-mode fibre cables. He now heads a group working on advanced direct detection systems and receivers operating at gigabit/s information rates.

David Payne obtained a B.Sc. in Electrical Engineering from the University of Aston in Birmingham in 1971. He then joined the Post Office Research Department and worked on analogue and digital systems design with emphasis on interfacing problems. He was also involved in early work on microprocessor-controlled automatic test equipment. In 1979, he obtained an M.Sc. in Telecommunications Systems from the University of Essex. He then joined the optical-fibre communications project at BTRL and was responsible for single-mode fibre splicing and connectors. During this time, he was also involved in the establishment of the BT single-mode fibre specification. Since then, he has had responsibility for wavelength multiplexing components and is currently head of a group investigating the application of single-mode fibre technology for local networks.

Michael Reeve graduated from the University of Durham in 1973 with a degree in Applied Physics, and joined the Post Office Research Centre in the same year. He has worked on several aspects of optical-fibre communications, including modal properties, strength and cable design. Recently he has worked on the BT single-mode fibre specification and the design of a new type of optical cable—the blown-fibre cable. He is currently head of a group concerned with optical-cable and component design.

Time-Division Multiple-Access Systems for Satellite Communications

J. R. LEWIS, B.TECH., C.ENG., M.I.E.E, and D. T. ELLIOTT, B.SC, PH.D.†

UDC 621.391.63

International digital communications via satellite are shortly to enter a new era when the INTELSAT and EUTELSAT 120 Mbit/s time-division multiple-access systems, providing service to America, Europe and the Middle/Far East, are introduced in 1986. This article outlines the fundamental concept of TDMA and describes the system in some detail. It includes a description of the TDMA burst structure, the manner in which the system is synchronised and maintained, and reviews the network interface, and the earth-station requirements.

INTRODUCTION

Time-division multiple-access (TDMA) is a technique which permits many users to gain access to a single satellite repeater (transponder) simultaneously. Up to now, the technique most widely used for this purpose has been frequency-division multiple-access (FDMA), a technique that involves transmitting a number of continuous frequency-spaced carriers (usually frequency modulated) through a satellite transponder. Although this access method is entirely satisfactory, and permits many users to utilise the transponder, it does not make best use of the available satellite power (and hence capacity) because the transponders must be operated below their maximum, or saturated, output power owing to the multi-carrier arrangement.

With TDMA, however, the satellite transponder can be operated at, or very close to, saturation, with each user utilising all or part of the available transponder bandwidth and power for very short periods of time. As a consequence, improvements in capacity can be made. Moreover, because the transmissions occur in short bursts, this scheme is particularly suited to digital modulation and, hence, to the transmission of digital information. The purpose of this article is to introduce readers to the principles and operation of TDMA.

Although straightforward in principle, TDMA is a complex scheme to implement owing to the stringent demands placed on earth-station equipment by the high operating speeds, the burst-like nature of the system, and the complex protocols necessary to ensure correct system operation. For these reasons the application of TDMA has, to date, been limited. A high-speed TDMA system (operating with a transmission rate of 60 Mbit/s) was first proved experimentally in 1978/79 during field trials carried out by INTELSAT (the global International Telecommunications Satellite organisation) using an INTELSAT IV satellite; commercial systems (operating at various rates) have, to date, been limited to a handful of American domestic satellite systems, and a domestic Japanese system. However, TDMA will become much more widespread when INTELSAT introduces a 120 Mbit/s TDMA system into its global network in 1986 (via INTELSAT V satellites) and, at about the same time, EUTELSAT (the European Telecommunications Satellite operating agency) commences operation with its TDMA system (also 120 Mbit/s) on EUTELSAT 1.

During the system definition phase for these systems, both INTELSAT and EUTELSAT recognised that there were, in general, many ways in which a TDMA system can be designed to meet a particular set of requirements. As a consequence of this, both organisations set up, in 1979, international working groups charged with the responsibility of writing the overall system specification for their respective systems. The information on which these working groups based their deliberations came from a variety of sources, including the TDMA field trial carried out by INTELSAT, the European Orbital Test Satellite (OTS) test programme and the results of a large amount of in-house study by various administrations.

At a fairly early stage in the evolution of the two specifications[1, 2], it was recognised by the European telecommunication administrations that there would be significant equipment cost and operational advantages if the INTELSAT and EUTELSAT specifications could be essentially the same. Harmonisation of the two specifications therefore became a primary aim of the EUTELSAT group and, mainly as a result of a strong European contribution to the INTELSAT work, eventually a reality. The two specifications, which were finally completed around mid-1981, are identical wherever possible. The only differences that exist result from unavoidable differences of network topology and circuit signalling.

The aim of this article is to introduce the principles of TDMA by reference to the detailed characteristics of the EUTELSAT/INTELSAT 120 Mbit/s TDMA systems, and as such it represents a medium-depth description of these systems. Where reference is made in the article to 'the 120 Mbit/s systems', it is the common aspects of the EUTELSAT and INTELSAT system specifications that are being referenced.

The article starts with a general outline of the principles of TDMA operation, and then goes on to cover burst structure, synchronisation, housekeeping, terrestrial network interfacing, and earth-station requirements. The final section on implementation looks in a little more detail at the reasons for adopting TDMA, and discusses the time-scales currently envisaged for the introduction of these systems.

GENERAL OUTLINE OF TDMA OPERATION

Fundamentals

The fundamental concept of TDMA is quite straightforward in that a number of users access the same transmission channel in turn, each transmitting for a short period of the total time. Fig. 1 illustrates the signals at the input to the

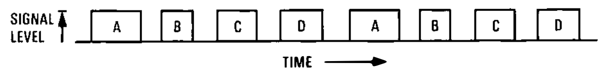


FIG. 1—Input to satellite amplifier assembled from four users

† Satellite Systems Division, British Telecom International

satellite amplifier with four users in a system. It can be seen that transmissions arrive in a pre-assigned sequence, the only difference between them being the length of the individual transmissions. It should be noted that the horizontal axis in this figure represents time and not frequency.

The frequency characteristics of these transmissions will depend on the modulation method used, but it could well be that each burst will occupy the whole of the satellite transponder bandwidth for the short time that it is transmitted. However, as long as each station maintains the instants of its transmissions so that its bursts do not overlap those from other stations at the satellite, then each station can use the satellite independently of the other. This multiple use (access) of the satellite transponder on a time-division basis is called *TDMA*.

An essential feature of TDMA is that information is transmitted in bursts, rather than continuously, and the way in which this is achieved in the earth-station equipment (TDMA terminal) is illustrated in Fig. 2. In this figure, it

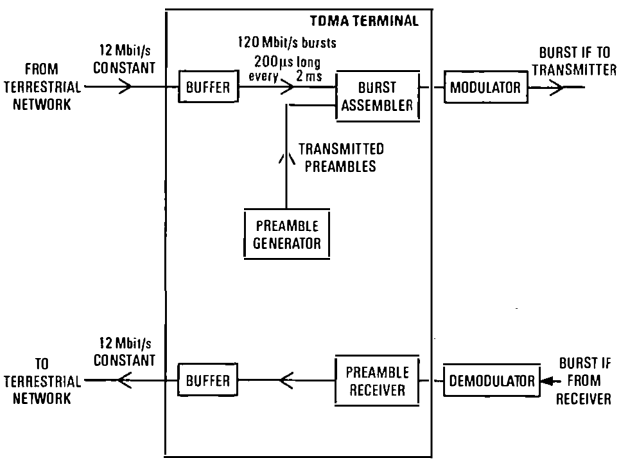


FIG. 2—Fundementals of TDMA terminal operation

is assumed that the interface with the terrestrial network is digital; any analogue-to-digital conversion that may have been necessary is assumed to have taken place prior to the TDMA terminal. Further, an arbitrary 12 Mbit/s bit rate has been chosen so that there is no indication as to whether the signal represents a number of multiplexed voice signals, a television signal, a data stream or whatever. This is to stress that the fundamentals of TDMA operation do not depend in any way on the intelligence represented by the bits that are carried.

From Fig. 2 it can be seen that the input stream is first fed into a buffer store at a continuous rate, for example, 12 Mbit/s, then read out of the buffer at a much higher rate; for example, 120 Mbit/s. This is the rate at which transmissions to the satellite take place. Since the output rate is greater than the input rate, it cannot be continuous, but is in bursts; in this example, one burst is read out and transmitted every 2 ms. Thus, in order to transmit all the bits being read into the buffer, bursts must be 200 μs in length; that is, 200 μs long bursts at 120 Mbit/s occurring once every 2 ms result in an effective throughput bit rate of 12 Mbit/s.

Before being transmitted to the satellite, each of these bursts has added to it a few more bits called the *preamble*. The preamble enables communicating TDMA terminals to receive the bursts correctly and its function is therefore supervisory rather than traffic carrying. The preamble is generated within the TDMA terminal, as indicated in Fig. 2, and is combined with each burst of traffic bits in such a way that the whole package is fed to the modulator as a single burst. The intermediate frequency (IF) carrier output from the modulator is correspondingly in bursts, and is passed to the frequency converter and transmitter for transmission to the satellite in the appropriate super high frequency (SHF) band. The reverse procedure takes place on the receive side, with the received preamble being removed and processed before the traffic bits are passed to the buffer. Transmission between the earth and the satellite in burst mode is an essential characteristic of TDMA.

TDMA Frame

The obvious question arising from the above description is how is it possible to ensure that bursts do not overlap? The answer lies in the provision of a synchronisation system based on the use of *reference bursts*. These bursts are transmitted by one or more reference stations, which may be either specially provided earth stations, or normal earth stations with extra facilities. The reference bursts are received by all participating stations, each of which ensures that its own subsequent bursts arrive at the satellite at a fixed pre-determined time relative to each reference burst. This procedure is discussed in some depth later.

The reference bursts need only be very short, about the same number of bits as the preamble already mentioned, and are transmitted at the same rate as each participating station transmits its traffic bursts (every 2 ms in the example). The total ensemble of signals arriving at the satellite repeater therefore comprises a reference burst, followed by traffic bursts from each participating station in a pre-determined order. This signal ensemble is commonly referred to as the *TDMA frame*, and the rate at which it repeats (2 ms in the 120 Mbit/s system) as the *frame length*. Factors influencing the choice of frame length include frame efficiency and acceptable information delay. The overall frame structure is illustrated in Fig. 3, which shows traffic

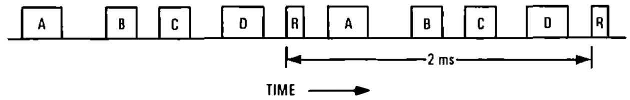


FIG. 3—TDMA frame structure

bursts from stations A, B, C and D plus a reference burst R. In reality, there are likely to be more than just four participating stations, and it is probable that many stations will transmit more than one burst per frame. In addition, in the 120 Mbit/s system, there are two almost identical reference bursts per frame for system security purposes.

There is one final point worth making in this outline of TDMA operation. It was stated that by means of the synchronisation system each station transmits its bursts relative to the reference burst, resulting in one burst being transmitted every 2 ms (for example). Now it is clear that 2 ms will be perceived as a slightly different period by every participant, depending on the accuracy of the clock used to measure it. However, as a result of the synchronisation technique used, the 2 ms which define the TDMA frame length is that determined by the reference station. Therefore, because the number of bits in each burst is fixed, the overall rate at which every participant transmits over the TDMA system is in fact controlled by the reference station. This important feature of TDMA has consequences for the interface with a terrestrial network, as discussed later.

BURST STRUCTURE

This section looks in some detail at the structure of the traffic bursts and the reference burst. In general, the structure of the bursts is dependent on the particular TDMA system in which they are used, and so this section is heavily inclined towards the 120 Mbit/s system. However, where features are common to any TDMA system this is emphasised. In addition, the last subsection looks briefly at features of other systems which are not included in the 120 Mbit/s system.

In order to describe burst structure, it is convenient to introduce the commonly used unit of measurement, the *symbol*. A symbol is a number of bits transmitted together, the number depending on the type of modulation used. The 120 Mbit/s system uses four-phase phase-shift keying in which each symbol represents 2 bits. 120 Mbit/s is therefore 60 Msymbol/s, making each symbol about 16·5 ns. This modulation technique is discussed in more detail later. The two parallel bit streams (each of 60 Mbit/s) fed to and from the modems in this arrangement are termed the *P* and *Q* streams.

The following is a description of the burst structure used in the 120 Mbit/s system. Each burst comprises a preamble followed by up to eight traffic carrying sub-bursts.

Preamble

The purpose of the preamble is to enable TDMA terminals to receive bursts correctly. The structure of the preamble in the 120 Mbit/s system is shown in Fig. 4. It comprises a 176 symbol carrier and bit timing recovery (CBTR) sequence, a 24 symbol unique word, eight symbols that are used for teletype order wires, an 8 symbol service channel, and two 32 symbol voice order wires. This gives an overall length of 280 symbols, which is about 4·6 μs or about 0·2% of a frame.

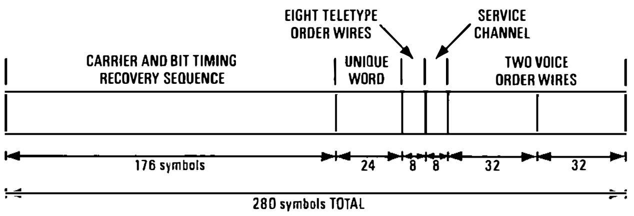


FIG. 4—Preamble structure for the 120 Mbit/s TDMA system

Carrier and Bit Timing Recovery Sequence

The purpose of the CBTR sequence is, as the name implies, to enable each demodulator to lock on to the precise frequency and bit rate of each received burst. This is necessary because any demodulator has to receive bursts from a number of different transmitting stations, and because small variations of frequency and bit rate from one burst to another always exist.

Accurate frequency tracking of each individual burst is necessary to maximise the signal-to-noise ratio of the received IF signal, which subsequently minimises the error rate in the demodulated signal. Accurate clock extraction is necessary both to maximise the probability of correctly interpreting the value of each received bit and to provide a clock signal for initial processing of the received signal.

The nature of the CBTR will depend on the type of modulation used, but in the 120 Mbit/s system it comprises 48 symbols of unmodulated carrier, followed by 128 symbols of carrier phase reversals. This is produced by the following modulating sequence:

P bit stream	1	1	1	..	1	0	1	0	1	..	0	1
Q bit stream	1	1	1	..	1	0	1	0	1	..	0	1
Symbol number	1	2	3	..	48	49	50	51	52	..	175	176

Unique Word

The unique word is a fixed sequence of bits that enables receiving stations to determine the exact location of information within a burst. It is necessary because, as a result of timing tolerances within the system, a receiving station cannot be certain of the precise location in the frame of any particular burst. By receiving the unique word (using a correlation detector), the presence of the burst is confirmed and a precise timing reference is established, which enables correct decoding of the information carried in the burst. The length and pattern of the unique word are chosen so that the probability of it being replicated by other information or noise sequences is suitably low.

In the 120 Mbit/s system, four different unique words are used, but each is derived from the following 24 symbol sequence:

P bit stream	0 1 1 1 1 0 0 0 1 0 0 1	0 1 1 1 1 0 0 0 1 0 0 1
Q bit stream	0 1 1 1 1 0 0 0 1 0 0 1	0 1 1 1 1 0 0 0 1 0 0 1
Symbol number	1 12	13 24

This particular sequence is known as ***unique word 0*** (UW0) and is used for most bursts. It can be seen that UW0 is made up of two identical consecutive 12 symbol sequences, with the P and Q bit streams in each sequence also being the same. Other unique words are derived from this by complementing (that is, inverting) one or both of the second 12 bit sequences of UW0. Thus, unique word 1 (UW1) has the second 12 bits of the P stream inverted, unique word 2 (UW2) has the second 12 bits of the Q stream inverted, and unique word 3 (UW3) has the second 12 bits of both P and Q inverted. The application of these other unique words is discussed below. The advantage of using this complementing method to differentiate between different unique words is that one correlation detector can be used to detect all of them, since a complete negative correlation is just as valid as a complete positive correlation in terms of detecting the position of the unique word. The reason for defining the unique words as two consecutive sequences derives from the need for them to perform a dual function. As well as burst position information, they provide a means of resolving the phase ambiguity that exists at the demodulator as a result of the modulation scheme used. The first 12 symbol sequence is used for this purpose, and is described in more detail later.

Service Channel

The service channel is used for sending system control messages to other stations in the system and, as such, is peculiar to the 120 Mbit/s system. The service channel itself is eight symbols in length, but the messages conveyed on it are all 32 bits (that is, 16 symbols) in length, which means that each message has to be transmitted over a number of frames. In addition, it is important to ensure that some of the messages are received without error and so a very secure transmission arrangement has been adopted. This involves transmitting only two bits (one symbol) of the message in each frame, but repeating those two bits eight times (hence the 8 symbol length of the service channel). Majority-decision logic is then used at the receiver. It therefore takes 16 frames to transmit the entire 32 bit message and this 16-frame period is known as a *multiframe*. So that the service message can be received correctly, a means of identifying the start of the multiframe must be provided; this is done by using one of the special unique words described previously. Thus, in the first frame of each multiframe (that is, every 16 frames), all traffic stations transmit UW3 instead of UW0, and this is referred to as the *multiframe marker*. Details of the different messages carried in the service channel are given later under 'Housekeeping'.

Order Wires

Order wires, also known as *engineering service circuits*, are provided on many satellite systems to enable easy and reliable contact between operators at different earth stations, and have been provided on the 120 Mbit/s system so that they provide a similar capacity on each burst to that available on each frequency-modulated (FM) carrier in the INTELSAT FDMA system. There are eight teletype order wires, carried in a single 8 symbol channel, and two voice order wires. Again, further details of these channels are given later.

Sub-Bursts

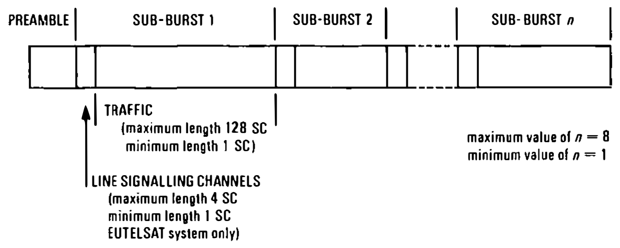
Sub-bursts are the traffic carrying elements of the bursts in the 120 Mbit/s system, and subdivision of the traffic in this way is another particular feature of this system rather than an essential characteristic of TDMA. There are two different types of sub-bursts, defined to match the types of traffic that are carried. Digital speech interpolation (DSI) sub-bursts carry mainly telephony traffic where most, though not necessarily all, of the channels have been passed through a DSI process. This process, which in general terms allows more than one input channel to be carried on one satellite channel, is described in more detail in the section on DSI. The first satellite channel in each DSI sub-burst is known as the *assignment channel*, which is necessary for DSI operation, and this is followed by up to 127 satellite channels carrying traffic. Each satellite channel is 64 symbols (128 bits) in length, which, with a repetition rate (that is, frame length) of 2 ms, provides an effective rate of 64 kbit/s over each satellite channel. A maximum sub-burst length of 128 satellite channels was selected on utilisation considerations, and the practical minimum number of channels, although theoretically one is about 30 because of DSI characteristics. Both of these points are discussed further in the section on DSI.

Digital non-interpolated (DNI) sub-bursts are used to carry traffic which cannot for various reasons be subjected to interpolation. They can again be anywhere between one and 128 satellite channels long, although there is obviously no need for the first channel to be an assignment channel.

A further feature of sub-bursts (both DSI and DNI) is the line-signalling channel (LSC). This is used on the EUTELSAT system to carry the out-of-band line signals associated with the digital R2 signalling system, but it has not been defined for the INTELSAT system because R2 signalling is not used on any circuits carried on this system. The LSC consists of between one and four satellite channels, depending on sub-burst length, located at the beginning of the sub-burst. This again is discussed in more detail later. Overall, a burst in the 120 Mbit/s system comprises a preamble, followed by between one and eight sub-bursts, the maximum number of eight being based on traffic planning considerations. There is no system constraint on the mix of DSI and DNI sub-bursts that can be used, and the actual number of sub-bursts in the burst will, in general, be determined by traffic levels. Overall burst structure is shown in Fig. 5.

A maximum-size burst comprising a preamble plus eight 128 channel sub-bursts, each with a four satellite channel LSC, will be 67 864 symbols long, which is about 1·1 ms or 57% of a frame.

Finally, it should be noted that the INTELSAT system permits the use of forward error correction (FEC) on selected sub-bursts. This provides a means by which system performance can be improved, but with a corresponding reduction in the effective information throughput of the system. The FEC overhead is approximately 14% per sub-burst on which it is used.



SC: Satellite channel (each satellite channel is 64 symbols long)

FIG. 5—Burst structure for the 120 Mbit/s TDMA system

Reference Bursts

The most essential function of the reference burst in any TDMA system is to provide a timing reference for system synchronisation, and to meet this function it need comprise only a suitable CBTR sequence and a unique word. A basic reference burst of this type conveys no information other than the definition of an instant in time. In most TDMA systems, however, there is a need to convey additional information in the reference burst and so additional channels to meet particular system needs are added to the two basic sequences. This is the case in the 120 Mbit/s system, where order wires, a service channel and a control channel are provided, in a similar manner to the normal traffic stations burst preamble. In fact, the reference burst is identical in structure to the traffic-burst preamble, but with an additional 8 symbol control and delay channel (CDC) added at the end. The CDC is used to send synchronisation control messages and the synchronisation delay value to each participating station. Details of these messages are given in the section on synchronisation. Each message is 32 bits long and is transmitted over 16 frames with eightfold repetition per frame, exactly the same as the service message format described previously.

The reference station uses the same multiframe for both service-channel and CDC messages, and, like normal traffic stations, a special unique word is used to define the start of this multiframe. It has already been noted that there are two reference bursts per frame (transmitted from two separate reference stations for security purposes), and one of them uses UW1 to indicate the start of the multiframe, while the other uses UW2. Both use UW0, the same as the traffic stations, for other frames. This arrangement provides a simple means by which receiving stations can distinguish between traffic bursts and each of the two reference bursts, as well as identifying the start of multiframe.

Other Burst Features

The paragraphs above have described the features of the bursts on the 120 Mbit/s system, some of which are essential to any TDMA system, and some which are optional. There are, however, other elements that could feature in a TDMA system, and probably the most significant of these is a *demand assignment request* channel. This is associated with the way the *burst time plan* is organised. Briefly, the burst time plan describes the arrangement of bursts in the frame, and has to be centrally controlled and notified to each station to ensure correct transmission. This plan can be of two basic types. In the 120 Mbit/s system, a fixed, or pre-assigned plan is used which remains in use for a long period, typically 12 months. This implies a fixed capacity at each station over the period of the plan. However, in some circumstances, where capacity requirements at any station vary considerably over short periods, a dynamic plan is used, with frame capacity being allocated by a central station in response to requests from participating stations. This demand-assigned arrangement requires a channel by which capacity can be requested, plus an equivalent channel from the controlling station (which will probably be a reference station) with which to respond to such requests.

Overall Frame Structure

Overall, the frame comprises one or two reference bursts plus a large number of traffic bursts with spaces between them. The size of these spaces will, in general, reflect how close to maximum capacity the system is operating, with a situation where all bursts are virtually consecutive representing a saturated system. However, in some cases, spaces between bursts are unavoidable because of the constraints of burst planning. This is particularly true where transponder hopping is used (as described later), and the ratio of utilised frame time to total frame time is often referred to as *frame fill efficiency*. Regardless of how good the burst planning is, there is one factor which always leads to some loss of efficiency; this is the need for *guard times* between bursts. These empty spaces are a necessary feature of the burst time plan because of the inevitable inaccuracy of burst position that results from small uncertainties in the synchronisation system. The length of guard times is dependent mainly on the synchronisation scheme used, but also to some extent on the origin of the two bursts being separated. For example, in the case of one station transmitting two consecutive bursts, it is possible that no guard time need be allocated because of the total certainty at the transmitting station of where the first burst ends and the second starts. A further possible feature of the overall frame structure is acquisition windows, which are special empty spaces used by all stations for acquiring initial synchronisation; these are described more fully in the following sections on acquisition and synchronisation.

SYNCHRONISATION

General

A fundamental requirement of a TDMA system is that the individual bursts are maintained in their assigned frame position so that they do not corrupt one another. This is achieved by ensuring that the bursts are transmitted from each earth station at a precise moment so that they arrive at the satellite sequentially and at exactly the right instant. The sequence, which is repeated once every TDMA frame, is arranged to maximise the use of the available satellite capacity, and is referred to as the *burst time plan*. Each station has its own receive and transmit time plans, which are sub-sets of the burst time plan, and these are stored in a memory within the TDMA terminal.

The start of each TDMA frame is defined by the transmission of a reference burst from the reference station according to timing information derived from a stable timing source. The reception of the unique word contained in this burst by each station in the system defines a specific instant, known in the 120 Mbit/s system as the *start of receive frame* (SORF), and this provides a timing reference for the receive time plan. The timing reference for the transmit time plan is known in the 120 Mbit/s system as the *start of transmit frame* (SOTF), and is derived from the SORF by applying a suitable delay referred to as D_n. The value of D_n is unique to each station in the system and depends upon the distance from the satellite to the station (the satellite range).

Nearly all modern day communications satellites are maintained in geostationary orbit; that is, an orbit in which the satellite moves at the same angular velocity as the Earth. To an observer on the Earth the satellite should, therefore, appear to be stationary. However, because of irregular gravitational effects and the small forces exerted by the moon and the sun, the satellite tends to drift from its nominal position, both in latitude and longitude. Small thrusters on board the satellite are used to conteract this drift and maintain its position within acceptable limits. Viewed from the Earth, this movement appears as near-sinusoidal variations in the antenna pointing angles and the satellite range. The period of the sinusoid is approximately 24 hours and the magnitude is usually at a minimum midway between position-correction manoeuvres, which occur at approximately monthly intervals. This variation in the satellite range is of direct concern to TDMA operation because it results in direct variations in the value of D_n. In the INTELSAT system, satellite movement is generally restrained to within $\pm 0 \cdot 1°$ in both North/South and East/West directions, resulting in a maximum variation in D_n of about 1 ms.

It is the function of the synchronisation system to constantly refresh the value of D_n at each station, and the more frequently this is done the more accurate the synchronisation. The result of more accurate synchronisation is that frame guard times can be reduced, and this improves frame-fill efficiency. There are several ways in which the value of D_n can be obtained during steady-state TDMA operation; they fall into two broad categories known as *closed-loop* and *open-loop synchronisation.*

Closed-Loop Synchronisation

Satellites with beam coverage patterns arranged so that the stations accessing them are capable of receiving their own transmissions are ideally suited to closed-loop synchronisation. By observing the error in the frame location of one of its bursts (that is, relative to the reference burst), each station determines its own value of D_n so that the burst assumes its correct position in subsequent frames. Thus, a closed control loop as shown in Fig. 6 is established. Because

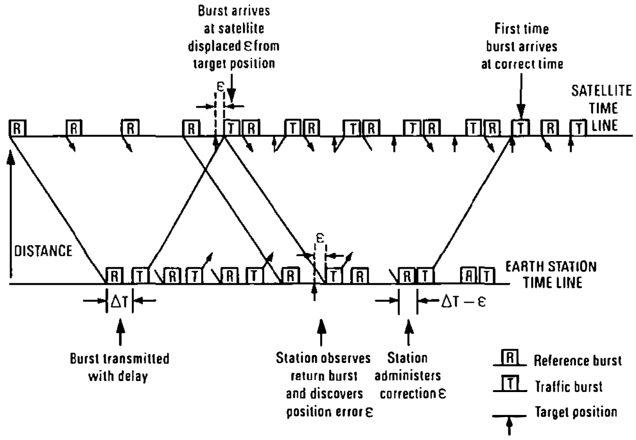


FIG. 6—Direct loop-back correction of traffic burst position

of the long propagation delay between the earth station and the satellite, it is important that the error is not determined until the previous correction has had time to have an effect.

Feedback closed-loop synchronisation techniques extend the closed-loop principle to include a second co-operating station. This is necessary when operation is taking place in a spot-beam environment where, in general, stations cannot receive their own transmissions. The co-operating station, which is most likely to be the reference station, determines the burst-position error and relays the necessary correction back to the synchronising station.

Open-Loop Sychronisation

Open-loop sychronisation is based upon a calculated value for the delay, D_n. From an accurate knowledge of the earth station location and satellite orbit, the range, and consequently the value ofD_n, can be readily calculated. Whilst the earth station location remains constant, the satellite position varies considerably as discussed above. There are various methods by which the satellite location can be determined; these fall into two basic categories: direct measurement and prediction.

Direct Measurement

(*a*) *Triangulation* This technique relies upon three simultaneous satellite range measurements from three geographically-remote earth stations. From a knowledge of their location, the real-time satellite location can be determined, the accuracy being dependent upon the accuracy of the measured range, the earth station locations, and the degree of their relative separation. Ranging measurements can be made very accurately, typically to within about 10 m. Earth station locations can be determined to within a similar accuracy.

Processing of the measurements can be carried out at a central location, probably the reference station, and the information broadcast to each synchronising station in the system.

For operational use, this method of synchronisation demands considerable system discipline since it relies upon real-time measurement and processing.

(*b*) *Antenna Pointing Angles* From a knowledge of the antenna pointing angles and the satellite range at one earth station, the satellite location can be readily determined. This technique is considerably simpler than the triangulation method because it only involves one station, generally the reference station. However, it is inherently less accurate due to errors incurred in the measurement of the antenna pointing angles.

Prediction

There are a number of very sophisticated computer programs capable of predicting the various orbital perturbations that cause the variations in the satellite position. By using a number of accurate tracking measurements performed routinely during the life of a satellite these programs are capable of predicting the satellite's trajectory to within a few metres over a period of many weeks. However, accurate prediction is not possible during an orbital manoeuvre because the thrust from the satellite's thrusters can only be predicted to within about 10%. Consequently, although very accurate for most of the time, prediction techniques alone are not secure enough for the operational environment.

Initial Acquisition

When a closed-loop synchronisation scheme is used, it is necessary to employ an initial acquisition scheme to determine an initial value ofD_n, and consequently an initial burst position error, so that the loop can gain lock. This value ofD_n does not have to be as accurate as that during the synchronisation phase; it is merely required to place a short synchronising burst within a relatively large acquisition aperture within the TDMA frame. The short burst need only consist of a CBTR sequence and unique word, and the aperture could be either a special acquisition window in the frame or one of the apertures into which the acquiring terminal would normally transmit a traffic burst. Open-loop acquisition, based upon the techniques of open-loop synchronisation, is perhaps the most effective technique for this application. Because the accuracy demanded during acquisition is lower than during normal synchronised operation, much greater measurement errors can be tolerated. Obviously, the size of the vacant acquisition aperture will determine the actual accuracy required for the particular application.

Another closed-loop initial acquisition method available, is based on the transmission of very low power *acquisition* signal into the TDMA satellite transponder. The signal is transmitted at such a level that it does not cause significant interference to other TDMA users, but is sufficient to enable its reception with a correlation detector, and hence the determination of an initial value forD_n.

The 120 Mbit/s Systems

The choice of synchronisation system for the 120 Mbit/s TDMA system had to satisfy a number of clear requirements to meet the needs of a reliable operational system. These requirements can be summarised as follows:

(*a*) it must be capable of operating in a spot beam environment (for example, INTELSAT V);

(*b*) traffic terminals must be as simple as possible, therefore much of the necessary processing must be concentrated in the reference terminals; and

(*c*) the reference terminals must be highly reliable since they determine the overall system reliability.

After a number of different possible system combinations had been explored, it was decided to adopt a feedback closed-loop synchronisation system using open-loop acquisition. For satellite position determination, INTELSAT will use a combination of triangulation and prediction, whilst EUTELSAT intends to use the reference station's antenna pointing angles.

To provide the highly reliable reference station facility, two reference stations access each transponder, one transmitting primary reference bursts and the other secondary reference bursts. Both primary and secondary bursts contain nominally the same control and synchronisation information, the secondary being used only in the event of a failure in the primary. Thus system disruption will occur only in the unlikely event of simultaneous failure at both reference stations.

Acquisition and Synchronisation

To keep the traffic terminal as simple as possible, all processing of acquisition and synchronisation information is performed at the reference stations. The traffic terminal merely obeys the *control* code instructions that it receives via the CDC, which forms part of the reference burst. Each station is given a new control code and associated delay (D_n) once every control frame, which is approximately 1 s in duration. Up to 32 stations can be addressed from one reference burst, although a reference station can transmit a different CDC in the reference bursts of each transponder. The traffic terminal implements the given value ofD_n and transmits according to the received control code. Four control codes are available, having the following meanings:

(*a*) *Initial Acquisition Phase 1* (*IAP1*) This code is transmitted prior to a traffic station attempting initial acquisition. It instructs the terminal to transmit only its *short burst*, which is the preamble of its principal burst. The principal burst is used by the reference station for synchronisation. The reference station determines an initial value for the delay based upon open-loop acquisition calculations that will place the short burst in the middle of the assigned frame allocation of the principal burst.

(*b*) *Initial Acquisition Phase 2* (*IAP2*) Upon reception of the short burst, the reference station changes the *IAP1* code to *IAP2*, instructing the acquiring terminal that it can maintain transmission and that it is under the control of the reference station. This phase continues until the reference station has placed the short burst into the correct frame position (that is, in the position of the preamble of the principal burst), according to the time plan, by adjusting the value of D_n.

(*c*) *Synchronisation Phase* (*SYNC*) When the short burst is at its assigned frame position, the control code *IAP2* is replaced by the *SYNC* code. This instructs the terminal that it is now SYNCHRONISED and that it can commence transmitting all of its bursts. This code represents the steady-state synchronisation phase.

(*d*) *Do Not Transmit* (*DNTX*) This code is transmitted by the reference station if it does not wish the terminal to transmit any bursts. It is only used in extreme circumstances when a major problem has made the continuation of normal system operation impossible.

By means of the above process, the reference station can control the acquisition and synchronisation of all traffic stations in the system.

SYSTEM HOUSEKEEPING

Most transmission systems employ housekeeping protocols of one form or another to ensure correct and orderly operation, and, with complex systems such as TDMA, these can be a very significant part of the system. Different TDMA systems have slightly different requirements and it is impossible to cover all conceivable facilities here. However, the procedures provided on the 120 Mbit/s systems, which are described in some detail, are typical of those to be found in use elsewhere.

Most of the facilities provided on the 120 Mbit/s systems fall into one of three broad categories: those which assist smooth interworking of a particular terminal both with the rest of the system and with the terrestrial network; those which protect the system from malfunction of one terminal; and those which enable degradation of system transmission quality to be identified and rectified. In addition to the procedures in each of these categories, the EUTELSAT system is provided with certain operational facilities to help with initial system set-up and terminal testing.

Burst Time Plan Management

Burst time plan management comprises a set of protocols in the interworking category and is arguably the most important of all the housekeeping procedures. The burst time plan (BTP) is an overall system plan which specifies the timing and duration of every burst and the traffic carried on them. Each terminal has to be provided with its own BTP, a subset of the overall system plan, and this forms the basis of the program which is loaded into the terminal to control its operation. This terminal BTP is referred to as its *master time plan* (MTP). The MTP contains all the details necessary for a terminal to participate in the system, including satellite details, frequency and polarisation assignments, terminal identification information, burst position and duration information, order-wire allocations, sub-burst types, details of circuits in each sub-burst, signalling-channel information and alarm identification numbers.

A sub-set of the MTP is the *condensed time plan* (CTP). This includes all the elements of the MTP which a terminal must get right in order that it will not disrupt system operation; for example, burst timing and duration. This is in contrast to the other elements of the MTP where an error will result perhaps in a particular circuit not being established. Because of the importance of a terminal being loaded with error-free CTP information, procedures have been defined to enable CTP data to be transmitted to a terminal either on order wires or via the public switched telephone network. At the receiving terminal, this information is combined with essential local equipment configuration information, loaded into the terminal, read out of the terminal and retransmitted to the originating authority (INTELSAT/ EUTELSAT) for checking before it can be used. The data is transmitted in blocks, each with a checksum, to assist identification of errors, and CCITT† X25 protocols are used for link control transmission error detection/correction. This verified data is stored in background memory in the terminal until it is required.

Burst Time Plan

When it is either desirable or necessary to implement a new BTP, the traffic terminal is instructed to transfer control from the current BTP to the BTP stored (and previously verified) in the background memory. In the 120 Mbit/s TDMA systems, specific procedures have been included to ensure that all stations change their time plan at exactly the same instant. This is referred to as a *synchronous BTP change* and permits a time plan change to take place without traffic being disrupted.

In both the INTELSAT and EUTELSAT systems, a BTP change is initiated by remote control from the operations centre. This operations centre controls, co-ordinates and maintains the TDMA system, and utilises data, voice and telegraph links with the reference stations for this purpose. Once the BTP change has been initiated, however, it progresses under the control of the principal reference station. This controlling reference station informs firstly other reference stations that it is to proceed with a BTP change, and then all of the traffic stations under its control by using the service channel. The message used for this purpose also contains details of the new BTP number to be implemented.

On receipt of this message, the traffic stations signal to the reference stations (also via the service channel) that they are READY TO CHANGE. Provided all stations indicate that they are ready to change, then the reference stations begin a countdown sequence, enabling stations to synchronise the instant at which the change takes place. The time taken from initiation of a BTP change by the control centre to traffic stations implementing their receive and transmit time plans is less than 20 s. If, however, at any stage in this process acknowledgements are not received within a given period of time, the BTP change is terminated. A re-try is not permitted for a period of approximately 1 minute.

The 120 Mbit/s systems also permit certain subsets of the facilities outlined above to be used. For example, in the case where just the BTP number at a terminal requires changing (all other data remaining unchanged) it is possible for the reference station to transmit what is referred to as an *abridged condensed time plan* (ACTP) which causes the existing time plan to be repeated, but with the new plan number. Thus, the entire contents of the CTP are read back, checked and retransmitted to the reference station for verification. This facility is therefore useful in permitting the controlling authority (that is, INTELSAT or EUTELSAT) to send short messages, but in return to receive the complete CTP, perhaps for verification. Traffic stations may also make use of the countdown sequence procedure provided by the reference station to assist with traffic reconfigurations. Although not crucial to the system's operation, this technique does offer the possibility of achieving rearrangement of terrestrial/satellite circuit mappings synchronously with the remote end, thus providing a no-break change-over capability.

Monitoring and Alarms

In any TDMA system, it is important for housekeeping purposes that transmissions from stations are monitored to ensure that transmissions from one station do not interfere with those of another. It is particularly important in a high-speed TDMA system where transmissions are separated from each other only by very short periods of time (typically

† CCITT—International Telegraph and Telephone Consultative Committee

fractions of a microsecond). The 120 Mbit/s systems have been designed with monitoring facilities both at the reference station and the traffic stations.

At the reference station, all of the traffic and reference bursts received by that station are monitored. In the event that one or more of these bursts is not in its allocated position (for more than 32 multiframes; that is, 1·024 s) and could therefore be interfering with other carriers (and in the worst case a reference station), the reference station sends a *selective do not transmit* (*SDNTX*) message to the offending station via the service channel. On receipt of a *SDNTX* message, the station must either switch to redundant equipment or cease transmissions of that burst. If the problem persists, the reference station transmits a second *SDNTX* to the station, in which case certain pre-defined sets of its transmissions must cease. The problem is brought to the attention of station operators by means of a prompt maintenance alarm.

System performance between terminals is also monitored by traffic stations, and this information is passed between terminals by means of the service channel. These messages, or alarms, relate to inter-station performance and are referred to as the *unique word loss alarm* and the *high BER alarm*. They are instigated as follows.

A unique word loss alarm is generated when the unique word in a particular burst is lost for more than 0·5 s. This is important since none of the information can be correctly decoded without knowledge of the unique word. A message is sent to the appropriate station, via the service channel, every second until the unique word is re-established. At the offending earth station, a prompt maintenance alarm is generated, and a signal passed to the terrestrial network to indicate that the satellite transmission system is temporarily inoperative. Likewise, at the receiving station, a signal is forwarded to indicate that a fault has occurred downstream.

A high BER alarm is generated when the receiving station determines (from the unique word) that a BER of $10^{-3}$is exceeded. This BER is significant when dealing with pulse-code modulation (PCM) encoded speech and is used as a threshold in the terrestrial transmission network. As a consequence, it is also used for monitoring purposes on the satellite link. The alarm generated is sent every 4 s until the BER performance improves. In this case, local alarms are generated whilst information is passed to the remote end by means of specified bits in the frame structure of the hierarchical rate employed; for example, CCITT Recommendation G732.

After any loss of incoming signal or cessation of transmission, for example following receipt of *SDNTX*, an *alarm indication signal* (AIS) is passed to the terrestrial network in accordance with CCITT maintenance philosophy. This is carried by an *all ones* signal in the data stream (in place of the normal signal), and indicates that a problem has been identified. Subsequent maintenance alarms are inhibited by means of an *upstream failure indication* (UFI) signal. At the remote end, the loss of the incoming signal is similarly detected and signalled to the terrestrial network. It should be stressed that the generation, detection and extension of signals both in the terrestrial and satellite parts of the link are an important consideration in any transmission system, and that the INTELSAT and EUTELSAT TDMA systems have been designed to comply with CCITT Recommendations regarding maintenance philosophy.

In addition to the housekeeping facilities described above, reference stations are also equipped with TDMA system monitor (TSM) equipment. This monitors general system performance and can be used to assist in the setting up of TDMA terminals and the diagnosis of system faults. It should be noted, however, that this monitor is not part of the reference terminal equipment (RTE), which is concerned specifically with correct system operation. The TSM can monitor relative power levels, the frequency and position of a burst, the transponder operating point, and provide an estimate of the error rate.

SERVICE CHANNEL MESSAGES

Many of the control and alarm messages described above are sent via the service channel. The complete set of messages transmitted in this way is

(*a*) the value of delay, D_n, being used,
(*b*) the BTP identification numbers,
(*c*) the codes for the synchronisation of BTP changes,
(*d*) the unique word and BER alarms, and
(*e*) in the case of reference stations, the *SDNTX* message.

When no service messages are being sent, an ineffective message is transmitted to avoid the possibility of spurious messages being incorrectly detected.

As described previously, the service channel information is transmitted by using the eight symbols (16 bits) set aside for this purpose in the preamble of both the traffic and reference stations. The messages themselves occupy 32 bits and are, therefore, carried not in one but over a number of frames. In order to ensure a very high confidence in the service messages being detected correctly, only one symbol (2 bits) is transmitted in any one frame. This is transmitted with eightfold redundancy. At the receiver, majority logic is used to determine the original symbol transmitted. A complete message is thus transmitted over 16 frames, or 1 multiframe (32 ms).

The service channel message format is illustrated in Fig. 7 and can be seen to be in three parts: a function code, a data

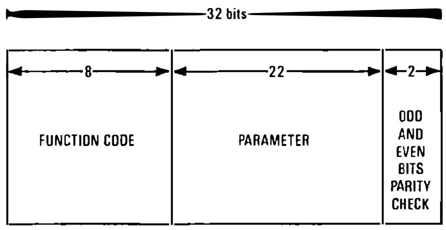


FIG. 7—Service channel format

block and some parity bits. This format permits the various messages described to be transmitted. Owing to the relatively slow transmission of any single message (32 ms) and the possible need to transmit several messages at about the same time, it is necessary to establish a transmission priority rule. The highest priority is given to messages concerning correct operation of the system, that is, the value of transmission delay used, whilst those relating to a BTP change and the alarms are given increasingly lower priority. Fig. 8 illustrates the service channel messages sent and the transmission priority used.

Engineering Service Circuit Facilities

In addition to the service-channel facilities described above, the 120 Mbit/s TDMA systems also provide a means for earth stations to communicate with each other, with reference stations and with the operations centres by means of ESCs. Facilities are provided for both voice and telegraph service circuits, which can be used for co-ordination and maintenance of the TDMA network and for the exchange of BTP data.

The 120 Mbit/s systems provide for two independent voice order wires (VOW) and up to eight independent telegraph (TTY) order wires on each transmitted traffic and reference burst. In total 16 VOWs and 32 TTY transmissions can be in progress at any one time. These circuits are used primarily by terminal operators to co-ordinate commissioning, operation and fault isolation of the equipment.

Type of Message	Contents of the Parameter Block (22 bits)	Transmission Priority
Transmission of delay used (all terminals)	Value of D_n used	First
Confirm 'ready for change' (selected terminals)	New burst time plan number	Second
Unique-word-loss alarm (from traffic terminals to traffic terminals)	Destination terminal number and burst number	Third
High BER alarm (from traffic terminals to traffic terminals)	Destination terminal number and burst number	Fourth
Ineffective message	Not used	Fifth
Request 'ready for change' (reference station to selected traffic terminals)	Destination terminal number and new burst plan number	(Note 1)
Notification of time plan change (reference station to selected traffic station)	Countdown sequence	(Note 1)
SDNTX (reference station to selected traffic station)	Destination terminal number and burst number	(Note 1)

Note 1: Not applicable for traffic terminal

FIG. 8—Service channel messages

Voice information is encoded using 32 kbit/s delta modulation and transmitted using the 32 symbols in the preamble sequence (32 symbols at a repetition rate of 2 ms corresponds to a bit rate of 16 ksymbols/s, or 32 kbit/s). Calls are routed automatically by means of switching computers at each traffic station. These operate in a similar manner to a telephone exchange seizing free outgoing channels when requested and signalling to the distant end to seize a free channel in the reverse direction. This operation is also monitored by all other switching computers. At the end of a call, a *clear down* signal is transmitted, making the channels available for new calls. Call routeing is thus carried out using standard order wire signalling procedures. The eight telegraph order wires operate in a similar fashion to that described above, but are transmitted by using the eight symbols in the preamble sequence. Each TTY (50 bit/s) is oversampled at 1 kHz (500 symbols/s) and transmitted using one of the eight possible symbols every frame. Facilities are also provided for establishing engineering service circuits (both VOW and TTY) between stations which are not able to receive each others' bursts directly.

Summarising, TDMA systems, are in general, complex systems requiring a high degree of housekeeping to ensure correct system operation. In the 120 Mbit/s systems this is achieved by means of secure BTP management procedures, in-service monitoring by traffic and reference stations, with engineering service circuits provided to assist in the diagnosis and correction of faults.

TERRESTRIAL NETWORK INTERFACE

This section considers the interface between the TDMA equipment located at an earth station and the terrestrial network that feeds traffic to it. In this context, the TDMA equipment is that which carries out the functions indicated earlier in this article, an important point to establish because, in practice, the interface equipment is often considered (and procured) as part of the TDMA terminal.

The functions carried out at the interface vary widely from one TDMA system to another. In the simplest case, the interface might merely accept traffic from a terrestrial digital transmission system, probably remove line coding, and feed the output of complete first-, second- or third-order multiplexers to the TDMA equipment. Alternatively, if an analogue terrestrial system is used, the interface will perform analogue/digital conversion. At the opposite extreme, the interface might incorporate, in effect, a small exchange, accepting digital or analogue traffic from a number of services and performing switching/selection at individual channel level in order to generate a bit sequence of the format required for the particular TDMA system.

It is also worth pointing out that the terrestrial networks can be widely varying in scale. In the case of a TDMA system designed for business users, the earth station might be located at the user premises with the terrestrial network being a cable from the terminal to the office upstairs. On the other hand, for a system such as the 120 Mbit/s system, the earth station is remotely located and the terrestrial network comprises a very long transmission system back to the international switching centre.

Regardless of the characteristics of any particular interface, it is possible to identify three general facets which will always have to be provided for. These are the traffic interface (that is, that which handles the bits that represents intelligence passing between two users of the system), the signalling interface (which handles bits representing call and system control information) and the timing interface. The following sections consider these three aspects separately, again giving details appropriate to the 120 Mbit/s systems. Before going into these details, however, it will be helpful to present some general information about the terrestrial network arrangements for the two 120 Mbit/s systems.

120 Mbit/s TDMA equipment will be located at BTI earth stations at Goonhilly in Cornwall and at Madley in Herefordshire. Traffic will be routed to them from Keybridge and Mondial international switching centres in London by using digital transmission systems operating at both 120 Mbit/s and 140 Mbit/s. The TDMA equipment is designed with inputs at 2048 kbit/s in the standard CCITT primary multiplex format and demultiplexing from 140 Mbit/s and 120 Mbit/s to 2 Mbit/s is implemented at the earth stations' repeater station.

Traffic Interface

The 120 Mbit/s system was designed primarily for carrying telephony traffic derived from public switched telephony networks, and to maximise system capacity for this type of traffic, DSI was incorporated into the terrestrial interface. However, the need to carry small proportions of non-telephony traffic was also recognised and so facilities for bypassing the DSI process for some channels are provided. In practical terms, the interface part of the TDMA equipment is modular, each module being commonly referred to as a *terrestrial interface module* (TIM). In theory TIMs may be of two types: either DSI, where interpolation can be applied to some or all of the input channels, or DNI, where interpolation is not provided at all. In practice, for flexibility reasons, BTI is procuring only DSI TIMs since these can be used in any of three ways; that is, all channels DSI, all channels DNI, or a mixture of DSI and DNI in any proportion.

The traffic processed by each TIM is formed into a sub-burst, of which each TIM transmits only one per frame. However, this may well carry traffic for more than one destination and so, on the receive side, a TIM must be able to handle more than one sub-burst, the maximum being eight. A sub-burst can be up to 128 satellite channels long and will usually comprise a pool of DSI channels followed by the DNI channels. The maximum number of input channels varies from 240 (that is, eight primary multiplexes) for a completely DSI TIM to 128 for a completely DNI TIM. For mixed DSI/DNI TIMs, the maximum number will depend on the ratio of DSI to DNI channels.

One important feature of the TIM concept is that there is no correspondence between sub-burst and primary multiplex format, meaning that, in general, the TDMA system does not offer transparency above 64 kbit/s. However, the need for such transparency for small proportions of traffic has been recognised and can be provided as indicated below.

Digital Speech Interpolation

It is appropriate at this point to say a few words about DSI. Firstly, it should be stressed that DSI is not an essential part of TDMA, but was included as a feature of the 120 Mbit/s system in order to ensure that high capacity was achieved from the system. DSI is one of a number of circuit multiplication techniques and it utilises the fact that speech is only present on each half of a telephone conversation for around 40% of the time, taking into account natural pauses in speech for breaths, thinking, etc. as well as the time when the other party is speaking. This *activity* as it is technically known, is monitored by a voice activity detector (VAD) in the DSI equipment which examines each 8 bit PCM-encoded sample of every input channel. If the VAD detects a signal above a threshold value on any input channel for a number of samples, it interprets that channel as being ACTIVE, if not, the channel is considered INACTIVE. The ACTIVE/INACTIVE condition of each input channel is passed to an assignment control processor, which arranges for active channels to be assigned to satellite channels. Satellite channels are not disconnected from input channels when the input channel becomes inactive, but only when that satellite channel is required for another active input channel. This minimises the amount of disconnection that occurs, and means that during non-busy periods, there may not in fact be much interpolation occurring. However, even during busy periods, it is important from a circuit quality point of view to avoid disconnecting an input circuit for very short speech pauses. For this reason there is a hangover time built into the VAD during which it continues to indicate an ACTIVE condition for an input circuit, even though the signal level has reduced below the activity threshold.

As well as making input channel to satellite channel connections, the assignment control processor has to inform the remote TDMA terminal of these connections, and to perform this function a DSI assignment channel is provided at the beginning of each DSI sub-burst.

Each input channel is given a 3-digit decimal (8-digit binary) number, as is each satellite channel, and an *assignment message* comprises a pair of these numbers. In all, three assignment messages are carried in the assignment channel each frame; this means that input channel to satellite channel associations can be updated at the rate of three per 2 ms.

To ensure that these assignment messages are correctly received, powerful forward error correction is used on the assignment channel. The code used is a rate 23/12 Golay code, which is capable of correcting up to three errors in each 12 bit block. It is the use of this code which results in the maximum limit of three assignment messages per frame. The format of the assignment channel is shown in Fig. 9 where it is also shown that there are 36 unuseable or dummy bits in each 128 bit assignment channel.

It is evident from the foregoing description that the assignment process, that is, activity detection, channel allocation and assignment message transmission, must take a finite time to implement and, therefore, in order to prevent severe front-end clipping of each new spurt of activity, each input channel is delayed in the DSI unit. This delay is adequate to allow for conjestion of assignment messages on the assignment channel during busy periods, and it also allows each assignment message to be transmitted one frame earlier than the first sample of the newly active channel, so giving time for the distant terminal to establish correct connections. This is particularly important where, as in many sub-bursts on the 120 Mbit/s system, the DSI sub-burst carries traffic for many destinations. Any one destination is able, by receiving the assignment channel, to determine which satellite channels in the following frame carry input channels that it must receive. In this way, each destination need only process a fraction of the satellite channels each frame, resulting in a simplification of the equipment.

Despite the implementation of delay in the DSI unit, there are times when the number of active input channels exceeds the number of satellite channels available to carry them, with the result that the fronts of speech spurts become clipped. This situation is known as *freeze-out*. The degree of freeze-out that occurs on any DSI system is dependent on both the overall DSI gain (that is, the ratio of input channels to satellite channels) and the VAD characteristics. The 120 Mbit/s system has been designed such that clipping of more than 50 ms occurs on less than 2% of voice spurts.

In order to soften the onset of freeze-out, the 120 Mbit/s system has an additional overload feature. When overload occurs (that is, the number of active input channels exceeds the number of satellite channels), the system generates extra, or overload satellite channels by reducing from the normal 8 bits-per-sample satellite channel capability to 7 bits-per-sample. The spare bits are assembled into channels in a predetermined manner, resulting in up to 16 extra channels being generated (dependent on sub-burst size). This in effect decreases the DSI gain for a short time, so reducing the chance of freeze-out. During overload, the least significant bit of input channel samples is omitted, with the result that there is a minimal effect on transmission quality. Receiving TDMA terminals are informed of the use of overload by the occurrence of satellite channel numbers in the assignment messages which are outside the normal range.

Non-Interpolating Interface

DNI channels are incorporated into the TDMA system to accommodate any services where interpolation cannot be permitted. In large quantities, they can be processed by dedicated DNI TIMs, but as it is expected that the quantities will be small, a DNI capability has been incorporated into the DSI TIM. In this case, the channels bypass the interpolation process and are grouped together at the end of the sub-burst generated by the TIM. This means that DNI channels have a permanent input-channel-to-satellite-

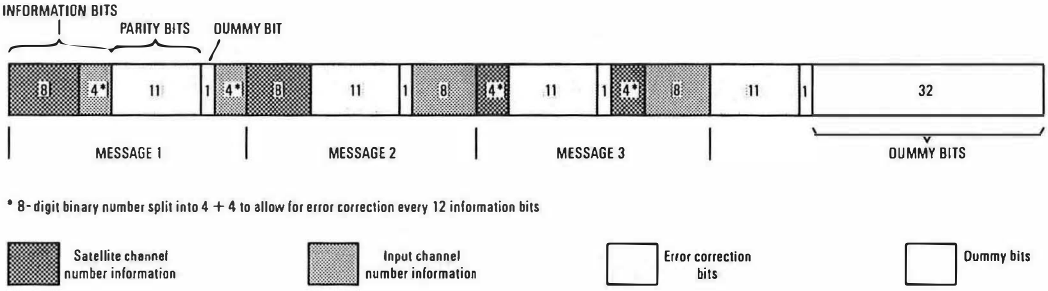


FIG. 9—Format of the DSI assignment channel on the 120 Mbit/s system

channel allocation, unlike DSI where the allocation is constantly changing depending on the availability of satellite channels in the DSI pool. An important aspect of DNI is that it is not dynamic; each input channel to the TIM is designated as DSI or DNI and remains that way until the TIM is reconfigured. This means that DNI or DSI selection on a circuit by circuit basis must take place at the switching centre in London.

Another important special feature of the DNI TIM is that it may be configured to provide transparency at higher bit rates. The TDMA system specification permits up to 8·192 Mbit/s to be handled in this way, this being the maximum rate that can be carried on a 128 satellite channel sub-burst. However, this rate is of little use to BT, being below the 8·448 Mbit/s second-order multiplex level, and so BT's TDMA equipment will have a maximum 2·048 Mbit/s transparency capability. This is implemented by transferring a complete primary multiplex onto 32 satellite channels within the sub-burst.

Signalling Interface

Signalling in the 120 Mbit/s system is handled in each TIM alongside the traffic to which it relates and so the signalling interface is conceptually rather than physically separate to the traffic interface.

An important aspect of the signalling interface is that it is one of the few areas where the INTELSAT and EUTELSAT systems differ, a situation brought about because of the different signalling systems used for European and intercontinental traffic.

At the start of the INTELSAT system, most circuits will use the CCITT No. 5 signalling system. Since this is an in-band system, it does not place any special requirements on the TDMA system, other than the need to ensure that the DSI VAD is designed to avoid clipping of the signals. When signalling systems CCITT No. 6 and CCITT No. 7 come into use, there will again be no problem because of the availability of DNI channels to carry the common signalling channel, if required.

In Europe, the situation is different because the majority of traffic uses the R2 regional signalling system in the digital line signals version. Again there is no difficulty with the in-band inter-register signals, but special arrangements have to be made to carry the line signals. Over the terrestrial link, these are carried as the 'a' and 'b' bits of time-slot (TS) 16, but to avoid carrying unused bits, the TDMA system does not accommodate TS16 and TS0.

Instead, a special R2 line signalling channel (LSC) was defined. This comprises up to four satellite channels located at the front of each sub-burst. Each of these satellite channels can carry the line signalling for up to 60 input channels, using the format shown below.

P Bit stream	0	1	Y_1	Y_3	a_1	a_2	..	a_n	..	a_{59}	a_{60}
Q Bit stream	1	0	Y_2	Y_4	b_1	b_2	..	b_n	..	b_{59}	b_{60}
Symbol number	1	2	3	4	5	6				63	64

Note: Symbols 1 and 2 are not used and carry the fixed seqence shown
Y_1, Y_2, Y_3 and Y_4 are used for alarm messages associated with LSC failure
a_n and b_n are the line signalling bits relating to input channel number *n*

This represents the first of the four channels, carrying signals for input channels numbered 1 to 60. These channel numbers are known as *international channel numbers*, and are a TDMA system number for each of the input channels, as distinct from *terrestrial channel number*, which is the designation allocated to each channel by an administration. The other feature of each LSC channel is the four bits used to carry alarm messages. These alarm messages are sent between terminals when difficulties occur with LSC transmission, and consequently enable alarm signals to be extended back to the main international switching centres.

Timing Interface

The concept of a timing interface arises because of the inevitable difference in frequency between the clock which controls transmission over the TDMA system and that which controls the flow of traffic to and from the terminal over a terrestrial system.

It has already been pointed out that the flow of information over the TDMA system between one TDMA terminal and another is controlled by the reference station. To illustrate this, consider a station transmitting a 6000 bit burst on a system working at a nominal frame rate of 2 ms. This gives a nominal bit rate over the TDMA system of 3·0 Mbit/s. However, if the reference station is transmitting reference bursts every 2·001 ms for instance, the station's effective transmission rate drops to 2·9985 Mbit/s.

This situation assumes great importance if there is a digital terrestrial link between the earth station and the international switching centre. The clock used on this link will probably be derived from a master network clock and hence will be totally independent of the TDMA reference station clock. In the above example, traffic using this network clock would be fed to the earth station at 3 Mbit/s, resulting in a slip of information at the point where the two clocks interface.

This slip is an unavoidable feature of any interface between two digital bit streams driven from independent clocks, since it is impossible to ensure that they operate at identical frequencies. An interface between two bit streams operating at nominally the same rate is called a *plesiochronous interface*. By international agreement (CCITT Recommendation G811), the master clocks of synchronous telecommunication networks[3] are operated with a frequency stability of 1 part in 10^{11} and the TDMA reference station clocks will be provided to this standard. This causes a maximum offset at the TDMA interface of 2 in 10^{11}, which translates as approximately one PCM frame (that is, 125 μs or 256 bits) every 70 days, or one PCM multiframe (2 ms, or 4096 bits) every 3½ years.

The rate at which slips will occur in practice is determined when the TDMA equipment is designed. Factors affecting the rate are: the effect on traffic (that is, is it better to suffer small disturbances often or long disturbances less frequently); and storage requirements (the longer the period between slips the greater must be the interface capability to store surplus bits). Slips of whole frames or multiframes will generally be selected to minimise PCM re-synchronisation problems, and one advantage of the multiframe slip is that it happens so rarely that there will be opportunities to reset the traffic at outages due to other causes, for example, planned maintenance or sun transit outages, so effectively eliminating plesiochronous slip as a degradation mechanism.

The other aspect of the timing interface is the effect of satellite movement. Although communication satellites are geosynchronous, slight changes in the satellite position cause a periodic change of range from any earth location, resulting in a change of perceived received frequency at the earth station caused by the Doppler effect. This mechanism adds to the clock frequency effect, but it is cyclic and, over a day, integrates to zero. It does not therefore result in clock slip, but does require more storage to be provided at the clock interface. In the 120 Mbit/s system, a storage capability of 2·2 ms is provided to accommodate this effect.

EARTH STATION REQUIREMENTS

The previous sections of this article have described in some depth the system characteristics of the INTELSAT and EUTELSAT 120 Mbit/s TDMA systems. This section discusses briefly those elements of the earth station equipment which are related specifically to the use of TDMA; namely, the burst-mode modem (including a description of the digital modulation technique) and the earth station equalisation requirements (that is, amplitude, group delay and path length). A general description of the earth-station characteristics, for example, antenna size and frequency of operation is also given.

Burst-Mode Modem

One of the most critical elements in any TDMA system is the burst-mode modem. The modem (modulator/demodulator) accepts digital data and control signals and outputs an intermediate frequency (IF) carrier (in bursts) which is modulated in sympathy with this information, and vice versa.

The modulation scheme used in the 120 Mbit/s systems is quadrature phase shift keying (QPSK), a commonly used digital modulation method. In this scheme, data bits are taken two at a time and mapped onto one of four permissible carrier phase states. These phases are spaced at 90° intervals and permit one pair of bits (or 1 symbol) to be represented by a carrier with an essentially constant amplitude and a clearly defined phase at the sampling instant. Information is thus transmitted as a stream of symbols, the transmitted symbol rate being half the transmission bit rate; that is, approximately 60 Msymbol/s.

A QPSK modulator operating with a square wave input produces a frequency spectrum which extends to infinity. Clearly, in an environment of close channel spacing (such as there is in any practical satellite system) the energy must be limited to the boundaries of the satellite channel in order to minimise the interference to users in adjacent transponders, and this gives rise to the need for channel shaping filters. In the INTELSAT and EUTELSAT systems, the satellite channels are nominally 72 MHz wide and are spaced 80 MHz and 83·3 MHz apart respectively.

The main factors which influence the choice of these filters are tolerable interference, both to and from adjacent transponders, and intersymbol interference (ISI). The former arises from unavoidable spectral overlap due to the modulation method used, and the effect of passing the signal through a non-linear amplifier (that is, one which contains a non-linear characteristic). The latter refers to unwanted distortion of the current symbol arising from the pulse responses of past and future symbols, and is due to the memory associated with the satellite channel and the channel shaping filters. This ISI can be almost totally eliminated in a linear channel by selecting filters which satisfy the Nyquist criteria. These criteria determine that filters with a certain symmetry, for example, raised cosine filtering (RCO), should be used and that to maximise the received power the receive filter should match the incoming spectral shape. In a non-linear channel, however, considerable amounts of ISI can result and, as a consequence, system design becomes a very complex process.

Computer simulations and field trials carried out during the TDMA system study phase demonstrated that the optimum arrangement for the 120 Mbit/s TDM systems was 40% roll-off raised cosine filtering, split equally between the modulator and demodulator. Futhermore, if a square wave signal is applied to the input of the modulator, it is necessary to compensate for its inherent frequency characteristics by including $x/\sin x$ compensation. This ensures that the spectrum at the input to the satellite amplifier over the bandwidth of interest is essentially that of the chosen channel shaping filter. Bursts of carrier, modulated in sympathy with the incoming data stream and, having the desired frequency response, are thus generated and, following frequency translation and amplification, transmitted to the satellite.

The burst-mode demodulator must accept these bursts of incoming carrier and make a decision as to the most likely original sequence transmitted, all in the presence of additive Gaussian noise and all during a very short space of time, typically several microseconds. To do this, the demodulator must first derive a reference carrier (carrier recovery), secondly, establish the phase orientation relative to the transmitted signal (carrier phase ambiguity resolution) and, finally, derive timing from the incoming signal (clock recovery).

Carrier recovery involves the demodulator generating an accurate phase reference which is synchronised to the incoming stream such that changes in the phase of this signal can be assessed. The reference carrier is derived by multiplying the incoming phase-modulated IF carrier by four (to remove the phase modulation) and then dividing by four to obtain a carrier of the correct frequency. Having done this, it is necessary to establish the phase relationship between the demodulated reference carrier and that originally transmitted. In the 120 Mbit/s system, this is achieved by using the unique word transmitted in the preamble sequence. It will be recalled that this is made up of a 12 bit sequence, which is present in both the P and Q channels. These unique words will always appear orthogonal to each other (irrespective of the recovered carrier phase sense) and, consequently, it is possible to establish the correct phase orientation of the demodulated reference carrier, and to invert as necessary the P and Q streams for the burst

An alternative carrier phase resolution technique frequently used in TDMA systems employing QPSK is to use differential encoding whereby the data is encoded so that only changes in the carrier phase are significant (as opposed to absolute phase). This removes the need for a reference phase, although it does have a slight penalty in terms of performance (0·4 dB) since an error in one symbol will affect the next symbol as well.

In addition to recovering the carrier, it is necessary to derive clock timing from the incoming signal so that the information can be sampled accurately. This enables small variations in timing resulting from satellite motion (Doppler) and transmit-clock inaccuracies to be tracked. In practice, clock recovery is normally carried out either by detecting the zero crossings of the baseband information or by passing the IF signal through a square-law device and using envelope detection. It can be appreciated, however, that an important requirement of any TDMA system is that both carrier and clock recovery should be carried out as quickly as possible. In the 120 Mbit/s system, these recovery procedures are carried out in less than 176 symbols, which is the period allocated for carrier and bit timing recovery purposes.

After the demodulation process, the received signal is sampled at the symbol clock rate midway through the duration of the symbol, and a decision made as to the actual phase transmitted. In phase-shift keying, an error will occur when the amplitude and phase characteristics of the noise or interference cause the received signal to be shifted into a region associated with a different symbol. In the case of ideal QPSK, the required carrier-to-noise ratio (C/N) to yield one errored bit in a million is 13·5 dB. In practice, however, owing to the need to allow for implementation margins, distortions introduced by the non-linearities (particularly when close to saturation) and equalisation considerations, this value is usually nearer to 17 dB.

Equalisation Considerations

A consequence of using digital modulation, particularly at very high rates, is the need to ensure that the up and down chains in the earth station have essentially flat group delay and amplitude characteristics over the bands of interest. This is to ensure that they do not impact significantly on the overall system performance. In the 120 Mbit/s systems, the tolerable group delay and amplitude characteristics over most of the wanted channel are specified to be a maximum of ± 0·15 ns and ± 0·25 dB respectively. These are difficult requirements to meet at the frequencies concerned, and require the use of fairly sophisticated equalisers. Additional amplitude and group delay equalisation must also be incorporated in both the earth station up and down chains to compensate for the filter responses of the satellite input and output multiplexer filters respectively.

In addition to equalisation of the frequency characteristics, it is also necessary to equalise signal path lengths. This requirement arises because of the very high bit rates being used at which delay through the equipment has a significant impact on the times at which bursts arrive at the satellite. For example, at 60 Msymbol/s, one symbol has a duration of 16 ns (corresponding to a free-space propagation distance of 5 m, or approximately 9 m in coaxial cable). In the 120 Mbit/s systems, the difference between electrical lengths of any two signal paths, including redundant equipment and chains, is specified to be less than 32 ns (2 symbols). This ensures that earth station inaccuracies do not contribute significantly to the burst guard times necessary, although these figure could be relaxed in TDMA systems operating either at more modest transmission rates, or when larger burst guard times can be used.

A further requirement for accurate timing comes from the need for terminals to carry out transponder hopping. This is a technique in which a terminal transmits or receives bursts to or from more than one transponder in a single frame. It is carried out in accordance with the burst time plan and is a particularly useful facility for increasing system flexibility. It has an impact, however, on the path length equalisation required since the instant at which bursts arrive at the hopping switch (for any given signal path) must also be carefully controlled. In the 120 Mbit/s TDMA systems, the relative delay between all possible signal paths to (and from) the antenna feed and the point at which transponder hopping is employed has to be less than 100 ns.

General Earth Station Characteristics

Finally, it is perhaps appropriate to mention the overall earth station characteristics employed with the INTELSAT and EUTELSAT TDMA systems. These systems have been designed to use very large earth stations, and to meet the digital performance criteria given in CCIR† Recommendation 522[4, 5]. These criteria are quoted in terms of bit error rate (BER) which should not exceed:

(*a*) 1 in 10^6 for more that 20% of any month;
(*b*) 1 in 10^4 for more than 0·3% of any month; and
(*c*) 1 in 10^3 for more than 0·01% of the year.

Initially, INTELSAT will introduce 120 Mbit/s TDMA at 6/4 GHz using INTELSAT V satellites over the Atlantic and Indian ocean regions[6]. Frequency reuse is employed and is achieved by both spacial and polarisation discrimination (left-hand and right-hand circular polarisation being used). The TDMA system will operate with Standard A earth stations, which have a dish diameter of about 30 m and a figure of merit (that is, gain relative to system noise) of 40·7 dB/K. These stations are required to transmit very high equivalent isotropic radiated power (EIRP) of approximately 90 dBW.

The EUTELSAT TDMA system will operate solely in the 14/11 GHz band (but with two orthogonally polarised beams), and provide coverage over most of central Europe. This system will use earth stations similar to the Standard C type, having a dish diameter of 14–18 m, a figure of merit of 39·0 dB/K and EIRPs of between 85 and 88 dBW.

IMPLEMENTATION

This article has provided a detailed description of the technical aspects of TDMA. To conclude, it is useful to review the reasons for adopting such a complex technique, the time-scales involved and the capacity that it will provide.

For INTELSAT, the decision to adopt TDMA was not easily made. The probable high cost of equipment weighed against it, the FDMA system was firmly established and working well, and other techniques were being suggested to overcome foreseeable capacity problems around the mid-1980s. Probably the most compelling reason for the decision to adopt TDMA was that it is a digital system, making it compatible both with the digital transmission systems which are finding increasing application in inland telecommunication networks and also with the services that will be offered on the emerging integrated services digital network (ISDN). Of course, it is not necessary to go to TDMA to provide digital transmission facilities, it would be quite feasible to operate a number of digital carriers through one transponder in FDMA. Unfortunately, such an arrangement requires that the travelling wave tube (TWT) amplifier on the satellite be operated in a linear region to avoid excessive intermodulation, and this requires operation at an output power well below its maximum. The result is a loss of downpath power (relative to that possible with single-carrier operation) and a consequent reduction in overall transponder capacity. It was clear, therefore, that in order to meet INTELSAT's capacity requirements with a digital system, it was necessary to adopt TDMA.

For EUTELSAT, the adoption of TDMA was not so difficult. With a European satellite system not then in operation, the problems of changing from one operating system to another did not exist. It was decided from the outset that the European system should be compatible with future digital networks and TDMA was the obvious technique to use.

The dates at which the two TDMA systems will be introduced have slipped somewhat from the original schedule. The advanced nature of the systems requires complex equipment both at the normal traffic terminals and the reference station terminals. In addition, to ensure that equipment from different suppliers will work together properly, any apparent omissions or ambiguities in the system specifications have had to be discussed internationally before decisions could be made. Both of these aspects tend to make manufacturing time-scales difficult to meet.

The current expected start dates for TDMA operation are late 1985 for INTELSAT (although it is expected that a pre-operational period will get underway somewhat earlier) and early 1986 for EUTELSAT. BTI has contracts with DCC Ltd. for the supply of terminals for both of these systems, and these will be ready for pre-operational service in the middle of this year.

For both INTELSAT and EUTELSAT, there will only be two or three participants at the outset, but in the two or three years after the TDMA systems start working, it is expected that many more administrations will join in so that, during the latter part of the 1980s, quite large proportions of international telecommunications traffic will be carried on TDMA systems.

The capacity of the TDMA system is dependent to a large extent on the type of traffic that will be carried. This is because DSI, which effectively doubles the capacity of channels on which it is used, cannot be applied to circuits intended for lease or for carrying digital data, and does not give any advantage on circuits carrying modem data. However, the basic capacity of the 120 Mbit/s TDMA system is about 1700 satellite channels in a 72 MHz bandwidth transponder, and on current estimates of the proportions of non-voice traffic, DSI will lift this to about 3000 one-way channels per transponder.

When INTELSAT starts TDMA operation, it will be with two transponders on the Major Path 2 satellite which provides transatlantic service. Over the following two or three years, two transponders on each of two other transatlantic service satellites, and two transponders on an Indian

† CCIR—International Radio Consultative Committee

Ocean region satellite (providing service between Europe and East Africa and the Middle/Far East) will be allocated to TDMA operation. As TDMA traffic levels build up, a further two transponders on the Indian Ocean region satellites will be made available for TDMA with the result that towards the end of the 1980s INTELSAT will have a total TDMA capacity of about 30 000 channels.

In Europe, EUTELSAT will start TDMA operation with three 72 MHz transponders on the new EUTELSAT 1 satellite (previously called *ECS*), and this will increase to a maximum of seven by the late-1980s, giving an intra-Europe TDMA capacity of around 21 000 channels by that date. BTI is a major member of both INTELSAT and EUTELSAT and currently plans to use TDMA capacity on four of the five satellites. TDMA terminals will therefore be required on four aerials located at both Goonhilly and Madley earth stations. This commitment will give BTI the international transmission capability to carry all digital telecommunication services (up to 2 Mbit/s) to any country that participates in the system.

SUMMARY

This article has outlined the fundamentals of TDMA and has provided a medium-depth description of the INTELSAT and EUTELSAT 120 Mbit/s TDMA systems. This has included a description of the burst structure, acquisition and synchronisation procedures, interface and housekeeping functions, and the reasons for the implementation of the systems. Both of these systems will become fully operational during 1986, marking the start of widespread international digital communications in the fixed satellite services in Europe, America and the Middle/Far East.

References

[1] INTELSAT TDMA/DSI System Specification, BG-42-65E (Rev. 2), June 1983.

[2] EUTELSAT TDMA/DSI System Specification, ECS/C-11-17 Rev 2., July 1983.

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

[4] CCIR Recommendation 522-1, Recommendations and Reports of the CCIR 1982. Volume IV Part 1, XVth Plenary Assembly, ITU, Geneva 1982.

[5] LEWIS, J. R. Factors Involved in Determining the Performance of Digital Satellite Links. *Br Telecommun. Eng.*, Oct. 1984, **3**, p. 174.

[6] EATON, R. J., and KIRBY, R. J. The Evolution of the INTELSAT V System and Satellite. *Post Off. Electr. Engs. J.*, Apr. 1977, **70**, p. 2, and July 1977, **70**, p. 68.

Biographies

John Lewis studied at the University of Bradford from 1968 to 1972 and gained an honours degree in Electrical Engineering; he later obtained the Diploma of Imperial College in communications. After graduating, he joined British Telecom International, and has worked on many aspects of satellite system design. He was until recently Head of the Digital Systems Group in the Satellite Services Division. In 1979, he was appointed by CCIR to act as correspondent to CCITT SG XVIII with the responsibility for ensuring that due account was taken of satellite system requirements during development of ISDN error-performance objectives.

David Elliott graduated with a first-class honours degree in Electrical and Electronic Engineering from Queen Mary College, University of London, in 1978. He spent a further three years carrying out research into surface and bulk wave piezo-electric resonators, and was awarded a Ph.D. in 1982. Since joining BTI, in 1981, as an Executive Engineer, he has been involved in studies of digital satellite systems.

British Telecom Press Notice

Prestel for Singapore

Singapore has chosen British Telecom's (BT's) world-leading Prestel system in a £15M deal with Marconi, a GEC company, to develop the world's most advanced combined viewdata and teletext service. The contract is the biggest ever for an exported viewdata system.

Singapore Telecoms' unique Teleview service will use telephone links and television signals. The high capacity of television will enable the service to carry high-definition pictures, graphics and characters from the Chinese language in picture form.

Teleview is due to start public service to more than 1000 customers in March 1987. It will have advanced facilities such as electronic messaging, two-way Telex, Photo Videotex and terminals in public buildings linked to the Singapore telephone network's directory enquiry centres. By 1990, an estimated one million telephone customers could be connected to an enhanced service offering directory enquiry facilities in all public telephones.

Sir George Jefferson, Chairman of BT, said: 'This is the most exciting development in viewdata since GEC and BT joined to research and produce Prestel, the world's first and most successful public system. This contract has been won on technical and commercial merit against fierce international competition. Once again British Prestel technology has been proved to be the best in the world. Prestel terminal standards have been well proven technically and commercially and, significantly as this contract shows, those standards can be upgraded to provide a host of new features.'

At the heart of the service will be three GEC 4190 computers, each capable of handling 2000 simultaneous telephone calls. Prestel's exclusive software programs will be used and Marconi will supply adaptors, terminals and television transmission equipment. EASAMS, GEC's project management subsidiary and GEC Computers are also supplying expertise and equipment as part of the contract.

The Teleview system is designed to operate on both telephone links and standard 625-line television transmissions. Customers will receive signals on their terminals via the television link, using the telephone to send out their instructions and response frames. The service will also operate on telephone links alone in both directions in the event of interruptions to the television signal.

Initially, Marconi are supplying 1100 terminals, 800 for residential use with hand-held keypads and 300 keyboard versions for business use. Their contract also includes editing terminals, some equipped with full typewriter-style keyboards. Each terminal will double as a powerful 256 Kbyte random-access memory personal computer.

The Teleview service will offer viewdata across all standards from basic text, to alpha-mosaic graphics, alpha-geometric illustrations, dynamically redefinable character sets, and full-colour high-quality picture Prestel or Photo Videotex by using the high transmission capacity of television.

Singapore becomes the eleventh country to purchase Prestel technology from Britain. The others are the Netherlands, Switzerland, Hong Kong, Malaysia, Australia, Austria, Belgium, Italy, New Zealand and West Germany.

A Review of the Applications of Different Types of Television Codecs

G. A. GERRARD†

UDC 621.397.6 : 621.38.037.33 : 621.38.037.37

The rapid digitalisation of telecommunication networks and the emergence of new video services has brought about an increased interest in the use of television codecs. Although the development and application of television codecs have been inhibited by the lack of international standards for the digital transmission of television signals, it is felt that enough experience has been gained to give some idea of the likely fields of application of the different types of television codecs. This article, therefore, explores the likely use of codecs in such fields as broadcast, satellite and cable television, and videoconferencing. This is done by highlighting the constraints placed upon application by such factors as the form of coding, network configuration and the type of service to be offered. The advantages and problems inherent in the digital transmission of television signals are discussed, and the article concludes by speculating on future requirements for television codecs.

This article is based on a paper presented at the Institution of Electrical Engineers (IEE) Third International Conference on Telecommunication Transmission, London, and is reproduced with the permission of the IEE.*

INTRODUCTION

Work on the digital coding of television signals has been going on for many years, and the techniques for digital coding and for reducing bit rate are fairly well known. There is now a much more complete understanding of the television scanning process and of how pre- and post-sampling filtering affects picture quality[1]. Analogue-to-digital and digital-to-analogue converters operating at video sampling rates are now available as integrated circuits, and this considerably simplifies the implementation of complex coding schemes. Most of the work in this field has been done by broadcasting organisations and telecommunications authorities since the lack of any internationally agreed standard for the digital transmission of television signals has inhibited the commercial development of codecs.

There is now a renewed interest in the digital transmission of television signals because telecommunication networks are becoming digital and digital techniques are becoming more common in television studios. This is particularly so now that agreement has been reached for the component coding of television signals in studios (CCIR‡ Recommendation 601). The time therefore seems opportune to review some of the applications which are now emerging for television codecs, for the transmission of both broadcast television and other video services such as cable television and videoconferencing.

CODECS USING PULSE-CODE MODULATION

It is convenient to begin this review by considering the requirements for the pulse-code modulation (PCM) coding of composite colour-television signals such as the PAL** System I signal used in the UK. This signal has a bandwidth of 5·5 MHz, and so a minimum sampling frequency of about 12 MHz is required to prevent aliasing when realisable designs of low-pass filter are used. In order to ensure smooth grey-scale reproduction, at least 8 bits per sample are required; and this results in a bit rate of around 100 Mbit/s so that the fourth level of the digital hierarchy, or 140 Mbit/s, would be required for transmission. The PCM coding of a television signal results in a high bit rate, but it has the advantage of being the most analogue-like and signal-transparent form of digital television coding. For this reason, it is particularly suitable for the coding of television signals for transmission over digital plant in a predominantly analogue network where the performance requirements are specified in analogue terms.

The existing network provided by British Telecom (BT) for the broadcasting authorities has all analogue interfaces, and the links are provided by analogue transmission systems. In BT's main trunk network, transmission systems are now increasingly being provided by means of digital plant and, on occasion, it may be convenient to use such plant as links in the television network. In that case, codecs will be required to provide suitable analogue interfaces with the existing television network. Codecs for this application have to meet the well-established analogue performance requirements for a main television link[2]. This means that the codec has to pass the vertical interval test signals for monitoring performance. The codec must also allow for the transmission of television signals with sound-in-syncs and teletext, and be suitable for NTSC and SECAM television signals which have to pass over the inland network to television studios where standards converters are located. It is also important that links using codecs can be connected in tandem and still meet the performance requirements of the UK television hypothetical reference circuit[2]. 9 bit PCM enables these requirements to be met and is technically feasible. The resulting bit rate for a sampling frequency of 13·9 MHz (a tenth of the digital channel rate) is 125 Mbit/s; a number of 2 Mbit/s signals can therefore be multiplexed with the digital television signal in the codec to bring the gross bit rate up to 140 Mbit/s and thereby use the channel more efficiently.

An impairment of the digital path which needs to be considered in connection with codec applications is the error ratio. The subjective effect on picture quality depends on the coding process used and the error distribution. In this respect, PCM coding appears to be most tolerant, a random error ratio of about 10^{-7} being just perceptible for 8 or 9 bit PCM.

† Trunk Services, British Telecom National Networks

* GERRARD, G. A. A Review of the Applications of Different Types of Television Codecs. IEE Conference Publication No. 246, 1985, pp. 282–285.

‡ CCIR—International Radio Consultative Committee

** PAL, SECAM and NTSC refer to standard systems for encoding colour-television signals

CODECS USING BIT-RATE REDUCTION SCHEMES

Reference 3 indicates some of the techniques which may be used to reduce the bit rate required for a television signal. These techniques take advantage of the inherent redundancy in a television picture and the differing tolerances of the human visual system to different picture impairments. The price paid for this reduction in bit rate is increased codec complexity, a reduced tolerance to errors occurring in the transmission channel and increased distortion of the television waveform. This distortion does not necessarily impair the picture quality since the coding process is generally optimised for both the signal to be transmitted and the visual characteristics of the human observer. However, the waveform distortions introduced on conventional analogue television test signals by these more sophisticated forms of coding are not, in general, directly related to analogue distortions, and test signals are therefore of limited value in objectively measuring the performance of such codecs. Thus, these codecs are not well suited for use in a predominantly analogue network where performance is monitored by conventional analogue test signals.

Differential PCM

Differential PCM (DPCM) is a commonly used technique for reducing the bit rate and is used in many television codecs either on its own or in combination with other techniques. DPCM operates by transmitting the difference between the true sample value of the signal and a prediction based on a previous sample or samples. The simplest implementation of DPCM for the transmission of a composite colour-television signal is to use a sampling frequency of three times the colour subcarrier frequency and transmit the difference between samples spaced three picture elements apart[4]. For broadcast television picture quality, the difference signal needs to be coded to at least 5 bit resolution, and this gives a bit rate for a PAL or SECAM signal of 66·5 Mbit/s. A rate of 2 Mbit/s is suitable for the transmission of a number of sound programme circuits, and a convenient package is formed if a 2 Mbit/s signal is multiplexed with the digital television signal in the codec[5] to give a gross bit rate of 68 Mbit/s. Access to the digital network at 68 Mbit/s can be provided in the UK, but this is not an internationally agreed hierarchical level. The picture impairment introduced by this type of coding[6] is equivalent to about half that allocated to the UK television hypothetical reference circuit. For broadcast-quality pictures, the use of this coding is therefore restricted to applications where only one coding and decoding process would be required. Television signals with sound-in-syncs are satisfactorily transmitted by this codec, but conventional full line and vertical interval test signals, although transmitted, are of limited value in objectively assessing performance, for the reasons previously discussed.

A similar type of codec to the above is particularly useful for the trunk transmission of cable television signals[7]. In this implementation, two television signals, two television sound circuits and three radio sound circuits are multiplexed for transmission at 140 Mbit/s.

Straightforward third-previous-element DPCM is vulnerable to errors in the transmission channel. A single bit error manifests itself as a bright or dark streak from the occurrence of the error to the end of the television scanning line. Practical implementations, as in references 5 and 7, use a hybrid form of DPCM which greatly reduces the visibility of errors.

Sub-Nyquist Sampling and DPCM

The subjective picture impairment introduced by predictive coding can be reduced by using a more complex prediction algorithm and increasing the number of bits in the transmitted difference signal. If the resolution of the transmitted difference signal is increased to six bits, then the sampling frequency needs to be at a sub-Nyquist rate in order to keep the resulting bit rate within 68 Mbit/s. The energy in a television signal is concentrated at multiples of the line frequency and it is possible to sample at a sub-Nyquist rate provided the frequency is chosen so that the alias components interleave between the spectral lines. The video-frequency components can be recovered by a comb filter which is proportioned to have maximum attenuation at odd multiples of half-line frequency. A codec employing these techniques[8] uses sub-Nyquist sampling at a frequency of 8·86 MHz (twice the PAL colour subcarrier frequency) and 6 bit DPCM, and gives a video bit rate of 53·2 Mbit/s. A number of data channels are multiplexed with the digital television signal in order to bring the gross bit rate up to 68 Mbit/s. These data channels can be used for the transmission of sound programme circuits via NICAM† equipment since the codec does not transmit television signals with sound-in-syncs. This codec has been optimised for the transmission of PAL signals, and the prediction algorithm uses eight samples taken from both the present and previous fields of the television picture. An important consideration was that up to four coding and decoding processes in tandem should not produce perceptible degradation of the picture.

This codec has been used in a joint BBC*/BT 140 Mbit/s digital transmission pilot scheme between London and Birmingham in which the equipment carried BBC1 and BBC2 services to the Midlands and to the North[9]. The objective of the scheme was to assess the technical and operational difficulties of going digital in a broadcasting environment.

Each decoder incorporates an optional forward error corrector that uses a Reed Soloman code and is capable of correcting error bursts of up to 67 bits. A comprehensive error monitoring system is linked to the error corrector, and one of the objects of the trial was to see whether a powerful error corrector of this type is necessary and effective on practical links.

Blanking-Interval Suppression, Sub-Nyquist Sampling and DPCM

Further reduction in bit rate can be achieved by only coding the active line period of a television signal. In combination with sub-Nyquist sampling and DPCM, this technique[10] allows transmission of broadcast-quality signals at 34 Mbit/s. This bit rate appears particularly suitable for transmission over satellite circuits since it enables one television signal and associated sound circuits to be accommodated in half a transponder. In this application, some form of error correction may be necessary since satellite circuits may have a much poorer performance in respect of errors than terrestrial links.

Conditional Replenishment

In videoconferencing, the less stringent demands on motion rendition enable temporal redundancy in the picture to be exploited; this requires a frame store in both coder and decoder. A well-proven technique is conditional replenishment, in which only those parts of the picture that change significantly frame by frame are transmitted. The resulting non-uniform information rate is smoothed by a buffer store and by a coding algorithm that adapts to the rate of information in the picture. The aim is to effect a graceful reduction in picture quality with increasing motion of the picture, within the constraints of the transmission channel. A transmission rate of 2 Mbit/s has been recommended as a

† NICAM—Near instantaneously companded audio multiplex
* BBC—British Broadcasting Corporation

standard for videoconferencing in Europe, and codecs were built to operate at this standard as part of a European collaborative project, known as *COST 211*, which ran from 1971–1982. Further developments have enhanced the facilities provided by this codec and it is now used for videoconferencing on national and international routes[11]. Basically, the codec provides for the transmission of PAL, SECAM and NTSC signals together with audio and data signals at bit rates from 2 Mbit/s down to 768 kbit/s (twelve 64 kbit/s channels). The design of the video synchronisation and transmission framing structure is such that a 525-line/60 Hz field rate/1·5 Mbit/s version of the codec is compatible with a 625-line/50 Hz field rate/2 Mbit/s version of the codec. No error correction is incorporated in the decoder, but the coding algorithm has been chosen to be error resilient.

Conditional replenishment at a transmission rate of 2 Mbit/s does not produce broadcast-quality pictures since, for videoconferencing, low transmission costs are more important than full broadcast quality. However, at only slightly higher bit rates, for example, 8 Mbit/s, vastly improved motion rendition is possible, and picture quality appears to be comparable with electronic news gathering (ENG) video recording equipment.

Transform Coding

In transform coding, a television picture is divided up into a large number of blocks, usually of 8 × 8 or 16 × 16 picture elements, and the magnitude of the samples in each block is operated on by a suitable transform. This results in a spectral-like presentation of the television signal, which can be transmitted with less bits per sample than the original PCM samples of the signal because the higher-order coefficients of the transform do not need to be quantised to the same accuracy as the lower-order terms.

Transform coding techniques are expensive in terms of the real-time computational hardware required to implement them and do not appear to offer advantages compared with predictive coding techniques for the transmission of broadcast-quality television pictures. However, combined with other coding techniques, such as conditional replenishment, they enable television pictures to be transmitted at really low bit rates, down to 64 kbit/s, and of adequate quality for videoconferencing applications.

CODECS FOR COMPONENT-CODED TELEVISION SIGNALS

CCIR Recommendation 601 outlines the sampling frequencies and form of coding recommended for digital studio equipment using component signals. For the standard digital interface between main digital studio equipment, and for international programme exchange, the sampling frequencies of the luminance and colour difference signals are related in the ratio 4:2:2, where the figures represent multiples of 3·375 MHz. The form of coding to be used is 8 bit PCM and the resulting bit rate when the three signals are multiplexed is 216 Mbit/s. Predictive coding techniques and sub-Nyquist sampling can be used to reduce this bit rate to 140 Mbit/s for transmission over 140 Mbit/s transmission channels. Alternatively, the multiplexed component signal could be transmitted without any bit-rate reduction at a possible 280 Mbit/s interface on 565 Mbit/s transmission systems. This method would have the advantage of preserving the picture quality inherent in the 4:2:2 standard and would allow post-production processing of the received signal to be carried out with less impairment to the picture.

CONCLUSION

This article has discussed some of the significant developments in television codecs and their application for the point-to-point transmission of television signals. The efficient coding of picture material allows significant reductions to be made in bit rate and therefore in transmission costs, but the degree to which this can be carried out depends on the application. At one extreme there is videoconferencing, where the less stringent requirements on motion rendition enable the bit rate to be reduced to very low values—2 Mbit/s or even 64 kbit/s. Broadcast television is at the other extreme; here picture quality needs to be at least as good as that offered by existing analogue transmission media and, when the possibility of high-definition television is being proposed, possibly needs a bit rate of 565 Mbit/s.

For the transmission of composite colour-television signals in a mixed analogue and digital network, 140 Mbit/s 9 bit PCM appears necessary if the same signals and operational flexibility as in a completely analogue network are required. In a predominantly digital network, 68 Mbit/s appears an appropriate rate for the transmission of composite colour television and multiplexed sound and data. For the transmission of component-coded signals, codecs may be used to allow transmission at 140 Mbit/s on existing systems. However, the future provision of transmission systems operating at higher bit rates—for example, 565 Mbit/s—may prove to be a more convenient and cost effective form of transmission for these signals.

References

1 TONGE, G. J. The Television Scanning Process. *SMPTE J.*, 1984, **93**, pp. 657–666.

2 MACDIARMID, I. F. Performance Requirements of Links for Colour Television. IEE Conference Publication No. 46, 1968, 6.8.1–6.8.4.

3 DEVEREUX, V. G., and JONES, A. H. Pulse-Code Modulation of Video Signals: Codecs Specifically Designed for PAL. *Proc. IEE*, 1978, **125**, pp. 591–598.

4 THOMPSON, J. E. Differential Coding for Digital Transmission of PAL Colour Television Signals. IEE Conference Publication No. 88, 1972, pp. 26–32.

5 SEXTON, M. J. A Codec for Television Programme Distribution in a Digital Transmission Network. IEE Conference Publication No. 193, 1981, pp. 124–127.

6 REDSTALL, M. W., and WHITE, T. A. Subjective Quality of a 70 Mbit/s Digital Codec for Colour Television. *Proc. IEE*, 1983, **130**(F), pp. 477–483.

7 MOGENSEN, G., SCHARØE PETERSON, B., and STEFFENSEN, H. H. DOCAT-Digital, Optical CATV Trunk System. IEE Conference Publication No. 220, 1982, pp. 257–260.

8 JONES, A. H., and KITSON, D. M. Digital Transmission System for Television, Sound and Associated Data. IEE Conference Publication No. 240, 1984, pp. 327–330.

9 KEARSEY, B. N., and GOOCH, P. V. The Transmission of Broadcast Services over 140 Mbit/s Bearers. IEE Conference Publication No. 240, 1984, pp. 331–337.

10 RATLIFF, P. A. Digital Coding of the Composite PAL Colour Television Signal for Transmission at 34 Mbit/s. *BBC Eng.*, 1980, **115**, pp. 24–35.

11 NICOL, R. C., and DUFFY, T. S. A Codec System for Worldwide Videoconferencing. *Professional Video*, 1983, **10**, 36–42.

Biography

Alan Gerrard joined the Post Office Research Station at Dollis Hill in 1967 as an Experimental Officer, and worked there on various aspects of analogue television and the digital encoding of television signals. He transfered to the facsimile systems group in 1975, and in 1976 moved to the BT Research Laboratories at Martlesham Heath, where he worked on faster and improved facsimile communications equipment. After promotion, he took up his present post in 1979 as head of television systems development in the Trunk Services Engineering Division of BT National Networks.

4TEL in TXK1 Local Exchanges

D. ROSLING, and R. RUNDEL†

UDC 621.395.342

This article describes experience in Severnside District of using the 4TEL Automated Subscriber Loop Test System, supplied by Teradyne Ltd., to replace the automatic line insulation tester (ALIT) in TXK1 crossbar local exchanges. The article describes some of the problems encountered, and how the redundant ALIT was modified to make the 4TEL equipment more efficient and cheaper to purchase.

INTRODUCTION

Early in 1983, the then Bristol Telephone Area began trials of a new American-designed system for testing customers' lines, supplied by Teradyne Ltd. The system, known as the *4TEL Automated Subscriber Loop Test System*, comprised a central main computer installed at the Bristol Repair Service Centre, and remote dial-and-test units fitted at distant satellite telephone exchanges.

The remote units, known as *central office line testers* (COLTs), seize and test customers lines via the normal test access circuits. Tests can be carried out overnight by using a routine mode in which lines are automatically tested in the sequence 0000–9999. The results are stored in the central computer and can be printed out when required. Alternatively, individual lines can be tested on demand, and the results displayed on a visual display unit.

One of the exchanges chosen for the trial was the TXK1 crossbar exchange at Henbury, which had a multiple of 10 500 lines. Investigations showed that a COLT with one dial circuit could test only about 4000 circuits each night. Another dial circuit was added at extra cost, but still only about 8000 lines per night could be tested. The problem was caused by the method used to access line circuits in a TXK1 exchange.

Each COLT dial circuit was connected to a test transmission relay group (TTRG), and dialled a four-digit code to seize each line circuit, including all spare lines. The two dial circuits tended to operate simultaneously and dial consecutive numbers. At the linemarker stage of call set-up, only one terminating call at a time can be processed. The coincident calls sometimes led to an apparent system failure and, consequently, a fault print-out on the exchange teleprinter. Fig. 1 shows the original equipment configuration.

The call-count meters in the exchange indicated the following increases in call rate in the routers used by the 4TEL system:

† Severnside District, British Telecom Local Communications Services

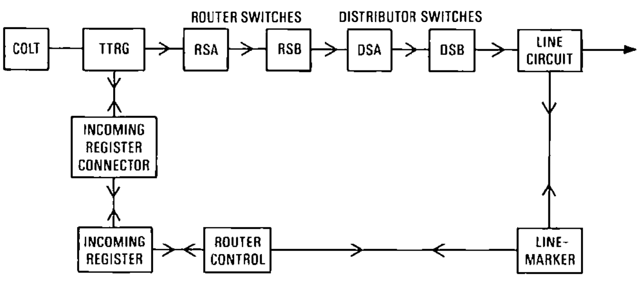


FIG. 1—Original method of interfacing the COLT to a TXK1 exchange

Incoming register connector	Operations doubled
Router controls	Operations doubled
'Preference choice' incoming registers	Operations ten times normal

The massive increase in the use of registers caused concern because the wear, and subsequent failure, of pulsing relays is a major service hazard in TXK1 exchanges. It was therefore suggested that the automatic line insulation tester (ALIT), made redundant by the 4TEL system, could be used to set up the 4TEL calls. The ALIT is, in essence, a register that has the ability to test lines, and the great advantage of being able to increment calls sequentially without re-dialling. Using the ALIT as an interface between the 4TEL and the router control (see Fig. 2) would eliminate the extra wear on the registers.

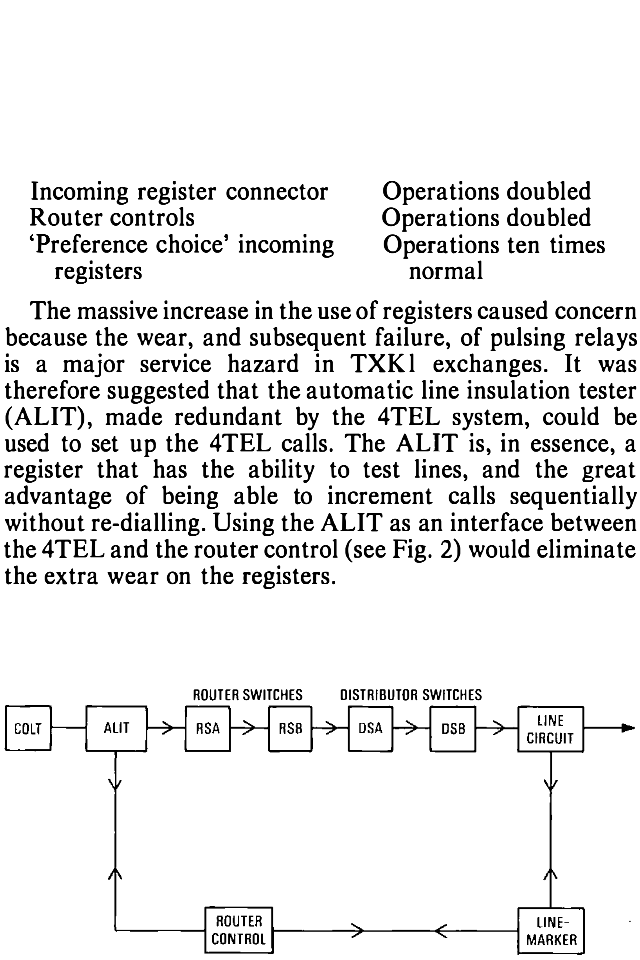


FIG. 2—New method of interfacing

Preliminary investigations also indicated that time could be saved by changing the COLT control program. In Strowger exchanges, the 4TEL program includes some call incrementing. The COLT dials via the test selector access to the first outlet of the test final selector bank. After the test has been completed, the wipers are stepped on to the second outlet. This continues up to digit 9, when the COLT clears and redials the number ending in '0'. It was thought that this increment feature of 4TEL could be used in a TXK1 exchange. Calculations showed that, by using only one dial circuit, more than 10 000 tests per night should be possible.

DEVELOPMENT

The ALIT is normally set up for a test programme by code digits dialled from a remote access point into relay stores, as follows:

First digit	Type of start; for example, immediate or delayed
Second digit	Insulation test limits
Third digit	10 000-line group to be tested
Fourth digit	1000-line start group to be tested (for example, 2000)
Fifth digit	100-line start group to be tested (for example, 2300)
Sixth digit	1000-line finish group (for example, 3000)
Seventh digit	100-line finish group (for example, 3599)

The tens and units counting stores are automatically set to zero, ready to increment from there. In the example given, the ALIT would have tested from 2300–3599.

The first stage of the development was to list the COLT requirements, check the COLT signalling arrangements, and then modify the ALIT accordingly. The COLT had to be able to

(*a*) set up a call to any number, and increment if necessary for the routine mode;
(*b*) control the increment;
(*c*) receive information on line status, that is, FREE, BUSY, number unobtainable, etc;
(*d*) test the line if FREE;
(*e*) monitor the line if BUSY; and
(*f*) clear on completion of the test programme.

The first two digits of the original ALIT programming were not necessary as the type of start was always going to be immediate and the COLT would do the testing. The first two ALIT information stores were therefore strapped out, so that the digits dialled by the COLT became:

First digit	10 000-line group to be tested
Second digit	1000-line group to be tested
Third digit	100-line group to be tested
Fourth digit	10-line group to be tested
Fifth digit	unit number to be tested

The rewiring work that allowed the COLT to operate the tens and units stores for digits four and five was a major task. The OPERATE leads that originally went to the finish stores of the ALIT had to be diverted to the tens and units stores. This enabled the COLT to pre-set the tens and units stores to any required number. As spare relay contacts were used, only four extra diodes were required to carry out these changes. The increment feature of the ALIT stayed much the same as before, but, instead of it being automatic, the COLT controlled it. Thus the COLT initiates a one-step increment, checks that one step has occurred, and then progresses with testing.

Once the fifth set-up number has been received, the ALIT reverses the potential of the dialling wires. The COLT recognises this signal, and sends an earth to the ALIT on the START leads. The ALIT then acts as a register and seizes a free router control. When the router control has interrogated the line circuit required, the relevant information on line status is returned to the ALIT. If the line is FREE, then the call is completed and the ALIT signals to the COLT. If the line is BUSY, then the call is completed and the ALIT signals the two facts to the COLT. Where there is number unobtainable (NU) tone, change number interception (CNI) or equipment engaged tone (EET), this information is passed from the ALIT to the COLT on separate leads.

When a call is completed from the ALIT to a line circuit, a test is made by the COLT. To facilitate this, the test wires of the ALIT were diverted from the original testing element to the COLT test circuit. The line test takes about two seconds, and then the COLT can either increment or clear down.

If the line under test is BUSY, then the call is still completed. This is not a normal feature of the ALIT, but was achieved by giving the ALIT a class-of-service identical to the TTRG. Thus the router control overrides the *busy* signal, and the call is set up as if the line is FREE. If the COLT monitoring facility is in use, then the line being tested is checked. If voice-frequency modulation is present, the COLT recognises this as a successful call in progress. This information can then be shown in the results. If monitoring is not required, the COLT can increment immediately.

A call is not set up to a line circuit where there is NU tone, CNI, or EET. In these cases, the information on line conditions is stored, and the COLT increments or clears down.

When the COLT has finished its tests, an earth signal is sent to operate the ALIT clear-down relay.

OPERATIONAL EXPERIENCE

The modified ALIT has been in use since mid-July 1984, and has proved to be very reliable. Fig. 3 shows a comparison

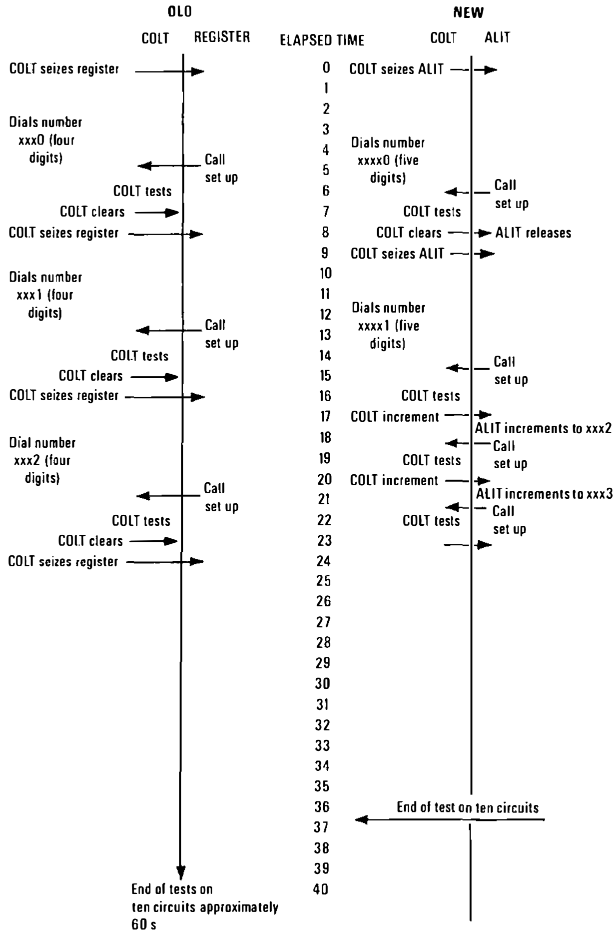


FIG. 3—Comparison of test timings between the original and new methods

between the old and the new system for testing ten lines, and shows an approximate saving of 22–24 seconds per ten tests. This saving means that, under the new arrangement, only one dial circuit in the COLT needs to be used to test 10 000 lines each night. Assuming that all lines are working, and no re-tests are required, then the time required to test 10 000 lines is approximately 10 hours.

As previously explained, the COLT can increment 1–9, but then clears and re-dials to 0. Changes to the software could make the COLT increment from 0–9 and, possibly, 0–99. These modifications would enable the time required to test 10 000 lines to be reduced to approximately 8·8 and 8·0 hours, respectively.

CONCLUSION

The new ALIT–COLT interface has proved a success. The wear and tear on registers is back to normal, and the nightly 4TEL routine can now test 10 000 plus lines per night.

Financial as well as time savings have been achieved as only one dialler instead of two is required. The modifications take about three days to complete, and the extra equipment needed is one standard relay, six diodes, and the cabling.

This new system is being adopted in other large TXK1 exchanges in Severnside District, and interest has been shown from other Areas/Districts suffering from the same problems with the 4TEL system.

Biographies

David Rosling joined the then Post Office as a Youth-in-Training in 1956. He completed two years of national service in the Royal Corps of Signals and returned to the Post Office as a Technician 2A on subscriber apparatus maintenance duties. Later, as a pool Technical Officer, he worked on UAX7 maintenance, TXK1 maintenance and special faults investigation. In 1981, he became an Assistant Executive Engineer, supervising Strowger and TXK1 exchanges. Since 1984, his work load has concentrated on TXK1 exchanges, with the additional duties of the new Severnside District TXK1 Quality of Service and Liaison Officer.

Ray Rundel joined the Post Office as an apprentice in 1953. He served for two years with the Royal Corps of Signals in Kenya, and returned to the Post Office to work on customer apparatus maintenance in 1957. Since his promotion to Technical Officer, he has been an engineering instructor and worked on Strowger maintenance. At present, he is working on crossbar maintenance.

Book Review

Mathematical Topics in Telecommunications. Volume 1, Optimisation Methods in Electronics and Communications; and Volume 2, Problems of Randomness in Communication Engineering. Edited by K. W. Cattermole, and J. J. O'Reilly. Pentech Press. Vol. 1, 168 pp. 72 ills. £18·00; Vol. 2, 350 pp. 112 ills. £26·00.

These two volumes are the first in a series aimed at extending the knowledge of mathematical techniques available to professional telecommunication engineers. Edited by Ken Cattermole and John O'Reilly of the Telecommunications Group at Essex University, the source material is derived from a series of colloquia held at Essex University. Each chapter covers a distinct topic and is written by a different author, usually an academic but occasionally a practising engineer. The presentation is on the whole good, with a diversity of contributions successfully brought together to give a good composite coverage of the subject's themes. The mathematical content often reaches post-graduate level, and this may not always appeal to a general readership. However, the intention is to give visibility to techniques which do have practical use, and numerous examples of their application to common problems ensure that this aim is achieved.

Part 1 of the first volume provides a good mathematical foundation to the calculus of variation and the optimisation of multi-variable functions. After a general treatment of surfaces and the meaning of maxima, minima and boundary conditions in multi-dimension space, the Euler-Lagrange equation is established. In a lighthearted discourse which may appeal to the fellwalkers amongst readers, the mathematical representation of a public footpath on a mountainside is derived! Of more practical value, the use of such optimisation techniques in the design of filters is then given good coverage. Other examples of particular interest deal with the problems encountered in the design of optical communications systems and design in the presence of Poisson noise (in contrast to the more familiar problems of dealing with band-limited Gaussian noise). Having introduced the reader to the optimisation of problems which may be conveniently represented by multi-variable algebraic functions, the authors provide, in Part 2, coverage of the numerical, iterative techniques which are more likely to be encountered in practice. Direct search and gradient techniques are covered, with derivation of Newton-Raphson and Gauss-Newton methods. Consideration of boundary constraints leads on to the more general quasi-Newton methods. A number of worked examples are given for the optimisation of linear networks and computer-aided filter design.

Almost all aspects of communication theory are influenced by parameters which are of a probabilistic, rather than of a determinate nature, and Volume 2 in this series covers such problems. It is divided into four parts. An introduction discusses the theoretical basis for probabilistic techniques. The concept of probability generating functions, spectral characterisation of cyclo-stationary random processes, Markov chains and the tackling of boundary constraints and approximations are covered. It is not long before the practising engineer will realise that many problems cannot be dealt with by using such simple probability distributions (for example, Gaussian, Binomial) and that the parameters of such distributions are themselves subject to random fluctuations or discontinuities. In an attempt to provide enlightenment in such matters, one section of this volume is devoted to compound randomness and another to the treatment of non-linear operations on stochastic processes. Examples of compound randomness are found in the modelling of traffic in networks, the representation of the detected signal in an optical communication receiver and the modelling of burst errors in digital transmission systems. Examples of non-linear operations are found in the extraction of a timing signal from a baseband digital pulse-amplitude modulation (PAM) signal, the noise spectra associated with quantising noise and the mathematical modelling of the intermodulation products in a single-channel-per-carrier satellite communication system. The fourth section of this volume is devoted to digital line codes, their representation and analysis. A method of obtaining the power spectral density characterisation of specific codes is presented and the design of an improved 7B8B line code is discussed.

These two volumes do cover a great deal of ground and, for those interested in expanding their repertoire of mathematical techniques in the specific areas considered, they can be firmly recommended. However, they are books for the specialist, and the mathematical content is in general too high for the casual reader.

R. C. Ward

Maintenance-Free Gas-Recombination Cells

J. A. O'CONNOR† and R. J. DOODY*

UDC 621.353.2 : 621.39

This article describes the applications in British Telecom (BT) for maintenance-free lead-acid gas-recombination cells. It also describes the electrical testing and the materials and components analysis involved prior to granting design approval for sealed lead-acid cells, with specific reference to the 6 V 100 A h unit, which is the BT standard. Although it is agreed that there is no substitute for in-service experience, BT is satisfied that its testing procedures give a high degree of confidence that the designed life of ten years will be achieved. There are over 500 power systems employing recombination cells in 6 V units installed to date; indications are that the required reliability will be met.

INTRODUCTION

Early in 1980, work was started within Chloride Power Standby Ltd, Chloride Technical Ltd, and British Telecom (BT), on a new range of sealed lead-acid stationary cells for use in BT's digital telephone exchanges. The reasons for this development are perhaps worth stating here. At that time, the normal approach to providing DC power for switching or transmission equipment was to install plant to meet a forecast power demand (usually for 10–20 years) at the site. Thus, for any given period, this meant that a considerable element of provision was made ahead of demand. With the increasing pace of development in communications equipment, this approach was becoming less cost effective. It would clearly be advantageous if equipment could be installed complete with its own DC power system, designed to operate efficiently with that equipment.

The requirement was for a DC power system that offered:

(*a*) compatibility with the proposed layout of the switching equipment,

(*b*) good AC/DC efficiencies,

(*c*) flexibility of installation capability,

(*d*) universal application, including use in apparatus sited in customers' premises, and

(*e*) a cost-effective approach to the lifetime power requirements of the switching equipment; that is, parallel installation with telecommunications equipment.

It was clear that a new design of battery would be necessary. The need for flexibility indicated that the size of the battery would need to be limited, and large batteries, with capacities up to 15 000 A h, would no longer be required. Also, the lead times and the expense of large installations were not acceptable. Compatibility and cost effectiveness indicated that a lifetime of ten years in service would be acceptable, compared to current lifetime expectancy of 20 years in service obtainable with Planté cells. A sealed cell was required so that the provision of a separate battery room, with the consequent cost penalties involved, could be avoided, and for it to be compatible with an office environment for use in modern customer-based communications equipment. It was also clear that a new AC/DC conversion technique which would allow a reduction in the physical size of rectifier would be required, and a switch-mode rectifier design was adopted for this application.

The initial draft specification for the battery called for:

(*a*) floating at 2·27 V per cell continuously, with no periodic boost charging, and no boost charging after a discharge;

(*b*) a low internal resistance;

(*c*) a 24-cell battery with a capacity of 2 kW for one hour, operating within the limits of 54·5 V and 46 V;

(*d*) compatibility with the likely operating environments, including requirements for fire resistance; and

(*e*) a 3-cell monobloc with a footprint of 200 mm by 210 mm, height no more than 280 mm, and a maximum weight of 23 kg per unit. These dimensions were specified to enable installation in both the standard BT and commercial 19 inch rack practices.

To meet these parameters, a prismatic design based on the starved-electrolyte oxygen-recombination principle was offered. This approach was also adopted by Tungstone Products Ltd. and Gates Energy Products Ltd. BT were already using the Gates wound design of recombination cell in another application.

APPLICATIONS

The power system designed for System X exchanges is shown in Fig. 1. This approach has now been used in a variety of applications within BT, and further applications, both on new and existing exchange equipment practices, are in course of development and deployment. A typical installation is

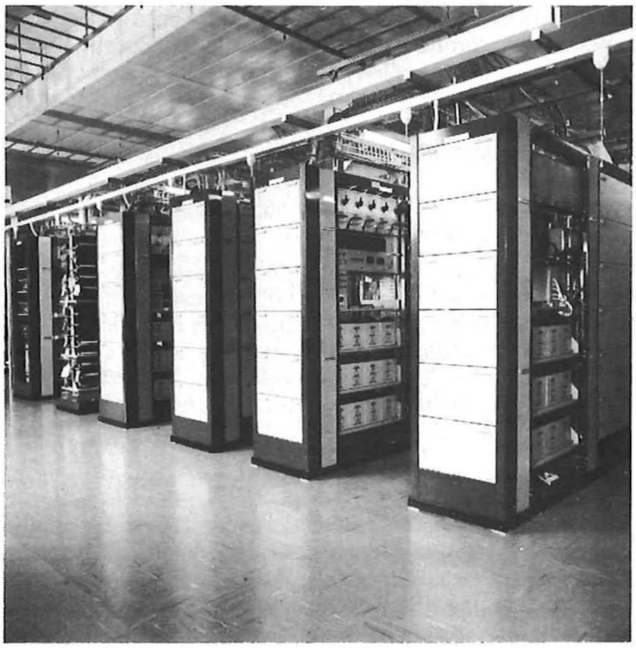


FIG. 1—Typical digital exchange power system

† Energy, Transport and Accommodation Department, British Telecom Local Communications Services

* Materials and Components Centre, British Telecom Development and Procurement

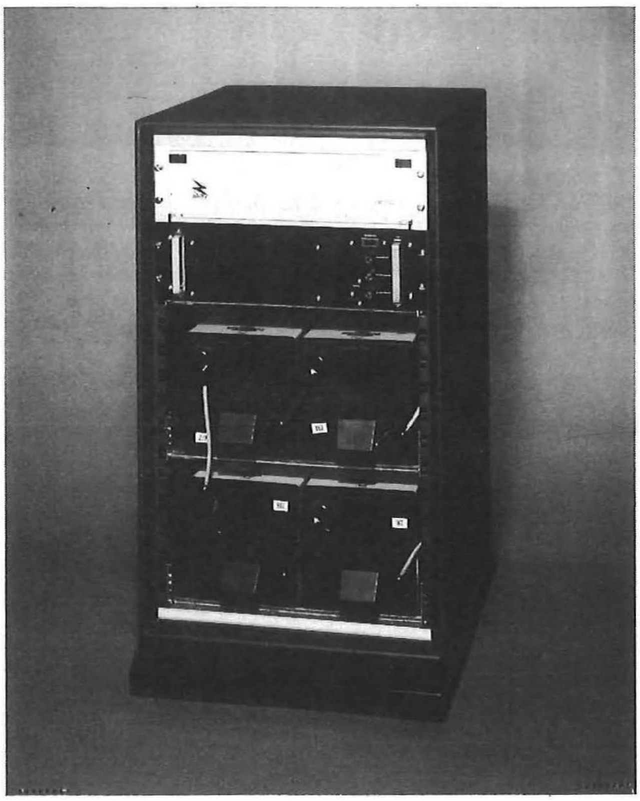


FIG. 2—Typical customer-sited power system installation

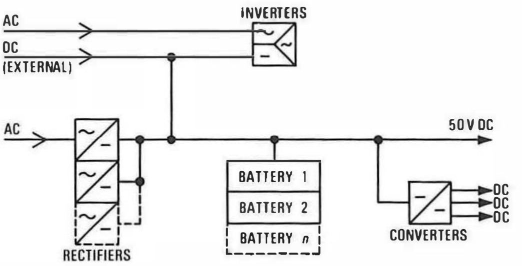


FIG. 3—Electrical layout

shown in Fig. 2.

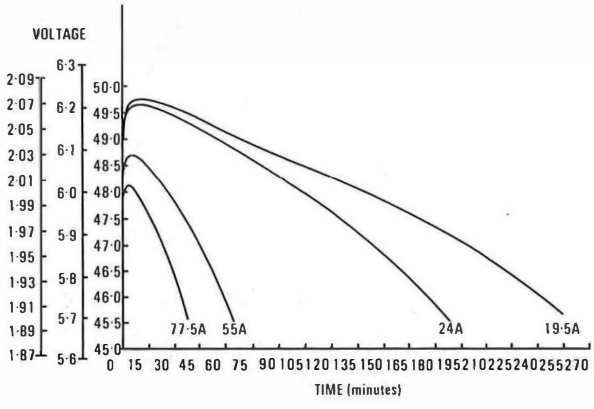
Most of these systems use the battery in a direct DC-coupled configuration (see Fig. 3). In this configuration, the battery acts as a smoothing filter and is on-line immediately in the event of mains-input failure. Modern digital equipment is more sensitive to mains disturbances, and this approach gives a high level of attenuation of these disturbances.

TYPE APPROVAL TESTING

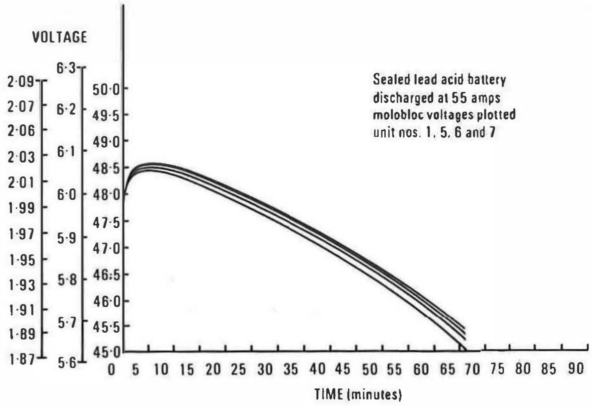
With no relevant data to rely on, a team was set up to evaluate and monitor the aspects of the designs offered; that is, the construction of the cells with regard to the integrity of the various mechanical components used, and the electrical performance and stability of the cells in service under the specified operating conditions. Testing commenced in January 1980.

ELECTRICAL TESTS

The electrical tests consisted of float charging 24 cells in series at 54·5 V. Five discharges, at the one-hour rate, were performed at weekly intervals. After six months to a year on float charging, the batteries were discharged again. Recharging was effected by using rectifiers set to 54·5 V, with the current limited to 15% of the rated ampere-hour capacity of the cells. Battery and cell voltages were monitored by



(*a*) Discharge curves for 6 V 100 A h monobloc



(*b*) Unit discharge voltages, including best and worst units

FIG. 4—Typical discharge curves

using a data logger. Fig. 4 shows some typical results of constant current and constant power discharges. Early results indicated that the power available more than met the requirements, and that cell-to-cell variations were not excessive. Trials were set up at a variety of operational sites to confirm the laboratory results. These gave some important information, which is discussed later.

During normal operation, these cells emit small amounts of gas through the re-sealable vents. During the above electrical tests, the small amounts of gas emitted were collected and sampled by using the apparatus shown in Fig. 5. The gas was shown to be predominantly hydrogen. This work resulted in a proposal for the definition of 'sealed' as applied to lead-acid stationary batteries. The proposal defines the maximum permissible gas emission as 5% of the Faradaic equivalent of the trickle-charge current under float operation, averaged over 24 cells for 1000 hours. The gas is assumed to be 100% hydrogen.

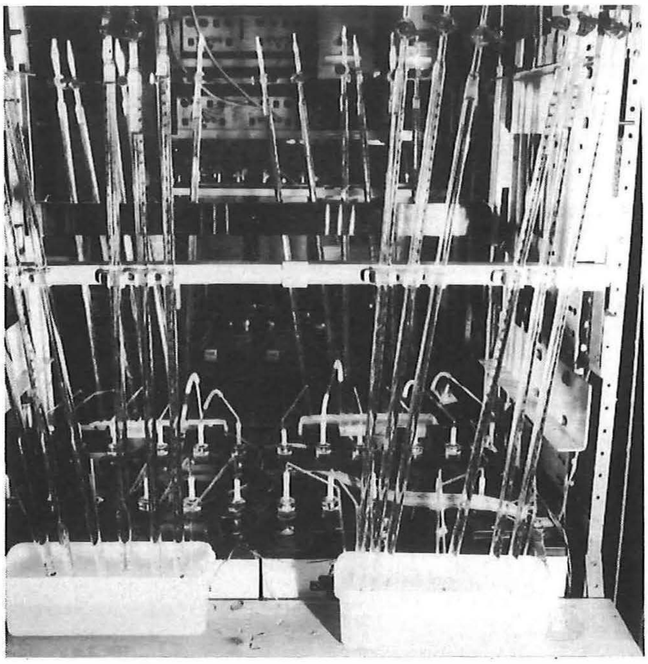
Charge/discharge cycling tests were not carried out as BT does not have any applications that involve the regular discharging of batteries. Testing of the cells for use on starting diesel-engined standby generators is in progress. Early results at −10°C indicate that the design is eminently suited to this application.

MATERIALS AND COMPONENTS TESTING

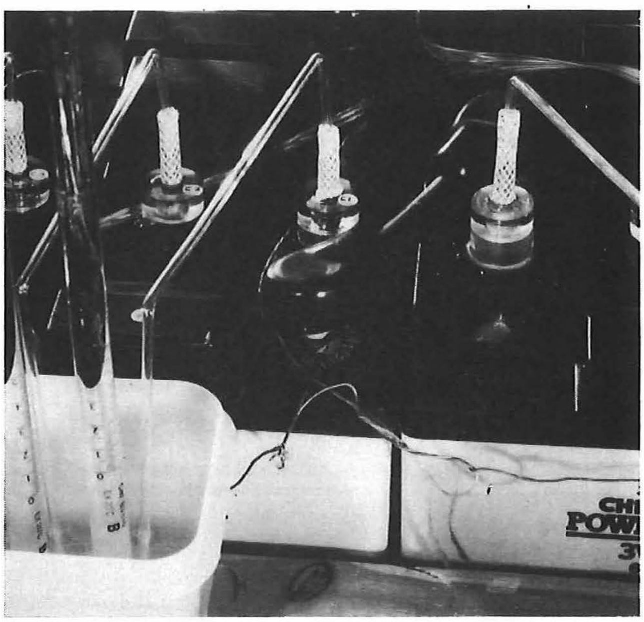
The materials testing was undertaken at BT's Materials and Components Laboratory in Birmingham.

Component Testing

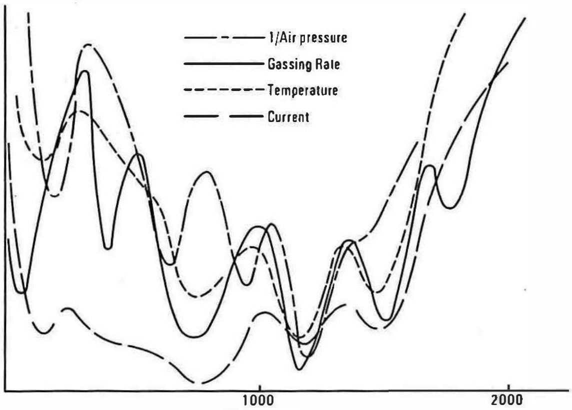
Component testing involved an analysis of all the major components to determine whether there were any manufacturing or design defects, and vibration testing of the assembled cells. Fig. 6 shows the main components analysed. The



(*a*) General arrangement



(*b*) Gas collection apparatus



(*c*) Variation in gassing rate

FIG. 5—Gas collection

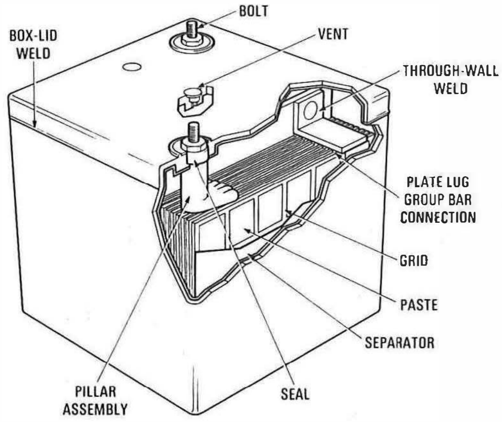


FIG. 6—Diagram of cell showing components analysed

pillar/group-bar sub-assemblies were examined for casting defects, especially where cast-on strap techniques were used. The group-bar-to-plate lug connections were also examined. Fig. 7 shows some typical results. The lid-box seal was checked for leaks by using a helium leak test, normally used on electrical components such as capacitors.

The pillar seals were closely examined and a testing procedure for corrosion resistance was devised. Pillar assemblies were partially immersed in a tray of sulphuric acid, and made anodic to the acid. By observing the spread of peroxidation up the pillar with time, a measure of the effectiveness of the seal design is obtained. Fig. 8 shows some typical results.

Grid cross-sections were examined after three years in service to establish likely rates of grid corrosion. Fig. 9 shows the results from the two grid alloys currently in use. The paste structures were also examined after three years float service to find out if there was any loss of structure. This loss of structure is associated with loss of capacity, but does not necessarily show up in electrical tests until the degradation is far advanced. This effect should not happen until after some years in service. Scanning electron micrograph techniques were used to give an early indication of such loss of structure. The analysis indicated no serious loss of structure after three years in service. Fig. 10 shows some typical results.

Mechanical Testing

The following mechanical tests were carried out on units (or cells) in three planes; namely, unit on its base, unit on its side (plates vertical), and unit on its side (plates horizontal). One unit only was used for each test in each plane.

(*a*) Vibration test to British Standard BS 2011 Part 2.1 Fc

Constant acceleration	$2g$ (19·62 m/s^2)
Constant displacement	3 mm peak
Sine swept	5–150 Hz
Crossover frequency	13 Hz
Duration	30 minutes

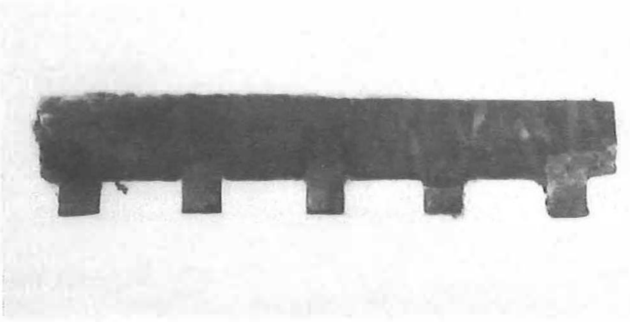
(*b*) Drop test to British Standard BS 2011 Part 2.1 Ed

2 × 100 mm drops onto solid floor

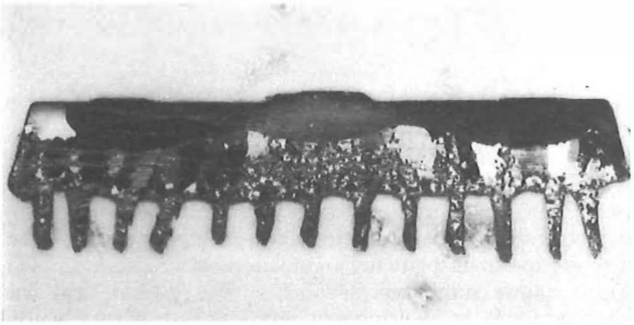
(*c*) Bump test to British Standard BS 2011 Part 2.1 Eb

1000 continuous bumps
25*g* severity
6 ms half sine pulse

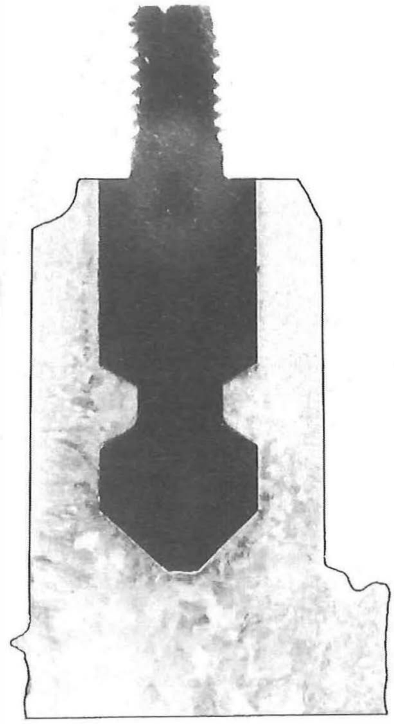
These tests proved to be very searching, and weaknesses in the mechanical design of the prototypes were clearly



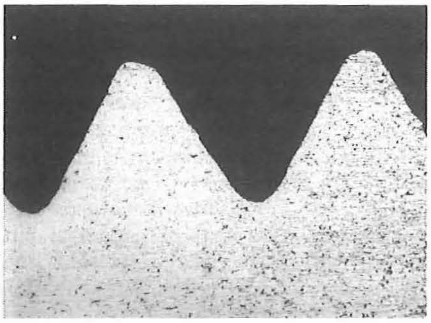
(*a*) Section of group bar—note poor adhesion of centre tab



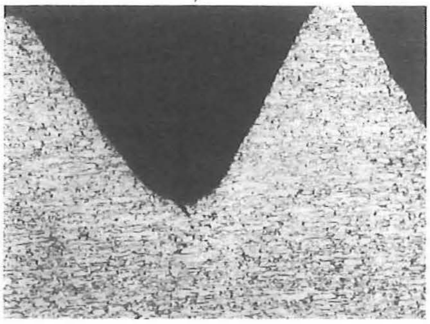
(*b*) Section of group bar—note good adhesion of tabs



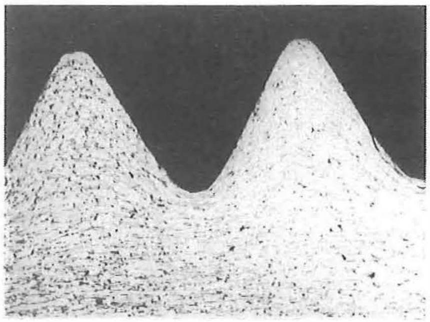
(*c*) Section through pillar—no porosity visible



(*d*) Section through pillar screw note horizontal strata, cut thread



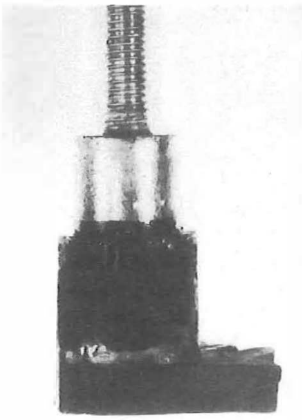
(*e*) Section of failing pillar screw note crack in the root of the thread



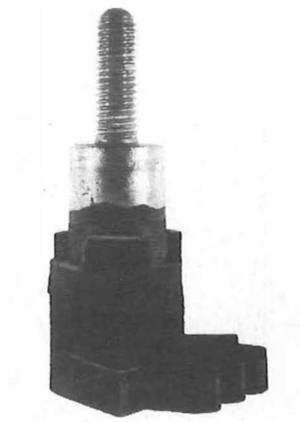
(*f*) Section through rolled thread note rearranged strata avoiding cracks

FIG. 7—Typical component test observations

(*a*) Test arrangements

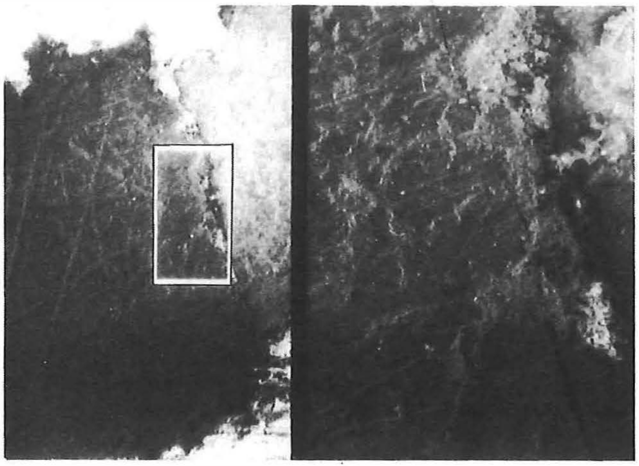


(*b*) Peroxidation on pillar after three months—note sharp edge of peroxidation layer

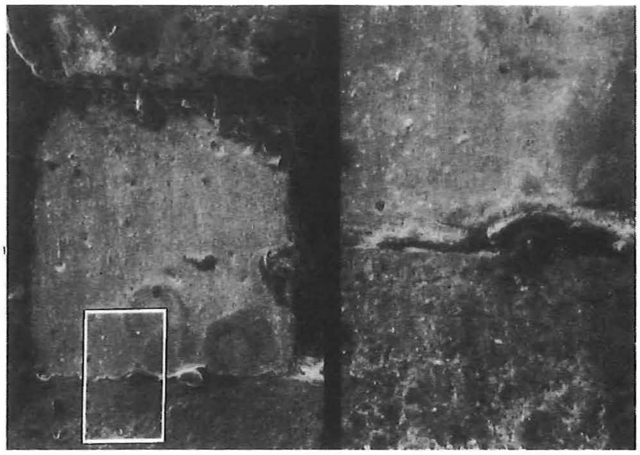


(*c*) Peroxidation on pillar after two months—note signs of peroxidation spreading into seal area

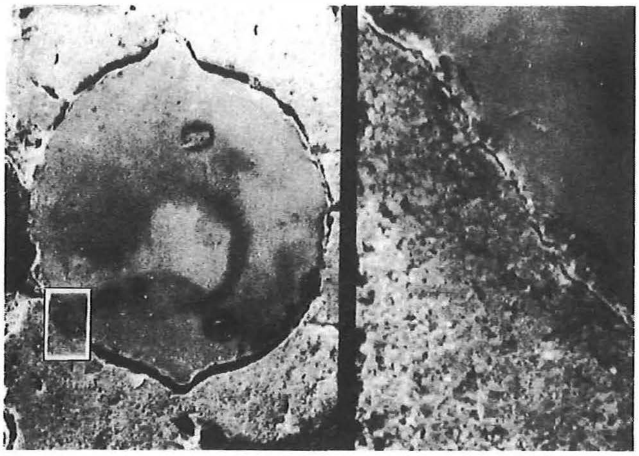
FIG. 8—Pillar seal tests



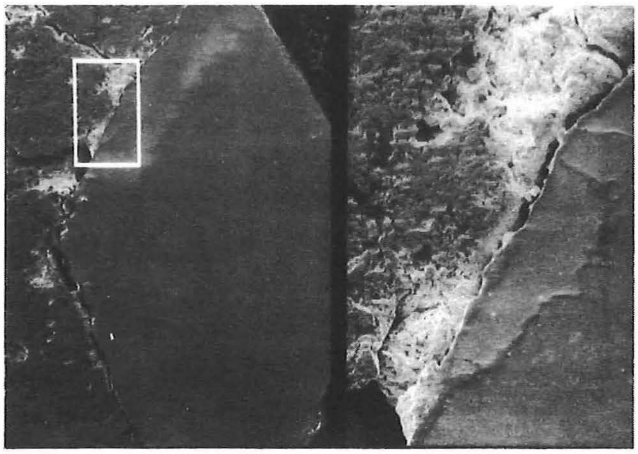
(*a*) Section of pure lead grid from new cell



(*b*) Section of pure lead from 4-year-old cell



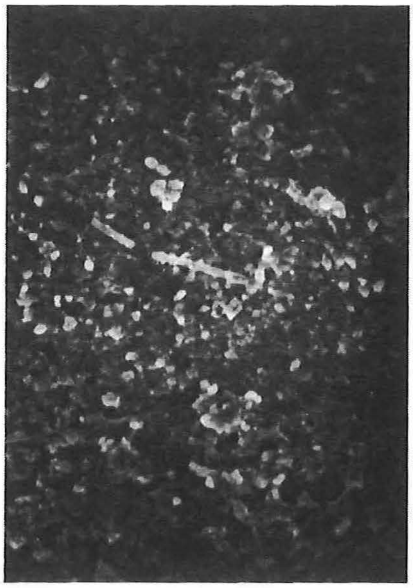
(*c*) Section of lead-calcium-tin grid from new cell



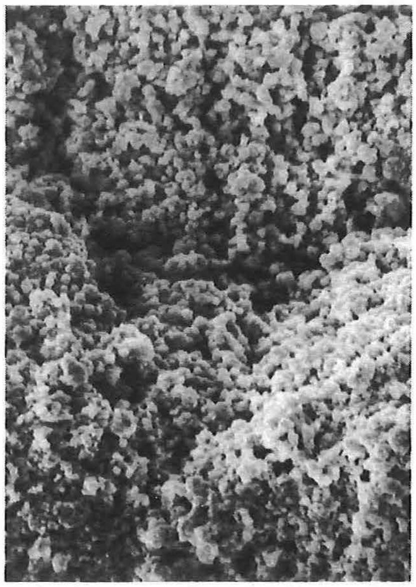
(*d*) Section of lead-calcium-tin grid from 3-year-old cell

FIG. 9—Examination of grid cross-sections

(*a*) Paste sample from pure lead grid after 4 years on float



(*b*) Paste sample from defective cell note lack of surface porosity



(*c*) Paste sample from 3-year-old lead-calcium-tin grid

FIG. 10—Scanning electron micrograph of positive paste structure

identified by failure of the affected parts. This confirmed the findings of component analysis of the parts made of lead, and of electrical tests at high rates of discharge.

CONCLUSIONS

Sufficient data has been obtained to give a high level of confidence that the final designs meet BT's requirements. There is also sufficient data to show that these new designs are sensitive to the characteristics of the charging rectifier. With a varying load demand, the trickle charge current is, possibly, too low to compensate for the capacity lost in supplementing the rectifier output. Consequently, a rectifier with a fast response to changes in load or input voltage must be used, or the battery will slowly lose capacity. The sealed battery/switch-mode rectifier power system adopted by BT is now being used in a variety of applications. With over 500 plants in service, only a few minor battery problems have been reported, most of which were associated with the container mouldings.

The data and results now available indicate that a ten year service life has a high probability of achievement. The likely modes of failure can be summarised as follows:

(*a*) *Electrolyte drying out* The gas tests give an estimate well in excess of ten years.

(*b*) *Grid corrosion* The results on pure lead and lead-calcium-tin alloys give no cause to doubt that the expected life will be achieved. There has been no evidence of any sudden loss of capacity after discharges, or overheating during recharging. There is also no evidence of passivating corrosion layers building up around the grid surface where lead-calcium-tin grids are used.

(*c*) *Pillar corrosion* The pillar seal assemblies have been tested under fairly severe conditions with no indications of premature failure.

(*d*) *Corrosion of internal lead connections* It has been BT's experience over many years, with flooded-electrolyte designs of cells, that corrosion of internal connections can be a serious problem. Normally, corrosion seriously degrades cell performance where there is a fault or weakness in the lead surface, or a junction between dissimilar antimony alloys. These faults are much more likely to occur where manual lead burning has been used during manufacture. In the sealed lead-acid designs, all series connections (that is, pillars, group bars, and through-wall welds) are machine cast, and can therefore be reliably quality assured. The tests indicate that the design of these components meets the requirements. Manual lead burning is confined to parallel connections (that is, the plate-group bar junctions), where the quality of burn is more readily controlled. Failure in service of these parallel connections is not nearly as serious, from a battery reliability point of view, as is failure of a series connection.

(*e*) *Loss of paste activity* The paste structures have been examined after up to three years service, and there is little evidence of serious degradation. It normally requires a large loss of structure before a significant loss of capacity occurs.

FUTURE DEVELOPMENTS

For the future, it is foreseen that there will be a continuing expansion in the use of these cells into the areas which traditionally used flooded-electrolyte designs of cell, but, in addition, the authors feel that the sealed battery will also create its own new markets. There is still considerable scope for development in lead-acid battery technology, particularly for further increases in energy and power density. The authors feel that the sealed-recombination technique is only one of many in the area of sealed stationary battery development.

ACKNOWLEDGEMENTS

The authors would like to thank their many colleagues in BT for their assistance in this project, and in particular those at BT's Birmingham Materials and Components Centre, who made an invaluable contribution.

Biographies

Jim O'Connor joined BT in 1968, and specialised in the maintenance aspects of power equipment for BT's switching and transmission systems. His experience covers rotating machine DC and AC converters, diesel generating sets, rectifier equipment and most recently lead-acid batteries. In 1979, he transferred to the power system design section, where he has responsibility for the design and development of all batteries used by BT.

Bob Doody has been with BT for about 17 years. He has spent most of that time in analytical chemistry, but has a broad material science background from expertise gathered in metallurgy and related disciplines. He is currently a member of the Materials and Components Centre based at the Birmingham laboratories. The laboratories provide a centre of expertise on all aspects of materials and components used in BT systems, equipment and apparatus. The centre liaises with equipment/component designers, purchasers, manufacturers, users and materials suppliers, providing support to all sections of BT.

New Concepts in the Design of Manholes

P. J. KING, B.ENG., DIP.GEOTECH., M.I.E.I.†

UDC 621.315.233

In keeping with advances in materials technology and the latest thinking on the design of reinforced-concrete structures, British Telecom has decided to adopt a new method of designing reinforced-concrete manholes. As a result, a new type of manhole is being introduced. However, because of the requirement to comply with the relevant British Standards, this manhole is more complex in design and construction than those that have been used in the past. This complexity, compounded with the desire to produce the most economical structure, would involve the engineer in long and tedious calculations by conventional methods. An obvious solution was to write a computer program which would effectively and quickly deal with the complexity and economy of the design, and thus free the engineer for more productive work.

INTRODUCTION

British Telecom (BT) uses manholes beneath carriageways for various purposes, such as to facilitate cable jointing, to house transmission equipment etc. A typical manhole is made of reinforced concrete; is 2 m high, 3 m long and 2 m wide; and has a hole in the roof that allows access to the chamber and 'windows' in the walls that allow cables to enter.

Manhole design has passed through a number of historical phases. It began with the old National Telephone Company's manholes, which were simply brick-walled pits roofed with slabs of York stone. Sometimes the York stone slabs were supported by rolled-steel joists. However, York stone, like unreinforced concrete, has very little strength in tension. When a stone is supporting a load, it is required to bend; this causes tensions to be set up in the stone and so cracks occur. Thus unreinforced concrete or York stone are not safe materials with which to build manholes, especially if the manhole is to support a modern carriageway.

Between 1900 and the mid-1930s, manhole sizes increased and roofs were changed to unreinforced concrete placed on boiler plates, which were in turn supported with rolled-steel joists. These types of manholes have their own associated problems; namely, as the concrete is unreinforced and the boiler plates act simply as a shutter for the concrete, all the load is carried by the joists. Rusting of the joists has meant that many of these manholes are now reaching the end of their useful lives and are having to be found and replaced.

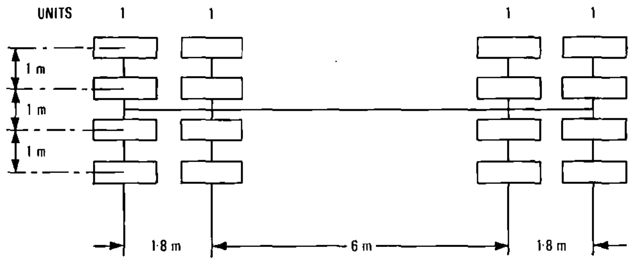
These designs of manholes were followed by a *straight-bar* design, which was built of reinforced concrete using only straight reinforcing bars. During the Second World War, in an attempt to save steel, the so-called *Harding* or *bent-bar* design was introduced. This design was for a manhole of reinforced concrete with a number of reinforced concrete roofing beams[1]. After the war, the straight-bar design was reintroduced, and is still in use. These reinforced-concrete manholes have performed, on the whole, adequately as engineering structures. Where there have been problems, they have been identified as material failures—for example, the use of high-alumina concrete—or construction faults such as poor-quality concrete or incorrect placing of reinforcement. Only one type of problem—the cracking radiating from duct openings—can perhaps be attributed to design inadequacy.

The design, as such, has stood the test of time; however, significant changes have taken place. Some local authorities and some consulting engineers who have had an interest in the siting of BT's manholes have insisted that they should be designed to the standard of British Standard Code of Practice for Reinforced Concrete Structures (CP 110)[2]; and this requirement rules out the straight-bar design. Resolving such differences introduces a delay into underground works that is advantageous to avoid. A manhole that would meet the CP 110 requirements wherever possible was therefore designed. The latest reinforced-concrete manhole, designed to the relevant British Standards, is the subject of the remainder of this article.

As many factors have to be considered in both the analysis and the economic design of a manhole, the process is both long and tedious; therefore, a computer program that would take the burden of this work away from the engineer was written. The manual design of a manhole to conform to British Standards could take a civil engineer a couple of weeks; however, by using a suitable computer program, this time can be reduced to a few minutes.

APPLIED LOADS

The road vehicle recommended for use as the basis for the load analysis of the manhole by British Standard BS 5400 (Steel, Concrete and Composite Bridges)[3] is known as the *HB vehicle* (see Fig. 1). All roads and bridges are designed



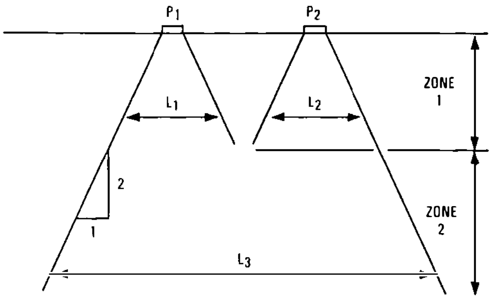
1 Unit = 10 kN
Note: The diagram shows a plan view of the wheels of the vehicle

FIG. 1—Highway bridges Type HB unit loading

† Local Lines Services, British Telecom Local Communications Services

to take this basic vehicle; however, depending on the type of road, the vehicle will have a certain weight on each axle. For example, motorways are designed to take 45 units of HB; this means that each axle of the vehicle takes 450 kN, each unit representing 10 kN per axle.

When the HB vehicle is placed on top of a manhole, the load on the axles is dispersed through the carriageway. The Department of Transport[4] recommends two ways of finding the resultant load on the roof and walls of the manhole: the wheel loads can be dispersed at a slope of 2 vertically to 1 horizontally (Fig. 2), or, when the depth of cover exceeds 1 m, the loads can be dispersed according to the Boussinesq



P_1 and P_2 represent wheel loads.
Zone 1: P_1 or P_2 uniformly distributed over length L_1 or L_2, respectively.
Zone 2: $P_1 + P_2$ uniformly distributed over length L_3

FIG. 2—Dispersal of wheel loading

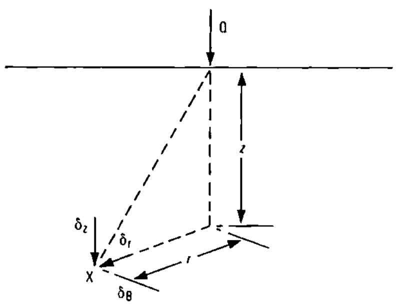
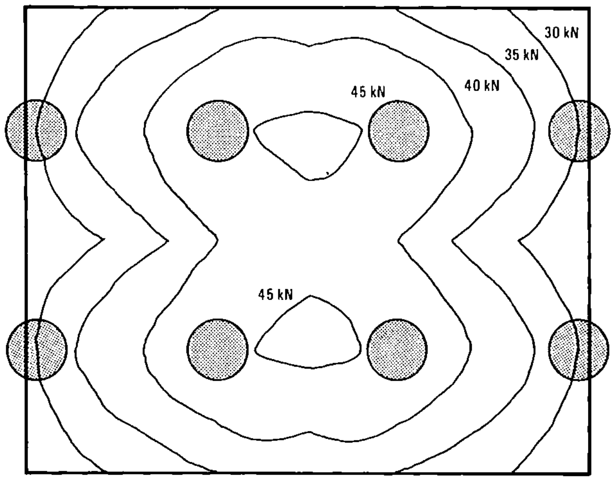


FIG. 3—Stresses due to a point load

formula[5]. This formula, when applied to soil mechanics, allows the calculation of the stresses, which occur at any depth and in any direction within the soil mass, resulting from a point load on the surface (see Fig. 3). If the roof of the manhole is divided into a number of lines at right angles and spaced at 0·25 m centres to form a grid, the Boussinesq formula can be used to calculate, at every intersection point of the grid, the stress due to the wheels of the HB vehicle (see Fig. 4). These loads can be averaged over the entire roof, and the result, combined with the dead weight of the soil above the roof, is then used as the working load of the roof.

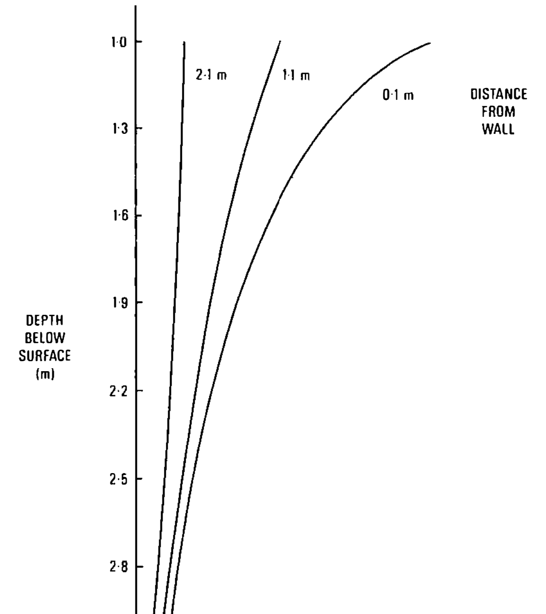
Similarly, a grid can be generated for the wall, and the same procedure followed to find the load on the wall due to the HB vehicle (Fig. 5). The stresses at the nodes can then be averaged, and the result, added to the stress due to the soil, gives the working load for the wall.



Note: Internal dimensions: 3·0 m × 2·0 m.
Depth beneath surface = 1·5 m.
Units of HB = 45.

The wheels are in a symmetrical position about the centre lines of the roof. Circular contact areas shown

FIG. 4—Load contours on manhole roof



Note: The graph shows the load on the wall (1 m below surface) due to the approach of the HB vehicle (45 units) at right angles to the plane of the wall

FIG. 5—Load on manhole wall

STRUCTURAL DESIGN

After the loads on the roof and on the wall have been calculated, the response of the structure can be examined. The Hillerborg method[6] was used as a means of mathematically modelling this response. The Hillerborg method has many advantages, not least among them is the ease with which it can be computerised.

The manhole is envisaged as acting as a monolithic structure composed of three elements: a portal frame in the short direction, a portal frame in the long direction, and a closed

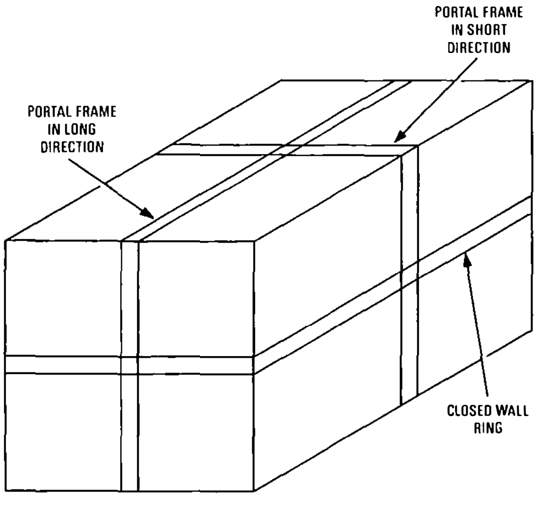


FIG. 6—Section of the main elements

ring around the walls (see Fig. 6). The load carried by each element can easily be derived by splitting the design loads into their directional components. The floor is designed as a simply-supported slab resting on the four walls.

As it is necessary to find the worst loading condition for each element of the structure, a number of different load cases must be taken into consideration. The first load case is when there is full load on both the walls and the roof; this is the normal traffic condition. The second load case is where there is full load on the walls, but only soil load on the roof; this occurs when the top of the manhole has been cordoned off to allow entry to the hole. The third case is when there is no traffic load on either the wall or on the roof. The computer program examines each of the elements, under these load conditions, to produce a bending-moment envelope; this envelope gives the greatest moment experienced by each element.

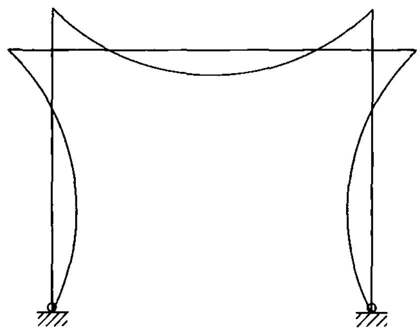
COMPUTER PROGRAM

Very little information is required to use the computer program to design a manhole. Soil conditions and other variable factors are not needed as the program assumes the worst possible case; that is, density of soil is 23·8 kN/m^3, and density of concrete 33·0 kN/m^3. These values, however, can be overridden and particular values, pertaining to a specific case, can be used when the appropriate information is available.

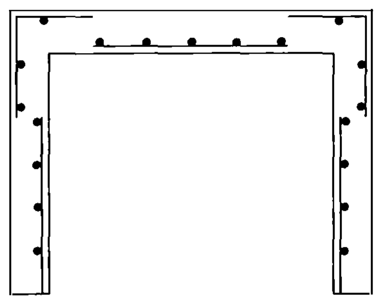
The program operates in an interactive manner and prompts the operator for the requisite information: the basic internal dimensions, the depth below the surface to the roof and the number of units of HB load that the road is designed to take. The program produces a basic manhole design in five minutes on a small business computer. Furthermore, it prints out a full bar bending schedule, complete with sizes and lengths ready for cutting, at the end of the design. An engineer using manual methods would take over a week to produce this same information.

The computer program designs the manhole on the assumption that the cube crushing strength of the concrete grade being used is 20 N/mm^2 and that the tensile strength of the steel is 425 N/mm^2. Again, however, there is a facility to change these parameters; for example, a ready-mixed concrete of 30 N/mm^2 may be used with mild steel (tensile strength 250 N/mm^2). The program first calculates the loads on the roof and on the walls, then resolves the loads into their directional components and calculates the bending-moment envelope. Based on these calculations, it gives the operator the choice of four different diameters for the reinforcing bars; the size chosen is used throughout the design for simplicity in placing.

Subroutines, based on CP 110, are used to determine the concrete thickness of each of the elements and the amount of steel required. The results of these calculations are printed out in the form of the number of bars to be used, the sizes of the bars and the spacing between each bar. Because the bending-moment envelopes are retained in the computer's memory, the most economical length of reinforcing bars to fit the envelopes can easily be calculated (see Fig. 7).



(*a*) Typical bending-moment diagram for a portal frame



(*b*) Idealised steel arrangement to fit the bending-moment diagram

FIG. 7—Relationship between reinforcement steel and bending-moment diagram

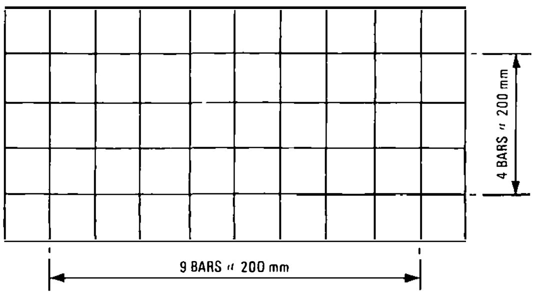
The bar schedule is printed at the end of the output, and contains the number, length and bending dimension of each bar to be used in the construction. This enables the drawing office staff, once they become familiar with the design, to quickly draw a manhole to British Standard requirements.

Where duct entry holes are needed in the structure, the standard methods of designing holes, based on the recommendations of the Cement and Concrete Association[7], are used.

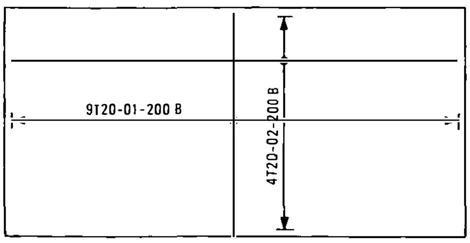
The result of each step in the design process is printed out by the computer. This allows the engineer to compare the design of the most highly stressed portions of the structure, as given by the computer, with a fairly basic manual design. This feature has been incorporated so that random bugs or errors in the program can be identified. The checking of all computer designs is a very necessary procedure, as the widespread use of computers, especially in the field of structural design, could lead to an unquestioning confidence in the results produced. This over-confident attitude can result in expensive and embarrassing failures, as emphasised by a number of recent cases in the civil engineering industry.

DRAWING

The introduction of this design procedure will necessitate the use of new drawing practices, as the design is more complex than that used at present. The main difference is that in the old style of drawing all the bars in a run were shown, whereas in the new method only one bar of a run is shown (see Fig. 8). The new style of drawing will bring BT into line with the methods used in the construction industry.



(*a*) Old-style reinforcement drawing



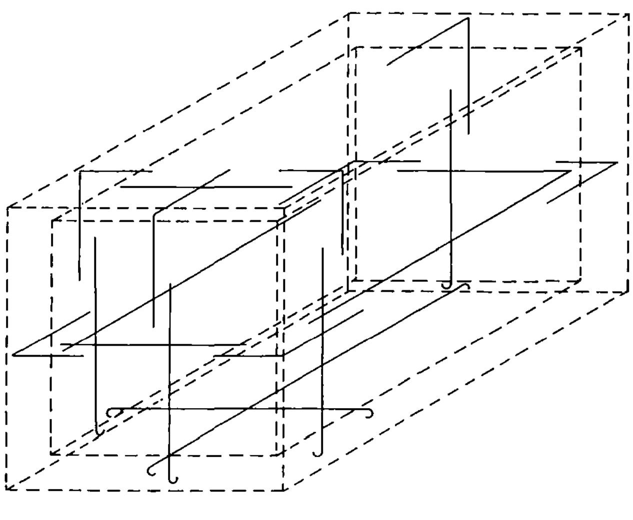
Note: '9' (or '4') refers to the number of bars, 'T' is the grade of steel used, '20' is the bar diameter, '02' is the bar mark, '200' is the spacing between bars, 'B', for bottom, refers to the bar position

(*b*) New-style reinforcement drawing

FIG. 8—Old- and new-style reinforcement drawings

CONCLUSION

This new type of manhole design (see Fig. 9) will enable BT to produce manhole designs and drawings which are in agreement with current specifications on reinforced-concrete design, and thus satisfy the requirements of the local authorities. The printed output, once it has been checked by an engineer, is in a form that can be readily used by the drawing office to produce a working drawing. This will, initially, take more time than the present method, but the time should be greatly reduced as drawing office staff become more familiar with the design.



Note: The diagram shows one each of the principal reinforcement bars. Ducts and holes have been ignored

FIG. 9—Three-dimensional view of a typical new manhole design

References

[1] HARDING, J. P. The Economic Design Of Manholes. Parts 1, 2 and 3. *Post Off. Electr. Eng. J.*, Apr. 1941, **34**, p. 1; July 1941, **34**, p. 85 and Oct. 1941, **34**, p. 133.

[2] British Standards Institution Code of Practice for Reinforced Concrete Design. CP 110: 1972.

[3] British Standards Institution Steel, Concrete and Composite Bridges. BS 5400: 1978.

[4] Department of Transport Technical Memorandum (Bridges) BE 1/77: Standard Highway Loadings, p. 39.

[5] CRAIG, R. F. Soil Mechanics. Van Nostrand Reinhold, 1976, p. 158.

[6] HILLERBORG, A. Strip Method of Design. Viewpoint Publications, 1975.

[7] The Concrete Society Standard Reinforced Concrete Details. 1973, p. 26.

Bibliography

British Standards Institution Bending Dimensions and Scheduling of Reinforcement for Concrete. BS 4466: 1981.

The Concrete Society Standard Method of Detailing Reinforced Concrete. 1983.

Biography

P. J. King is a temporary Executive Engineer in the External Plant Division of BT Local Communications Services. He graduated from University College, Cork, in 1979, and subsequently received a diploma in geotechnical engineering from Middlesex Polytechnic. He joined the then Post Office after leaving college as a consultant on the design of underground structures. For the past three years, he has been involved in day-to-day civil engineering consultancy, in the implementation of the new British Standard manhole, and in the structural and maintenance aspects of deep-level tunnels.

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.

Software Engineering
28–30 August 1985
Imperial College, London

Measurements for Telecommunication Transmission Systems
27–28 November 1985
Institution of Electrical Engineers

Computerised Quality Assurance
23–26 March 1986
University of Sussex, Brighton

Speech Input/Output; Techniques and Applications
24–26 March 1986
Institute of Education, London
Synopses by:15 October 1985

Software Engineering for Telecommunication Switching Systems
14–18 April 1986
Eindhoven, The Netherlands

COMMUNICATIONS 86
13–15 May 1986
Birmingham Metropole Hotel, Birmingham
Abstracts by: 23 September 1985

IEEE International Conference on Communications '86, 1450 Don Mills Road, Don Mills, Ontario, Canada M3B 2X7.

International Conference on Communications '86—'Integrating the World through communications'
22–25 June 1986
Toronto, Canada
Complete manuscript and abstract by: 1 November 1985

Institution of Electronic and Radio Engineers, 99 Gower Street, London WC1E 6AZ.
Telephone: 01–388 3071.

Networks and Electronic Systems
17–19 September 1985
Forum Hotel, London

Technology Management
2–4 October 1985
Grand Hotel, Eastbourne

Land Mobile Radio
10–13 December 1985
Churchill College, University of Cambridge

Statistics for Industry (UK) Ltd., 14 Kirkgate, Knaresborough, North Yorkshire HG5 8AD.
Telephone: 0423 865955.

Introduction to Reliability Analysis
23 October 1984
Bloomsbury Crest Hotel, London

Online Conferences Ltd., Pinner Green House, Ash Hill Drive, Pinner, Middlesex HA5 2AE.
Telephone: 01–868 4466.

Cellular Communications International
5–7 November 1985
Wembley Conference Centre, London

EUROCON 86 General Secretariat, 11 Rue Hamelin, F-75783 Paris Cedex 16, France.

Advanced Technologies and Processes in Communication and Power Systems (EUROCON 86)
21–23 April 1986
Paris, France

IMPORTANT NOTICE
TO ALL BRITISH TELECOM AND
POST OFFICE READERS

PLEASE RETURN THE GOLD-COLOURED FORM INCLUDED WITH THE APRIL 1985 ISSUE REGARDING DISTRIBUTION OF THE JOURNAL

AS SOON AS POSSIBLE

British Telecom Press Notices

British Telecom Commissions World's Most Powerful Exchange

British Telecom International (BTI) has successfully brought into service the world's most powerful telephone exchange, which uses the most up-to-date digital technology. The exchange is an enhancement of the international exchange at Keybridge House, Vauxhall, London. It can now handle nearly six times more calls, and has a capacity of 800 000 call attempts an hour. The exchange has been installed to cater for the continuing rapid growth in international telephone calls. In addition, the new exchange, code-named *TXD20*, is now providing for international digital operation, and creating a link between British Telecom's (BT's) growing inland digital network and similar networks overseas.

Commenting on this latest development, Mr. Anthony Booth, Managing Director of BTI, said: 'This powerful new exchange puts BT ahead of the rest of the world. It confirms that we are in the vanguard of the world drive towards integrated digital networks, which are essential for Britain's future economic prosperity. This enhancement of our international communications capability will enable us to aid international trade and invisible exports. It will act as a bridge between existing analogue networks and future digital services, carrying BT into the 1990s and beyond. Using the latest technology, it demonstrates BTI's expertise in providing international communications services, in which Britain leads the world. This exchange is the first phase of our programme of replacing our existing equipment with the latest digital technology. Eventually BTI expects to have three more digital international switching centres comparable to this in operation, and we plan to evaluate tenders later this year for the next exchange to come into service in 1988.'

At present, BTI operates six international exchanges using analogue electromechanical (crossbar) switching; one of these uses stored-program control. They are all used for international direct dialling and, together, they provide service between 30 million telphones in the UK and about 525 million throughout the world.

The new exchange was supplied by Thorn Ericsson Telecommunications Ltd. and is based on the L. M. Ericsson AXE1O design using the APZ212 processor. At present, it provides capacity for 13 800 circuits, and interworks with existing analogue networks as well as new digital systems. In its original version, it handled up to 144 000 call attempts an hour. The new APZ212 computer control system has now been installed, and occupies less than a quarter of the space taken by the original processor, which it replaces. This provides the increased call-handling capacity, which will be fully utilised when the expansion of the exchange to its maximum size of 50 000 circuits is completed next year. In addition, the switching matrix has been substantially enlarged and common-channel digital signalling provided on transmission links to other countries.

Most of the work of processor replacement and other changes during the last nine months were carried out without the continuing operation of the exchange being affected.

The exchange is used primarily to switch international direct dialling calls on the busiest routes, between the UK and the USA, Canada, South Africa, Australia, Japan, Hong Kong, and ten countries in western Europe. It is also being linked to BTI's existing exchanges to carry calls to and from other countries, transit calls (from one country to another via the UK), and calls set up by telephonists at BTI's international operator centres.

Claudivs Converse

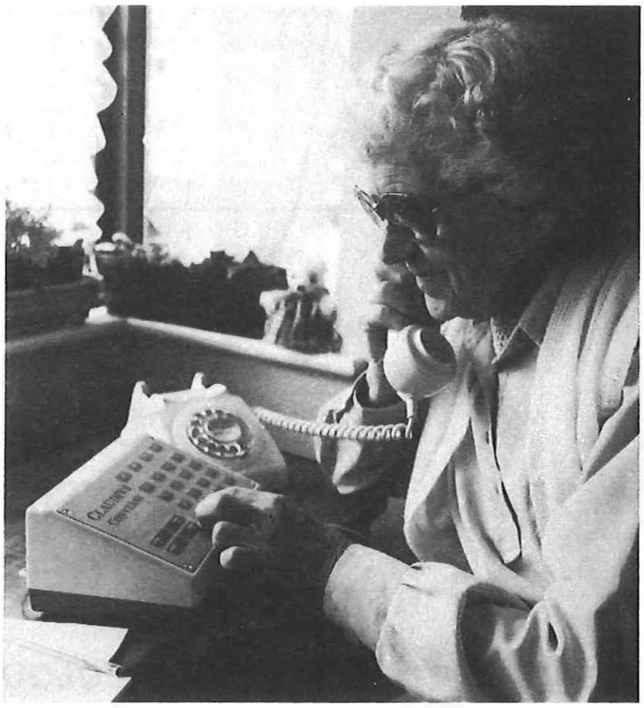
A new British Telecom (BT) device is helping people suffering from speech impairment to communicate. The device, known as *Claudivs Converse* (which stands for *calling line announcements using digitally integrated voice synthesis*), enables disabled people to select words and phrases transmitted by the equipment's built-in voice; and even mute people are able to hold simple conversations.

The unit, which takes its name from a Roman emperor who had a voice defect, can be connected either to a telephone line, or used on its own to broadcast words, sentences or messages through its loudspeaker. The device offers a combination of words giving access to a total of 64 possible phrases; and its four red emergency buttons enable the emergency services to be called quickly and provide automatically the caller's home address.

The unit does not have to be plugged into the telephone network, but can be used independently, as for instance in a hospital. Thus it can be used as the operator's voice, so that patients who cannot speak can still request assistance.

The customer chooses phrases he or she wants to be able to transmit, usually with the help of a speech therapist, or health visitor. These are then recorded for the customer and transferred digitally on to a speech synthesis microchip. The chip is then installed into the customer's unit, which is then ready for use. Male or female voices can be reproduced, in any language.

Claudivs Converse is one of the projects supported by BT Action in the Interests of the Disabled (BTAID), which was launched in 1983 to identify the special needs of all disabled people, and to expand the range of services for them. The device was invented by engineers at BT's Test Equipment Design Centre at Eastbourne, Sussex. A. P. Besson, a division of Crystalate Electronics, based at Hove, in Sussex, has been granted a non-exclusive licence by BT to manufacture the equipment. The licence permits manufacture of Claudivs Converse in the UK, and world-wide sales.



Beattie Brooks, who lives in Littlehampton, Sussex, and lost her voice two years ago after a throat operation, was one of the first people in the country to use Claudivs Converse

Improved Dial-Up for Prestel

Users of Prestel are to get an even more reliable dial-up to its computers this year over a new access network which will also be used by Telecom Gold, British Telecom's (BT's) electronic mail service. This new access network will provide users with high-speed access to Prestel, Telecom Gold, BT's national data network Packet SwitchStream (PSS) and other computers operated by the Value Added Systems and Services Division of BT Enterprises (BTE). The new access network will replace the existing Prestel communications links, and is being ordered from Telematics International by BT National Networks, which will install and operate it on behalf of BTE. In addition to providing a fast response—less than half a second—the access network will extend the current availability of Prestel at local call rates.

Commenting on the new development, Mr. Richard Hooper, BTE's Chief Executive for Value Added Systems and Services, said: 'We are providing our customers with a new stepping stone to value-added network services which will be faster, friendlier, more flexible and more reliable. They will be able to link up more quickly and easily with Prestel and Telecom Gold, and will have great freedom in the choice of terminals they can use. The Telematics system provides us with another route to the PSS with features of particular value to Prestel. Existing Prestel terminals will be able to use the new access network without modification and, if they incorporate automatic identification, they will be linked directly to Prestel. Users whose terminals do not have automatic identification will be presented with a menu of choices—Prestel, Prestel Editing and Telecom Gold, which will grow as BTE adds other value-added services it is now planning.'

The new network will provide PSS access nodes at 60 locations throughout the UK, and offer more than 96% per cent of the country's telephone users connection at local call rate. Dial-up calls to a node will connect the calling terminal through the local telephone network to a Lion System 'intelligent' modem, and then through a Timeplex statistical multiplexor to a data channel operating under high-level data link control (HDLC). This will route the call to a Telematics packet unit operating as an X25 interface to a packet-switching exchange on the PSS. The call will then be forwarded either to the host computer or to Prestel or Telecom Gold.

The access network, when complete next year, will include Telematics interfaces at seven locations. The access link will support two-way operation at bit rates of 1200/75, 300/300, 1200/1200 and 2400/2400, all in full duplex. Error correction will be available as an optional extra for terminals capable of implementing the necessary software. All standard asynchronous terminals will be supported on the network notably Prestel and ASCII teletypewriters and visual display units. Currently, the PSS supports the full X25 protocol, and development is under way to permit the direct connection of hosts operating under IBM BSC3270 and SNA protocols.

Teletex Takes Off

Teletex, Britain's launch pad for information technology, has been officially inaugurated by Ron Back, Managing Director of British Telecom (BT) National Networks and Mr. Geoffrey Pattie, Minister of State for Industry and Information Technology. Teletex, 'the electronic postman', is a new automatic high-speed message transmission service that can send a full page of text in a few seconds—thirty times faster than Telex. It enables users to type ordinary letters, messages and other documents on memory typewriters, word processors or other computers and to send them to similar equipment elsewhere, to produce high-quality replicas of the original. Setting up calls, transmission and reception will be fully automatic.

The start of the Teletex service was announced at a seminar in London organised by the Government-sponsored Teletex Awareness Group, which includes representatives of equipment manufacturers as well as BT and the Department of Trade and Industry. Mr Back said: 'Teletex will be a major benefit for British business, speeding the flow of correspondence, orders, invoices and other documents, and reducing reliance on paperwork. It will bring the electronic office closer still. Every business with a telephone line and a Teletex-compatible word processor, personal microcomputer, or memory electronic typewriter is now able to plug into this powerful new communications medium. It is fast, easy and cheap, and fully interconnected with the worldwide Telex network and its 1·5 million users. It will be cheaper and more convenient than the post for a great deal of business correspondence.'

BT has published the first Teletex directory to encourage customers to make maximum use of the new service. This lists all users of the system and their numbers and gives instructions for using the service. BT's primary role in the Teletex service is to provide the networks. Customers have a telephone line to which their terminal is connected. Calls to another Teletex terminal are made either over the public switched telephone network or over Packet SwitchStream (PSS), BT's public data network. Which network customers use will depend on volume: heavy users may find the PSS more economical.

A wide variety of equipment is already available capable of being used as Teletex terminals. This includes the MerlinTex adaptor available from BT Business Systems. It adds a Teletex capability to the Merlin M4000 and M3300 word processors, and can be used with some other companies' text machines.

British Telecom Launches Testing Service for Microchips

British Telecom (BT) has introduced the most advanced service in Britain today for commercially testing microchips. BT's Materials and Components Centre (MCC) is offering other organisations access to its state-of-the-art testing equipment—the Sentry/Schlumberger Series 80 linear/analogue tester for large-scale integrated circuits. This equipment was recently installed by the MCC for testing the wide variety of components used by BT.

BT's tester is the only Series 80 in the country available for commercial use by other companies. This facility offers UK manufacturers cost-effective access to a sophisticated testing package. It is suitable for testing a wide range of both linear and mixed linear/digital devices, such as combined codecs/filters, tone generators, diallers, modems, comparators operational amplifiers and digital-to-analogue converters.

The BT equipment is configured to provide 32 analogue input/output channels, with measurement accuracies down to 0·1 mV, 250 pA and less than 1 ns. It provides 32 digital input/output channels, with a 2 MHz data rate and 50 ns timing resolution, is supported by a comprehensive software library and can be used for production testing, go/no-go component checks and detailed evaluation of new devices.

Organisations requiring more details of the service should contact the MCC, 310 Bordesley Green, Birmingham B9 5NF (021–772 2361, Ext. 2174). BT has also appointed MTL Microtesting Ltd. as marketing agents for the service.

The Associate Section of the Institution of British Telecommunications Engineers and its National Committee

G. LYALL, B.A., ENG.TECH., A.M.I.ELEC.I.E.†

INTRODUCTION

The Associate Section of the Institution of British Telecommunications Engineers (IBTE), originally known as the *Junior Section*, was set up in 1931; in 1971, the National Executive Committee (NEC) was formed. The Associate Section attempts to further the interest and technical education of its members, who are drawn from engineering technical grades (below level 1) in British Telecom (BT). It achieves this aim by inviting notable people to lecture on telecommunications and allied subjects at meetings, arranged as often as possible, of members and their guests. These meetings are designed not only to be instructive and enjoyable, but also to provide members with the opportunity to get together socially.

The Associate Section was also set up so that members could exchange information and ideas among themselves, not only on work within BT but also on allied subjects. Home microcomputers greatly interest many Associate Section members, to the extent that local home-computer groups have been set up in some areas under the aegis of the IBTE. Once a year, most Associate Section Centres try to combine business with pleasure by taking a day trip to a place of interest, such as an electronics factory, in the morning, followed by a lighter visit in the afternoon.

ORGANISATION

The Associate Section has 25 000 members and is organised into Local Centres based at present on Telephone Areas. A second tier is based on Regions—for example, Scotland, North East etc—and the NEC, an elected body that meets quarterly, reports to an annual meeting with delegates from all the Regions. The NEC also serves as a communications point between the parent body—the Senior Section/Council—and the Associate Section.

In order to stimulate activities nationwide, the NEC is responsible to the membership for promoting and encouraging national competitions; presenting awards for achievement; encouraging the writing of technical and non-technical essays; and for promoting the advancement of telecommunications engineering and allied subjects among members.

To achieve these aims, the NEC is composed of the following officers:

(*a*) a Chairman and Vice-Chairman;
(*b*) a Secretary and Assistant Secretary;
(*c*) an Editor, who produces the Associate Section's quarterly *National News Sheet*;
(*d*) a quiz organiser, who organises a nationwide technical quiz culminating in a prestige final every April;
(*e*) a Projects and Visits Secretary, who organises a nationwide projects competition open to all members; and
(*f*) a Treasurer, responsible for finance.

These officials are elected at the annual general meeting of the Associate Section held every May and their names can be found in the National Associate Section Directory.

A President and Vice-President are appointed from senior management within BT Headquarters to assist and advise the NEC. The Associate Section can take great encouragement from the fact that, as a body, it is held in such regard by BT that such capable men as its President(s) are encouraged to give the Associate Section their time and resources.

COMPETITIONS

The competitions organised by the Associate Section are:

(*a*) *The Bray Trophy* This is a handsome award, named after the first President of the NEC, Dr. P. Bray, and is made to the winners of the final of the annual Technical Quiz, perhaps the best-known event.

(*b*) *The Cotswold Trophy* This trophy, donated by the Gloucester Centre, is presented on an annual basis to an individual member, Centre or Region that the NEC and President(s) think has done most to further the aims of the Associate Section.

(*c*) *The NEC Project Competition* This is again an annual trophy, and is held to encourage members to undertake a project, preferably on subjects with a telecommunications bearing; computer projects are included, although most computer software is judged under a new competition (see (*d*)).

(*d*) *The London Trophy* This is presented by BT London for the best microcomputer software and/or associated programs.

(*e*) *The Anning Award* This is named after Joe Anning, a much respected, past member and Projects Organiser, and was instituted to encourage Trainee Technicians (Apprentices) (TT(A)s) and Trainee Technicians (Improvers) (TT(I)s) to prepare and compete in a national competition to find the best in the country.

Five of the seven TT(A)s who entered the 1984 Anning Award. The winner was Mr. R. Ryan from London South East (pictured with clock)

† Edinburgh Telephone Area, British Telecom Local Communications Services

This last contest ends with a visit to London for the Regional winners and a final informal interview prior to the April final of the Technical Quiz, where the prestigious trophy is presented. With the changes in recruitment in BT, it may well be that the eligibility for entry to this competition could change to include all recruits to BT in the membership of the Associate Section. Thus, the Associate Section not only offers members a chance to meet on an informal basis to learn and discuss a variety of worthwhile topics, but a chance to compete nationwide in varied competitions.

ELIGIBILITY AND SUBSCRIPTION

Eligibility for membership of the Associate Section broadly means engineers below the Managerial and Professional Structure Band A (MPSA), although with the changes in job categories in BT now taking effect, already other grades, that is, staff from Traffic Division, are joining as Affiliated Members. The NEC will be looking at eligibility as BT develops.

Current subscriptions for membership of the Associate Section are between 4p and 15p monthly, depending on the number of activities the Local Centre is engaged in. Subscriptions are deducted from members' pay either weekly or monthly as the case may be.

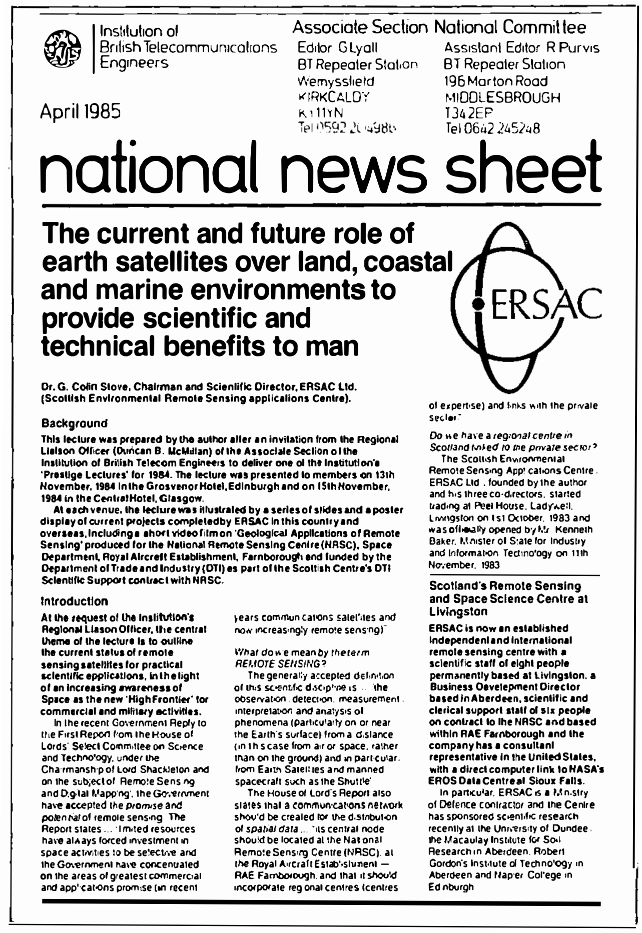
PUBLICATIONS

The quarterly *National News Sheet* is distributed to members and through it they are informed of technical and non-technical features from all the Regions.

Over 80% of the members of the Associate Section also subscribe to *British Telecommunications Engineering*, the journal of the IBTE, and form a large part of the readership. The editors of *British Telecommunications Engineering* strive to include articles that are likely to be of interest to members of the Associate Section, and of particular interest to students is the model answer *Supplement*. (It should be noted that subscription to *British Telecommunications Engineering* is *not* included as part of the subscription to the Associate Section.)

CONTACT

If you would like to know more about the Associate Section, please contact the local Secretary in your Area or District. His or her name will be in your local official directory.



Institution of British Telecommunications Engineers

Associate Section National Committee

Editor G Lyall, BT Repeater Station, Wemyssfield, KIRKCALDY, KY1 1YN, Tel 0592 204986

Assistant Editor R Purvis, BT Repeater Station, 196 Marton Road, MIDDLESBROUGH, TS4 2EP, Tel 0642 245248

April 1985

national news sheet

The current and future role of earth satellites over land, coastal and marine environments to provide scientific and technical benefits to man

ERSAC

Dr. G. Colin Stove, Chairman and Scientific Director, ERSAC Ltd. (Scottish Environmental Remote Sensing applications Centre).

Background

This lecture was prepared by the author after an invitation from the Regional Liaison Officer (Duncan B. McMillan) of the Associate Section of the Institution of British Telecom Engineers to deliver one of the Institution's 'Prestige Lectures' for 1984. The lecture was presented to members on 13th November, 1984 in the Grosvenor Hotel, Edinburgh and on 15th November, 1984 in the Central Hotel, Glasgow.

At each venue, the lecture was illustrated by a series of slides and a poster display of current projects completed by ERSAC in this country and overseas, including a short video film on 'Geological Applications of Remote Sensing' produced for the National Remote Sensing Centre (NRSC), Space Department, Royal Aircraft Establishment, Farnborough and funded by the Department of Trade and Industry (DTI) as part of the Scottish Centre's DTI Scientific Support contract with NRSC.

Introduction

At the request of the Institution's Regional Liaison Officer, the central theme of the lecture is to outline the current status of remote sensing satellites for practical scientific applications, in the light of an increasing awareness of Space as the new 'High Frontier' for commercial and military activities.

In the recent Government Reply to the First Report from the House of Lords' Select Committee on Science and Technology, under the Chairmanship of Lord Shackleton and on the subject of 'Remote Sensing and Digital Mapping', the Government have accepted the *promise* and *potential* of remote sensing. The Report states ... 'limited resources have always forced investment in space activities to be selective and the Government have concentrated on the areas of greatest commercial and applications promise (in recent years communications satellites and now increasingly remote sensing)'.

What do we mean by the term REMOTE SENSING?

The generally accepted definition of this scientific discipline is ... 'the observation, detection, measurement, interpretation and analysis of phenomena (particularly on or near the Earth's surface) from a distance (in this case from air or space, rather than on the ground) and in particular, from Earth Satellites and manned spacecraft such as the Shuttle'.

The House of Lord's Report also states that a communications network should be created for the distribution of *spatial data* ... 'its central node should be located at the National Remote Sensing Centre (NRSC), at the Royal Aircraft Establishment — RAE Farnborough, and that it should incorporate regional centres (centres of expertise) and links with the private sector'.

Do we have a regional centre in Scotland linked to the private sector?

The Scottish Environmental Remote Sensing Applications Centre, ERSAC Ltd, founded by the author and his three co-directors, started trading at Peel House, Ladywell, Livingston on 1st October, 1983 and was officially opened by Mr Kenneth Baker, Minister of State for Industry and Information Technology on 11th November, 1983.

Scotland's Remote Sensing and Space Science Centre at Livingston

ERSAC is now an established independent and international remote sensing centre with a scientific staff of eight people permanently based at Livingston, a Business Development Director based in Aberdeen, scientific and clerical support staff of six people on contract to the NRSC and based within RAE Farnborough and the company has a consultant representative in the United States, with a direct computer link to NASA's EROS Data Centre at Sioux Falls.

In particular, ERSAC is a Ministry of Defence contractor and the Centre has sponsored scientific research recently at the University of Dundee, the Macaulay Institute for Soil Research in Aberdeen, Robert Gordon's Institute of Technology in Aberdeen and Napier College in Edinburgh.

The Associate Section's *National News Sheet*

Biography

Graeme Lyall joined the Post Office in 1971 as a TT(A); after two years as a Technical Officer-in-Training with the Circuit Provision Control in the Dundee Telephone Area, he became a Technical Officer on transmission construction, based in Kirkcaldy, and is now working on transmission planning in Edinburgh. He serves as Editor of the *National News Sheet*. He gained a degree from the Open University two years ago and is a member of the Territorial Army.

Notes and Comments

MAILING THE JOURNAL TO HOME ADDRESSES

All Full Members of IBTE, and Associate Section Members and other employees of British Telecom and the Post Office who subscribe to the *Journal*, are reminded that they should complete the gold-coloured form concerning the mailing of Journals to members' and readers' home addresses that was inserted in the April 1985 issue, and return it without delay. This will ensure that from the October 1985 issue your copy of the *Journal* will be sent to your home. Further copies of the form can be obtained from the editorial office. It is important that your pay group reference is correctly quoted on the form; the pay group reference is given on your pay slip.

CONTRIBUTIONS TO THE JOURNAL

Contributions of articles to *British Telecommunications Engineering* are always welcome. Anyone who feels that he or she could contribute an article (either 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.

EDITORIAL OFFICE—CHANGE OF ADDRESS

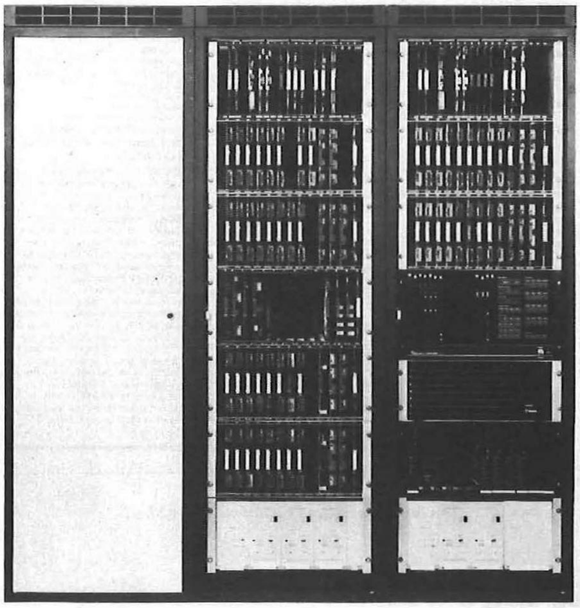
The editorial office of the *Journal* has moved to the following address: *British Telecommunications Engineering*, NN/CMkt2.2, Room 107, Intel House, 24 Southwark Bridge Road, London SE1 9HJ. (Telephone: 01–928 8686 Extn. 2233.) All correspondence relating to editorial matters ('letters to the editor', submissions of articles and educational papers, requests for authors' notes etc.) should be sent to the Managing Editor or Deputy Managing Editor, as appropriate, at this address.

Product News

Monarch IT440

British Telecom (BT) Business Systems's new Merlin Monarch IT440 introduces a new era of voice-data-text communications. Based on the successful Monarch telephone system, of which over 10 000 units have been installed, and the experience gained with the UXD5, BT's small public exchange system, the IT440 builds on this to provide a system that brings business communications and computing closer to the reality of convergence.

The new system, developed by BT engineers, provides voice switching for up to 440 extensions and flexible data and text switching for up to 240 users. The design is modular so that the system can grow to meet specific needs; this modularity also means that voice and data switching are kept separate to avoid data traffic affecting voice switching, and affords greater security to both voice and data traffic.



Merlin Monarch IT440

The Monarch IT440 uses the proven hardware technology from the Monarch 250 allied to the dual-processor control system from the UXD5 to provide an advanced voice switch. This, together with stand-by batteries and zoned power supplies, ensures security of operation for the voice switch. The stand-by batteries and power supply are also based on the experience gained with the UXD5 and with Sytem X, and are installed in a cabinet *en suite* with the switching equipment. Regulatory requirements state that telephone systems with direct dialling-in must have a stand-by power supply, and this requirement is met by the power cabinet on the IT440. The equipment is housed in three cabinets, and an additional free-standing connection frame provides all extension and line cabling facilities.

The data and text switching facilities use the combined features of Merlin DatelNet 500 data-over-voice transmission equipment, which allows full duplex speech and data transmsission over ordinary two-wire PABX extension cabling, and Merlin Datelmux 5500, which provides the data switching capability. Terminal-to-terminal data or text switching uses named calling between terminals, and integral modem access provides ease of conversion for communication over analogue circuits.

The new system can have up to four operator consoles and offers improved call-logging information, which, when combined with the new Merlin CM8000 Call Management System, enables full details of public and private voice network calls and internal calls to be recorded

The Monarch IT440 represents a significant advance in business switching systems whilst using proven technology. It therefore provides a low-risk route to companies wishing to progress to integrated systems.

Existing Monarch 250 users will also be able to benefit from the development of the IT440. A sister version, the Monarch SE440, which will be available later in 1985, will enable Monarch 250C1 and C2 users to enhance their systems to the same size and software capability as the IT440, but as a single processor version and without integral stand-by power and data switching.

Merlin TX72 Featurephone

The Merlin TX72 Featurephone is a new terminal from British Telecom Business Systems to complement the Monarch telephone system. The TX72, a low-profile terminal of compact, modern design, gives users single-key access to the many Monarch facilities. It has a 21-digit liquid-crystal display to provide information to the user, and includes some additional features of its own including on-hook dialling, a microphone mute control, link extension and repeat last number.

The Merlin TX72 has 21 dedicated feature keys, each of which gives immediate access to a specific facility on the Monarch system at the touch of a button. This simplifies operation by the user and encourages maximum usage of the wide range of facilities offered by Monarch. The dedicated feature keys are organised into four groups by common feature: diversion and callback; call hold and retrieve; wait on busy and call-handling facilities (conference, shuttle and recall). The display shows which feature is in use at any time and provides prompts to assist operation.

The TX72 has its own number store for short-code dialling. Up to 40 telephone numbers, each of up to 21 digits, can be stored, entered and changed by the user; again the display shows the stored number being called. This store is in addition to the centrally stored short-code feature of the Monarch.

Merlin TX72 Featurephone

The link extension feature provided by the TX72 is useful where two people are working closely together. It enables a second extension identity to be programmed in by the user so that direct calls can be made by single-button operation. Additionally, calls can be diverted to the linked number by operation of a single key.

Merlin Pentara 100

British Telecom has built on the experience gained of over 40 000 Herald telephone systems in operation by designing its successor—the Merlin Pentara 100. This new system provides all the features of its predecessor and more in a completely redesigned package.

The immediately obvious difference is the range of new low-profile terminals, which incorporate programmable function buttons, integral monitor speaker and variable ringing pitch. Facilitites incorporated in the range also include full two-way loudspeaking and a liquid-crystal display to provide operational information to the user. The less obvious differences lie in the software of the system, which provides many of the features.

Apart from such regular features as abbreviated dialling, call diversion, conference and hands-free calling, the Pentara allows users to carry out their own changes in facilities. And, to ensure that changes are correctly implemented, the system includes speech synthesis to provide spoken messages to the user. This feature also allows users to check the facilities available on a particular telephone at the time of use.

Other facilities available with the Pentara 100 system that are designed to aid operation in business situations are call queuing, call hold and ring back when free. This last feature avoids the frustrations of having to try repeatedly to call a busy extension by doing this job automatically and ringing the caller when the extension is free.

The Merlin Pentara 100 has capacity for up to 16 exchange lines and 76 extensions, although there is some flexibility in these maxima depending on individual requirements. The system can be configured either as a key system, where any terminal can answer incoming calls or, by providing an operator's terminal, as an operator-controlled system.

Merlin Pentara 100

New Merlin Modem—V32 Modem for Two-Wire Operation at 9600 bit/s

British Telecom Business Systems has introduced a new modem, the Merlin Datel Modem 4962X, to join its popular Fourth Generation X-series. This product is the first designed to meet CCITT Recommendation V32. The heart of the new modem is a new custom large-scale integration chip-set developed at the British Telecom Research Laboratories. The modem, a significant breakthrough in the design of modems, provides duplex operation at 9600 kbit/s over two-wire circuits by using echo-cancelling techniques. The modem can also be used on connections through the public switched telephone network (PSTN).

The Datel Modem 4962X offers a wide range of features to the data-communications user including 9600/4800 bit/s synchronous or asynchronous operation, automatic speed and bit-rate selection to match the remote end, local or remote soft-strap configuration, and powerful automatic adaptive equaliser and echo canceller. The modem also has automatic calling and answering as standard features for operation on the PSTN, and provides comprehensive and powerful diagnostics.

The Merlin Fourth Generation is a family of British designed and manufactured modems meeting CCITT standards and covering data rates from 300–9600 bit/s. Fourth Generation X-series modems feature commonality of design and construction. The modularity enables members of the family to share common cases and common racking; this means that users can have a mixture of modem types together in the same shelf unit. A requirement for upgrading the speed of operation can thus be met by simply changing a card. The designs incorporate the latest technology, including LSI and microprocessor control techniques, to give high performance and powerful features with small physical size and power dissipation. All members of the family incorporate comprehensive diagnostic capabilities and are suitable for operation on private circuits and the PSTN.

Merlin CM8000 Series Call Management System

British Telecom Business Systems' new advanced call management and information system, the CM8000 series, offers a wide and flexible range of call analysis facilities for identifying, controlling and allocating business telephone costs.

The CM8000 is the management accountant's dream as it provides a detailed information analysis as a printed report. The system records all calls, and telephone usage can be broken down by department and even by extension, a boom for allocating costs, establishing budgets and identifying underutilised extensions. The nature of calls, whether they are local, long distance or via the operator, at cheap, standard or peak rate, their average duration and total cost can be identified. A peak/standard traffic comparison shows users how much they would have saved had all peak rate calls been made during standard rate time.

The system maintains an up-to-the minute on-line company telephone directory with details of surname, department and extension, as well as details of telephone usage. A hard copy can be printed out from this for use as an internal telephone directory. An account coding report allows the user of the CM8000 system to identify and allocate the costs of all calls made in connection with a specific contract or on behalf of a client, subject of course to the availability of data from the telephone system.

The great advantage of the CM8000 series is that it gives choice and flexibility to the user. Fourteen separate analysis programs are available to enable users to run their telephone systems in the most effective and efficient way. Users can tailor the software to suit their own reporting requirements and the modular character of the equipment enables additions to be made. The system gives a choice of three call-recording capacities, ranging from 15 000 to 850 000 calls, and so a version in the CM8000 series is available to suit every business.

The CM8000 series is based on the successful 16 bit Merlin M4000 series microcomputer; it can therefore take advantage of that system's ability to run the Concurrent CP/M operating system. Word processing using WordStar can be run concurrently with the storage of call records on the CM8202 and concurrently with storage and processing on the CM8211 and CM8221 versions. The system can therefore be used for writing reports while calls are being recorded and processed.

British Telecommunications Engineering

If you wish to subscribe to *British Telecommunications Engineering*, please complete the relevant section of the order form below and send it to the address given. British Telecom (BT) and British Post Office (BPO) staff should complete the upper section and other subscribers the lower section. A photocopy of this form is acceptable.

SUBSCRIPTION ORDER (BT and BPO STAFF)

To: *British Telecommunications Engineering Journal,*
Room 107, Intel House, 24 Southwark
Bridge Road, London SE1 9HJ.

Please arrange for me to receive *British Telecommunications Engineering*. I have completed the form opposite authorising deductions from pay to cover the special annual subscription of £2·04.

Date .... Signature ....

Name and initials ....
(in block capitals)

IBTE Associate Section membership (Note 3) YES/NO

Home address (Note 1) ....

....

....

....

Pay group reference (Note 2) ....

British Telecommunications Engineering: Authority for Deduction from Pay

I, ...

(name in full, surname first in block capitals), authorise the deduction from my salary or wages, until further notice, of the sum of 17p per month/4p per week for payment to the Managing Editor, *British Telecommunications Engineering*, on my behalf.

Date .... Signature ....

Official address ....

....

....

Rank .... Pay No. (Note 2) ....

District/Area/Dept. ....

Pay group reference (Note 2) ....

Notes: 1 The *Journal* will be sent to your home address.
2 Your pay number and pay group reference are printed on your salary slip.
3 Indicate whether you are a member of the Associate Section of the Institution of British Telecommunication Engineers (IBTE). Full Members of the IBTE receive the *Journal* as part of their IBTE subscription.

YEARLY SUBSCRIPTION ORDER (NON-BT/BPO STAFF)

To: *British Telecommunication Engineering Journal,*
Room 107, Intel House, 24 Southwark
Bridge Road, London SE1 9HJ.

Please supply 4 quarterly issues of *British Telecommunications Engineering*. I enclose a cheque/postal order for the sum of £6·00 (Canada and the USA: $10) to cover the yearly subscription.

(Cheques and postal orders, payable to **'BTE Journal'**, should be crossed '& Co.' and enclosed with the order. Cash should not be sent through the post.)

Name ....

Address ....

....

....

Please state with which issue you wish your subscription to commence (April, July, October or January) ....

INDEX TO ADVERTISERS

Advertisements

Communications should be addressed to Mr A. J. Pritchard, The Advertisement Manager, *BTE Journal*, Room 104, 90–91 Wood Street, London, EC2V 7JS.

No responsibility is accepted by the Journal for any of the private or trade advertisements included in this publication.

Institution of British Telecommunications Engineers

CENTRAL LIBRARY

The IBTE Library has over 2500 books, collected since the earliest days of the Institution. Many of the books are on engineering subjects related to telephony and telegraphy, whilst others cover a wide field of general technical interest.

An abbreviated catalogue is bound in with the *Supplement* included with this issue of the *Journal*, and will be convenient for members wishing to know which recent books are available. A full catalogue including all the books of historical interest is available from the Librarian, IBTE, 2–12 Gresham Street, London EC2V 7AG.

Members can borrow books from the Library by completing a Library requistion form, available from the Librarian or Local-Centre Secretaries. Alternatively, the panel below, or a photocopy can be used. The form should be returned to the Librarian at the address above; a self-addressed label must be enclosed.

In recent years, the IBTE Library has been underused and its future is now dependent upon increased borrowings.

IBTE LIBRARY REQUISITION FORM

For use only by members of the Institution of British Telecommunications Engineers

Author ....

Title ....

....

Catalogue No. .... Date ....

Name ....

Official Address ....

....

....